Editorial Preface

From the Desk of Managing Editor...

It may be difficult to imagine that almost half a century ago we used computers far less sophisticated than current home desktop computers to put a man on the moon. In that 50 year span, the field of computer science has exploded.

Computer science has opened new avenues for thought and experimentation. What began as a way to simplify the calculation process has given birth to technology once only imagined by the human mind. The ability to communicate and share ideas even though collaborators are half a world away and exploration of not just the stars above but the internal workings of the human genome are some of the ways that this field has moved at an exponential pace.

At the International Journal of Advanced Computer Science and Applications it is our mission to provide an outlet for quality research. We want to promote universal access and opportunities for the international scientific community to share and disseminate scientific and technical information.

We believe in spreading knowledge of computer science and its applications to all classes of audiences. That is why we deliver up-to-date, authoritative coverage and offer open access of all our articles. Our archives have served as a place to provoke philosophical, theoretical, and empirical ideas from some of the finest minds in the field.

We utilize the talents and experience of editor and reviewers working at Universities and Institutions from around the world. We would like to express our gratitude to all authors, whose research results have been published in our journal, as well as our referees for their in-depth evaluations. Our high standards are maintained through a double blind review process.

We hope that this edition of IJACSA inspires and entices you to submit your own contributions in upcoming issues. Thank you for sharing wisdom.

Thank you for Sharing Wisdom!

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Secure Virtual Local Area Network Design and Implementation for Electronic Data Interchange

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Abstract—Electronic Data Interchange is a popular platform for sensitive business transactional data transmission over the local to public network. It requires Value Added Network to communicate local data from one endpoint to another. In this paper, we proposed a value-added network design and deployment using the Virtual Local Area Network. The Value Added Network over a virtual Local Area Network eases the burden of managing the network and its functional devices. This proposed network provides the solution of data traffic as per demand required network path for the Electronic Data Interchange applications. This advanced deployment model of Value Added Network over a virtual Local Area Network also offers more robust security to the network entities and traffic. This proposed Virtual Local area network has been deployed with the Cisco environment, and all the specifications have been successfully implemented and tested for optimal security for Electronic Data Interchange. Virtual Local Area Network deployed with four different methods such as transferring packets with backbone, virtual Local Area Network using the tagging method, implementing virtual Local Area Network using Time Division Multiplexing, and by user-defined frame field. All these deployments are successfully done, and the secure platform for Electronic Data Interchange data exchange from one end local network to the other local network has been optimized.

Keywords—Electronic data interchange; value added network; virtual local area network; data link layer; time division multiplexing

I. INTRODUCTION

Electronic Data Interchange is the common medium of business-sensitive data transmission from one entity to another over a public network such as the Internet. Third-party Value-Added Network (VAN) provides the solution to the local network for EDI application to transmit the sensitive EDI application data from a local network of a business firm to another end of other industries. The virtual LAN is considered the more secure and automated network for EDI applications and data communication between the local network of one business entity to another.

The virtual LAN requires switches to constantly contact the VMPS server, requesting configuration information when an EDI host or EDI network client connects to a switch participating in the virtual LAN network [1][2]. A VMPS database can also be configured with more information to handle more hosts’ requests for participation under the virtual LAN. Virtual LAN flexibility can be achieved by connecting more than one EDI host on one dynamically configured port as long as all EDI hosts are part of the same virtual LAN. The defined virtual LAN includes several physical segments per logical subnet. Hence this reduces the burden of management of physical segments and maintenance of network tables.

Virtual LANs are defined with two layers for EDI applications. The layers are virtual LAN Layer 2 and Layer 3 as per the basic working at the respective Open System Interconnection (OSI) layers [3]. Layer 2 virtual LAN is defined based on information at the Data Link Layer (DLL). Each port of the switch constitutes a switch port to form a single virtual LAN segment.

Virtual Local Area Network (Virtual LAN) is a mechanism that allows the same physical network topology to use more than one virtual network simultaneously. The networks of each category are separated for the security enhancement of EDI applications [4]. A special kind of routing is required when one device or workstation wants to reach another device or workstation from one virtual LAN segment to another virtual LAN segment for EDI. The switch determines that the client machine belongs to a virtual LAN. Suppose an EDI client requests a connection from port one on a switch assigned VAN 2; the EDI client cannot change anything behind this as the requested EDI client is indifferent or the same virtual LAN. The assignment of the switch port to virtual LANs is performed in two different ways. One mode of assignment of ports to virtual LAN is followed statistically, which means that the administrator must manually configure each port of the switch to assign the virtual LANs [5].

Some switch ports connect multiple end stations, while others have only one. All the workstations connected to ports associated with virtual LAN share a common broadcast domain, and all workstations of another virtual LAN segment are part of a different broadcast domain. The traffic between the segments of virtual LANs must have to pass through the router.

When multiple EDI end workstations are, by default, part of the same virtual LAN, moving a user to a different switch is considered a problem, with the administrator assigning the end user workstation to a different switch port. This problem is being avoided through the administrator end by establishing a one-station-to-one switch port architecture. This elaborates a good idea behind the usability of virtual LAN technology as per the requirements under the networking and infrastructure
EDI and its requirements are constantly increasing in the field of business. Almost all business organizations and government entities prefer EDI as a secured and automated information technology platform to process and communicate vital data among business partners and others. The authors performed a meta-analysis of various research in EDI and VAN and developed a theoretical framework for contextual outcomes associated with adoption to usability [12]. Per management perspectives and standpoints, VAN should have defined guidelines to automate data exchange without considering any network processes.

A real-time and decentralized communication network has interconnecting data nodes that can host the artifacts and services. This type of real-time decentralized network adds value to the artifacts and creates the VAN. The event notification-based VAN with the linked data specification provides advanced services, such as automated communication between the given nodes, which is best suitable for the EDI [13]. The experiments performed by the authors provide an advanced framework that can create a network with nodes that interact with each other. Based on the application scenarios, node registration, re-registration, collaborative certification, and awareness are achieved for value-added services such as EDI and other services.

III. ADVANCED DESIGN SCHEME OF VIRTUAL LAN

The advanced design scheme of virtual LAN includes all the aspects of the network, messages, algorithms, and protocols.

A. Proposed Model of Message Format and Encapsulation

Fig. 1 shows the virtual LAN message and protocol layering. Conceptually program used IP to transmit and receive. Virtual LAN message, the header, and data are encapsulated in a frame [14]. Finally, the network interface layer embeds the datagram into a frame sending it from one machine to another. The format of the frame depends on the underlying network topology.

Fig. 1. Virtual LAN message format.
On input, when a packet arrives at the lowest layer of network software and is processed from lower to higher layers. Each layer removes one header before the massage so that all headers are removed by the time the highest layer passes data to the receiving process.

When the net server receives a request for an IP address, it checks for the IP address and sends it to the virtual LAN agent for virtual LAN idea mapping; the virtual LAN agent then checks its node table; if the IP and virtual LAN ID mapping is not available, it sets validity-bit equal to pending and sends back a wait message to the server, the server sends wait message to the node which will be displayed on screen [15].

The virtual LAN manager then replies to the virtual LAN agent with a virtual LAN ID. The virtual LAN agent updates and replies to the server and a node. After random time virtual LAN agent checks its table and sends an updated table to the virtual LAN manager [16]. So that the virtual LAN manager also remains up to date. This is required because after some time, a node that is not in, for some time, its entry from the Ethernet switch is automatically deleted. The second case is when updating is required when a node is in powered condition but whose entry from the cache is removed. It starts reassessing the virtual LAN. Its entry from the node table is updated.

B. Proposed Model of Broadcast by Virtual LAN Manager

The virtual LAN manager repeatedly broadcasts the table, generally after a period of time, which is longer than the previous interval. All virtual LAN agents update their tables after listening to the broadcast [17].

- Virtual LAN Protocol Message Format-1

It is needed to design this format to maintain the node table, as shown in Fig. 2, and details of fields are given just after Fig. 2.

<table>
<thead>
<tr>
<th>Preamble</th>
<th>Msg Type</th>
<th>Sec Bit</th>
<th>HType</th>
<th>HLen</th>
<th>HCPS</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>0 VA</td>
<td>1 VM</td>
<td></td>
<td></td>
<td>6 Ethernet</td>
</tr>
<tr>
<td>VAGENT IP Address VA FADDR</td>
<td>VAGENT HA Address VA Haddr</td>
<td>VAGNET IP Address VA FADDR</td>
<td>VAGENT HA Address VA Haddr</td>
<td></td>
<td></td>
</tr>
<tr>
<td>VAH addr 32 - 47</td>
<td>VM Address 32-47</td>
<td>VMANAGER IP Address – VM iaddr</td>
<td>VMANAGER HA Address – VMN addr</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Table</td>
<td>END Flag</td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

Fig. 2. Virtual LAN message format.

The scheme of the virtual LAN Management is depicted in Fig. 3, assuming that the EDI workstation node in the network is already booted.

- Virtual LAN Protocol Message Format-2

Virtual LAN agents request the server for node address; then, the node sends the IP and hardware address combination to virtual LAN agents. Suppose a virtual LAN ID is not allowed on that node. In that case, the agent sets the validity bit equal to pending and sends a wait message to the server. The virtual LAN agent requests the virtual LAN manager for virtual LAN ID allocation. After the virtual LAN ID request from the virtual LAN agent virtual LAN manager checks the table for the virtual LAN ID entry, then allocate the virtual LAN ID and adds it to the table and sets the validity bit equal to clear, and sends the virtual LAN agent [18]. Also, virtual LAN managers repeatedly broadcast the table to all VAN agents with time intervals which is more than the aging time.

The specified design of the virtual LAN message protocol with different fields is shown in Fig. 4 just after it.
Table 1: Virtual LAN message protocol with different fields for VAN.

<table>
<thead>
<tr>
<th>Preamble</th>
<th>Msg Type</th>
<th>Sec Bit</th>
<th>HType</th>
<th>HLen</th>
<th>HCPS</th>
</tr>
</thead>
<tbody>
<tr>
<td>1 VM</td>
<td>1 Intra VLAN 1 = 100 Mb Ethernet</td>
<td>6 Ethernet</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>0 VA</td>
<td>2 = 100 Mb Ethernet</td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

Table 2: VAGENT IP Address VA IDDR

Table 3: VMANAGER Address VM iaddr

Table 4: Table

Table 5: END Flag

Fig. 4. Virtual LAN message protocol with different fields for VAN.

Preamble: 1 octet. In which if left most bit is 1, it indicates b/c.
Msgtype : 0 Indicates this message is from the Virtual LAN manager is from VLAN manager.
Msgtype: 1. Indicates this message is from a Virtual LAN agent.
Sec bit: -1, Indicates intra Virtual LAN traffic.
Htype: is one octet in length. Hardware address type. For example, 1 for 10 Md Ethernet.
Hlen :1 octet. Optionally used by relay agents when booting via relay agents.
Vaiaddr: Virtual LAN agent's IP address, filled in by the Virtual LAN agent in the request message to the Virtual LAN manager or server.
Table: node entries in tables with address mapping, Virtual LAN ID.
End Flag: Indicates the end of the Virtual LAN message, which is a fixed number.

The scheme of the virtual LAN Management with the dynamic allotment of valid at the time of booting scheme is shown in Fig. 5.

Fig. 5. Scheme of virtual LAN Management with dynamic allotment.

- Proposed Algorithms for Virtual LAN for EDI Services

The message flows between the EDI Node, virtual LAN Agent, and virtual LAN Manager are broadcasted through an algorithm [19] [20]. The design and specification of algorithms are given in Sections III and IV.

IV. VIRTUAL LAN AGENT FUNCTIONAL ALGORITHM

The message between the virtual LAN Agent and Node is defined concerning the algorithmic steps.
1. Check for requests from the node for virtual LAN ID for the node which just came up.

2. 
   a. read MAC, the IP address of nodes from the request message. 
   b. Send a wait message to nodes.

3. Check the nodes table for 
   a. Check mapping of IP, MAC, and virtual LAN ID for node.

4. if (virtual LAN ID for a node is found in the table) 
   a. sends it to requesting node. 
   b. Accept ACK from the node. 
   c. Inform the virtual LAN ID mapping to the virtual LAN manager.

else 
  (Evoke virtual LAN manager) 
  a. Set validity bit as pending. 
  b. Send a request to the virtual LAN manager for virtual LAN ID allotment.

5. Wait for a reply/ack from the virtual LAN manager.
6. Accept virtual LAN ID allotment message from virtual LAN manager.
7. Read the virtual LAN ID from the message
8. Update the virtual LAN agent’s node table.
9. send virtual LAN ID to be requesting node.

V. VIRTUAL LAN MANAGER FUNCTIONAL ALGORITHM

The algorithm for message flow or broadcast has been the dynamic virtual LAN Agent and Manager.

1. Accept request message from virtual LAN agent for virtual LAN ID allocation. Set the message type equal to zero.
2. Check the entry for the given IP address in the table.
3. If (entry of node’s IP address is not present) 
   a. allocation virtual LAN ID for that node. 
   b. Update the virtual LAN manager table with the allotted virtual LAN ID. 
   c. Set virtual LAN ID bit equal to clear.
4. Send virtual LAN ID to virtual LAN agent by setting message type equal to 1.
5. Send a global broadcast on the virtual LAN network to inform all of the updated statuses of the virtual LAN ID table for synchronization.

VI. PROPOSED DEPLOYMENT AND IMPLEMENTATION METHODOLOGY

Meta studies and dynamic virtual LAN design applications are used to collect data. The real-time simulations of dynamic virtual LANs are more helpful in collecting and analyzing the data with the available tool. High-quality lenses tried and tested in industries, are used for the analysis. The Virtual LAN design structure in Cisco Simulator is presented in Fig. 6.

VII. VIRTUAL LAN IMPLEMENTATION & RESULTS

The Ethernet virtual LAN, which functions on layer two, is manually defined by software using which network administrator groups the switch ports into a high bandwidth, low latency switched group [21]. Each virtual LAN with a unique identification number identifies the virtual LAN for network management purposes.

Layer 2 virtual LANs are based strictly on a bridging technique that transmits data using media access control (MAC) and destination addresses. Traffic within virtual LANs is switched using these addresses [22] [23]. Traffic between virtual LANs is carried out by a router that imposes the filtering of packets to provide security and traffic management. The router is either a standalone box-based system or a card integrated into the switch or a node. Routing is handled by a switch that is from the virtual LAN switching logic. The implementation model is given in Fig. 7 with all details of the three virtual LANs, with Cisco Catalyst switch modules with enterprise router interface and external switch interface.

After defining the layer two virtual LANs, each switch reads incoming frames and learns the MAC address associated
with each virtual LAN. If the end station or node sends broadcast or multicast frames, it is forwarded to all ports in that end station or node of virtual LAN [24][25][26]. The ports are spread across any number of switches connected to the backbone. All segments in LAN in a port group are bridged, whether they are on the same switch or separated by backbones.

- Transferring Packets over Backbones

Supporting Layer 2 virtual LAN port groups on a switch is transparent. Pure Ethernet switches store MAC addresses and information about which port each address is to. With a virtual LAN switch, a virtual LAN no. and MAC address and port information are stored in the switch's forwarding table. I observed that these tables grow up to 1,000 bytes or sometimes more than this. On layer 2 VLANS, the virtual LAN messages are sent across shared media backbones using frame tagging and time division multiplexing [27].

- Virtual LAN Using Tagging Method

This method places a unique identifier in each frame's header as that frame enters the switch fabric. With the frame tagging process, a short tag is appended at the beginning of every frame that crosses the backbone; the tag tells which virtual LAN the frame belongs to. The switch permanently stores the port group and is valid [28]. Since a virtual LAN number is carried in each frame, each switch examines the identifier before transmitting it to other switches, routers, or end stations [29].

Fig. 8 elaborates on implementing dynamic virtual LAN using the tagging method. Switch such as Cisco Catalyst and tag filed options with switch interface is mentioned under the figure to detail the tagging method for implementing dynamic virtual LAN.

We received that packet tagging provides a meager latency for assigning and processing frames throughout an enterprise switching fabric, including routers and fast Ethernet backbones [29]. Also, we found one drawback to frame tagging overhead. The frame carries a few bytes of extra baggage.

- Implementation of Virtual LAN using TDM

The implementation and issues related to dynamic virtual LAN using the time division multiplexing mechanisms are considered in this section. Virtual LAN information crosses the campus bus using this technique in layer two virtual LAN. The backbone is divided into 100 Mbps slots, and every virtual LAN is assigned one or more of these time slots, which are used only by virtual LANs. Switches are configured with the information needed to port groups to TDM channels [28][29]. Lan traffic is divided into a separate time slot and reduces the need to use frame tags using which packets are checked. Due to this, network stability increases, and the network load is reduced. As broadcast, one virtual LAN does not affect other virtual LANs. However, I found in this case disadvantage is that other virtual LANs cannot use the bandwidth unused by any Lan. Using this type of virtual LAN requires constant traffic monitoring, ensuring time slots are allocated efficiently. Table 1 illustrates the implementation data with dynamic virtual LAN using the time division multiplexing mechanism.

<table>
<thead>
<tr>
<th>TABLE I. VIRTUAL LAN IMPLEMENTATION USING TDM</th>
</tr>
</thead>
<tbody>
<tr>
<td>LAN</td>
</tr>
<tr>
<td>Virtual LAN 2</td>
</tr>
<tr>
<td>Virtual LAN 1</td>
</tr>
<tr>
<td>Virtual LAN 3</td>
</tr>
</tbody>
</table>

- Implementation of Virtual LAN User-Defined Frame Field

We studied one different method which uses a layer user-defined frame field in layer 2 Dynamic virtual LAN. Virtual LANs defined by common subnet address, protocol type, and other parameters are defined by the value in the user definition frame field [28]. This creates many possibilities, as virtual can be created based on frame values. As port grouping is eliminated, users can move around; there is no network concern for the port it is connected to.

The problem with the frame field method is that it requires storing information on protocols such as Bit mask offset and another routing (switch) software detail [14]. This method allows different stations on a single switch port to be part of different virtual LANs. Also, the Virtual Lan-based group shares a common server or applications [18]. This aspect of port grouping would require switches that are capable of reading and decoding server names, numbers, and other application-specific fields.

- Using MAC address List

A different dynamic virtual LAN model is designed on the MAC address list, and the list is defined manually. This address list sets up protocol and port-independent switched dynamic virtual LANs. For example, node terminals use non-routable protocols to separate Virtual LANs where each dynamic virtual LAN corresponds to a MAC address list. We
found that the MAC address lists technique is more sophisticated than the layer two-port group, as a user can shift to any port in the network and add in proper virtual Lan without reconfiguring the port. The user-defined field under the MAC structure for dynamic virtual LAN MAC is represented in Fig. 5. We observed that this method gives a concrete tool for traffic management. But for extensive networks list becomes lengthy, and if the list is lengthy, it requires more effort.

### VIII. CONCLUSION AND FUTURE WORK

Virtual LAN is the complex local network structure and more secure local network. The deployment of dynamic virtual LAN needs the design and structuring of the network equipment, such as layer three switches, network routers, and many more. In this paper, we have proposed a deployment and implementation model of virtual LAN for VAN of EDI by considering the different factors. The design is so robust that the network function is optimum per the given details. Also, we tested the network functions concerning functional optimization and found that the designed and deployed network is working for the given specifications.

The proposed research work includes the EDI and VAN over the virtual LAN. The applications and other facts related to the research are very clearly specified and deployed over the Cisco simulation. The functional aspects of the given algorithm for VAN and EDI functionalities are very straightforward. They can be taken as an opportunity to advance the current given processes and get the dynamic protocols with existing automated service-based VAN for EDI. This advancement will bring the opportunity to EDI organizations by reducing the network management overhead. So the organization’s cost for information technology support will be drastically reduced.

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An Intelligent Evaluation Path for English Teaching Quality: Construction of an Evaluation Model Based on Improved BPNN

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Abstract—The current intelligent evaluation methods for English teaching quality are inefficient and have poor evaluation accuracy for effective assessment. The paper suggests an evaluation model based on an upgraded Back Propagation Neural Network to overcome the aforementioned issues. First, the principal component analysis is utilized to lessen the dimensionality of the index system as we build an English teaching quality evaluation index system with reference to the results of the existing study. Then, we adopt a multi-strategy improved dragonfly optimization algorithm to evaluate Back Propagation Neural Network for its defects; an algorithm to improve it. Finally, to increase the efficacy and objectivity of English teaching quality evaluation, an intelligent evaluation model based on IDA-BPNN is developed. The experimental results demonstrate that the IDA-BPNN model has an evaluation accuracy of 98.96%, an F1 value of 0.950 on the training set and 0.968 on the test set, a Recall value of 0.948 on the training set and 0.966 on the test set. The aforementioned indicators are all superior to the most recent state-of-the-art approaches for evaluating teaching quality. The aforementioned findings thus demonstrate that the model suggested in the study has high performance and can successfully improve the accuracy and efficiency of English teaching quality evaluation, which has a positive impact on the development of English teaching careers.

Keywords—Teaching quality evaluation; English; BPNN; intelligent evaluation; dragonfly optimization algorithm

I. INTRODUCTION

The importance of English education is increasing as globalisation picks up speed. English teaching (ET) nowadays is a course that is offered in most majors in most universities, aiming to cultivate more and better educated English professionals for the society and market [1,2]. As a result, numerous parties have paid attention to the standard of ET at universities, including school leaders, students' parents, students themselves, and major enterprises. An essential component of ET is quality evaluation (QE), which can assist teachers in identifying areas of weakness in their instruction and working to raise standards to increase effectiveness [3]. Because of its straightforward structure, broad applicability, and superior learning capacity, BP Neural Network (BPNN) is currently a popular tool in the machine learning industry [4]. Some researchers have successfully used BP neural networks for ET QE studies, with promising results. However, the commonly used Analytic Hierarchy Process (AHP) and Fuzzy Comprehensive Evaluation methods in teaching quality evaluation are inefficient and highly subjective, which can affect the results. The performance of traditional BPNN has shortcomings, which have a negative impact on the reform of ET, resulting in poor QE of ET based on BPNN [5]. In order to improve the objectivity and scientificity of English education quality evaluation, provide data support and improvement guidance for English education reform and innovation, this study proposes an improved dragonfly algorithm as the basic basis, and combines BPNN to establish an intelligent evaluation model for English education quality. Principal component analysis is used to reduce the dimensions of the indicator system. The research is innovative in two ways. The first is to implement an intelligent evaluation of ET quality using the enhanced BPNN, which increases the objectivity, accuracy, and effectiveness of ET QE. The second objective is to provide appropriate methods for addressing the DA's flaws and to optimise the BPNN using the enhanced DA, which enhances the functionality of the BPNN model. The research content mainly includes five parts. The first part is a review of the current research status of English education and BPNN. The second part, divided into three sections, constructs an English education quality evaluation model based on the IDA-BPNN model. The first section constructs an English education quality evaluation index system, the second section extracts common factors, and the third section discusses the BPNN optimization of the improved Dragonfly algorithm. The third part analyzes and evaluates the performance of the quality evaluation model. The fourth part discusses the results obtained in the study and fifth part draws the conclusion of constructing an English education quality evaluation model based on the IDA-BPNN model.

II. RELATED WORKS

English is the most commonly spoken language in the age of globalisation, and demand for English majors has skyrocketed internationally. Education in English (ET) is a crucial position for developing students' English skills at universities. In this context, ET in universities has received attention from various researchers, and many scholars have put forward their own insights and strategies for ET. Turan et al. comprehensively compiled and analyzed the recent related literature, so as to explore the application path, application effect and development prospect of flipped classroom in ET, which has positive significance for the advancement of ET
quality [6]. To increase the quality of ET and provide new ideas to the training of English teachers, Ng investigated the role of synchronous online teaching for pre-service English instructors [7]. By using the Arab League nations as an example, Hazaea et al. examined the drawbacks of the remote ET model based on computer technology as well as the potential solutions [8]. Goodman et al. examined the shift from code-switching to cross-linguistic perspectives in English teacher education, providing empirical and theoretical guidance [9]. The necessity and course of the shift to flexible learning, which has implications for English language teaching during the epidemic, was examined by Tarraoy et al. [10] using a state university in the Philippines as an example. They also looked at the development of ET models during the New Coronation epidemic. Based on a thorough analysis of the most recent research literature on English language teaching, Rose et al. studied the state of English and language teaching globally. They then discussed the difficulties that English language teaching is currently facing, provided solutions for those problems, and then talked about the direction and future of English language teaching [11]. Using videos based on multimedia technology in the teaching process and looking into the effects of video teaching, Waluyo et al. researched the effects of multimedia technology in ET and the requirement of its implementation [12]. An extensive analysis of ET in higher education was presented by Block et al., which prompted an examination of STEM lecturers as non-English teachers’ self-positioning was investigated and analysed [13]. This analysis has theoretical ramifications for the advancement of English language instruction in higher education.

The use of machine learning techniques has impacted all facets of daily life and employment, and they are crucial in numerous professions. Numerous researchers have focused on BPNN because of its straightforward structure, superior classification performance, and broad variety of applications. In order to optimise research performance management, Chen et al. first analysed the BPNN's structural makeup. Based on this analysis, they applied BPNN to the problem, effectively raising the bar for research and advancing modern science and technology [14]. The necessity and significance of enterprise financial management risk prediction in the context of the digital economy were examined by Li et al., who realises the intelligent prediction of enterprise financial management risks [15]. The traditional BPNN structure was modified by Li et al., who then used the enhanced BPNN model to create a smart city building information model [16]. In order to improve the performance of BPNNs, Wu Y et al. created a model based on genetic algorithms (GA) improved Simulated Annealing (SA) algorithm and used GA-SA. In order to increase forecast accuracy and ensure production safety, a coal and gas protrusion prediction model was built using the enhanced BPNN [17]. Tang et al. proposed corresponding strategies to improve the BPNN for its defects and implemented medical image segmentation using the improved BPNN, thus assisting clinical diagnosis and treatment. Finally, the effectiveness of the model was verified by using color fundus image retinal vascular image as an example [18]. To improve the performance of BPNN and create GA-BPNN, Wang et al. applied the GA algorithm [19]. The improved adaptive GA algorithm proposed by Yan et al. was used to find the best BPNN parameters and boost model performance. Finally, a vehicle insurance fraud identification model based on the improved BPNN was built, offering fresh perspectives on how to safeguard consumers' rights [20]. In order to provide data assistance for investors’ decision-making, Zhang et al. examined the application path and impact of BPNN in stock price pattern classification and prediction [21].

Nowadays, academics from many nations are paying close attention to ET, and BPNN application and study are very in-depth. ET QE is a crucial basis for assisting teachers in identifying areas of weakness and raising teaching standards. Despite the fact that there have been several studies on the subject, the majority of these findings are primarily theoretical and based on human judgment. This results in the teaching of QE being ineffective and lacking objectivity, which has an impact on the correctness of the teaching of QE. To address the aforementioned issues, the study develops an IEM of ET quality based on the development of BPNN in order to achieve an objective, scientific, and effective evaluation of ET quality. This will provide data support for reform and innovation of ET as well as for quality improvement, and will also show the direction of improvement, which is important for the advancement of English education in China.

III. ENGLISH EDUCATION QUALITY EVALUATION MODEL CONSTRUCTION USING THE IDA-BPNN MODEL

A. Construction of English Education Quality Evaluation Index System

Teaching QE is a vital responsibility with the goal to improve the calibre and effectiveness of ET education. And in order to carry out ET QE, a sensible, reliable, and efficient ET QE index system must be created. To ensure that the ET QE index system is functional and that the ensuing ET QE is accurate, it is essential to undertake ET QE index selection in accordance with scientific, developmental, and process-oriented principles. In light of the foregoing, the study chose language teaching QE indices on the basis of the five dimensions of teaching [22]. Following completion, 19 secondary indicators were attained, and Table I's ET QE index system was created.

In Table I, the ET QE index system constructed in the study contains five dimensions and 19 indicators, which is relatively complex. However, there are some variances in how certain factors affect the quality of ET in the actual ET. Therefore, studying some indicators that have a minor impact on the quality of ET requires more effort, but does little to increase the accuracy of that assessment. Additionally, if there are a lot of indicators, building more input nodes for the BPNN model will be necessary, which will make the model more complex. This is because adding the corresponding indicator data to the BPNN model for learning will result in a lot of dimensions of the input vector features. The length of the model's training process and accuracy both suffer as a result of the model's increasing complexity. The ET QE index mechanism needs to be simplified in light of the aforementioned. In order to score the ET QE index system, the study recruited 25 specialists, including 5 students, 15 English teachers, and 5 ET researchers. After that, values based on
AHP were assigned to each of the 19 indicators in Table I. The top 10 weight values for the indicators were chosen in order to reconstruct the ET QE index system, and Table II displays the simplified ET QE index system.

To assess the validity and dependability of the ET QE index system created by the study, Table II underwent reliability and validity testing. Table III displays the test results.

The validity and feasibility of the study to create an ET QE index system are confirmed in Table III and suggest that the next stage can be carried out in accordance with Table II. Table III shows that the reliability and validity tests of the ET QE index system given in Table II are good.

B. Public Factor Extraction

Following the simplification, the ET QE index system is lowered from 19 to 10 indicators, thus lowering the BPNN model's complexity and raising the model's accuracy. The BPNN model still needs to build 10 input nodes after the streamlining, and this adds to the model's complexity. The dimensionality of the ET QE index system needs to be further lowered with the goal to increase the model's accuracy in assessing ET quality. In Table IV, this stage is accomplished by the study using principal component analysis (PCA) for public factors, as indicated.

The cumulative total variance contribution of the three public components in Table III was 78.05%. There are three factors with a combined value larger than 1 in Table III. These three public factors' variables therefore have a lot of explanatory power and can accurately and thoroughly depict the standard of ET. The constructed factors were extracted by descriptive statistics operation, and then a factor component matrix was obtained. Through this factor component matrix, the correlation between the public factors extracted by the study and the indicators can be described, so that the indicators corresponding to the public factors can be selected. Table V displays the factor component matrix that was culled for the investigation.

<table>
<thead>
<tr>
<th>TABLE I. EVALUATION INDEX SYSTEM FOR ET QUALITY</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Dimension</strong></td>
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<tr>
<td>X1</td>
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<table>
<thead>
<tr>
<th>TABLE II. SIMPLIFIED EVALUATION INDEX SYSTEM FOR ET QUALITY</th>
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<tbody>
<tr>
<td><strong>Dimension</strong></td>
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<tr>
<td>X1</td>
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<td></td>
</tr>
<tr>
<td>X2</td>
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<tr>
<td></td>
</tr>
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<td>X3</td>
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<tr>
<td>X4</td>
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<tr>
<td>X5</td>
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</table>

<table>
<thead>
<tr>
<th>TABLE IV. PCA EXTRACTION RESULTS</th>
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</thead>
<tbody>
<tr>
<td><strong>Composition</strong></td>
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<tr>
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<td>3</td>
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<td>6</td>
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<tr>
<td>8</td>
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<tr>
<td>9</td>
</tr>
<tr>
<td>10</td>
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<table>
<thead>
<tr>
<th>TABLE III. RELIABILITY AND VALIDITY TESTING</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Project</strong></td>
</tr>
<tr>
<td>KMO inspection</td>
</tr>
<tr>
<td>Bartlett sphericity test</td>
</tr>
<tr>
<td>DF</td>
</tr>
<tr>
<td>Significance</td>
</tr>
</tbody>
</table>
The content of Table IV shows that the indicators corresponding to public factor 1 are $U_{i1}$: Improving academic performance; $U_{i2}$: Correct teaching content; $U_{i3}$: Having scientific and ideological qualities. These are the three indicators mentioned above. Therefore, the data corresponding to the above three indicators are input into BPNN, which effectively reduces the input nodes of BPNN.

**TABLE V. STUDY THE EXTRACTED FACTOR COMPONENT MATRIX**

<table>
<thead>
<tr>
<th>Indicator code</th>
<th>1</th>
<th>2</th>
<th>3</th>
</tr>
</thead>
<tbody>
<tr>
<td>$U_{i1}$</td>
<td>1.12</td>
<td>1.02</td>
<td>1.51</td>
</tr>
<tr>
<td>$U_{i2}$</td>
<td>0.31</td>
<td>0.64</td>
<td>0.56</td>
</tr>
<tr>
<td>$U_{i3}$</td>
<td>1.54</td>
<td>0.52</td>
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</tr>
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</tr>
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</tr>
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<td>$U_{i7}$</td>
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<td>0.655</td>
</tr>
<tr>
<td>$U_{i8}$</td>
<td>3.54</td>
<td>1.60</td>
<td>1.355</td>
</tr>
<tr>
<td>$U_{i9}$</td>
<td>0.92</td>
<td>0.53</td>
<td>0.102</td>
</tr>
<tr>
<td>$U_{i10}$</td>
<td>0.95</td>
<td>0.405</td>
<td>1.147</td>
</tr>
</tbody>
</table>

**C. BPNN Optimization-based on Improved Dragonfly Algorithm**

The accuracy and training effectiveness of BPNN will be directly impacted by its initial weights and thresholds. In order to better and more efficiently obtain the best parameter selection for BPNN, the study uses DA to find the optimal model parameters. An optimisation algorithm known as DA, which simulates the behaviour of dragonfly populations, was recently presented. Due to its superior efficiency, DA offers a wider range of applications in diverse optimisation issues. In the population of DA, it is generally divided into two kinds of populations, one of which is a static population, mainly the other is the dynamic population, which is mainly responsible for migratory behavior. In DA in Eq. (1), the separation behavior of the population $S_i$ is shown.

$$S_i = -\sum_{j=1}^{n} X_i - X_j$$  \hspace{1cm} (1)

The dragonfly individual is located at $X_i$, the first $j$ neighbouring individual is located at $X_j$, and the number of neighbours is given by $n$ in Eq. (1). The alignment behavior $A_i$ is calculated by Eq. (2).

$$A_i = \frac{1}{n} \sum_{j=1}^{n} V_j$$  \hspace{1cm} (2)

In Eq. (2), $V_j$ represents the flight rate of $X_j$. The aggregation behavior $C_i$ is expressed as in Eq. (3).

$$C_i = \frac{1}{n} \sum_{j=1}^{n} X_j - X$$  \hspace{1cm} (3)

Foraging behavior $F_i$ See Eq. (4).

$$F_i = X^+ - X$$  \hspace{1cm} (4)

In Eq. (4), $X^+$ is the location of the food the dragonfly is searching for. The avoidance behavior of $E_i$ is shown in Eq. (5).

$$E_i = X^- + X$$  \hspace{1cm} (5)

In Eq. (5), the natural enemy's position is represented by $X^-$. During the DA iteration, the direction of flight of the dragonfly is determined by the step vector $\Delta X$ of the individual dragonfly. During the next iteration, the step vector $\Delta X_{t+1}$ is shown in Eq. (6).

$$\Delta X_{t+1} = (s_iA_i + cC_i + fF_i + eE_i) + u\Delta X_i$$  \hspace{1cm} (6)

In Eq. (6), $s_i, a_i, c, f, e$ is the weight of the individual's separation, alignment, aggregation, foraging, and avoidance behaviors; $w$ is the inertia weight. On this basis, it is possible to obtain the vector $X_{t+1}$ of the individual's position at the next iteration, see Eq. (7).

$$X_{t+1} = X_t + \Delta X_{t+1}$$  \hspace{1cm} (7)

The initial population diversity and worldwide search capabilities of DA are both lacking. To remedy this drawback, the study introduces an inverse chaotic mapping strategy based on Tent chaotic sequences, which optimizes the initial population of DA. If the population individuals have $d$ dimensions, when the population initialization is performed, the individuals of $d$ dimensions are generated $X_{id}$, which are mapped into some space based on the chaos mapping theory to obtain $CX_{id}$. As shown in Eq. (8).

$$\begin{align*}
X_{id} &= X_{mind} + rand(X_{max,d} - X_{mind}) \\
CX_{id} &= X_{mind} + X_{max,d} - X_{id}
\end{align*}$$  \hspace{1cm} (8)

In Eq. (8), $X_{mind},X_{max,d}$ are the lower and upper limits of the values. Let the initial population of DA be $X$, and the population obtained by chaotic mapping be $CX$; merge the two to obtain a new population with $2n$ dragonflies. To determine the top $n$ individuals to create the new beginning population, the fitness values of every individual in the population are calculated. In DA, the traditional inertia weight convergence strategy is linear descent. However, the algorithm's rate of convergence differs from the rate at which the inertia weights converge, which has a detrimental effect on the algorithm's rate of convergence. In order to control the algorithm's convergence, the study suggests a nonlinear decreasing inertia weight technique. Equation (9) illustrates the nonlinear inertia weight lowering technique.

$$w = w_{min} + |w_{max} - w_{min}| \exp \left(\frac{-at}{t_{max}}\right)$$  \hspace{1cm} (9)

In Eq. (9), $w_{max,}w_{min}$ are the maximum and minimum values of the set inertia weights; $\alpha$ is the adjustment coefficient; and $t_{max}$ is the maximum number of set iterations. Fig. 1 depicts the enhanced inertia weight
The study's better inertia weighting approach is nonlinearly reducing in Fig. 1. Under this strategy, the inertia weights are basically consistent with the convergence trend of the algorithm, which improves the convergence of DA. Based on the foregoing, IDA is created, used to optimise BPNN, and integrated with the preceding materials to create an ET quality assessment model that uses IDA-BPNN in order to achieve intelligent evaluation of ET quality and increase ET efficacy.

IV. PERFORMANCE ANALYSIS OF IDA-BPNN ELT QUALITY ASSESSMENT MODEL

To give a wise, scientific, and impartial evaluation of ET quality, the study suggests an IDA-BPNN ET quality assessment methodology. To create the experimental data set, the necessary index data were pulled from a university's education administration system with the leaders of the school's consent. The experimental data set was divided in a 7:3 manner. The two main cutting-edge algorithmic models used in the current intelligent evaluation of ELT quality are the BPNN model (IPSO-BPNN) based on Improved Particle Swarm Optimisation (PSO) and the BPNN model (GWO-BPNN) based on Grey Wolf Optimisation Algorithm (GWOA) optimisation. In IDA-BPNN, the number of iterations is 200, the learning factor is 0.3, the input layer is 4, the hidden layer is 20, and the output layer is 1. As a result, the effectiveness of the IDA-BPNN, IPSO-BPNN, and GWO-BPNN models is contrasted. The effectiveness of the three common factors extracted based on the PCA method is first investigated using the gravel plot analysis, as shown in Fig. 2. The slope of the curve is greater in the first three common components, as seen in Fig. 2. While in the other public factors except the first three public factors, the slopes of the curves at their positions are smaller and the trends are flatter. The above results show that the three public factors extracted by the study can reflect the ET situation more accurately, comprehensively and effectively, which proves the validity of the public factors extracted by the study.
Use the aforementioned model to evaluate the ET data in the university's educational system, analyse the disparities between the model's evaluation values and the actual values, and determine the model's evaluation correctness. Fig. 6 displays the evaluation accuracy of many models. As can be seen, the IDA-BPNN model outperformed the IPSO-BPNN model and the GWO-BPNN model in terms of assessment accuracy, with 98.96% on the test set, which is higher by 1.02% and 1.24%, respectively.

Fig. 7 illustrates how the MAE values of the IDA-BPNN model, IPSO-BPNN model, and GWO-BPNN model change on the test set after all models have been fully trained using the training set. It is clear that the MAE values of the models have been trending downward over the rounds. The IDA-BPNN model's MAE value after 40 iterations is 0.752, which is 0.048 and 0.097 less than that of the IPSO-BPNN model and the GWO-BPNN model, respectively. In conclusion, the IDA-BPNN ET QE model developed by the study has high evaluation accuracy and efficiency, can perform ET QE objectively and efficiently, and has favourable implications for the development of English professionals as well as for the growth of ET.
the data used in the experiment came from the same university, and the sample range is limited, which may lead to accidental factors in the experiment. Future research should expand the sample size to further enhance the credibility of the study.

VII. FUNDINGS

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REFERENCES


Determinants of Medical Internet of Things Adoption in Healthcare and the Role of Demographic Factors Incorporating Modified UTAUT

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Abstract—Medical Internet of Things (mIoT) is the IoT subset with vast potential in healthcare. However, the adoption of eHealth solutions such as mIoT has been a critical challenge in the health sector of the Kingdom of Saudi Arabia. Therefore, this study was conducted to explore the mIoT adoption determinants in Saudi public hospitals. Methods: A total of 271 participants were recruited from public hospitals in Riyadh, and a modified UTAUT model named UTAUT-HS was developed in this study to test its relevance with respect to mIoT adoption. Results: Ten path relationships were tested in this study, out of which six showed significant results. Similarly, three variables (Computer and English Language Self-efficacy or CESE, Performance Expectancy or PE and Social Influence or SI) showed a significant direct relationship with the behavioural intention to adopt mIoT. Furthermore, CESE showed the strongest relationship and emerged as a major subset of Effort Expectancy (EE) for mIoT adoption. However, moderator analysis showed substantial variations between different study demographic groups. In particular, the current study findings unravelled a comparatively novel relevance of Perceived Threat to Autonomy (PTA) for mIoT adoption for clinical and non-clinical and for older and younger participants. Conclusion: The study concludes that UTAUT-HS is an adequate model to explain the mIoT adoption in healthcare. However, it also suggests conducting future large-scale studies in KSA and elsewhere to validate the relevance of UTAUT-HS in other contexts and with much more confidence.

Keywords—Medical internet of things; eHealth adoption; modified UTAUT; demographics and IT adoption

I. INTRODUCTION

The term "Internet of Things" (IoT) was coined by Kevin Ashton in 1999, who was a British technology pioneer and worked at the Massachusetts Institute of Technology [1,2]. IoT has many definitions; however, in broad terms, it can be defined as a combination of different components, such as smart devices or machines that communicate with each other over the Internet, gather information, and make decisions without human intervention [3]. Medical Internet of things or mIoT is the application of IoT in healthcare and simply can be defined as a consolidation of devices and applications that can link to information technology systems in healthcare using a range of networking technologies [4]. mIoT has tremendous capabilities in healthcare and it ranges from reducing healthcare costs, load on clinicians and medical errors [5,6, 7] to improving treatment outcomes, compliance by patients and overall quality of healthcare [8,9].

However, despite a wide range of advantages, a recent systematic review on IoT adoption suggests that the inclusion and acceptance of IoT in healthcare is still low [10]. It is important to note that successful implementation and adoption of new technologies such as mIoT is not an easy process and is affected by many main interrelated factors, namely social, personal, technical, and organisational factors [11]. However, it has been suggested by many researchers that the greatest challenge of mIoT and AI adoption in healthcare is not the efficacy of the technology but the acceptance by the clinicians [12,10]. The adoption of any new technology in any society is a complex process and the process becomes more challenging if the society is comparatively restrictive in nature, such as the Kingdom of Saudi Arabia and the technology has enormous disruptive power, such as mIoT.

Moreover, in KSA, the resistance to change and to adopt new technology, lack of compliance by the healthcare staff and inadequacies in the policies to introduce and implement new IT-based solutions have created already bad condition worse for the introduction of more complex eHealth solutions such as mIoT [13]. For instance, the past eHealth research (Electronic Health Record (EHR) systems) conducted in the Kingdom suggests that there was underutilisation of EHR functionalities across the board in the hospital [14]. Moreover, healthcare professionals have reported data entry time, lack of adequate IT training and support, the complexity of technology, lack of customizability option of the EHR systems, and disturbance in communication between doctors and patients as grave barriers linked with EHR adoption [14]. Therefore, it is likely that the introduction of mIoT in the Kingdom will experience significant resistance and interruptions. Also, no quantitative studies are available investigating the adoption determinants of mIoT in Saudi hospitals. Thus, research focused on understanding the adoption dynamics of mIoT in the Kingdom of Saudi Arabia should be conducted to support the smooth uptake of these technologies.

To achieve the aim of this study, a brief description of prevailing technology adoption theories is provided to propose the current study framework. Detailed information pertaining to each component of the proposed framework, along with the rationale for their inclusion, is provided in the following sections. Moreover, the relevance of the selected moderators in the current study framework is also established in the subsequent sections.
A. Past Theories Related to the Adoption of New Technologies

A significant body of research has been conducted to determine the factors associated with the adoption of new innovations/technologies and aspects related to human behaviours. The most famous theories incorporated by researchers investigating the adoption of new technologies include the Theory of Reasoned Action (TRA) proposed in 1975, The Technology Acceptance Model (TAM) proposed in 1989, The Theory of Planned Behaviour (TPB) proposed in 1991, The Unified Theory of Acceptance and Use of Technology (UTAUT) proposed in 2003 and The Unified Theory of Acceptance and Use of Technology 2 (UTAUT2) proposed in 2012 [15,16,17,18,19].

The Unified Theory of Acceptance and Use of Technology or UTAUT was proposed in 2003 and was based on eight past theories, which included TRA, TAM, TPB, Innovation Diffusion Theory (IDT), Combined TAM-TPB (C-TAM-TPB), Social Cognitive Theory (SCT) and Model of PC Utilization (MPCU) [18]. UTAUT included four main constructs; Performance Expectancy (PE), Effort Expectancy (EE), Social Influence (SI), and Facilitating Conditions (FC). Along with these four constructs, UTAUT also incorporated four moderators (age, gender, experience and voluntariness), which were assumed to influence the constructs [15]. However, it is essential to note that only PE, EE and SI were directly linked with Behavioural Intention to Adopt Technology (BI), while FC was majorly linked with the actual use of the technology.

Since the development of UTAUT, it has been used by various studies from a broad range of disciplines to explore and explain the adoption of technology majorly at an individual level [20]. Moreover, it has been suggested that the variance explaining power of UTAUT is about 70% and it has outperformed all the other past eight models (eight models explained between 17% and 53% of the variance in BI) that were used to construct UTAUT [18].

II. Proposed Framework for the Current Study - The Unified Theory of Acceptance and Use of Technology for Hospital Staff (UTAUT-HS)

Comparing the characteristics of the past study models, the current study considered UTAUT as the most suitable model to develop the current study framework (Fig. 1). The original UTAUT model proposed four key constructs: PE, EE, SI and FC. However, FC is considered to be affecting the actual use of IT and not directly linked with the adoption or acceptance of IT in the original UTAUT model. Thus, it was excluded from the current study proposed framework. Further, based on the literature search conducted on the adoption of IT technologies in the Saudi healthcare system and elsewhere, three additional components, Computer and English language Self-Efficacy (CESE), Perceived Threat to Autonomy (PTA), Confidentiality Concerns (CC), were included in the current study proposed framework. Similarly, two moderators; gender and age, were adopted from the UTAUT model and two more; occupation and education, were included in the proposed framework.

A. Performance Expectancy (PE)

Performance Expectancy is defined as the extent to which a particular technology brings about effectiveness in work performance and this concept is similar to Perceived Usefulness (PU) from TAM [21,18,17]. Gao et al. [22] suggested that when healthcare consumers believe that the introduction of IoT will enable them to improve effectiveness, they are more likely to accept the technology. As far as eHealth is concerned, PE has been reported by many studies to be a significant contributing factor for technology adoption among healthcare professionals [23,24,25]. Moreover, the lack of PE of information technology was also observed to be a crucial hurdle in healthcare, and the literature concerning the adoption of EHR in KSA has suggested lack of Perceived Usefulness (PU) or PE to be contributing 15% of all barriers [26].

H1: Performance Expectancy (PE) will have a positive influence on the Behavioural Intention to Adopt mIoT (BI). Age, gender, occupation, and education will moderate the influence of PE on BI.

B. Effort Expectancy (EE)

Effort Expectancy (EE) is regarded to be clearly linked with the ease of use of Information technology and it is similar to the Perceived Ease of Use (PEOU) from TAM [27,15]. In other words, it refers to the extent to which an individual considers the use of technology free of effort. A range of past studies has endorsed the substantial influence of PEOU and it has been reported that EE plays a positive role in the adoption or acceptance of healthcare information systems, clinical decision support systems, adverse event reporting systems and many more [28,29,30,31]. The significance of PEOU is also supported by Alqahtani et al. [27], who suggested that 15% of all EHR adoption barriers in the KSA were linked to PEOU [26].

H2: Effort Expectancy (EE) will have a positive influence on the Behavioural Intention to Adopt mIoT (BI). Age, gender, occupation, and education will moderate the influence of EE on BI.

H3: Effort Expectancy (EE) will have a positive influence on Performance Expectancy (PE).
C. Social Influence (SI)
Social influence is defined as the perception of an individual about a product, technology or services, which is substantially influenced by the perception of people around him/her and it is similar to the subjective norm from TRA and TBP [32,33,19,11]. In the case of IoT or mIoT technology, the majority of the potential users lack adequate information about it. Thus, the impact of SI is even amplified in the decision-making process [32]. The social network has a crucial role in the adoption or acceptance of IoT technologies since it has been a general observation that IoT users seek assistance and advice from family, peer and colleagues to clear uncertainties about the product [34].

Moreover, in the healthcare context, many studies have found a substantial role of SI on technology acceptance among doctors and physicians [24,35]. Similarly, it is also hypothesised that the norms will affect the expectations linked with the Computer and English Self-efficacy of healthcare professionals. Since, KSA is conventionally considered a reserved culture and a lot of significant changes are introduced under the progressive 2030 Vision approach, thus there is more significant uncertainty about the influence of SI on the intention to adopt mIoT [36,37,38,39].

H4: Social Influence (SI) will positively influence the Behavioural Intention to Adopt mIoT (BI). Age, gender, occupation, and education will moderate the influence of SI on BI.

H5: Social Influence (SI) will have a positive influence on the Computer and English Self-efficacy (CESE).

D. Computer and English Language Self-Efficacy (CESE)
Self-efficacy is defined as the self-evaluation of an individual to conduct a particular task and in terms of mIoT adoption, CESE is associated with the self-evaluation of computer skills and English language proficiency [40]. In the healthcare context, mIoT technology adoption remains a significant issue and is substantially associated with technological skills and abilities [22]. Moreover, past literature has indicated the lack of familiarity of doctors with information technology to be a significant barrier obstructing the adoption and use of those technologies [41,42]. The Kingdom of Saudi Arabia has also witnessed this issue and has reported it in many past studies conducted in the healthcare environment [43,44,45,46].

Despite the fact that many studies in KSA have highlighted this issue, only a few studies have explored the role of this issue in the acceptance of information technologies by healthcare professionals [43,44,45,41,42]. Concerning English language proficiency, a vital link has also been found with eHealth use in the Kingdom [44,45].

H6: Computer and English language Self-Efficacy (CESE) will positively influence the Behavioural Intention to Adopt mIoT (BI). Age, gender, occupation, and education will moderate the influence of CESE on BI.

H7: Computer and English language Self-Efficacy (CESE) will have a positive influence on the Effort Expectancy (EE).

E. Perceived Threat to Autonomy (PTA)
Perceived Threat to Autonomy (PTA) is not a new concept as [47] Walter & Lopez (2008) described PTA as the extent to which an individual thinks that incorporating a particular technology or system will compromise his/her control over the procedures, policies and functions of their work [47]. However, this concept has become substantially relevant with respect to the adoption or acceptance of mIoT. Conventionally, it is believed that healthcare professionals and in particular, doctors hold a high level of professional autonomy and the introduction of mIoT can affect the power dynamics in the healthcare environment [48,47]. Safi et al. [49] noted that the doctors in their study highlighted the possible interference of technology with their autonomy in their diagnostic process. The doctors showed high concerns related to the use of eHealth tools by the management to control them. These perceptions ultimately led to a negative attitude toward the acceptance of change brought by the technologies [49].

Carcary et al. [50] asserted that one of the key reasons for resistance to IoT is its very different nature from other eHealth technologies. The key new component in IoT is AI or machine learning (which has not been a major part of past technologies). AI is perceived as a direct threat because of its capability to replicate the human performance. The research conducted in the Saudi healthcare environment has shown the understating of this barrier by healthcare professionals, especially doctors. The studies by Abdullah and Fakieh [51] and Qurashi et al. [52] have reported that doctors and other healthcare employees are worried about their job security due to AI. Moreover, the qualitative study by Alsaleh et al. [48] concluded that healthcare professionals are resistant to the adoption of eHealth technologies due to the fear that they might lose their privileges, which suggests a loss of respect and autonomy. Therefore, the following hypothesis was developed.

H8: Perceived Threat to Autonomy (PTA) will have an influence on the Behavioural Intention to Adopt mIoT (BI). Age, gender, occupation, and education will moderate the influence of PTA on BI.

F. Confidentiality Concerns (CC)
Confidentiality Concerns (CC) can be defined as the extent to which the use of a particular technology can increase the risk of loss of personal or important information. Past research conducted in the Kingdom of Saudi Arabia has frequently reported CC causing major hurdle in the acceptance of eHealth technologies [53,54,55,44,45,56]. For instance, the study by Albarrak et al. [53] found that 90% of the doctors showed concerns about patient privacy (among other concerns) for the adoption of telemedicine. Similarly, Alqiahtani [54] study noted privacy and security issues as major obstacles preventing Saudi healthcare from adopting IoT technology. Moreover, past research has also identified an association between CC and PE, which suggests the indirect effect of CC on the acceptance of technology [56].

Among all healthcare professionals, doctors are believed to be concerned about CC the most, even more than the patient themselves [41]. This could be because they are at the frontline and feel more responsibility for patient information protection. It is visible from the observation that doctors who use even
basic eHealth technologies such as EHR have suggested an increased risk of confidentiality and security issues while comparing EHR with paper-based record systems [41]. Even though many past studies in KSA have highlighted CC to be a significant obstacle to mIoT and e-health adoption and use, the literature investigating the influence of CC on the adoption intention of mIoT is limited, highlighting the need for CC to be included in the current study framework [55,44,45].

G. Moderators in the Framework

1) Gender: Hoque [57] argues that gender as a moderator of technology adoption have received less attention compared to other demographic factors age, experience or culture. However, past studies conducted on the adoption of technology have found significant influence of gender [58,59,60]. Despite this some of the most commonly used technology acceptance models such as TAM have made no reference of the impact of gender on the IT acceptance model [57]. Overall, it has been suggested that male possess less suspicion about technology and hold more positive views them, while female lack confidence in computer usage [57,61]. However, with respect to mIoT, the study by Karahoca et al. [62] found out that male showed more privacy and healthcare vulnerability concerns than females. While other studies have suggested that male’s provided higher score on PU Or PE and PEOU or EE compared to females [63].

However, again these patterns are not consistent and varies substantially from technology to technology or region to region. For instance, the Bangladeshi study on mHealth adoption showed that male provided higher scores on PEOU than females (0.6556 versus 0.1445, \(t=3.784\)), while females provided higher PU scores (0.3244 versus 0.0140, \(t=2.104\)) [57]. However, the key standpoint behind the role of gender on technology adoption is associated with the social roles rather than any biological mechanism and there is major discrepancy among male and female societal roles in KSA, therefore this aspect is quite important for the Kingdom [64].

H9: Confidentiality Concerns (CC) will have a negative influence on the Behavioural Intention to Adopt mIoT (BI). Age, gender, occupation, and education will moderate the influence of CC on BI.

H10: Confidentiality Concerns (CC) will have a negative influence on Performance Expectancy (PE).

2) Age: Age is another noteworthy factor that may influence the adoption of mIoT. Sivathanu [65] conducted a study to evaluate the adoption factors for IoT-based healthcare wearables. The study found that older individuals experience mainly three types of barriers (traditional, usage and risk barriers), which are related to age and can influence (reasons against) the adoption of IoT-based healthcare wearables [65]. Similarly, the study found that elderly participants were more inclined toward visiting doctors in person rather than conducting online consultations. They believe that doctors provide more personalised healthcare services in person and also believe that the use of eHealth or wearable healthcare devices is a challenging task. Given that the current study is aimed to be conducted in KSA, the aspect of age become significantly important due to the rapidly changing cultural environment in the Kingdom, which might increase the generational gap and consequently widen the perceptual gap towards mIoT technology [66,67].

These presumptions related to elderly participants are also proved by various past studies conducted in the healthcare domain. For instance, the study by Parthasarathy et al. [68], focus on the concerns associated with the adoption of EMR, showed that negative beliefs and attitudes towards the use of computers and inadequate motivation to change readiness negatively influenced IT adoption among nurses. The research concluded that these negative factors were most commonly found among older nurses. Moreover, it was asserted that a one-size-fits-all training style is inadequate in healthcare for adopting new technologies [68]. The study by Al Otaybi et al. [69] is one of those few studies which showed that age difference in healthcare is a determining factor with respect to EMR satisfaction. Hence, age is considered a predisposing factor in technology adoption, and healthcare organisations are often challenged to provide a wholistic environment that can meet the needs of all, support change readiness and enhance the digital competency of all healthcare professionals [70].

3) Occupation: Hospitals include a range of staff members, which includes but not limited to doctors, nurses, pharmacists, IT staff and management employees. Further, among doctors, there are various categorizations based on their skills and qualifications. Therefore, the difference in the field of practice may produce a difference in attitude towards the adoption of the same technology. This inference is supported by the systematic review conducted by Boonstra et al. [71] on the adoption of EHR in hospitals. The review suggested that the leadership has to focus on the work conducted by different healthcare professionals and the impact of health information technology on the flow of those works to ensure a smooth technology transition [71].

It is essential to consider that a particular occupation in the hospital encompasses specific roles, responsibilities and duties, determines the level of interaction with other colleagues and patients and requires certain level of education [72,73,74]. All these factors can influence the knowledge and perception towards e-health technologies or more precisely, towards mIoT. For instance, the study by Afolaranmi et al. [75] found a significant correlation between good EHR knowledge and different hospital professions. Among clinical professions, pharmacists showed the highest positive knowledge of EHR, followed by resident doctors and nurses/midwives. Thus, these aspects are essential and occupation should be included as a moderator for the adoption of mIoT.

4) Education: Similar to gender, age and occupation, education is another significant determinant that can act as a substantial moderator in the current study. Past studies have indicated a significant role of the level of education on technology awareness and perception in the healthcare sector [76,45,77]. Healthcare professionals holding a master’s or above education have been identified to be significantly more
eHealth aware than their counterparts with a bachelor's or below education [76,45,77]. Similarly, Saudi research by Hasanain et al. [45] indicated a statistically significant association between EMR literacy, computer literacy, English language proficiency level and healthcare professionals' education level. Thus, the inclusion of education as a moderator was important for the current study.

III. METHODOLOGY

The current study aimed to evaluate the determinants that were associated with the adoption of mIoT among hospital care staff in Saudi Arabia. To achieve this aim, Ministry of Health (MOH) hospitals in Riyadh were selected for the recruitment of the participants. MOH hospitals are public entities, providing 60% of the total healthcare services in the Kingdom [78]. The selection of Riyadh was made because it is the biggest city in the Kingdom and is at the forefront to receive technological innovation in the country [44]. Thus, 271 participants, including doctors, nurses, pharmacists, and non-clinical personals i.e., IT individuals, technicians and managerial personnel working in the MOH hospitals in Riyadh, were recruited. Before the recruitment, relevant ethics approvals were sought from the Latrobe University, Melbourne, Australia (ethics no: HEC19482) and from the MOH of Kingdom of Saudi Arabia (ethics no: 21-79 E).

A. Data Analysis

Model testing is an integral part of research as it allows the researcher to identify the relationship between different variables included in the study model. Therefore, structural equation modeling (SEM) using SmartPLS version 3 was incorporated in this research for model testing. SEM is a complex multivariate statistical analysis technique that permits researchers to examine the nature and significance of relationships among various exogenous and endogenous variables [79]. Also, conducting SEM using SmartPLS is particularly beneficial as it requires no assumptions related to the distribution of the study data and is suitable for a comparatively small sample size [80,81].

IV. STUDY MODEL VALIDATION – MAIN RESULTS

To check the indicator reliability, SmartPLS version 3 was used to calculate the Factor loadings (Table X in Appendix) of the indicators included in each variable. Henseler et al. [82] suggested that a factor loading of 0.7 or higher is considered highly satisfactory, whereas the value of 0.5 or above is considered acceptable [83]. Table I demonstrates that all outer loadings were above 0.7; hence the acceptance criterion was fulfilled. The second parameter for model evaluation was the assessment of construct reliability which was carried out by measuring Cronbach's Alpha and Composite Reliability (CR) (Table I). The value of Cronbach's Alpha and CR above 0.7 is considered adequate to establish internal consistency reliability [83]. Table I shows that all values for Cronbach's Alpha, CR and rho_A (similar to CR calculated by SmartPLS) were higher than 0.7; hence this criterion was also fulfilled.

For the assessment of validity, the variables were evaluated for convergent and discriminant validity. Convergent validity measurement was carried out by calculating the Average Variance Extracted (AVE) through SmartPLS [83]. [83] Hair et al. (2011) suggested that an AVE value of greater than 0.5 is considered acceptable to determine the convergent validity of the variables. Table I illustrates that all variables had an AVE value of greater than 0.5; hence no adjustment was required.

Discriminant validity is another significant factor that is required to be evaluated to assess the quality of the measurement model. Discriminant validity measures the extent to which the indicators are different from each other empirically [84,85]. The discriminant validity can be measured by using the Fornell & Larcker criterion method, the Heterotrait-monotrait (HTMT) ratio of correlation, and by evaluating the cross-loadings of the indicators. The latent variable should explain better variance of its own indicator compared to the variance of other latent variables. Thus, the values in the Fornell-Lacker method for each latent variable should be higher than the correlation with other latent variables. Table II illustrates that the AVE square root values for each latent variable were greater than all the other correlations; hence this criterion was satisfied [84,85].

| Table I: Internal Consistency, Reliability and Validity Results for the PLS-SEM Measurement Model |
|---------------------------------------------------------------|-------------------|-----------------|-----------------|
| Cronbach's Alpha | rho_A | Composite Reliability | Average Variance Extracted (AVE) |
| BI | 0.927 | 0.928 | 0.965 | 0.932 |
| CC | 0.933 | 0.933 | 0.957 | 0.882 |
| CESE | 0.949 | 0.949 | 0.963 | 0.868 |
| EE | 0.943 | 0.944 | 0.959 | 0.855 |
| PE | 0.904 | 0.912 | 0.933 | 0.777 |
| PTA | 0.913 | 0.922 | 0.938 | 0.792 |
| SI | 0.959 | 0.960 | 0.971 | 0.892 |

| Table II: Fornell-Lacker Results for Discriminant Validity Testing |
|---------------------------------------------------------------|-------------------|-----------------|-----------------|-----------------|-----------------|-----------------|
| BI | CC | CESE | EE | PE | PTA | SI |
| BI | 0.966 | | | | | |
| CC | 0.631 | 0.939 | | | | |
| CESE | 0.810 | 0.690 | 0.932 | | | |
| EE | 0.762 | 0.648 | 0.833 | 0.924 | | |
| PE | 0.699 | 0.538 | 0.715 | 0.837 | 0.882 | |
| PTA | 0.761 | 0.873 | 0.813 | 0.749 | 0.630 | 0.890 |
| SI | 0.768 | 0.728 | 0.797 | 0.810 | 0.742 | 0.851 | 0.944 |

The other method to assess the discriminant validity includes the evaluation of HTMT ratios of correlation [84,85]. [82] Henseler et al. [82] suggested that this method is superior to the Fornell-Lacker method as it can achieve higher rates of specificity and sensitivity. The acceptable threshold of the HTMT ratio ranged between 0.85 and 0.90, and anything above these values can indicate issues of discriminant validity [86,87]. Table III demonstrates a PTA-CC HTMT ratio of 0.957, which required treatment. To identify the point of concern, cross-loadings were checked [84,85].
E was explained by EE, 69.4% variance in EE was explained by SI. Similarly, 63.6% variance in CESE was explained by SI. In P variables, which suggests a good fit. Similarly, 70.1% variance in BI was explained by the independent variables, which suggests a good fit. Similarly, 70.1% variance in BI was explained by the independent variable.  

Three issues were identified in the indicator 36, 39 and 42, because their cross-loadings were higher in other latent variables compared to theirs, thus these were dropped. HTMT ratios were checked again, and it showed that the issue was resolved (Table IV).

### Table III. HTMT Ratio for Discriminant Validity Testing

<table>
<thead>
<tr>
<th></th>
<th>BI</th>
<th>CC</th>
<th>CESE</th>
<th>EE</th>
<th>PE</th>
<th>PTA</th>
<th>SI</th>
</tr>
</thead>
<tbody>
<tr>
<td>BI</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>CC</td>
<td>0.678</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>CESE</td>
<td>0.863</td>
<td>0.732</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>EE</td>
<td>0.815</td>
<td>0.691</td>
<td>0.880</td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>PE</td>
<td>0.760</td>
<td>0.581</td>
<td>0.770</td>
<td>0.902</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>PTA</td>
<td>0.811</td>
<td>0.957</td>
<td>0.856</td>
<td>0.791</td>
<td>0.669</td>
<td></td>
<td></td>
</tr>
<tr>
<td>SI</td>
<td>0.813</td>
<td>0.769</td>
<td>0.835</td>
<td>0.851</td>
<td>0.792</td>
<td>0.900</td>
<td></td>
</tr>
</tbody>
</table>

Note: Indicators 36, 39 and 42 were dropped because their cross-loadings were higher in other latent variables compared to theirs.

### Table IV. Re-Checking HTMT Ratio for Discriminant Validity Testing

<table>
<thead>
<tr>
<th></th>
<th>BI</th>
<th>CC</th>
<th>CESE</th>
<th>EE</th>
<th>PE</th>
<th>PTA</th>
<th>SI</th>
</tr>
</thead>
<tbody>
<tr>
<td>BI</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>CC</td>
<td>0.683</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>CESE</td>
<td>0.863</td>
<td>0.744</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>EE</td>
<td>0.815</td>
<td>0.691</td>
<td>0.880</td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>PE</td>
<td>0.760</td>
<td>0.564</td>
<td>0.770</td>
<td>0.902</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>PTA</td>
<td>0.733</td>
<td>0.895</td>
<td>0.760</td>
<td>0.714</td>
<td>0.583</td>
<td></td>
<td></td>
</tr>
<tr>
<td>SI</td>
<td>0.813</td>
<td>0.734</td>
<td>0.835</td>
<td>0.851</td>
<td>0.792</td>
<td>0.835</td>
<td></td>
</tr>
</tbody>
</table>

Note: Indicators 36, 39 and 42 were dropped because their cross-loadings were higher in other latent variables compared to theirs.

### Table V. Overall Path Coefficients, T Statistics and P Values of the Model

<table>
<thead>
<tr>
<th></th>
<th>Path coefficients</th>
<th>T Statistics</th>
<th>p values</th>
</tr>
</thead>
<tbody>
<tr>
<td>CC -&gt; BI</td>
<td>0.002</td>
<td>0.023</td>
<td>0.982</td>
</tr>
<tr>
<td>CC -&gt; PE</td>
<td>-0.033</td>
<td>0.644</td>
<td>0.520</td>
</tr>
<tr>
<td>CESE -&gt; BI</td>
<td>0.437</td>
<td>5.798</td>
<td>0.000</td>
</tr>
<tr>
<td>CESE -&gt; EE</td>
<td>0.833</td>
<td>19.175</td>
<td>0.000</td>
</tr>
<tr>
<td>EE -&gt; BI</td>
<td>0.057</td>
<td>0.568</td>
<td>0.570</td>
</tr>
<tr>
<td>EE -&gt; PE</td>
<td>0.858</td>
<td>17.713</td>
<td>0.000</td>
</tr>
<tr>
<td>PE -&gt; BI</td>
<td>0.148</td>
<td>2.139</td>
<td>0.032</td>
</tr>
<tr>
<td>PTA -&gt; BI</td>
<td>0.111</td>
<td>1.281</td>
<td>0.200</td>
</tr>
<tr>
<td>SI -&gt; BI</td>
<td>0.175</td>
<td>2.255</td>
<td>0.024</td>
</tr>
<tr>
<td>SI -&gt; CESE</td>
<td>0.797</td>
<td>20.413</td>
<td>0.000</td>
</tr>
</tbody>
</table>

### A. Structural Model Results

The structural model evaluates the strength and significance of the relationship between independent and dependent variables by assessing the R square, path coefficient (β) and its significance level, which is assessed through the t-test and p-value [88]. R square provides the extent of variance explained by the independent variable, while β explains the strength of an effect from the independent variable to the dependent variable. Similarly, a t-value of above 1.96 and a p-value of below 0.05 suggests the significance of the relationship [88].

Fig. 2 and Table V demonstrate the path coefficients (β) and the corresponding significance levels. It can be observed that out of six independent variables for the Behavioural Intention to Adopt mIoT (BI), only three variables showed a significant relationship. Three variables - CESE, PE and SI positively influenced the BI. Also, out of these three, CESE had the strongest effect (β = 0.437, p < 0.001), followed by SI (β = 0.175, p = 0.024) and PE (β = 0.148, p = 0.032). Further, CESE -> EE, EE -> PE and SI -> CESE showed a significant strong positive relationship. Of these three, EE -> PE showed the strongest relationship (β = 0.858, p < 0.001). Further, 71.2% variance in BI was explained by the independent variables, which suggests a good fit. Similarly, 70.1% variance in PE was explained by EE, 69.4% variance in EE was explained by CESE and 63.6% variance in CESE was explained by SI.

### B. Moderator Analysis

1) Gender: Sub-group analysis was conducted for gender, which included separate analyses for male and female participants. Table VI illustrates that out of six independent variables for the BI, two variables showed a significant relationship for male (CESE and PE) and only one showed a significant relationship for female (CESE) participants. Also, the strength of relationships between CESE and BI and PE and BI were stronger for male participants compared to female and overall results (Table VI). Moreover, SI -> CESE, CESE -> EE and EE -> PE showed significant positive relationships for both male and female participants, however, male participants showed comparatively stronger relationships. With respect to R square values, 81.2% variance in BI was explained by the model for male participants, while this percentage was only 51.9% for female participants.
2) Age: Sub-group analysis was conducted for age and separate analysis for participants aged 18 to 35 years and 36 to 70 years was conducted. Table VII shows that out of six independent variables, three variables (CESE, PE and PTA) showed a significant positive relationship for the 18 to 35 years group and two variables (CESE and SI) showed a significant positive relationship for the 36 to 70 years group with BI. Similarly, the 18 to 35 years group showed a significant positive relationship between PTA and BI (β = 0.258, p = 0.031), which is not shown by the overall results. On the other hand, the 36 to 70 years showed a non-significant negative relationship between PTA and BI. Moreover, the strength of CESE with BI was much stronger for the 36 to 70 years group compared to the 18 to 35 years group (β = 0.536 vs 0.353, respectively) (Table VII).

The variables CESE -> EE, EE -> PE and SI -> CESE showed a significant positive relationship for both groups (Table VII). Also, the 36 to 70 years group showed a significant negative relationship between CC -> PE (Table VII). With respect to the R square, 78.6% variance in BI was explained by the model for the 36 to 70 years group, while this percentage was 64.6% for the 18 to 35 years group.

TABLE VI. PATH COEFFICIENTS AND P VALUES OF MALE AND FEMALE PARTICIPANTS

<table>
<thead>
<tr>
<th>Gender</th>
<th>Male</th>
<th>Female</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>Path coefficients</td>
<td>p Values</td>
</tr>
<tr>
<td>CC -&gt; BI</td>
<td>-0.021</td>
<td>0.844</td>
</tr>
<tr>
<td>CC -&gt; PE</td>
<td>-0.064</td>
<td>0.306</td>
</tr>
<tr>
<td>CESE -&gt; BI</td>
<td>0.537</td>
<td>0.000</td>
</tr>
<tr>
<td>CESE -&gt; EE</td>
<td>0.877</td>
<td>0.000</td>
</tr>
<tr>
<td>EE -&gt; BI</td>
<td>0.072</td>
<td>0.544</td>
</tr>
<tr>
<td>EE -&gt; PE</td>
<td>0.867</td>
<td>0.000</td>
</tr>
<tr>
<td>PE -&gt; BI</td>
<td>0.178</td>
<td>0.045</td>
</tr>
<tr>
<td>PTA -&gt; BI</td>
<td>0.115</td>
<td>0.364</td>
</tr>
<tr>
<td>SI -&gt; BI</td>
<td>0.103</td>
<td>0.362</td>
</tr>
<tr>
<td>SI -&gt; CESE</td>
<td>0.825</td>
<td>0.000</td>
</tr>
</tbody>
</table>

3) Education: Sub-group analysis was conducted for education and bachelor and above (high education) and diploma and below (low education) groups were analysed. Table VIII shows that out of six independent variables, two variables showed a significant relationship with BI for both groups. However, the nature of variables varied between groups – CESE and PE showed a significant positive relationship with BI for the high education group. In contrast, CESE and SI showed a significant positive relationship with BI for the low education group. Moreover, the strength of CESE with BI was much stronger for the low education group compared to the high education group (β = 0.442 vs 0.366, respectively). The variables CESE -> EE, EE -> PE and SI -> CESE showed a significant positive relationship for both groups. With respect to the R square, 82.2% variance in BI was explained by the model for the low education group, while this percentage was 60.5% for the high education group.

TABLE VIII. PATH COEFFICIENTS AND P VALUES OF HIGH AND LOW EDUCATION GROUPS

<table>
<thead>
<tr>
<th>Education groups</th>
<th>Bachelor’s and above</th>
<th>Diploma and below</th>
</tr>
</thead>
<tbody>
<tr>
<td>Education groups</td>
<td>Path coefficients</td>
<td>p Values</td>
</tr>
<tr>
<td>CC -&gt; BI</td>
<td>-0.010</td>
<td>0.933</td>
</tr>
<tr>
<td>CC -&gt; PE</td>
<td>-0.026</td>
<td>0.775</td>
</tr>
<tr>
<td>CESE -&gt; BI</td>
<td>0.366</td>
<td>0.001</td>
</tr>
<tr>
<td>CESE -&gt; EE</td>
<td>0.851</td>
<td>0.000</td>
</tr>
<tr>
<td>EE -&gt; BI</td>
<td>0.050</td>
<td>0.720</td>
</tr>
<tr>
<td>EE -&gt; PE</td>
<td>0.773</td>
<td>0.000</td>
</tr>
<tr>
<td>PE -&gt; BI</td>
<td>0.236</td>
<td>0.007</td>
</tr>
<tr>
<td>PTA -&gt; BI</td>
<td>0.110</td>
<td>0.372</td>
</tr>
<tr>
<td>SI -&gt; BI</td>
<td>0.139</td>
<td>0.231</td>
</tr>
<tr>
<td>SI -&gt; CESE</td>
<td>0.728</td>
<td>0.000</td>
</tr>
</tbody>
</table>

4) Occupation: Sub-group analysis was conducted for occupation and clinical and non-clinical groups were analysed. Table IX shows that the variables CESE and SI showed a significant positive relationship with BI for the clinical group, while CESE, PE and PTA showed a significant positive relationship with BI for the non-clinical group. Moreover, the strength of CESE with BI was stronger for the non-clinical group compared to the clinical group (β = 0.525 vs 0.424, respectively). Also, the non-clinical group a significant positive relationship between CC -> PE (Table VII). With respect to the R square, 84.5% variance in BI was explained by the model for the non-clinical group, while this percentage was 60.7% for the clinical group.

TABLE VII. PATH COEFFICIENTS AND P VALUES OF 18 TO 35 AND 36 TO 70 YEARS OLD PARTICIPANTS

<table>
<thead>
<tr>
<th>Age</th>
<th>18 to 35</th>
<th>36 to 70</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>Path coefficients</td>
<td>P Values</td>
</tr>
<tr>
<td>CC -&gt; BI</td>
<td>-0.020</td>
<td>0.856</td>
</tr>
<tr>
<td>CC -&gt; PE</td>
<td>0.076</td>
<td>0.423</td>
</tr>
<tr>
<td>CESE -&gt; BI</td>
<td>0.353</td>
<td>0.000</td>
</tr>
<tr>
<td>CESE -&gt; EE</td>
<td>0.826</td>
<td>0.000</td>
</tr>
<tr>
<td>EE -&gt; BI</td>
<td>0.048</td>
<td>0.708</td>
</tr>
<tr>
<td>EE -&gt; PE</td>
<td>0.760</td>
<td>0.000</td>
</tr>
<tr>
<td>PE -&gt; BI</td>
<td>0.162</td>
<td>0.043</td>
</tr>
<tr>
<td>PTA -&gt; BI</td>
<td>0.258</td>
<td>0.031</td>
</tr>
<tr>
<td>SI -&gt; BI</td>
<td>0.106</td>
<td>0.338</td>
</tr>
<tr>
<td>SI -&gt; CESE</td>
<td>0.751</td>
<td>0.000</td>
</tr>
</tbody>
</table>
TABLE IX. PATH COEFFICIENTS AND P VALUES OF CLINICAL AND NON-CLINICAL GROUPS

<table>
<thead>
<tr>
<th>Occupation</th>
<th>Clinical</th>
<th>Non-clinical</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>Path coefficients</td>
<td>p Values</td>
</tr>
<tr>
<td>CC -&gt; BI</td>
<td>0.077</td>
<td>0.356</td>
</tr>
<tr>
<td>CC -&gt; PE</td>
<td>-0.013</td>
<td>0.848</td>
</tr>
<tr>
<td>CESE -&gt; BI</td>
<td>0.424</td>
<td>0.000</td>
</tr>
<tr>
<td>CESE -&gt; EE</td>
<td>0.779</td>
<td>0.000</td>
</tr>
<tr>
<td>EE -&gt; BI</td>
<td>0.006</td>
<td>0.973</td>
</tr>
<tr>
<td>EE -&gt; PE</td>
<td>0.839</td>
<td>0.000</td>
</tr>
<tr>
<td>PE -&gt; BI</td>
<td>0.131</td>
<td>0.235</td>
</tr>
<tr>
<td>PTA -&gt; BI</td>
<td>-0.100</td>
<td>0.368</td>
</tr>
<tr>
<td>SI -&gt; BI</td>
<td>0.314</td>
<td>0.019</td>
</tr>
<tr>
<td>SI -&gt; CESE</td>
<td>0.764</td>
<td>0.000</td>
</tr>
</tbody>
</table>

V. MODEL DISCUSSION

This study model explained 71.2\% variance in BI, which demonstrates a good fit and shows that the majority of the factors predicting mIoT adoption were included in the model. Also, the variance explained by the current study model is very similar to the potential of the original UTAUT model [89]. In total, ten path relationships were tested in this study and results concluded six significant relationships (hypothesis supported), out of which three variables showed a significant direct relationship with the behavioural intention to adopt mIoT.

Three variables - CESE, PE and SI positively influenced the BI. Also, out of these three, CESE had the most potent effect (β = 0.437, p < 0.001), followed by SI (β = 0.175, p = 0.024) and PE (β = 0.148, p = 0.032). The current study did not show the significant effect of EE on BI. However, a very strong effect of CESE on EE (β = 0.833, p < 0.001) was observed in this research. This concludes that computer and English language competence is a major sub-determinant of the overall Effort Expectancy of mIoT use. There is very limited research available regarding IoT adoption among healthcare professionals; thus, the current study's findings were compared with IT adoption by healthcare professionals. The strong influence of CESE or self-efficacy is supported by [90,91,92]. These studies concluded that self-efficacy, including but not limited to computer skills, had a strong impact on the Perceived Ease of Use and subsequent indirect effect on the adoption of technologies.

The findings regarding the strongest influence of CESE (considering CESE as a major sub-part of EE) on BI are also well aligned with the previous research. Gagnon et al. [24] conducted a study with physicians and concluded that EE or Perceived Ease of Use had the strongest impact on EHR acceptance. Similarly, Chen & Hsiao [90], conducted a study with hospital staff and concluded similar results with respect to health information systems adoption. However, the Saudi studies conducted on nurses and physicians suggested that PU or PE was more important than PEOU or EE in determining professionals’ acceptance of EHR [93,54]. The reason for this discrepancy could be due to the nature of the technology. EHR systems are mainly operated by humans and require effort to enter data. However, mIoT majorly relies on wearable devices to upload and AI to analyse data without human involvement, which suggests the transformation of EE from the user point of view.

Similarly, another reason could be due to the unfamiliarity of the participants with the technology, as the hospitals are not using it on a full scale. Thus, they perceived computer and English skills as the most crucial component (as these are the most integral component of all previous technologies). This inference is supported by the previous research by Alsahafi et al. [94], where EE was the most significant predictor of behavioural intention to use National Electronic Health Records by the respondents who were actually non-users of the technology in KSA. This proposes a key finding of this study as the previous dominance (major predictor) of PE or even overall EE on IT adoption is not supported in this one of the early mIoT acceptance research in KSA. Hence, other reported factors of EE, such as time required for data entry, interruption in workflow, the influence of technology on the communication between professionals and patients and many others (El Mahalli [14]), are not important for the adoption of mIoT for this study cohort. This is an essential consideration for policymakers as understanding key adoption determinants can tailor the efforts in the right direction and even assist product developers about the consumers’ expectations.

The second significant determinant for the adoption of mIoT reported in this study was SI (β = 0.175, p = 0.024). However, SI also showed a strong effect (β = 0.797, p < 0.001) on the CESE (which showed the strongest effect); thus, it is just to conclude that the total effect of SI on the adoption would be more. This inference is supported by Tsourela & Nerantzaki [95], where SI was found to influence PEOU or EE and PU or PE, which influence attitude, and behavioural intention to adopt IoT technologies. SI is very important for the adoption of mIoT, especially in KSA [96]. The reasons for this are the nature of the technology (wearable devices, apps providing live data feed to the patients, improving control of patients over their health data, etc.) and Saudi society's nature. IoT products can be considered health and fashion products with aesthetic qualities, which could allow customers to articulate their characters and values [95]. This is supported by Yang et al. [97], where social image (SI) showed the strongest (b = 0.303, t-value = 4.66, p < 0.001) influence on the customers’ intention to use of wearable devices.

With respect to society, Saudi Arabia is considered a conservative and enclosed community and the citizens are substantially influenced by their native culture and therefore prefer face-to-face interaction to virtual one [98]. However, this has changed due to the ruler’s (Prince Muhammad bin Salman) progressive approach and the COVID-19 epidemic. There is a difference in the findings of the research conducted before and after COVID-19 in KSA and post-COVID-19 research shows that Saudi health consumers are happy with the transition toward e-health during the epidemic [99,100,101]. Thus, it is reasonable to conclude that the influence of Saudi society has inclined more towards e-health or IoT. Hence, SI is a significant predictor of mIoT adoption. This conclusion has substantial value for the IT managers and individuals.
responsible for the introduction of new technology in Saudi hospitals. Considering the role and value of SI, they can develop strategic plans highlighting the links between the role of mIoT and SI in the hospital to support the smooth transition towards mIoT.

Performance expectancy showed the lowest influence on BI ($\beta = 0.148, p = 0.032$), however, a very strong effect of EE on PE ($\beta = 0.858, p < 0.001$) was reported in this study, which the CESE also influenced (CESE $\rightarrow$ EE, $\beta = 0.833, p < 0.001$). This relationship supports the TAM factors developed by Davis as it shows the strong impact of EE or PEOU on PE or PU. This is also supported by the previous Saudi research on EMR and EHR adoption with healthcare professionals [93,92]. However, the strength of the relationship between PE and BI in the current study is comparatively weaker than the previous studies conducted in Saudi Arabia [54,93]. The reason for this discrepancy could be due to the composition of this study cohort (comparatively fewer doctors and more non-clinical participants) or could be the nature of the technology. As mIoT is different from EMR or EHR, it is quite probable that participants cared more about the computer and English skills directly influencing EE rather than the performance concerning adoption. This is supported by Chen & Hsiao [90], where PU has substantially less impact than the PEOU on health information system acceptance by healthcare professionals.

A. Gender and Study Model

Sub-group analysis of the model showed differences in the relationships due to gender, which is supported by the original UTAUT model [18]. Out of six independent variables for the BI, two variables showed a significant relationship for male (CESE and PE) and only one showed a significant relationship for female (CESE) participants. Past research on gender and technology suggests that males incline more toward task accomplishment than females; therefore, as an illustration, PE tends to be more significant for males, whereas females are more concerned with effort in adopting new technology [18]. Also, the strength of the relationship between CESE and BI and PE and BI were stronger among male participants compared to female and overall results. The study by Hoque [57] found similar results where male participants showed stronger relationships than females ($0.6556$ versus $0.1445, t = 3.784$) between PEOU and m-Health adoption.

This indicates that EE, as suggested by other researchers, is a substantial factor for the adoption of mIoT among women. However, for men, both EE and PE were important. This is also supported by Tubaishat [92], where male Jordanian nurses were found to have a 0.19 higher PU of EHR than females, controlling for other variables in the model. Similar findings regarding the stronger link between PU and intention to use mHealth among men compared to women were also shown by a Western study suggesting a universal moderating trend of gender on technology adoption [102].

B. Age and Study Model

Sub-group analysis of the model was conducted for age and a separate analysis for participants aged 18 to 35 years (younger) and 36 to 70 years (senior) was conducted. The study findings showed that out of six independent variables, three (CESE, PE and PTA) showed a significant positive relationship for the younger group, and two (CESE and SI) showed a significant positive relationship for the senior group with BI. Moreover, the strength of CESE with BI was much stronger for the senior group compared to the younger ($\beta = 0.536$ vs 0.353, respectively). This stronger relationship for the older group is well supported by Venkatesh and Morris [103], who suggested that EE or CESE is more salient for older participants for IT adoption. The weaker computer skills of the middle-aged and older participants expressed through computer anxiety and less computer self-efficacy [104,105] could be the main reason for this stronger relationship. It is to assume that the participants will put more emphasis (hence a stronger relationship) on the determinants that retain the most significance for them.

PE did not show a significant relationship with BI for the senior participants, and it could be due to low mIoT knowledge among them or due to lower expectations from mIoT performance. As Al Otaiby et al. [69] showed, healthcare participants above 50 were more satisfied with EMR performance in KSA than their blow 35 years counterparts. SI only showed a significant relationship with BI for the senior participants. The reason for this could be the rapid cultural transformation occurring in KSA, which has influenced the youth, but the elderly are still intact with their culture and rely on each other, and value the opinion of the people when it comes to technology adoption [94]. The influence of aging and the role of social influence on the adoption of new technologies is also evident in the Western culture, where elderly participants expressed feelings of inadequacy in comparison to younger generations and lack of social interaction as a major barrier to the use new IT technologies [106].

Apart from these, PTA was another major factor that showed a moderate positive relationship with BI ($\beta = 0.258, p = 0.031$) for the younger group. In contrast, the other group showed a non-significant negative relationship between PTA and BI. This is a significant finding of our study. In contrast to past research [48, 51] PTA is not a barrier to mIoT adoption for younger participants, but it might still be for the middle-aged and elderly. This discrepancy could be explained by the power difference dynamics in the hospital environment. Younger staff in KSA, especially nurses, usually hold working or sub-managerial positions in the hospitals and they are often mismanaged or scolded by senior staff or their managers [107]. Increasing control of the top management and reducing the privacy of the behaviour conducted by the senior staff through mIoT might give a sense of transparency and reassurance to the junior employees. Hence, they perceived PTA as a significant positive determinant for adoption, which can assist them in reporting unseen misbehaviour to the top management [108].

On the other hand, senior staff might believe that mIoT can expose their professional misconduct and or can reveal their professional in-competencies and hence perceive PTA as a barrier. Also, the senior group showed a significant negative relationship between CC $\rightarrow$ PE, which aligns with the previous research related to security concerns [13,10]. Moreover, the lack of significant effect of PE on BI for the senior group might also be explained by this finding as CC did not show any significant relationship for the overall or any other sub-group analysis.
C. Education and Study Model

The sub-group analysis of education showed that out of six independent variables, two variables showed a significant relationship with BI for both groups. However, the nature of variables varied between groups – CESE and PE showed a significant positive relationship with BI for the high education group (bachelor and above). In contrast, CESE and SI showed a significant positive relationship with BI for the low education group (diploma and below). This discrepancy could be due to the difference in the role of professionals in Saudi hospitals based on their education. It is most likely that participants with higher education were working in managerial or assistant managerial positions; as noted by Aboshatijaq & Alharbi [109] 72% of nursing managers had master’s qualifications. Managers have a specific focus on performance and that is why the high-education group showed a significant effect of PE for BI [110]. Moreover, the strength of CESE with BI was much stronger for the low-education group than the high-education group (β = 0.442 vs 0.366, respectively). Again, it could be due to the varied computer skills among our study participants (high education better skills and vice versa), which is also supported by past research [111,45].

D. Occupation and Study Model

Sub-group analysis for occupation showed that out of six independent variables, two variables (CESE and SI) showed a significant relationship with BI for the clinical group, and three (CESE, PE and PTA) showed for the non-clinical group. This difference is well explained by the nature of the job of the clinical professionals [112]. Compared to other professionals in the hospitals doctors, nurses and other clinical staff are directly connected with the patients and have more potential for interaction with their colleagues; thus, SI holds substantial significance for them. In contrast, the non-clinical staff, which includes the administrators and IT personnel, have the duty of care for performance [74,110] and hence, they prioritised PE.

Moreover, the strength of CESE with BI was stronger for the non-clinical group than the clinical group (β = 0.525 vs 0.424, respectively). This could be due to the better understanding of the non-clinical participants about the computer skills required for mIoT use. Also, the non-clinical group showed a significant positive relationship between PTA and BI (β = 0.340, p = 0.017), while the clinical group showed a non-significant negative relationship (β = -0.100, p = 0.368). Again, this is a noteworthy finding of our study as it shows that the non-clinical group, which included administrators and IT personnel perceived PTA as a significant facilitator for BI as it can increase their control and support transparency in work done mainly by clinical professionals [112,108]. While as previously supported and hypothesised by the current study, PTA is a negative determinant of BI for the clinical participants, which can compromise their professional freedom and autonomy in the hospital [48].

VI. Study Contributions

This study makes a substantial contribution to the subject of mIoT adoption by hospital care staff in KSA. The findings and inference concluded in this research hold critical importance for the Saudi Ministry of Health as it is the first-ever quantitative study conducted on mIoT in Saudi hospitals and it aligns with the inclusion of the latest technology in healthcare, which is an integral part of the Saudi 2030 vision. Apart from that, the study findings are important for hospital administrators, IT managers, mIoT developers, vendors, and researchers. The following sub-sections elaborate the significance for each stakeholder.

It is strongly advised to the mIoT developers and vendors to incorporate an option of multiple languages in the mIoT systems. The current study participants showed a very strong relationship between CESE and mIoT adoption. Thus, this factor should be addressed and be prioritised.

It is recommended that the hospital administrators tailor the mIoT training program according to the demographic needs of the staff. The current study showed a disparity between the hospital staff regarding mIoT adoption as different demographic groups have shown different priorities and interests towards mIoT adoption determinants. To support equality and equity and to avoid conflict of interest among hospital staff, this study suggests the following.

It is advised to give special focus to the female, middle-aged hospital staff with low education, such as diploma and below, and clinical professionals in the hospital during awareness and training program sessions. It is likely that these identified groups might require some extra assistance to understand the concepts related to mIoT. It is also possible that they might not ask for assistance by themselves due to peer pressure and perceived shame. Thus, it is possible, two versions (beginner and intermediate) of mIoT awareness and training program sessions should be introduced in the hospital, so the participants can choose according to their self-perceived abilities.

Based on demographic differences, the study participants showed varying preferences and expectations toward mIoT. For instance, young and non-clinical participants have shown PTA as a significant facilitator for the adoption, while older and clinical participants have indicated otherwise. Thus, this element should be considered carefully and moderated opinion should be promoted regarding PTA to accommodate both groups. Although PTA is a relatively new determinant for the adoption of e-health, it holds substantial importance for mIoT due to the nature of the technology. Therefore, hospital administrators have to be very careful about their presentation and explanation to the staff.

VII. Limitations and Research Implications

This research provides a substantial basis to future mIoT studies in the Kingdom of Saudi Arabia and elsewhere in the Gulf region. The study incorporated a modified UTAUT model and named it UTAUT-HS. The model showed significant variance (above 70%), which indicated that it was a good fit to explain the mIoT adoption behaviour. In addition, the model also showed some interesting findings, which as per the researchers' knowledge, have yet to be observed in any mIoT studies.

These findings include the most decisive role of CESE in adopting mIoT. Though CESE can be considered a subset of EE, based on the current study findings, it has narrowed the scope of EE to CESE, which can help future researchers tailor
their research direction toward more targeted needs. However, the current study must assert that the UTAUT-HS model should be tested with a much larger sample (including many doctors from different specialisations and hospitals) as the current study speculate some other EE concerns (which are not shown in the current study) from the physicians. Furthermore, PTA is another factor that is initially not included in any of the original past technology adoption models (TRA, TPB, TAM, UTAUT, etc.). Now, this factor is substantially crucial with respect to mIoT and again, it should be tested with a large sample, including a sufficient number of doctors.

Also, it is suggested that more direct questions (e.g., 'mIoT including AI can make me redundant etc.) should be included in PTA, which may overlap with the job security factor or perhaps another construct of job security can be included in the model to explore the severity of this matter. Apart from the current research, some other recent qualitative studies have highlighted these concerns. Another noteworthy finding concerning the UTAUT-HS model was the moderators' significant influence, including age, gender, education, and occupation. The current study found a wide range of significant differences among different demographic groups. While most of the differences were related to the degree (in one direction) to which a factor was known or perceived (e.g., males knew more about mIoT than females), some were in the opposite direction. PTA positively influenced BI for the younger participants, while it was non-significantly negative for the older counterparts.

Unexplored and unreported contrasting findings concerning a comparatively sensitive construct such as PTA can have grave consequences for the stakeholders responsible for introducing mIoT in the Kingdom. Thus, it is suggested that future studies include all potential demographics as moderators in the model to identify these conflicting perceptions. However, as asserted before, a much larger sample size would be required to execute these suggestions. Last, but not least, the UTAUT-HS may have universal relevance concerning mIoT adoption. Thus, researchers conducting studies in healthcare outside the Kingdom, particularly in the Western world, are suggested to employ UTAUT-HS to test its relevance.

VIII. CONCLUSION

The study concludes that CESE, PE, PTA and SI are the significant determinants that can influence the adoption of mIoT among the hospital care staff in the Kingdom of Saudi Arabia. The model (UTAUT-HS) included in the study showed a 71.2% variance in BI, which demonstrated a good fit and showed that the majority of the factors predicting mIoT adoption were included in the model. Among all determinants, CESE demonstrated the most substantial effect suggesting that computer and English language competence is a significant sub-determinant of the overall Effort Expectancy for mIoT adoption and should be prioritised during the development and introduction of these technologies. However, the element of PTA cannot be disregarded as it provided critical insights related to occupation and power dynamics in the hospitals and respective attitudes towards mIoT adoption. Future large-scale studies are recommended in KSA and elsewhere to validate the relevance of UTAUT-HS for mIoT adoption in the healthcare sector.

REFERENCES


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Apache Spark in Riot Games: A Case Study on Data Processing and Analytics

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Abstract—This case study examines Riot Games’ use of Apache Spark and its effects on data processing and analytics. Riot Games is a well-known game production studio. The developer Riot Games, best known for the well-liked online multiplayer game League of Legends, manages enormous volumes of data produced daily by millions of players. Riot Games handled and analyzed this data quickly using Apache Spark, a distributed computing technology that made insightful findings and improved user experiences. This case study explores Riot Games’ difficulties, the company's adoption of Apache Spark, its implementation, and the advantages of utilizing Spark's capabilities. We evaluated the drawbacks and advantages of adopting Spark in the gaming sector and offered suggestions for game creators wishing to embrace Spark for their data processing and real-time analytics requirements. Our study adds to the increasing body of knowledge on the use of Spark in the gaming sector and offers suggestions and insights for both game producers and researchers.

Keywords—Riot games; Apache Spark; data processing; real-time analytics; distributed computing technology

I. INTRODUCTION

Riot Games, a prominent player in the online gaming industry since its inception in 2006, has achieved unprecedented success with its flagship game, League of Legends, which has garnered a staggering 100 million active players worldwide. The company faces the daunting task of managing and extracting insights from the massive volume of data generated by the game, which includes player behavior, gameplay patterns, and performance metrics.

To address this challenge, Riot Games has adopted Apache Spark, a cutting-edge distributed computing framework that efficiently processes large-scale data. The study [1] outlines the advantages of Apache Spark, such as its ability to handle both batch and streaming data and its ability to seamlessly integrate with other big data tools. By utilizing Apache Spark, Riot Games can effectively process and analyze the vast amounts of data generated by League of Legends to derive insights that can be used to improve the game and enhance the player experience continuously.

Moreover, Apache Spark's in-memory processing capabilities allow for faster data processing and analytics, making it a suitable choice for Riot Games. In their research, [2] discusses the benefits of in-memory computing for data-intensive applications, such as online gaming. By leveraging Apache Spark's in-memory processing capabilities, Riot Games can process data more quickly and efficiently, allowing real-time analysis of player behavior and gameplay patterns.

In summary, Riot Games has harnessed the power of Apache Spark to effectively process and analyze the vast amounts of data generated by League of Legends. By doing so, the company can continuously improve the game and provide a better experience for its players. In this study, we propose to explore the use of Apache Spark in the gaming industry through a case study analysis of Riot Games.

The remaining sections of this study are organized as follows. In Section II, a detailed description about role of Hadoop framework in processing Riot games is discussed. Literature review is presented in Section III. Section IV describes various aspects of Riot game and its processing with big data. The suggested framework is provided in Section V. The conclusion of the paper is provided in Section VI.

II. APACHE SPARK IN RIOT GAMES

A. Apache Spark

Apache Spark is an open-source distributed computing framework that offers a fast and efficient method for processing large volumes of data. Developed by the Apache Software Foundation, Apache Spark has emerged as a popular alternative to Hadoop Map Reduce for big data processing, providing a more versatile and flexible interface. In their research paper, [1] outlines the advantages of Apache Spark, such as its ability to handle both batch and streaming data and integrate with other big data tools seamlessly. Apache Spark's scalability and fault tolerance capabilities make it ideal for processing large volumes of data, even during hardware failures or network disruptions. Additionally, its easy-to-use interface enables users to process data in various programming languages, including Java, Scala, and Python. In [3] discuss the advantages of using Apache Spark for large-scale data processing, such as its ability to perform iterative algorithms efficiently and its support for in-memory computing.

B. Hadoop

Apache Hadoop is an open-source distributed computing platform that enables the processing large-scale data sets using thousands of independent machines and large amounts of data.
The platform includes a framework that offers tools for distributed processing, such as Hadoop Distributed File System (HDFS) for distributed storage and Apache MapReduce for distributed processing. Hadoop was initially based on Google MapReduce and the Google File System (GFS). Hadoop's distributed computing capabilities have made it popular for big data applications, such as social media analysis, predictive modeling, and machine learning. Its open-source nature and a large community of contributors have also contributed to its popularity. In their research, [4] describes HDFS's architecture and design, including its scalability and fault tolerance capabilities. The study [5] provides an overview of Hadoop's features and advantages for big data analytics, its limitations, and potential future developments.

C. Architecture of Riot Games

The architecture of Riot Games can be broadly divided into four main components, as described in [6]:

Game Servers: The game servers manage the game logic, handle player inputs, and update the game state. Riot Games uses a distributed server architecture, where multiple servers work together to run game sessions. This architecture allows for better scalability and fault tolerance, as the load can be distributed across various servers. If one server fails, the game session can be migrated to another server [7]. API Platform Architecture and High-level Architecture are shown in Fig. 2, and Fig. 3, respectively.

Matchmaking: Riot Games' matchmaking system pairs players together for games. The matchmaking system uses a complex algorithm to match players based on various criteria, such as skill level, playing history, and other factors. This ensures players are matched with opponents of similar skill levels, leading to more enjoyable gameplay experiences. As outlined in a [8], Riot Games' matchmaking system is continuously being updated and refined to provide the best possible experience for players.

Data Storage: Riot Games uses a variety of data storage solutions to manage the vast amounts of data generated by their games. This includes databases for player accounts and game data and distributed file systems for storing game assets like textures and sound files. As described in [9], Hadoop Distributed File System (HDFS) is used for processing and analyzing large volumes of data.

Game Clients: The game clients are the software that runs on players' computers or devices, allowing them to interact with the game servers. Riot Games' game clients are built using a combination of C++, Lua, and other technologies and are designed to be highly responsive and optimized for low-latency gameplay. As outlined in [10] developed a network optimization system called "Riot Direct" to improve network performance and reduce latency for players. Overall, Riot Games' architecture is designed to provide a scalable and fault-tolerant platform for their online games, ensuring that players enjoy seamless gaming experiences.

D. Apache Spark in Riot Games

Riot Games has effectively leveraged Apache Spark, an open-source distributed computing framework, for various data analysis tasks supporting their online game services. According to [11], Riot Games has utilized Apache Spark to analyze large volumes of data from their online game, League of Legends, to extract insights on player behavior, gameplay patterns, and performance metrics. The real-time processing capabilities of Apache Spark have enabled Riot Games to analyze player behavior and game performance in real time, allowing them to quickly identify and address issues that may impact the game experience. Furthermore, [12] shows that Riot Games has utilized Apache Spark to build machine-learning models that can improve game performance and player experience.

In addition, Riot Games has used the stream processing capabilities of Apache Spark to analyze real-time game data streams, helping the company quickly detect and respond to issues such as server downtime and player disconnects. This was demonstrated in [13], where Apache Spark was used by Riot.
Fig. 2. Riot game data processing and analysis architecture.

Fig. 3. API platform architecture [14].
Games detect and diagnose network issues in real time, enabling them to respond and mitigate the issues quickly. Overall, Apache Spark has been a vital tool for Riot Games in analyzing and processing the massive volumes of data generated by their online games, enabling them to improve the gaming experience for players while maintaining the stability and reliability of their online game services. How Riot Games Reworked Its Software Infrastructure Using Open Source: Even under the best conditions, developing a software architecture supporting hypergrowth is challenging. Riot Games faced several obstacles last year, including the need for the company to switch from a traditional SQL database to Hadoop and to create better real-time monitoring tools.

Graphic Settings and Client-Side Logic are shown in Fig. 4(a) and Fig. 4(b), respectively [17], [18]. The company is behind the hugely well-liked online game League of Legends and has offices in Santa Monica and St. Louis. The game, released four years ago, currently boasts 32 million active players who play for over a billion hours each month. However, the game's craze was giving their software team fits. Barry Livingston, Director of Engineering at Riot Games, said, "It took us too long to gain insights into our software performance, and our database servers were sluggish." "You wouldn't think a five-year-old company would have legacy software issues, but we did because we have proliferated," he said in a July presentation at the St. Louis Stampede conference. For its data warehouse, Riot Games started with a monolithic SQL platform. It necessitated numerous manual, custom-coded procedures. Most of the reporting was done in Excel, and queries were developed in MySQL. Game screen is shown in Fig. 5 whereas, Fig. 6 shows the working of Database.
Riot Games’ senior employees concluded that they needed new software data architecture last year since they had reached a critical point. "The time to arrive at insights was taking too long, and ours. The solution required too many dev team members to make changes to our data schemas or other updates," the developer claimed. They also had two more objectives they sought to achieve. The first goal was to democratise data access by making player statistics and gameplay analysis available to a larger group of our workers. For our employees to more easily do updates and address issues, we also needed to produce meaningful insights about the game’s underlying software components.
It was necessary to integrate Hadoop, a cloud-based data warehouse, and an end-to-end automated software development pipeline to improve the software infrastructure. The Hadoop transformation involved several add-on programmers and tools for different purposes:

- **Honu**: A pipeline for collecting streaming logs and processing events.
- **Analysis and visualization of BI on Platform Workflow job scheduler Oozie**.
- **Data warehouse and queries using Hive**.
- **Chef**: Code distribution and configuration administration.
- **Version control and Programme tracking of GitHub**.
- **Build system management with Jenkins Service discovery process, eureka**.

### III. LITERATURE REVIEW

Apache Spark is a distributed computing framework that has gained significant popularity in big data processing and analytics. It is known for its speed, scalability, and flexibility and has been widely adopted by companies across various industries. In particular, the gaming industry has been a significant adopter of Apache Spark due to its ability to handle large volumes of gaming data generated by millions of players worldwide.

Riot Games, the developer of the popular game League of Legends, is one such company that has adopted Apache Spark for its data processing and analytics needs. Riot Games has a massive player base generating vast amounts of data related to gameplay, user behavior, and other metrics. The ability to process and analyze this data in real time is essential to maintaining a high-quality gaming experience for players. Fig. 7 shows Riot with Spark & Hive [9].

In a case study [15], Riot Games discussed its implementation of Apache Spark for data processing and analytics. Riot Games used Spark to handle real-time data processing, providing insights into game server performance and player behavior. The company also used Spark’s machine learning capabilities to develop predictive models that improved the gaming experience for players. Other studies have also demonstrated the effectiveness of Apache Spark in gaming data processing and analytics. For example, [16] used Spark to process and analyze massive amounts of gaming data generated by online games. Their study found that Spark could effectively identify user behaviors, preferences, and patterns, allowing game developers to improve game design and enhance the player experience.

Similarly, [17] used Spark to analyze user behavior data in mobile games. Their study demonstrated that Spark could be used to identify patterns in user behavior, helping game developers optimize game design and improve user engagement. The literature suggests that Apache Spark is a valuable tool for gaming data processing and analytics. Spark in Riot Games’ data processing and analytics pipeline has enabled the company to handle large volumes of data in real time, identify issues quickly, and develop predictive models that enhance the player experience. Spark in the gaming industry has also been shown to be effective in identifying patterns in user behavior, optimizing game design, and improving user engagement.

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![Fig. 7. Riot with spark & hive [9].](image-url)
IV. OBSERVATION

This research paper explores the use of Apache Spark in the gaming industry, focusing on Riot Games. By analyzing a case study and reviewing the literature on the topic, we found that Spark is a powerful tool for handling extensive data generated by games and improving the player experience. Our analysis revealed that Spark's real-time data processing capabilities enable quick issue identification and predictive modeling, leading to more accurate decision-making and better user engagement. Furthermore, our literature review showed that Spark had been widely adopted in the gaming industry for data processing and analytics, with numerous studies demonstrating its effectiveness in identifying user behavior patterns and preferences, optimizing game design, and improving user engagement. Overall, our research provides valuable insights into the benefits of Apache Spark in the gaming industry, demonstrating its potential for data processing and analytics. The findings of this study can help game developers adopt Spark for their data processing and analytics needs, leading to a better user experience and improved business outcomes [19-21].

1) Problem definition: Riot Games encountered several data-related challenges, including: Massive data volume: The game generates a staggering amount of data, including player interactions, in-game events, and performance metrics, resulting in petabytes of data.

- Real-time processing: Riot Games required the ability to process and analyze data in real-time to derive actionable insights promptly.
- Scalability: Traditional data processing solutions struggled to handle the scale and complexity of Riot Games' data, leading to performance bottlenecks and increased processing times.
- Diverse data formats: The data generated by the game encompassed structured, semi-structured, and unstructured formats, making it challenging to handle with conventional tools.

2) PC variability: Hardware and OS profiles are significantly different even with regions:

- OS and patch models
- CPU
- Memory
- Video Card (GPU)
- Video Card Memory
- Drivers

3) Client-side logic: The client-side logic of Riot Games' games is complex and varies depending on the specific game. However, the author can provide some general information about how client-side logic works in online multiplayer games like those developed by Riot Games.

At a high level, the client-side logic of online multiplayer games is responsible for managing the game state on the player's computer or device. This includes the player's position in the game world, health and mana levels, inventory and equipment, and any other relevant data. The client-side logic communicates with the game server, which is responsible for managing the game state for all players. The server sends updates to the client about the state of the game, such as the position of other players, the location of items, and the outcome of actions like attacks or spells.

4) Game load screen: Here are some potential ways that Riot Games could improve game load times:

- Optimize game assets: One way to improve load times is to optimize the game's assets, such as textures and models. By reducing the size of these assets without sacrificing quality, the game can load faster.
- Implement asynchronous loading: Asynchronous loading allows the game to load resources in the background while the player does something else. This can help reduce load times by allowing the player to start playing sooner.
- Use compression: Compressing game files can reduce their size, leading to faster load times. Riot Games could explore using compression algorithms like LZMA or Zlib to reduce the size of game assets.
- Prioritize loading: Riot Games could prioritize loading critical assets required for gameplay, such as maps and character models. By loading these assets first, the player can start playing sooner.
- Improve network performance: Load times can also be affected by the quality of the player's internet connection. Riot Games could work to improve network performance by optimizing their netcode and using content delivery networks (CDNs) to reduce latency.

V. SUGGESTED FRAMEWORK

In this study, we propose to explore the use of Apache Spark in the gaming industry through a case study analysis of Riot Games. Our proposed work will involve the following steps:

1) Conducting a comprehensive literature review to identify the existing research on using Apache Spark in the gaming industry. This will involve reviewing academic papers, industry reports, and other relevant sources.

2) Collecting and analyzing data on Riot Games' use of Apache Spark for data processing and analytics. This will involve interviewing key stakeholders, reviewing internal documentation, and analyzing gaming data.

3) Identifying the key challenges and successes of using Apache Spark in the gaming industry. This will involve analyzing the impact of Spark on key performance indicators such as user engagement, revenue, and game design.
4) Providing recommendations for game developers looking to adopt Apache Spark for their data processing and analytics needs. This will involve identifying best practices and potential use cases for Spark in the gaming industry.

5) Evaluating the potential for Apache Spark to be used in other areas of the gaming industry beyond data processing and analytics, such as fraud detection and customer support.

Overall, our proposed work will provide valuable insights into the use of Apache Spark in the gaming industry, focusing on Riot Games. By conducting a case study analysis and literature review, we aim to provide practical guidance and recommendations for game developers looking to adopt Spark for their data processing and analytics needs. The findings of this study can help drive better decision-making and improved business outcomes in the gaming industry.

Method: Riot Games turned to Apache Spark, a robust distributed computing framework designed for big data processing and analytics, to address these challenges. Spark offers the following advantages:

- Speed and scalability: Spark’s in-memory processing and distributed computing capabilities enabled Riot Games to handle large datasets with significant speed improvements over traditional batch processing systems.
- Real-time analytics: Spark Streaming, a component of Apache Spark, facilitated real-time data processing, enabling Riot Games to monitor and respond to in-game events and player interactions in near real-time.
- Fault tolerance: Spark’s resilient distributed datasets (RDDs) ensured fault tolerance, enabling continuous processing and reducing the risk of data loss.
- Versatile data processing: Spark’s support for various data formats, including structured (Spark SQL), semi-structured (Spark DataFrames), and unstructured (Spark Streaming), made it an ideal choice for handling Riot Games’ diverse data requirements.
- Rich ecosystem: Spark’s extensive library ecosystem, including machine learning (MLlib) and graph processing (GraphX), offered additional capabilities for advanced analytics and insights.
- Experimental work: Riot Games adopted a multi-phased approach to implementing Apache Spark, focusing on the following key areas:
  - Data ingestion: Data from various sources, including game servers, user interactions, and telemetry, were ingested into a data lake using Apache Kafka and Apache Flume.
  - Data processing pipeline: Apache Spark was integrated into the existing pipeline, leveraging its ability to handle batch and real-time streaming data processing.
  - Data analytics: Spark SQL and Spark Data Frames were utilized to perform ad-hoc queries, generate reports, and extract valuable insights from the processed data.
  - Machine learning: Spark’s MLlib library was employed to develop and deploy machine learning models for player behavior analysis, fraud detection, and more.

VI. CONCLUSION

Our research paper examined the use of Apache Spark in the gaming industry, with a case study analysis of Riot Games. Our study showed that Spark could provide significant benefits to game developers regarding data processing and analytics, enabling them to gain valuable insights into user behavior and game performance. We identified the challenges and successes of using Spark in the gaming industry and provided recommendations for game developers looking to adopt Spark for their data processing and analytics needs. Our research contributes to the growing body of research on using Spark in the gaming industry, providing practical guidance and insights for game developers and researchers alike. With its well-liked games and burgeoning e-sports industry, Riot Games, a well-known video game developer, has a bright future. Providing speed and scalability for real-time analytics and machine learning applications, Apache Spark, a distributed data processing engine, is poised to play a significant role in the big data landscape.

REFERENCES


Deep Learning-based Method for Enhancing the Detection of Arabic Authorship Attribution using Acoustic and Textual-based Features

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Abstract—Authorship attribution (AA) is defined as the identification of the original author of an unseen text. It is found that the style of the author’s writing can change from one topic to another, but the author’s habits are still the same in different texts. The authorship attribution has been extensively studied for texts written in different languages such as English. However, few studies investigated the Arabic authorship attribution (AAA) due to the special challenges faced with the Arabic scripts. Additionally, there is a need to identify the authors of texts extracted from live stream broadcasting and the recorded speeches to protect the intellectual property of these authors. This paper aims to enhance the detection of Arabic authorship attribution by extracting different features and fusing the outputs of two deep learning models. The dataset used in this study was collected from the weekly live stream and recorded Arabic sermons that are available publicly on the official website of Al-Haramain in Saudi Arabia. The acoustic, textual and stylometric features were extracted for five authors. Then, the data were pre-processed and fed into the deep learning-based models (CNN architecture and its pre-trained ResNet34). After that the hard and soft voting ensemble methods were applied for combining the outputs of the applied models and improve the overall performance. The experimental results showed that the use of CNN with textual data obtained an acceptable performance using all evaluation metrics. Then, the performance of ResNet34 model with acoustic features outperformed the other models and obtained the accuracy of 90.34%. Finally, the results showed that the soft voting ensemble method enhanced the performance of AAA and outperformed the other method in terms of accuracy and precision, which obtained 93.19% and 0.9311 respectively.

Keywords—Authorship attribution; acoustic features; fusion approach; deep learning; CNN; ResNet34

I. INTRODUCTION

As Authorship attribution (AA) refers to the process of identifying the authorship of a document that has not yet been seen, given a list of potential writers and a list of documents that have already been described and tagged by each potential author [1, 2]. AA can be seen as a classification problem, when the training is done on a given set of training texts of known authors (labels). Using a collection of characteristics taken from the text in the training set, we can identify the authors for the testing set. The main characteristics employed during the classifier's training are the style, sentiment, and subject [3]. The authors' individual writing styles are examined in order to extract the stylometry features. Although the writing style of the author might vary depending on the topic, certain persistent and unchecked habits, but the writing styles of the authors cannot be too different over the time [1]. Previously, the authors’ attribution was conducted by hand to recognize the authors of unseen texts, but with the huge amount of the texts available online, it is hard to perform this task manually. Several studies addressed this issue in the literature such as [4].

Different statistical and machine learning-based techniques were recently applied on AA [4]. These techniques included Naive Bayes [5, 6], Support Vector Machine (SVM) [7–12], Bayesian classifiers [13], k-nearest neighbor [14, 15], and decision trees [16]. The authorship attribution for texts written in English, Spanish and Chinese has been studied well in the literature; however, less attention was given to the texts written in Arabic because of the complexity of Arabic scripts [17].

To address the Arabic authorship attribution (AAA) issues, several machine and deep learning methods have been applied. For instance, Al-Sarem et al. [17] applied the ensemble machine learning methods with multi-attribute decision making method (TOPSIS) that identifies the base classifier on two Arabic enquires (Fatwa) datasets. The findings showed that AdaBoost and Bagging methods achieved the highest accuracy for the two used datasets. Similarly, Al-Sarem, Alsaeedi, and Saeed [18] applied deep learning-based methods for AAA. The findings showed that the performance of deep learning method outperformed the state-of-art methods.

A recent study by Alqahtani and Dohler [19] reviewed the authorship attribution for Arabic texts. They found that the findings of AAA tasks vary based on the used dataset and features. Also, it was found that few challenges were faced when dealing with pre-processing of Arabic texts because of the scripts’ concatenative morphology. For Arabic scripts, all the datasets presented in the previous studies were in texts format and the majority of the currently published studies rely on machine learning techniques. Few studies have previously examined the effectiveness of deep learning for AAA tasks. Therefore, the purpose of this study is to address this gap by
examining how deep learning techniques can be used for enhancing Arabic authorship attribution detection. In addition, to the best of our knowledge, we found that no previous studies investigated the authorship attribution of texts extracted from livestream broadcasting and the recorded Arabic speeches. Therefore, the main contributions of this paper are:

- Scraping data from livestream weekly livestream Arabic sermons as well as the recorded sermons and available publicly on the official website of Al-Haramain in Saudi Arabia (https://www.alharamain.gov.sa/ last access: May 8, 2023).
- Extracting acoustic features and stylometric features directly from the livestream broadcasting and the recorded mp3 files.
- Combining the extracted features using different fusion approaches and Applying different deep learning models for AAA.
- Conducting a rigorous analysis on the applied DL models in terms of different performance evaluation metrics.

This paper is organized as follows: Section II gives an overview of the state-of-the-art methods for detecting authorship attribution. The description of the proposed model's architecture can be found in Section III. Section IV describes the research methodology, including the dataset, preprocessing methods, evaluation metrics, experimental design, and evaluation process. The experimental results were presented in Section V. In Section VI, we summarize the contributions and conclude the paper.

II. RELATED STUDIES

Several studies addressed the authorship attribution for Arabic language. For instance, Jambi et al. [20] investigated the feasibility of predicting authorship in Arabic short-microblog content using modern classifiers. To forecast the accuracy of the chosen classifiers, they used three frequently language features—character, lexical, and syntactic—in an incremental approach. Another developing area of machine learning is deep learning, which performs better than classical machine learning in several areas [21] and does not need the use of feature engineering. Although authorship attribution has been a topic of discussion in different languages such as English [22], German [23], Spanish [24], and Chinese [25], only a few studies have specifically addressed the authorship attribution in the context of Arabic [16].

Different deep learning models have been studied in the literature, including the Deep Belief Network (DBN), Convolutional Neural Network (CNN), Recurrent Neural Network (RNN), and Long Short-Term Memory (LSTM). Deep neural networks have recently been used to build authorship attribution systems [26–29]. Usually, input for neural networks takes the form of a list of words or a string of letters. Most techniques focus on lexical aspects, even though lexical-based language models are not scalable when dealing with authorship covering a range of themes [30, 31]. The syntactic features are based on content, more robust, and effective against topic volatility. Sari et al. [26] utilized continuous representation via a neural network together with a classification layer to determine the authorship. In [32], the IMDB62 dataset was used and 94.8% accuracy rate was achieved. According to [30, 33], CNN model was used for authorship identification. They tested their approach using a variety of text data, including emails, reviews, blogs, and tweets, and they were able to get outstanding accuracy scores between 85.0% and 95.0%. The authorship-attrribution problem of brief postings was addressed by Shrestha et al. [28] using CNN based on a string of character n-grams. They assessed their methodology using the dataset shown in [34].

To address the issue of authorship attribution, Hitschler et al. [35] used a single author from an anthology reference corpus [36] using POS tags and CNN approach. To achieve proper results from a corpus of scientific papers, they replaced unusual terms with their POS tags in order to create a better generalization. The researchers in [37] used a number of embedding structures based on character, word, n-gram, and POS tags in a CNN model to accurately identify the writing styles used by authors of tweets and posts on Weibo and Twitter. Zhang et al. [29] proposed to encode the syntax parse tree of phrases which contributed towards addressing the authorship attribution problem. By combining lexical characteristics and CNN, they were able to achieve accuracy results of 96.16%, 81.00%, and 56.73% on the IMDB62, CCAT50, and Blogs50 datasets, respectively. Jafarikianabad and Hua [30] used similar datasets and achieved accuracy values of 73.83% and 82.35% on the Blogs50, CCAT50 datasets, respectively, by encoding the syntactic and semantic structures of sentences in texts, and employing a hierarchical neural network based on attention. In [38], a large corpus of blogs with author-provided demographic information was used to study how writing style and content vary by age and gender. It identifies significant differences in vocabulary use and topics among different groups and shows how they can be used for authorship attribution. Moreover, Murauer and Specht [39] applied multiple well-established pre-trained language models to reach better generalization outcomes on the authorship attribution problem. Specifically, they employed variety of language models such as bidirectional encoder representations from Transformers (BERT) [40], a pre-trained general-purpose language representation of DistilBERT [41], and a robustly optimized BERT pre-trained model (RoBERTa) [42] on a large number of extremely diverse authorship-attrribution datasets.

Post-authorship attribution is an unresolved research issue because of the numerous inherent uncertainties that text snippets present. On the foundation of a convolutional neural network, a technique for character-level authorship identification was suggested by Modupe et al. [43]. The suggested approach can effectively extract lexical, syntactic, and structural representations from a given post to determine the authorship of dubious materials. Four benchmark experimental datasets were used to compare the performance of this approach to thirteen state-of-the-art approaches.

A character-level RNN was used by Bagnall [44] to model the language of each author in the training set received from the PAN 2015 author verification problem [45]. Additionally, Shrestha et al. [46] used a three-layer CNN model to determine who posted a tweet. The suggested model was used at the
character level since tweets are generally brief in duration. Furthermore, a hierarchical attention-based neural network was also employed by Verma and Srinivasan [47]. The semantic and syntactic structural features are extracted and encoded using the suggested model. Next, document representations were created by combining these features. They used a collection of convolutional layers in their attention-based model to represent a phrase at the word level. While the structural patterns were recorded using an attention based RNN. Moreover, RNN was used by Jafariakinabad et al. [48] to identify a document's syntactic patterns. They used various combinations of DL models to explore both long-term and short-term dependency (POS) tags in phrases. They concluded that the LSTM-based POS are faster than CNN-based POS in capturing the authorial writing style of short texts. Further, Rhodes [49] investigated the effectiveness of the CNN model using a dataset that included 28 English novels written by 14 writers and eight English books written by six authors, both taken from the PAN2012 Authorship Identification competition. The model was used at the sentence level and achieved an accuracy rate of 76% for the books dataset and 20.52% for the PAN2012 English novels.

Ruder et al. [50] examined how different CNN setups affected the impact of attributing various post types. The best performing technique was a character-level CNN, which were compared to other conventional approaches. In addition, a feedforward neural network language model was applied by Ge et al. [51] to train a classifier to attribute a dataset with only a little amount of data. In comparison to n-gram baselines, the model trained a representation for each word using a window of four grams and achieved an accuracy of 95%.

According to the conducted review of the literature, there are few state-of-the-art techniques for identifying Arabic language writers, especially for the text extracted from livestream broadcasting and the recorded speeches. Most of the existing studies investigated the machine learning methods for AAA. Therefore, this study aims to investigate how to apply deep learning for solving Arabic authorship attribution using religious texts extracted from livestream broadcasting and recorded speeches. The dataset was collected from the official website of Al-Haramain for several authors (scholars) in Saudi Arabia.

III. PROPOSED APPROACH

As mentioned earlier, this paper aims to enhance the performance of the authorship attribution detection. For this purpose, a combination of acoustics and stylometric features were extracted and combined, and deep learning methods were applied. Generally, the proposed approach consists of the following stages: feature extraction, data preprocessing, detection and classification, and ensemble learning-based techniques.

Fig. 1 shows the overview of the proposed approach used for extracting the required features and applying the classification. The full description of the proposed approach stages are given in the subsections.
A. Extraction Stage

In this stage, a set of acoustic, textual and stylometric features are extracted from the weekly live stream of Arabic sermons and the recorded speeches in mp3 files that are available publicly on the official website of Al-Haramain. We collected the dataset from this source because the Modern Standard Arabic (Fusha) is used in all sermons’ texts and recorded speeches. The google cloud APIs were used to extract audio files from the livestream videos and to transcribe speech to text. The final transcription is stored in JSON format and then passed to the next stages. Fig. 2 shows the process of extracting the acoustics and stylometric features using google cloud APIs. The process of extracting the raw textual data from the audio file after converting it to FLAC format is summarized in Algorithm 1. Also, Table I shows a fragment of JSON file obtained as the output of the Google cloud API and the JSON request file fired in Google cloud. For determining the SampleRateHertz of an audio file, we used Apache Tika tool1.

Algorithm 1: Extraction textual data from audio files

Input: livestream Arabic sermons ArS; recorded mp3 files AudioF

Output: JSON file

Initialize: FileMetadata = []; // empty list

For (every ArS and AudioF) do

    ConvertToFlac( ); // convert file to FLAC format

    FileMetadata ← GetMetadata() // get file metadata including

    // SampleRateHertz by

    // Apache Tika tool

    JSON_file = ConstructJSON (FileMetadata)

End

Return JSON_file

B. Data Preprocessing

1) Preprocessing the textual data: Before feeding the raw textual data into the proposed DL model, a set of preprocessing techniques were applied. This is a necessary step to enhance the quality of the DL model. AL-Sarem et al. [52] investigated the importance of the preprocessing techniques in increasing the performance of ML models. The preprocessing phase includes data cleansing techniques, stemming and tokenization techniques. In terms of the used cleansing techniques, since the textual data is obtained directly from the JSON files that were generated by the Google speech-to-text APIs, the only necessary steps are fixing the spelling errors, stop word removal, and completing the words that might truncated due to the non-complete video frame splitting (this is because the video frames had a fixed length and we set them to 30 second per a frame). In addition, we performed the stemming process using the porter stemmer as recommended in [53].

2) Preprocessing the audio files: We treated the audio files as the source for obtaining the acoustic features such as waveforms, spectrogram, spectral roll-off and chroma features. The audio waveforms are represented as spectrograms, which depict the intensity of a signal over time at various frequencies. Fig. 3 to 6 show a sample of waveform, spectrogram, spectral roll-off and chroma images that were obtained when we passed the same sliced audio signal to the acoustics feature extractor.

1 https://tika.apache.org/
TABLE I. JSON FILES: THE LEFT COLUMN PRESENTS A FRAGMENT OF JSON REQUEST FILE, MIDDLE COLUMN PRESENTS THE JSON OBTAINED FILE, AND THE LAST COLUMN A TRANSLATION OF THE OBTAINED TRANSCRIPT TO ENGLISH

<table>
<thead>
<tr>
<th>JSON File Request</th>
<th>JSON file Obtained</th>
<th>Translation to English</th>
</tr>
</thead>
<tbody>
<tr>
<td>{ &quot;config&quot;: { &quot;encoding&quot;:&quot;FLAC&quot;, &quot;sampleRateHertz&quot;: 44100, &quot;Language Code&quot;: &quot;ar-SA&quot;, &quot;enableWordTimeOffsets&quot;: false }, &quot;audio&quot;: { &quot;uri&quot;:&quot;~/audio.flac&quot; } }</td>
<td>{&quot;results&quot;:[]}</td>
<td>“Say, &quot;O Prophet, “The enjoyment of this world is so little, whereas the Hereafter is far better for those mindful ‘of Allah’. And none of you will be wronged ‘even by the width of the thread of a date stone. Beware, O Muslim, from being deceived by this world, as well as avoiding negligence, lust, and whims. God Almighty said, O people, God's promise is true.”</td>
</tr>
</tbody>
</table>

C. Detection and Classification Stage

For the detection and classification stage, the textual data that were extracted as JSON file by the google speech-to-text APIs and prepared at the second stage were fed into the deep learning-based models. DL-based techniques possess the main advantage that no feature engineering is needed [54, 55]. In the training phase, DL classifiers extract and obtain useful features from the input data directly. In this section, we described the models that were applied in this study such as CNN architecture and its pre-trained ResNet34. In addition, this section explains how we used multi-input word embedding as a text representation. For the acoustic features, the pretrained ResNet34 model was fine-tuned and trained using the spectrogram images dataset obtained from the audio files. Then, the outputs of both models (multi-input CNN-based models and the pre-trained ResNet34 model) were combined. Finally, different ensemble methods were used to obtain more accurate results.

1) Multi-input word embedding model: The concept of word embedding (WE) refers to a technique of a text representation in which words having the same meaning are represented similarly. Word2vec, GloVe, and FastText are some recent word-embeddings that are commonly used with ML and DL models. In the proposed multi-input channel CNN model, we used the Keras embedding layer. The proposed model processes textual data in three channels with four, six, and eight grams of input. The word-embedding vector has an output dimensions size of 100 with a maximum length of sequences (input length) computed directly from the input textual data.

2) Architecture of the proposed CNN model: As shown in Fig. 1, the word embedding layer was connected to three parallel CNN blocks. Each block (channel) composes of the following:
• The length of input sequences.
• A layer of embedding set with 100-dimensional real-valued representations.
• A convolution layer with 32 filters and a kernel size set to the number of words to be read simultaneously.
• Max Pooling layer to consolidate the output from the convolutional layer.
• Flatten layer to reduce the dimensional output before concatenation phase.

Finally, we concatenated the output from the three channels into a single vector and processed it through a dense layer and an output layer. Table II summarizes the configuration of each CNN block.

<table>
<thead>
<tr>
<th>CNN Block</th>
<th>Output Dimension</th>
<th>Kernal Size</th>
<th>Filter</th>
<th>Dropout Rate</th>
<th>Activation Function</th>
</tr>
</thead>
<tbody>
<tr>
<td>Block No. 1</td>
<td>100</td>
<td>4</td>
<td>32</td>
<td>0.5</td>
<td>relu</td>
</tr>
<tr>
<td>Block No. 2</td>
<td>100</td>
<td>6</td>
<td>32</td>
<td>0.5</td>
<td>relu</td>
</tr>
<tr>
<td>Block No. 3</td>
<td>100</td>
<td>8</td>
<td>32</td>
<td>0.5</td>
<td>relu</td>
</tr>
</tbody>
</table>

3) ResNet34 model: ResNet34 model is an image-based pre-trained CNN model with 34 convolution layers, one MaxPool, and one average pool layer. Unlike the CNN-based models which commonly experience vanishing gradients during backpropagation, the skipping connections in the ResNet34 [56] are used to solves this issue.

As shown in Fig. 7, the first convolutional layer in ResNet34 model has filter with kernel size of 7x7 followed by a MaxPool layer (the stride is set to 2, which is indicated as “/2” in the Figure). Next, a group of convolution layers were connected using skip connection which is jumping every two layers. The layer of the first group (colored in grey) has kernel size of 3x3 and filters of 64. This layer is repeated three times and layered between the MaxPool layer and the layers of the second group. The layers of the second group have filter of 128 and Kernal size of 3x3 and these are get repeated on this time four times keeping the same skip connection length. In this manner, we continue until we reach the avg_pooling and softmax functions. Each time, the number of filters gets doubled.

4) Ensemble learning-base fusion strategies: Ensemble learning is a technique for merging the outputs of different ML models to improve the overall performance [57-59]. The empirical results show that when different ML/DL models work together, the performance is improved compared to the performance of standalone single model. Armed with this concept, two ensemble learning strategies: the hard voting and soft voting were implemented to combine the output of the multi-input channel CNNs models and the ResNet34 model.

a) Hard voting approach: The hard voting approach follows the majority voting concept in which a class with the most n votes is considered as the final output class. In general, each classifier $c \in C$, makes its own prediction, and a vector size of n store the results, where the number of classifiers that participate in voting pool: $[C_1(x), C_2(x), \ldots, C_n(x)]$. The output class $y$, is then, predicted by applying the formula presented in Equation (1).

$$y = \text{mode}[C_1(x), C_2(x), \ldots, C_n(x)]$$ (1)

b) Soft voting approach: Unlike the hard voting, soft voting determines the output class by projecting probability $p$ of all classifiers [57]. Then, the average probability is computed for each class as follows:

$$P_{\text{max}}(i_n|x) = \frac{1}{n} \sum_{k=1}^{n} P_{\text{mean}}(i_j|x)$$

$$Y = \text{argmax}[P_{\text{mean}}(i_0|x), P_{\text{mean}}(i_1|x), \ldots, P_{\text{mean}}(i_n|x)]$$ (2)

Then, taking into consideration the greatest probability, the output class $y$ is determined according to the formula in Equation (2) as follows:

$$y = \text{argmax}[P_{\text{max}}(i_0|x), P_{\text{max}}(i_1|x), \ldots, P_{\text{max}}(i_n|x)]$$ (2)

IV. EXPERIMENTS

This section presents and analyses the finding results of the proposed model and the ensemble learning-based fusion techniques that applied to dataset of textual and acoustic-based features.

A. Experimental Setup

In this study, an Intel(R) Core (TM) i9-10980HK CPU @ 2.40 GHz and an NVIDIA GeForce MX graphics card run on Windows 11 with 32 GB RAM to implement the proposed methods. All the tested DL models were encoded using Python 3.9 using the Jupyter Notebook (available online: https://jupyter.org). It is also provided by Anaconda distribution (available online: https://www.anaconda.com). In
addition, we employed the open-source PyTorch library (available online: https://pytorch.org) to implement the ResNet34 and tune the multi-input channel CNN models. Since the main aim of the proposed model is to enhance the performance of classifying the authors of Arabic scripts and live stream or recorded Arabic sermons, a set of different DL architectures over 35 epochs was trained. Later, we compared the performance of these models to the proposed model. In addition, the models applied the Adam optimizer and the cross-entropy loss function at 1e-3 learning rate. The Spectrogram images of the audio signal with size of (224 × 224) pixels and batch size of 64 were used for training the ResNet34. Table III summarizes the hyperparameters used for models training.

<p>| TABLE III. CONFIGURATION PARAMETERS OF THE RESNET34 |</p>
<table>
<thead>
<tr>
<th>Parameters</th>
<th>Values</th>
</tr>
</thead>
<tbody>
<tr>
<td>Epochs</td>
<td>35</td>
</tr>
<tr>
<td>Batch Size</td>
<td>64</td>
</tr>
<tr>
<td>Learning rate</td>
<td>1e-3</td>
</tr>
<tr>
<td>Optimizer</td>
<td>Adam Optimizer</td>
</tr>
<tr>
<td>Spectrogram image size</td>
<td>(224x224)</td>
</tr>
<tr>
<td>Loss function</td>
<td>Cross-entropy</td>
</tr>
</tbody>
</table>

B. Dataset

As mentioned earlier, the dataset was scrapped directly from the official website of Al-Haramain in Saudi Arabia (https://www.alharamain.gov.sa/) and its official YouTube channel (https://www.youtube.com/watch?v=o5tC9aWaQ80).

The size of scrapped data was about 2 GB. Table IV shows some statistical characteristics of the dataset. Since the dataset is imbalanced in terms of number of videos (see Fig. 8), we assumed a Gaussian distribution over the candidate authors.

In addition, the dataset contains 14,912 spectrogram images size of (224x224) representing five distinct classes. In the experimental part, we split the dataset into 70% for training, 20% for validation, and 10% for testing.

<p>| TABLE IV. SOME STATISTICAL CHARACTERISTICS OF DATASET |</p>
<table>
<thead>
<tr>
<th>Author</th>
<th>No. of videos</th>
<th>Average video length in (Min)</th>
<th>Total Size in (MB)</th>
</tr>
</thead>
<tbody>
<tr>
<td>AAI AlShaikh</td>
<td>42</td>
<td>18.24</td>
<td>344</td>
</tr>
<tr>
<td>Ahmed Hameed</td>
<td>20</td>
<td>15.38</td>
<td>153</td>
</tr>
<tr>
<td>Albaejan</td>
<td>43</td>
<td>17.32</td>
<td>308</td>
</tr>
<tr>
<td>Alhudaify</td>
<td>68</td>
<td>24.55</td>
<td>775</td>
</tr>
<tr>
<td>AlQaseem</td>
<td>49</td>
<td>20.46</td>
<td>482</td>
</tr>
</tbody>
</table>

C. Performance Metrics

For any classification problem, the performance of ML and DL models can be evaluated by computing the classification accuracy, precision, recall, and F1 score. In this paper, to precisely assess the proposed method, the experiments were conducted and validated using 5-fold cross-validation method. The confusion matrix (refer to Table V for more details) is presented to demonstrate how the statistical metrics (accuracy, precision, recall, and f1-score) can be computed.

\[
\text{Accuracy} = \frac{TP + TN}{TN + FP + FN + TP} \quad (3)
\]

\[
\text{Precision} = \frac{TP}{TP + FP} \quad (4)
\]

\[
\text{Recall} = \frac{TP}{TP + FN} \quad (5)
\]

\[
F1 - \text{score} = \frac{2 \times \text{Precision} \times \text{Recall}}{\text{Precision} + \text{Recall}} \quad (6)
\]

| TABLE V. CONFUSION MATRIX |
|---------------------------|---------------------------|
| Actual Negative | Predicted Negative | Predicted Positive |
| True negative (TN)      | False negative (FN)      | True positive (TP) |

Fig. 8. Distribution of video length as per an author class.

V. RESULTS

The subsections below summarize the findings of the experiments performed on the used dataset. We first presented the results obtained by employing only the extracted textual data. Then, the performance of ResNet34 model was highlighted. Later, we showed the results of fusing both features using the two fusion strategies. It is important here to mention that the results presented in the following subsections are the average value of each experiment that was repeated five times independently.

A. Evaluation with Textual Data

To show the impact of the proposed CNN model with multi-input channels, different kernel size input layer was implemented. Table VI shows the performance of the CNN model with different input channels. The results of CNN model with the multi-input channel are highlighted at the bottom of the table. In addition, the structure of the proposed CNN model with multi-input channel is depicted on Fig. 9.

The results show that the use of textual data yields an acceptable performance in terms of all measuring metrics. However, when the CNN model was restructured and the multi-input blocks were added to the original CNN model, a notable improvement was observed in terms of all metrics which encourage us to keep this structure and test it after the fusion with ResNet34 model.
The acoustic features that were extracted from the audio files are fed into the ResNet34 model. As we mentioned earlier, several acoustic features can be extracted as images from each audio file such as waveforms, spectrogram, spectral roll-off and chroma features. The spectrogram-based images were used. To illustrate the performance of ResNet34, two other pretrained DL models were also implemented namely Xception Model [60] and VGG-19 model [61]. Table VII shows the performance of the deployed DL models achieved when the spectrogram images were used as a training set. The results show that the ResNet34 model overcomes other models and yields accuracy of 90.34%, and 0.903, 0.899, and 0.901 in terms of precision, recall, and F1-score respectively.

### TABLE VII. Performance of CNN-Based Models Used in This Paper

<table>
<thead>
<tr>
<th>CNN model</th>
<th>Accuracy</th>
<th>Precision</th>
<th>Recall</th>
<th>F1-Score</th>
</tr>
</thead>
<tbody>
<tr>
<td>VGG19</td>
<td>79.12%</td>
<td>0.789</td>
<td>0.801</td>
<td>0.795</td>
</tr>
<tr>
<td>Xception</td>
<td>85.31%</td>
<td>0.850</td>
<td>0.860</td>
<td>0.850</td>
</tr>
<tr>
<td>ResNet34</td>
<td>90.34%</td>
<td>0.903</td>
<td>0.899</td>
<td>0.901</td>
</tr>
</tbody>
</table>

### C. Fusion Model

In the previous subsections, the experimental results proved superiority of multi-input channel CNN model and ResNet34 model in detection and classification the authors’ class using the textual data and the acoustic-based features. The next step is to examine the impact of fusing both models using various learning strategies. Both the hard and soft voting approaches have been investigated. Table VIII presents the performance results of the proposed model with respect to the used ensemble learning strategies.

### TABLE VIII. Performance Results of the Proposed Model with Respect to the Ensemble Learning Strategies

<table>
<thead>
<tr>
<th>Ensemble Learning</th>
<th>Accuracy</th>
<th>Precision</th>
<th>Recall</th>
<th>F1-score</th>
</tr>
</thead>
<tbody>
<tr>
<td>Hard voting</td>
<td>92.75%</td>
<td>0.9258</td>
<td>0.9287</td>
<td>0.9272</td>
</tr>
<tr>
<td>Soft voting</td>
<td>93.19%</td>
<td>0.9311</td>
<td>0.9271</td>
<td>0.9291</td>
</tr>
</tbody>
</table>

---

Fig. 9. Architecture of the proposed CNN model with the multi-input channel.

### TABLE VI. The Performance of CNN Models with Different Input Channels

<table>
<thead>
<tr>
<th>CNN model</th>
<th>Accuracy</th>
<th>Precision</th>
<th>Recall</th>
<th>F1-Score</th>
</tr>
</thead>
<tbody>
<tr>
<td>Block No. 1</td>
<td>85.26%</td>
<td>0.856</td>
<td>0.853</td>
<td>0.853</td>
</tr>
<tr>
<td>Block No. 2</td>
<td>85.10%</td>
<td>0.851</td>
<td>0.851</td>
<td>0.851</td>
</tr>
<tr>
<td>Block No. 3</td>
<td>85.42%</td>
<td>0.856</td>
<td>0.854</td>
<td>0.854</td>
</tr>
<tr>
<td>The proposed Model</td>
<td>86.57%</td>
<td>0.864</td>
<td>0.861</td>
<td>0.861</td>
</tr>
</tbody>
</table>

### B. Evaluation with Spectrogram Images

The acoustic features that were extracted from the audio files are fed into the ResNet34 model. As we mentioned earlier, several acoustic features can be extracted as images from each audio file such as waveforms, spectrogram, spectral roll-off and chroma features. The spectrogram-based images were used. To illustrate the performance of ResNet34, two other pretrained DL models were also implemented namely Xception Model [60] and VGG-19 model [61]. Table VII shows the performance of the deployed DL models achieved when the spectrogram images were used as a training set. The results show that the ResNet34 model overcomes other models and yields accuracy of 90.34%, and 0.903, 0.899, and 0.901 in terms of precision, recall, and F1-score respectively.
In addition, the results show the superiority of the combined models over the individual models. The results show that soft voting approach clearly outperforms the hard voting approach in terms of accuracy and precision and both approaches obtained similar results in terms of recall and F1-score. As a result, the soft voting approach is the recommended assembling strategy.

VI. CONCLUSION

This study proposed an Arabic authorship attribution approach that includes five main stages: feature extraction, data preprocessing, detection and classification, and ensemble learning-based fusion strategies. In the first stage, a set of acoustic, textual and stylometric features were extracted from different Arabic live stream sermons and recorded Arabic speeches files for five authors. In the data preprocessing, several techniques were applied for data cleansing, stemming and tokenization. The audio waveforms were represented as spectrograms, which depict the intensity of a signal over time at various frequencies. Next, in the detection and classification stage, the extracted data were fed into the deep learning-based models (CNN architecture and its pre-trained ResNet34). Then, hard and soft voting ensemble methods were applied for combining the outputs of the applied models to improve the overall performance. The experimental results showed that the use of textual data with CNN yields an acceptable performance in terms of all evaluation metrics. Then, when the acoustic features were extracted from the audio files and fed into the ResNet34 model, the results show that the ResNet34 model overcomes other models and yields accuracy of 90.34%, and 0.903, 0.899, and 0.901 in terms of precision, recall, and F1-score respectively. Finally, when the outputs of the multi-input channel CNNs models and the ResNet34 model were combined using hard and soft voting ensemble methods, the results show the superiority of the combined models over the individual ones. The results show that soft voting approach clearly outperforms the hard voting approach in terms of accuracy and precision (93.19% and 0.9311 were obtained respectively). Future works can investigate the application of the proposed model on different datasets and apply different CNN architectures and pre-train models. Different fusion methods can be applied and lead to obtain more enhancements for the detection of AAA.

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CONFLICTS OF INTEREST

The authors declare no conflict of interest.

REFERENCES


Speech-Music Classification Model Based on Improved Neural Network and Beat Spectrum

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Arts College of Sichuan University, Chengdu 401331, China²

Abstract—A speech-music classification method according to a developed neural system and beat spectrum is proposed to achieve accurate classification of speech-music through pre-emphasis, endpoint detection, framing, windowing and other steps to preprocess and collect vocal music signals. After fast Fourier transforms and triangle filter processing, the Mel frequency cepstrum coefficient (MFCC) is obtained, and a discrete cosine transform is performed to obtain the signal MFCC characteristic parameters. After calculating the similarity of feature parameters through cosine similarity, the signal similarity matrix is obtained, based on which the vocal music beat spectrum is obtained. The residual structure is optimized by adding Swish and max-out activation functions, respectively, between convolutional neural network layers to build residual convolution layers and deepen the number of convolution layers. The connected time series classification (CTC) is used as the objective loss function. It is applied to the softmax layer to build a deep optimization residual convolutional neural network for speech-music classification model. The pitch spectrum of vocal music is used as the input information of the model to realize the vocal music classification. The experiment proves that the classification accuracy of the design model is higher than 99%; when the iteration reaches 1200, the training loss approaches 0; when the signal-to-noise ratio is 180dB, the sensitivity and specificity are 99.98% and 99.96%, respectively; the accuracy of voice music classification is higher than 99%, and the running time is 0.48 seconds. It has been proven that the model has high classification accuracy, low training loss, good sensitivity and special effects, and can effectively achieve the classification of speech-music.

Keywords—Vocal music; classification model; beat spectrum; feature parameter extraction; cosine similarity; convolutional neural network

I. INTRODUCTION

With the rapid development of the network and computer field, multimedia information such as audio and video has gradually become the mainstream of information. Audio media is the most important media form besides visual media, accounting for over 20% of the total information [1]. Faced with the huge scale of media databases and the large amount of audio content generated by users daily, it is also difficult for people to find the information they want [2]. The traditional text-based conventional information retrieval technology has been unable to meet the users' retrieval requirements, so how to effectively retrieve this information is a critical error to be resolved. Audio classification is an effective means to achieve rapid audio information retrieval [3]. Speech and music are two of the most common signals in audio. The classification of speech and music signals not only plays an important role in audio retrieval but also plays an important role in speech recognition, beat tracking and other fields [4]. For instance, in speech signal processing, first, determine the type. If it is a speech type, the following steps can evaluate the language, gender, etc. [5]; If it is music, the following steps can evaluate the music type, beat tracking, etc. [6]. This shows that the classification of voice and music is very important, and accurate classification can set a solid base for the next retrieval and other work.

In recent years, many scholars at home and abroad have done a lot of research on audio classification issues, such as vocal music, and have made certain research achievements. For example, Sun HF et al. [7] proposed a vocal music classification algorithm based on zero crossing rate and spectrum. After endpoint detection and subsection preprocessing of voice and music signals, this algorithm combines each audio segment's zero crossing rate and spectrum amplitude characteristics for classification and recognition processing. Finally, the classification of voice and music signals is realized by calculating the probability of becoming voice or music; Zhang X L [8] and others combined the residual network and random forest methods to convert the vocal music audio data of one-dimensional time domain signal into two-dimensional data form of Mel spectrogram and pre-trained the residual network to obtain a network model with high accuracy as the characteristic extractor. The network figure is used to extract the deep features in the audio, and then the random forest algorithm is used to achieve the vocal music classification in the audio; In [9] first proposed an adaptive Mel filtering algorithm to extract the Mel spectrum with higher discrimination and then proposed a cyclic residual network structure, combined with migration and fine-tuning methods, to construct a cyclic residual network spectrum classifier, aiming at the problem that the classification accuracy of voice, music and audio signals in a small sample environment needs to be improved urgently. Combining adaptive Mel filtering algorithm and cyclic residual network spectrum classifier, an audio signal classification model mainly used in small sample environment is generated, through which vocal music in audio can be classified. Although the above three methods play a role in classifying audio, such as vocal music, they also have some defects. The first method ignores the extraction of signal features before the vocal music classification. Although the last two methods extract the signal features of vocal music, they do not consider the music’s rhythmic characteristics in the
audio, which leads to inaccurate extracted features, and thus affects the classification results of the overall audio signal.

The neural network is an algorithm network composed of multiple neurons with strong self-learning features, associative storage performance and the ability to figure out optimal answers at high speed [10]. The convolutional neural network is a network obtained by improving the neural network. It is a feedforward neural network with convolution calculation and depth structure. It belongs to one of the representative algorithms of depth learning. It has the ability to represent learning and can translate and classify the input information according to the hierarchical structure [11]. Beat is an organization form representing a fixed unit's time value and the law of strength and weakness in audio. Its focus is on the relationship between the strength and weakness of each audio bar, as well as the length of time between beats. The beat in music has the characteristics of a periodic cycle. Although some research has focused on music classification, with the popularity of streaming music services, real-time music classification has become increasingly important. For this reason, according to the characteristics of neural network and beat, this paper proposes a vocal music classification model based on improved neural network and beat spectrum, that is, first obtain the beat spectrum of the audio signal of vocal music, and input it into the classification model as the intake characteristic of the audio classification figure. The classification model uses a convolutional neural network, takes the connection timing classification as the objective optimization function, and integrates the residual structure into the convolution layer; Swish and max-out activation functions will be used to improve it, and finally, an end-to-end vocal music classification model based on deep optimization residual convolutional neural network will be formed. This model has good classification performance and strong applicability.

II. CLASSIFICATION MODEL OF VOCAL MUSIC

A. Feature Extraction of Vocal Music Signal Based on Beat Spectrum

Voice and music are two very important audio signals, but they have different characteristics. In the music signal, there are periodic laws with a strong beat, while most of the voice is irregular, and the beat is not obvious. Beat refers to the combination rule of downbeat and downbeat, which is composed of downbeat and downbeat in a certain sequence. In music, the beat uses a strong, weak relationship to organize the music. Therefore, the characteristics of vocal music signals are extracted based on the beat spectrum.

1) MFCC feature parameter extraction of vocal music signal: Generally, if the vocal music signal is directly analyzed, the signal resolution will be very low. However, if the voice parameters of the vocal music signal are extracted and analyzed, the recognition rate will be greatly improved [12]. Mel frequency cepstral coefficients (MFCC), to a certain extent, simulate the characteristics of speech processing by the human ear [13], have good performance in auditory perception, and also have good robustness in the case of channel noise and spectral distortion [14]. The principle of MFCC is to use a group of band-pass filters to filter the sound signal according to the frequency bands of different frequencies and the bandwidth length to obtain the output signal of each band-pass filter [15]. For example, the output signal obtained by filtering the vocal music signal through the band-pass filter is used as the basic feature of the vocal music signal, which can be used as the characteristic parameters of the vocal music signal after transformation and processing [16].

In order to facilitate the study of the human ear's perception characteristics of different frequencies of speech, Mel frequency can be used. 1Mel is 1/1000 of the tone perception degree of 1000Hz. Formula (1) is the conversion formula of frequency $f$ and Mel frequency $F$:

$$F = 3322.23 \log(1 + 0.001f)$$  

(1)

MFCC is proposed based on the above Mel frequency concept. The process of extracting MFCC feature parameters of vocal music signals is represented in Fig. 1.

![Fig. 1. Extraction flow chart of MFCC feature parameters of speech and music signals.](image)
The specific steps for extracting MFCC feature parameters of vocal music signals are as follows:

Step 1: Read the audio file of the vocal music signal

Input the vocal music audio file into the Matlab tool, read the data and sampling frequency of the audio file using the audio read function, select the wav format for the audio file, and record the read vocal music signal as \( r(n) \).

Step 2: The pre-emphasis processing of vocal music signal

Since the travelling wave propagation distance of treble is smaller than that of bass, and the energy is smaller than that of bass, it is easy to be masked by bass, so it is necessary to pre-emphasize the high-frequency signal. The pre-emphasis filter \( \delta \) is used to amplify the high-frequency signal to avoid the loss of high-frequency signal. The transfer function \( H(s) \) of filter \( \delta \) is Formula (2):

\[
H(s) = 1 - \delta s^{-1}
\]  

(2)

Where, \( \delta \in [0.9, 1] \) is generally 0.97 or 0.95.

The pre-emphasis function is Formula (3):

\[
R(n) = r(n) - \delta r(n-1)
\]  

(3)

In the formula, \( n \) is the number of data points in the vocal music signal, usually 1 to 200K.

Step 3: Vocal music signal endpoint detection

Endpoint detection is an important part of vocal music recognition. Effective endpoint detection methods can not only reduce data storage and processing time but also eliminate noise interference in silent segments, making vocal music recognition more accurate. In this paper, short-time energy is used to detect the endpoint of the vocal music signal, and the start frame and end frame of the whole audio segment are determined to reduce the amount of data and calculation of subsequent audio feature extraction and improve the stability of the system.

Set the short-time energy of each frame of data in the music signal to \( \varepsilon \) as the Formula (4):

\[
\varepsilon = \sum_{m=1}^{u_b} y^2(m)
\]  

(4)

Where, \( y(m) \) is the amplitude of the \( m \)th frame of the vocal music signal, \( u_b \) is the frame length, and \( m = 1, 2, \ldots, b \).

Step 4: Vocal music signal framing operation

If the voice of the whole vocal music signal is directly Fourier transformed, it is easy to cause a loss of timing information. However, suppose the signal is divided into several frames, and a fast Fourier transform (FFT) is performed on each frame with a fixed duration. In that case, the loss of information can be effectively avoided. Because a speech signal is a non-stationary time-varying signal, its quasi-stationary feature can only be considered as a stationary process in a short period (generally speaking, the speech signal between 10ms and 30ms is considered a stationary signal). Therefore, the voice signal can be divided into one short time segment. Each short-time segment is called a frame, and each frame contains \( N \) sampling points, usually 256 or 512 for \( N \); thus, the framing operation is completed.

Step 5: Windowing processing of vocal music signals

After the vocal music signal is divided into frames, the periodicity of a small signal segment will not be obvious. At this time, a Hamming window needs to be added so that the vocal music signal in a window will show periodicity. In order to avoid losing the dynamic change information of voice and music signals, there must be an overlap area between every two adjacent frames. The length of the overlap area is generally 1/2 or 1/3 of that. Multiply each frame by the Hamming window so that the continuity of each frame’s left and right ends can be increased.

Set the vocal music signal of the \( m \) frame as \( R(n,m), n = 0,1, \ldots, N - 1 \), and the vocal music signal \( R'(n,m) \) after Hamming window processing is the Formula (5):

\[
R'(n,m) = R(n,m)[0.54 - 0.46\cos(2\pi n / (N - 1))]
\]  

(5)

Where, \( 0 \leq n \leq N - 1 \).

Step 6: Fast Fourier Transform (FFT) of speech-music Signal

The function of the FFT transform is to convert the time domain of the vocal music signal to the frequency domain [17], which can better reflect the characteristics of the signal and facilitate its analysis.

The frequency spectrum of the \( m \)th frame of vocal music signal after FFT transformation is Formula (6):

\[
R(k,m) = \sum_{n=0}^{N'} R'(n,m)\exp(-2\pi nk / N')
\]  

(6)

Where, \( N' \) is the window width of FFT. After taking the modulus square of the spectrum of the vocal music signal, the power spectrum of the vocal music signal can be obtained.

Step 7: Filtering operation of vocal music signal

The obtained power spectrum of vocal music signal is filtered by a triangle filter with Mel frequency average distribution to obtain a group of coefficients \( d_1, d_2, \ldots \), which are the energy output by each filter and also MFCC coefficients.

Step 8: Discrete Cosine Transform of speech-music Signal

MFCC coefficients have a high correlation. Discrete Cosine Transform (DCT) can reduce the dimension of their
correlation. Calculate the MFCC coefficient using the discrete cosine transform to obtain the \( L \) dimension MFCC parameter \( A_l, l = 1, 2, \cdots, L \), as shown in Formula (7):

\[
A_l = \sum_{k=1}^{p} \log(d_k) \cos \left( (kl - (1/2)l)(\pi / p) \right)
\]

(7)

Where, \( p \) is the number of triangular filters. After DCT, MFCC characteristic parameters of vocal music are obtained.

2) **Beat spectrum of vocal music signal**: Beat spectrum is a measure of acoustic self-similarity and a function of time delay. A highly structured or repetitive concert has a strong beat spectrum peak, which reveals the relative intensity of rhythm and a specific beat. Therefore, different kinds of rhythms of the same beat can be distinguished. The pitch spectrum of vocal music first needs to use cosine similarity to calculate the similarity between two pairs of MFCC feature parameters of vocal music [18] to obtain a similarity matrix and then obtain it by evaluating the autocorrelation of the similarity matrix.

The MFCC parameter of vocal music is regarded as the feature vector of the vocal music signal. The alikeness among them is evaluated by calculating the cosine value of the angle between the two vectors. The cosine value of \( 0^\circ \) is 1, while the cosine value of any other angle is not greater than 1, and the minimum value is -1. Music has the characteristics of rhythm, which makes it repeatable to calculate its similarity, while voice does not. The cosine value is Formula (8):

\[
\cos \theta = \frac{(A_i(i)A_j(j))}{\|A_i(i)\| \times \|A_j(j)\|}
\]

(8)

In the formula, \( A_i(i) \) and \( A_j(j) \) represent the two eigenvectors of the vocal music signal respectively, and \( \cos \theta \) represents the calculated cosine value.

Using the cosine parameter of the included angle of the feature vector, the similarity matrix of the vocal music signal is evaluated through Formula (9). The autocorrelation of its similarity matrix calculates the beat spectrum of the signal to obtain the Formula (10):

\[
X(i, j) = \frac{(A_i(i)A_j(j))}{\|A_i(i)\| \times \|A_j(j)\|}
\]

(9)

\[
C(g, h) = \sum_{i,j} X(i, j)X(i + g, j + h)
\]

(10)

Where, \( A_i(i) \) and \( A_j(j) \) are the eigenvectors of the \( i \) and \( j \) frames, respectively, \( X(i, j) \) is the similarity matrix, \( C(g, h) \) is its symmetric matrix, and the beat spectrum \( C(h) \) is obtainable by adding them by row or column.

**B. Speech-Music Classification on the Basis of Convolutional Neural Network**

The neural network has the characteristics of distributed storage, parallel processing and self-learning skill, so it has a wide application view in data processing, pattern identification, classification and other areas. Convolutional Neural Networks (CNN), as an improved neural network, is one of the commonly used speech-music recognition models at present. Its unique convolution structure ensures the translation invariance of speech-music signals in time and frequency domains.

1) **Convolution neural network**: CNN is an algorithm for learning multilayer neural network structure, which is created of the input layer, convolution layer, pooling layer, full connection layer and output layer [19].

The sparse interaction and parameter sharing of CNN is able to decrease the number of training principles and network complexity [20]; the invariance of data scaling and translation is conducive to optimizing the network structure, making the model have a stronger generalization ability in feature extraction [21]. The network structure of CNN is shown in Fig. 2.

![Fig. 2. Structure of convolutional neural network](image-url)
CNN convolution layer has the characteristics of weight sharing and local connection [22]. Set $Q_{a(i')c(j')}$ as the $i'$ neuron on the input $a$ feature plane, and output the connection weight value between the $j'$ neuron on the $c$ feature plane. Then there is Formula (11):

$$Q_{a(i')c(j')} = Q_{a(i'+1)c(j'+1)} = Q_{a(i'+2)c(j'+2)}$$  \hspace{1cm} (11)

In CNN, the output feature surface of each roll-up layer uniquely corresponds to the output feature surface of the pooled layer. The commonly used activation function is the Sigmoid function [23]. Through pooling operation, the features of vocal music signals are further extracted. Common pooling methods include the mean and maximum pooling methods [24]. After the convolution pooling operation, the obtained features enter the full connection layer. Each neuron in the fully connected layer is connected to each neuron in the previous layer, and the fully connected layer can receive all local information in the previous layer.

2) Residual convolutional neural network for depth optimization: Using the idea of a residual network for reference, the input of the upper layer is directly transmitted to the lower layer by adding a congruent mapping layer. This paper combines the input feature $x$ beyond the convolution operation with the outcome after the convolution operation as the intake of the subsequent activation function. Then it outputs new information to build an optimized residual convolution layer. Aiming to raise the number of layers of the depth convolution neural network, Swish and max-out activation functions are introduced to optimize the residual structure, which can directly transmit the input information of the network to the output and improve the degradation and gradient disappearance problems of the network.

The formula of the activation function Swish is $f(x) = x \times \text{sigmod}(x)$, which is non-monotonic, smooth and unsaturated.

The function of the activation function max-out is similar to that of the activation function layer in the network structure of the convolution layer and pooling layer. It is attached in front of each output neuron. The highest value of the node outcome of the same group of the hidden layer is taken, and the output of each node in the hidden layer is calculated as Formula (12):

$$W = \varphi(o), o = \varphi(x + \varphi)$$  \hspace{1cm} (12)

In the formula, the linear activation vector $\boldsymbol{o}$ of the neuron is calculated by inputting the sample data vector $x$, the weight coefficient vector $\varphi$ and the offset term coefficient $\varphi$. Then the nonlinear transformation is performed through the max-out activation function, where $\varphi(o)$ represents $\max(0, o)$. Maxout divides the $M$ hidden layer units of the specified layer into $I$ groups. If each group contains $V$ units, the output of the $i$ group is Formula (13):

$$W_i = \max_{v=0}^{V-1} z_{i,v}, i = 0, 1, \ldots, I - 1$$  \hspace{1cm} (13)

Since the piecewise linear function can fit any convex function with any precision, and the $V$ hidden layer neuron nodes taken by max-out also have the characteristics of linearity, the operation of taking the maximum value also has the characteristics of piecewise linearity, where the number of segments is related to the size of $V$ value. The max-out activation function has a very strong fitting ability and can fit any convex function [25].

The building of the deep optimization residual convolutional neural network is shown in Fig. 3.
Since the max-out function has no weight sharing, it is easy to overfit, so only the max-out activation function is added to the external output of the residual structure. In contrast, the internal convolution layer activation function uses the Swish function to replace the ReLU function. Although both Swish and ReLU function images have no upper and lower bounds, they can increase the nonlinear mapping performance of the network model from the input feature space to the high-dimensional space. The difference is that Swish has the characteristics of non-monotone and smooth. When the sample data of the input dimension is negative, the gradient of the ReLU activation function is 0, and the network model parameters cannot be updated effectively, resulting in the problem of gradient disappearance. However, the first derivative of the Swish activation function is smooth and will not produce saturation. The network model parameters can be updated normally to avoid the disappearance of the gradient, which can improve the training effect of the depth model to a certain extent.

The above depth optimization residual convolutional neural network process can be expressed as follows: Firstly, the acquired vocal music signal beat spectrum is input into the convolutional neural system. Following the convolution process, batch normalization and Swish activation function processing, it is summed with the original input that has not undergone the convolution operation of this layer. The output results obtained are used as the input of the max-out activation function of the next layer, thus forming a group of optimized residual convolutional network structures; a complete depth optimization residual convolutional neural network model is constructed by superimposing the layers of multilayer optimization residual convolutional network structure.

3) Classification of the connection sequence: In the process of vocal music recognition, vocal music generates a real value of training during training, which is in comparison with the anticipated value in the vocal music type. The loss function in the output layer is used to anticipate the degree of inconsistency between the anticipated and the real value. The loss function represents the robust performance of the established model. The smaller the loss function, the better the robust performance of the model.

CTC is a loss function which is used to measure the difference between the input sequence data and the real output after passing through the neural network [26]. CTC introduces an empty node, which does not require full alignment of voice frames. CTC is applied to the speech-music classification in this paper. As the objective function of the softmax layer, CTC will optimize the likelihood between the input and output target sequences.

CTC adopts the maximum likelihood function as Formula (14):

$$B(q) = \sum_{(x,z) \in q} B(x, z)$$

(14)

The CTC loss function is defined as Formula (15):

$$B(q) = -\ln \prod_{(x,z) \in q} P(z|x) = - \sum_{(x,z) \in q} \ln P(z|x)$$

(15)

Where, $P(z|x)$ represents the probability of the output sequence $Z$ for a given input $X$, and $q$ is the training set. When the input is given, the function of CTC is to find the output sequence with the highest probability.

4) Speech-music classification model based on DCNN-CTC: The convolution layer and pooling layer structure in the convolution neural network can help the model accurately recognize the slight deformation and displacement of the input features. As an end-to-end structure, CTC can resolve the issue of sequence misalignment in the recognition procedure of vocal music. However, the disadvantage of the shallow neural network model is that the feature information extracted by the convolution neural network is not significant enough, which affects the final classification and recognition effect. Therefore, this paper proposes a new DCNN-CTC classification model based on the optimized depth residual convolutional neural network. The model raises the number of convolution layers through the residual structure and constructs the end-to-end model structure of the deep convolution neural network, which is created of six convolution layers, a pooling layer and two filled connection layers. The DCNN-CTC vocal music classification model is shown in Fig. 4.

![Speech-music classification model of DCNN-CTC.](image-url)
When using the DCNN-CTC vocal music classification model for training, the pitch spectrum of the vocal music signal is first input into the depth convolution neural network. The convolution layer uses the residual structure. After six layers of convolution operations, the pitch spectrum of the original vocal music is extracted for depth features so that the local information of the audio features can be fully integrated. The step size of the convolution layer is set to $1\times1$. The first convolution layer in the residual structure is to reduce the dimension of the vocal music signal. The size of the convolution core is $1\times1$. Then the output of the convolution layer is imported to the pooling layer for sampling through maximum pooling. The full connection layer of the two layers has 1024 nodes, and its activation function is maxed out. The softmax layer adopts an end-to-end CTC structure to classify the input vocal music. The end-to-end CTC structure can further improve the robustness of the model, speed up decoding, and obtain the state output of features after classification. The supervised gradient descent method is used for model training in this paper.

III. EXPERIMENTAL ANALYSIS

Take two audio segments in a radio station as the experimental object, and the time length is the 20s. In order to confirm the validity of this type, the test accomplished a classification test on this segment of vocal music. First, the vocal music signal is pre-emphasized and windowed by frames. Set the frame length to 30, the frameshift to 15, and add 0 if the frame length is less than 30. The window function is Hamming window. After MFCC feature parameter extraction and cosine similarity calculation, the vocal music beat spectrum is obtained. Input the acquired vocal music beat spectrum into the model in this paper. The convolution layer of the model in this paper is set as 6 layers. After three residual block structures, the extracted characteristics are compressed and extracted through the maximum pooling method. Finally, they are transferred to the softmax layer by two filled connection layers, which use the CTC loss function. The supervised random gradient descent method is used for model training, with 128 iterations and an initial learning rate of 0.01. After the experiment, the classification results of this segment of vocal music are shown in Fig. 5 and Table I.
<table>
<thead>
<tr>
<th>The name of the audio</th>
<th>The classification results</th>
</tr>
</thead>
<tbody>
<tr>
<td>Audio 1</td>
<td>Speech signal</td>
</tr>
<tr>
<td>Audio 2</td>
<td>Music signals</td>
</tr>
</tbody>
</table>

Due to the periodicity of the music signal, it has a strong beat, while most of the voice is irregular, and the beat is weak. It is perceivable from Fig. 5 that the peak value in the vocal music signal obtained by using the model in this paper corresponds to its main rhythm component, and the amplitude of different peaks reflects the strength of the corresponding rhythm of the signal. In addition, the peak value change of some vocal music signals with a strong sense of rhythm will be relatively obvious. In contrast, the peak value change of vocal music signals with a weak sense of rhythm will be slightly weak, which is able to invert the main features of the vocal music signal better. It can be seen from the identification diagram of audio 1 signal that the signal has no regularity and a weak beat; it can be seen from the identification diagram of audio 2 signal that the signal changes periodically, with a regular period from 0s to 9.6s and a regular period from 9.7s to 20s, and the beat is strong.

Through the above analysis, it shows that this model is effective for the vocal music classification.

In order to measure the effect of endpoint detection on the vocal music signal mentioned in this paper, the experiment collected a section of vocal music signal on the radio station. The sampling frequency is 8kHz, the accuracy is 16bit, and the duration is 30s. The result of endpoint detection of this signal is shown in Fig. 6.

It is perceivable from Fig. 6 that after endpoint detection of the vocal music signal using the model in this paper, the audio period with low head and tail energy of the signal is removed. The start frame and end frame of the whole audio signal are determined, which can effectively reduce the amount of data and calculation of subsequent audio feature extraction and improve the stability of the model.

To judge the excellence of MFCC characteristic parameter extraction proposed in this model, an audio file intercepted from the radio station is input into the Matlab tool in the experiment. The audio file data and sampling frequency are read using the audio read function, and the audio file is in wav format. Set the audio frame length to 256 points, frameshift to 80, filters to 24 groups, and dimensions to 12. Use this model to extract the MFCC feature parameters of this audio, and the results are shown in Fig. 7.
In Fig. 7, comparing the collected original signal with the signal after MFCC feature parameter extraction, we can see that the collected original signal has 15K data points, and the audio signal data points after MFCC parameter extraction are only 480. After amplification and comparison, the audio signal waveform after MFCC parameter extraction is more consistent with the trend and trend of the original signal waveform; after MFCC parameters are extracted, 480 data points completely reflect the characteristics and change trend of 15K data points. From this, it can be seen that the MFCC feature parameter extraction proposed in this model can reflect the characteristics of the entire audio signal, which is conducive to subsequent analysis of the signal and lays a good foundation for the next step of speech-music classification.

The convolution layer is an important part of feature learning and extraction in the convolutional neural network structure. Different convolution layers will lead to different feature representations. Therefore, the experiment tests the performance of the residual convolutional neural network with depth optimization when the convolution layer is 3, 4, 5 and 6 layers. The preprocessed vocal music signal data set is divided into a training data set and a test data set at a ratio of 7:3 for experiments to determine the number of convolution layers with the best classification performance. The experiment sets the input length, the characteristic dimension and the number of iterations obtained by the full connection layer as 150, 3600 and 1200, respectively. It obtains the classification accuracy and training loss of the depth-optimized residual convolutional neural network under different convolution layer structures, as shown in Fig. 8 and Fig. 9.
It is perceivable from Fig. 8 that for the depth-optimized residual convolutional neural network figure with a 6-layer convolution layer structure, when the number of iterations is 600, the classification accuracy has reached more than 99%; when the number of iterations is 1200, the classification accuracy has reached 99.87%, approaching 100%; while for the depth optimized residual convolutional neural network model with other hierarchical structures, the performance in classification accuracy is slightly worse, which shows that, When six convolution layers are set in the depth optimization residual convolutional neural network model, it has a strong classification ability.

It is perceivable from Fig. 9 that when the number of iterations is between 400 and 600, the training loss of the deep optimization residual convolutional neural network decreases the most, especially for the convolution neural network with six convolution layers; the training loss is only about 0.01 at this time. By the time the number of iterations is 1200, the training loss is close to zero, and the training loss of other hierarchical networks is inferior to that of the convolution neural network with six convolution layers; at the same time, it also shows that the network can better avoid the overfitting phenomenon.

In conclusion, when the convolution layer of the depth optimization residual convolutional neural network model proposed in this method is 6 layers, it performs well both in classification performance and training loss. It is superior to other hierarchical convolution neural networks, with good classification performance.

For classification models, sensitivity and specificity are two important performance evaluation indicators. Sensitivity represents the proportion of all positive samples to be paired, which measures the recognition ability of classification.
models to positive samples; Specificity represents the proportion of all negative samples divided into pairs and measures the recognition ability of classification models to negative samples. The sensitivity and specificity are Formula (16) and (17), respectively:

\[
sensitive = \frac{TP}{P} \quad (16)
\]

\[
specificity = \frac{TN}{N} \quad (17)
\]

Where, \(P\) represents a positive sample, \(N\) represents a negative sample, \(TP\) represents a positive sample classified as a positive sample, and \(TN\) represents a negative sample classified as a negative sample.

The experiment selects 100000 vocal music audios from the audio database of the radio station as the sample set, of which 46000 are voice audio, set as positive samples, and 54000 are music audio, set as negative samples. Under the conditions that the signal-to-noise ratio is 20dB, 40dB, 60dB, 80dB, 100dB, 120dB, 140dB, 160dB, and 180dB, the model in this paper is tested from the perspective of sensitivity and specificity. The final results are shown in Fig 10.

It is perceivable from Fig. 10 that under different signal-to-noise ratios, the method in this article has a good performance both in sensitivity and specificity. When the signal-to-noise ratio is 100dB, the sensitivity and specificity reach 99%. When the signal-to-noise ratio is 180dB, the sensitivity and specificity reach 99.98% and 99.96%, respectively. It is perceivable that the model in this article has good recognition ability and classification effect for speech-music.

To measure the function of the model in this article further, for vocal music classification, the experiment selects different types of audio from the audio library of the radio station as the sample set; the sampling rate is 32kHz, the format is PCM format, the whole number of demos is 30min, in which the voice duration is 18min, including the pronunciation of men and women of different ages, different speakers, and different sentences; The music lasts for 12 minutes, including various instrumental music, such as violin, piano, zither, flute, etc. 70% of the samples in the sample set were chosen as training demos and 30% as test demos. The experiment compares the classification model of zero crossing rate and spectrum in reference [7], the classification model of the residual network and random forest in reference [8], and the classification model of Meyer spectrum and cyclic residual in reference [9] with the model in this paper, and analyzes the classification results of different models for vocal music, as presented in Table II.

<table>
<thead>
<tr>
<th>Methods</th>
<th>Audio type</th>
<th>The total number of samples</th>
<th>Number of corrected samples</th>
<th>Number of wrong samples</th>
<th>Classification accuracy /%</th>
<th>The average classification accuracy /%</th>
<th>Running time /s</th>
</tr>
</thead>
<tbody>
<tr>
<td>Model in reference [7]</td>
<td>Voice</td>
<td>273</td>
<td>265</td>
<td>8</td>
<td>97.07</td>
<td>95.47</td>
<td>0.75</td>
</tr>
<tr>
<td></td>
<td>Music</td>
<td>147</td>
<td>138</td>
<td>9</td>
<td>93.88</td>
<td>97.23</td>
<td>0.59</td>
</tr>
<tr>
<td>Model in reference [8]</td>
<td>Voice</td>
<td>273</td>
<td>269</td>
<td>4</td>
<td>98.53</td>
<td>95.92</td>
<td>0.67</td>
</tr>
<tr>
<td></td>
<td>Music</td>
<td>147</td>
<td>141</td>
<td>6</td>
<td>95.92</td>
<td>94.87</td>
<td></td>
</tr>
<tr>
<td></td>
<td>Music</td>
<td>147</td>
<td>135</td>
<td>12</td>
<td>91.84</td>
<td>91.84</td>
<td></td>
</tr>
<tr>
<td>In this paper, the model</td>
<td>Voice</td>
<td>273</td>
<td>272</td>
<td>1</td>
<td>99.63</td>
<td>99.82</td>
<td>0.48</td>
</tr>
<tr>
<td></td>
<td>Music</td>
<td>147</td>
<td>147</td>
<td>0</td>
<td>100.00</td>
<td>100.00</td>
<td></td>
</tr>
</tbody>
</table>
It can be seen from Table II that different models are used to classify different types of vocal music. The accuracy of the model in this paper for music recognition reached 100%, and there was only one error in speech recognition, with an average classification accuracy of 99.82%. It has been proven that the proposed model has good classification performance and the running time is the shortest compared to other algorithms, only 0.48s. While the average classification accuracy of the reference [7] model for vocal music is 95.47%, the running time is 0.75 seconds, the mean classification correctness of the reference [8] model is 97.23%, and the running time is 0.59 seconds. The mean classification correctness of the reference [9] model is 93.35%, and the running time is 0.67 seconds. It can be seen from the above data that compared with the other three models, the vocal music classification in this model is the most accurate, and the classification speed is the fastest. It has a good classification function and convergence effect.

IV. DISCUSSION

The speech-music classification model based on improved neural networks and beat scores is a research aimed at improving the accuracy and efficiency of music classification. This study designed a novel model that achieved impressive results by combining an improved neural network architecture and beat spectrum feature extraction method.

Firstly, the model in this study performed well in terms of music classification accuracy, achieving an accuracy rate of over 99%. This means that the model can highly accurately classify different types and styles of music, providing users with more accurate music recommendations and personalized experiences.

Secondly, after 1200 iterations of training, the training loss of the model approaches zero. This indicates that the model can fully learn and adapt to music datasets, and has strong learning and generalization abilities.

In addition, the model exhibits very high sensitivity and specificity at high signal-to-noise ratio (180dB), reaching 99.98% and 99.96%, respectively. This means that even in noisy environments, the model can still accurately recognize and classify music, providing users with stable and reliable classification performance.

Finally, the study also focuses on the runtime of the model. After optimization, the running time of the model is only 0.48 seconds, which means that the model has high efficiency and practicability, and can quickly classify music in real-time applications.

In summary, the speech-music classification model based on improved neural networks and beat scores is an exciting research topic. By designing models with high accuracy, low training loss, high sensitivity, and special effects, and completing classification tasks in a short running time, this study has made significant contributions to the development of music classification and provided strong support for music recommendation systems and other related applications.

V. CONCLUSION

With the development of cloud storage and Internet technology, more and more multimedia data, such as audio, has entered people's lives. In order to save local storage space, many individuals and enterprise users store multimedia data in the cloud, which increases the technical burden of multimedia data retrieval. Classifying audio is an effective means to achieve fast retrieval. As the two most important types of audios - voice and music, their classification has important application value in content-based audio retrieval, video retrieval and summarization, and voice document retrieval. It is an important preprocessing work in sound signal processing. Therefore, this paper proposes a vocal music classification model based on an improved neural network and beat spectrum. This model combines CTC and residual network design ideas, proposes an improved depth optimization residual convolutional neural network structure, and introduces the beat spectrum as the model's input. The model in this paper effectively solves the problem of model overfitting and improves classification accuracy. The innovation work of this paper mainly includes:

1) Design the pitch of vocal music. In the characteristic extraction of the original audio signal, the beat feature is added to enhance the precision of characteristic extraction.

2) The introduction of residual structure. The residual structure is introduced into the convolutional neural network to increase the number of convolution layers and increase the number of convolution layers to 6 layers, which deepens the depth of the network model, extracts the features of deeper vocal music signals, and better enhances the classification correctness of the network figure.

3) Introduction of CTC. The softmax layer of the convolutional neural network adopts an end-to-end CTC structure, which can solve the issue of sequence misalignment in the recognition procedure of vocal music, improve the robustness of the model, speed up decoding, and obtain the state output of features after classification.

Through experiments, it has been proven that the application of this model can effectively achieve accurate classification of speech and music, with small errors, high accuracy, and fast speed.

In order to further improve the performance of the design in practical applications, in-depth research will be conducted in the following aspects in the future:

1) Dataset expansion: Find more datasets for training and evaluating model performance. These datasets can contain different types and styles of music, as well as different language and cultural backgrounds.

2) Model architecture optimization: try different neural network architectures, such as Convolutional neural network (CNN), Recurrent neural network (RNN) or attention mechanism, to improve the modeling ability of the model for music features.

3) Multimodal learning: Combining audio data with data from other modalities (such as images or text) to improve the
performance of music classification models. For example, audio data can be combined with lyrics or album cover images for joint training.

**DATA AVAILABILITY**

The raw data supporting the conclusions of this article will be made available by the authors, without undue reservation.

**CONFLICTS OF INTEREST**

The authors declared that they have no conflicts of interest regarding this work.

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**REFERENCES**


A Method for Evaluating the Competitiveness of Human Resources in High-tech Enterprises Based on Self-organized Data Mining Algorithms

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Abstract—The level of human resources competitiveness of high-tech companies affects the efficiency and effectiveness of enterprises to a particular extent. To achieve sustainable development of high-tech enterprises, an evaluation method of human resource competitiveness of high-tech enterprises based on a self-organized data mining algorithm is proposed. The fuzzy clustering algorithm is used to select five first-level indexes for the evaluation of HR competitiveness of high-tech companies, including human capital power, human resources policy incentive power, and human resources performance manifestation power, and to construct the initial evaluation indicator setting. The self-organized data mining algorithm is used to identify the key attributes related to the human resource competitiveness of high-tech companies within the initial assessment indicator setup, reduce the complexity of the indexes and construct the final rating index system. The multi-level fuzzy evaluation method is applied to calculate the evaluation index weights and fuzzy evaluation matrix to obtain the assessment results of HR competitiveness of high-tech enterprises. The experimental results show that the information contribution rate of the evaluation index system constructed by this method is higher than 95%, which can accurately evaluate the human resource competitiveness of high-tech enterprises.

Keywords—Self-organized data mining algorithm; high-tech enterprises; human resources; competitiveness evaluation; multi-level fuzzy evaluation method

I. INTRODUCTION

Human resources are the physical and mental workers who can promote social and economic development and create material and spiritual wealth for society [1, 2]. High-tech enterprises are organizations based on high technology. As creativity, information, and technology become more and more constituents of products, the knowledge content in products and services increases and the development of enterprises depends to a large extent on the ability to create knowledge and the level of technology possessed, thus human resources as carriers of knowledge and skills become strategic resources of enterprises [3-5]. The competitiveness of the human resources of a high-tech enterprise refers to the ability to integrate the strengths of the human resources of a high-tech enterprise with the process of human resource integration, which mainly emphasizes the ability to attract, invest, develop, utilize, and generate economic performance of HR of high-tech companies under the effective human resource management mechanism, to achieve the rapid and sustainable development of high-tech enterprises and open the gap with other high-tech zones [6]. However, high-tech enterprises still have the problem of a low overall level of human resource competitiveness in human resource management, so it is of considerable theoretical and practical importance to research how high-tech enterprises manage human resources and integrate human resource quality to bring into play the competitive advantage of enterprises.

Collins [7] examined how scholars in the field of strategic human resource management have utilized the resource-based view of the firm to contend that an HR strategy centered around high commitment can result in a competitive edge by establishing exclusive and valuable employee-centered resources at the firm level. Nevertheless, previous research has not clarified why variations in employee-centered resources remain among firms that adopt the same HR strategy, nor have they taken into account the significance of aligning employee-centered resources with other organizational abilities. The author suggested that the cognitive abilities of the CEO, social capital, and human capital can help clarify when a pursuit of a high commitment HR strategy leads to an increase in employee-centered resources and when firms can effectively handle and use them for competitive advantage. Utilizing Grey Relational Analysis (GRA) to analyze data from the CGSS and the China Health Statistics Yearbook for 2013 and 2015, Peng et al. [8] investigated the determinants of public satisfaction with the health system in China. The findings revealed that the percentage of total healthcare expenditure allocated by the government is the most critical factor for public satisfaction in both years. Moreover, out-of-pocket expenses were identified as a significant factor in 2013, while hospital beds per thousand population were crucial in 2015. The importance of healthcare workforce per thousand population increased from 2013 to 2015. The study’s results indicated a transition in priority from economic affordability to more people-centric services in recent years. However, the research also highlighted persistent regional disparities and gaps that require attention in future healthcare reform endeavors. Geng and Hengxin [9] aimed to shed light on the evolution and contemporary environmental management practices of industrial parks in China. Industrial parks have played a pivotal role in China’s economic advancement, particularly in terms of attracting foreign investment. The paper outlined the characteristics of industrial parks in China, delineated the principal site selection criteria, and highlights crucial factors for foreign investors. It also underscored the necessity of pursuing sustainable development in industrial parks, given their historical contribution to environmental pollution. The authors advocated for the
adoption of eco-industrial development as a fresh developmental approach for industrial parks. Overall, the paper contributed to a more comprehensive comprehension of the current state of affairs and management outlooks concerning the growth of industrial parks in China.

In order to understand the competitiveness of enterprises, provide direction for human resource improvement in high-tech enterprises, and compare with competitors, the evaluation method of human resource competitiveness of high-tech enterprises based on a self-organized data mining algorithm is investigated. Follow the following approach for research:

1) Build an evaluation index system for human resource competitiveness of high-tech enterprises, comprehensively consider the weights and contributions of different indicators, and form a comprehensive evaluation result to provide a comprehensive and objective evaluation of human resource competitiveness.

2) Based on self-organizing data mining algorithms, key indicators are selected to determine human resource competitiveness evaluation indicators suitable for high-tech enterprises.

3) Evaluate the human resource competitiveness of high-tech enterprises by constructing a factor set, determining the weights of each indicator layer, constructing grade evaluation standards, determining membership relationships, constructing a fuzzy evaluation matrix, and determining a fuzzy subset vector;

4) Design experiments to verify that the proposed method can accurately evaluate the human resource competitiveness of high-tech enterprises.

The idea of self-organized data mining is the first milestone in developing mining theory and method. The self-organized data mining algorithm can use complete and incomplete induction algorithms to achieve the automatic selection of the optimal model. Through the research of this method, it is hoped to optimize human resource allocation, motivate employee motivation, improve employee satisfaction and retention rate, and promote the formation of innovation and learning culture. These benefits help enterprises improve their human resource management level, enhance competitiveness and sustainable development capabilities.

II. MATERIALS AND METHODS

A. Construction of the Evaluation Index System for Human Resources Competitiveness of High-Tech Enterprises

In constructing the evaluation index system for the human resources competitiveness of high-tech enterprises, the fuzzy clustering algorithm is used to select evaluation indexes. Cluster analysis is a young branch of numerical taxonomy [10], which classifies indexes according to the degree of affinity of representative sample indexes in nature. In the assessment of the identifying the components of HR competitiveness of high-tech enterprises, because of the degree of reflection of various evaluation elements on the evaluation of HR competitiveness of high-tech enterprises, the boundaries of each other are not obvious. They have a certain degree of fuzziness [11], so employing fuzzy mathematical methods to deal with them is more appropriate.

The stages of using a fuzzy clustering algorithm to select the assessment indexes of HR competitiveness of high-tech enterprises are as follows.

Step 1: Characterize the theoretical domain

The assessment elements of HR competitiveness of high-tech enterprises are used to be classified as a thesis domain, 

\[ B = \{b_1, b_2, \ldots, b_m\} \]

where \( b_i \) is characterized by a set of rating data, i.e.,

\[ b_i = \{b_{i1}, b_{i2}, \ldots, b_{ij}, \ldots, b_{in}\} \]

Step 2: Determine the fuzzy relationship

The importance of various evaluation elements is fuzzily scored according to the specified scoring criteria to form the evaluation element scoring matrix \( X(x_{ij}) \). On this basis, the scoring matrix is normalized to calculate the correlation coefficient \( r_{ij} \) between different evaluation elements and establish the fuzzy similarity matrix \( R \) on the domain \( B \). The most common method to determine the similarity coefficient \( r_{ij} \) is the closeness method. The closeness of \( u_i \) to \( u_j \) is

\[ r_{ij} = N(x_i, x_j) \]

When \( N \) takes the distance closeness,

\[ r_{ij} = 1 - \sum_{k=1}^{m} |x_{ik} - x_{jk}| \]

Step 3: Calculate the fuzzy equivalence matrix

This fuzzy similarity relation matrix \( R = [r_{ij}]_{n \times n} \) satisfies only self-reflexivity and symmetry but not transferability. To perform fuzzy clustering analysis, a Boolean multiplicative transfer closure operation is performed on \( R \) up

\[ R^k = R^{2k} \quad (k = 2, 4 \ldots 2^n) \]

\[ R^* = R^k = t(R) \]

\( t(R) \) is the fuzzy equivalence relation matrix, and the fuzzy equivalence relation matrix \( t(R) \) satisfies self-reflexivity, symmetry, and transferability [12].

Step 4: Clustering

The fuzzy equivalence matrix \( r(R) \) is used to find its intercept matrix under different thresholds \( \phi \), and the intercept matrix \( \phi \) is used to analyze the similarity of the evaluation elements [13, 14] so that the indexes of each evaluation element can be clustered.

Step 5: Determine the optimal threshold \( \phi \)
The intercept matrix of the fuzzy equivalence relation matrix \( t(R) \) under different thresholds \( \phi \) is
\[
t(R) = \hat{\lambda}(r_{ij})
\]
where
\[
\phi(r_{ij}) = \begin{cases} 
1, & r_{ij} \geq \phi \\
0, & r_{ij} < \phi
\end{cases}
\]
(1)

Different values of \( \phi \) will get different truncation matrix \( R_\phi \). Generally, when \( \phi \) gradually decreases from large to small, the classification becomes coarser from fine, forming a dynamic cluster analysis picture. Through this cluster analysis picture, the best \( \phi \) value is selected according to the actual problem's needs, the classification objects' master-slave ranking can be realized, and the identification and analysis of the components of the HR competitiveness of high-tech companies can be realized.

Based on the principles of comparability, operability, the composition of qualitative and quantitative, dynamism, comprehensiveness, and complexity [15-18], the comprehensive index system of initial high-tech enterprises' HR competitiveness is constructed by combining the specifications of high-tech companies themselves. This index system mainly consists of five primary indexes, 16 secondary indexes, and 32 tertiary indexes, as presented in Table I.

The human resource competitiveness of high-tech enterprises mainly consists of five parts: human capital power, human resource environment attractiveness, human resource policy incentive power, human resource investment competitiveness, and human resource performance manifestation power. With the goal of reflecting both the existing competitiveness and the future competitive potential, Human resource environment attractiveness, human resource policy incentive power, and human resource investment competitiveness are the direct manifestation of human resource competitiveness of high-tech zones, and human capital power is the bridge between human resource environment attractiveness, human resource policy incentive power, and human resource investment competitiveness and human resource performance manifestation power.

B. Selection of Key Indexes Based on Self-organized Data Mining Algorithm

Since the initial human resource competitiveness evaluation index system of high-tech enterprises contains a large number of indexes, using all the indexes to evaluate the competitiveness of high-tech enterprises will consume a lot of time and space resources, so in order to improve the evaluation efficiency, it is necessary to analyze the indexes in the initial human resource competitiveness evaluation index system of high-tech enterprises, identify the key attributes related to the competitiveness of high-tech enterprises, and design the evaluation system of human resources competitiveness of high-tech enterprises based on this.

In the process of identifying the key attributes related to the competitiveness of HR in high-tech companies, the objective system analysis (OSA) way in the self-organized info analysis algorithm is used to extract the key attributes [19]. The objective system analysis method in the self-organized data mining algorithm automates the following steps objectively.

1) Divide the observed sample data into training and detection sets.
2) Generate the system to be selected at each stage by different variables and growing complexity [20].
3) Estimate the unknown parameters on the training set for the parameters of high-tech enterprises' human resource competitiveness evaluation system.
4) Select some optimal human resource competitiveness evaluation systems for high-tech enterprises at each stage using data from the testing set.
5) When repeating steps (2) to (4), the complexity of the assessment setup of HR competitiveness of high-tech enterprises grows gradually as the number of HR of high-tech companies increases. Otherwise, the final optimal evaluation system of human resources competitiveness of high-tech companies is selected.

According to the basic principles of the OSA algorithm, the key attributes related to the competitiveness of HR of high-tech enterprises are identified in the following steps.

1) Divide the initial data set of HR competitiveness evaluation indexes of high-tech enterprises \( W \) into two groups with the same sample size \( \gamma \) and \( \hat{\lambda} \) so that \( W = \gamma \cup \hat{\lambda} \). The data lengths of \( \gamma \) and \( \hat{\lambda} \) are both \( N \), and \( P = \{1, 2, \ldots, m\} \), \( Q = \{1, 2, \ldots, N\} \).

2) Let \( k = 1 \). For the \( i \)-th variable (initial evaluation index data of high-tech enterprises' human resource competitiveness), parameter estimation is performed by the least squares method on the full set of samples [21] to obtain:
\[
x_i = \gamma_0 + \lambda \cdot x_0 \quad i \in P
\]
(2)

Then perform parameter estimation using least squares on the data sets \( \gamma \) and \( \hat{\lambda} \), respectively, to obtain:
\[
x_i' = \gamma_0' + \lambda' \cdot x_0 \quad i \in P
\]
(3)

Calculate the minimum deviation criterion value [22].
\[
r_{n_i} = \frac{\sum_i (x_i'^{(k)} - x_i^{(k)})^2}{N_i} \quad i \in P
\]
Let \( \eta_i = \min(r_{n_i}) \)
Let \( \hat{k} = k + 1 \)
(4)
Then carry out the parameter estimation for the independent variables in Formula (5) using the least squares on $k$ and obtain a system of k-Formulas.

Calculate the minimum deviation criterion value for each attribute variables $x_1, x_2, \ldots, x_i (i = 1, \ldots, r \in P)$. least squares estimate the parameters on the full set of samples to obtain a system of k-Formulas.

Within the $n$ independent variables, take any k different attribute variables $x_1, x_2, \ldots, x_i (i = 1, \ldots, r \in P)$, least squares estimate the parameters on the full set of samples to obtain a system of k-Formulas.
In Formula (8), calculate $\eta_{ki}, \eta_{kj}, \ldots, \eta_{kr}$ according to Formula (4), denoted as
\[
\eta_k = \min(\eta_{i_{j\cdots k}}).
\]

(4) Compare $\eta_k$ with $\eta_{k-1}$; if $\eta_k \leq \eta_{k-1}$, go back to step (3); otherwise, stop the algorithm and record the minimum deviation criterion value $\eta_{k-1}$. The variables in the set of Formulas corresponding to the minimum deviation criterion value of $\eta_{k-1}$ are the characteristic variables of the system [23]. The attributes corresponding to these characteristic variables are the key attributes for evaluating the competitiveness of HR of high-tech enterprises.

The obtained key attributes of the human resource competitiveness of high-tech companies are used to construct the final applied assessment index setup of HR competitiveness of high-tech enterprises, according to which the fuzzy comprehensive assessment process is applied to design the assessment project of human resource competitiveness of high-tech enterprises and obtain the evaluation results of human resource competitiveness of high-tech enterprises.

C. Evaluation of Human Resource Competitiveness of High-tech Enterprises

In the process of evaluating the competitiveness of human resources in high-tech enterprises, there are several elements to be assumed, and they are able to be divided into various layers and groups [24]. When conducting a comprehensive evaluation of such an object, in order to facilitate the distinction of the act and effect of the ultimate assessment of distinct elements and to receive the data that all elements collected more comprehensively, the number of elements is divisible to various groups based on particular features [25], with a small number of factors in one category and a comprehensive evaluation of each type first, followed by a high-level synthesis of the evaluation results between the classes. Therefore, the assessment of HR competitiveness of high-tech enterprises is achieved by a multi-level fuzzy comprehensive evaluation method.

a) Construction of Factor Sets

The theoretical factor domain $U$ is divided into $m$ disjoint subsets according to different attributes, i.e.,
\[
B = \bigcup_{j=1}^{m} B_j,
\]
and $B_i \cap B_j = \phi(i \neq j)$.

The first-level indicator set is further divided, and the second-level indicator set is further divided and called the first-level indicator set; each first-level indicator is called the second-level indicator set. Similarly, a set of indicators can be obtained at three or even more levels.

b) Determination of the Weights of Each Index Layer

The design of the weights of each level of indexes mainly uses the hierarchical analysis method to calculate the weights of tertiary, secondary, and primary indexes, in turn. The comparison matrices are constructed for different indicator levels, and the weights of each indicator are calculated [26], and then the consistency test is performed. If they pass the test, the weights can be determined; if they do not pass the test, the comparison matrix is reconstructed, the weights of each indicator are calculated, and the consistency test is conducted again until they pass the test so that the weights of each indicator can be scientifically determined.

c) Construction of Grade Evaluation Criteria

The evaluation level set is a collection of various evaluation results that evaluators may make about the competitiveness of HR in high-tech enterprises, i.e.,
\[
V = \{v_1, v_2, \ldots, v_e\}.
\]

d) Determination of the Affiliation Relationship to Construct the Fuzzy Evaluation Matrix

The fuzzy assessment matrix for every element of the second level is created based on the third-level indexes. The factors of the third level index $B_{ijk}$ are evaluated according to the rank of the evaluation set $V = \{v_1, v_2, \ldots, v_e\}$, and the affiliation degree of $B_{ijk}$ to $V_e$ is evaluated by $R_{ie} (i = 1, 2, \ldots, n; e = 1, 2, \ldots, f)$, thus forming the evaluation matrix of the factors of the second level index $R_{ij}$.

\[
R_{ij} = \begin{bmatrix} R_{i1} & \cdots & R_{ie} \\ \vdots & \ddots & \vdots \\ R_{i1} & \cdots & R_{ie} \end{bmatrix} \tag{9}
\]

$R_{ij}$ denotes the fuzzy matrix corresponding to the factors of the 3rd level subset under the factor $i$ at the 2nd level, $i$ shows the number of elements of the 3rd level indexes under the 2nd level indexes, $e$ is the number of human resource competitiveness level of high-tech enterprises, where:

$R_{ie} (i = 1, 2, \ldots, n; e = 1, 2, \ldots, f)$

denotes the affiliation degree of the $f$-th level comments made to the $i$-th evaluation index, i.e., later than using professionals to score the assessment outcomes, it can get $V_{ie}$ level comments $V_e$ for the $i$-th evaluation index, then:

\[
R_{ie} = \frac{V_e}{h} \tag{10}
\]

Where, $h$ is the number of experts.
e) Determination of the Fuzzy Subset Vector

The single-factor assessment matrix \( R_j \) and the weight vector \( A_j \) of the three-level index set \( U = \{ U_{ik} \} \) are used to do the fuzzy comprehensive assessment. The fuzzy integrated assessment vector of the three-level indexes is attained and normalized (”~” indicates normalization, same below).

\[
P_{ij} = A_{ij} \cdot R_{ij} \sim (b_{ij}^e), (i = 1, \ldots, n; j = 1, \ldots, m; e = 1, 2, \ldots, f)
\]

Where, \( p_{ij}^e \) is the value of the affiliation function of the comprehensive evaluation \( P_{ij} \) taken at the \( e \)-th comment. The fuzzy operation ”\( \cdot \) ” here uses the \( M(\cdot, \wedge) \) operator. That is,

\[
p_{ij}^e = \wedge(a_{ij}^e, r_{ij}^e), (i = 1, \ldots, n; j = 1, \ldots, m; e = 1, 2, \ldots, f)
\]

where \( \wedge = a + b - a \cdot b \).

In the same way, the comprehensive evaluation of the second-level indicator set and the first-level indicator set is done again. Eventually, the single-factor assessment matrix \( R \) and the weight vector \( A \) of the first-level indicator \( B = \{ B_i \} \) are used to do the fuzzy comprehensive assessment, and the comprehensive assessment vector \( P \) of the first-level indicator is attained and normalized; according to the above model, the fuzzy comprehensive evaluation results of human resources competitiveness of high-tech enterprises \( P_{ij}, P_i, P \) can be obtained, which are the fuzzy subset vectors on \( V \).

\[
\begin{aligned}
P_{ij} &= A_{ij} \cdot R_{ij} = (P_{i1} \cdots P_{im}) \\
P_i &= A_i \cdot R_i \\
P &= A \cdot R
\end{aligned}
\]

Thus, the results in assessing the HR competitiveness of high-tech enterprises are obtained.

III. RESULTS

A. Experimental Setup

To confirm the applied impact of the HR competitiveness evaluation method of high-tech companies according to the self-organized data mining algorithms studied in this article in the evaluation of the human resource competitiveness of actual high-tech enterprises, eight high-tech enterprises in a first-tier city are used as the research objects. The overview of the research objects is shown in Table II.

The human resource competitiveness of each research subject is evaluated using the methodology of this paper, and the outcomes attained are shown hereby.

B. Info Clustering Results

In this paper, the fuzzy clustering algorithm is used to obtain the initial evaluation index system by clustering the research object’s human resource-related data. Assuming that several human resource-related data points are random distribution, as shown in Fig. 1(a), where the noise points occupy 8% of the data, the number of clusters in this paper is set to 5. The clustering results of human resource-related data obtained by the method in the paper are shown in Fig. 1(b).

Analysis of Fig. 1 shows that the boundaries of various evaluation elements of the evaluation of HR competitiveness of high-tech enterprises are not obvious and have certain fuzziness. The method in this paper applies the principle of cluster analysis that “the indicators with similar properties are classified into one category, and the indicators with large differences are classified into different categories, so that the indicators within the category have high homogeneity, and the indicators between the categories have high heterogeneity,” which can better shield the impact of noise and outlier data, and accurately reflect the spatial set characteristics of HR-related data. The initial assessment indicator arrangement of HR competitiveness of high-tech enterprises has been constructed, which has a positive impact on improving the application efficiency of human resource-related data.

C. Selection of Key Indexes

The method in this paper uses self-organized data mining algorithms to identify key attributes related to the competitiveness of high-tech enterprises on the basis of the initial evaluation index system of high-tech enterprise's human resource competitiveness, thus generating the final used evaluation index system for high-tech enterprise's human resource competitiveness, as shown in Table III.

In order to analyze the scientific of the evaluation index system of HR competitiveness of high-tech companies finally used, the principle of data variance describing the information content of evaluation indexes is used as the basis to set the analysis criteria for the scientific of evaluation index system construction. \( In \) and \( S \) are taken as the information contribution rate of the filtered evaluation indexes to the initial evaluation indexes and the covariance matrix of the evaluation index data, respectively, then.

\[
In = \frac{trS_s}{trS_h}
\]

In Formula (13), \( tr \) is the trace, and \( S \) and \( h \) are the number of filtered evaluation indexes and the number of initial evaluation indexes, respectively.

Formula (13) can describe the ratio between the overall variance of the screened evaluation indexes and the overall variance of the initial evaluation indexes so that the information on the initial indexes described by the screened evaluation indexes can be obtained. In general, if the screened indexes can reflect more than 90% of the initial indexes, it means that the evaluation index system constructed by the screened evaluation indexes is scientific.
The overall variance of the assessment indicators screened by way of this paper and the overall variance of the initial 32 evaluation indexes are brought into Formula (13) to obtain the results of the scientific analysis of the evaluation index system constructed by way of this article, as presented in Fig. 2.

Analyzing Fig. 2, we can get that after identifying the key attributes of the evaluation indexes in the initial evaluation index system, and the information contribution rate reaches more than 95\% when the number of evaluation indexes reaches 17, which indicates that the assessment indicator setup constructed by the method of this paper has high scientific. Although the information contribution rate also increases under the condition that the number of evaluation indexes continues to increase, the complexity and redundancy of the evaluation index system also increase.

**D. Evaluation Results**

Taking enterprise A as an example, the evaluation index weights are calculated using this paper's method, and the results are shown in Fig. 3.

Analyzing Fig. 3, it can obtain that the weight of each level of the evaluation index is calculated using the method of this paper, and the highest weight of human capital power (B1) reaches 0.26. The lowest weight of human resource investment competitiveness reaches 0.15 (B4).

Fig. 4 shows the evaluation results of the methods in this article, studies [7], and [8] for each research object, as well as the comparison between the evaluation results of this article's method and the actual human resource competitiveness of each research object.

Analyzing Fig. 4, it can be seen that the human resource competitiveness of the selected research object does not have significant advantages and is generally at a moderate level. There is a significant deviation between the evaluation results obtained using the methods of studies [7] and [8] and the actual human resource development potential of the research object. This method can effectively obtain the evaluation results of the human resource competitiveness of the research object, and the evaluation results obtained are basically consistent with the actual human resource development potential of the research object. The major cause for the variance in the assessment outcomes of enterprise B maybe since the main direction of enterprise B is new materials and application technology and advanced manufacturing technology. The two technologies are related but relatively opposed to each other, so there is a certain duplication and redundancy in acquiring human resource information, which causes a certain deviation in the final evaluation results. Still, this deviation is within a manageable range. The above data fully demonstrate that the method of this paper can evaluate the competitiveness of HR of high-tech enterprises more accurately.

### TABLE II. OVERVIEW OF RESEARCH OBJECTS

<table>
<thead>
<tr>
<th>Research object</th>
<th>Survey</th>
</tr>
</thead>
<tbody>
<tr>
<td>Enterprise A</td>
<td>The main research direction is electronics and information technology</td>
</tr>
<tr>
<td>Enterprise B</td>
<td>Main research directions are new materials and application technology and advanced manufacturing technology</td>
</tr>
<tr>
<td>Enterprise C</td>
<td>The main research direction is modern agricultural technology</td>
</tr>
<tr>
<td>Enterprise D</td>
<td>The main research direction is aerospace technology</td>
</tr>
<tr>
<td>Enterprise E</td>
<td>The main research directions are new technologies for environmental protection and new energy and efficient energy-saving technologies</td>
</tr>
<tr>
<td>Enterprise F</td>
<td>The main research direction is nuclear application technology</td>
</tr>
<tr>
<td>Enterprise G</td>
<td>Main research directions are bioengineering and new medical technology</td>
</tr>
<tr>
<td>Enterprise H</td>
<td>The main research direction is marine engineering technology</td>
</tr>
</tbody>
</table>

(a) Random distribution of HR related data points.
(b) Clustering results.

Fig. 1. Clustering results of human resource-related data.
### Table III. The Final Evaluation Index System Of Human Resources Competitiveness of High-tech Enterprises

<table>
<thead>
<tr>
<th>Target layer</th>
<th>First-level indexes</th>
<th>Secondary indexes</th>
<th>Tertiary indexes</th>
</tr>
</thead>
<tbody>
<tr>
<td>Human capital (B1)</td>
<td></td>
<td>Number of human resources (B11)</td>
<td>Total number of employees</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td>Proportion of R&amp;D personnel</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td>Proportion of personnel with master's degree</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td>Proportion of personnel with doctor's degree</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td>Proportion of personnel with senior professional titles</td>
</tr>
<tr>
<td>Attractiveness of HR and environment (B2)</td>
<td></td>
<td>Perfection of human resources market (B21)</td>
<td>Human resource flow system</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td>Human resources entrepreneurship service satisfaction index</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td>Per capita disposable income</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td>Contribution rate of high-tech industry</td>
</tr>
<tr>
<td>HR policy incentives (B3)</td>
<td>Attraction of HR policy (B31)</td>
<td></td>
<td></td>
</tr>
<tr>
<td></td>
<td>Incentive degree of human resources policy (B32)</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Human resources investment competitiveness (B4)</td>
<td>Perfection of education and training system (B41)</td>
<td>Training input index</td>
<td></td>
</tr>
<tr>
<td></td>
<td>Investment in innovation (B42)</td>
<td>Funds for technical development activities</td>
<td></td>
</tr>
<tr>
<td></td>
<td></td>
<td>Per capita scientific research funds</td>
<td></td>
</tr>
<tr>
<td>Human resources performance demonstration (B5)</td>
<td>Scientific and technological achievements (B51)</td>
<td>Independent intellectual property rate technical income</td>
<td></td>
</tr>
<tr>
<td></td>
<td>Economic performance (B51)</td>
<td>Per capita GNP</td>
<td></td>
</tr>
<tr>
<td></td>
<td></td>
<td>Per capita profit and tax</td>
<td></td>
</tr>
</tbody>
</table>

![Fig. 2. Scientific analysis results of evaluation index system.](image)

![Fig. 3. Calculation result of weight of first-level evaluation index.](image)
E. Discuss

In today's fiercely competitive business environment, the human resource competitiveness of high-tech enterprises is crucial for innovation and sustainable development. However, accurately assessing and measuring human resource competitiveness has always been a challenging task. Traditional evaluation methods often rely on subjective judgment and limited data, making it difficult to fully reflect the actual situation of the enterprise. Therefore, the evaluation method based on self-organizing data mining algorithm provides a more scientific and objective way for relevant research to evaluate the human resource competitiveness of high-tech enterprises.

The advantage of self-organizing data mining algorithm lies in its ability to automatically discover and select key evaluation indicators from a large amount of data, without the need for prior assumptions or manual intervention. This method can help identify factors that play a crucial role in human resource competitiveness, which may be overlooked or less noticeable in traditional methods. Through self-organizing data mining algorithms, key indicators that have a significant impact on the human resource competitiveness of high-tech enterprises can be extracted from massive data, providing a more accurate and comprehensive basis for evaluation.

In addition, evaluation methods based on self-organizing data mining algorithms can provide more reference and decision support. By analyzing and mining the selected key indicators, we can gain a deeper understanding of the correlation and degree of impact between these indicators. This in-depth analysis can help understand the complex relationships in human resource management and provide more targeted improvement and optimization suggestions for enterprises. For example, by identifying a positive correlation between employee satisfaction and innovation ability, companies can place greater emphasis on improving employee satisfaction to promote innovation and enhance competitiveness.

Overall, the evaluation method for human resource competitiveness of high-tech enterprises based on self-organizing data mining algorithms has obvious advantages and potential benefits. It can automatically discover and select key indicators from a large amount of data, provide comprehensive, objective, and accurate evaluations, and provide direction for improvement and optimization for enterprises. However, when applying this method, attention needs to be paid to issues such as data quality, algorithm selection, and comprehensive analysis of results. By overcoming these challenges and limitations, we can better evaluate the human resource competitiveness of high-tech enterprises, improve their competitiveness and sustainable development capabilities.

IV. Conclusion

This paper studies the assessment method of HR competitiveness of high-tech enterprises based on a self-organized data mining algorithm that constructs an evaluation index system by using a fuzzy clustering algorithm and self-organized data mining algorithm and uses multi-level comprehensive fuzzy evaluation method to comprehensively evaluate the human resource competitiveness of high-tech enterprises, so as to determine the level of human resource competitiveness of high-tech enterprises, thereby judging the strength and weakness of human resource competitiveness of high-tech enterprises and discovering the defects of high-tech enterprises, which can provide a reference for measuring and improving the competitiveness of high-tech enterprises' human resources, and provide a reference for promoting the comprehensive, all-round and coordinated development of high-tech companies' HR competitiveness. With the arrival of the era of Big data, more and more data exist in unstructured forms, such as text, image and audio data. Future research can explore how to use Natural language processing, image recognition, sound analysis and other technologies to incorporate these unstructured data into the evaluation model to obtain more comprehensive evaluation results.

REFERENCES


A Novel Internet of Things-enabled Approach to Monitor Patients’ Health Statistics

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Abstract—Leveraging Internet of Things (IoT) technology in healthcare systems improves patient care, reduces costs, and increases efficiency. Enabled by IoT, telemedicine allows remote patient monitoring, vital sign tracking, and seamless data accessibility for doctors across multiple locations. This article presents a novel IoT-enabled approach that utilizes artificial neural networks with radial basis functions to detect patients’ positions. This real-time tracking mechanism operates even without cellular connectivity, providing timely diagnoses and treatments. Our research aims to develop a smart and cost-effective healthcare approach, revolutionizing patient care. Mathematical analysis and experiments confirm the effectiveness of our proposed method, particularly in predicting patient location for the upcoming smart healthcare solution.

Keywords—Internet of things; healthcare; telemedicine; artificial neural network

I. INTRODUCTION

In today's world, with the ever-increasing advancement of technologies, sharing data related to old systems is cumbersome and often weak. For example, sharing health data with others in old systems is difficult due to different formats and parameters and is not suitable for the urgent needs of modern users [1]. In addition, the relationship between healthcare experts and patients has always been a one-way path. For patients and experts, the current system is incredibly slow, inflexible, and unclear. These problems are equally evident in all stages. When a patient needs services, health plans are used to determine the amount of money they will pay. To determine this fee, as opposed to the agreement between the patient and the health plan, the health plan must validate the services received from the provider and then share the results with the provider [2]. This only happens if the application provider is online. For a provider to be considered online, a complex agreement must be negotiated, which adds significant overhead to the provider's administrative costs. Part of these costs are related to billing and insurance costs, which include activities such as updating the database and maintaining records of the services provided [3]. Normally, this whole process takes between one to two weeks if done electronically and three to five weeks if done on paper. In addition, this process is full of spots where it is possible that an error can occur. In order for sponsorship to actually take place, many people have to check multiple old agreements compared to multiple records. The result is an ineffective and unclear process that causes profiteers and patients to feel confused and pessimistic [4].

Recent advancements in wearable health technologies have opened new avenues for telemedicine and have made it possible to monitor one's health remotely and continuously. These sensors can collect a wide range of data, including heart and respiratory system data and several more vital signals. Using artificial intelligence and signal processing techniques, these signals can be automatically analyzed and categorized. In the event of detecting unusual signal patterns, notifications are promptly sent to both patients and doctors. This innovative approach to telemedicine provides a valuable and rewarding experience for individuals needing medical assistance [5]. Internet of Things (IoT), humanitarian logistics, meta-heuristic algorithms, nonlinear complementarity, silicon nanostructure arrays, machine learning, Markov-modulated regime-switching market, and artificial intelligence play crucial roles in the field of telemedicine, revolutionizing healthcare delivery and improving patient outcomes.

The integration of IoT in telemedicine allows for remote patient monitoring, real-time data collection, and seamless communication between healthcare providers and patients. This connectivity enhances the accessibility and effectiveness of healthcare services, especially in remote or underserved areas [6-8]. Humanitarian logistics leverages technology to optimize the delivery of medical supplies and resources during emergencies or humanitarian crises. It ensures timely and efficient distribution, minimizing supply chain disruptions and saving lives [9]. Meta-heuristic algorithms provide efficient solutions to optimization problems in telemedicine, helping healthcare providers in resource allocation, scheduling, and decision-making. These algorithms can optimize complex systems and improve the utilization of healthcare resources [10, 11]. Nonlinear complementarity and silicon nanostructure arrays are emerging technologies with potential for telemedicine. They enable advanced diagnostics, drug delivery systems, and personalized medicine, enhancing the accuracy and effectiveness of medical treatments [12, 13]. Machine learning and artificial intelligence enable predictive analytics, risk assessment, and decision support systems in telemedicine [14-16]. They can analyze vast amounts of patient data, identify trends, and assist healthcare professionals in making informed decisions [17-21]. The markov-modulated regime-switching market model provides insights into the dynamics of telemedicine markets, helping stakeholders understand market behavior and optimize healthcare service provision [22].

Furthermore, telemedicine makes medical services more available at lower costs by using modern communication tools. Additionally, telemedicine virtually provides easy access to medical services for people in rural areas [23]. This easy access cannot be limited to just one country, and different people in different countries can use these medical services. Doctors and

*Corresponding Author
patients can communicate with each other online using computers or smartphones. Doctors can check the patient’s file and test results remotely. Patients also visit their doctor at their homes and receive diagnoses and treatment through telemedicine software without waiting for an appointment. Easy and reliable use, smart characteristics, no time loss, high security, and low cost are some advantages of telemedicine [24]. Although telemedicine is still a new concept for users and doctors, advances in technology and medical innovation have expanded its use. Despite the provision of many advantages through this technology, its use is expected to increase in the coming years. “Telemedicine” includes maintenance, care, diagnosis, consultation, and treatment, while at this stage, the focus is on transferring medical data and educational goals. Also, in a more comprehensive and complete view, it can be said that, in general, “telemedicine” is the use of medical and communication technologies to exchange any data, including data, voice or video communications between a doctor and a patient or a doctor and healthcare experts in separate geographical locations as well as creating the possibility to exchange medical, healthcare, research and educational concepts [25].

The convenience of using remote medical services, low waiting time, high-quality medical diagnosis and treatment, and cost reduction are among the advantages of this technology. Also, the presence of the patient’s medical record online makes it possible to diagnose and treat the disease more accurately and with higher quality. Another advantage is the lower costs of telemedicine compared to the traditional method. With this method becoming common, insurance companies also will include telemedicine services in their insurance plans, and this process will lead to a reduction in the costs of telemedicine services. Telemedicine services only require a webcam (or smartphone) and a secure online system to connect the patient with the doctor and store medical records. In addition to ensuring the patient’s privacy and safe keeping of medical records, the professional competence of doctors should also be ensured through the examination of documents [26]. Moreover, the challenges facing the current medical care system, which is often accompanied by cumbersome laws that slow down medical services, can include data dispersion in the health system, patient disobliging, misdiagnosis, and false medical data, security risks for patient data, and lack of transparency. Researchers and healthcare professionals are faced with a number of new challenges as they attempt to understand and utilize new developments in the field of information technology. Wireless communication platforms and their application to the development and interconnection of smart devices have revolutionized the landscape where the first electronic healthcare systems were conceived. The use of network-enabled devices has become widespread, covering a wide range of items from household items and cars to health and care management systems in what has been referred to as IoT [27].

The purpose of modern information and communication technologies (ICTs) in the healthcare system is traditionally to provide promising solutions that facilitate the efficient delivery of healthcare services, referred to as e-health, including personalized diagnostic, telemedicine, and electronic record tools. In the developed world, however, the rapid increase in longevity has led to the fact that an increasing proportion of the population is over the age of 80. Consequently, traditional healthcare systems need to be designed and developed in a more cohesive and ubiquitous manner in order to provide excellent patient-centered services [28]. In recent years, the emergence of wearable gadgets and smartphones has enabled IoT technology to revolutionize healthcare by changing a traditional hub-based model to a personalized one. With the effective implementation of IoT into customized healthcare services, preventive care becomes timelier and safer, costs are reduced, and patient-centered care is improved. As IoT becomes increasingly prevalent, highly customized healthcare services will be able to provide enhanced and customized access to extensive healthcare information and facilitate clinical decision-making for each patient through the use of unobtrusive, continuous monitoring and sensing [29]. We conduct research on the remotely monitored healthcare system as it plays a vital role in the medical field. A change in lifestyle is also leading to new diseases appearing in the modern age. Further, the COVID-19 epidemic shows the global community our deficiencies regarding healthcare services. A large number of people need health care services at the same time, and adequate resources, equipment, and space are required. Remote health monitoring offers several advantages for elderly and ill patients who are unable to visit medical facilities for periodic checkups. This will ultimately improve the efficiency of the entire healthcare system and allow critical patients to receive the necessary care. The main contributions of the paper are centered around addressing the energy efficiency and decision problems related to location tracking for remote health monitoring. The specific contributions can be summarized as follows:

- Problem identification and analysis: The paper identifies and analyzes the energy efficiency and decision challenges associated with location tracking in the context of remote health monitoring. By understanding these challenges, the researchers lay the groundwork for developing solutions that optimize energy consumption and improve decision-making processes.

- Development of a location tracking system: The researchers design and develop a system that is capable of collecting patients’ location data for remote health monitoring. This system incorporates a mobility management model to efficiently track the patients’ movements and gather relevant location information.

- Feature extraction based on movement: In order to accurately predict the patients’ location, the paper introduces a novel approach for feature extraction. By analyzing the movement patterns of the patients, relevant features are extracted from the location data. This step is crucial for improving the accuracy of location prediction.

- Utilization of radial basis function neural network: The paper proposes the use of radial basis function neural network as a learning algorithm to process the extracted features and predict the patients’ location.
• Enhanced prediction of patient location: The combination of the mobility management model, feature extraction, and radial basis function neural network learning enables an efficient and accurate prediction of patients' locations. This advancement in location tracking facilitates seamless information gathering of patients' health data in remote monitoring scenarios.

II. RELATED WORK

A mobility-aware, IoT-enabled healthcare security framework has been proposed by Moosavi, et al. [30]. The system consists of three main components: interconnected gateways that provide robust mobility, an ad-hoc protocol that supports resuming sessions, and a certificate-based DTLS handshake protocol for user authentication and authorization. Smart gateways serve as an intermediary computing layer between cloud services and IoT devices. With the computing layer, there is no need to reconfigure the device for ubiquitous mobility. The framework is demonstrated using simulations and a full prototype of both hardware and software. Compared to existing approaches, this framework significantly reduces the delay between end users and smart gateways by 17% and the amount of communication overhead by 25%.

Data generated by medical devices with sensors is often referred to as big data, which is a combination of structured and unstructured data. The complexity of the data makes it difficult to analyze and process big data in order to find relevant information to aid decision-making. Security of data is an essential requirement for healthcare big data systems. This issue was addressed by Manogaran, et al. [31] by creating a new architecture for incorporating the IoT into healthcare for storing and processing scalable sensor data (big data). The suggested architecture is based on two main underlying models, namely Grouping and Choosing (GC) and Meta Fog-Redirection (MFR). MFR collects and archives sensor data (big data) generated by a variety of sensors. The proposed GC architecture integrates fog computing and cloud computing. MapReduce-based prediction models are also used to predict heart disease. Based on performance measures such as f-measure, precision, robustness, and efficiency, the proposed architecture and prediction model are evaluated.

Pal, et al. [32] have proposed an IoT-based system for controlling access to constrained healthcare resources. Using this approach, trusted users will have access to the services while unauthorized users will not be able to access valuable resources. The authorization design utilizes attributes, roles, and capabilities in a hybrid manner. Role membership is assigned based on attributes, and permissions are evaluated based on attributes. Capabilities are granted by membership in roles. User capabilities may be granted according to additional attributes when accessing specific services provided by IoT devices. Consequently, the number of policy instances necessary for specifying access control criteria is significantly reduced. It is implemented and evaluated based on an initial proof-of-concept. There was minimal additional overhead associated with this approach when compared to other solutions that incorporate access control functions within the Internet of Things.

IoT technology has revolutionized the healthcare industry, resulting in smart medical applications. A system for monitoring urine-based diabetes (UbD) at home has been presented by Bhatia, et al. [33]. In order to predict and monitor diabetes-related urinary tract infections, a four-layer system is proposed. Diabetes measures are regularly tracked and a prediction procedure is conducted to enable precautionary steps to be taken at an early stage of the disease. Furthermore, a Recurrent Neural Network (RNN) was used to obtain a probabilistic measure of UbD monitoring by calculating the Level of Diabetic Infection (LoDI), a measure of diabetes infection. Simulation results demonstrated that the proposed system was more accurate, robust, and consistent than state-of-the-art decision-making techniques.

In the design of any information and communication technology (ICT) infrastructure, interoperability and connectivity play an important role. By integrating IoT technologies into medical systems, people will be able to receive timely, cost-effective, and ubiquitous healthcare services. Currently, a complex ecosystem of devices, databases, and communication technologies provides a wide range of healthcare services. Healthcare information systems must integrate and collaborate with these technologies and resources to ensure that healthcare is provided efficiently. By providing connectivity for billions of devices worldwide, IoT technology improves healthcare delivery and quality. The potential benefits of IoT-based connectivity for healthcare are discussed by Zeadally and Bello [34]. An overview of recently implemented IoT-based healthcare systems is provided. Lastly, they discuss the potential uses of the IoT for enhancing healthcare solutions.

A new IoT-based reliable healthcare monitoring approach was proposed by Jacob, et al. [35]. To collect vital parameters, such as deviations in body temperature, multiple sensor nodes are implanted into the body of a patient as a first step. After preprocessing the dataset in order to eliminate redundant and irrelevant attributes, the normalization procedure is applied. Cancers are classified based on their features using a convolutional neural network (CNN). Based on sensor input, the CNN model calculates the cancer risk of a patient. Once the results have been received by the hospital management, they are analyzed. Rivest-Shamir-Adleman (RSA) encryption is primarily used in this study in order to achieve high assurance security, simplicity of deployment, and simplicity of use. A modified version of the RSA algorithm, utilizing the double encryption-decryption process and n prime numbers, is presented in this paper. Based on experimental results, the proposed method demonstrated superior performance when compared to other approaches.

III. PROPOSED METHOD

A. Network Model

System performance and network architecture are critical to the success of an IoT-based healthcare system. The ability to process large amounts of data quickly and accurately is essential for IoT-based healthcare systems to work properly and provide accurate diagnoses and treatments. Additionally, the network architecture of an IoT-based healthcare system must be robust enough to handle the large amounts of data
being transferred between various devices and systems. By placing sensors in the patient's body and transmitting the data to the monitoring end, various processes are undertaken, which require careful observation of the locations of the sensors, the hardware configurations, and the control of the energy usage of the monitoring devices. An industry-specific IoT environment incorporating significant parameters was used to design the network architecture for optimal data transmission and energy consumption according to the patient or node locations.

Assumptions and approximations are outlined below.

- Sensor nodes operate continuously and communicate via cellular networks.
- According to the Rayleigh distribution, the signal magnitude varies randomly or fades.
- Rayleigh fading channels are considered throughout the communication process.
- Continuous sensing, transmission, and reception are characteristics of sensing operation.
- Patients' bodies are continuously monitored by heterogeneous sensors that collect data such as temperature, blood sugar, etc.
- Patients are equipped with sensors, and they are mobile.

As illustrated by Fig. 1, the present design comprises a mobility management strategy for collecting location information, followed by a radial basis function neural network-based learning method used to predict the node's location to ensure uninterrupted communication. Fig. 2 illustrates the network architecture. In this system, mobility management is used to determine the patterns of movement of nodes as a set of locations. As shown in Fig. 2, these data are used to train the proposed model with location feature maps for each time instant. Based on the training output, nodes or patients' positions are predicted and localized. Localized nodes collect observations estimated from the previous data gathered over time.

B. Mathematical Modeling

The proposed system begins with tracking patient location information. Thus, a patient movement scheme is designed to estimate at first the positions of N patients based on their movements. The number of locations (n) is assumed to be fixed in proximity to the individual's residence or the locations he/she frequently visits. It is also assumed that there are m unknown places that must be investigated to determine how patients move two-dimensionally. Throughout the range of transmission radius R, sensing devices can communicate omnidirectionally with each other. The direction and speed experienced by a patient from any fixed position can be used to determine their relative locations. Sensor location changes over time are primarily a function of rapid node movements, and displacement is measured as a Gaussian random variable with mean 0 and variance 1. The following equations can be used to illustrate the randomness of the movement:

\[ u(r) = \eta u(r - 1) + (1 - \eta)\phi_u + \gamma u\sqrt{1 - \eta^2}h_u(r-1) \]  

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In the above Equations, \( u(r) \) represents time-dependent displacement, whereas \( d(r) \) represents displacement direction concurrent with the movement of the sensor nodes. In this situation, the probability of the sample depends on \( \eta \) ranging from 0 to 1 and on the proportional average speed based on the scale factor of \( r \rightarrow \infty \).

C. Pattern Definition and Feature Assignment

The random movement of patients makes it difficult to recognize certain patterns. Patients move from one location or provider to another, and their care may be fragmented, making it hard to determine a consistent care pattern or evaluate the effectiveness of interventions. Additionally, patients may not have the same healthcare access, making it difficult to identify patterns. However, using historical location records, relationships are formed, and feature mapping is conducted. This feature mapping can then be used to identify potential gaps in care and areas where healthcare providers can intervene to improve patient outcomes. It can also help reduce the burden of fragmented care by helping providers identify the most effective treatments. In a high-dimensional space, locations of patient i can be mapped at different time intervals t, as follows:

\[ L_{x_i} = \{x_i(1), x_i(2), ..., x_i(r)\} \]  

In a g-dimensional space, patients' positions can be mapped separately in the x- and y-axes. By plotting points on the graph in the x- and y-axes, it is possible to map out the position of each patient within the given space. This allows for more accurate tracking and analysis of the patient's movements over time. Thus, both feature spaces can be expressed as follows:

\[ W_{x_i} = \{W_{x_i}(1), W_{x_i}(2), ..., W_{x_i}(r - g + 1)\} \]  

\[ W_{y_i} = \{W_{y_i}(1), W_{y_i}(2), ..., W_{y_i}(r - g + 1)\} \]  

The feature vectors assigned to each patient are bound in the g dimension and stored in feature spaces WX_i and WY_i, respectively. These features provide sensor movement characteristics and can be used as training data for predicting a sensor's future location.

D. Feature Learning

Machine learning techniques are the most effective in terms of optimizing and approximating functions accurately, making them a valuable tool for prediction. Among the various machine learning approaches used to train systems, the radial basis function neural networks are unique in their ability to approximate functions and classify them. As shown in Fig. 3, the radial basis function neural networks are able to linearize nonlinear data.

Fig. 3 illustrates an architecture in which three layers are present: input, hidden, and output. The input layer takes the input data and passes it on to the hidden layer. The hidden layer processes the input data with mathematical operations and passes the processed data to the output layer. The output
layer produces the desired output, such as a prediction or classification. A radial basis function is used in the hidden layer to convert the nonlinear to linear data. In most cases, nonlinear kernels are used as Gaussian kernels in radial basis function neural networks based on Euclidean distance. The radial basis function neural network is trained to learn the input characteristics in the form of \( v = 1, 2, \ldots, (r - t(g - 1)) \), which are output components to be \( Q_{x(t)} \) and \( Q_{y(t)} \).

E. Place Estimation

The last step in our approach involves obtaining predictions based on our assumptions. The predicting procedure is initiated, and the position is estimated if the mobility management solution cannot identify the point of interest because of excessive communication interference. At first, the model considers the initial coordinates of the last place and searches for the next possible location. During the prediction process, this assumption was based on the non-availability of the model for mobility management. According to our assumptions, \( m \) is the predicted number of locations or steps associated with our experiment and \( V_p(p) \) denotes an input vector for \( p = 1, 2, \ldots, m \). Suppose that the output results for \( P = X, Y \) during the prediction will be \( S_p(p) \). Therefore, it can be calculated as follows:

\[
S_p(p) = \sum_{j=1}^{r} k_j \phi(V_p(p), c_j)
\]  

In other words, with respect to \( S_p(p) \), the anticipated position \( i \) individual can be shown by:

\[
\hat{L}_i(r + p) = \hat{x}_i(r + p), \hat{y}_i(r + p)
\]

In this manner, the latest position corresponds to the following location on the axis of \( \hat{x}_i(r + p), \hat{y}_i(r + p) \). Neurons are updated periodically in order to predict future locations in accordance with certain rules. The incoming points for the \( p \)th vector are expressed matrixial in the following way.

\[
V_x(p) = \begin{bmatrix} x_1(r - t(g - 1)) \ldots x_i(r) \end{bmatrix}
\]

\[
V_y(p) = \begin{bmatrix} y_1(r - t(g - 1)) \ldots y_i(r) \end{bmatrix}
\]

This is our setup for converting high-dimension states into low-dimension ones in order to predict them. The prediction outcome is as follows:

\[
\hat{L}_i = \{\hat{L}_i(r + 1), \hat{L}_i(r + 2), \ldots, \hat{L}_i(r + m)\}
\]

In order to determine the prediction error, the actual number of patient places traveled by the mobility management scheme can be compared to the prediction error. According to the prediction-based algorithm, known locations can be expressed as follows:

\[
L_i^0 = \{L_i^0(r + 1), L_i^0(r + 2), \ldots, L_i^0(r + m)\}
\]

In this regard, the expected error value can be calculated as follows:

\[
e(p) = \frac{1}{N} \sum_{i=1}^{N} (L_i^0 - \hat{L}_i)^2
\]  

Fig. 1. Flowchart of the proposed method.
This section presents the numerical analysis and provides a detailed analysis of the experimental findings to evaluate the effectiveness of the proposed mechanism. The experiments were conducted using a MATLAB simulator, and Table I outlines the parameters used for the simulation tests. The dataset consisted of 2400 data points, capturing position information for 600 locations over different time periods, which were used for both training and testing purposes. To assess the accuracy of the suggested technique, an average error was calculated, as depicted in Fig. 4. In the traditional method, a simple range error is employed as the basis for the mobility management model. As the number of prediction steps increases, the average error becomes more pronounced. The error rate rises when the number of prediction steps is reduced and when an individual remains outside the range of the cellular network for an extended period. However, our proposed method demonstrates fewer errors compared to conventional methods. Unlike the traditional approach, which relies on a fixed number of prediction steps and exhibits increased error rates when the patient is out of range for longer durations, our method leverages machine learning algorithms to better predict the patient’s movements, resulting in a reduction in the average error rate.

### Table I. Simulation Parameters

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Network dimensions</td>
<td>80 x 80 m²</td>
</tr>
<tr>
<td>Number of simulations</td>
<td>500</td>
</tr>
<tr>
<td>Simulation time</td>
<td>1000 seconds</td>
</tr>
<tr>
<td>Number of prediction steps</td>
<td>20</td>
</tr>
<tr>
<td>Range of sensors</td>
<td>30 m</td>
</tr>
<tr>
<td>Locations for training</td>
<td>500 locations</td>
</tr>
</tbody>
</table>
Furthermore, the proposed method is compared to a conventional mobility management model in Fig. 5. The primary objective of our research is not only to predict patients’ locations but also to monitor their health with precision. Therefore, it is crucial to verify that the developed system can obtain accurate and timely patient data to a significant extent. Fig. 5 illustrates that our prediction model provides additional data and information compared to traditional approaches. As a result, the proposed approach enables more accurate and real-time prediction of patient positions.

Additionally, Fig. 6 shows the minimal response time achieved by our proposed algorithm compared to the conventional method [36]. The results demonstrate that our algorithm can handle a greater number of requests in less time.

This improved performance is attributed to the efficient utilization of resources and optimal request routing. The use of at least 10 prediction steps and a minimum time interval of one second between locations contribute to the algorithm's superior performance. It is important to note that modifying these parameters may result in a smoother transition between the performance of the proposed method and the traditional approach.

The experimental results provide quantitative evidence of the proposed mechanism's effectiveness in terms of efficiency, accuracy, and response time. The findings highlight the superiority of our approach over conventional methods, emphasizing the potential impact and benefits it can offer in the field of healthcare monitoring and patient management.
V. CONCLUSION

In this paper, a neural network-based radial basis function algorithm is presented for the prediction of a patient's location in a remote healthcare system. The movement characteristics of the patients are trained using a radial basis function neural network, and their locations are predicted whenever they move outside of their range of cellular devices. The training of the system is based on a mobility management model. Simulated results indicate that the proposed method is faster and consumes less energy than the existing method. Also, the proposed work shows good results regarding the average error for predicting the position. As a result, the proposed system can reliably and accurately track the locations of patients while maintaining low energy consumption and fast response times. This makes the proposed system an ideal choice for applications in healthcare and monitoring patient mobility, as it offers optimal performance. While radial basis function neural networks have faster training times, their classification process can be slower compared to multilayer perceptron networks. This is primarily due to the computation of the radial basis function in each node of the hidden layer before the input sample vector can be classified. To overcome this limitation and enhance the system's ability to learn new categories and update the classification system in real-time, we propose incorporating transfer learning in future work. Transfer learning is a technique that utilizes pre-trained models as a starting point for learning new features while preserving the existing knowledge. By leveraging transfer learning, we intend to take advantage of the knowledge acquired by the back-end network of the radial basis function neural network. This approach allows for a more efficient adaptation of the network to new categories without disregarding the valuable information that has already been learned.

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REFERENCES


Automated Modified Grey Wolf Optimizer for Identification of Unauthorized Requests in Software-defined Networks

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Abstract—Software Defined Networking (SDN) is utilized to centralize network control within a controller, but its reliance on a single control plane can make it vulnerable to attacks such as DDoS. This highlights the importance of developing effective security mechanisms and using proactive measures such as detection and prevention strategies to mitigate the risk of attacks. Many DDoS attack detection technologies within SDN focus on detecting and mitigating the attack once it has occurred in the controller, which leads to more seconds of exposure, diminished precision, and high overhead. In this work, we have developed an Automated Modified Grey Wolf Optimizer Algorithm (AMGWOA) to design the detection of this malicious activity in an SDN environment to prevent the attack in the controller. Our methodology involves the development of the AMGWOA, which incorporates a mechanism to facilitate the blocking of malicious requests while reducing detection time and minimizing the use of storage and data resources for detection purposes. The results obtained show that our model performs well, with an ability to minimize a very large number of malicious requests in a minimum of time of less than 1 second compared to Grey Wolf Optimizer and particle swarm optimization algorithms evaluated using the same datasets.

Keywords—Software-defined networks; security; DDoS attacks; metaheuristic algorithms; Grey Wolf Optimizer

I. INTRODUCTION

Prior to very recently, communication service providers (CSPs) used proprietary physical equipment and devices to carry out network activities, making security a crucial component of wireless communication systems. This typical network design does not provide a scalable and manageable solution for such big and complicated networks, and as user needs rise, more hardware devices are needed to satisfy consumer expectations [1].

With 5G and edge computing, software-defined networking (SDN), which separates the control plane from the data plane, can be used to deliver more adaptable and dynamic services across the wireless communication network [2].

Although this SDN technology has several significant advantages, such as flexibility and economical, effective administration, it also introduces new risks [3]. The SDN controller serves as the brain of the system. The entire network will be at risk if the controller is compromised, or worse, destroyed. The SDN paradigm is vulnerable to DDoS attacks from malicious users, according to a number of recent research studies [4] [5] [6]. This attack is characterized by a large number of puppet hosts controlled by the controller launching an attack on the targeted system, rapidly depleting its resources and threatening its continued operation.

When a DDoS attack affects an SDN network, the switches generate a flood of incoming packet messages for the controller to process. This places a strain on the controller’s assets, causes the switch routing table to grow and potentially compromises the integrity of the encrypted connection between the controller and the switches. This has the potential to bring the whole SDN network down.

If the DDoS happens in SDN, communication channels might be quickly blocked, and controller resources would be used up, drastically reducing service quality.

In this research, we provide a novel model based on the Grey Wolf optimization technique that acts in an automated way to prevent DDoS attacks in the SDN network.

Mirjalili [7] describes the Grey Wolf Optimizer (GWO), a novel population-based algorithm motivated by the hunting strategies of a wolf pack. While other evolutionary computation-based methods, such as particle swarm optimization (PSO), fast evolutionary programming (FEP), and the gravitational search algorithm (GSA), achieve comparable performance, GWO has the advantage of requiring fewer adjustment parameters [7].

The design of an improved algorithm based on an automated modified grey wolf (GWO) is the contribution of this paper. This modified GWO will identify the most malicious requests and facilitate blocking them by the SDN Controller, which will minimize the risk of dealing with a critical DDoS attack, minimize the latency, and maintain the continuous availability of the controller for legitimate users.

This work is organized as follows: The background of the research is provided in Section II. Section III gives a brief summary of previous work. Section IV presents the design of our proposed algorithm (AMGWOA) for DDoS detection in SDN. Section V contains the experimental methodology and findings. Section VI is the concluding section of the paper.
II. BACKGROUND

A. Overview of the SDN Concept

The control and infrastructure layers of conventional networks function as a single unit. The control layer is responsible for determining the best route for data packets to take across the network, while the Infrastructure layer is responsible for carrying out those instructions. In software-defined networking (SDN), the Infrastructure and control layers are two distinct entities, with the control layer controlling several data planes. Through “softwarization,” it can centrally monitor and regulate the network. The architecture of this new software technology is shown in Fig. 1.

![Fig. 1. A three-layer software-defined networking (SDN) architecture.](image)

1) **Application layer:** It contains network applications like firewalls, load balancing, monitoring, and routing. The management plane is in charge of establishing regulations and guidelines.

2) **Control layer:** It is responsible for setting up the forwarding devices. The program that interacts with the hardware and software parts of the network in an SDN is called a controller. It is a focal point of the network since it coordinates information transfer between northbound and southbound APIs and connects the data plane and application plane.

3) **Infrastructure layer:** In the data plane, a physical network architecture is defined. Switches and routers are examples of forwarding devices, and they can communicate with one another using either wired or wireless means. In the data plane, the header, match, and actions fields constitute the main part of forwarding tables. The ternary content addressable memory (TCAM) contains the flow entries into tables for the data plane. Another name for it is a forwarding plane (FP).

4) **Northern connection:** This connection interface is called the northbound interface. Communication between the management plane and the control plane. It gives the southbound interface low-level instructions. It’s also known by its alternate name, the Management to Control Plane Interface (MCPI). As of yet, there are no agreed-upon protocols for the northbound interface [8, 9].

5) **Southern connection:** The southern interface is known as the southbound interface. It provides a means of communication between the control and infrastructure layers, made possible by a protocol called OpenFlow. The control plane and data plane are separated by the OpenFlow standard protocol for SDNs [10].

B. DDoS Attacks in Software Defined Networks

when the processing power of the network is centralized, one central point of vulnerability is created. In simpler terms, the network will fail if the state of the SDN controller is compromised or cannot fulfill the requests of the switches. The DDoS attack in SDN illustrated in Fig. 2 aims to overwhelm the target host resources in order to disrupt the benign host.

The following possible attack scenarios could be carried out by attackers:

1) **Attack on the application plane:** The attack takes place via the applications that are present in the application plane. The rogue application uses up all the resources, hurting honest users.

2) **Attack on the controller:** The controller would be able to handle all of the packet requests sent by the attacker, which would result in a malfunction. As a result, all requests from respectable users suffer.

3) **The transmission path of information transfer between the control plane and the data plane:** An attacker could attempt to assault the communication channel connecting the control and data planes by sending many Packets as requests.

4) **Attack using a table overflow:** Innocent users are harmed when an attacker utilizes phony IP sources to fill the switch flow table to its maximum capacity. As a result, a decent traffic sender will be denied access to the services.

![Fig. 2. DDOS attack scenario in SDN an example.](image)

C. Overview of the Grey Wolf Optimizer

Grey wolves tend to move and hunt in packs of 10 to 17 individuals, and this social behavior inspired GWO, a population-based metaheuristic algorithm [11],[12],[13],[14],[15]. Grey wolves have a distinct social hierarchy. Wolves at the top of the social hierarchy of a pack are called alphas. In the hierarchy, Beta, Delta, and Omega are ranked second, third, and fourth, respectively. When to get up, where to sleep, and when to go hunting are all decisions made by the Alpha.

The rest of the wolves must follow the decision of the Alpha. Beta wolf provides assistance to the Alpha wolf when
making decisions and will take over if the Alpha dies or is incapacitated.

The Beta defers to the Alpha’s decision but gives orders to lower-ranked wolves. Sentinels, scouts, elders, and caregivers are all examples of Delta wolves. The Omegas are at the bottom of the list and should be consumed last. Omega wolves take orders from all wolves. In the same pack, the delta wolf, in turn, dominates omega but follows the instructions of alpha and beta.

Grey wolves are sociable animals with many shared traits, including group hunting. Grey wolves will follow, pursue, and approach their prey first. The target will then be pursued, encircled, and harassed until it stops moving.

Wolves will then attack their prey in the final phase of the hunt.

The GWO algorithm mimics two social behaviors typical of wolves: social hierarchy and collective hunting. Each of the individual wolves represents various strategies for achieving optimal performance. Alpha (α) is the best possible response, while beta (β) is the second, and delta (δ) is the third choice. These three competitors are leading and being followed by the hunt. Every other option is assumed to be the omega (ω) solution. The wolves encircling behavior is modeled analytically using Equation (1).

\[ \vec{Y}(t + 1) = \vec{Y}_p(t) + \vec{B} \cdot \vec{E} \]  

(1)

\( \vec{Y}_p \) is the position of the prey, \( \vec{Y} \) is the position of the grey wolf, and \( \vec{E} \) is as stated in equation (2), \( t \) is the iteration number, \( \vec{B} \) and \( \vec{D} \) are coefficient vectors as defined in equations (3) and (4).

\[ \vec{E} = \| \vec{D} \cdot \vec{Y}_p(t) - \vec{Y}(t) \| \]  

(2)

\[ \vec{B} = 2\alpha \vec{r} - \vec{a} \]  

(3)

\[ \vec{D} = 2\vec{r}_2 \]  

(4)

where \( \alpha \) is reduced linearly from 2 to 0 over iterations and \( \vec{r}_1 \) and \( \vec{r}_2 \) are random vectors in \([0, 1]\). The alpha, beta, and delta are said to have superior knowledge of the likely whereabouts of prey in order to mimic the hunting behavior of grey wolves. Once the best search agents’ locations have been determined (alpha, beta, and delta), the positions of the other wolves will be updated accordingly. The wolves’ positions are updated in accordance with equation (5).

\[ \vec{Y}(t + 1) = (\vec{Y}_1 + \vec{Y}_2 + \vec{Y}_3)/3 \]  

(5)

Where \( \vec{Y}_1 \), \( \vec{Y}_2 \) and \( \vec{Y}_3 \) are defined in equations (6), (7) and (8).

\[ \vec{Y}_1 = \vec{Y}_a - \vec{B}_1 \cdot (E_\alpha) \]  

(6)

\[ \vec{Y}_2 = \vec{Y}_\beta - \vec{B}_2 \cdot (E_\beta) \]  

(7)

\[ \vec{Y}_3 = \vec{Y}_\delta - \vec{B}_3 \cdot (E_\delta) \]  

(8)

where \( \vec{Y}_a \), \( \vec{Y}_\beta \) and \( \vec{Y}_\delta \) are the positions of the first three solutions, \( \vec{B}_1 \), \( \vec{B}_2 \) and \( \vec{B}_3 \) are defined in equations (6), (7) and (8) and \( E_\alpha \), \( E_\beta \), and \( E_\delta \) are defined in equations (9), (10) and (11).

\[ E_\alpha = |\vec{D}_1 \cdot \vec{Y}_a - \vec{Y}| \]  

(9)

\[ E_\beta = |\vec{D}_2 \cdot \vec{Y}_\beta - \vec{Y}| \]  

(10)

\[ E_\delta = |\vec{D}_3 \cdot \vec{Y}_\delta - \vec{Y}| \]  

(11)

Where \( \vec{D}_1 \), \( \vec{D}_2 \) and \( \vec{D}_3 \) are as defined by equation (4). The parameter \( a \), which controls the balance of exploration and exploitation, is updated based on equation (12).

\[ A = 2 - \frac{2t}{M} \]  

(12)

where \( t \) is the number of iterations and \( M \) is the maximum number of iterations. The pseudocode for the GWO algorithm is represented by Algorithm 1.

### Algorithm 1: Grey Wolf Optimizer

1. **Input:** \( \vec{Y}_i(i=1,2,...,n) \); \( a \), \( \beta \) and \( \delta \), best wolves \( \vec{Y}_a \), \( \vec{Y}_\beta \) and \( \vec{Y}_\delta \)
2. **Output:** \( \vec{Y}_a \)
3. **while** \( t < M \)
4. **for** each wolf
5. **Update the current wolf position using equation (5)**
6. **end for**
7. **Update a, B and D**
8. **Compute the fitness of all search agents**
9. **Update \( \vec{Y}_a \), \( \vec{Y}_\beta \) and \( \vec{Y}_\delta \)**
10. \( t = t + 1 \)
11. **end while**

### III. Related Works on DDoS Detection in SDN

There has been a lot of discussion about the security risks that SDN faces. DDoS attacks are the most frequent and well-known SDN attacks. Numerous DDoS detection algorithms have been proposed thus far, but only a selected few are presented here.

#### A. The Detection Methods Based on Information Entropy

In [16] DDoS is detected by evaluating the unpredictability of incoming packets and using two elements: window size and threshold. The entropy of packets is calculated and if it goes below a threshold, the attack is detected. This strategy merely identifies the DDoS attack, but does not eliminate it.

Authors in [17] proposed a fusion entropy method. In this method, the benefits of log energy entropy and information entropy are combined to achieve complementarity. Attackers can take advantage of fusion entropy’s ease of detection and the transparency of its entropy value variations. Since it is challenging to discern between normal network traffic and low-rate DDoS attacks when they occur, this method makes it more challenging to detect low-rate DDoS assaults.

Low-rate and high-rate DDoS attacks against the controller are both detectable using an entropy-based DDoS attack detection method, which the authors of [18] and [19] assess in terms of detection rate (DR) and false-positive rate (FPR),

[86]
as well as whether the attacks originate from a single host, multiple hosts, or both. Eight different scenarios were tested, each representing a different level of traffic rate during a distributed denial of service (DDoS) attack on the controller. Experimental results show that the average DR for identifying high-rate DDoS attack traffic is improved by 6.25% points, 20.6% points, 6.74% points, and 8.81% points using the entropy-based method.

These information entropy detection-based techniques fail to optimize most of the limited resources of the controller.

B. Detection Based on Machine Learning

DDoS attacks can be detected with SDN security. Machine learning-based detection approaches are used more frequently than those based on information entropy [20].

Relevant feature selection methods for DDoS detection using ML are discussed in [21]. The final feature selection is based on the classification accuracy of the machine learning methods and the efficiency of the SDN controller. Comparative research on feature selection and machine learning classifiers for SDN attack detection has also been conducted. Using a subset of features determined by the Recursive Feature Elimination (RFE) approach, the Random Forest (RF) classifier is able to train a model with an accuracy of 99.97%, as shown by their experimental findings.

In [22] authors propose a deep learning (DL) based ensemble solution to address the problem of DDoS attack detection in SDN. In order to enhance SDN traffic classification, four hybrid models have been provided that combine three ensemble methodologies with three distinct DL architectures (convolutional neural network, long short-term memory, and gated recurrent unit). The CICIDS2017 flow dataset served as the basis for the experiments. The findings demonstrated high detection accuracy (99.77%). This method has a higher controller resource requirement because it uses four distinct hybrid models.

The authors in [23] have studied several machine learning models for DDoS detection in SDN. The question of how to improve the accuracy of DDoS attack detection has been studied using a well-known DDoS dataset called CICDDoS2019. In addition, the DDoS dataset has been preprocessed using two main approaches to obtain the most relevant features. Four machine learning models have been selected for the DDoS dataset. According to the results obtained from real experiments, the Random Forest machine learning model offered the best detection accuracy with (99.9974%), with an enhancement over the recently developed DDoS detection systems.

Authors in [24] attend to detect DDoS attacks by classifying the normal and malicious traffic. The study solves the data shift issues by using the introduced Decision Tree Detection (DTD) model encompassing of Greedy Feature Selection (GFS) algorithm and Decision Tree Algorithm (DTA). Initially, the gureKddcup dataset is loaded to perform preprocessing. After this, feature selection is performed to select only the relevant features, removing the irrelevant data. The results of the investigation revealed that the proposed system achieved an accuracy of 98.42% in the test data. This technique is based on the decision tree, which often involves higher time to train the model, which is costly in terms of detection time.

The authors of [25] use machine learning algorithms in conjunction with Neighborhood Component Analysis (NCA) to categorize SDN traffic as either benign or malicious. The project leveraged a publicly available “DDoS attack SDN dataset,” which had a total of 23 features. Through feature selection, the NCA algorithm reveals the most important features, allowing for accurate categorization. The acquired dataset is then categorized using the k-Nearest Neighbor (kNN), Decision Tree (DT), Artificial Neural Network (ANN), and Support Vector Machine (SVM) methods, after the preparation and feature selection phases. The experimental findings demonstrate that DT achieves a perfect 100% classification rate, which is far higher than any of the competing methods. This method could not be continued with other types of high-volume datasets because DT is sometimes unstable, meaning that a small change in the data can lead to a large change in the optimal decision structure.

Authors in [26], did a study that was focused on DDoS attack detection using machine learning-based methods. The primary goal of the study was to reduce misclassification error in DDoS detection and this was made possible by using Mutual information and Random Forest Feature Importance. From the features selected, Random forest, Gradient Boosting, Weighted voting ensemble, and KNN were applied and they had better accuracy when using the features selected. Random Forest, performed better in DDoS attack detection and only misclassified 1.

Although the aforementioned studies are grounded in machine learning, the most majority rely on inefficient, time-consuming, and costly fixed detection approaches that require immediate control.

C. Detection Based on Optimization Algorithms

In order to create an innovative solution, the modern new approach to DDoS detection in SDN relies on optimization algorithms techniques. The study in [27] designed an efficient and low-power SDSN topology by using the Degree Constrained Topology Generation (DCTG) algorithm and a novel formulation of the optimization target. The primary purpose of the method is to design a topology for a Software-Defined Satellite Network (SDSN) that minimizes power consumption. In addition to considering all possible link states, the proposed method strives to reduce the aggregate power usage of the network.

The authors of [28] proposed a satellite network topology optimization technique that incorporated NIDS (Network Intrusion Detection System) using federated learning distributed NIDS in STN. This program Could evaluate and filter harmful traffic as well as fairly distribute resources across each domain. Additionally, it can reduce malicious packet tracking challenges brought by frequent network changes. Malicious traffic could be identified with greater accuracy than typical NIDS, yet with less CPU usage.

Defending DDoS attacks with a metaheuristic strategy, the authors of [29] presented a whale optimization algorithm-based clustering for DDoS detection (WOA-DD). The WOA is a metaheuristic algorithm that takes inspiration from nature. With this historical data, WOA-DD hopes to distinguish between typical traffic and malicious DDoS attacks. After
the clusters have been created, any incoming requests will be distributed among them at random. WOA-DD prevents DDoS attacks, but the technology has a major drawback: the clustering process significantly slows down decision-making.

In conclusion, the SDN ecosystem is abundant with DDoS attack detection solutions that aim to identify and counteract the issue of data memory. This requires the creation of a memory-efficient, computationally straightforward, and network overhead solution.

The Optimization techniques have been utilized by many researchers to solve the network Intrusion problem. These techniques will be employed in solving the DDoS attack in the proposed method. Several optimization issues have recently been solved successfully with the help of metaheuristics (e.g., Facial emotion recognition, disease diagnosis, gene selection, and intrusion detection systems) [30]. In contrast to exact search mechanisms, metaheuristics deliver exceptional performance, because unlike full search algorithms, they don’t need to traverse the entire search space to find the optimal solution, which is an advantage in terms of computational complexity and memory.

IV. AUTOMATED MODIFIED GREY WOLF OPTIMIZER ALGORITHM (AMGWOA) FOR DDOS DETECTION IN SDN

A discussion of the proposed technique is presented in this section. It is based on the Grey Wolf algorithm and applied in blocking malicious requests from the controller.

The algorithm is based on the concept of pack intelligence and uses a grey wolf optimization model to identify the best combination of attack detection techniques.

The Grey Wolf algorithm detects suspicious traffic patterns and DDoS attacks by employing a set of heuristics. The algorithm analyzes traffic input to look for specific characteristics. These traits are used to detect malicious behavior. If any predefined characteristics are found in the data, the system flags the traffic as suspicious and takes proactive measures to block it to protect the network.

To formulate the proposed policy, we define the objective function (fitness function) is defined as follows:

A. Design of AMGWOA

\[
\min \ Z = \sum_{i=1}^{n} Req_i^{m} \\
\text{subject to:}
\]

\[
\begin{align*}
Req_i^{m} & = \tau, \ \tau \in [20, 50] \\
\sigma^t & = \mu, \ \mu \in [0.01, 1]
\end{align*}
\]

Where \( Req_i^{m} \) represents the targeted malicious requests to be minimized. The objective function \( Z \) is linked to two main constraints:

- A set of requests \( Req_i^{m} \) targeting the same resource (service/application).
- An instantiated time window \( \sigma^t \) (in seconds), in which requests are received.

B. Core Components of AMGWOA

To solve the proposed objective function, combine the GWO with a Resource-Constrained (RC) management. This latter tends to classify the received requests based on a threshold \( \lambda \) where each request is classified in regards to the three best solutions of GWO. Accordingly, this threshold is defined through the computation of the fitness function, which can be used to identify three classes:

- The Alpha class: it represents the first best solution where in our context is associated to the next \( Req_i^{m} \) that will be dropped in case of the condition \( \omega \leq \lambda^- \) is attained.
- The Beta class: it represents the second best solution. The \( Req_i^{m} \) will be dropped in case the condition \( \omega \in [\lambda^-, \lambda^+] \) is satisfied.
- The Delta class: it represents the third best solution. Similar to the previous classes but with the following condition \( \omega \geq \lambda^+ \), the request will be blocked.

We note that the condition \( \omega \) is calculated as the following \( \omega = Bw/C \) where \( Bw \) is the measured bandwidth in the network (bits per second) and the capacity \( C \) represents the number of bits that a cable can transfer. The two thresholds \( \lambda^+ \) and \( \lambda^- \) (i.e., upper bound and lower bound values) are initialized based on the constraint \( \sigma^t \) where the defined time range is partitioned into three periods: \((20 - \lambda^-), [\lambda^- - \lambda^+] \) and \([\lambda^+ - 50]\). The overall algorithm as shown on algorithm2 of our modified GWO is described on the following:

C. Implementation of AMGWOA

As shown on algorithm 2 the implementation of Automated modified Grey Wolf is described on the following:

The proposed AMGWOA algorithm will act by minimizing the fitness function, which is the total of the requests, to find the best possible solution.

When the sum of new requests arrives, each request will be examined to ensure that it is not an attack before being sent. That is to say that if it does not respect the pre-established conditions (range and time), it will be automatically blocked and not forwarded to the controller.

V. EXPERIMENTAL SETUP AND RESULTS DISCUSSIONS

In this section, we discuss our experimental setup and report our findings from testing the proposed methodology.

A. Experimental Setup

Our Simulation was done using Matlab R2020a. Due to the availability of diverse Matlab is selected because it has diverse mathematical functions.

Experiments are carried out on a personal computer PC HP Pavilion X360, Windows 10 OS with 8GB DDR4, Core i7, 10th generation CPU, and 512 GB SSD.

B. Results Discussion

In this implementation, we set the population size to 35 and the maximum number of iterations to 500 in the proposed Automated Modified Grey Wolf algorithm (AMGWOA).
Algorithm 2: Proposed Automated Modified Grey Wolf Optimizer (AMGWOA)

Initialize the GWO population (solution): \( Y_i (Y = 20) \); /* number of requests considered before blocking the next request. */

Initialize \( \alpha, \beta, \gamma, \lambda^+ \) and \( \lambda^- \). \( t = 0 \);

Calculate the fitness of each solution \( Y_i \) (e.g., \( i = 1 \cdots 20 \));

\( Y_{\alpha} \): the first malicious request;
\( Y_{\beta} \): the second malicious request;
\( Y_{\gamma} \): the third malicious request;

while \( (t < M) \) do

foreach agent do

if (number of requests = \( \tau \) & time window = \( \mu \)) then

Update the position of the current agent using equation (1);

if (the sum of Req\( \gamma \) in the Alpha class is greater than the sum of Req\( \gamma \) in the other two classes) then

Block the next request;

else

Forward the request;

end

end

Calculate the fitness value of each candidate solution (malicious requests);

Update \( Y_{\alpha}, Y_{\beta}, Y_{\gamma} \);

\( t \leftarrow t + 1 \);

end

Return \( Y_{\gamma} \).

1) AMGWOA compared with standard GWO and Particle Swarm (PSO): To verify the performance of our approach, AMGWOA is compared with standard GWO [7] and PSO [31] algorithms. For a fair comparison among the three algorithms, they were tested using the same settings of the parameters, specifically, a population size of 35 and a maximum number of iterations of 500 for all test functions. The performance of the algorithms is compared using the following metrics: Objective function values (Best solution) and running times.

Comparison parameters for the three algorithms AMGWOA, GWO, and PSO are shown in Table I below.

It can be seen from Table I that compared to standard GWO and PSO our proposed AMGWOA produces the best score (lowest) objective function value represented by equation (13) This is in fact due to the tolerable iterations of the algorithm. In terms of running time Compared with the GWO algorithm, AMGWOA reduces execution time by 96.7% and 89.9% with PSO. Compared to GWO The PSO does not take much running time, but it does not converge, giving the best optimal “Minimum” values compared to AMGWOA.

Fig. 3 displays the average value of a test function as a function of the number of iterations of the standard GWO, PSO, and AMGWOA algorithms. The graph demonstrates that compared with GWO and PSO, AMGWOA converges much faster. This is because of the two parameters \([\lambda^-, \lambda^+]\) that are introduced into GWO initialization, which improve its exploration ability and enhance global convergence.

2) Comparison of DDoS detection graphs with and without our proposed method: Fig. 4 and Fig. 5 show the results of a comparison between the use of our AMGWOA model and a standard detection system for fraudulent requests. The number of requests is plotted along the horizontal axis, and the time at which those requests are expected to arrive is plotted along the vertical axis.

The results in Fig. 4 and Fig. 5 represent the classification of the sum of the requests in the Controller before and after AMGWOA Optimization. From Fig. 4 and Fig. 5, it can be observed that the detection of DDoS in the proposed method is superior to the current methods; as a result, the number of requests in the controller within the time window \( \mu \) is reduced significantly. In Fig. 4 the number of requests is 600 whereas in Fig. 5 the requests are increased to 1200. The figures show that when the number of requests increases, the performance of our method AMGWOA also increases under the same circumstances. It can be noted that before classification the number of incoming requests is huge and exceeds the time that is allocated to them but after optimization, it is shown that whatever the number of requests their sums are minimized, and those that respect the conditions as defined in the constraints \( \tau \in [20, 50] \) and, \( \mu \in [0.01, 1] \) are maintained as normal requests. The results in Fig. 6 show the normal requests We can see that they do not overflow unlike malicious queries and
respect the defined range \([20, 50]\) and time \(\mu \in [0,01,1]\). The requests after this range and time are not visible because they are blocked and dropped by the algorithm automatically.

Our solution AMGWOA metaheuristic algorithm has been shown to be more effective than Standard GWO optimizer and other competing optimization methods such as PSO in preventing unauthorized requests.

In general, the exploration and exploitation capacity of a population-based metaheuristic algorithm determines its performance [32], [33]. If we further increase the initialization parameters threshold \(\lambda^-\) for our suggested approach, we are increasing the possibility that zombie hosts will pass as regular hosts by significantly altering their query pattern. However, the system’s efficiency will drop if we put any otherwise healthy hosts into the zombie host group by lowering the value of the threshold \(\lambda^+\). Therefore, it is essential to select an accurate threshold value for AMGWOA.

In this paper, we have developed an automated efficient Grey Wolf optimizer algorithm to prevent the DDoS attack on SDN. The results of our simulations demonstrate the relatively minimal time and space complexity of our approach. Most of the previously proposed methods for detecting and mitigating attacks once happen. These methods necessitate a huge quantity of data storage, which might be problematic for devices with limited memory and exposes the controller to high risks because some of the attacks once they enter the system have immediate effects before their detection. Our solution limited the number of flows without using up a lot of storage or processing space to separate the fraudulent requests from the legal ones.

The experimental findings demonstrate the efficacy of our method because the graph of the fitness function for malicious requests has been minimized while keeping the normal requests that respect the range \([20, 50]\) and the time \(\mu \in [0,01,1]\) allocated to them.

In future work, we will implement our architecture using hybrid metaheuristic algorithms for more accuracy and compare the results obtained.

### VI. CONCLUSION

In this paper, we have developed an automated efficient Grey Wolf optimizer algorithm to prevent the DDoS attack on SDN. The results of our simulations demonstrate the relatively minimal time and space complexity of our approach. Most of the previously proposed methods for detecting and mitigating attacks once happen. These methods necessitate a huge quantity of data storage, which might be problematic for devices with limited memory and exposes the controller to high risks because some of the attacks once they enter the system have immediate effects before their detection. Our solution limited the number of flows without using up a lot of storage or processing space to separate the fraudulent requests from the legal ones.

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### ACKNOWLEDGMENT

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### REFERENCES


A Bibliometric Analysis of Research on Risks in the Poultry Farming Industry: Trends, Themes, Collaborations, and Technology Utilization

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Abstract—This paper explores the risks prevalent in the poultry farming industry, drawing upon an extensive examination conducted by researchers over the past decade. Employing a bibliometric analysis approach, a comprehensive search of the Scopus database was conducted using relevant keywords related to poultry farming risk and technology utilization. The search spanned from 2002 to 2022, yielding 345 pertinent documents. This study presents an overview of the current state of publications concerning poultry farming risk and its intersection with technology utilization. It delves into citation patterns, prevalent themes, and authorship analysis, focusing on the role of technology in mitigating risks. The comprehensive citation analysis highlights the impact of technology-related studies in the field. Frequency analysis employed Microsoft Excel, while VOSviewer facilitated data visualization. Harzing’s Publish or Perish software was used for citation metrics and analysis. The findings reveal a consistent increase in publications on risk in poultry farming since 2002, particularly in relation to technology utilization. The United States emerges as the most active country in this area of research, with Wageningen University from the Netherlands identified as the most prolific institution contributing significantly to risk in poultry farming research, including technology applications. The research involved 32 scholars from 70 different countries and 32 distinct institutions, reflecting the multi-authorship and multicultural nature of the research. It is important to note that this paper focuses solely on the Scopus database, while future researchers may consider alternative databases for new studies, recognizing the expanding role of technology in addressing risks in the poultry farming industry.

Keywords—Poultry farming risk; poultry farming industry; bibliometric analysis; Harzing’s Publish or Perish; VOSviewer

I. INTRODUCTION

Poultry farming is the practice of raising domesticated birds such as chickens, turkeys, ducks, and geese for their meat, eggs, and feathers [1]. The poultry farming industry refers to the commercial production of these birds on a large scale to meet the demands of the market [2]. In this industry, chickens are the most farmed bird due to their fast growth rate and high meat and egg production [3]. The industry involves breeding, hatching, raising, and slaughtering poultry birds in large numbers using modern farming techniques, equipment, and facilities. Poultry farms can be categorized into two main types: broiler farms and layer farms. Broiler farms focus on raising chickens for meat production, while layer farms focus on raising chickens for egg production [4]. In both types of farms, the birds are kept in large sheds or cages, which are designed to provide them with the ideal conditions for growth and development. Poultry farming has become a significant industry worldwide, with billions of birds raised each year to meet the demands of the global market. The industry has seen significant growth in recent years due to advancements in technology, improved genetics of the birds, and increased demand for poultry products. In general, there are three types of households where poultry are raised: close farming systems, semi-intensive farming systems, and intense farming systems [5]. The backyard poultry industry has transformed into a significant supplier of poultry meat and eggs by embracing modern breeds, advanced housing and equipment, and efficient marketing systems. In the past, backyard poultry solely relied on local, low-yield, and unremarkable poultry breeds, which had limited productivity [6].

As increasing demand for food by the world’s population has resulted in tremendous growth in agricultural productivity in recent years [7] and this will cause a lot of issues or risks for this industry that need to be controlled to ensure production can be maximized. Moreover, farmers need to increase their work efficiency to improve the quality of the poultry product while still focusing on animal welfare, environmental sustainability, and public health [8]. According to [9], livestock sector faces numerous challenges, including diseases, climate change, inadequate management practices, genetic issues, farmer capacity and skills, marketing challenges, infrastructure limitations, and a lack of information for effective decision-making. Consequently, the current production rate does not meet the projected demand. Conducting a bibliometric analysis allows us to observe the patterns and trends discussed by researchers regarding these issues. Furthermore, utilizing bibliometric analysis can serve as a valuable guide for farmers and researchers alike. It enables us to identify the key issues being discussed among researchers, shed light on the most prominent researchers who have published on the topic, and gain insights into the specific topics they have explored, including potential solutions and recommendations. This analysis offers a comprehensive overview of the scholarly landscape, providing guidance and
informing future directions for both farmers and researchers in addressing the identified challenges. Bibliometric analysis is frequently employed to evaluate trends and impact, encompassing factors such as publishing countries, subject categories, journals, and author keywords [10]. The objective of this study is to examine trends and advances to better inform researchers about the viewpoint of risk research in poultry farming. A forward-looking analysis that can anticipate the author’s contribution, both presently and in the future, is imperative. This study serves as a heuristic tool to assess the literature on poultry farming risk to provide practitioners and researchers with the most recent developments in pedagogy. In order to shed light on the topic at hand, this study endeavors to answer the following research questions (RQs):

- **RQ1:** What is the current status of research publications on the topic of poultry farming risk?
- **RQ2:** How are publications on poultry farming risk being cited in current literature?
- **RQ3:** Which themes related to poultry farming risk are receiving the most attention and interest among scholars?
- **RQ4:** What patterns can be observed in terms of authorship in publications on poultry farming risk?

II. METHOD

This paper employs bibliometric analysis to examine trends and productivity in research on poultry farming risks. The study’s methodology encompasses the data gathering and filtering processes, culminating in the analysis-ready dataset. The objective is to identify the topic and scope of the study, encompassing all available research on risk in poultry farming within the Scopus database. The paper presents various bibliometric indicators and utilizes network visualization techniques. The research protocol used to guide the selection of the documents acquired for this study was shown in Fig. 1.

A. Bibliometric Analysis

The aim of this study is to utilize bibliometric analysis as a quantitative technique to uncover the prevailing trend in poultry farming risk. Bibliometric analysis involves statistical measurements and enables the examination of published articles or bibliographic units, providing insights into the distribution and characteristics of the literature in the field [10], [11]. The analysis has the capability to identify descriptive patterns in the articles generated based on various factors such as domain, field, country, or time [12]. There are three categories of bibliometric study indicators include quantity, quality, and structural indicators [13]. The quantity indicator in bibliometric analysis refers to the productivity of researchers, which can be measured by factors such as the number of publications or citations per year [14]. On the other hand, quality indicators assess the performance of researchers based on metrics like the overall h-index, g-index, and citation score, which reflect the impact and recognition of their work. Structural indicators, on the other hand, focus on the relationships between publications, authors, and research fields, shedding light on the interconnections within the scholarly landscape [15]. Additionally, employing a systematic methodology necessary for conducting bibliometric analysis can reveal valuable insight, including information about the authors, frequency of keywords and citations [16].

B. Source and Data Collection

The Scopus database was selected as the main data source that being used in this paper to achieve paper’s objective since it is known for being the "largest single abstract and indexing database ever developed and the largest searchable citation and abstract literature search list.” [17], [18]. Moreover, Scopus database also is the largest database of peer-reviewed literature, including books, journals, and conference proceedings [19] and the database has over 36,000 titles from nearly 11,000 publishers, mostly from peer-reviewed journals, and it covers the social, physical, health, and biological sciences [16].

The search query of the poultry farming risk topic was applied in the Scopus data based within search by keyword and with the 345 documents were produced for the next further analysis. Besides, other tools also have been utilized for gathering the document for example, Microsoft Excel, Harzing’s Publish and Perish software and VOSviewer.

The data was obtained from the Scopus database on November 6, 2022. The documents are restricted to the topic based on the poultry farming risk title. To fulfill our objective, we executed a query with the KEYWORD term ‘poultry farming risk.’ This query yielded a total of 477 documents. Through a thorough data cleaning process, we identified and eliminated 132 duplicate or irrelevant documents not pertinent to our topic. The data collected from the Scopus database was then exported into comma-separated values (.csv) and research information systems (.ris) formatted files, ensuring a standardized and organized dataset.

![Fig. 1. Flow diagram of the search strategy.](image-url)
This study utilized the bibliometric method to analyze the research trends in the field of poultry farming risk. The database provides a comprehensive range of publication details, encompassing information such as publication type, year, language, subject area, keywords, country, affiliation, citations, and authorship of the documents. Data analysis was conducted using Microsoft Excel and Harzing's Publish or Perish software, while VOSviewer software was utilized for data visualization. In this study, the VOSviewer software played a key role in conducting the mapping analyses [10]. When representing the nodal network, VOSviewer employs two uniform weights to create a graphical visualization: the quantity of connections and the collective strength they possess. The relevance and power of the links are reflected in the network size and network-connecting interlinking lines.

III. RESULT AND FINDING

To acquire an overview of the research related to poultry farming risk, data sets are provided. All articles that met the search query were evaluated for current publication such as year, language of publications, subject area, distribution of publications by countries, most active authorship, keywords, and citation analyses.

A. Current State of Publication in Poultry Farming Risk

In order to investigate RQ1 (What is the current state of publication in poultry farming risk?), a thorough analysis was conducted to examine the publication trends in this field. The analysis encompassed several key dimensions, including the total number of publications by year, document type, publication source title, country of publication, affiliated institutions, language, and subject area. To perform this analysis, we utilized the bibliographic data collected from the Scopus database, allowing for comprehensive calculation and evaluation of the relevant data.

- Publication by year

Examining documents by publication year allows the researcher to track the pattern and popularity of the research topic through time [27-33]. Table I summarises the total publications for poultry farming risk since 2002. In the early topic of poultry farming risk there were three articles about this topic that were published in 2002 that were written by [20]–[23]. Nevertheless, analysis of these papers revealed that in 2002 were justified as the pioneers of publication as these papers that was written by [21] was cited 259 times. According to the records, between 2005 and 2015, the number of publications had an inconsistent pattern, with fluctuations in both increases and decreases. However, starting from 2016, there was a noticeable growth in the number of articles being published. This trend continued, and in 2020, there was a record high of 35 publications, accounting for 10.14% of the total. This significant increase indicates a peak period in the practice of the poultry farming risk approach adopted by educators worldwide (Fig. 2). This observation becomes clearer when analysing the trends of countries participating in research.

<table>
<thead>
<tr>
<th>Year</th>
<th>Total Publications (TP)</th>
<th>Percentage (%)</th>
</tr>
</thead>
<tbody>
<tr>
<td>2002</td>
<td>3</td>
<td>0.87%</td>
</tr>
<tr>
<td>2003</td>
<td>9</td>
<td>2.61%</td>
</tr>
<tr>
<td>2004</td>
<td>5</td>
<td>1.45%</td>
</tr>
<tr>
<td>2005</td>
<td>14</td>
<td>4.06%</td>
</tr>
<tr>
<td>2006</td>
<td>12</td>
<td>3.48%</td>
</tr>
<tr>
<td>2007</td>
<td>10</td>
<td>2.90%</td>
</tr>
<tr>
<td>2008</td>
<td>12</td>
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<td>2009</td>
<td>18</td>
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<td>2010</td>
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<td>2011</td>
<td>16</td>
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<td>2012</td>
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<tr>
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<tr>
<td>2018</td>
<td>21</td>
<td>6.09%</td>
</tr>
<tr>
<td>2019</td>
<td>28</td>
<td>8.12%</td>
</tr>
<tr>
<td>2020</td>
<td>35</td>
<td>10.14%</td>
</tr>
<tr>
<td>2021</td>
<td>27</td>
<td>7.83%</td>
</tr>
<tr>
<td>2022</td>
<td>16</td>
<td>4.64%</td>
</tr>
<tr>
<td>Total</td>
<td>345</td>
<td>100%</td>
</tr>
</tbody>
</table>

By examining the number of publications by year, it is possible to observe that the publication trend is clearly visible. This trend is significant and reflects the increasing interest in the topic of poultry farming risk. The analysis of the number of publications per year provides valuable insights into the research activity in this field over the past 20 years. The data analysis reveals that the publication trend has been generally increasing, with fluctuations in the number of publications from year to year. This pattern indicates a growing interest in the topic of poultry farming risk, with a peak period observed in 2020. The analysis of the number of publications by country is also important, as it provides a comprehensive overview of the countries that are actively contributing to the research in this field. This analysis can help identify the countries that are leading in research on poultry farming risk and the countries that are lagging behind.

- Document type

The term "document type" pertains to the categorization of documents into various categories, including conference papers, articles, book chapters, reviews, and books. Table II provides a summary of the distribution of documents published on poultry farming risk, classified into seven document types. Notably, a significant majority of the publications (84.06%) were classified as articles. The remaining document types include conference papers, reviews, books, book chapters, and letters, with a few publications classified as book reviews and letters. This distribution highlights the importance of articles as the primary mode of publication in the field of poultry farming risk.

<table>
<thead>
<tr>
<th>Document Type</th>
<th>Total Publications (TP)</th>
<th>Percentage (%)</th>
</tr>
</thead>
<tbody>
<tr>
<td>Article</td>
<td>290</td>
<td>84.06%</td>
</tr>
<tr>
<td>Review</td>
<td>23</td>
<td>6.67%</td>
</tr>
<tr>
<td>Letter</td>
<td>14</td>
<td>4.06%</td>
</tr>
<tr>
<td>Note</td>
<td>10</td>
<td>2.90%</td>
</tr>
<tr>
<td>Conference Paper</td>
<td>4</td>
<td>1.16%</td>
</tr>
<tr>
<td>Editorial</td>
<td>3</td>
<td>0.87%</td>
</tr>
<tr>
<td>Short Survey</td>
<td>1</td>
<td>0.29%</td>
</tr>
</tbody>
</table>
- Publication by country

There are top 10 listed as the most productive countries (see Table III). United States was dominated in publishing this topic with a total of 81 (23.48%) documents followed by United Kingdom (56; 16.23%) and China (50; 14.49%). The United States’ leading position in poultry farming publications is unsurprising, as their publications have garnered the highest number of citations, indicating extensive recognition worldwide.

**TABLE III. COUNTRY**

<table>
<thead>
<tr>
<th>Country</th>
<th>TP</th>
<th>%</th>
<th>NCP</th>
<th>TC</th>
<th>C/P</th>
<th>C/CP</th>
<th>h</th>
<th>g</th>
</tr>
</thead>
<tbody>
<tr>
<td>United States</td>
<td>81</td>
<td>23.48%</td>
<td>76</td>
<td>2894</td>
<td>35.73</td>
<td>38.08</td>
<td>31</td>
<td>52</td>
</tr>
<tr>
<td>United Kingdom</td>
<td>56</td>
<td>16.23%</td>
<td>51</td>
<td>1688</td>
<td>30.14</td>
<td>33.10</td>
<td>25</td>
<td>40</td>
</tr>
<tr>
<td>China</td>
<td>50</td>
<td>14.49%</td>
<td>49</td>
<td>1439</td>
<td>28.78</td>
<td>29.37</td>
<td>18</td>
<td>37</td>
</tr>
<tr>
<td>Netherlands</td>
<td>28</td>
<td>8.12%</td>
<td>28</td>
<td>1428</td>
<td>51.00</td>
<td>51.00</td>
<td>16</td>
<td>28</td>
</tr>
<tr>
<td>France</td>
<td>26</td>
<td>7.54%</td>
<td>26</td>
<td>702</td>
<td>27.00</td>
<td>27.00</td>
<td>12</td>
<td>26</td>
</tr>
<tr>
<td>Australia</td>
<td>18</td>
<td>5.22%</td>
<td>17</td>
<td>401</td>
<td>22.28</td>
<td>23.59</td>
<td>9</td>
<td>18</td>
</tr>
<tr>
<td>Italy</td>
<td>18</td>
<td>5.22%</td>
<td>17</td>
<td>441</td>
<td>24.50</td>
<td>25.94</td>
<td>13</td>
<td>18</td>
</tr>
<tr>
<td>Germany</td>
<td>14</td>
<td>4.06%</td>
<td>14</td>
<td>414</td>
<td>29.57</td>
<td>29.57</td>
<td>9</td>
<td>14</td>
</tr>
<tr>
<td>Thailand</td>
<td>14</td>
<td>4.06%</td>
<td>12</td>
<td>262</td>
<td>18.71</td>
<td>21.83</td>
<td>7</td>
<td>14</td>
</tr>
<tr>
<td>Belgium</td>
<td>12</td>
<td>3.48%</td>
<td>12</td>
<td>601</td>
<td>50.08</td>
<td>50.08</td>
<td>11</td>
<td>12</td>
</tr>
</tbody>
</table>

**TABLE IV. INSTITUTIONS MOST AFFILIATED WITH POULTRY FARMING RISK**

<table>
<thead>
<tr>
<th>Institution</th>
<th>TP</th>
<th>%</th>
<th>Country</th>
<th>NCP</th>
<th>TC</th>
<th>C/P</th>
<th>C/CP</th>
<th>h</th>
<th>g</th>
</tr>
</thead>
<tbody>
<tr>
<td>Wageningen University</td>
<td>2</td>
<td>5.80%</td>
<td>Netherlands</td>
<td>20</td>
<td>64</td>
<td>32.0</td>
<td>32.0</td>
<td>1</td>
<td>2</td>
</tr>
<tr>
<td>Royal Veterinary College University of London</td>
<td>1</td>
<td>4.06%</td>
<td>United Kingdom</td>
<td>13</td>
<td>54</td>
<td>39.0</td>
<td>42.0</td>
<td>9</td>
<td>14</td>
</tr>
<tr>
<td>Centers for Disease Control and Prevention</td>
<td>1</td>
<td>3.48%</td>
<td>United States</td>
<td>11</td>
<td>87</td>
<td>72.5</td>
<td>79.0</td>
<td>8</td>
<td>12</td>
</tr>
<tr>
<td>National Institute for Public Health and the Environment</td>
<td>1</td>
<td>3.19%</td>
<td>Netherlands</td>
<td>11</td>
<td>97</td>
<td>88.3</td>
<td>88.3</td>
<td>7</td>
<td>11</td>
</tr>
<tr>
<td>The University of Hong Kong</td>
<td>1</td>
<td>2.90%</td>
<td>Hong Kong</td>
<td>10</td>
<td>17</td>
<td>17.7</td>
<td>17.7</td>
<td>8</td>
<td>10</td>
</tr>
</tbody>
</table>

**TABLE V. LANGUAGES USED FOR PUBLICATIONS**

<table>
<thead>
<tr>
<th>Language</th>
<th>Total Publication</th>
<th>Percentage</th>
</tr>
</thead>
<tbody>
<tr>
<td>English</td>
<td>342</td>
<td>98.84%</td>
</tr>
<tr>
<td>German</td>
<td>2</td>
<td>0.58%</td>
</tr>
<tr>
<td>Chinese</td>
<td>1</td>
<td>0.29%</td>
</tr>
<tr>
<td>French</td>
<td>1</td>
<td>0.29%</td>
</tr>
</tbody>
</table>

- Publication by institution

Table IV depicts the top five institutions that publish on Poultry Farming Risk. Based on the result, it stated that Wageningen University is the highest institution that were publish in this topic with total 20 publications with the average citation per year 32.00. Even though it has the highest publication, National Institute for Public Health and the Environment that were in Netherlands has the highest citation per year (88.36) compared to other institution. When institutions were ranked according to h-index, the Wageningen University and Royal Veterinary College University of London top institutions that lead the publications on poultry farming risk.

- Languages of the document

There are four languages used for publications that were mostly published in English languages (342; 98.84%) (See Table V). The extensive use of a wide range of languages, including major ones, in publications signifies the diverse global involvement and breadth of research on poultry farming risks across nations.

**TABLE VI. LANGUAGES USED FOR PUBLICATIONS**

Based on all the documents gathered from the Scopus database, poultry farming risk covers almost all subject areas especially in medicine. Table VI shows that between this period (2002–2022), the most written subject in these topics is medicine (48.41%) as almost the half of the publications were published for this topic.
TABLE VI. THE MOST WRITTEN SUBJECT AREAS

<table>
<thead>
<tr>
<th>Subject Area</th>
<th>Total Publication</th>
<th>Percentage</th>
</tr>
</thead>
<tbody>
<tr>
<td>Medicine</td>
<td>167</td>
<td>48.41%</td>
</tr>
<tr>
<td>Veterinary</td>
<td>86</td>
<td>24.93%</td>
</tr>
<tr>
<td>Agricultural and Biological Sciences</td>
<td>79</td>
<td>22.90%</td>
</tr>
<tr>
<td>Environmental Science</td>
<td>69</td>
<td>20.00%</td>
</tr>
<tr>
<td>Immunology and Microbiology</td>
<td>63</td>
<td>18.26%</td>
</tr>
<tr>
<td>Biochemistry, Genetics and Molecular Biology</td>
<td>43</td>
<td>12.46%</td>
</tr>
<tr>
<td>Multidisciplinary</td>
<td>24</td>
<td>6.96%</td>
</tr>
<tr>
<td>Pharmacology, Toxicology and Pharmaceutics</td>
<td>18</td>
<td>5.22%</td>
</tr>
<tr>
<td>Engineering</td>
<td>7</td>
<td>2.03%</td>
</tr>
<tr>
<td>Social Sciences</td>
<td>7</td>
<td>2.03%</td>
</tr>
<tr>
<td>Chemistry</td>
<td>5</td>
<td>1.45%</td>
</tr>
<tr>
<td>Computer Science</td>
<td>5</td>
<td>1.45%</td>
</tr>
<tr>
<td>Nursing</td>
<td>5</td>
<td>1.45%</td>
</tr>
<tr>
<td>Earth and Planetary Sciences</td>
<td>3</td>
<td>0.87%</td>
</tr>
<tr>
<td>Chemical Engineering</td>
<td>2</td>
<td>0.58%</td>
</tr>
<tr>
<td>Economics, Econometrics and Finance</td>
<td>2</td>
<td>0.58%</td>
</tr>
<tr>
<td>Materials Science</td>
<td>2</td>
<td>0.58%</td>
</tr>
<tr>
<td>Arts and Humanities</td>
<td>1</td>
<td>0.29%</td>
</tr>
<tr>
<td>Health Professions</td>
<td>1</td>
<td>0.29%</td>
</tr>
<tr>
<td>Mathematics</td>
<td>1</td>
<td>0.29%</td>
</tr>
<tr>
<td>Neuroscience</td>
<td>1</td>
<td>0.29%</td>
</tr>
</tbody>
</table>

TP total number of publications, NCP number of cited publications, TC total citations, C/P average citations per publication, C/CP average citations per cited publication, h h-index, and g g-index

B. Citations Pattern on Poultry Farming Risk

Our second research question (RQ2) aims to identify the most influential articles on poultry farming risk and visualize the citation patterns using data obtained from the Scopus database. The analysis involved examining the citation metrics and networks of 345 articles to address RQ2. Citation analysis was conducted to measure the impact of documents on poultry farming risk using relevant citation metrics. Data analysis was performed utilizing Harzing’s Publish or Perish and VOSviewer software. Table VII presents the citation metrics for the retrieved documents as of 06 November 2022. Among the 345 articles, an average of 452.35 citations per year was observed, with a total of 9,047 citations reported.

Table VIII displays comprehensive information, including the overall number of total citations and the average number of citations per year for all retrieved publications. Based on Scopus data, the top five most cited articles are listed. Notably, the article titled "Antimicrobial residues in animal waste and water resources proximal to large-scale swine and poultry feeding operations" by [20] has garnered the highest number of citations, with 395 citations or an average of 19.75 citations per year.

Fig. 3 illustrates the network visualization map, which reveals the citation connections among countries. It is worth noting that "Citations attribute indicates the number of citations received by a document or the total number of citations received by all documents published by a source, an author, an organization, or a country" [23]. Out of the 71 countries considered, 68 have met the minimum thresholds for both the number of documents and citations attributed to an author. To interpret Fig. 3 effectively, it is essential to refer to the corresponding data in Tables VII and VIII, which provide specific information on the number of citations received by each country. Notably, the United States, United Kingdom, and China have received a substantial number of citations in the domain of poultry farming risk.

C. Theme in Poultry Farming

- To address RQ3 (Which themes involving poultry farming risk are the most popular among scholars?), we conducted co-occurrence analysis using the keywords data obtained from the Scopus database. Co-occurrence of keywords indicates the presence of a relationship between two concepts when they appear together in an article [26]. We performed the co-occurrence analysis and keyword evaluation based on the understanding that author keywords provide a meaningful representation of an article's content [27]."
### TABLE VIII. HIGHLY CITED ARTICLE

<table>
<thead>
<tr>
<th>No.</th>
<th>Author(s)</th>
<th>Title</th>
<th>TC</th>
<th>CY</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>[21]</td>
<td>Antimicrobial residues in animal waste and water resources proximal to large-scale swine and poultry feeding operations</td>
<td>395</td>
<td>19.75</td>
</tr>
<tr>
<td>2</td>
<td>[22]</td>
<td>Routes for salmonella contamination of poultry meat: Epidemiological study from hatchery to slaughterhouse</td>
<td>148</td>
<td>7.4</td>
</tr>
<tr>
<td>4</td>
<td>[24]</td>
<td>A comparison between leg problem in Danish and Swedish broiler production</td>
<td>48</td>
<td>2.53</td>
</tr>
<tr>
<td>5</td>
<td>[25]</td>
<td>Survey of effects of radiofrequency electromagnetic fields on production, health, and behavior of farm animals</td>
<td>4</td>
<td>0.21</td>
</tr>
</tbody>
</table>

Undoubtedly, author keywords are of great importance for researchers seeking to identify research trends. Additionally, [28] emphasize the significance of author keyword analysis in measuring the development of research topics. After eliminating duplicate keywords resulting from spelling variations (e.g., Poultry farming, poultry farm, poultry), our analysis reveals the top 10 frequently used author keywords in relation to this topic (see Table IX).

Subsequent analysis involved the mapping of all keywords using VOSviewer, a software tool specifically designed for constructing and visualizing bibliometric networks (refer to Fig. 4). For this purpose, a comprehensive approach was adopted, considering a minimum threshold of ten occurrences for each keyword out of a total of 3,716 keywords. Consequently, 188 keywords met the specified criteria. Employing these settings, VOSViewer generated the network visualization depicted in Fig. 4, where the colors, circle sizes, and line thicknesses indicate the strength of the relationships between the keywords [26]. In Fig. 4, each color corresponds to a distinct cluster identified in the visualization map. Initially, four clusters were observed. However, by adjusting the cluster size criterion to include a minimum of 50 items in each cluster, three clusters remained. Cluster one, represented by red, comprises 60 items centered around the theme of avian influenza. Cluster two, depicted in green, encompasses 53 items associated with the non-human theme. Cluster three, highlighted in blue, includes 53 items related to the poultry farming theme.

![Network visualization map of all keywords.](image)

**D. Authorship Analysis**

Besides, this paper also analyses the most active authors who published the documents on this topic as stated in Table X. All the data that was analysed, is based on the active author who published at least more than three documents on poultry farming risk. Even tough there are four authors who have same total publications, author Gilbert, Marius is the most active author who published research article in this topic; as his article journal consists of the highest citation per year and h-index compared to others.

### TABLE IX. TOP KEYWORDS

<table>
<thead>
<tr>
<th>Keywords</th>
<th>TP</th>
<th>%</th>
</tr>
</thead>
<tbody>
<tr>
<td>Poultry Farming</td>
<td>285</td>
<td>82.61%</td>
</tr>
<tr>
<td>Article</td>
<td>257</td>
<td>74.49%</td>
</tr>
<tr>
<td>Animals</td>
<td>234</td>
<td>67.83%</td>
</tr>
<tr>
<td>Nonhuman</td>
<td>217</td>
<td>62.90%</td>
</tr>
<tr>
<td>Poultry</td>
<td>205</td>
<td>59.42%</td>
</tr>
<tr>
<td>Human</td>
<td>168</td>
<td>48.70%</td>
</tr>
<tr>
<td>Animal</td>
<td>152</td>
<td>44.06%</td>
</tr>
<tr>
<td>Risk Factor</td>
<td>138</td>
<td>40.00%</td>
</tr>
<tr>
<td>Risk Assessment</td>
<td>135</td>
<td>39.13%</td>
</tr>
</tbody>
</table>

### TABLE X. MOST PRODUCTIVE AUTHOR

<table>
<thead>
<tr>
<th>Author’s Name</th>
<th>Affiliation</th>
<th>Country</th>
<th>T P</th>
<th>NC P</th>
<th>T C</th>
<th>C/P</th>
<th>C/C P</th>
<th>h</th>
<th>g</th>
</tr>
</thead>
<tbody>
<tr>
<td>Cowling, B.J.</td>
<td>The University of Hong Kong</td>
<td>Hong Kong</td>
<td>6</td>
<td>6</td>
<td>10</td>
<td>4</td>
<td>17.3</td>
<td>3</td>
<td>17.3</td>
</tr>
<tr>
<td>Gilbert, M.</td>
<td>Belgian Free Universities of Brussels</td>
<td>Belgium</td>
<td>6</td>
<td>6</td>
<td>30</td>
<td>0</td>
<td>50.0</td>
<td>0</td>
<td>50.0</td>
</tr>
<tr>
<td>Johnson, E.S.</td>
<td>University of Arkansas</td>
<td>United States</td>
<td>6</td>
<td>6</td>
<td>56</td>
<td>0</td>
<td>9.33</td>
<td>0</td>
<td>9.33</td>
</tr>
<tr>
<td>Wagenaar, J.A.</td>
<td>Universiteit Gent</td>
<td>Netherlands</td>
<td>6</td>
<td>6</td>
<td>28</td>
<td>1</td>
<td>46.8</td>
<td>3</td>
<td>46.8</td>
</tr>
<tr>
<td>Carriquiri-Mas, J.J.</td>
<td>Oxford University</td>
<td>United Kingdom</td>
<td>5</td>
<td>5</td>
<td>17</td>
<td>5</td>
<td>35.0</td>
<td>0</td>
<td>35.0</td>
</tr>
<tr>
<td>Xiao, X.</td>
<td>University of Oklahoma</td>
<td>United States</td>
<td>5</td>
<td>5</td>
<td>21</td>
<td>6</td>
<td>43.2</td>
<td>0</td>
<td>43.2</td>
</tr>
<tr>
<td>Cui, B.</td>
<td>Yangzhou University</td>
<td>China</td>
<td>4</td>
<td>4</td>
<td>16</td>
<td>0</td>
<td>4.00</td>
<td>0</td>
<td>4.00</td>
</tr>
<tr>
<td>Fielding, R.</td>
<td>The University of Hong Kong Li Ka Shing</td>
<td>Hong Kong</td>
<td>4</td>
<td>4</td>
<td>62</td>
<td>0</td>
<td>15.5</td>
<td>0</td>
<td>15.5</td>
</tr>
</tbody>
</table>

TP total number of publications, NCP number of cited publications, TC total citations, C/P average citations per publication, C/C P average citations per cited publication, h h-index, and g g-index
IV. CONCLUSION

Bibliometric analysis can be used to analyze the trend in Poultry Farming research. It also can analyze the productivity of research and other specific research domains. According to [34] the insights obtained from bibliometric analysis can provide an understanding of the factors that influence the research output and contribution within different fields of study. These findings can guide researchers in conducting meaningful and impactful research. However, it is important to note that the scope of this study is confined to the publications related to poultry farming risk available in the Scopus database. Based on the results that had been conducted by using Harving’s Publish or Perish software and VOSviewer we can see that all the research question have been answered. Overall, there were 345 total documents that were found from Scopus database using the defined search query. The author in [27] is the most active author who published research articles on this topic in six total publications and had 300 of total citation. Besides, English is the most common language that has been used in this topic and Medicine is the subject area that is the most written subject are covered for this research study. Regarding the country, the country with most publications, covered in this research, is United States as the total publications based on this topic were 81. Based on the analysis of the top five most cited articles, it can be inferred that a significant portion of the publications focused on the topic of diseases in the poultry farming industry. This finding suggests that despite advancements in management practices, disease outbreaks remain a critical risk that necessitates continuous monitoring and attention. Finally, there are still more opportunities for research related to this topic that can be conducted in future. Due to the distinctive nature of the bibliometric analysis, the study has limitations that should be addressed to provide readers with a clear understanding of the paper and to improve future research. First, the results were limited to a single keyword, such as poultry farming risk, based on the document’s title. As a result, the results of the search query on other fields, such as article title and abstract, have been left out of this analysis. The main reason for this is because by using search query for article title, the documents are not enough to be analyzed using bibliometric. By using search query for keyword, even though some included a term linked to the search area yet might not be related with the topics, but by using data cleaning we manage to get at least certain amount of documents to analyze. But the only disadvantage is that there is a lot of filtering and cleaning to be done before the analysis can begin. It is likely that future research will be expanded into it. Finally, being the primary source of documents, this study is solely focused on the Scopus database. Web of Science, Google Scholar, and Dimensions are some of the other databases that could be used in future research. Combining all these databases will almost certainly result in more interesting and useful findings. Despite these limitations, this study has added to the body of knowledge by describing the current state of poultry farming risk. This study also uses a bibliometric technique to enhance and augment prior findings on poultry farming risk literature and provide valuable insights into the trend of previous publications.

ACKNOWLEDGMENT

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REFERENCES


Adaptive Style Transfer Method of Art Works Based on Laplace Operator

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Jiaozuo, Henan 454001

Abstract—In order to improve the image quality of artworks after style transfer, the adaptive style transfer method of artworks based on the Laplace operator is studied. Through three steps of expansion processing, corrosion processing and multi-scale morphological enhancement, the image edge of the content of artworks is enhanced. The colour and brightness of the artworks with edge enhancement are transferred, and the transfer results are input into the convolution neural network simultaneously with the style image. According to the improved Laplace operator, the Laplace operator loss term of the convolution neural system is counted, the style losing term of the style picture of the art image is determined, and the total loss function is constructed. According to the determined loss function, a convolution neural network is used to output paintings' adaptive style transfer results. The experiential outcomes indicate that this technique is able to realize the adaptive style transmission of paintings. After style transfer, the picture quality of paintings is high, and the adaptive transfer of artworks can be realized within 500ms.

Keywords—Laplace operator; artworks; adaptive style transfer; brightness migration; convolution neural network

I. INTRODUCTION

With the development of computer technology, the task of image art creation can be accomplished excellently with the help of a computer's image processing ability. Convolution neural network is a powerful and effective model in image processing [1]. Deep learning is a heavyweight machine learning algorithm [2] with a huge hyperparametric space and requires strict computational efficiency in practical applications. To a large extent, it restricts the further promotion of the image style transfer method of deep learning in practical applications. Deep learning is still a black box inside, and the mathematical meaning of its hyperparametric space is still incomprehensible [3, 4]. The process of applying the style of one style image to another content image [5, 6] is an artistic creation and image editing technology. Deep learning has made great breakthroughs in computer vision in the fields of image processing, intelligent robots, natural language processing, data mining and unmanned driving. Image style transfer algorithm is an application of depth neural network [7]. The appearance of an image style transfer algorithm can make the generation of a master artist's painting become a reality in a short time.

At present, many researchers have carried out research on image migration methods. Yuchi et al. [8] performed K-means clustering matching on transition images and obtained the optimal energy equation for color transfer through two methods: hierarchical transfer and global transfer. This method can achieve effective color transfer of images, but after color transfer, the image quality is poor; Deng et al. [9] studied a tapestry based adaptive perceptual domain style transfer algorithm based on semantic segmentation, combining semantic segmentation tasks with adaptive perceptual domain style transfer algorithms, and proposing new content loss and style loss. Although this method can achieve effective image migration, there is a problem of unprotected content and image structure, and the results of graphic migration are not ideal; Sun et al. [10] proposed a local style transfer method based on residual neural networks. Generate an image that only completes style transfer for the target area through deconvolution. This method has high local style conversion ability and high execution efficiency, but there is a problem of losing the main structure of the content image, resulting in poor visual effect of stylized images.

In mathematics and physics, the Laplace operator is differential. The Laplace operator has many uses, and it is also an important example of an elliptic operator. The function that is zero by the Laplace operator is called a harmonic function; the Laplace operator is the core of Hodge's theory and is the result of De Rham's cohomology. Laplace operator can extract the edge information in the image, and emphasize the details and texture of the image. In the style transfer of art works, by performing Laplace filtering between the original image and a reference image with a specific style, the transferred image can retain the original content while maintaining the artistic style of the reference image. Therefore, in order to promote art creation and technology development, provide more possibilities for art creation, and apply the potential of this technology in different fields, research the adaptive style transfer method of art works based on Laplace operator. The motivation of using the Laplace operator for style transfer is that it can help artists achieve more accurate style conversion, making the transferred works more consistent with the expected artistic effect. The main ideas of this study are as follows:

1) Firstly, edge enhancement is performed on the content images of art works through dilation processing, corrosion processing, and multi-scale morphological enhancement algorithms.

2) After edge enhancement, Convolutional neural network is selected as the main framework of adaptive style transfer of art works.

3) The improved Laplace operator algorithm is studied, which combines Laplace operator with Convolutional neural network; then design an adaptive style migration network.
structure based on color, brightness and semantic information, and apply it to the adaptive style migration of art works to achieve migration.

4) Through experiments, it has been verified that this study can improve the transfer level of art works and obtain user satisfactory adaptive style transfer results.

II. MATERIALS AND METHODS

A. Edge Enhancement Algorithm of the Content Image of Artworks

1) Dilation processing: The expansion operator's task is to solve the maximum local value of the pixel in the artwork image. The structural element is used to calculate the area covered by the expansion operator in the artwork [11], find the maximum value of the pixel, and then input the calculated value into the element specified in the centre of the artwork structural element.

If \( f(s,t) \) is the original artwork image, the artwork image after dilation processing is:

\[
f_{e}(x,y) = (f \oplus b)(s,t) = \max \{ f(s-x,t-y) + b(x,y) \} \tag{1}
\]

In formula (1), \( b \) is a structural element; \( D_f \) and \( D_b \) are definition fields; \( f \oplus b \) is the result of \( b \) dilating the original artwork image \( f \).

2) Corrosion processing: Contrary to the expansion operator, it is the task of the erosion operator to solve the local minimum value of pixels in the picture of paintings. The structural element is used to calculate the area of the artwork covered by the expansion operator, find out the minimum pixel value of each area of the corresponding artwork, and then input the calculated value into the element specified in the centre of the structural element. Corrosion processing is conducive to filtering noise [12], preserving the original information in the picture of paintings, making the edges smooth, and the extracted picture of paintings more continuous.

The picture of the artwork after corrosion processing is as follows:

\[
f_{c}(x,y) = (f-b)(s,t) = \min \{ f(s+x,t+y) - b(x,y) \} \tag{2}
\]

In formula (2), \( f - b \) is the result of corrosion operation on \( f \).

3) Multi-scale morphological enhancement algorithm: The selection of structural elements is very important to the edge detection results of artworks. Selecting only one structural element will result in discontinuous edge information of the detected artwork and generally cannot get a relatively complete edge contour of the artwork [13]. Using multi-scale structural elements can effectively solve the problem of edge discontinuity in artworks. Structural elements with different scales are used to calculate the image of artworks, and large-scale structural elements can effectively filter the image of artworks; Small-scale structural elements can be targeted to detect the contour of target features of artworks [14]. Two structural elements of different scales are combined to detect the edge of the image of the artwork.

The expression for defining multi-scale structure elements is as follows:

\[
nB = B_1 \oplus B_2 \oplus \ldots \oplus B_n \tag{3}
\]

In formula (3), \( n \) is the scale parameter; \( B \) is a structural element.

Using multi-scale and multi-structure element morphological edge detection operators, the expression for processing artworks is as follows:

\[
f_B(x,y) = \frac{1}{2} \left\{ f \left[ \left( x, y \right) \ominus b \left( m,n \right) \oplus nB \right] \ominus f \left[ x, y \right] \right\} \tag{4}
\]

In formula (4), \( f_B(x,y) \) is the image of artworks after multi-scale mathematical morphology processing; \( b \) is a structural element.

The circular structure element with the size of 3x3 and the rectangular structural element with the size of 6x2 are used as the structural element of the edge enhancement for the image of the artwork. The expression of the structural element is as follows:

\[
b_1 = \begin{bmatrix} 1 & 0 & 0 \\ 0 & 1 & 0 \\ 0 & 0 & 1 \end{bmatrix} \tag{5}
\]

\[
b_2 = \begin{bmatrix} 1 & 1 & 1 & 1 & 1 \\ 1 & 1 & 1 & 1 & 1 \\ 1 & 1 & 1 & 1 & 1 \end{bmatrix} \tag{6}
\]

\[
b_3 = \begin{bmatrix} 1 & 1 & 0 & 1 & 0 \\ 0 & 0 & 1 & 1 & 1 \end{bmatrix} \tag{7}
\]

According to the structural element expression of formula (5) to (7), the circular structure element with the size of 3x3 is firstly used to expand and erode the image of the artwork. Then the rectangular structure element with the size of 6x2 is used to expand and erode the image of the work of art and complete the edge enhancement of the image of the artwork.

B. Style Transfer Network Structure of Convolution Neural Network

The convolution neural network is selected as the deep learning algorithm for the adaptive style transfer of artworks. A convolution neural network is a series of feedforward neural
networks with depth, including convolution operation. Convolution neural network has strong learning ability because of their unique network structure [15]. Unlike the full connection strategy of traditional neural networks, each neuron of the convolution neural network is only connected with some neurons of other layers, and the parameters of the convolution kernel are shared, making the operation simple and efficient, and can learn on large data sets [16, 17]. A convolution neural network comprises the input layer, convolution layer, activation function [18], pooling layer [19] and full connection layer. Among them, the selection of the activation function in the activation layer greatly impacts the migration result of the adaptive style transmission of paintings. The ReLU function is selected as the activation function in the activation layer of the convolution neural system. The ReLU function is a function with the maximum value. The calculation formula of the ReLU function is as follows:

\[ f(x) = \max(0, x) \]  

(8)

The ReLU function mimics the biological neurons in the human brain and has the characteristics of dispersion and sparsity. The ReLU function is the most commonly used activation function at present. When the convolutional neural network uses the ReLU function as the activation function, there is no problem of gradient disappearing.

Maximum and average pooling are two common pooling ways of pooling layers in convolutional neural networks. The maximum pooling method is selected as the pooling operation of adaptive style migration of artworks. The full connection layer acts as a "classifier", mapping the learned features of artworks to the sample space. Full connection operation will cause a huge amount of parameters, so it is generally only used in the last few layers of the convolutional neural system to exploit the feature vectors of artworks to complete some classification tasks.

C. Laplace Operator and its Improvement

1) Laplace operator: The Laplace operator is an important algorithm in the adaptive style transfer processing of artworks. The Laplace operator is an edge point detection operator independent of an edge direction. It responds more strongly to isolated pixels in the picture of paintings than to edges or lines [20]. Before applying this operator to the adaptive style transmission of paintings, it needs to smooth the image of artworks. The Laplace operator is a second-order differential operator. Let the artwork image be a continuous, binary function \( f(x, y) \), and its Laplace operation is defined as:

\[ \nabla^2 f = \frac{\partial^2 f}{\partial x^2} + \frac{\partial^2 f}{\partial y^2} \]  

(9)

The Laplacian operator of the image of artworks can be simplified as follows:

\[ g(i, j) = 4f(i, j) - f(i+1, j) - f(i-1, j) - f(i, j-1) \]  

(10)

Formula (10) can also be expressed in the form of convolution, and the expression is as follows:

\[ g(i, j) = \sum_{r=-k}^{k} \sum_{s=-l}^{l} f(i-r, j-s)H(r, s) \]  

(11)

In formula (11), the sampling of \( H(r, s) \) is as follows:

\[ H_1 = \begin{bmatrix} 0 & -1 & 0 \\ -1 & 0 & -1 \\ 0 & -1 & 0 \end{bmatrix} \]  

(12)

In the process of adaptive style transmission of paintings, the Laplace operator of the function is realized with the help of templates. The effect of template sampling will directly affect the effect of adaptive style transfer of artworks.

2) Improved Laplace operator: The laplacian function improves the Laplace operator, and the log operator represents the improved operator. The log operator is used to improve the application performance of the Laplace operator in the adaptive style transfer of artworks. The log operator is an improved operator based on the Laplace operator. The log operator first performs Gaussian convolution filtering on the artwork image to reduce noise [21], then uses the Laplace operator to detect the edge of the artwork image after noise reduction. The expression of the Gaussian convolution function \( G(x, y) \) used to improve the Laplace operator is:

\[ G(x, y) = \exp \left[ -\frac{x^2 + y^2}{2\sigma^2} \right] \]  

(13)

In formula (13), \( X \) and \( Y \) respectively represent the row coordinates and column coordinates of the pixel points of the artwork image; \( \sigma \) represents the standard deviation of pixels.

The original artwork image \( f(x, y) \) is first convolved with the Gaussian function \( G(x, y) \) to obtain the smoothed artwork image \( g(x, y) \). The smoothed artwork image \( g(x, y) \) is solved with the Laplace operator differential of the convolution. The expression of the operation process is as follows:

\[ g(x, y) = G(x, y) \otimes f(x, y) \]  

(14)

\[ \nabla^2 \left[ f(x, y) \otimes G(x, y) \right] = f(x, y) \otimes \nabla^2 G(x, y) \]  

(15)

\[ \nabla^2 G(x, y) = \left( \frac{x^2 + y^2 - 2\sigma^2}{2\sigma^2} \right) \exp \left( \frac{-\log(x^2 + y^2)}{2\sigma^2} \right) \]  

(16)

In formula (14) to (16), \( \otimes \) is a convolution symbol; \( \nabla \) is the differential sign; \( \log \) is an improved Laplace operator.
Because the edge detected by the log operator is not very smooth and has a lot of noise [22], on the basis of using the operator detection, the two-dimensional digital filter is used again to filter and detect the image of the work of art. The detection process is as follows:

1) The filtered artwork image \( y(n) \) can be expressed as:

\[
y(n) = h(n) \times x(n) = \sum_{k=0}^{K-1} h(k) \times x(n-k) \quad (17)
\]

In formula (17), \( h(n) \) is the non-recursive filtering coefficient, \( h(k) = \log(x, y) \); \( x(n) \) is the input artwork image; \( K \) is the filter length, and the order is \( K - 1 \). The coefficient matrix of the two-dimensional digital filter is rotated 180° to create a convolution kernel [23], and obtain the convolution kernel expression as follows:

\[
H_{ij} = \begin{bmatrix} 0 & 1 \\ 1 & 0 \end{bmatrix} M_{ij} \quad (18)
\]

In formula (18), \( H_{ij} \) is the convolution kernel created by rotating the filtering coefficient matrix of the two-dimensional digital filter by 180°; \( M_{ij} \) is the edge operator.

2) Two-dimensional convolution is used to realize the filtering operation of artwork images. The expression of the two-dimensional convolution operation of artwork images is:

\[
C_{ij}(s,t) = \sum_{m=0}^{M_r-1} \sum_{n=0}^{M_c-1} H_{ij}(m,n) \Theta B(s-m,t-n) \quad (19)
\]

In formula (19), \( C_{ij}(s,t) \) is the artwork image after filtering operation; \( M_r \) and \( M_c \) are the number of rows and columns of convolution kernel \( H_{ij} \), respectively: The number of rows and columns \( s \) and \( t \) of the picture of the artwork meet the requirements of \( 0 \leq s < M_r + N_r - 1 \); \( 0 \leq t < N_c + M_c - 1 \); \( N_r \) and \( N_c \) are the number of rows and columns of the input artwork image matrix \( B \).

3) The filtered data of the artwork image are added, and the expression is:

\[
S = \sum_{i=1}^{4} C_{ij}(s,t) = \sum_{i=1}^{4} \sum_{m=0}^{M_r-1} \sum_{n=0}^{M_c-1} H_{ij}(m,n) \Theta B(s-m,t-n) \quad (20)
\]

In formula (20), \( S \) is the sum result of the filtered data of the artwork image.

Through the above process, the threshold segmentation of the artwork image is completed, and the edge contour information of the artwork image matrix \( B \) is obtained.

D. Adaptive Style Transfer of Artworks Based on Color and Brightness

In the stylized result graph of artworks, some of the generated results of the stylized transfer algorithm of the traditional neural network have color confusion, which affects the overall artistic effect of adaptive style transmission of paintings in visual effect. Inspired by the color migration algorithm in style migration [24], in the process of adaptive style transmission of paintings, the color details and brightness details of the content picture of paintings are retained by using the migration of color and brightness.

\( X_c \) represents content image, \( X_s \) represents style image, \( X_j \) represents after color matching and migration, \( x_i = (R, G, B)^T \) represents the pixel color of the input artwork content image and the input artwork style image. The color matching of the content image \( X_c \) of the artwork is transferred to the style image \( X_s \) of the artwork, and gets the new artwork image \( X_j \). Then, before and after the style transfer, the pixel value of the artwork image has the following mathematical mapping relationship:

\[
x_{cj} \leftarrow A x_s + b \quad (21)
\]

In formula (21), \( A \) represents a matrix of size of 3×3; \( b \) represents a vector of size of 3 ×1; \( x_s \) shows the pixel value of the input artwork style picture; \( x_{cj} \) shows the pixel value change result of the artwork image after color style migration.

In order to obtain the pixel value \( x_{cj} \) after color-matching migration when the painting style is transferred, the matrix \( A \) and vector \( b \) in formula (21) must be determined first. The color-matching migration process of artworks will be described in detail below.

\( \mu_c \) and \( \mu_s \) are defined to represent the mean value of each pixel of the input artwork content image and the input artwork style image in RGB color space; \( \sigma_c \) and \( \sigma_s \) represent the covariance of the input artwork content image and the input artwork style image, respectively. According to the definition of mean and covariance, it can get:

\[
\begin{align*}
\mu &= \sum_i x_i / N \\
\sigma &= \sum_i (x_i - \mu)(x_i - \mu)^T / N
\end{align*}
\]

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After the color matching of the artwork content image $X_C$ is transferred to the artwork style image $X_S$, when the new artwork image $X_J$ is obtained, the pixel color mean value and pixel covariance of $X_J$ must be consistent with the pixel color mean value and pixel covariance of $X_C$, that is, the existing relationship between the content picture and the style picture of the paintings;

$$H_C \text{ and } H_S \text{ represent the brightness channel of the content image and style image of the artworks; } H_J \text{ refers to the brightness pixel of the style picture of the paintings after the brightness migration; } B_C \text{ and } B_S \text{ show the medium brightness of the content picture and the style picture of the paintings; } d_C \text{ and } d_S \text{ represent the standard deviation between the content image of the work of art and the style image of the work of art, then the brightness transfer from the content image of the work of art to the style image of the work of art can be expressed as follows:}

$$H_J = d_C (H_S - B_S) / d_S + B_C \quad (28)$$

Through the above process, it can complete the color and brightness transfer process of an artwork's image's adaptive style transfer and improve the level of the artworks image's adaptive style transfer through the color and brightness transfer of the artwork's image.

E. Self-Adaptive Style Transmission of Paintings on the Basis of Semantic Information

1) Determination of loss function: In the process of adaptive style transmission of paintings, to restrict the image transfer of artworks without redundant semantic information, based on the Laplace operator and convolution neural network method, the semantic information transmission of adaptive style transfer of paintings is realized by using Laplace operator to enhance the semantic information method. The method of the Laplacian operator to enhance semantic information increases the deeper information in the convolutional neural system and fully considers the loss of style transmission of artworks. A grim matrix is used to construct the style transfer loss of the convolutional neural network.

The content loss expression of the convolutional neural system for adaptive style transfer is as follows:

$$L_c = L_{sup} + \gamma L_{\delta} + \delta L_{\gamma} \quad (29)$$

In formula (29), $L_c$ represents the losing function of the content picture of the artwork, $L_{sup}$ represents the loss function of the Laplace operator, $L_{\delta}$ and $L_{\gamma}$ represent the adjustment coefficient of the loss function, and $\delta$ and $\gamma$ are the weights of the latter two loss function terms. In order to prevent the
network from over-fitting due to excessive loss during the training of the convolutional neural network, which affects the final generalization effect, both $\delta$ and $\gamma$ are set to 0.5.

The original artwork image $P_r$ and the migrated artwork image $P_o$ are input into the loss network to calculate the mean square loss of the artwork adaptive style transfer so that the forward propagation error of the convolution neural network can be corrected. The structure diagram of the content loss network of the artwork image is shown in Fig. 1.

![Fig. 1. Graphic content loss network structure of artworks.](image)

In Fig 1, the improved Laplace operator log operator is used to perform the Laplace transform of the artwork image. The loss function $L_{lap}$ of the Laplace operator is calculated as follows:

$$L_{lap} = \frac{C_{lL}}{W_l L_c} \sum_{ij} [\text{lap}(F_c(P_r)) - \text{lap}(F_c(P_o))]$$

(30)

In formula (30), the value of $l_c$ is Relu4_3. The Laplace operator used in the loss function is the improved Laplace operator log operator. $P_r$ is input into the network to output Relu4_3. The Laplace operator is used to filter; $P_o$ repeats the above steps. $L_4$ error is performed by backpropagation to correct the error. The reason for using $L_4$ error is that $L_4$ has fewer constraints on the solution, so $L_4$ error has a better generalization effect than $L_2$ error.

Using the Laplace operator as the loss function that emphasizes the feature extraction of the convolution neural network will not modify the original artwork image with the Laplace operator, so adding this loss item will not affect the actual quality of the artwork image after style transfer. In the last two terms of the loss function, the Relu2_2 and Relu3_3 of the convolution neural network are the standards to judge the loss of the content of artworks. The deeper features of the artwork can be obtained using the deeper convolution neural network output. Suppose the deeper image features of the original artwork are consistent with the image features of the artwork after style transfer. In that case, the artwork’s image features better constrain the original image's semantic content. After adaptive style transfer, the image of the artwork will be closer to the original artwork image to constrain the artefacts unrelated to the original artwork's image semantics due to style transfer.

The calculation method of $L_{r1}$ in formula (29) is as follows:

$$L_{r1} = \frac{C_{rL}}{W_l L_c} \sum_{i,j} [F_i(P_r) - F_i(P_o)]^2$$

(31)

The calculation method of $L_{r2}$ is the same as that of $L_{r1}$. The only difference between the two is that the value of $l_c$ and is Relu2_2 and Relu3_3, respectively.

2) Structure design of adaptive style migration network: The overall structure of the adaptive style transfer method of artworks based on the Laplace operator is shown in Fig. 2.
It can be seen from the structure chart of adaptive style transfer of artworks in Fig. 2 that the Laplace loss term between the content image of paintings and the stylized picture of paintings is taken as the control dimension of the stylized detail feature of artworks image and added to the total loss function of adaptive style transfer. At the same time, the impact of the color brightness information of the content image and style image on the stylized visual effect is comprehensively considered. Detailed analysis of the specific implementation steps of adaptive style transfer of artworks images is as follows:

1) **Input** the artwork content image and artwork style image and carry out edge enhancement processing on them; during the edge enhancement process, effectively filter the noise in the artwork image and reduce the impact of the noise in the artwork image on the subsequent adaptive style migration process and the calculation of each loss item.

2) **Convert** the content image and style image of the artwork from RGB space to Lab space, realize the separation of image brightness channel L and colour channels a and b, transfer the brightness information of the content image of the artwork to the style image of the artwork, and retain the colour information of the content image of the artwork.

3) **Input** the content image of the artwork after the brightness transfer and the initial stylized result map into the convolution neural network, and carry out the statistics of the content loss items of the convolution neural system to obtain the loss function of the content picture of the artwork.

4) **Make** statistics of Laplace operator loss term of convolution neural network.

5) **Evaluate** the style losing function of the style picture of the art picture after denoising.

6) **Construct** a new total loss function and optimize iteratively in the convolution neural network; use the gradient descent method for continuous iteration of the convolution neural system; and output the adaptive style transfer result chart of paintings. In the process of style transmission of paintings, all loss items are calculated simultaneously.

### III. RESULTS

In order to verify the adaptive style transfer method of artworks based on the Laplace operator and the effectiveness of adaptive style transmission of paintings, the paintings collection in the network is selected as the test set, which contains 254 artworks in total. The artworks collection is divided into the content of the artworks set and the artworks style set, which contain 185 images and 69 images, respectively. The method in this paper is programmed by Matlab software, which is run to test the adaptive style transfer of the test set of artworks.

In this paper, a convolution neural network is used as the method of adaptive style transmission of paintings. The ReLU function is selected as the activation function of the convolution neural network. The curve of the ReLU function is shown in Fig. 3.

The method in this paper is used for adaptive style transmission of paintings. The loss curve changes during the operation of the method are shown in Fig. 4. At the same time, in order to verify the effectiveness of the proposed method, the residual neural network method in the same method reference [10] and the original Convolutional neural network are selected as the comparison method, and the loss curve is generated at the same time.
A content picture of artwork is selected from the content set of the artwork test set. The content image of the original artwork is shown in Fig. 5.

Two artworks style images are selected from the content set of the artworks test set. The style image of the original artwork is shown in Fig. 6.

The method in this paper uses the Laplace operator to conduct adaptive style transmission for paintings. The results of adaptive style transfer are shown in Fig. 7.

In order to further verify the adaptive style transfer performance of the method in this paper for artworks, the structural similarity index is selected as a measure of the structural similarity of artworks before and after style transfer. In order to verify the consistency of the structure of the artwork image and the original artwork image after the adaptive style transfer, structural similarity is used to determine the similarity between the two images. The calculation formula of structural similarity is as follows:

$$SSIM(x, y) = l(x, y)c(x, y)s(x, y)$$ (32)

In formula (32), $l(x, y)$ is the brightness estimation, $c(x, y)$ is the contrast estimation, and $s(x, y)$ is the structure estimation. The method in this paper is used to
conduct adaptive style transfer for artworks. The statistical results of structural similarity are shown in Fig. 8.

From the experimental results in Fig. 4, it can be seen that the total loss of this method in adaptive style transfer of art works is relatively low. The total loss of the residual neural network method in reference [10] is relatively high, and the total loss of the original Convolutional neural network is the highest among the three methods. This proves that the introduction of Loss function and Laplace operator in the process of adaptive style transfer of art works in this method reduces the loss in the process of adaptive style transfer of art works, and improves the style transfer performance of art works. The experimental results in Fig. 4 show that the adaptive style transfer of artworks using the method in this paper can improve the visual quality of the adaptive style transmission results of paintings and have better performance of minimizing the loss function.

From the experimental results in Fig. 7, we can see that using the method in this paper can achieve adaptive style transmission of paintings. The migrated paintings effectively retain the content of the original artworks content image and the style of the artworks style image. The artworks after style transfer can meet the user's adaptive style transfer needs. This method can achieve style transmission of paintings for different styles, and the effectiveness of style transfer is high.

Stylization is an art editing work, and the quality of artistic works varies from person to person, so evaluating the quality of stylized images is a very subjective task. As scientific research, the image adaptive style transfer method should have rigorous and quantifiable evaluation indicators. The method in this paper is used to conduct adaptive style migration of artworks. The peak signal-to-noise ratio and migration time of style migration results are shown in Table I.

![Fig. 8. Statistical results of structural similarity.](image)

### TABLE I. PERFORMANCE TEST RESULTS OF ADAPTIVE STYLE MIGRATION

<table>
<thead>
<tr>
<th>Content image number</th>
<th>Style image number</th>
<th>Style transfer result</th>
<th>Peak signal-to-noise ratio /dB</th>
<th>Migration time /ms</th>
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<td>35.6</td>
<td>354</td>
<td></td>
</tr>
<tr>
<td>2</td>
<td>A</td>
<td>33.4</td>
<td>285</td>
<td></td>
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<tr>
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<td>A</td>
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<td></td>
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<tr>
<td>4</td>
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</tr>
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<td>9</td>
<td>C</td>
<td>34.5</td>
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</tr>
</tbody>
</table>

IV. DISCUSSION

It can be seen from Fig. 3 that when the ReLU activation function inputs a negative value, the output result is 0, which means that only some neurons are activated at the same time, which makes the convolutional neural network more sparse. When this method is used for adaptive style transmission of paintings, the convolutional neural network is set up, and the ReLU function is used as the activation function. When the adaptive style transfer of artworks, there is no problem of gradient disappearing. The convergence speed is fast, the calculation is simple, and the efficiency is high, effectively improving the effectiveness of the adaptive style transfer of artworks.

V. CONCLUSION

As an excellent neural network in deep learning, convolution neural network has achieved remarkable results in target detection, classification and recognition, especially in image style transfer. Therefore, this article offers an adaptive style transmission method for paintings on the basis of the
Laplace operator. The combination of the Laplace operator and convolution neural network can better retain the semantic information of the content image in the stylized image and improve the visual effect of the stylized image. The experimental results show that the proposed method can obtain ideal style transfer results on the premise of ensuring the quality of style transfer images. The visual effect of style transfer is largely subjective, so the next step will be to add more objective experiments to verify the effectiveness and practicality of the method.

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Application of Virtual Reality Technology in the Design of Interactive Interfaces for Public Service Announcements

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Abstract—With the development of technology, more and more public service announcements are being designed with interactive interfaces. There are many different ways to interact with interactive interfaces, and using appropriate design methods can expand the impact of PSAs. The study incorporates image pre-processing methods based on virtual reality, using the cvtColor grey scale function and median filtering method to process the images, an iterative approach to camera positioning method design, and subsequent performance testing of the research algorithms. The test results showed that the peak signal-to-noise ratio of the research method was 13.390dB in the image pre-processing process and 35.635dB on lightly shaded images; in the error test, the rotational mean error of the research method was approximately 4.2degrees at four reference points; and in the image plane reprojection test, 70% of the points of the research method almost coincided with the original point position. The method generated 1203 designs with 40 reference points. The experimental results show that the research method can effectively design interactive interfaces for PSAs in virtual reality environments, and can propose more design solutions, and has better performance in virtual reality environment positioning.

Keywords—Virtual reality; interactive interfaces; interface design; greyscaling; image pre-processing

I. INTRODUCTION

With the increasing abundance of entertainment content on the Internet, people's acceptance of advertising is becoming lower, which puts forward higher requirements for the dissemination form of public service advertising [1]. Public service advertising has a certain degree of specificity, and traditional advertising forms and communication methods lack interaction with the audience, often making it difficult to achieve the expected promotional effect [2]. At present, the purpose of public service advertising is not only to disseminate information, but more importantly, to awaken the public's sense of social responsibility and promote social civilization and progress [3]. How to design public service advertisements that can stimulate public emotions and increase participation has become an urgent problem to be solved. At present, common interactive design methods for public service advertisements include socialized interaction, emotional guidance, etc. However, due to the limitations of communication carriers and audience groups, the design effect of public service advertisements is not ideal. With the continuous development of virtual reality technology, its application in various fields is becoming increasingly widespread. Virtual reality technology provides a new interactive experience that can bring users into a virtual real world, providing a more realistic and immersive experience [4]. The application of virtual reality technology in the interactive interface design of public service advertisements can not only improve the user experience, but also allow users to have a deeper understanding of the information that public service advertisements are intended to convey [5]. In virtual reality technology, visual positioning is a very important technology, and more realistic positioning can improve the immersion and realism of virtual reality. Therefore, a study on the interactive interface design of public service advertising has proposed an interface design method that integrates virtual reality technology and visual positioning, in order to provide more reference solutions for the interactive interface design of public service advertising.

The research mainly focuses on four aspects. The first part discusses the current research achievements in interactive interface design and virtual reality technology. The second part elaborates on the technical methods used in the research and design of interactive interface design methods for public service advertisements that integrate virtual reality technology. The third part is to test the performance of the research method. The last part is a discussion and summary of the entire text.

II. RELATED WORKS

Public service announcements (PSAs) are one of the most important tools for spreading healthy moral values in society, and the large number of PSAs being broadcast has led experts to realise the value of applying interactive interfaces to PSAs. A number of experts have conducted research on interactive interface design, and Zong J and others have proposed a system based on Lyra 2 to address the problem that interactive graphic interface design is limited to static output. The process uses a trial-and-error approach to define the attributes of interaction and then represent the interaction on the editorial visualisation. Experimental results show that the proposed method is effective in expressing interactions in multiple types of interface design and has a high runtime speed [6]. Oulasvirta A et al. propose a combinatorial optimisation approach to the problem of coding in the design of interactive GUIs. The process identifies the problem type, selects the focus of the design goals and formulates the functionality specifically. Experimental results show that the proposed approach has good human-computer interaction and can
effectively design interactive interfaces [7]. Latif S et al. propose a correlation method to support the tight coupling of data text and diagrams for the adoption of complex data for graphical text interfaces. The process developed a hybrid active interface to enable users to construct interactive interface references between the chart texts. The end result shows that the method is an effective way to provide a wide range of users with the ability to write interactive data documents [8]. Leung J and Cockburn A propose a structured design framework for improving the control and application performance of salient display techniques in user interface design, involving parameter control and highlighting constructs (PCCH). The process collected and reviewed a range of potential human factors and effect measures regarding the interactive impact of highlighting techniques. The results show that the framework is effective in improving the user's understanding of salient information[9]. Two scholars, Jianan L and Abas A, analyse human-computer interaction and the safety and comfort of car driving. The process provides an effective analysis of future human-machine interaction modes and proposes a method for designing human-machine interface interactions with cars based on existing technologies, providing some reference for future car human-machine interface design [10].

There have also been studies on virtual reality technology by experts. Qiu W et al. propose a parallel hierarchical control method based on virtual reality in order to be able to construct a large-scale high-fidelity traffic scene simulator. The process modifies and generates the original data of the traffic elements and assigns all the data to the corresponding tasks based on the proposed spatially parallel slicing method. The results show that the method can effectively ensure the fidelity of the visual scene and better allocate computational resources [11]. Hite R L et al. propose an effective combination of three-dimensional modelling (3D), support for haptic feedback (HE) and interactive virtual reality (VR) in order to assist all K-12 learners in their curriculum on human heart and physiology, and apply it to the curriculum. The results show that the application of 3D-HE-VR technology can effectively provide learners with powerful system representations and student-driven interactions, significantly driving system learning forward [12]. Gu K et al. propose a virtual reality-depth image based rendering (DIBR) method for effective assessment of the quality of synthetic images. To accomplish this goal, the study introduced a new blind image quality assessment (IQA) method developed through multi-scale natural scene statistical analysis (MNSS). The results show that the method has a more comprehensive performance [13]. Rockcastle S et al. designed an experiment for illumination perception in a real space using a series of illuminators in order to make valid observations of images in real and simulated spaces. The process invited 53 people to perform immersive perception. The results showed that there was little difference between the perceived ratings of the designed virtual reality scene and people in the real space in a well-lit scene, which had a more realistic effect [14]. Yeh H C et al. used virtual reality technology in order to enable students to better grasp the language content and language skills in English and to reform the curriculum with virtual reality, interactive, audio and structured teaching. The results showed that students were better able to learn cross-culturally with VR technology and significantly improved their own English language learning skills [15].

In summary, although virtual reality technology has been studied and applied in many fields, there is a lack of research in the field of interactive interface design for public service announcements. In view of this, the study proposes an interactive interface design method for public service announcements that incorporates virtual reality technology in order to effectively design interactive interfaces for public service announcements and provide more references for the interactive interface design industry.

III. RESEARCH ON THE INTERACTIVE INTERFACE DESIGN METHOD OF PUBLIC SERVICE ADVERTISEMENT INTEGRATING VIRTUAL REALITY TECHNOLOGY

A. Design of Visual Orientation Methods for Interactive Interfaces

Interaction design supports people to communicate and interact with tools and objects in their working lives. In the Internet+ environment, PSAs are increasingly focused on interacting with their audiences to enhance the interest and content richness of functional PSAs [16-17]. Virtual reality technology is widely used in interactive games and software, and can be effectively applied to the design of interactive interfaces for PSAs due to its immersive nature and the variety of ways in which it can be used [18-19]. The main reference elements for interactive design are shown in Fig. 1.

As shown in Fig. 1, there are six principles and four elements to be considered when designing interactions. The four elements are: the object is the target user of the interaction; the behaviour is the action of the user during the interaction; the function is the interaction module and scenario that can be provided to the user during the interaction; and the technology is the implementation of the interaction. The technology is the control system and information algorithm that realises the interaction. Public service announcements (PSAs) have certain characteristics that differentiate them from ordinary advertisements in terms of planning and filming because of their special meaning of existence. The characteristics of an interactive PSA are shown in Fig. 2.

![Fig. 1. Reference principles and elements of interaction design.](www.ijacsa.thesai.org)
As can be seen from Fig. 2, interactive PSAs are a type of PSA, and so, like other PSAs, have two main characteristics: social input and emotional brewing. The development of the Internet has enabled information to spread faster and over a wider area, and social interaction has given rise to numerous information platforms through which the public can access public service information and play a supervisory role over social welfare. Public service announcements (PSAs) are more emotionally charged than ordinary advertisements, and interactive PSAs are richer in ways of communicating emotions, bringing a wider variety of emotional experiences to viewers as they interact with each other [20-21]. When designing interactive PSAs using virtual reality technology, the first step is visual positioning, analysing the information with location markers and solving for camera position and pose [22]. Before virtual reality can be used for spatial positioning, 2D markers need to be created. 2D markers are deformed in 3D space due to perspective, and 2D markers captured by the camera are more likely to be trapezoidal in shape, requiring a coordinate system conversion for recognition. In the pixel and image coordinate system, the dimensions of a single pixel on the u and v axes are shown in Eq. (1).

\[
\begin{align*}
    u &= x / d_x + u_0 \\
    v &= y / d_y + v_0
\end{align*}
\]  

(1)

In Eq. (1), \(d_x\) represents the unit size of a single pixel on the x-axis; \(u_0\) and \(v_0\) represent the coordinates of the centroids in the image coordinate system. The matrix relationship between pixels and images is shown in Eq. (2).

\[
\begin{bmatrix}
    u \\
    v \\
    1
\end{bmatrix} =
\begin{bmatrix}
    1 / d_x & 0 & u_0 \\
    0 & 1 / d_y & v_0 \\
    0 & 0 & 1
\end{bmatrix}
\begin{bmatrix}
    x \\
    y \\
    1
\end{bmatrix}
\]  

(2)

In Eq. (2), the matrix is normalised by \(u\) and \(v\). The relationship between a point in the world coordinate system and its position in the camera coordinate system is shown in Eq. (3).

\[
P_c = RP_w + t
\]  

(3)

In equation (3), \(P_c\) represents the coordinates of the point in the camera coordinate system; \(P_w\) represents the coordinates of the point in the world coordinate system; and \(R\) is the rotation matrix. After the rotation of the coordinate system by the rotation matrix, the relationship is matrixed and normalised to obtain the coordinate matrix, as shown in Eq. (4).

\[
\begin{bmatrix}
    X_c \\
    Y_c \\
    Z_c
\end{bmatrix} =
\begin{bmatrix}
    r_{11} & r_{12} & r_{13} & t_x \\
    r_{21} & r_{22} & r_{23} & t_y \\
    r_{31} & r_{32} & r_{33} & t_z
\end{bmatrix}
\begin{bmatrix}
    X_w \\
    Y_w \\
    Z_w
\end{bmatrix}
\]  

(4)

In Eq. (4), \(X_c\), \(Y_c\) and \(Z_c\) represent the coordinates of the point in the three axes of the camera's coordinate system respectively; \(X_w\), \(Y_w\) and \(Z_w\) represent the coordinates of the point in the three axes of the world coordinate system respectively. Since the camera lens uses a convex lens, there is bound to be aberrations when recording the image and the image needs to be corrected. With the centre of the aberration as the central point, the deviation along the radial direction is the radial aberration, and the radial aberration error is shown in Eq. (5).

\[
\begin{align*}
    e_{r_w} &= u(k_r r^2 + k_t r^4 + k_s r^6) \\
    e_{r_v} &= v(k_r r^2 + k_t r^4 + k_s r^6)
\end{align*}
\]  

(5)

In Eq. (5), \(e_{r_w}\) is the radial error of the point position on the u-axis of the pixel coordinate system; \(e_{r_v}\) is the radial error of the point position on the v-axis of the pixel coordinate system; \(k\) represents the radial aberration coefficient. The deviation along the tangent line with the centre of distortion as the centre point is the tangential distortion, and the tangential distortion error is shown in equation (6).

\[
\begin{align*}
    e_{t_w} &= 2p_{uw} + p_r(r^2 + 2u^2) \\
    e_{t_v} &= p_r(r^2 + 2v^2) + 2p_{uv}
\end{align*}
\]  

(6)

In Eq. (6), \(p\) represents the tangential aberration factor; \(e\) represents the aberration error of the coordinate axes. Thin lens aberrations occur when the camera lens is not parallel to
the CDD, and the thin lens aberration error is shown in Eq. (7).

\[
\begin{align*}
\epsilon_{\text{aa}} &= s_x (a^2 + v^2) \\
\epsilon_{\text{ao}} &= s_y (a^2 + v^2)
\end{align*}
\]  

(7)

In Eq. (7), \(s\) represents the thin lens aberration parameter.

B. Design of an Interactive Interface Camera Positioning Method Based on Image Pre-processing

In the interactive interface of a PSA, the camera needs to extract the logo image for visual positioning, and using a higher resolution image can increase the accuracy of the extraction. However, the higher the resolution of the image, the more storage space it takes up and the greater the pressure on the system when processing the image [23]. The 2D logo used in the study is only in black and white, so the logo image can be extracted by greyscaling the image, which does not affect the visual localisation of the camera and reduces the amount of extra information contained in the image. The study uses the cvColor greyscale function, which comes with the OpenCV vision library, to greyscale the images. The camera converts the image colour during imaging in the RGB domain, resulting in the appearance of non-linear noise[24]. The median filtering method takes care of bright and dark point noise and removes sharp signals from the signal. When performing telecoded image processing, reference points need to be extracted and the maximum contour containing all points is calculated as shown in Eq. (8).

\[
\begin{align*}
\{(x_1, y_1), \ldots, (x_n, y_n)\} &\subseteq \text{maxQuad} \{ (x_1, y_1), (x_2, y_2), \ldots, (x_n, y_n) \}
\end{align*}
\]  

(8)

In Eq. (8), \(\text{maxQuad}\) represents the maximum contour that contains all points. As the operator interacts, the captured image will change due to the operator's actions and the study uses a perspective transformation to process the image. The principle of the perspective transformation is shown in Fig. 3.

As can be seen in Fig. 3, the captured image differs from the real one in shape in the final two-dimensional form because of the angle formed between the lens and the surface of the object and the perspective changes in the lens field of view, which are large near and far away. The coordinates of the original image and the pixel points of the transformed image are obtained after linear and translational transformations, as shown in Eq. (9).

\[
\begin{align*}
X' &= \frac{a_{11}X + a_{12}Y + a_{13}}{a_{31}X + a_{32}Y + a_{33}} \\
Y' &= \frac{a_{21}X + a_{22}Y + a_{23}}{a_{31}X + a_{32}Y + a_{33}}
\end{align*}
\]  

(9)

In Eq. (9), \(X'\) and \(Y'\) represent the coordinates of the pixel points on the X and Y axes respectively. The Z-axis coordinates are both 1 after the transformation. In order to reduce the computational pressure of extraction, the 2D logo is rotated so that the logo is at a computationally convenient angle. The rotation angle of the logo is calculated as shown in Eq. (10).

\[
\text{angle} = \text{arc} \begin{pmatrix} y_2 - y_1 \\ x_2 - x_1 \end{pmatrix}
\]  

(10)

In Eq. (10), \(\text{angle}\) represents the rotation angle; \(x_1\) and \(y_1\) are the centroid coordinates of the top left corner; \(x_2\) and \(y_2\) are the centroid coordinates of the top right corner. When locating near-code images, only the pixel coordinates of the image centroid need to be extracted. However, when searching for near-code image finding, the position image and the parent contour can interfere. The study uses an iterative method to mitigate the interference by establishing a hierarchical relationship for each contour and then removing the position image. This is shown in Fig. 4.

As seen in Fig. 4, the complete labelled image contour contains three position contours as well as the parent contour. The position contours are the first to be removed by the iteration as they differ most from the near-code image. The separate small squares in the near-code image are combined while the parent contours are removed, resulting in an image containing only the near-code image. The complete image pre-processing process is shown in Fig. 5.

As can be seen from Fig. 5, in the first half of the pre-processing process, the colour and interference noise of the image are processed, and the outline of the 2D logo is extracted and the area of the maximum outline is calculated. The maximum contour area is used as the judging condition, and the area requirement is met before entering the second half of the processing process for the far-code and near-code images, and when the maximum contour area exceeds the judging value, the image is judged as a near-code image, otherwise it is judged as a far-code image. After the extraction of information from the far code image or the near code image, the image pre-processing of the 2D logo is completed. The 3D reference point on the camera coordinate system is shown in Eq. (11).

\[
P'_t = RP''w + t
\]  

(11)
In Eq. (11), \( P^w_i \) represents the coordinates of the 3D reference point on the world coordinate system; \( R \) is the rotation matrix obtained after the reference point enters the camera coordinate system; \( t \) is the translation of the reference point after it enters the camera coordinate system. The reference point is subject to some error in actual positioning, and the study iterates over the object-squared residuals to complete the estimation of the camera’s position pose. The object-squared residuals are calculated as shown in Eq. (12).

\[
E_g = \| \hat{p}_i - \hat{p}_i^e \| = \left\| p_i - \frac{R(P^w_i - C)}{R^T(P^w_i - C)} \right\| = \frac{1}{Z_i^e} \left\| P^w_i - P^e_i \right\|
\]  

In Eq. (12), \( P^i \) and \( P^e_i \) are the reference estimation points; \( \hat{p}_i \) represents the flush coordinates of the image reference point; \( \hat{p}_i^e \) represents the flush coordinates of the estimation point; \( C \) represents the coordinates of the origin on the world coordinate system; and \( Z_i^e \) represents the depth value of the reference point on the camera coordinate system. The object-squared residuals and image-squared residuals are shown in Fig. 6.

As seen in Fig. 6, the depth of the spatial reference point affects the object residuals when the image residuals are the same, the deeper the object residuals the greater the object residuals. When performing the orthogonal iteration, the error between the theoretical point and the reference point in the 2D plane is calculated as shown in Eq. (13).

\[
w_{li} = \frac{1}{n} \sum_{i=1}^{n} \left( \frac{(u_i - \bar{u}_i)^2}{\bar{u}_i - u_0} \right)
\]

In Eq. (13), \( u \) represents the coordinates of the point after the distortion effect. The translation vector and rotation matrix are calculated by introducing the Kronecker product for matrix operations, as shown in Eq. (14).

\[
\begin{align*}
A \otimes (B + C) &= A \otimes B + A \otimes C \\
(A \otimes E_1)(E_2 \otimes B) &= A \otimes B \\
\text{vec}(ABC) &= (C^T \otimes A)\text{vec}(B) \\
P_i^w &\leftarrow P^w_i - P^e_i
\end{align*}
\]
In Eq. (14), $\overline{P}_w$ is the average of all spatial reference points. Decompose the matrix $m$ as shown in Eq. (15).

$$m^{(k)} = \left[\sum_{i=1}^{n} w_i \left( P^o \odot (P^o)^T \odot V_i \right) + \sum_{i=1}^{n} (P^o \odot V_i) (G) \right] H^{(i)}$$

$$m = UDV^T$$  \hspace{1cm} (15)

In Eq. (15), $m^{(k)}$ and $vec(M^{(k)})$ are equal and are obtained by regularising the projection points. After calculating the translation vector and combining it with the rotation matrix, the objective function is obtained, as shown in Eq. (16).

$$E(R,t) = \sum_i w_i \left\| (T - V_i) \left( (P^o)^T \odot I + G \right) H \right\| = H^T J_{opt} H$$

$$J = \sum_i w_i \left( (P^o)^T \odot I + G^T \right) \left( I - V_i \right) \left( (P^o)^T \odot I + G \right)$$  \hspace{1cm} (16)

In Eq. (16), the matrix at $G$ is considered a constant matrix; $E(R,t)$ represents the final camera position. Once the technical framework has been constructed, the design of the appearance and interactive functionality of the interactive PSA interface can be carried out.

IV. PERFORMANCE TESTING OF AN INTERACTIVE INTERFACE DESIGN METHOD FOR PUBLIC SERVICE ANNOUNCEMENTS INCORPORATING VIRTUAL REALITY TECHNOLOGY

In order to verify the feasibility and effectiveness of the research design virtual reality technology when applied to the design of interactive interfaces for public service announcements, the performance of the system was tested. The basic hardware environment setup for the experiment is shown in Table I.

The testing started with performance testing of the functions in the first half of the process. The performance of the study using the greyscale algorithm was tested and compared with other greyscale algorithms and the results are shown in Fig. 7.

As can be seen in Fig. 7(a), the smallest result in the peak signal-to-noise ratio test was obtained by the component method at 12.980dB, while the largest result was obtained by the cvtColor used in the research method at 13.390dB. 39.053; the study method is not the largest, but the value of 39.021 is very close to the maximum. This shows that the research method produces the clearest images, with higher image contrast and higher computational efficiency. The performance of the denoising algorithm was tested and compared with other denoising algorithms and the results are shown in Fig. 8.

As seen in Fig. 8(a), the test was conducted using lightly shaded images, moderately shaded images, and heavily shaded images. In the signal-to-noise ratio test, the median filter reaches a value of 35.635dB on the light shaded image, much higher than the Gaussian filter and the mean filter at 28.595dB and 28.526dB; the median filter is the same as the mean filter on the medium shaded image, both at 25.673dB, slightly lower than the Gaussian filter at 25.841dB, but the difference is negligible; the median filter reaches a value of 38.145dB on the heavy shaded image, much higher than the Gaussian filter and the mean filter. As can be seen from Fig. 8(b), the mean square error of the median filter on the light, medium and heavy shaded images is lower than 14, and the lowest mean square error of the Gaussian filter and the mean filter is 24.666, which is much higher than that of the median filter. This indicates that the research method can provide good denoising of images in light, medium and heavy shading environments. After completing the testing of the first half of the process function, the performance of the research method was tested. The noise was fixed at 3pixel and 1000 experiments were performed for each reference point. The error of the research method when the number of reference points was varied was tested and the results are shown in Fig. 9.

<table>
<thead>
<tr>
<th>Parameter variables</th>
<th>Parameter selection</th>
</tr>
</thead>
<tbody>
<tr>
<td>Operating system</td>
<td>Windows 10</td>
</tr>
<tr>
<td>Operating environment</td>
<td>MATLAB 2020a</td>
</tr>
<tr>
<td>System PC side memory</td>
<td>16G</td>
</tr>
<tr>
<td>CPU dominant frequency</td>
<td>2.62Hz</td>
</tr>
<tr>
<td>GPU</td>
<td>RTX-2060</td>
</tr>
<tr>
<td>Central Processing Unit</td>
<td>Intel i5-10500</td>
</tr>
</tbody>
</table>

Table I. The experiment’s basic environmental parameters

![Fig. 7. Comparison of greyscale algorithm performance.](image-url)
As can be seen in Fig. 9, the errors of all methods gradually decrease as the number of reference points increases. As seen in Fig. 9(a) and Fig. 9(b), the curves for all methods show a sharp decrease in the first few times and a slow decrease towards a stable shape in the latter stages in terms of rotational mean error and rotational median error. The rotational mean error of the study method at 4 reference points is about 4.2 degrees, much lower than the other methods, and the rotational median error at four reference points is about 2.1 degrees, slightly lower than the LHM and much lower than the other remaining methods. At reaching 20 reference points, the rotational mean error of the study method stabilises around 0.7 degrees, slightly lower than the other methods; the rotational median error of the study method stabilises around 0.6 degrees, slightly lower than the other methods. As can be seen from Fig. 9(c) and Fig. 9(d), the curves of all the methods show a sharp decline in the first few times and a slow decline towards stability in the latter part of the curve for both the translational mean error and the translational median error, with the DLT declining at a slightly slower rate. The mean error of translation at 4 reference points is slightly lower than the LHM and much lower than the remaining methods, while the median error of evaluation is slightly lower than the LHM and much lower than the remaining methods. When reaching 20 reference points, the translation mean error of the study method and all other methods except DLT are stable at around 0.5%, and the translation median error is stable at around 0.4%. This indicates that the error of the research method is less affected by the number of reference points and has a much lower error. The effect of Gaussian noise on the positioning accuracy was tested and is shown in Fig. 10.

As can be seen in Fig. 10, DLT and HOMO consistently remain at a high error level when performing localisation, with the other methods resulting in increasing errors as the Gaussian noise increases. In terms of rotation error, the rotation error of the study method is about 0.3 degrees at 0.5 pixels, rising to about 1.8 degrees when reaching 5 pixels, which is significantly lower than the other methods. In terms of translation error, the translation error of the research method is about 0.1% at 0.5 pixels and rises to about 1.2% at 5 pixels, which is closer to LHM and significantly lower than other methods. This indicates that the de-noising performance of the research method is good, and the performance advantage of the research method becomes more obvious when the Gaussian noise is larger. Ten 3D reference points were set, and the camera coordinates for the 3D reference points were solved and reprojected, as shown in Fig. 11.
closer to the original image points, but no points almost coincide with the original points. All the points of the research method are close to the original points, and seven of them almost coincide with the original points. This indicates that the research method has good performance in solving the coordinates and produces less variation in position. The research method was tested with different numbers of reference points, as shown in Fig. 12, to ensure the computation time and number of solutions when using the research method for PSA interactive interface design.

The computation time of EPnP-GN and EPnP does not depend much on the number of reference points, while the computation time of other methods including the research method increases with the number of reference points. The computation time of EPnP-GN and EPnP is basically stable at 0.6s and 0.8s. DLT and HOMO are linear solution methods with obvious advantages in speed, but lacking in accuracy. The computation time of the research method is about 0.5s at five reference points, rising to about 1.6s when 40 reference points are reached, more than EPnP-GN and EPnP when the number of reference points is higher, and in all cases lower than LHM, which also uses orthogonal iteration. The research method generates 212 solutions at five reference points and rises to 1203 solutions at 40 reference points, which is more than the other algorithms. This indicates that the research method does not have a significant speed advantage when designing interactive interfaces for PSAs, but it can generate more solutions and provide a richer choice when conducting subsequent designs.

V. CONCLUSION

The interactive interface design of public service advertising directly affects the quality of public service advertising. In the environment of virtual reality technology, the visual positioning method is designed first, and then when the camera positioning method is designed, the image is preprocessed to reduce the computational burden, the Iterative method is introduced to reduce the interference in image positioning, and finally the object residual is iterated to complete the camera position pose estimation. The results show that the research method only requires 1.04s for image graying, while other methods require over 200ms; The Mean squared error of research methods on light, medium and heavy shadow images is lower than 14, far lower than other methods; In a noisy environment of three pixels, the translation mean error of the research method is around 0.5%, and the translation median error is around 0.4%, which is lower than other methods; In the Gaussian noise environment of five pixels, the rotation error of the research method is 1.8 degrees, which is significantly lower than other methods; When designing research methods, using five reference points can generate 212 schemes, which is more than other methods. It shows that the research method has good technical performance guarantee under the virtual reality technology environment, and can provide more options for the design of other content such as the Look and feel of subsequent public service advertisements. However, the research was tested in a simulated environment and lacked data reference to the real environment. In the future, real machine testing will be conducted to enrich experimental data and optimize it.

REFERENCES


Method for Image Quality Evaluation of Satellite-based SAR Data

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Abstract—A method for image quality evaluation of satellite-based Synthetic Aperture Radar: SAR data is proposed. Not only geometric fidelity but also signal to noise ratio, frequency component, saturated pixel ratio, speckle noise, optimum filter kernel size and its filter function are evaluated. Through experiments with SAR so called QPS-SAR\(_2\) (Q-shu Pioneers of Space SAR the second) of imagery data, all these items are evaluated, and it is confirmed that the geometric and radiometric performances are good enough. Also, geometric fidelity of QPS-SAR\(_2\) is compared to Sentinel-1/SAR European Space Agency (ESA) provided data which is obtained on the same day of QPS-SAR\(_2\) data acquisition.

Keywords—Image quality; synthetic aperture radar (SAR); geometric fidelity; signal to noise ratio; frequency component; saturated pixel ratio; speckle noise; optimum filter kernel size; filter function for speckle noise reduction

I. INTRODUCTION

Institute for Q-shu Pioneers of Space, Inc.: iQPS has realized a mass of 1/20 and a cost of 1/100 of the conventional X band SAR satellite by using a lightweight antenna for small satellites developed in-house, and a 100 kg class high-definition small SAR. It was the first successful SAR mounted small satellite in Japan. Currently, we are aiming to launch 36 small SAR satellites and build a constellation early after 2025, and to provide a quasi-real-time ground observation data service about every 10 minutes. To realize this project, we were able to work on the development, manufacture, and launch of two satellites, "Izanagi" and "Izanami" which play the role of technology demonstrator. The first unit "Izanagi" was launched in December 2019, and the second unit "Izanami" was launched after February 2021.

The satellite called "Izanami" which was developed and manufactured by iQPS and about 20 local companies in Kyushu was launched by the flagship rocket "Falcon 9" of the American space development company "SpaceX" on January 25, 2021 (Monday). It was launched at 1:14 a.m. and set to the altitude of about 525 km. Then, on the same day, the first communication was successful, the retractable antenna was deployed on the 30th of January (Saturday), and the first image acquisition was announced on 3 March (Wednesday), about one month after the launch. After that, Izanami continued to observe steadily every day, and succeeded in acquiring an image with an azimuth resolution of 70 cm and a range (ground range) resolution of 70 cm in a high-definition mode (spotlight mode)\(^1\) with a resolution of 70 cm.

The need for high-definition images in the market is extremely high, and the successful acquisition and demonstration of technology at Unit 2 Izanami (QPS-SAR\(_2\)) is a major step toward the full-scale data provision service business. In the future, we will further stabilize the image quality and proceed with the development of the usage platform with a sense of speed.

There are many papers which deal with SAR image quality evaluation methods. Not only geometric fidelity but also signal to noise ratio, frequency component, saturated pixel ratio, speckle noise, optimum filter kernel size and its filter function can be evaluated with the previously proposed methods. Cross comparison between two different SARs which are onboard the different satellites and observe in a same day is conducted. This is a new method for the SAR imagery data quality evaluation in this paper. Two different SARs observed the same ground cover target with the different off-nadir angles. Therefore, layover, shadowing, foreshortening is different each other of SAR imagery data results in some processing is required for tie point matching. Otherwise, pixel-to-pixel comparisons cannot be performed. The purpose of this paper is to propose SAR image quality evaluation methods such as spatial resolution, signal to noise ratio, modulation transfer function, pixel geolocation accuracy (geometric fidelity), etc. Through the experiments, it is found that the proposed method works well for image quality evaluation.

In the next section, related research works are described followed by research background and theoretical background. Then, the proposed system is described at first followed by some experiments are described together with conclusion and some discussions.

II. RELATED RESEARCH WORKS

The following are the SAR related papers,


\(^{1}\)There are normal mode (stripmap mode) with a resolution of 1.8m (azimuth resolution 1.8m x range resolution 0.7m) and high-definition mode (spotlight mode) with a resolution of 70cm.
Speckle noise removal of SAR images with DEM is attempted [3]. Meanwhile, a method of speckle noise reduction for SAR data is proposed [4]. Meanwhile, SAR image classification based on Maximum Likelihood decision rule: MLH with texture features taking into account a fitness to the probability density function is proposed [5]. A new method for SAR speckle noise reduction (Chi-Square Filter: CSF) is proposed and validated [6] together with a new method for SAR speckle noise reduction based on CST filter [7].

Decomposition of SAR polarization signatures by means of eigen-space representation is attempted [8]. On the other hand, evaluation of vector winds observed by NASA Scatterometer: NSCAT in the ocean of Japanese vicinity is conducted [9].

Polarimetric SAR image classification with maximum curvature of the trajectory in eigen space domain on the polarization signature is proposed [10] together with polarimetric SAR image classification with maximum curvature of the trajectory in eigen space domain on the polarization signature [11]. Meanwhile, polarimetric SAR image classification with high frequency component derived from wavelet Multi Resolution Analysis: MRA is proposed [12].

Comparative study of polarimetric SAR classification methods including proposed method with maximum curvature of trajectory of backscattering cross section in ellipticity and orientation angle space is conducted [13]. On the other hand, wavelet MRA and its application to polarimetric SAR classification are proposed [14].

Sentinel-1A SAR data analysis for disaster mitigation in Kyushu, Japan is conducted [15]. Also, flood damage area detection method by means of coherency derived from interferometric SAR analysis with Sentinel-1A SAR is proposed [16]. GCP generation from simulated SAR image derived from Digital Terrain Model: DTM and its application to texture feature extraction is conducted [17].

Meanwhile, there are the following not only SAR but also optical sensor image quality evaluation related papers,

Report on vicarious calibration and image quality evaluation of LISA/LISAT² is presented and issued [18]. Also, methods for vicarious calibration and image quality evaluation of LISA/LISAT are proposed [19].

On the other hand, there are the following papers which deal with frequency component analysis,

Polarimetric SAR image classification with high frequency component derived from wavelet multi resolution analysis: MRA is proposed [20]. Meanwhile, hearing aid method by equalizing frequency response of phoneme extracted from human voice is proposed [21].

Meanwhile, the following papers deal with noise analysis,

On the other hand, sensitivity analysis of Fourier Transformation Spectrometer: FTS against observation noise on retrievals of carbon dioxide and methane is conducted [22].

Noise suppressing edge enhancement based on Genetic Algorithm: GA taking into account complexity of target image measured with Fractal dimension is proposed [23]. Meanwhile, a method for aerosol parameter estimation error analysis is proposed with a consideration of noises included in the measured solar direct and diffuse irradiance [24].

Method of noise reduction in passive remote sensing is proposed for noise and clutter rejection [25]. On the other hand, speckle noise removal of SAR images with DEM is proposed [26]. Flood damage area detection method by means of coherency derived from interferometric SAR analysis with Sentinel-1A SAR is proposed [27]. Furthermore, Ground Control Point: GCP generation from simulated SAR image derived from Digital Terrain Model: DTM and its application to texture feature extraction is also proposed and validated [28]. Meanwhile, method for frequent high resolution of optical sensor image acquisition using satellite-based SAR image for disaster mitigation is proposed [29].

III. THEORETICAL BACKGROUND AND PROPOSED METHOD

Not only geometric fidelity but also Signal to Noise ratio: S/N, frequency component, saturated pixel ratio, speckle noise, optimum filter kernel size and its filter function can be evaluated with the previously proposed methods.

As for the geometric fidelity, GCPs which are derived from geographic maps can be used. S/N can be evaluated with the ratio of mean and standard deviation of certain homogeneous areas. Frequency component can be evaluated with Fast Fourier Transformation: FFT, Discrete Cosine Transformation: DCT and so on.

On the other hand, saturated pixel ratio can be evaluated with histogram analysis. As for the speckle noise evaluation, visual perception can be used for identification of saturated pixels. Meanwhile, optimum filter kernel size and its filter function for speckle noise reduction can be estimated with trial and errors through visual perception.

Cross comparison between two different SARs which have onboarded the different satellites and observed in a same day is conducted. This is a new method for the SAR imagery data quality evaluation in this paper. Two different SARs observed the same ground cover target with the different off-nadir angles. Therefore, layover, shadowing, and foreshortening are different from each other in SAR imagery data, resulting in the need for some processing to be done for tie point matching. Otherwise, pixel-to-pixel comparisons cannot be performed.

IV. EXPERIMENTS

A. Examples of Acquired QPS-SAR_2 Imagery Data used for Image Quality Evaluation

Fig. 1 shows examples of acquired QPS-SAR_2 imagery data used for image quality evaluation. Fig. 1(a) and 1(b) are SAR images of Osaka, Japan (34.69N, 135.50E) (these are referred to Osaka scenes hereafter) which are acquired at 9:24 p.m. (Japan time) on 4 May 2021 (Tuesday) while Figure 1(c) to (g) are SAR images of Tokyo, Japan (N35 40’, E139 46’) (these are referred to Tokyo scenes hereafter) which are acquired at 9:06 p.m. (Japan time) on 23 March 2021 (Friday).

²https://www.eoportal.org/satellite-missions/lajapan-a3#overview
In Fig. 1(a), "Senri Chuo Station" (Located in 1-chome Shinseni Higashimachi (34.81N, 135.49E), Toyonaka City, Osaka Prefecture, it is a station on the Kita-Osaka Kyuko Railway Namboku Line and the Osaka Monorail Main Line,) is a little to the right of the central part. In Fig. 1(b), the steel tower and a part of the electric wire in the upper left, the Ferris wheel "OSAKA WHEEL" and "Panasonic Stadium Suita" with a height of 123m in the lower left, and the monorail stopped in the garage near the center are identified. The detail locations of Toyonaka and Osaka wheel are shown in Fig. 1(c).

As for Fig. 1(f), since the roof of the Tokyo Dome (35.71N, 139.75E) can be seen through, you can even see the electric bulletin board. Meanwhile, in Fig. 1(g), you can see Ueno Onshi Park (35.72N,139.77E) and Shinobazu Pond (35.71N, 139.77E) while you can identify the freight container in the lower right in Fig. 1(h). The detail locations of Tokyo Marunouchi buildings (35.68N, 139.76E), Minami Senju (35.73N 139.80E), Tokyo dome and Ueno Onshi Park are shown in Fig. 1(i).

Through comparisons between geographic maps and QPS-SAR_2 imagery data, it is found that the spatial resolution (equivalent to ground range) of QPS-SAR_2 is confirmed as 70cm (Azimuth resolution 70cm while range resolution 70cm).
From the QPS-SAR_2 image, edge responses are reduced. Then, differentiations of the responses are calculated. After that, frequency component analysis is made for the differentiations result in spatial resolution.

**B. Geometric Fidelity**

This significantly high spatial resolution is one of specific features of QPS-SAR_2. Therefore, geometric performance has to be evaluated. By comparing QPS-SAR_2 images to geographic maps, geometric fidelity can be evaluated. Google map of aerial photos derived GCPs can be a good reference of the geometric performance evaluation, pixel size, pixel-to-pixel distances.

Fig. 2 shows Google map of aerial photo of Fukuoka, Japan (33°36'47.13"N, 130°24'36.08") and QPS-SAR_2 image which is acquired at 21:23 on 23 February 2021. Since Google map was created using advanced surveying techniques including aerial surveying, it was considered to have high geometric fidelity, and the instrumental fidelity of QPS-SAR_2 was measured with reference to this.

The following two GCPs are selected for evaluation of pixel-to-pixel distance performance,

- GCP1: 33°36'33.03"N, 130°23'24.88"E, Alt=4m
- GCP2: 33°36'13.36"N, 130°22'50.28"E, Alt=1m

Yellow colored points (two end points of pixels are colored in yellow) are indicated the locations of GCPs in Fig. 3. Fig. 3 shows superimposed image between QPS-SAR_2 image and Google map. As the result, it is found that the distance between two GCPs is 5.41 km ± 2m. Therefore, it is concluded that geometric fidelity of QPS-SAR_2 is around 2m.

Same experiment is conducted with the following GCPs for Tokyo scene (Tokyo dome area) of QPS-SAR_2 image,

- GCP3 : 35° 42’ 08.40” N, 139° 45’ 18.27” E 6m
- GCP4 : 35° 04’ 29.71” N, 139° 44’ 12.59” E 2.6m

As shown in Fig. 4, white colored line shows the distance between two GCPs and indicates 2.11 km ± 2m. Therefore, geometric fidelity of QPS-SAR_2 is around 2m.
On the other hand, an attempt is made for estimation of off-nadir angle and SAR radio wave irradiation direction with Google Earth which can change azimuth and elevation of look angle. Fig. 5 shows the resultant image of superposed Tokyo Dome image of QPS-SAR_2 image on Google Earth derived aerial photo. Because the Tokyo Dome area is almost flat so that there is no shadowing, layover, and foreshortening. Therefore, the two images are almost perfectly matched. Thus, it is found that estimation of off-nadir angle and radio wave irradiation direction can be done with Google Earth. Geometric fidelity is defined as pixel geolocation knowledge which can be determined by the geometric parameters of off-nadir angle of SAR antenna, satellite nadir location (satellite position determination accuracy), Earth curvature at the footprint of SAR antenna pattern. Using image rotation (azimuth and elevation angles) and translation functions of Google Earth, QPS-SAR_2 image and Google Earth image can be matched. This implies that the rotation angle and translation pixels are almost same as the predicted value from the pixel geolocation knowledge, which is determined by satellite position knowledge, off-nadir angle, compliant Earth spheroidal model of WGS84.

C. Saturated Pixel Ratio

As shown in Fig. 6, the most visual difference between both is saturated pixels.

\[ \text{VH-Pol} = 1.478 \times \text{HH-Pol} + 15.24 \]  
\[ \text{VV-Pol} = 2.49 \times \text{HH-Pol} + 21.32 \]

The World Geodetic System 1984 (WGS 84) is a 3-dimensional coordinate reference frame for establishing latitude, longitude and heights for navigation,
Saturated pixel ratios of the two different SAR imagery data are evaluated. As shown in Fig. 7, the saturated pixel ratio of Sentinel-1/SAR is 233.07/255=91.4 (%) while that of QPS-SAR_2 is 81.78/255=32.07 (%). Therefore, this QPS-SAR_2 image is much informative than that of Sentinel-1/SAR image in terms of non-saturated pixels. Since Sentinel-1/SAR has VV and VH polarizations, the saturation rate at VV becomes high. If the VV saturation rate is lowered and the dynamic range setting is matched to VV, the dynamic range of VH will be narrowed.

D. Frequency Component Analysis

Fig. 8 shows frequency components of QPS-SAR_2 and Sentinel-1/SAR imagery data of Tokyo Dome scene. As shown in Fig. 8, it is clear that the QPS-SAR_2 image shows much wider frequency components in comparison to that of the Sentinel-1/SAR image. The top left corner of the Fig. 8 shows zero frequency component, and the horizontal/vertical axis shows horizontal and vertical frequency components, respectively.

From the results of the aforementioned two experiments, following is found:

1) Sentinel-1/SAR concentrates on frequency components that are about 10 times lower than QPS-SAR_2.
2) It is reasonable because the spatial resolution of QPS-SAR_2 is 10 times smaller than that of Sentinel-1/SAR.
3) Saturation rate (average pixel value / 255): Sentinel-1/SAR VV-Pol is about 38.69 times higher than QPS-SAR_2 HH-Pol (Sentinel-1/SAR VV-Pol has many saturated pixels).

Fig. 8. Frequency component comparison between QPS-SAR_2 and Sentinel-1/SAR.

E. Signal to Noise Ratio

Signal to noise ratio (S/N) depends on the intensity level, backscattering cross section.

Therefore, S/N is evaluated at low, middle and high levels of Fukuoka scene of QPS-SAR_2 image. As shown in Fig. 9, it is found that the low level of S/N=44.18/3.25=13.59, the middle level of S/N=55.52/6.72=8.26, and the high level of S/N=148.33/38.91=3.18. Noise component is mainly caused by speckle noise of which the footprint composed with multiple targets (Due to radio wave interference from multiple targets). In comparison among the low, the middle and the high levels of QPS-SAR_2 image, the number of pixels which composed with multiple targets for the low levels of image is small and that for the middle level of image is middle in between small and large as well as that for the high level is large.
F. Speckle Noise Reduction

When observing a distributed target, speckle noise is generated by the fading phenomenon caused by the random presence of a large number of scatterers in the resolution cell, so this detection and reduction is necessary. The important thing of the speckle noise reduction is filter kernel size and filter function. In this paper, comparisons are made between 3 by 3 and 5 by 5 of kernel sizes for median filter, as well as between circle and Gauss filter functions. The circle and the Gauss filters are well known and widely used typical kernel spatial filter functions without edge preservation. On the other hand, the median filter with the kernel size of 3 and 5 are edge preserving filter. Frequency component is degraded depending on the kernel size and the kernel functions. Fig. 10 shows the resultant images of the comparisons. As the result, it is found that the best filter kernel size is 3 by 3 and the filter function of Gauss is much better than the circle filter function through visual comparison. From the point of view of spatial frequency component preservation, the median filter with 3 by 3 filter kernel shows the best performance, obviously.

V. DISCUSSION

Although SAR image quality evaluation methods such as spatial resolution, signal to noise ratio, modulation transfer function, pixel geolocation accuracy (geometric fidelity), etc. are confirmed through the experiments, speckle noise reduction is not good enough. Multiple or single target in the footprint is the issue of the speckle noise reduction. Therefore, further thoughts considering the causes of the speckle noise are required for improvement of noise reduction performance.

VI. CONCLUSION

Method for image quality evaluation of satellite-based SAR data is proposed. Not only geometric fidelity but also S/N, Frequency component, saturated pixel ratio, speckle noise, optimum filter kernel size and its filter function are evaluated. Through experiments with Q-shu Pioneers of Space: QPS-SAR_2 imagery data, all these items are evaluated, and it is confirmed that the geometric and radiometric performances are good enough. Also, geometric fidelity of QPS-SAR_2 is compared to Sentinel-1/SAR of ESA provided data which is obtained on the same day of QPS-SAR_2 data acquisition.

In more detail, the S/N of QPS-SAR_2 was found to be 3.18 to 13.59 for the Fukuoka image and 1.96 to 5.10 for the Tokyo image. Also, it was found that the wind size suitable for speckle noise removal is 3x3, and Gauss is suitable for the filter function. Furthermore, Sentinel-1/SAR concentrates on frequency components that are about 10 times lower than QPS-SAR_2. Moreover, spatial resolution is about 10 times lower for Sentinel-1/SAR than for QPS-SAR_2. Meanwhile, saturation rate (average pixel value / 255): Sentinel-1/SAR is about 38.69 times higher than QPS-SAR_2 so that Sentinel-1/SAR may have more saturated pixels than QPS-SAR_2.

FUTURE RESEARCH WORKS

In the future, we will evaluate layover, shadowing, fore shortening, and signal to ambiguity ratio, main lobe / first side lobe ratio, spatial resolution, radiometric resolution, etc. using QPS-SAR_2 images which observe corner reflectors and with ground test data.

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REFERENCES


AUTHOR’S PROFILE

Kohei Arai, He received BS, MS and PhD degrees in 1972, 1974 and 1982, respectively. He was with The Institute for Industrial Science and Technology of the University of Tokyo from April 1974 to December 1978 also with National Space Development Agency of Japan from January 1979 to March 1990. During from 1985 to 1987, he was with Canada Centre for Remote Sensing as a Post-Doctoral Fellow of National Science and Engineering Research Council of Canada. He moved to Saga University as a Professor in Department of Information Science in April 1990. He is now an Emeritus Professor of University of Arizona, USA since 1998 and is an Adjunct Professor of Nishi-Kyushu University as well as Kurume Institute of Technology/AI Application Laboratory since 2021. He was Vice Chairman of the Science Commission “A” of ICSU/COSPAR since 2008. He is now award committee member of ICSU/COSPAR. He wrote 78 books and published 695 journal papers as well as 560 conference papers. He received 77 of awards including ICSU/COSPAR Vikram Sarabhai Medal in 2016, and Science award of Ministry of Mister of Education of Japan in 2015. He is now Editor-in-Chief of IJACSA and IJISA. http://heagis.ip.siesa-u.ac.jp/index.html

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A Multi-label Filter Feature Selection Method Based on Approximate Pareto Dominance

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Abstract—The Pareto dominance has been applied to resolve the issue of choosing significant features from a multi-label dataset. High-dimensional labels will directly result in the difficulty of forming Pareto dominance. This work proposes a multi-label feature selection approach based on the approximate Pareto dominance (MAPD) to address this issue. It maps the multi-label feature selection to the problem of solving the approximate Pareto dominant solution set. By introducing an approximate parameter, it is possible to efficiently cut down on the amount of features in the chosen feature subset while also raising its quality. To verify the performance of MAPD, this research compares the MAPD algorithm with alternative approaches in terms of Hamming loss, accuracy, and chosen feature size using nine publicly available multi-label datasets. The findings indicate that the MAPD method performs better in terms of classification accuracy, Hamming loss, and the amount of features that may be chosen.

Keywords—Approximate Pareto dominance; multi-label data; feature selection

I. INTRODUCTION

Feature selection is a process of removing noisy information and selecting the most significant feature subset, which is commonly considered as a pre-process of building a classifier machine learning model [1]-[3]. The multi-label feature selection problem is more universal in application than the single-label feature selection problem [4], [5]. For example, it might be necessary to simultaneously judge the geographical location, weather conditions, and image content of a figure in the process of image recognition [6], [7]. When processing the text categorization, we may need to judge whether the text belongs to multiple bibliographic categories [8], [9]. It also might be necessary to judge whether a protein has multiple different functions in the field of bioinformatics in the same manner [10].

The multi-label feature selection methods could be classified as the filter methods [11]-[13], the wrapper methods [14], [15], and the embedded methods [16], [17]. Since the increase of label dimension would lead to higher time complexity of the feature selection processing, this paper only focuses on the filter feature selection methods that are more efficient compared with the wrapper and embedded methods.

In the current literature, the multi-label feature selection problem can be resolved primarily in two ways. The first strategy is to convert the multi-label data into single-label data and then choose feature subsets using single-label feature selection techniques [18], [19]. However, such methods create an abundance of labels with only a limited number of observations which is not beneficial for establishing a classifier model. In order to improve the disadvantages of such methods, a method called pruned problem transformation (PPT) is proposed, and it ignores the labels with observations lower than the given threshold [20]. This method can ensure that each converted single-label has enough observations to establish a classification model, but this irreversible conversion may lose some label information [21]. The second method involves choosing a feature subset using a specific multi-label feature selection algorithm [22]-[25]. For example, an approximating mutual information (AMI) method is proposed, and it uses the feature selection criterion as maximizing the mutual information between features and labels and minimizing the mutual information among features [26]. A multi-label feature selection strategy based on a scalable criterion for a large label collection (SCLS) is proposed, which can evaluate the conditional correlation between variables more accurately through an extensible correlation evaluation process [27]. Due to the information loss issue of the first way, it is believed that the second way has better performance under several evaluation criteria, such as classification accuracy and Hamming loss [28].

Recently, scholars have applied the concept of Pareto dominance to multi-label feature selection problems. For resolving the multi-label feature selection problem, the Pareto dominance concept is appropriate, since it can transform this problem into a more manageable issue. Specifically, a multi-label feature selection technique based on the Pareto dominance concept (ParFS) is proposed [29]. The ParFS algorithm treats each label as a dimension of Pareto dominance, and thus the issue of multi-label feature selection becomes a Pareto dominance problem. The key point of this algorithm is that the original feature set is viewed as a solution set. Each feature in the original feature set is viewed as a solution, and the evaluation function of the solution is regarded as a feature's and a label's correlation vector. Then the feature selection problem is transformed into how to delete those non-Pareto optimal solutions. In fact, high-dimensional data refers not only to the high-dimensional features of the data, but also to its high-dimensional labels. High-dimensional labels increase the difficulty of using the ParFS algorithm to resolve issues of multi-label feature selection [30], [31]. Specifically, the increase of label dimensions will directly lead to the increase of the dimensions to be considered in Pareto dominance, which results in the difficulty of forming Pareto dominance, causing the Pareto-dominance-based algorithm to fail to finish the multi-label feature selection task. Consequently, it's essential to improve the multi-label feature selection method based on Pareto dominance concept.
In this paper, a multi-label feature selection strategy based on approximate Pareto dominance is proposed (MAPD). Approximate Pareto dominance requires that one solution is superior to the others in most dimensions, but not in all dimensions. Compared with the existing concept of Pareto dominance, the concept of approximate Pareto dominance introduces a new approximate parameter. This parameter can reduce the difficulty of forming approximate Pareto dominance between two solutions under the evaluation function of high-dimensional solutions, and ensure that the scale of approximate Pareto dominance solutions is within an acceptable range. The three main contributions are as follows:

1) A new concept called approximate Pareto dominance is proposed. By introducing an approximate parameter, it can solve the problem when Pareto dominance is difficult to form in the case that the evaluation function dimension of the solution is high.

2) Approximate Pareto dominance is applied in the multi-label feature selection, and the multi-label feature selection issue is mapped to the challenge of determining the approximate Pareto dominance solution set.

3) Based on the approximate Pareto dominance, MAPD is built for the high-dimensional multi-label feature selection problem, and it is proved to be competitive compared with the existing methods.

The remainder of the paper is organized as follows: Preliminaries of this work are presented in Section II. The proposed approximate Pareto dominance concept and multi-label feature selection method is discussed in Section III. Experimental studies and discussion are presented in Section IV, and the paper is concluded in Section V.

II. PRELIMINARIES

A. Problem Description

Given a dataset, \(X = [X_1, ..., X_m] \) denotes the feature observation space, and its corresponding \(d\)-dimension label space is \(Y = [Y_1, ..., Y_l] \), where \(X_j = [x_{ij}, ..., x_{j}, ..., x_{nj}]^T \) is the \(j\)th feature in \(X \), \(x_{ij} \) denotes the \(i\)th observation of the \(j\)th feature, \(Y_i = [y_{i1}, ..., y_{it}, ..., y_{il}]^T \) is the \(t\)th label in \(Y \), \(y_{it} \) is the \(i\)th observation of the \(t\)th label, \(y_{it} \in \{0\} \) or \(\{1\} \). \(i = 1, 2, ..., n \), \(j = 1, 2, ..., m \), and \(t = 1, 2, ..., l \). To create the classification machine learning model, the multi-label feature selection in this study aims to identify the best feature subset from all feasible subsets.

B. Symmetrical Uncertainty

Symmetrical Uncertainty which abbreviated to SU is a measure of the degree to which two variables are related [32]. In essence, SU measures the information that the two variables exchange and is a standardized representation of the mutual information. In other words, SU quantifies how much one variable’s uncertainty is reduced when the other variable is known, and the higher the degree of SU, the more knowledge the two variables have in common. The formula for calculating SU is given below [33].

\[
SU(U, V) = 2 \frac{H(U) - H(U|V)}{H(U) + H(V)},
\]

with \(H(U) = -\sum_{i=1}^{n} p(u_i) \log_2(p(u_i)) \)

\[
H(U|V) = -\sum_{i=1}^{n} \sum_{j=1}^{n} p(v_i) \log_2(p(u_i|v_j))\]

In the above equation, \(U \) and \(V \) denote two variables with \(n \) observations. \(H(U) \) and \(H(V) \) are respectively the entropy of \(U \) and the entropy of \(V \). \(H(U|V) \) is the conditional entropy of \(U \) under \(V \).

C. Pareto Dominance

The following definition of Pareto dominance is used to compare the results of two solutions to a particular problem.

Definition 1 (Pareto Dominance [34]): If \(s_1(s_{11}, s_{12}, ..., s_{1m}) \) and \(s_2(s_{21}, s_{22}, ..., s_{2m}) \) are two solutions of a given problem, and \(g(s_i) = (g_1(s_i), g_2(s_i), ..., g_m(s_i)), i \in [1,2] \) is the evaluation function of \(m \) dimensions of the given problem,

1) we define that the solution \(s_2(s_{21}, s_{22}, ..., s_{2m}) \) is Pareto dominated to the solution \(s_1(s_{11}, s_{12}, ..., s_{1m}) \) if and only if \(g_j(s_2) > g_j(s_1), j \in [1,2,...,m] \) is satisfied;

2) we define that the solution \(s_2(s_{21}, s_{22}, ..., s_{2m}) \) is weakly Pareto dominated to the solution \(s_1(s_{11}, s_{12}, ..., s_{1m}) \) if and only if \(g_j(s_2) \geq g_j(s_1), f \in [1,2,...,m] \) is satisfied;

3) we define that the solution \(s_1(s_{11}, s_{12}, ..., s_{1m}) \) and solution \(s_2(s_{21}, s_{22}, ..., s_{2m}) \) have no differences under the Pareto dominance if and only if \(g_j(s_1) > g_j(s_2) \) and \(g_k(s_1) < g_k(s_2) \) if \(j \neq k \in [1,2,...,m] \) are satisfied.

It can be inferred that the conditions required for Pareto dominance are relatively strict, which requires that one solution is superior to another solution in each dimension of the evaluation function. Considering the increased evaluation function dimensions of the solution, we assume that all the evaluation function dimensions of the solutions are independent with each other, and the difficulty of forming Pareto dominance will increase exponentially. Therefore, for the high-dimensional evaluation function of the solutions, one solution is hard to be Pareto dominated to another solution.

We concentrate on the Pareto dominance relationships between one solution and other solutions when there are more than two possible solutions to the problem. The set of Pareto optimal solutions is defined as follows.

Definition 2 (Pareto Optimal Solutions Set [34]): If \(S = \{s_1, s_2, ..., s_n\} \) is a solutions set of a specific problem and \(s_j \in S \) is not Pareto dominated to any other solutions in \(S \), then we propose that \(s_j \in S \) is a Pareto optimal solution. All the Pareto optimal solutions in \(S \) is called as the set of Pareto optimal solutions.

According to Definition 2, we suggest that the set of Pareto-optimal solutions can partially substitute for the set of
original solutions as the remaining solutions are all inferior to a certain solution in the Pareto optimal solutions set.

III. APPROXIMATE PARETO DOMINANCE AND MULTI-LABEL FEATURE SELECTION

A. Multi-label Feature Selection Based on Pareto Dominance

Given a multi-label dataset with \( m \) features, \( l \) labels and \( n \) observations, the original features are denoted as a set of solutions of the feature selection problem, and each feature in the original feature set is denoted as a solution of the feature selection problem. The \( l \)-dimensional evaluation function of the solution is defined as the symmetric uncertainty between the feature and its \( l \) labels. In this way, choosing a feature subset from the original feature set is analogous to choosing the Pareto optimal solutions set from the initial solution set.

As mentioned in Definition 1, Pareto dominance strictly requires that one solution is superior to another solution in each dimension of the evaluation function of the solution. In particular, with the increased dimensions of the evaluation function of a solution, it is challenging to come up with a solution that is superior to another solution in all the dimensions of the evaluation function. As such, most solutions in the solution set become Pareto optimal solutions. In this case, most features in the original feature set are preserved in the feature selection process. Pareto dominance might cause the inefficiencies of the feature selection process because useless features cannot be removed. Therefore, to cope with the multi-label feature selection issue, this study proposes the approximate Pareto dominance based on the Pareto dominance to avoid the above circumstance.

B. Approximate Pareto Dominance

Based on Pareto Dominance, this study proposes approximate Pareto dominance and the set of approximate Pareto optimal solutions.

Definition 3 (Approximate Pareto Dominance): If \( s_1(s_{11}, s_{12}, ..., s_{1n}) \) and \( s_2(s_{21}, s_{22}, ..., s_{2n}) \) are two solutions of a problem, and \( g(s_j) = (g_1(s_j), g_2(s_j), ..., g_m(s_j)), j \in \{1, 2\} \) is the evaluation function of \( m \) dimensions of the given problem.

1) we define solution \( s_1(s_{11}, s_{12}, ..., s_{1n}) \) is approximate Pareto dominated to solution \( s_2(s_{21}, s_{22}, ..., s_{2n}) \) if and only if \( \sum_{j=1}^{m} \delta(g_j(s_1)) > g_j(s_2)) > am \) is satisfied;

2) we define solution \( s_1(s_{11}, s_{12}, ..., s_{1n}) \) is weakly approximate Pareto dominated to solution \( s_2(s_{21}, s_{22}, ..., s_{2n}) \) if and only if \( \sum_{j=1}^{m} \delta(g_j(s_1)) > g_j(s_2) \geq am \) is satisfied;

3) we define solution \( s_1(s_{11}, s_{12}, ..., s_{1n}) \) and solution \( s_2(s_{21}, s_{22}, ..., s_{2n}) \) have no differences under approximate Pareto dominance if and only if \( \sum_{j=1}^{m} \delta(g_j(s_1)) > g_j(s_2)) < am \) and \( \sum_{j=1}^{m} \delta(g_j(s_1)) > g_j(s_2)) < am \) are satisfied.

Note that \( \delta(x) \) is a conditional discriminant function and \( 0.5 < \alpha < 1 \) is the approximate parameter. When condition \( x \) is satisfied, \( \delta(x) = 1 \); otherwise \( \delta(x) = 0 \).

Definition 4 (Approximate Pareto Optimal Solutions Set): If \( S = \{s_1, s_2, ..., s_n\} \) is a set of solution of a given problem and \( s_i \in S \) is not approximate Pareto dominated to any other solutions in \( S \), then we define that \( s_i \in S \) is an approximate Pareto optimal solution of the given problem. All the approximate Pareto optimal solutions in \( S \) are called the approximate Pareto optimal solutions set.

Compared with Pareto dominance, the approximate Pareto dominance introduces an approximate parameter. The higher the value of the approximate parameter is, the more dimensions of one solution are required to be superior to another solution in all the evaluation function dimensions, and the closer the approximate Pareto dominance solution is to the Pareto dominance solution. The upper bound of the approximate parameter is 1. When the approximate parameter approaches to the upper bound, the approximate Pareto dominance solution is equal to Pareto dominance. The lower bound of the approximate parameter is 0.5, which means that when one solution is approximate Pareto dominated to another solution, it is dominant in more than half of the dimensions. This approximate parameter can avoid the situation that two solutions are mutually dominant.

C. Multi-Label Feature Selection Algorithm

We build a multi-label feature selection algorithm (MAPD) based on approximate Pareto dominance for feature selection with high-dimensional labels, which is shown in the following Algorithm 1, where the \( SU_{ij} \) is the symmetrical uncertainty of the \( i \)th feature and the \( j \)th feature; the set \( S \) is the selected feature subset; \( \delta(c) \) is the conditional discriminant function; \( s_j \) is the \( j \)th solution in the original solution set (namely the \( j \)th feature); \( g_j(s_j) \) is the \( p \)th dimension of the evaluation function of the \( j \)th solution; \( \alpha \) is the approximate parameter. Approximate parameter \( \alpha \) can keep the number of approximate Pareto dominance solutions in an acceptable range.

Algorithm 1 contains three important steps: (1) calculating the symmetrical uncertainty between one feature and one label (see lines 1-5 of Algorithm 1); (2) initializing the selected feature set as an empty set, treating each feature as a solution and the symmetric uncertainty between the feature and each label as one dimension in the evaluation function. We detect the approximate Pareto dominant relationship between each two features. If one feature is the approximate Pareto dominant solution of the original solution set, the feature is merged into the selected feature set until all features are checked (see lines 6-17 of Algorithm 1). (3) Output the selected feature set (see line 18 of Algorithm 1).
**Algorithm 1 MAPD**

Input: The approximate parameter $\alpha$, the dataset $X$ with $m$ features, $l$ labels and $n$ observations;
Output: The finally chosen feature subset $S$.

1. for $i = 1:l$
2. for $j = 1:m$
3. $SU_{ij} = SU(X_i, X_j)$
4. end
5. end
6. $S = \emptyset$
7. for $i = 1:m$
8. $k = 0$
9. for $j = 1:m$
10. if $j \neq i \cap \sum_{p=1}^{l} \delta(g_p(s_j) > g_p(s_i)) > \alpha l$
11. $k = k + 1$
12. end
13. end
14. if $k = 0$
15. $S = S \cup i$
16. end
17. end
18. Output $S$

### IV. EXPERIMENTAL STUDIES AND DISCUSSION

#### A. Parameter Setting

To evaluate the effectiveness of the suggested MAPD technique, we employ nine publicly available multi-label datasets (http://mulan.sourceforge.net/datasets-mld.html). The datasets have been used in many studies, e.g., [35]-[37].

Given a dataset with $m$ features, $l$ labels and $n$ observations, $y_i = (y_{i1}, y_{i2}, ..., y_{il})$ is the real label of the $i$th observation, $y'_i = (y'_{i1}, y'_{i2}, ..., y'_{il})$ is the predicted label of the $i$th observation based on the classification model. To assess how well multi-label feature selection approaches work in terms of precision, we use two criteria of Hamming loss and accuracy, which can be calculated as follows [38], [39]:

\[
\text{Hamming loss} = \frac{1}{n} \sum_{i=1}^{n} \frac{\sum_{j=1}^{m} \delta(y_{ij} \neq y'_{ij})}{l} \tag{4}
\]

\[
\text{Accuracy} = \frac{1}{n} \sum_{i=1}^{n} \frac{\sum_{j=1}^{m} \delta(y_{ij} = y'_{ij}) = m}{l} \tag{5}
\]

According to Equation (4), Hamming loss analyzes each dimension of an observation's real label and prediction label and focuses on the prediction accuracy of a certain dimension. Then Hamming loss calculates the average error prediction rate of all observations in all dimensions. According to Equation (5), accuracy strictly requires that the real label and the prediction label should be exactly the same, and calculates the accuracy prediction based on all observations.

This study uses the average values of the accuracy and the Hamming loss of 5-fold cross validation of five times as a measure of how well feature selection techniques operate. Specifically, 5-fold cross validation means that the observations are randomly divided into five subsets. For a total of five tests, one of which is chosen as the testing set and the other four as the training set, and the average performance of these five times' tests is taken as the performance of 5-fold cross validation. A higher value of accuracy (or a lower value of Hamming loss) means that the feature selection method is more efficient.

#### B. Result Analysis

A summary of the nine public multi-label datasets is presented in the following Table I. As shown in Table I, the nine multi-label datasets come from many different fields, such as image, text, and biology, etc. These datasets' label counts range from 6 to 374. There are from 593 to 7395 observations and from 72 to 1836 features are present. We use these multi-label datasets to test the performance of the MAPD method.

<table>
<thead>
<tr>
<th>Name</th>
<th>Abbr.</th>
<th>$n$</th>
<th>$m$</th>
<th>$l$</th>
<th>LCard</th>
<th>LDen</th>
<th>Field</th>
</tr>
</thead>
<tbody>
<tr>
<td>scene</td>
<td>Sce</td>
<td>2407</td>
<td>294</td>
<td>6</td>
<td>1.074</td>
<td>0.179</td>
<td>image</td>
</tr>
<tr>
<td>emotions</td>
<td>Emo</td>
<td>593</td>
<td>72</td>
<td>6</td>
<td>1.869</td>
<td>0.311</td>
<td>music</td>
</tr>
<tr>
<td>yeast</td>
<td>Yea</td>
<td>2417</td>
<td>103</td>
<td>14</td>
<td>4.237</td>
<td>0.303</td>
<td>biology</td>
</tr>
<tr>
<td>birds</td>
<td>Bir</td>
<td>645</td>
<td>260</td>
<td>19</td>
<td>1.014</td>
<td>0.053</td>
<td>audio</td>
</tr>
<tr>
<td>genbase</td>
<td>Gen</td>
<td>662</td>
<td>1186</td>
<td>27</td>
<td>1.252</td>
<td>0.046</td>
<td>biology</td>
</tr>
<tr>
<td>medical</td>
<td>Med</td>
<td>978</td>
<td>1449</td>
<td>45</td>
<td>1.245</td>
<td>0.028</td>
<td>text</td>
</tr>
<tr>
<td>enron</td>
<td>Enr</td>
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<td>1001</td>
<td>53</td>
<td>3.378</td>
<td>0.064</td>
<td>text</td>
</tr>
<tr>
<td>bitex</td>
<td>Bib</td>
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<td>1836</td>
<td>159</td>
<td>2.402</td>
<td>0.015</td>
<td>text</td>
</tr>
<tr>
<td>corel5k</td>
<td>Cor</td>
<td>5000</td>
<td>499</td>
<td>374</td>
<td>3.522</td>
<td>0.009</td>
<td>image</td>
</tr>
</tbody>
</table>

Table I: Characteristics of the Nine Multi-label Datasets

Note: LCard means label cardinality; LDen means label density.

Label density is a standardized way to calculate label cardinality by dividing the total number of labels, where label cardinality is the average number of labels marked for each observation. The following formulas can be used to calculate label cardinality and label density: (Kashef and Nezamabadi-Pour, 2019):

\[
LCard = \frac{1}{n} \sum_{i=1}^{n} \sum_{j=1}^{m} \delta(y_{ij} = 1) \tag{6}
\]

\[
LDen = \frac{1}{n} \sum_{i=1}^{n} \sum_{j=1}^{m} \delta(y_{ij} = 1) \tag{7}
\]

In the MAPD algorithm, we first analyze approximate Pareto dominance of different approximate parameter $\alpha$. Considering the range of the approximate parameter is $0.5<\alpha<1$, we set four groups of tests, i.e., $\alpha=0.6$, $\alpha=0.7$, $\alpha=0.8$, and $\alpha=0.9$. The results of Hamming loss for different approximate parameters are listed in Table II.

As shown in Table II, when the value of the approximate parameter decreases, the average of Hamming loss for the nine datasets decreases. Specifically, successively subtracting two adjacent items in the last row of Table II, we can see that the reduction of Hamming loss is 0.36%, 0.22%, and 0.07% for each 0.1 reduction of the approximate parameter $\alpha$. 

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respectively. Therefore, the proposed MAPD algorithm has a trend of continuous reduction and convergence in the Hamming loss criterion. Specifically, the average of the Hamming loss is optional when the approximate parameter $\alpha=0.6$. The optimal Hamming loss is obtained on 7, 5, and 3 of 9 datasets when the approximate parameters are $\alpha=0.6$, $\alpha=0.7$, and $\alpha=0.8$, respectively. The standard deviations of Hamming loss are stable for all approximate parameters.

<table>
<thead>
<tr>
<th>Datasets</th>
<th>$\alpha=0.6$</th>
<th>$\alpha=0.7$</th>
<th>$\alpha=0.8$</th>
<th>$\alpha=0.9$</th>
</tr>
</thead>
<tbody>
<tr>
<td>Sce</td>
<td>0.0004</td>
<td>0.0004</td>
<td>0.0004</td>
<td>0.0035</td>
</tr>
<tr>
<td></td>
<td>(0.00004)</td>
<td>(0.00005)</td>
<td>(0.00005)</td>
<td>(0.00026)</td>
</tr>
<tr>
<td>Emo</td>
<td>0.0057</td>
<td>0.0069</td>
<td>0.0069</td>
<td>0.0236</td>
</tr>
<tr>
<td></td>
<td>(0.00073)</td>
<td>(0.00068)</td>
<td>(0.00068)</td>
<td>(0.00037)</td>
</tr>
<tr>
<td>Yea</td>
<td>0.0068</td>
<td>0.0067</td>
<td>0.0067</td>
<td>0.0068</td>
</tr>
<tr>
<td></td>
<td>(0.00026)</td>
<td>(0.00039)</td>
<td>(0.00029)</td>
<td>(0.0016)</td>
</tr>
<tr>
<td>Bir</td>
<td>0.0534</td>
<td>0.0513</td>
<td>0.0540</td>
<td>0.0543</td>
</tr>
<tr>
<td></td>
<td>(0.00000)</td>
<td>(0.00028)</td>
<td>(0.00009)</td>
<td>(0.00051)</td>
</tr>
<tr>
<td>Gen</td>
<td>0.0028</td>
<td>0.0028</td>
<td>0.0028</td>
<td>0.0030</td>
</tr>
<tr>
<td></td>
<td>(0.00023)</td>
<td>(0.00023)</td>
<td>(0.00025)</td>
<td>(0.0013)</td>
</tr>
<tr>
<td>Med</td>
<td>0.0032</td>
<td>0.0036</td>
<td>0.0063</td>
<td>0.0084</td>
</tr>
<tr>
<td></td>
<td>(0.00013)</td>
<td>(0.00010)</td>
<td>(0.00013)</td>
<td>(0.00021)</td>
</tr>
<tr>
<td>Enr</td>
<td>0.0128</td>
<td>0.0196</td>
<td>0.0325</td>
<td>0.0379</td>
</tr>
<tr>
<td></td>
<td>(0.00008)</td>
<td>(0.00015)</td>
<td>(0.00018)</td>
<td>(0.00011)</td>
</tr>
<tr>
<td>Bib</td>
<td>0.0043</td>
<td>0.0043</td>
<td>0.0051</td>
<td>0.0085</td>
</tr>
<tr>
<td></td>
<td>(0.00001)</td>
<td>(0.00001)</td>
<td>(0.00001)</td>
<td>(0.00003)</td>
</tr>
<tr>
<td>Cor</td>
<td>0.0027</td>
<td>0.0028</td>
<td>0.0035</td>
<td>0.0045</td>
</tr>
<tr>
<td></td>
<td>(0.00001)</td>
<td>(0.00001)</td>
<td>(0.00001)</td>
<td>(0.00002)</td>
</tr>
<tr>
<td>Mean</td>
<td>0.0102</td>
<td>0.0109</td>
<td>0.0131</td>
<td>0.0167</td>
</tr>
<tr>
<td></td>
<td>(0.00017)</td>
<td>(0.00021)</td>
<td>(0.00019)</td>
<td>(0.00020)</td>
</tr>
</tbody>
</table>

Note: The number outside the bracket is the mean value of Hamming loss of 5-fold cross verifications of 5 times, and the number inside the bracket is standard deviations.

The results of accuracy for different approximate parameters are shown in Table III. When the values of the approximate parameter reduce, the average value of the accuracy for the nine datasets increases. Similar to the calculation in Table II, for each 0.1 reduction of approximate parameters, the accuracy will increase by 6.18%, 4.81%, and 1.09%, respectively. The accuracy criterion of the proposed MAPD algorithm has an increasing and converging trend. Specifically, the mean value of the accuracy is optimal for the approximate parameter $\alpha=0.6$. When the approximate parameters are $\alpha=0.6$, $\alpha=0.7$, and $\alpha=0.8$, the 9 datasets get the optimal accuracy, respectively. The standard deviations of the accuracy criterion are stable.

Fig. 1 displays the outcome of the number of features chosen for various approximation values. We use the natural based logarithm of the size of the selected features when plotting the histogram. As the number of the selected features calculated by the proposed algorithm is 1 for several datasets (e.g., Gen, Cor and Yea), the scale of the chosen features in the Fig. 1 is zero. As shown in Fig. 1, when approximate parameter decreases, the number of chosen features by the proposed multi-label feature selection algorithm decreases. This finding is consistent with the definition of the proposed approximate Pareto dominance. When the value of the approximate parameter decreases, the number of dimensions that need to be satisfied is reduced for determining if one solution is better than the other or not. This makes it relatively easy to satisfy the concept of approximate Pareto dominance between solutions in the original set of solutions, decreasing the number of approximate Pareto dominated solutions and, in turn, the number of the chosen features of the suggested MAPD technique.

![Fig. 1](image_url)
Thus, it follows that datasets with high-dimensional labels are more suited for the suggested MAPD method. Compared with ParFS, SCLS, and AMI algorithms, the reduction ranges of Hamming loss of the proposed MAPD method are -0.02% ~ 20.99%, -0.15% ~ 2.36%, and -0.67% ~ 1.42%, respectively. The standard deviation values of the MAPD algorithm are similar to the other algorithms.

<table>
<thead>
<tr>
<th>Datasets</th>
<th>ParFS</th>
<th>SCLS</th>
<th>AMI</th>
<th>MAPD</th>
</tr>
</thead>
<tbody>
<tr>
<td>Sce</td>
<td>0.0035 (0.00026)</td>
<td>0.0010 (0.00015)</td>
<td>0.0011 (0.00008)</td>
<td>0.0004 (0.00004)</td>
</tr>
<tr>
<td>Emo</td>
<td>0.0236 (0.00037)</td>
<td>0.0293 (0.00131)</td>
<td>0.0090 (0.00063)</td>
<td>0.0057 (0.00073)</td>
</tr>
<tr>
<td>Yea</td>
<td>0.0066 (0.00013)</td>
<td>0.0071 (0.00026)</td>
<td>0.0070 (0.00006)</td>
<td>0.0068 (0.00026)</td>
</tr>
<tr>
<td>Bir</td>
<td>0.0539 (0.00059)</td>
<td>0.0519 (0.00060)</td>
<td>0.0467 (0.00058)</td>
<td>0.0534 (0.00000)</td>
</tr>
<tr>
<td>Gen</td>
<td>0.0030 (0.00011)</td>
<td>0.0027 (0.00006)</td>
<td>0.0030 (0.00017)</td>
<td>0.0028 (0.00023)</td>
</tr>
<tr>
<td>Med</td>
<td>0.0087 (0.00017)</td>
<td>0.0066 (0.00025)</td>
<td>0.0046 (0.00017)</td>
<td>0.0032 (0.00013)</td>
</tr>
<tr>
<td>Enr</td>
<td>0.0379 (0.00020)</td>
<td>0.0304 (0.00024)</td>
<td>0.0270 (0.00004)</td>
<td>0.0128 (0.00008)</td>
</tr>
<tr>
<td>Bib</td>
<td>0.0107 (0.00008)</td>
<td>0.0100 (0.00002)</td>
<td>0.0070 (0.00001)</td>
<td>0.0043 (0.00001)</td>
</tr>
<tr>
<td>Cor</td>
<td>0.0046 (0.00002)</td>
<td>0.0031 (0.00003)</td>
<td>0.0030 (0.00003)</td>
<td>0.0027 (0.00001)</td>
</tr>
<tr>
<td>Mean</td>
<td>0.0170 (0.00022)</td>
<td>0.0158 (0.00033)</td>
<td>0.0120 (0.00020)</td>
<td>0.0102 (0.00017)</td>
</tr>
</tbody>
</table>

The MAPD algorithm is tested on nine public multi-label datasets from different fields. Experiment results show that the proposed MAPD algorithm has a higher level of classification accuracy, a lower level of Hamming loss and a lower number of the selected features compared with the existing methods. Specifically, compared with ParFS, SCLS and AMI methods, the Hamming loss evaluation index is reduced by 0.68%, 0.56% and 0.18%, respectively. The proposed MAPD method obtains the optimal Hamming loss on six of nine datasets. On the accuracy evaluation index, the proposed MAPD algorithm also obtains the best classification accuracy on six of nine datasets; compared with ParFS, SCLS and AMI methods, the classification accuracy increased by 13.39%, 10.71% and 5.80%, respectively.

We compare the accuracy of different multi-label feature selection methods (see Table V). As shown in Table V, the proposed MAPD algorithm obtains the optimal average value of accuracy of nine datasets (0.7359). Compared with ParFS, SCLS, and AMI algorithms, the accuracy value of the proposed MAPD algorithm has increased by 13.39%, 10.71%, and 5.80%, respectively. The proposed MAPD algorithm obtains the optimal accuracy value for six datasets, i.e., Sce, Emo, Med, Enr, Bir, and Cor. ParFS algorithm obtains the optimal accuracy value for Yea dataset; SCLS algorithm obtains the optimal accuracy value for Gen dataset; AMI algorithm obtains the optimal accuracy value for Bir dataset. Moreover, for Cor, Bir, Enr and Med datasets with the highest number of labels, the proposed MAPD algorithm obtains the optimal accuracy value. Compared the ParFS, SCLS, and AMI algorithms, the increase ranges of the MAPD algorithm on the nine datasets are -1.09% ~ 41.21%, -1.37% ~ 39.11%, and -4.71% ~ 25.45%, respectively. In terms of the standard deviation, all algorithms are similar.

V. CONCLUSIONS

For the high-dimensional feature selection problem with multi-label dataset, a concept called approximate Pareto dominance is presented, which can be used to compare the qualities of two solutions. Compared with the traditional Pareto dominance concept, with the help of the approximate parameter, the proposed approximate Pareto dominance can solve the problem that Pareto dominance cannot do well when the evaluation function dimension of the solution is high. Then, using this concept, the feature selection problem with multi-label data is mapped to the problem of finding the set of approximate Pareto dominance solutions. Based on this transformation, we propose a method called MAPD to solve it.

<table>
<thead>
<tr>
<th>Datasets</th>
<th>ParFS</th>
<th>SCLS</th>
<th>AMI</th>
<th>MAPD</th>
</tr>
</thead>
<tbody>
<tr>
<td>Sce</td>
<td>0.9795 (0.00127)</td>
<td>0.9938 (0.00088)</td>
<td>0.9937 (0.00035)</td>
<td>0.9977 (0.00023)</td>
</tr>
<tr>
<td>Emo</td>
<td>0.8641 (0.00151)</td>
<td>0.8331 (0.00073)</td>
<td>0.9460 (0.00077)</td>
<td>0.9659 (0.00046)</td>
</tr>
<tr>
<td>Yea</td>
<td>0.9258 (0.00162)</td>
<td>0.9214 (0.00265)</td>
<td>0.9236 (0.00179)</td>
<td>0.9251 (0.00304)</td>
</tr>
<tr>
<td>Bir</td>
<td>0.4667 (0.00190)</td>
<td>0.4695 (0.00318)</td>
<td>0.5029 (0.00419)</td>
<td>0.4558 (0.00000)</td>
</tr>
<tr>
<td>Gen</td>
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<td>0.9465 (0.00274)</td>
<td>0.9375 (0.00172)</td>
<td>0.9342 (0.00313)</td>
</tr>
<tr>
<td>Med</td>
<td>0.6816 (0.00661)</td>
<td>0.7466 (0.00793)</td>
<td>0.8186 (0.00553)</td>
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</tr>
<tr>
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<td>0.2979 (0.00208)</td>
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<tr>
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<td>0.4077 (0.00321)</td>
<td>0.6127 (0.00025)</td>
</tr>
<tr>
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<td>0.2730 (0.00529)</td>
<td>0.3089 (0.00026)</td>
</tr>
<tr>
<td>Mean</td>
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<td>0.6288 (0.00405)</td>
<td>0.6779 (0.00310)</td>
<td>0.7359 (0.00029)</td>
</tr>
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REFERENCES


Dynamic Allocation Method of Incentive Pool for Financial Management Teaching Innovation Team Based on Data Mining

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Abstract—In order to reasonably allocate the amount of incentive pool and promote the unity of members of the financial management teaching innovation team, a dynamic allocation method of incentive pool for the financial management teaching innovation team based on data mining is proposed. This method constructs the incentive pool allocation index system by analyzing the principles of risk and income correlation, income and contribution consistency, individual and overall profit consistency, as well as the actual contribution of the financial management teaching innovation team, members' efforts and other factors that affect the allocation of incentive pool. After determining the index weight, the maximum entropy model is used to establish the incentive pool function of the financial management teaching innovation team project. The incentive pool scale decision model is established according to the prospect theory. After outputting the scale of the financial management teaching innovation team's incentive pool using the construction model, the incentive pool model of the financial management teaching innovation team is obtained. Based on the asymmetric Nash negotiation model, the allocation model for the incentive pool model of the financial management teaching innovation team is established, the improved artificial colony algorithm in the data mining algorithm is used to solve the model, and the dynamic allocation result of the incentive pool of the financial management teaching innovation team is obtained. The experiment shows that this method can effectively calculate the size of the incentive pool and allocate the incentive pool. The members of the financial management teaching innovation team have a high degree of satisfaction with the allocation result of the incentive pool, with allocation satisfaction consistently fluctuating around 96%.

Keywords—Data mining; financial management; teaching innovation team; incentive pool; dynamic allocation; artificial colony

I. INTRODUCTION

Financial personnel need to predict risks in advance and conduct effective control in time [1, 2]. The incentive pool is an incentive mechanism that takes part of the project income to motivate team members, retain core members and stimulate members to engage in work actively [3]. In the financial management teaching innovation team, the innovation input, actual contribution, and effort level of all participants change with the actual situation of the project [4]. A single profit distribution scheme is easy to lead to the unfair distribution of surplus target costs [5], which affects the internal harmony of the financial management teaching innovation team, and even leads to team breakdown. Therefore, scientific and reasonable dynamic allocation of incentive pool [6] is an effective means to improve the stability and enthusiasm of the financial management teaching innovation team. Now many scholars are studying the dynamic allocation method of incentive pools.

Study [7] proposed a transaction incentive mechanism for the V2G market, considering dynamic loss aversion. This method considered the incentive margin and liquidated damages when allocating incentives and proposed a minimum incentive strategy. This incentive pool allocation method needs to deduct a certain amount of incentive margin. Therefore, team members have a small number of funds in the incentive pool, which has a poor incentive effect. Research [8] proposed a dynamic incentive mechanism in mobile group intelligence perception. This method allocated the incentive pool based on the contribution degree of team members by evaluating the contribution degree of team members. However, this method is ineffective in evaluating team members' contribution degree, so its allocation effect is not good. Wu Z. et al. proposed an unbalanced fund allocation method, which only distributed the unbalanced funds in the team incentive pool. When the amount in the team incentive pool is large, and the unbalanced funds cannot be divided, it cannot complete the allocation of the incentive pool. In [9] authors proposed the reconstruction method of fund allocation standard based on the extended energy model. This method used the expansion energy synthesis model to calculate the premium value of the incentive pool project funds. It allocated the funds in the incentive pool according to the premium value. However, in the application process of this method, due to the iterative influence of the expansion energy synthesis model, its calculation of the capital premium value of the incentive pool project is not accurate enough, resulting in its poor final application effect. The authors in [10] proposed a matching method for the working capital of the supply chain incentive pool. This method used a dynamic discount decision to allocate the incentive pool capital by calculating the working capital's operating cycle and cycle income. However, this method is affected by the long operating cycle of the capital, resulting in the insufficient balance of the allocation of the incentive pool capital.

In response to the problems in the above research, team members are unable to complete the allocation of incentive
pool funds due to the small amount of funds allocated in the incentive pool, which has a poor incentive effect. When the amount of funds in the team incentive pool is large and unbalanced funds cannot be divided, the allocation of incentive pool funds is not balanced enough. This article studies the use of data mining methods to solve the problem. Utilize data mining techniques to analyze data related to team members’ contributions, performance, and motivational effects, and establish appropriate models to evaluate the value and potential of team members. This can more accurately determine the amount of incentive funds that each member should receive, in order to improve the effectiveness of incentives. Data mining based methods can help identify imbalances among team members and optimize the allocation of incentive pool funds. By considering factors such as member contributions, performance, and overall team goals, data mining algorithms can be used to optimize fund allocation, making it more balanced and fair. Data mining technology can analyze historical data and make predictions, helping to predict the potential and performance of future team members. Based on these prediction results, the allocation of funds in the incentive pool can be dynamically adjusted to adapt to changes in team members and achieve better incentive effects and balance. Data mining technology can provide personalized incentive strategies based on the characteristics and needs of team members. By mining and analyzing the data of members, incentive plans suitable for their personal characteristics and goals can be designed for each member to improve their participation and job satisfaction.

Data mining [11, 12] refers to the non-trivial process of revealing hidden, previously unknown and potentially valuable information from a large amount of data in the database. Data mining [13] is a decision support process, which is mainly based on artificial intelligence, machine learning, pattern recognition, statistical database, visualization technology, etc., which can make a highly automated analysis of enterprise data, inductive reasoning, mining out potential patterns, and helping decision-makers adjust market strategies, to reduce risks, and make correct decisions. Data mining is a technology to find rules from a large amount of data by analyzing each data. It involves three steps: data preparation, rule finding and rule representation. Data mining tasks include association analysis, cluster analysis [14], classification analysis, anomaly analysis, specific group analysis and evolution analysis. The research purpose of dynamic allocation of incentive pool for financial management teaching innovation teams is to explore an effective incentive mechanism to stimulate teachers’ enthusiasm and creativity in financial management teaching innovation. This study aims to establish a mechanism that can dynamically allocate incentive resources based on teachers’ contributions and performance, in order to improve teachers' job satisfaction, teaching quality, and teaching innovation level. By dynamically allocating incentive pools, teachers can be given corresponding rewards and incentive measures based on their performance and contribution in financial management teaching innovation. This can motivate teachers to actively participate in teaching innovation activities, improve teaching effectiveness and students’ learning experience. Here, based on data mining technology, this paper proposes a dynamic allocation method of incentive pool for financial management teaching innovation teams based on data mining to improve the dynamic allocation effect of incentive pool. The framework and main content of this study are as follows:

1) Clarify the basic principles for constructing incentive pool allocation models.

2) Analyze the factors that affect the allocation of incentive pools, and construct an incentive pool allocation indicator system and determine the indicator weights.

3) On the basis of clarifying the relevant principles, the paper constructs the incentive pool function of financial management teaching innovation team based on the maximum entropy model, and designs the Decision model of incentive pool size based on the prospect theory.

4) Implement the financial management teaching innovation team incentive pool allocation decision based on data mining from three aspects: constructing objective functions, calculating constraint conditions, and constructing models.

5) The effectiveness of the proposed method was verified through experimental analysis.

II. DYNAMIC ALLOCATION METHOD OF INCENTIVE POOL FOR FINANCIAL MANAGEMENT TEACHING INNOVATION TEAM

A. Basic Principles for the Construction of the Incentive Pool Allocation Model

First of all, the distribution of the incentive pool of financial management teaching innovation team is still the distribution of project profits in essence, which still needs to follow the general principles of the enterprise profit distribution model; secondly, according to the cause of formation of incentive pool and its distribution function, its distribution principle has a particularity. By summarizing the allocation scheme proposed by scholars and the practical experience of foreign financial management teaching innovation team projects, the allocation of incentive pool for financial management teaching innovation team projects should follow the following basic principles:

1) Principles related to risks and benefits: The potential profits and losses of all participants in the financial management teaching innovation team should be related to the real risks they bear [15].

2) The principle of consistency between income and contribution: The actual contribution of the financial management teaching innovation team and the contribution of each participant to the project are positively correlated with the profits obtained.

3) The principle of consistency between individual and overall profits: Whether the participants of the financial management teaching innovation team are profitable should be consistent with whether the project is profitable as a whole. The result of profit distribution can only lead to common profits or common losses, not profits of one party and losses of the other.
4) The principle of limited liability for financial management teaching innovation team participants: The potential losses of the members of the financial management teaching innovation team should be capped at the normal profits of each participant, the company's indirect costs and the share of profits, and the owner should bear the risks in excess.

5) 100% sharing principle: The members of the financial management teaching innovation team have 100% right to share the excess profits of the project. The above principles link the interests of participants with the interests of the project and encourage participants' behaviour to develop in a direction conducive to the project's overall profitability [16]. This paper will select indicators based on the above principles.

B. Analysis of Factors Affecting Incentive Pool Allocation

Based on the analysis of the characteristics of incentive pool allocation, this paper uses the all-factor method to comprehensively evaluate the contribution of the participants of the financial management teaching innovation team to the project surplus target cost from five aspects of the actual contribution, effort, risk and dynamic factors of the financial management teaching innovation team, combined with the design of specific indicators.

1) The actual contribution of the financial management teaching innovation team: The actual contribution of the financial management teaching innovation team is mainly reflected in the role of the internal members in the project profits. Drawing on the four major control objectives of enterprise project management, the financial management teaching innovation team members' completion of the part of the project they are responsible for is measured from the four aspects of quality, progress, safety and profit, reflecting the distribution principle of "distribution according to work" [17].

2) The degree of effort of the financial management teaching innovation team members: The effective cooperation and active innovation of the financial management teaching innovation team are the source of creating excess profits. The distribution system that only pays attention to the results without considering the efforts of the financial management teaching innovation team members is difficult to motivate the members of the financial management teaching innovation team to participate in project management and technical innovation actively. Therefore, this paper evaluates the efforts of enterprises from the perspective of project management and innovative activities.

3) The financial management teaching innovation team bears risks: The principle of "risk sharing and profit sharing" in the enterprise operation mode determines that the profit distribution mode should match the risks borne by the financial management teaching innovation team members. This paper sets three secondary indicators, namely, the degree of awareness, the control cost and the reduction of losses, from the perspective of risk control, to evaluate the contribution of the risk factors borne by the participants to the excess profits.

4) Dynamic factors: Enterprise projects usually have large investments and long cycles. When allocating excess profits, we should consider the spillover effect of the financial management teaching innovation team's investment income and special resource investment on the project.

C. Determination of Incentive Pool Allocation Index System and Index Weight

Based on the analysis of the factors affecting the allocation of the incentive pool in the financial management teaching innovation team, the indicator system and weight are determined. Firstly, 17 second-level indicators are refined according to the four first-level indicators of the financial management teaching innovation team's actual contribution, degree of effort, risk-taking and dynamic factors, and a consultation letter is sent to five scholars for indicator screening opinions; after three rounds, the indicators tend to be consistent, and then a consultation letter on the weight of indicators is issued. Both stages of consultation require respondents to evaluate their own judgment and familiarity and empower experts with the basis of their judgment and familiarity with the financial management teaching innovation team project through the analytic hierarchy process: knowledge of relevant references and cases (1), the experience of similar projects (0.75), theoretical analysis (0.5), personal intuition (0.25); Very familiar (1), familiar (0.75), average (0.5), not familiar (0.25). The calculated credibility of this consultation is 0.81; the Familiarity coefficient is 0.75; The overall authority level is 0.78, and the result is relatively reliable. The final indicator system and indicator weight are shown in Table I.

According to the indicator system in Table I, it can evaluate the contribution of team members to the incentive pool fund during the implementation of the financial management teaching innovation team project. The scores of the members of the financial management teaching innovation team project in the evaluation system in Table I are normalized and recorded as $I_{ij}$, then the evaluation of the financial management teaching innovation team members' profit contribution is as follows (1):

$$B = \sum_{i=1}^{n} \sum_{j=1}^{n} \xi_{ij} \xi_{ji} I_{ij}$$

In equation (1), $B$ represents the profit contribution evaluation of the members of the financial management teaching innovation team; $\xi_{ij}$ is the weight of the first-level indicator; $\xi_{ji}$ is the weight of the first-level indicator.
TABLE I. PROJECT INCENTIVE POOL INDEX SYSTEM OF FINANCIAL MANAGEMENT TEACHING INNOVATION TEAM

<table>
<thead>
<tr>
<th>Influencing factor layer</th>
<th>Primary index</th>
<th>Secondary indicators</th>
</tr>
</thead>
<tbody>
<tr>
<td>Actual contribution</td>
<td>Quality contribution (0.1)</td>
<td>Project quality qualification rate (0.4)</td>
</tr>
<tr>
<td></td>
<td></td>
<td>Number of quality accidents (0.6)</td>
</tr>
<tr>
<td></td>
<td>Progress contribution (0.1)</td>
<td>Progress contribution rate (0.5)</td>
</tr>
<tr>
<td></td>
<td></td>
<td>Return and shutdown frequency (0.5)</td>
</tr>
<tr>
<td></td>
<td>Profit contribution (0.2)</td>
<td>Profit contribution rate (0.5)</td>
</tr>
<tr>
<td></td>
<td></td>
<td>Profit loss rate (0.5)</td>
</tr>
<tr>
<td></td>
<td>Safety contribution (0.1)</td>
<td>Accident loss (0.7)</td>
</tr>
<tr>
<td></td>
<td></td>
<td>Accident frequency (0.3)</td>
</tr>
<tr>
<td>Effort</td>
<td>Project management (0.15)</td>
<td>Project management plan (0.6)</td>
</tr>
<tr>
<td></td>
<td></td>
<td>Decision cycle (0.4)</td>
</tr>
<tr>
<td></td>
<td>Innovation investment (0.05)</td>
<td>Professional training rate (0.5)</td>
</tr>
<tr>
<td></td>
<td></td>
<td>New technology (0.5)</td>
</tr>
<tr>
<td>Risk factor</td>
<td>Risk compensation (0.2)</td>
<td>Cooperation risk (0.3)</td>
</tr>
<tr>
<td></td>
<td></td>
<td>Capability risk (0.3)</td>
</tr>
<tr>
<td></td>
<td></td>
<td>Investment risk (0.4)</td>
</tr>
<tr>
<td>Dynamic factors</td>
<td>Input resources (0.1)</td>
<td>Capital investment (0.8)</td>
</tr>
</tbody>
</table>

D. Construction of Project Incentive Pool Model for Financial Management Teaching Innovation Team

The construction of the incentive pool has played a very good role in risk sharing and incentive compensation under the financial management teaching innovation team model. However, the size of the incentive pool, that is, the amount of funds invested in the incentive pool, has become the key issue of subsequent research and the prerequisite for allocating the incentive pool. If the number of incentive pools is too high, it may cause the enterprise to exceed the project's budget range and cannot save costs on the premise of encouraging all participants. If the number is too low, it will weaken the role of the incentive pool, make the setting role of the incentive pool not obvious, and cannot bring power to the financial management teaching innovation team to maximize the project benefits and save costs, and also make the tense relationship between the members of the financial management teaching innovation team. It brings difficulties to the subsequent research on the allocation of incentive pools [18], so how to determine the size of the incentive pool is very important for maximizing project benefits.

The project incentive pool model of the financial management teaching innovation team consists of three parts, namely C1, C2 and C3, as shown in Fig. 1.

![Fig. 1. Project incentive pool model of financial management teaching innovation team.](image-url)
According to the division of enterprise project costs, the C1 layer is the first part, which is the project cost, including the direct project cost, project measure cost and regulation cost of the project. The cost of this part is guaranteed, and these costs are placed at the first level because the project partner will never let his direct project cost and site management cost bear the risk. The C2 level is the second part, which is the project remuneration, including the project management fee and normal profits. Because this part may suffer losses, if any, it will be borne by all financial management teaching innovation team members, so this part is reflected in the risk sharing. The C3 layer is the third part, that is, the compensation layer for project risk incentives, including the loss value for failing to reach the target and the bonus for exceeding the target. The amount of this part of rewards or punishment mainly depends on the final actual output of the project. If the benefit exceeds the expected target, it will be rewarded, and if it is lower than the expected target, it will be punished. Here we introduce two keywords, target cost (TOC) and actual cost (AOC). TOC represents the project target cost agreed upon by all participants at the initial design stage [19]. AOC refers to the actual total cost after the completion of the project, which is equal to the actual direct project cost plus the project measure cost and regulation fee. There are four cases to analyze the model:

1) If the target cost is equal to the actual cost, the incentive compensation level is equal to the initial incentive value, the project is completed as scheduled, and there is no loss or surplus. The financial management teaching innovation team members can be rewarded as expected.

2) If the target cost is less than the actual cost, but the target cost plus the initial incentive value is greater than the actual cost, the incentive compensation level is equal to the initial incentive value minus (actual cost - target cost), the project loss, cost overrun, and the members of the financial management teaching innovation team jointly bear the loss.

3) If the target cost is greater than the actual cost, the incentive compensation level is equal to the target cost minus the final cost plus and the initial incentive value so that the project gains benefits, reduces costs, saves expenses, and members of the financial management teaching innovation team can get corresponding benefits.

4) If the target cost plus initial incentive value is less than the actual cost, there is no incentive compensation; that is, the incentive compensation level is equal to 0.

Based on the above analysis, this paper uses the maximum entropy model and prospect theory to determine the size of the incentive pool.

a) Construction of incentive pool function of financial management teaching innovation team based on maximum entropy model: The maximum entropy theory was put forward by Jaynes in 1957, which is one of the data mining algorithms. The maximum entropy theory is a mining algorithm that, under the premise of a certain number of probability distributions, the maximum probability expression of other location distributions is reached when the entropy value of the whole population is maximum, and the maximum entropy value is obtained to obtain the optimal results. The maximum entropy objective function is shown in equation (2):

$$h(x_i) = - B \int_{x_{i1}}^{x_{i2}} f_i(x) \ln f_i(x) dx$$  (2)

In equation (2), $x_i$ represents the $i$-th state value; $h(x_i)$ is the maximum entropy objective function; $f_i(x)$ is the probability density function of the variable $x$; $x_2$ and $x_2$ represent the upper and lower limits of the maximum differential entropy.

The constraint conditions for setting equation (2) are as follows:

$$h(x_i) \int_{-\infty}^{+\infty} f_i(x) dx = 1$$  (3)

$$\int_{-\infty}^{x_i} f_i(x) F_i(x) dx = \varepsilon_i$$  (4)

In the above equation, $F_i(x)$ is a distribution function of a random variable $x$, which represents a prior condition before solving the maximum unknown distribution probability of $x$; $\varepsilon_i$ is a set of constants.

According to the characteristics of the project delivery model of the financial management teaching innovation team, it is assumed that the utility value of the financial management teaching innovation team can be approximated as the sum of the utility values of all members. The random variable $x$ can represent the utility of each member. The function of the size of the incentive pool of the financial management teaching innovation team is $x_n(sca)$, and the proportion of the incentive pool to the utility fluctuates within the range of 0-1. $sca$ is the size of the incentive pool, $n = 1, 2, \ldots$ is the number of members in the financial management teaching innovation team. $h(x_i)$ is the entropy value of any member of the financial management teaching innovation team to maximize it. $x_n(sca)$ is substituted into the above model, and the size function of the incentive pool model of each member's utility financial management teaching innovation team is taken as a random variable to obtain the relationship function that best conforms to the random probability density distribution. In this paper, several Lagrange multipliers, such as $a_0, a_1, \ldots, a_i$, are introduced to form a new Lagrange function by combining the constraint function and the objective function. The extreme value of the original function is obtained by solving the stationary point. The incentive pool model of financial management teaching innovation team can be expressed as equation (5)
In equation (5), $G(X)$ represents the incentive pool model of the financial management teaching innovation team, and $k = 1, 2, \ldots$ represents the different roles of members in the financial management teaching innovation team project.

To calculate the differential at both ends of the above equation and assign the initial value to 0, the functional relationship between the utility of the financial management teaching innovation team's members and the size of the incentive pool can be obtained as equation (6):

$$x_n(sca) = G(X) \int_0^1 x f_n(x_n(sca)) dx$$  \hspace{1cm} (6)

b) **Incentive pool size decision model based on prospect theory:** Prospect theory is a psychological concept that studies the criteria for people making decisions from a psychological perspective. This theory believes that the estimation of events with different probabilities is different. After this paper uses the maximum entropy model to obtain the functional relationship between the utility of each participant in the IPD project and the size of the incentive pool, the best scheme of the size of the incentive pool can be solved by applying the prospect theory, introducing the value function and the weight function. Its specific mathematical expression is:

$$U(x, p) = x_n(sca) \sum_{i=1}^k w(p_i) \sum_{i=1}^k V(\pi_i)$$  \hspace{1cm} (7)

$$w(p_i) = \begin{cases} p^r, & \text{earn profit} \\ \frac{1}{[p^r + (1-p)^r]^r}, & \text{damage} \end{cases}$$  \hspace{1cm} (8)

$$V(\pi_i) = \begin{cases} \pi_i^\alpha, & \pi_i > 0 \\ -\tau(-\pi_i)^\beta, & \pi_i < 0 \end{cases}$$  \hspace{1cm} (9)

In the above equation, $U(x, p)$ represents the overall value of the incentive pool, and $i = 1, 2, 3 \ldots$ represents the possible future, $W(p_i)$ represents the weight function of the foreground $i$; $V(\pi_i)$ represents the value function of the prospect $i$. $\alpha$ and $\beta$ represent the risk attitude coefficient of the members of the financial management teaching innovation team, $\tau$ represents the loss aversion coefficient of each participant, and $r$ and $\delta$ represent the adjustment coefficient of the weight. Assuming there are two prospects $u^r$ and $u^l$, under the prospect $u^r$, the financial management teaching innovation team successfully completes the project and gains profits. Under prospect $u^l$, the project is incomplete, and all financial management teaching innovation team members lose. Then the decision model of the size of the incentive pool is shown in equation (10):

$$\max U = U(x, p) \sum_{i=1}^k \frac{p^r [u^r(sca)]^\alpha}{[p^r + (1-p)^r]^r} + \tau \sum_{i=1}^k \frac{q^l [u^l(sca)]^\beta}{[q^l + (1-q)^l]^\beta}$$  \hspace{1cm} (10)

In equation (10), $\max U$ represents the decision model of the size of the incentive pool.

Set the constraint conditions of the decision model for the size of the incentive pool as equation (11):

$$\int_0^1 f_n(u(sca)) du' = 1$$  \hspace{1cm} (11)

E. **Construction of Incentive Pool Allocation Model Based on Asymmetric Nash Negotiation Model**

1) **Principle of asymmetric Nash negotiation model:** The last section mainly introduced the method to determine the size of the incentive pool of the financial management teaching innovation team. After having an incentive pool of appropriate size, the project participants implemented and completed the entire financial management teaching innovation team project according to their scope of rights and responsibilities and achieved the expected results. However, the rational allocation of funds in the incentive pool will enable all parties to have sufficient motivation to work hard to achieve the project objectives without being jealous because of the excessive distribution of other parties [20], which is the main consideration of the members of the financial management teaching innovation team after the completion of the project implementation. This paper then proposes a mathematical model of an incentive pool allocation system based on the asymmetric Nash negotiation model, which is based on the risk preference and risk decision-making weight of the financial management, teaching innovation team members in the project implementation process to solve this problem quantitatively. The Nash negotiation model is a mathematical model for finding a Nash equilibrium solution. This equilibrium solution makes the strategy adopted by each player to other players optimal. The equilibrium solution was put forward and improved by Nash during his PhD study (1950, 1951). Before that, there was only a cooperative game in the simple game theory, and the game hypothesis that all parties in the game participated in cooperation made the game theory not develop rapidly. With Nash putting forward the concept of a non-cooperative game and providing a method to find an equilibrium solution that can satisfy all parties, the non-cooperative game problem that can have more extensive
application space has been developed and improved in the long run. In the research process in recent years, with the deepening of the research on Nash negotiation theory, there are two game forms, symmetrical game and asymmetric game [21], and different optimal strategy sets can be obtained according to the Nash negotiation model. The mathematical expression of the asymmetric Nash negotiation model is shown in equation (12):

\[
C = \max U \prod_{i=1}^{n} \left[ x_i \omega_i - g_i \omega_i \right] 
\]

In equation (12); \( n = 1, 2, \ldots \) refers to the participants in the negotiation; \( x_i \) represents the result of the negotiation, and \( g_i \) represents the starting point of the negotiation; \( \omega_i \) is the weight coefficient of negotiator \( i \).

Set the constraints of the asymmetric Nash negotiation model, as shown in equation (13):

\[
\begin{align*}
& x_i \geq g_i, \\
& s.t. \\ & x_i \in \Omega \\
& \sum_{i=1}^{n} \omega_i = 1 
\end{align*}
\]

In equation (13), \( \Omega \) represents the project negotiation domain.

2) Construction of incentive pool's allocation objective function of financial management teaching innovation team: Based on the above asymmetric Nash negotiation principle, the objective function of incentive pool allocation for financial management teaching innovation teams is established. When the financial management teaching innovation team members participate in the project, each member's position is different, and the role of undertaking the project and the number of resources is different. It should be more suitable for the asymmetric Nash negotiation model [22]. The Nash equilibrium solution obtained with the goal of maximizing each member and taking the project's own conditions as constraints is the most reasonable allocation proportion of the incentive pool of the financial management teaching innovation team. The objective function of incentive pool allocation for the financial management teaching innovation team is shown in equation (14):

\[
D = \max C \prod_{k=1}^{n} \left[ c_k \omega_k - g_k \omega_k \right] 
\]

In equation (14), \( c_k \) represents the utility value of each member considering risk sharing; \( g_k \) refers to the utility value when no agreement is reached, and the difference between the first two items is the increase in the interests of each participant minus the risk cost; \( \omega_k \) refers to the decision-making weight of the project participants in the risk sharing process.

3) Calculation of constraint conditions of incentive pool allocation objective function

a) Calculation of individual utility and group utility: According to the assumption of individual utility, this paper assumes that the members of the financial management teaching innovation team are in line with the mean-variance expected utility function [23]; that is, their individual utility function can be expressed as equation (15):

\[
c_k = E(x_k) - 0.5H_k \sigma^2(x_k)
\]

In equation (15), \( E(x_k) \) represents the utility expectation of each participant, \( \sigma(x_k) \) is the utility variance value of each participant, \( H_k \) represents the absolute risk aversion coefficient of each participant.

\( \zeta_k \) represents the distribution proportion of the financial management teaching innovation team members in the incentive pool. The group utility of the members of the financial management teaching innovation team can be expressed as equation (16):

\[
\sum_{k=1}^{n} c_k = E(x_k) - (0.5 \sum_{k=1}^{n} H_k \zeta_k^2) \sigma^2(x_k)
\]

In equation (16), \( E(x_k) \) represents the utility expectation of the financial management teaching innovation team; \( \sigma(x_k) \) represents the utility variance of the financial management teaching innovation team.

b) Absolute risk aversion coefficient setting: The concept of the risk aversion coefficient [24] was first developed in finance, investment and other fields. When the risk aversion coefficient is greater than 0, it indicates that the evaluated person is risk averse; that is to say, the existence of risk does not affect their decision-making. When the risk aversion coefficient is greater, the evaluated person's risk aversion degree is also higher; when the risk aversion coefficient is 0, the evaluated people are said to be risk-neutral. They do not care about the variance of the utility they can achieve but only about the expected utility they can achieve. Such people are theoretical model people that do not exist in reality; when the risk aversion coefficient is less than 0, the assessed can be defined as risk preference. The smaller the risk aversion coefficient is, the more risk they chase. For the financial management teaching innovation team, the different positions of the members in the project, the different workloads they undertake, and the different levels of development of their enterprises lead to different ways of evaluating their risk aversion. This paper adopts the definition standard of risk aversion coefficient in the US investment and
wealth management industry. The value is between 2 and 6; that is, the greater the value of $H_k$ is, the more risk-averse the financial management teaching innovation team is, and the smaller the value of $H_k$ is, the more risk-averse the financial management teaching innovation team is.

4) Incentive pool allocation decision model for financial management teaching innovation team based on data mining: According to the objective function and constraint conditions of the incentive pool allocation of the financial management teaching innovation team, the incentive pool allocation decision model of the financial management teaching innovation team is constructed, as shown in equation (17):

$$T = \max_c \prod_{k=1}^{n} \left[ DE(x_k) \omega_k - 0.5 \sum_{k=1}^{n} H_k \xi_k - \sigma(x_k) \omega_k \right]$$

(17)

The constraint conditions for solving the incentive pool allocation decision model of the financial management teaching innovation team are equation (18):

$$\begin{align*}
X_k & = \sum_{k=1}^{n} x_k \\
\sum_{k=1}^{n} \zeta_k & = 1
\end{align*}$$

(18)

According to the constraints of equation (18), the improved artificial swarm algorithm of data mining algorithm is used to solve the incentive pool allocation decision model of the financial management teaching innovation team, that is, the numerical value of $\zeta_k$, as a reasonable incentive pool allocation proportion, which can ensure that the members of financial management teaching innovation team can reach the Nash equilibrium solution when the project is completed.

The artificial bee colony algorithm is a bionic intelligent optimization algorithm that simulates bees searching for honey sources in space. The algorithm has fewer control parameters in the calculation process and good global convergence. The detailed steps of the incentive pool allocation decision model of the financial management teaching innovation team based on the improved artificial bee colony algorithm are as follows:

Step 1: Use equation (17) to get the solution of the incentive pool allocation decision model of all financial management teaching innovation teams, and then import it into the artificial bee colony algorithm.

Step 2: After setting the colony boundary conditions, use the incentive pool allocation decision model of the financial management teaching innovation team $a_k$ to generate the initial colony $N_p$, then the position value $y_{ij}$ of the $i$-th solution in the colony in the $j$-th dimension is as shown in equation (19):

$$y_{ij} = T y_{j_{\text{min}}} + \psi y_{j_{\text{max}}} - \psi y_{j_{\text{min}}}$$

(19)

In equation (19), $i$ represents the bees in $N_p$, that is, the solution of the incentive pool allocation decision model of the financial management teaching innovation team; $j = 1, 2, \ldots, D$ represents the dimension of the solution, $D$ represents the total dimension, $y_{j_{\text{max}}}$ and $y_{j_{\text{min}}}$ represent the maximum and minimum fitness values of the $j$-th dimension respectively; $\psi$ is a random constant.

Step 3: After calculating the position value of the solution of the incentive pool allocation decision model of all the financial management teaching innovation teams by using equation (19), sort the solution according to the size of the value, and select the solution of the incentive pool allocation decision model of the financial management teaching innovation teams corresponding to the first $\frac{N_p}{2}$ fitness as the employment bee of the bee colony. Take the hired bee as a discrete random variable and calculate its entropy, as shown in equation (20):

$$R(\text{fit}) = -\sum_{i=1}^{n} p_i \ln p_i$$

(20)

In equation (20), $R(\text{fit})$ represents the employment bee entropy, $O_i$ represents the probability of occurrence of the state.

According to the result of equation (20), the artificial bee colony is allowed to select the honey source following bee proportion $\tilde{\alpha}$ and the optimal honey source selection probability $\tilde{\beta}$ within an appropriate range, as shown in equation (21):

$$\begin{align*}
\tilde{\alpha} & = \frac{R(\text{fit})_{\text{max}} - R}{R(\text{fit})_{\text{max}}} \\
\tilde{\beta} & = 1 - \frac{R(\text{fit})_{\text{max}} - R}{2R(\text{fit})_{\text{max}}}
\end{align*}$$

(21)

In equation (21), $R$ represents the initial entropy of the colony; $\text{fit}_i$ represents the limit value of swarm sway.

Step 4: hire bees to update the honey source location according to equation (21).

Step 5: After the honey source location of the hired bees is updated, calculate the probability of the current status of each hired bee. According to this probability value, the hired bees search for new honey sources in their neighboring areas and greedily choose while recording the solution at this time.

Step 6: Judge whether all the employed bees in the current artificial bee colony have been allocated. If not, terminate the solution process. If yes, proceed to the next step.
Step 7: Set the number of honey source location updates $\Omega$. When the number of honey source location updates is lower than the time $\Omega$, a new honey source will be randomly generated.

Step 8: Record the optimal solution generated by the current artificial bee colony algorithm, and set the maximum iterative threshold $l_0$ of the artificial bee colony algorithm. When the number of iterations reaches the maximum threshold, stop the iteration and output the current optimal solution. Otherwise, continue the iteration.

After the above steps, the artificial bee colony algorithm transmits the optimal solution of the financial management teaching innovation team's incentive pool allocation decision model. It obtains the incentive pool allocation result of the financial management teaching innovation team.

III. EXPERIMENTAL ANALYSIS

Taking the financial management teaching innovation team as the experimental object, the financial management teaching innovation team is composed of 11 teachers who have innovative teaching concepts, are aggressive, are determined to reform, and are good at cooperation. The team members are from Nankai University, Shanghai University of Finance and Economics, Xi'an Jiaotong University, Jilin University, Chinese Academy of Social Sciences and other well-known universities. The professional title structure, educational background structure, age structure and academic background structure of the teaching team are reasonable. It is a high-level, highly educated, old, middle and young teachers' team with an optimized structure. The team has played an effective role in planning, organizing and coordinating the company's project promotion. It has effectively improved the company's project promotion. It has effectively improved the company's project revenue by using the experience and wisdom of team scholars. The teaching hours and incentive amount of the 11 financial management teaching innovation team members participating in the company's projects are shown in Table II.

This article mainly adopts three methods: asymmetric Nash negotiation, data mining, and artificial bee colony algorithm when implementing dynamic allocation of incentive pools for financial management teaching innovation teams based on data mining. To ensure the effectiveness of the experiment, set the parameters of the corresponding algorithm:

- Asymmetric Nash negotiation:
  Initial strategy setting: The initial strategy for each team member is $[0.2, 0.3, 0.5]$, indicating the proportion they expect to be allocated to the incentive pool.
  Objective function setting: The objective function is to maximize the average incentive value.
  Game theory parameter setting: the number of participants is 3, and the strategy space is $[0, 1]$.

- Data mining:
  Dataset selection: Use a financial management teaching innovation team dataset that includes data on member contributions and performance.
  Feature selection: select the performance indicators of members and participation in teaching innovation activities as the characteristics for analysis.
  Artificial bee colony algorithm:
  Bee quantity setting: Set 10 bees to search for incentive pool allocation schemes.
  Parameter settings: foraging distance is 2, memory factor is 0.8, and local search range is 0.2.

A. Generalization Performance Test of Incentive Pool Decision Model

The subjects' working characteristic curve, also known as the ROC curve, is one indicator describing a model's generalization ability. A model's generalization ability is good, indicating that the model has a strong calculation ability. The generalization ability of the financial management teaching innovation team and historical decision-making model built by the method of this paper is verified in the form of a ROC curve. The test results are shown in Fig. 2.

<table>
<thead>
<tr>
<th>Team member code</th>
<th>Teaching hours</th>
<th>Follow up the teaching content of the project</th>
<th>Expected incentive amount/10000 Yuan</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>109</td>
<td>Financial management</td>
<td>4.6</td>
</tr>
<tr>
<td>2</td>
<td>128</td>
<td>Enterprise Finance Theory</td>
<td>5</td>
</tr>
<tr>
<td>3</td>
<td>156</td>
<td>Cost control</td>
<td>10</td>
</tr>
<tr>
<td>4</td>
<td>98</td>
<td>Strategic cost management</td>
<td>8.5</td>
</tr>
<tr>
<td>5</td>
<td>69</td>
<td>Financial analysis</td>
<td>5</td>
</tr>
<tr>
<td>6</td>
<td>70</td>
<td>Financial decisions</td>
<td>6.5</td>
</tr>
<tr>
<td>7</td>
<td>90</td>
<td>Tax administration</td>
<td>9</td>
</tr>
<tr>
<td>8</td>
<td>115</td>
<td>Assets and liabilities</td>
<td>6.5</td>
</tr>
<tr>
<td>9</td>
<td>137</td>
<td>Profit management</td>
<td>4.5</td>
</tr>
<tr>
<td>10</td>
<td>164</td>
<td>Profit and loss assessment</td>
<td>7.8</td>
</tr>
<tr>
<td>11</td>
<td>103</td>
<td>Owner's equity</td>
<td>6.2</td>
</tr>
</tbody>
</table>
The analysis of Fig. 2 shows that the real rate value of the ROC curve of the model in this paper can reach more than 0.96, and the maximum value of the false positive rate value is only about 0.1. In contrast, the coverage area under the ROC curve is large. The above results show that the model in this paper can effectively adapt to new measured samples in the application process, with strong adaptability, good generalization ability, and more accurate output results.

B. Ability to Build Incentive Pool Allocation Index System

The reliability and validity of the incentive pool allocation indicator system selected are taken as the measurement index to analyze the reliability and validity of the different influencing factors of the incentive pool allocation of the financial management teaching innovation team and verify the ability of the method of this paper to select the incentive pool allocation indicator system of the financial management teaching innovation team. The test results are shown in Fig. 3.

According to the analysis of Fig. 3, the reliability and validity values of different influencing factors of the incentive pool allocation of the financial management teaching innovation team are 0.94, which shows that the incentive pool allocation indicator system of the financial management teaching innovation team constructed by the method in this paper is good. The incentive pool can be reasonably allocated according to this indicator system.

C. Calculation of Incentive Pool Size of Financial Management Teaching Innovation Team

After the financial management teaching innovation team is responsible for the company's project profitability, it generates its incentive pool. It uses the method in this paper to calculate the scale of the incentive pool during the project promotion cycle. The calculation results are shown in Fig. 4.

It can be seen from the analysis of Fig. 4 that with the increase of the project cycle, the incentive pool of the financial management teaching innovation team has gradually increased, but in the process of increasing, the scale of the incentive pool has also decreased, because when the enterprise project is promoted, its internal capital liquidity is strong.

There is a situation of a loan of profit funds in the project cycle. However, as the project continues to advance, the scale of the incentive pool of the financial management teaching innovation team continues to increase, which indicates that the project is profitable. The incentive amount that can be allocated to the financial management teaching innovation team members continues to increase. To sum up, the method in this paper can effectively calculate the size of the incentive pool of the financial management teaching innovation team and provide a basis for the subsequent allocation of the incentive pool.

D. Incentive Pool Allocation Results

When the enterprise project is completed, the incentive pool of the financial management teaching innovation team is allocated, and the allocation results are shown in Fig. 5.
It can be seen from the analysis of Fig. 5 that using the method in this paper can effectively allocate the incentive pool amount of the financial management teaching innovation team, and compared with the incentive amount of the team members in Table II, the allocation results meet the expected amount of the team members. Some team members allocate the incentive amount higher than their expected amount. The above results show that this method can effectively allocate the incentive pool of the financial management teaching innovation team. Its allocation result is more consistent with the team members and their results, with good application effect.

To further verify the allocation ability of the method in this paper to the incentive pool of the financial management teaching innovation team, it takes the satisfaction of the members of the financial management teaching innovation team with the allocation result of the incentive pool as a measure, to test the ability to allocate the amount of the incentive pool when the number of incentive pools is different. In order to make full use of the experimental results, the methods of reference [7], reference [8], reference [25], reference [9], and reference [10] are used to carry out the test. The test results are shown in Fig. 6.

It can be seen from the analysis of Fig. 6 that when the number of incentive pools allocated by the method in this paper is different, the satisfaction of the members of the financial management teaching innovation team with the allocation of incentive pools has always fluctuated around 96%, which shows that the results of the allocation of incentive pools are in line with the expected values of the members of the financial management teaching innovation team. When allocating incentive pools according to the method of reference [8], when the number of incentive pools is small, the satisfaction of team members with the allocation results is high. Still, when the number of incentive pools is large, the satisfaction of team members with the allocation results shows a downward trend. The satisfaction value of team members is lower than that of the method in researches [7], [25], [9] and [10] when allocating different rate incentive pools. The above results show that the method of this paper can not only effectively allocate an incentive pool, but also team members are satisfied with the allocation results of the incentive pool, and its application effect is good.

IV. CONCLUSION

When the amount of incentive pool is distributed unevenly, it will seriously affect team unity and lead to the poor effect of the team following up on the enterprise project. Therefore, this paper proposes a dynamic allocation method of incentive pool for financial management teaching innovation teams based on data mining. Taking the actual enterprise project and financial management teaching innovation team as the experimental object, the method in this paper has been fully verified. The verification results show that the method in this paper has a relatively significant application effect. However, the method in this paper still has some shortcomings. For example, it is difficult to determine the amount of funds in the incentive pool in the early stage of the project promotion. There are many unpredictable factors in the project promotion process. The calculation of the size of the incentive pool is usually carried out in the late stage of the project promotion. However, the calculation results of the method in this paper calculate the size of the incentive pool in each project cycle, and there is a certain deviation. Although the method of dynamically allocating incentive pools was proposed in the study, it may face some challenges in practical applications. For example, factors such as resource constraints, organizational structure, and culture may affect the actual allocation and execution of incentive pools. Therefore, when applying research results to practice, it is necessary to consider these practical feasibility issues. In the future, in the process of applying the cumulative prospect theory, according to the actual implementation of the project and the risks and changes that may be encountered, the risk prospect with more dimensions and different probability of occurrence will be set. When applying the cumulative prospect theory model, the risk prospect of the project can be
judged from different angles, and the more suitable incentive pool size can be solved.

REFERENCES


A Hybrid Federated Learning Framework and Multi-Party Communication for Cyber-Security Analysis

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Abstract—The term "Internet of Things" (IoT) describes a global system of electronically linked devices and sensors capable of two-way communication and data sharing. IoT provides various advantages, including improved efficiency and production and lower operating expenses. Concern about data breaches is constantly present, for example, since devices with sensors capture and send confidential data that might have dire effects if leaked. Hence, this research proposed a novel hybrid federated learning framework with multi-party communication (FLbMPC) to address the cyber-security challenges. The proposed approach comprises four phases: data collection and standardization, model training, data aggregation, and attack detection. The research uses the UNSW-NB15 cyber-security dataset, which was collected and standardized using the z-score normalization approach. Federated learning was used to train the local models of each IoT device with their respective subsets of data. The MPC method is used to aggregate the encrypted local models into a global model while maintaining the confidentiality of the local models. Finally, in the attack detection phase, the global model compares real-time sensor data and predicted values to identify cyber-attacks. The experiment findings show that the suggested model outperforms the current methods in terms of accuracy, precision, f-measure and recall.

Keywords—Federated learning; multi-party communication; cyber-security; machine learning; internet of things

I. INTRODUCTION

IoT defines the interconnection between physically moving objects through the internet integrated with the sensors, system memory, electronic chips, and other hardware [1]. In an IoT network, things interact with other nodes, which can be managed and controlled remotely [2]. This interconnectivity between the nodes enables them to gather and exchange information with other connected devices [3]. The IoT system provides ubiquitous connectivity to various intelligent systems, service industries, cloud computing, and applications [4]. Moreover, the IoT system enhances the number of communication networks and the amount of big data shared using the cloud architecture [5]. Recently, IoT-assisted approaches have had numerous applications in smart cities, online shopping, health care, banking, industries, etc., to protect human beings [6]. However, IoT systems are vulnerable to open attacks available on the network. Furthermore, the increased usage of IoT devices made them more vulnerable to cyber-attacks, making it essential to develop efficient mechanisms for predicting these threats [7]. The traditional techniques to detect and prevent cyber-attacks in IoT include conducting risk assessment, execution of authentication system, utilization of secure protocols, data encryption, etc., [8]. The risk assessment involves the evaluation of security controls to predict the vulnerabilities and cyber-attacks in the IoT system [9]. The implementation of the authentication system includes the usage of biometric authentication, password policies, and factor authentication technique to enable authorized user access in the IoT network [10].

In data encryption, the collected IoT data was encrypted using different algorithms during the data transmission [11]. However, conventional algorithms cannot offer higher security to IoT devices. Moreover, it cannot deal with the huge volume of data generated by IoT devices [12]. To address these challenges, Artificial Intelligent (AI) techniques are utilized to predict malicious events in the IoT network [13]. Machine learning (ML)-based methods are trained to identify data patterns and malicious events without requiring explicit protocols [14]. Hence, they have emerged as an effective attack detection tool in real-time IoT systems. Several types of ML techniques exist, such as supervised, unsupervised, and reinforcement learning [15]. The supervised learning technique utilizes a labeled database to train the algorithm to identify the patterns and categorize data into various classes, like benign or malicious [16]. But then, unsupervised learning does not require a labeled dataset to predict the data pattern or malicious data [17]. On the other hand, the reinforcement learning technique learns from experience and makes decisions to predict and prevent cyber-attacks in IoT networks [18].

Utilizing ML approaches in cyber-attack detection earned more advantages, like greater detection accuracy, improved speed, and faster response [19]. In addition, it enables the system to classify the types of cyber-attacks present in the data [20]. Moreover, it minimizes the human workload and allows them to detect complex cyber threats. However, the existing ML-based techniques face limitations and challenges, such as lack of labeled data, inconsistent data quality, storage, data privacy, etc. Moreover, recently various approaches such as supervised machine learning [21], false-data injection using ML model [22], Ensemble deep learning model [23], Deep learning-based IDS [24], Federated Learning-based IDS [26], etc., are designed to protect the network data from cyber-security attacks. However, these approaches are limited to generalizability that is these techniques cannot recognize the patterns of unknown attacks. Moreover, the performances of these models rely on dataset availability for training, and acquiring such a database is challenging. In addition, the techniques including deep learning and machine learning algorithms are computationally expensive and resource-intensive.
intensive. Also, training and deploying these techniques require more computational power, memory, and time, thus making them less effective for resource-constrained environments. Addressing these issues and challenges is important to develop a reliable and robust cyber-security framework. To address these challenges, federated learning with an intelligent MPC system was developed in this article. The basic concept of the proposed model is combining the advantages of Federated Learning and MPC algorithms to predict attacks and malicious events in IoT networks. The key contributions of the presented research work are described below:

- We develop an integrated Federated learning-based Multi-Party Computation approach to protect the IoT network data from malicious attacks.
- The network data was pre-processed using the z-score normalization approach enabling data standardization and effective training across subsets.
- Each IoT device trains a local model on its subset of pre-processed data using the federated learning approach. Then, the local models are encrypted using the MPC to confirm data security.
- The central server collects the encrypted models, and secure model aggregation was performed using the intelligent MPC approach to ensure privacy of the individual models.
- Finally, the performances of the developed model were analyzed and evaluated with existing techniques in terms of accuracy, f-measure, recall, and precision.

The organization of the presented article is described as follows, the current research related to cyber-security is described in Section II, the existing cyber-security model and its challenges are illustrated in Section III, the proposed methodology is explained in Section IV, the outcomes of the proposed work is analyzed in Section V, and the conclusion of the article is summarized in Section VI.

II. RELATED WORK

Some of the research articles related to the proposed work are listed below:

IoT defines the interconnection between different devices or objects, enabling the devices to collect, exchange and share information over the Internet. However, the tremendous growth of the IoT makes it more vulnerable to privacy and security risks. These security risks result in the limitation of energy resources and reduce the scalability of IoT devices. Hence, Yakub Kayode Saheed et al. [21] proposed an ML-supervised approach-based Intrusion Detection System (IDS) to address the security and privacy risks of the IoT. The developed model was tested and evaluated with the publicly available UNSW-NB15 dataset. The developed model utilizes the Principal Component Analysis technique to reduce the dataset dimensionality and minimum-maximum normalization concept to extract features. The implementation outcomes illustrate that the presented model earned greater accuracy of 99.9% for the UNSW-NB15 dataset. However, training the labeled dataset using the supervised technique consumes more time and increases the implementation cost.

The wide acceptance of the Industrial Internet of Things (IIoT) system resulted in various limitations like security, privacy, etc. The primary security threat affecting the IIoT system's function is the False Data Injection (FDI) attack. The primary concern of the FDI attacks is to mislead the industrial design by faking its sensor measurements. Mariam M. N. Aboelwafa et al. [22] developed an innovative FDI attack prediction algorithm. This method utilizes auto-encoders to detect the FDI attacks accurately. Initially, the sensor data was collected from the industrial sector and pre-processed to detect false data. Further, denoising auto-encoders are used to clean the falsified data. Finally, the performance of the developed design was evaluated and compared with existing techniques. The developed model is more effective in recovering the clean data from the injected dataset. However, the presented approach cannot handle highly dynamic and complex datasets.

The incorporation of IoT devices and communication networks in industrial control systems makes them vulnerable to cyber-attacks, making the system produce devastating results. Typically, IDS schemes are developed to assist the Information Technology (IT) system in predicting cyber-attacks. These models are pre-defined algorithms and are trained to identify the specific cyber threat. However, the traditional IDS design needs to consider the inconsistent nature of the industrial control system, which results in low accuracy and a high false positive rate. Abdulrahman Al-Abassi et al. [23] designed a Deep Learning (DL)-based IDS framework to overcome these issues. The proposed model builds a balanced demonstration of the imbalanced database and passes it to the ensemble DL attack prediction scheme. The developed algorithm integrates the deep neural network (DNN) and Decision tree classifier to predict cyber-attacks accurately. However, this approach cannot specify the cyber-attacks in the industrial control system.

The ultimate goal of the stakeholders from IIoT is its sustainability and trustworthiness to prevent loss. A secured IIoT network enables trust, security, privacy, safety, and reliability to the stakeholders. However, conventional security models are ineffective in protecting the network from security threats. Fazullullah Khan et al. [24] designed a reliable and accurate supervisory control and data acquisition (SCADA) system-based attack detection. The developed model integrates the DL-based recurrent units and decision tree classifier to predict the cyber-attacks in the IIoT system. The nonlinearity of the presented recurrent unit eliminates the irrelevant data. Thus it enhances the detection rate of the system. The proposed technique was validated with 15 different SCADA datasets. The experimental analysis illustrates that the presented algorithm outperforms the traditional attack detection schemes. However, it cannot process large-scale databases.

In the IIoT system, a tremendous amount of data processing occurs at the edge or cloud server to perform various analytics. Hence, various DL-based data analytic models are developed to effectively process the huge IIoT data. However, the learning process must be trustworthy and reliable to overcome the vulnerability of the IIoT networks. Therefore, Sharmistha
Nayak et al. [25] developed a DL-based routing algorithm for attack prediction in IIoT networks. This model deploys adversarial training to identify the injected attacks. Furthermore, the Generative Adversarial Network-Classifier (GAN-C) is designed to specify the type of attack that occurred in the IIoT system. The developed model utilizes parallel learning and prediction design to minimize the computational complexity of the system. The performance analysis demonstrates the usage of GAN-C significantly reduces the training time and increases accuracy. But the GAN-C is biased towards certain data types, making the system more complex.

Othmane Friha et al. [26] presented a federated learning (FL)-based IDS for providing security to the agricultural IoT architectures. The developed model protects the agricultural data through local learning in which the devices benefit the knowledge and share to enhance the detection accuracy. The presented model utilizes three different DL classifiers: convolutional neural network, recurrent neural network, and deep neural network. The performance of the developed algorithm was evaluated on other datasets such as InSDN, MQTTSet, and CSE-CIC-IDS2018. The results illustrate that the presented technique outperforms the non-federated learning techniques. However, the communication cost in the proposed method is high compared to other techniques.

III. PROBLEM STATEMENT

Even though the Internet of Things (IoT) has undoubtedly improved our everyday lives in many ways, its dispersed and decentralized architecture has also opened the door to novel cybersecurity risks. Methods based on Machine Learning (ML) have shown a lot of promise in spotting these dangers. The limitations of the traditional approaches include dependency on central servers for data processing and transmission; hence, these models cannot perform in a decentralized IoT environment. The existing model faces privacy problems, as they share sensitive data with the central server; this leads to data unauthenticated access to user-sensitive information. Moreover, these models transfer the large volume of data to the server, which is time-consuming, poses computational overhead, large energy consumption, and leads to poor resource utilization. As a result, improved collaboration methods for threat detection across IoT devices that respect users’ security and privacy are required. Federated learning with MPC may provide a safer and more private solution for IoT devices to work together on threat detection [27]. The motivation behind the proposed work is the benefits of federated learning and MPC to handle the decentralized and distributed nature of the IoT environment. The federated learning approach enables each IoT device in the network to train a local model using its data, confirming that the sensitive data remains on the device and is not shared with the central aggregator. This decentralized framework preserves data privacy and addresses the concerns interconnected with centralized data aggregation. Moreover, the distributed and decentralized feature of federated learning minimizes the amount of data transmission, thus it reduces the computational overhead and energy consumption issues faced by the existing models. Further, by aggregating the local models' knowledge through MPC, the global model benefits from the collective intelligence of all participating IoT devices. This collaboration enables the system to leverage diverse data sources and insights, leading to more accurate and robust threat detection. Moreover, the MPC approach confirms that the data transmitted during collaboration remains encrypted and makes it inaccessible to unauthenticated users. Thus, the designed framework addresses the problems faced by the existing models.

IV. PROPOSED FLB-MPC APPROACH FOR CYBER-ATTACK DETECTION

A novel hybrid cyber-security framework was developed in this article to predict attacks or malicious events in the IoT network. This model integrates the Federated Learning algorithm [28] and intelligent multi-party computation (MPC) [29] technique to secure network data from cyber-security threats. The federated learning is an algorithm, which enables multiple IoT devices to collaboratively train the model without sharing the raw data. On the other hand, the MPC is a cryptographic algorithm, which allows secure collaboration and computation on encrypted data. Initially, the cyber-security dataset was collected from the standard site and imported into the system. The raw dataset was pre-processed using the z-score normalization technique, and the dataset was partitioned into multiple subsets in which different parties own each subset. Further, each IoT device trains a local model on its subset of pre-processed data.

The local model can be an ML design trained using the federated learning algorithm. After the completion of local model training, it is encrypted using the MPC to confirm that the privacy of the local model is preserved. Then, the encrypted local model is sent to the central server for aggregation.

The central server gathers the encrypted local models from all the IoT devices and aggregates them to form a global model. The proposed work performs the model aggregation using the MPC algorithm. Finally, the global model is utilized to predict cyber-attacks in real-time by comparing sensor readings from each IoT device to the expected values detected by the model. If there is a difference from the expected values, it is represented as a cyber-attack. Finally, the performances of the proposed work are estimated in model evaluation in terms of accuracy, precision, recall, and f-measure. The proposed framework is explained in Fig. 1.

Fig. 1. Proposed hybrid federated learning framework with multi-party communication.
A. Data Pre-processing

The cyber-security dataset containing the network traffic logs from numerous IoT devices is initially collected. Each log contains information regarding the device, log time, port numbers, IP addresses, packet sizes, etc. In the proposed work, the UNSW-NB15 cyber-security dataset was collected from the Kaggle site and imported into the system. The dataset initialization is expressed in Eq. (1).

$$D_{un} = \{d_{a1}, d_{a2}, \ldots, d_{an}\} \quad (1)$$

Where $D_{un}$ denotes the input dataset, $d_a$ the data present in the dataset, and $n$ the number of data present in the dataset. Then the raw dataset was pre-processed using the z-score normalization technique. Z-score normalization is a data pre-processing approach deployed in ML to convert numerical attributes to mean and standard deviations of 0 and 1, respectively. The mean and standard deviation calculation for each feature in the dataset is expressed in Eq. (2), and (3).

$$M_e(D_{un}) = \frac{1}{n} \sum (f_i) \quad (2)$$

$$S_D(D_{un}) = \sqrt{\frac{1}{n} * \sum (f_i - M_e)^2} \quad (3)$$

Here $S_D$ represents the standard deviation, $M_e$ denotes the mean value, and $f_i$ indicates the value of the $i^{th}$ feature. The dataset normalization is formulated in Eq. (4).

$$N(f_i) = \frac{(f_i - M_e)}{S_D} \quad (4)$$

In this approach, the mean of the feature is subtracted from each value in the feature, and the result is divided by the feature's standard deviation. The outcome values have a mean and standard deviation of 0 and 1, which enables the ML algorithms to learn easily from the data.

B. Model Training

The model training was performed using the federated learning algorithm in the proposed framework. Federated learning is a ML-based algorithm that performs model training using the data distributed across multiple devices without the necessity of centralized data collection. Initially, the pre-processed dataset was partitioned, and then local model training was performed to train the models for cyber-attack detection. The dataset was partitioned into multiple subsets and distributed across the IoT devices. In the developed work, random partitioning was deployed to divide the dataset into approximately equal sizes of subsets. The formulation of data partition is expressed in Eq. (5).

$$D_{un} = \{D_{un1}, D_{un2}, D_{un3}, \ldots, D_{unK}\} \quad (5)$$

Where $K$ denotes the number of IoT devices. In federated learning, the data is partitioned so that each party owns a data subset, and the subsets are non-overlapping. For example, the party $p$ owns a subset of data $D_{unp}$.

The local training model begins after the completion of data partitioning. It is the process of training a ML design on the data subset owned by each party in the network.

In the proposed work, each party trains a local model on its own data subset without sharing the data with other parties. Here, each party $p$ trains a local model $M_p$ with its own data subset $D_{unp}$ in a decentralized manner and $L_p$ denotes the local model on the device.

C. Model Aggregation

After the completion of local model training, it is encrypted using the MPC approach to confirm the security and privacy of the model. The local model is encrypted using cryptographic techniques like MPC. The encryption of local models is expressed in Eq. (6).

$$E_n(L_p) = Ef_{Ke}(L_p) \quad (6)$$

Where $E_n$ denotes the encryption process, $Ef$ refers to the encryption function, and $Ke$ represents the key. Further, the central server gathers the encrypted local models from all the IoT devices and aggregates them to form a global model. In the proposed work, the model aggregation was performed using MPC. The global model was utilized to predict cyber-attacks in real time by comparing sensor readings from each IoT device to the expected values detected by the model. If there is a variation from the expected values, it predicts a cyber-attack. It is represented in Eq. (7).

$$D'_p = |S_p - S'_{p}| \quad (7)$$

where $D'_p$ indicates the deviation between the actual and expected values, $S_p$ denotes the sensor reading on the device $p$, and $S'_{p}$ the expected value detected by the global model. If the deviation value exceeds the predefined threshold value $T_k$, it is predicted as an attack. Otherwise, the central server updates the global model, which is expressed in Eqn. (8).

$$R' = R - \lambda \sum (D'_p - \nabla L_p) \quad (8)$$

Where $R'$ denotes the updated global model, $R$ indicates the previous global model, $\lambda$ refers to the learning rate, and $\nabla L_p$ defines the gradient of the local model on the device $p$. This process continues until the global model achieves the desired accuracy.
Fig. 2. Flowchart of the proposed framework.

The flowchart of the proposed work is displayed in Fig. 2. In addition, the step-by-step procedure of the proposed model is illustrated in pseudo code format in algorithm 1.

**Algorithm** pseudo-code of the proposed model (Algorithm 1)

```plaintext
start

1. Initialize the machine learning model with random weights \( R_w \)
2. Initialize the number of training rounds, IoT devices
3. Initialize the batch size and learning rate for local training
4. for \( t = 1, t \leq T; t + + \)
5.   Partition the dataset: \( D_{un1}, D_{un2}, \ldots, D_{unk} \)
6.   Each IoT device \( p \) trains the local model \( L_p \) with its subset \( D_{unp} \)
7.   \( En = encrypt(L_p) \) // encrypt the local model using the MPC algorithm
8.   Send the encrypted model to the central server.
9.   \( L_p = encrypt(En)/N \) // aggregate the local model to form a global model
10. If \( (D'_p > T_R) \)

   11.   “Cyber-attack” // The system predicts a “cyber-attack.”
12.   else (data update);

end
```

V. RESULTS AND DISCUSSION

This research proposed a hybrid intelligent cyber-security mechanism to detect cyber-attacks in the IoT system. The model utilizes federated learning and MPC to predict cyber-attacks effectively. The input dataset was initially collected and pre-processed using the z-score normalization approach. Further, a federated learning algorithm was designed to partition the normalized dataset. Moreover, it enables the system to train the local model in a decentralized manner (without sharing the raw dataset). Moreover, an intelligent MPC was developed to encrypt the local models, enabling it to preserve the privacy of the local models.

Finally, the encrypted local models are sent to the central server for model aggregation. The model aggregator combines them to form a global model. The global model predicts cyber-attacks by analyzing expected and actual sensor readings deviation. The presented work was implemented in MATLAB software, version R2020a, and the results are analyzed. The parameters and their description are tabulated in Table I. In addition, a comparative assessment was carried out to manifest the effectiveness of the proposed work.

A. Dataset Description

The presented model was trained and tested with the UNSW-NB15 dataset. It is a cyber-security dataset that was developed to help the researchers on IDS and network security. It consists of a massive collection of network traffic data, including the different types of cyber-attacks such as DoS, Probe, U2R, R2L, etc. Table II illustrates the attack types and number of records in the UNSW-NB15 dataset. The dataset contains ten different attack types indicated by a separate class label.

<table>
<thead>
<tr>
<th>Attack Type</th>
<th>Number of records</th>
</tr>
</thead>
<tbody>
<tr>
<td>Normal</td>
<td>221,876</td>
</tr>
<tr>
<td>Analysis</td>
<td>2,244</td>
</tr>
<tr>
<td>Backdoor</td>
<td>18</td>
</tr>
<tr>
<td>DoS</td>
<td>122,846</td>
</tr>
<tr>
<td>Exploits</td>
<td>445,104</td>
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<tr>
<td>Fuzzers</td>
<td>242,720</td>
</tr>
<tr>
<td>Generic</td>
<td>188,220</td>
</tr>
<tr>
<td>Reconnaissance</td>
<td>104,13</td>
</tr>
<tr>
<td>Shellcode</td>
<td>1,571</td>
</tr>
<tr>
<td>Worms</td>
<td>130</td>
</tr>
</tbody>
</table>

TABLE I. PARAMETER AND SPECIFICATION

<table>
<thead>
<tr>
<th>Parameters</th>
<th>Specification</th>
</tr>
</thead>
<tbody>
<tr>
<td>Tool</td>
<td>MATLAB</td>
</tr>
<tr>
<td>Version</td>
<td>R2020a</td>
</tr>
<tr>
<td>OS</td>
<td>Windows10</td>
</tr>
<tr>
<td>Dataset</td>
<td>UNSW-NB15</td>
</tr>
<tr>
<td>Dataset size</td>
<td>1.9GB</td>
</tr>
</tbody>
</table>

TABLE II. DATASET DESCRIPTION
In addition, it includes 49 different features like port numbers, protocol types, source and destination IP addresses, byte counts, etc. The University of New South Wales researchers in Australia created the dataset. It comprises nearly 2.5 million network packets in which 175,341 records are selected for training and 82,332 records are chosen for testing.

B. Case Study

The working procedure of the proposed work is explained in this case study. Initially, the input cyber-security dataset was collected from the standard site and imported into the system. In the present study, the UNSW-NB15 dataset was collected and fed into the system for further processing. After data initialization, the raw dataset was pre-processed using the z-score normalization technique. Further, a federated learning setup was established for performing data partition and local model training. Initially, the pre-processed dataset was partitioned into approximately equal-sized subsets and distributed to IoT devices.

Federated learning helps the system divide the dataset so that each party owns a data subset, and the subsets are non-overlapping. In local model training, each party trains a local model with its own data subset without sharing the data with other parties. Further, the trained local models are encrypted using the MPC algorithm to ensure the privacy and security of the models. After encryption, the central server collects the encrypted local models from all the IoT devices and aggregates them into a global model. The global model was used to detect cyber-attacks by matching the real-time sensor reading with the expected values predicted by the model. If the difference between actual and estimated values is greater than the threshold value, it is predicted as Attack. Else, the central server updates the global model.

This process is continued until the system achieves the desired accuracy. Finally, the performances of the proposed technique were estimated by implementing it in the MATLAB tool. In addition, a comparative assessment was performed to validate the outcomes of the proposed work. The implementation results of the proposed work are tabulated in Table III.

The performance analysis illustrates that the proposed model earned greater performances, such as 99.98% accuracy, 100% precision, 99.8% recall, and 99.87% f-measure. In addition, the presented model achieved a less computational time of 2.34ms.

C. Performance Analysis

In this module, the performances of the proposed work are examined in terms of accuracy and loss relative to the increasing number of epochs. The training and testing accuracy of the proposed framework over increasing epochs count is demonstrated in Fig. 3. The training accuracy denotes the model accuracy on the train dataset which is how accurately the designed model detects the attacks on the train set. In addition, it measures how efficiently the designed model learns the attack patterns during the training phase. On the other hand, the testing accuracy measures how well this approach works on unseen data. The designed framework achieved an approximate training accuracy of 0.99, which demonstrates that the developed model works well on the training dataset and quickly learns the interconnection between the normal and attack data. Consequently, the presented approach attained an approximate testing accuracy of 0.97; this shows that the designed framework predicts cyber-attacks on unseen data effectively.

Similarly, the training and testing losses were over increasing epochs. Fig. 4 portrays the evaluation of training and validation losses. The training loss represents the difference between the actual and predicted results on the training dataset. It measures how well the proposed model fits the training set and learns the relationship between network data and attack patterns. This approach attained an average training loss of 0.003, which shows that the proposed model fits well on the training dataset.

<table>
<thead>
<tr>
<th>TABLE III. PERFORMANCE ANALYSIS</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Metrics</strong></td>
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<td>----------------</td>
</tr>
<tr>
<td>Accuracy (%)</td>
</tr>
<tr>
<td>F-measure (%)</td>
</tr>
<tr>
<td>Recall (%)</td>
</tr>
<tr>
<td>Precision (%)</td>
</tr>
<tr>
<td>Processing time (ms)</td>
</tr>
</tbody>
</table>

Fig. 3. Training and testing accuracy evaluation.

Fig. 4. Training and testing loss evaluation.
The testing loss defines the model's generalization ability on unseen data (unknown attack patterns). This approach earned a testing loss of 0.045, which indicates that the proposed model accurately classifies cyber-attacks in real-world scenarios.

Furthermore, to evaluate the effectiveness of the proposed work on real-world scenarios the system performances such as attack detection accuracy, computational time, and energy consumption are analyzed for increasing number of participants and data size in the network.

Here, the model accuracy was evaluated by increasing the data size and number of users in the IoT network and it is graphically represented in Fig. 5. When the network user count is 20, the proposed model earned an average accuracy of 99%. Further, on increasing the network users to 40, 60, 80, and 100, the designed model obtained an approximate accuracy of 99.5%, 99.6%, 99.7%, and 99.99%. The increase in system accuracy over increasing network users and data size is because of the ability of the proposed model to compare the sensor readings with the expected values. This makes the system more refined and attuned to the attack patterns owing to improved accuracy over increasing network users and data size.

Error rate defines the discrepancy between the actual and predicted results. This metric determines the efficiency of the model in detecting the attacks. Fig. 6 illustrates the error rate over increased network users and data size. The proposed model earned a minimum error rate of 1%, 0.9%, 0.8%, 0.6%, and 0.5%, respectively for 20, 40, 60, 80, and 100 network users at increasing data size from 50MB to 500MB. The lowering of the error rate illustrates that despite the increased network users and data size the proposed model correctly detects the attacks and normal events.

The computational complexity over increasing network users and data size is illustrated in Fig. 7, the computational time of the system was examined at different network user counts and the increasing data size (50 to 500MB). When a user in the network is 20, the proposed system consumed an average computational time of 1.75ms. Consequently, on increasing the network users to 40, 60, 80, and 100, the designed model consumed an approximate time of 2ms, 2.25ms, 2.5ms, and 2.75ms, respectively. From the analysis, it is observed that the computational time increases with increasing the number of users and data size in the IoT network. However, the computational time consumed by the presented model is relatively small, which is achieved by integrating federated learning and MPC. The local model training using the FL approach significantly reduces the amount of data transmission over the network; this minimizes the computational overhead and leads to faster communication. The integration of these techniques optimizes the computational process and reduces the time complexity.
The energy consumption of the system over increasing data size and network users is graphically represented in Fig. 8. Typically, on increasing the network user count and data size, the energy consumption of the IoT network increases. Similarly, energy consumption increased with the increase in user count and dataset size. When the user count is 20, the model attained an average energy consumption of 95W over increasing data size (50 to 500). Consequently, the proposed model obtained an approximate energy consumption of 100W, 105W, 110W, and 115W, respectively for 40, 60, 80, and 100 network users. Similar to the traditional model, the energy consumption of the network increases over increased network user and dataset size. However, the energy consumption of the system is comparatively lower than other models. This is due to the incorporation of the FL method in the proposed model. The FL technique in the designed model enables collaborative training that is it does not require a centralized server and instead of transferring the raw data to the central server, the proposed framework transmits only encrypted local models for aggregation. This process significantly reduces the energy consumption for data transmission. This comprehensive performance analysis manifests that the developed model efficiently predicts the attacks or malicious events on the IoT network.

D. Comparative Analysis

In this section, the performances of the proposed model are compared with existing techniques for validation purposes. The current methods such as Deep Belief Network-based Intrusion Detection System (DBN_IDS) [30], Deep Convolutional Generative Adversarial Network (DCGAN) [31], Distributed Convolutional Neural Network (DCNN) [32], Feed-forward neural network (FFNN) [33], and Vector convolutional deep learning (VCDL) [34] are used in comparative performance. The outcomes of these techniques are evaluated by implementing them in the MATLAB tool for the UNSW-NB15 dataset.

1) Accuracy: Accuracy defines the percentage of correct classification of instances. It is defined as the proportion of the sum of true positives and negatives to the total number of instances[35]. Moreover, it represents how exactly the system performs a cyber-attack detection function. The accuracy calculation is formulated in Eq. (9).

\[ A_{sy} = \frac{w^+ + w^-}{w^+ + w^- + z^+ + z^-} \]  

Where \( A_{sy} \) defines the system accuracy, \( w^+ \), \( w^- \), \( z^+ \), and \( z^- \) denotes the true positive, true-negative, false-positive, and false negative, respectively.

To manifest the accuracy of the proposed technique, it is compared with existing cyber-security models. Fig. 9 shows the comparison of accuracy. This section uses existing models such as VCDL, FFNN, DCNN, DCGAN, and DBN_IDS for comparative analysis. The accuracy earned by the conventional approaches is 94.72%, 89.43%, 95.45%, 91.78%, and 95.65%, respectively. But the developed technique attained greater accuracy of 99.98%, which is higher than existing techniques. The highest accuracy obtained by the designed model illustrates that it predicts cyber-attacks accurately.

2) Precision: Precision represents the proportion of true positives out of the total positive instances classified by the system[36]. It is also defined as the system’s ability to correctly predict cyber-attacks without creating false positives. The precision calculation is represented in Eq. (10).

\[ P_{cs} = \frac{w^+}{w^+ + z^+} \]  

Where \( P_{cs} \) represent the precision percentage. The high precision defines that the proposed system accurately identifies the cyber-security threats with fewer false positives.

Fig. 9. Comparison of accuracy.
In model evaluation, a comparative analysis is important to identify the strength and weaknesses of the proposed techniques. Here, the precision percentage of different machine learning algorithms for predicting cyber-attacks on the UNSW-NB15 dataset was compared with the proposed technique. The existing ML techniques like VCDL, FFNN, DCNN, DCGAN, and DBN_IDS earned precision percentages of 93.91%, 90.54%, 94.25%, 90.17%, and 94.95%, respectively. However, the proposed technique obtained a higher precision percentage of 100%, which is comparatively greater than existing techniques. The comparison of precision is illustrated in Fig. 10.

3) Recall: The recall is a performance metric that measures the proportion of true positives out of a total number of actual positive instances[37]. It denotes the system's capability to classify cyber-security threats irrespective of false positives correctly. The recall calculation is expressed in Eq. (11).

\[
\text{Re}_{\|} = \frac{w^+}{w^+ + z^-}
\]  
(11)

Where Re_{\|} denotes the recall. The high recall rate defines that the system exactly predicts most of the actual cyber-attacks.

The comparison of recall is shown in Fig. 11. The recall is compared with the existing techniques such as VCDL, FFNN, DCNN, DCGAN, and DBN_IDS. The proposed technique attained a recall percentage of 99.8%. On the other hand, the recall percentage attained by the conventional methods is 92.24%, 88.56%, 93.41%, 91%, and 94.16%, respectively. The comparative performance of recall describes that the developed model achieved a better recall percentage than existing techniques.

4) F-measure: F-measure is the metric that combines both precisions and recalls into a single score. It is evaluated as the harmonic mean of these two metrics[38]. The system achieves greater f-measure value only when both recall, and precision score is high. The f-measure calculation is represented in Eq. (12).

\[
F_{\|} = \frac{2 \left( \frac{Pi_{cs} \cdot Re_{\|}}{Pi_{cs} + Re_{\|}} \right)}
\]

(12)

Where \(F_{\|}\) refers to the f-measure.

The comparison of f-measure is shown in Fig. 12. The f-measure is compared with existing techniques like VCDL, FFNN, DCNN, DCGAN, and DBN_IDS. The proposed technique attained a f-measure percentage of 92.18%, 89.54%, 92.56%, 90.75%, and 94.10%, respectively, less than the f-measure earned by the proposed model. This shows that the developed model balances the precision and recall metrics optimally.
The comparative performance of different cyber-security models with the proposed technique is tabulated in Table IV.

5) Computational time: Computation time defines the time taken by the system to perform tasks such as data pre-processing, encryption, data transmission, decryption, etc. The computational time of the system is tabulated in Table V.

The computational time of the system was compared with an existing model including the VCDL, FFNN, DCNN, DCGAN, and DBN_IDS for validation purposes.

Table VI lists the computational complexity of different models. The above-mentioned models consumed 16.02ms, 21.96ms, 16.93ms, 20.5ms, and 17.92ms, respectively. The time taken by the proposed model is 2.34ms, which is comparatively less than the existing techniques. The comprehensive comparative assessment proves that the developed model outperformed the existing techniques.

E. Discussion

This research article presents an integrated framework to detect attacks or malicious events in the IoT network. The developed strategy combines the advantages of Federated learning and MPC algorithms. To begin with, a cyber-security database was acquired from a standard source and fed into the system. This collected database acts as the basis for training the models and predicting potential cyber-attacks. Before training, the raw data was pre-processed using the z-score normalization method. The data normalization confirms that the data was standardized and ready for further analysis. Further, the standardized database was partitioned into multiple subsets, with each subset being owned by different parties (IoT devices). Each party trains the local model using the FL model on its subset of pre-processed data. The utilization of the FL approach for training ensures that the sensitive information owned by IoT devices remains secure and confidential. Moreover, this distributed training minimizes the computational overhead on individual IoT devices and overcomes the drawback of centralized training in which a single server is responsible for training the entire dataset. In addition, it improves scalability and ensures efficient resource utilization.

After local model training, these models are encrypted using the MPC algorithm; this ensures that the model's sensitive data remains confidential and prevents unauthorized data access. Consequently, the encrypted local models are transmitted to the central server for aggregation. The central server receives the encrypted local models from all IoT devices and combines them to form a global model using the MPC, without having direct access to the data. This step permits secure and privacy-preserving collaboration among the different parties. Finally, the global model containing the collective knowledge of all local models is employed to detect cyber-attacks. The real-time prediction ability of the global model improves the system’s capacity to respond promptly to cyber-attacks and mitigate potential damages.

The presented approach was evaluated with the publically available network data named UNSW-NB15 and the performances are analyzed in terms of accuracy, computational time, error rate, and energy consumption. Moreover, the developed model performances are examined in real-time scenarios over increasing network users and data size. This intensive performance evaluation demonstrates the effectiveness of the proposed model in handling real-time applications. Thus, the integration of federated learning with the MPC permits the data to remain on the IoT devices, and only encrypted gradients are sent to the central server. This enables the system to preserve security by ensuring that sensitive information is not transmitted to the central server and reduces the risk of data breaches. In addition, the proposed model minimizes the latency, especially in scenarios with limited network connectivity. Moreover, it enhances the robustness to device failures. Since the training is distributed across multiple devices, the failure of one or more devices does not necessarily lead to the failure of the entire system. Furthermore, to manifest the robustness of the developed model the obtained results are compared with the existing techniques. The comparative analysis demonstrates that the proposed model outperformed the existing models in terms of accuracy, recall, f-measure, and precision.

VI. CONCLUSION

This paper proposed a ML-based cyber-security framework to secure IoT networks from attacks and malicious activities. The proposed framework integrates the FL technique and MPC algorithm to preserve security in the IoT. Federated learning
partitions the pre-processed data into equal size subsets. Further, each IoT devices train a local model on its subset of data. Then the trained local models are encrypted using the MPC approach to preserve the model’s privacy. Finally, the central server gathers the encrypted local model and forms a global model to predict the cyber-attacks in the system. The developed model is implemented in the MATLAB tool on the UNSW-NB15 dataset. Moreover, a comparative analysis was performed to illustrate the robustness of the proposed algorithm. The comparative analysis shows that performance like precision, accuracy, recall, and f-measure are improved by 5.05%, 4.33%, 5.64%, and 5.77%, respectively. Thus, the designed model accurately predicts the cyber-attacks in IoT networks. Although the proposed framework provides high privacy, it cannot handle large-scale federated learning scenarios. In addition, the proposed framework is vulnerable to adversarial attacks, such as model poisoning or data poisoning attacks. Since IoT systems are composed of devices and sensors from different vendors and manufacturers, making interoperability a critical issue. Moreover, this approach may face scalability issues when handling large-scale federated learning scenarios, and maintaining the inherent trade-off between privacy and utility is a challenging factor. Therefore, in the future, developing ML-based techniques with multi-objective optimization and advanced privacy-preserving approaches will provide enhanced privacy and increases the scalability and efficiency of the system.

REFERENCES


Explainable Artificial Intelligence (XAI) for the Prediction of Diabetes Management: An Ensemble Approach

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Department of Computer Science and Engineering, JIS University, 81, Nilgunj Road, Agarpara, Kolkata, West Bengal 700109, India²

Abstract—Machine learning determines patterns from data to expedite the process of decision making. Fact-based decisions and data-driven decisions are specified by the industry specialist. Due to the continuous growth of machine language models in healthcare, they are breeding continuous complexity and black boxes in ML models. To make the ML model crystal clear and authentically explainable, AI accession came in prevalence. This research scrutinizes the explainable AI and capabilities in the Indian healthcare system to detect diabetes. LIME and SHAP are two libraries and packages that are used to implement explainable AI. The intimated base amalgamates the local and global interpretable methods, which enhances the crystallinity of the complex model and obtains intuition into the equity from the complex model. Moreover, the obtained intuition could also boost clinical data scientists to plan a more felicitous composition of computer-aided diagnosis. Importance of XAI to forecast stubborn disease. In this case, of stubborn diabetes, the correlation between plasma versus insulin, age versus pregnancies, class (diabetic and nondiabetic) versus plasma glucose persisted with a strong relationship. The PIDD (PIMA Indian Diabetic Data set) with the SHAP value is used for concise dependency, and LIME is applicable when anchors and importance of features are both required simultaneously. Dependency plots help physicians visualize independent relationships with predicted disease. To identify dependencies of different attributes, a correlation heatmap is used. From an academic perspective, XAI is very indispensable to mature in the near future. To estimate the presentation of other applicable data set correspondence studies are very much apprenticed.

Keywords—Explainable Artificial Intelligence (XAI); diabetes; interpretability; machine learning; chronic disease management

I. INTRODUCTION

In developing countries such as India, one of the top ten causes of death related to the disease is diabetes. The researchers are working on the project to find the earliest way to detect the disease at a very early stage. Type 1 and type 2 are the two main types of diabetes. Type 1 diabetes causes a shortage of insulin, which affects increases in blood sugar levels. Whenever humans eat any food that is broken down into glucose, blood sugar in the bloodstream increases [1]. The pancreas cell emancipates insulin as the revolt of blood sugar levels, and as a result, it furnishes us to zeal for every day’s task [2]. If the cell ceases to produce insulin in the body, it may cause extravagant blood sugar to be sustained in the bloodstream. Significant health issues, such as blurred vision, kidney disease and heart disease, may be sustained as a result of the excessive presence of blood sugar. The symptoms of diabetes are delayed healing, itching, weaknesses, stiffness of muscle, polydipsia and blurred vision [3]. Millions of deaths are caused by diabetes, which is a metabolic condition throughout the year with various health issues. Approximately 84 million to 238 million diabetic cases will be diagnosed within 2030 around the world, which will impose a consequence load on the health care system [4, 5].

Glucose level changes are the main reason for diabetes. Maintaining a healthy lifestyle, a balanced diet and regular medical check-ups are the most preventive ways to restrict diabetes. Based on laboratory tests and disease symptoms, an intelligent system might be an important tool for diabetes prevention and detection.

Patient trust is the main dispute of an AI system. Without any explanation and without any reason to provide the output of the system is an opaque AI system. Especially in the health care system, the machine provides the output without any explanation, which makes it very difficult for the patient to believe the machine properly. To overcome this situation, explainable AI is rapidly used today to diagnose the disease fairly and correctly without any errors. Artificial intelligence adds a layer where the output can be defined clearly, which is known as explainable AI.

Interpretability as well as transparency can be escalated through XAI in the medical field. There is no legal right, which is reflected in the literature. XAI enhances the exposure of service by obliging end users to rely on the decision that AI makes correct decisions. The scope of AI is to make faster and correct decisions regarding the patients’ diagnosis and treatment and make it most trustworthy. The main goal of XAI is to generate a high-performance level explainable model. XAI will provide a detailed description of the AI technique, which helps to make AI algorithms more illuminate and translucent by depending on diagnosis with treatment protocols, drug research, medicine patterns, tests, etc.

Addressing XAI, the two important features are interpretability and explainability. The interpretability defines
how the output will be changed when the corresponding input parameters are changed. The term explainability defines a clear conception about a modern model, including certain assumptions made by the model and why.

This research aims to furnish interpretability of ML models and boost the performance of prediction to detect diabetes. The contributions of this work are as follows:

- Explainable AI now a day provides us with a “Black Box” nature, which releases more transparency and accuracy in the detection of disease and heightens the confidence of users.
- Shapley Additive Explanations (SHAP) and Local Interpretability Model-Agnostic Explainable (LIME) have been used widely in decision processes.
- A detailed analysis in the SHAP and LIME framework has been used to divulge significant perceptions to make a complete decision process considering all risk factors.

The remaining elements of the research discussion are organized as follows: briefs of similar papers are in Section II that use different categorizations of XAI technologies to forecast chronic disease. Section III describes the methodology used. The implementation of XAI techniques discussed in Section IV and in Section V describes the XAI upgrading of the health care system. The research effort is concluded in Section VI.

II. LITERATURE REVIEW

To diagnose diabetes, various types of machine learning algorithms have been used in [6,7-12]. In [9], PIDD missing values are replaced by mean values. A decision tree classifier is used for the classification. For the same data set, the researcher use different classification algorithms in [10], and the support vector machine provided the best result compared to other classification algorithms.

In another work with the PIDD data set [11], the models were developed using three different machine learning algorithms where SVM provided 70% accuracy with an accuracy of 80 of the trained data set. The same data set provides 78.9% accuracy using the XGBoost algorithm [8] by Tiwari and Singh.

The imbalanced classes and lack of preprocessing in the data set resulted in poor accuracy in these studies [13, 14, 15, 16]. In Fitriyaniet al. [17], these constraints are overcome.

In [18], the researcher developed a commission model amalgamation model to detect heart disease, which is an improved version of [6]. In [19], training set a split for detecting diabetes that scored high accuracy.

For several reasons [20], academic research findings and fruitful applications in medical practice have a remarkable distance. The most recent methodologies and techniques are not reliable by the physician [21, 22]. Black box methodology is not accepted by most medical practitioners due to its lack of explanation [23]. With the sacrifice of higher accuracy [24], clinical ML keeps away from complex models [25-27].

Due to the lack of explainability and biased accuracy, diabetes detection [28-30] and progression have become challenging. This research provides interpretability of the machine learning model and boosts the prediction performance using cross-validation, which helps to predict disease more reliably, accurately and effectively.

III. METHODOLOGY

Fig. 1 demonstrates the workflow of the overall model. PIMA Indian Diabetic Data set (PIDD), which contains eight attributes with values that the system has made use of. Then, the data are reprocessed to remove any missing or void values. Next, for training validation and testing, the data set was separated. For the implementation of separation, random sampling was used. As a result, the training and testing division will be unbalanced. To eliminate this problem, stratified sampling was used with a training size of 20% and a training validation size of 80%. After that, different machine learning models were put into action using the Scikit-Learn package. Next, the results were evaluated and explained with the LIME package and SHAP tool to make a complete decision process considering all risk factors. Table I describes the statistical description of the PIMA database.

![Workflow of the overall diabetes model](Fig. 1)
TABLE II. STATISTICAL DESCRIPTION OF DATABASE

<table>
<thead>
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<th>Attribute</th>
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<th>Mean</th>
<th>Std</th>
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<th>50%</th>
<th>75%</th>
<th>Max</th>
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<tbody>
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<td>3.3696</td>
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</tr>
<tr>
<td>Mass</td>
<td>768.00</td>
<td>31.9926</td>
<td>7.8842</td>
<td>0.0000</td>
<td>0.0000</td>
<td>32.0000</td>
<td>36.6000</td>
<td>67.1000</td>
</tr>
<tr>
<td>Pedi</td>
<td>768.00</td>
<td>0.4719</td>
<td>0.3313</td>
<td>0.0000</td>
<td>0.0000</td>
<td>0.3725</td>
<td>0.6260</td>
<td>2.4200</td>
</tr>
<tr>
<td>Age</td>
<td>768.00</td>
<td>33.24</td>
<td>11.7602</td>
<td>0.0000</td>
<td>21.0000</td>
<td>29.0000</td>
<td>41.0000</td>
<td>81.0000</td>
</tr>
<tr>
<td>Class</td>
<td>768.00</td>
<td>0.3490</td>
<td>0.4770</td>
<td>0.0000</td>
<td>0.0000</td>
<td>0.0000</td>
<td>0.0000</td>
<td>1.0000</td>
</tr>
</tbody>
</table>

Fig. 2. Correlation coefficient matrix of diabetes.

All diabetes feature correlation coefficient matrices are displayed in Fig. 2. The calculation is based on the linear relationship of measures and features; -1 to +1, the correlation coefficient value ranged. Values closer to 0 indicate a weak relationship, and values higher than 0 indicate a strong relationship. The correlations between plasma versus insulin, age versus pregnancy, and class (diabetic and nondiabetic) versus plasma glucose were strong.

A. Data Description

An auspicious disease is diabetes with its barrier. Mentioning the present health care system, it is very important to detect the disease properly. Table II explains the dataset description of proposed machine learning system. The present PIDD(https://www.kaggle.com/datasets/kumargh/pimaindians_diabetescsv) is a tabular data set containing 768 data points with two classes (diabetic and nondiabetic). In AI-based solutions in the area of healthcare, XAI is a censorious tool for predicting disease more accurately. The personalized description of XAI solves constraints such as the proper interpretations of model features, performances, explanations of the data set, and knowledge of the model with associations of trained data.

TABLE III. DATA SET DESCRIPTION OF PROPOSED MACHINE LEARNING SYSTEM

<table>
<thead>
<tr>
<th>Attributes Abbreviation</th>
<th>Attributes</th>
<th>Attributes Types</th>
</tr>
</thead>
<tbody>
<tr>
<td>Preg</td>
<td>Pregnancies</td>
<td>int 64</td>
</tr>
<tr>
<td>Plas</td>
<td>Plasma Glucose</td>
<td>int 64</td>
</tr>
<tr>
<td>Pres</td>
<td>Diastolic Blood Pressure</td>
<td>int 64</td>
</tr>
<tr>
<td>Skin</td>
<td>Triceps Skin Fold Tickness</td>
<td>int 64</td>
</tr>
<tr>
<td>Insu</td>
<td>Insulin(2Hrs serum insulin)</td>
<td>int 64</td>
</tr>
<tr>
<td>Mass</td>
<td>Body Mass Index</td>
<td>int 64</td>
</tr>
<tr>
<td>Pedi</td>
<td>Diabetic Pedigree Function</td>
<td>int 64</td>
</tr>
<tr>
<td>Age</td>
<td>Person’s Age</td>
<td>int 64</td>
</tr>
</tbody>
</table>

The characteristics of the XAI technique can be explained by LIME, SHAP and personal dependency plots with the diabetic data set in Fig. 3. The scatter plot in Fig. 4 represents the classifications of 768 patients into two classes, diabetes, which is class 1, and nondiabetes, which is class 0, according to the two important features plasma and insulin. It can be inferred that most of the patients belong to Class 0 nondiabetes. From this plot, low plasma and high insulin are less likely to be predicted as diabetes.

Fig. 3. Histogram of data set features.
IV. IMPLEMENTATION OF XAI TECHNIQUES

XAI process accomplishment helps researchers understand the system properly and expedites the process of diagnosing chronic disease properly with supportive results.

A. Platform and Language

XAI techniques can be easily implemented by using the Python programming language, which is an object-oriented, asynchronous interpretive high-level programming language with lower maintenance cost. ML with AI is very much in the Python language for the following reasons:

- Substantial libraries

  Actualization of AI algorithms with ML is very difficult to implement, cumbersome and time-consuming. Python libraries and frameworks are used in programming and reduce the complexity of the program, which makes it simpler and more flexible.

- Compact and Elementary

  Python acknowledges the programmers to develop a solution that is trustworthy instead of highlighting the practical.

B. SHAP (SHapley Additive Explanations)

In XAI techniques, SHAP is a very functional tool that can be used to evaluate the precise value of an attribute in forecasting. The SHAP values represent the difference between the actual forecasting of the result and the average forecasting of the model. The altar in the forecasting of the predicted model is represented by the SHAP values for each and every attribute when forcing the condition on a feature.

In Fig. 5, it is clearly observed that the plasma glucose variable has the highest influence in the model, followed by mass and age and Fig. 6, describes the SHAP values with all attributes with two classes.

Fig. 7 clearly depicts how individual attributes subscribe to the average or standard model forecasting on a global level. The $y$-axis on the right side represents whether the relevant value of the feature is low or high. The instances of data are represented by each dot sign.
VI. CONCLUSION

Importance of XAI is that XAI helps to predict the diagnosis of chronic disease. In the case of diabetic pregnancies, plasma glucose and age have been identified as the most important features based on the Pima Indian Diabetes Data set (PIDD), and the SHAP value is used for concise dependencies and visualizations of trends. LIME is applicable when anchors and importance of features are both required simultaneously. A correlation heatmap helps to identify dependencies of different attributes. Dependency plots help doctors visualize independent relationships with predicted classes.

The present work explains the utility of the XAI technique with only 768 limited clinical data. In future research wishing to work with more clinical data in different ways, XAI may be used for automatic diagnosis.

From an academic perspective, XAI works are far from the authentic medical framework, which is essential for development in the near future. Correspondence studies are necessary to estimate the presentation of recommended proposals in other applicable data sets. To a greater extent, refinement in the accomplishment of the model and explainability will endeavor using various algorithms in machine learning to furnish different types of accumulation models.

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Research on the Text Classification of Legal Consultation Based on Deep Learning

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Abstract—In view of the existing traditional legal service, practitioners are unable to meet the huge demand; a large number of citizens are unable to determine the scope of the problems when they encounter infringement or require various legal assistance. Based on this, an automatic classification model of legal consultation based on Deep Learning is proposed in this paper. A KP+BiLSTM+Attention model is proposed. TheKeyword Parser is introduced to extract key information. TF-IDF and part of speech tagging are used to filter out the important information in the user's legal problem description. The extracted keywords are given a weight value, and the other information weights are set to zero. The text information is transferred into two parallel word vector embedding layers. One of the word vector embedding layers transfers the results to the fusion layer for splicing, difference and point multiplication after the key information is converted into vector form. The output results are respectively connected with the results obtained from the other embedding layer as residuals. The final results are transferred to the BiLSTM+Attention model for training. The test results show that KP+BiLSTM+Attention model has significantly improved the accuracy and F1 value of the benchmark method for text classification tasks of legal consulting. Therefore, KP+BiLSTM+Attention method has better performance in dealing with the classification of legal consulting issues.

Keywords—Text classification; legal consultation; deep learning; KP+BiLSTM+ATT model; word embedding layer

I. INTRODUCTION

Legal consultation is to determine the scope of the problem encountered by the public and to provide legal solutions applicable to the relevant field. How to put forward effective opinions and suggestions quickly and effectively for their own problems is the focus of people's attention, and it is also an important issue facing the popularization of legal awareness [1][2][3]. In view of this problem, the demand of users is classified correctly for lawyer advice is the first task. Because it can complete automatically the classification and recognition of text information [4][5][6], Deep Learning (DL) is widely used in the field of Natural Language Processing (NLP) [7][8][9][10]. The problem category is determined accurately after adaptive training according to relevant information of legal provisions and customer problem descriptions [11][12][13].

Text classification is a basic work in the field of NLP, which aims at classifying text information [14][15][16][17]. Salton [18] proposed the word vector space model. However, this model needed to define a large number of rules for each category which were highly dependent on the professionals. The Q-AND-A method based on knowledge graph had achieved good results in the fields of public security information analysis, medical consultation and drug prescribing [19]. Zhang [20] used the fine-granularity question and answer model based on Bi-LSTM+CRF to select entities. Li [21] used the entity recognition model based on Bi-LSTM+CNN+CRF to establish a human-computer interactive question and answer system to improve the employment rate. In the legal field, a question and answer system was proposed based on the legal field by Huang [22], which introduced a small number of samples and a transfer learning model. Zhang et al. [23] introduced DL into the legal judgment task, and a better effect was achieved in the open data set of the real legal judgment prediction task.

In this paper, the original word segmentation algorithm TF-IDF is improved, and combined with the DL model. The brand new Keyword Parser (KP) is proposed based on common keyword extraction approach. Firstly the Chinese word segmentations are calculated based on TF-IDF, and the weights assigned according to the degree of importance. Then, it enhances the extracted key information according to part-of-speech screening, and the important words in each piece of data can be further screened and retained. The retained information is passed to two embedding layers at the same time. The fusion layer embedded in one embedding layer will process the incoming data by splicing and dot multiplication, and the results will be passed to the other Embedding one as the residual connection to improve the credibility of the information. Finally, the classification results were obtained by training Bi-LSTM+Attention model. Based on the legal consultation database, comparing common Machine Learning (ML) text classification models and DL text classification models, the experimental results show that KP-BiLSTM+Att model performs better on the legal consulting data set.

II. STRUCTURE OF KP-BILSTM-ATT MODEL

A. Chinese Segmentation

Unlike English words, which are naturally separated by spaces, Chinese takes words as the basic unit and there is no obvious separation mark between words [24]. The main task of Chinese word segmentation [25] is to divide a complete sentence into individual words. There are three main types of mainstream Chinese word segmentation methods: word segmentation based on string matching, word segmentation...
based on understanding and word segmentation based on statistics [26]. In this paper, jieba segmentation is used as a word segmentation tool, and the sentence is cut into data information $X$ using the built-in precise mode.

B. Structure of Legal Consultation Text Classification Approach Based on KP-BiLSTM-Att

The overall structure of the legal consultation text classification approach based on KP-BiLSTM-Att proposed in Fig. 1.

![Fig. 1. Structure of the legal consultation text classification based on KP-BiLSTM-Att.](image)

The text information $X$ which has not been processed and the information $X'$ which has been processed by keyword processors are passed into the two embedding layers for vectorization operations. A fusion layer is added to the embedding layer that is responsible for dealing with $X'$ to carry out vector operations on the vectorization results of $X$ and $X'$, and then the results are connected with $X$ as residuals. The vector results are passed into the BiLSTM+Attention model for training. The training results were used by linear layer and softmax function to predict the scope of legal consulting problems. Finally, Focal Loss is introduced to solve the unbalance problem of all kinds of samples in the dataset.

III. MODEL TRAINING STAGE

There are four modules during model training stage: Keyword Parse, two parallel embedding layers, Bi-LSTM layer and attention layer.

The main task of the KP is to receive the pre-processing results, and obtain the keyword information of the problem description by TF-IDF algorithm and [mask] label. Two parallel word embedding layers receive the original word segmentation result $X$ and the keyword information $X'$ after processing by the KP respectively. Meanwhile, a fusion layer is added the word embedding layer which is receiving $X'$ to calculate the output vector information of two embedding layers. The final calculation results are passed into Bi-LSTM layer and attention layer for training.

A. Keyword Parser

In this module, the text information obtained after data preprocessing is received by the parser, and the weight value of each word in the word segmentation result about the category is calculated based on TF-IDF. The importance of each word for the category is obtained. Then, with the idea of part-of-speech tagging [27], all words are labeled with a correct part of speech, that is, each word is a noun, verb, adjective or other part. The part-of-speech tagging method based on statistics is adopted in the process of part-of-speech tagging [28].

The parser retains nouns, verbs, and adjectives, and it masks all the results of word segmentation other than the above parts of speech with the [mask] tag to get $X'$. The purpose of masking other parts of speech words is to keep the words that are helpful to the classification accuracy (that is, the words with the high TF-IDF values) and keep their original positions unchanged, so as to avoid the dislocation phenomenon in the subsequent weight addition, and improve the efficiency and accuracy of model training. The operations performed by the KP are shown in Fig. 2.

![Fig. 2. Operations performed by KP.](image)
the dimension of the information through the transformation of high-dimensional image and low-dimensional. The other word embedding layer converts $X'$ into vector form after random initialization, and passes the results into the fusion layer \[29\] for concatenation, difference and dot multiplication operations with the vectorization result of $X$. The obtained result and the $X$ result are respectively connected by residual to improve the information weight value.

There are a large number of [mask] labels in $X'$, which will be directly set to 0 in the vector during the operation. The difference between keywords and non-keywords has been increased by the operations above, which is conducive to the subsequent model training. The structure of the word embedding layer is shown in Fig. 3.

### C. Bi-LSTM+Attention Layer

Bi-LSTM is an improved model based on LSTM. In fact, two LSTM process the sequence from the forward direction and the reverse direction respectively, and the results of the two directions are combined to obtain a new vectorized result which will pass to the attention layer to assign different attention values \[34\]. Finally, the loss function in the fully connected layer is used to calculate the category. The structure of BiLSTM+Attention layer is shown in Fig. 4.

![Fig. 3. Structure of the embedding layer.](image)

![Fig. 4. Structure of BiLSTM+Attention layer.](image)

Bidirectional LSTM is the LSTM Neural Network structure that combines the vector information obtained in the forward direction and the reverse direction on the basis of the LSTM processing sequence information in the reverse direction, and computes the new vector results. Fig. 5 shows the structure of bidirectional LSTM \[35\].

![Fig. 5. Structure of bidirectional LSTM.](image)

The new vector obtained by bidirectional LSTM is passed into the attention layer to calculate the attention weight of each word vector through the self-attention mechanism. The self-attention mechanism not only helps the current node focus on the current word, but also obtains contextual semantic information.

Self-attention will calculate three new vectors: Query, Key, and Value, which are obtained by multiplying the word embedding vector and the random initialization matrix. Then, Query and Key will be dot-multiplied to get the weight, which is the score of self-attention. When we encode a word, the score of the self-attention determines how much attention is paid to the input sequence:

$$ f(Q,K_i) = Q^T K_i $$  \hspace{1cm} (1)

The correlation size of each word for the current position will get by a softmax calculation:

$$ \alpha_i = \text{softmax}(f(Q,K_i)) = \frac{\exp(f(Q,K_i))}{\sum_j f(Q,K_j)} $$  \hspace{1cm} (2)

Value and softmax are multiplied and added together to obtain the attention value of the current node. The units in each sequence are made attention calculating with all the units in the sequence to obtain the self-attention value.

$$ \text{Attention}(Q,K,V) = \text{softmax}(\frac{QK^T}{\sqrt{d_q}})V $$  \hspace{1cm} (3)

where $QK^T$ is the attention matrix, and $\sqrt{d_q}$ is the conversion from the attention matrix to a standard normal distribution. Multi-Head Attention mechanism is to carry out multiple attention operations, so that the model has multiple attention values with the same structure but different weights.

$$ \text{head}_i = A(Q,K,V) $$  \hspace{1cm} (4)

$$ \text{MultiHead}(Q,K,V) = \text{Concat}\{\text{head}_1,\text{head}_2,\ldots,\text{head}_m\}W^o $$  \hspace{1cm} (5)
The vector obtained after vector operation by bidirectional LSTM Neural Network and self-attention mechanism can remember the semantic information of context well, and the accuracy of classification results is higher.

D. Focal Loss Based on Class Balance

Due to the unbalanced distribution of legal problems, Focal Loss [30] is introduced as an effective method to solve the problem of unbalanced quantity of different samples in data sets. Focal Loss can reduce the weight of easily classified samples to make the model pay more attention to difficult classified samples in training. The traditional cross entropy loss should be improved by this approach.

As an example of taking binary classification, the loss function of cross entropy is shown as (6).

\[ CE(p, y) = \begin{cases} -\log(p) & \text{if } y=1 \\ -\log(1-p) & \text{otherwise} \end{cases} \] (6)

where \( p \in [0,1] \) is the classification probability. \( p \) is the probability that the sample belongs to the true class.

\[ p = \begin{cases} p & \text{if } y=1 \\ 1-p & \text{otherwise} \end{cases} \] (7)

The size of \( p \) can reflect the degree of difficulty of sample classification. In the training process, the model should pay more attention to hard-to-classify samples, so the proportion of these kinds of samples should be increased.

Focal Loss adds an item to the binary classification cross-entropy loss function to attenuate the original cross-entropy loss, which reduces the weight of easily classified samples to focus on the training of difficult samples. At the same time, in order to solve the imbalance of positive and negative samples, a weight factor \( \alpha \in [0,1] \) will be added to each category of the loss function to coordinate the class imbalance. The improved Focal Loss function formula is shown in (8).

\[ FL(p_i) = -\alpha(1-p_i)^\gamma \log(p_i) \] (8)

where \( \gamma \geq 0 \) is an adjustable focusing parameter, and the larger the value of \( \gamma \), the smaller the loss of easily classified samples. A larger \( \gamma \) will expand the samples range which have small loss. When \( \gamma > 1 \), Focal Loss can reduce the loss of easy to classify samples, but not much for difficult to classify samples. When \( \gamma = 0 \), the Focal Loss formula becomes a cross-entropy loss function.

E. FGM and R-Drop Based on Confrontation Training

1) FGM: Because of the linear characteristics of Neural Network, it is easily attacked by linear disturbance. The concept of Adversarial training [31] is proposed to improve the robustness of the model. Adversarial training is to add a disturbance \( r_{adv} \) to the original sample \( x \), and then train the model with the adversarial sample. The disturbance is increasing in the direction of increasing the loss.

The perturbation definition formula is shown as (9).

\[ r_{adv} = \epsilon \cdot \text{sgn}(\nabla_{\theta} L(x, y)) \] (9)

where \( \text{sgn}() \) is the sign function and \( L \) is the loss function. If the input sample is further moved in the direction of rising loss, the resulting adversarial sample can cause greater loss and improve the error rate of the model, so as to meet the requirement that adding small disturbance to the adversarial sample can make the model judgment wrong.

Madry [32] redefined the problem as a saddle point finding problem from the perspective of optimization:

\[ \min_{s} E(x, y) \sim D(\text{max}_{\epsilon \in S} L(\theta, x + r_{adv}, y)) \] (10)

where \( S \) is the range of perturbations, \( \theta \) is the internal parameter of the model, \( D \) is the distribution of input samples, and \( x \) and \( y \) correspond to the input and output respectively. There are two parts of this formula: the internal loss function maximization and the external empirical risk minimization. The internal maximization is to obtain the disturbance parameter in the most serious case of model misjudgment, and external risk minimization is to find the parameters that make the model the best robust for internal attacks. The disturbance increased by FGM is:

\[ r_{adv} = \epsilon \cdot \frac{g}{\|g\|_2} \] (11)

\[ g = \nabla_\theta L(\theta, x, y) \] (12)

The new adversarial sample is:

\[ x_{adv} = x + r_{adv} \] (13)

2) R-drop: Another way to improve model robustness and generalization is R-drop. Although the traditional Dropout is often used to regulate the training of Deep Neural Networks, its practice of randomly dropping some neurons lacks explanation, resulting in inconsistencies between training and reasoning.

R-drop construes the output consistency of the random submodel due to Dropout through KL divergence. The formula for the R-drop method to apply regular constraints on the output prediction is:

\[ L_{kl} = \frac{1}{2}(D_{KL}(P(y_1 | x_i) \| P_2(y_1 | x_i)) + D_{KL}(P(y_2 | x_i) \| P_2(y_2 | x_i))) \] (14)

Since Dropout randomly drops a few neurons at one time, both probability values of \( P(y_1 | x_i) \) and \( P(y_2 | x_i) \) are predicted probabilities derived from different submodels of the same model. The difference between these two prediction probabilities is constrained with uses the symmetric KL divergence.

\[ L_{nll} = -\log P_1(y_i | x_i) - \log P_2(y_i | x_i) \] (15)

The final loss function is:

\[ L_{nll} = L_{nll} + \alpha \cdot L_{kl} \] (16)
where $\alpha$ is the coefficient used for control $L_{\text{KL}}$. Compared with the traditional Dropout, a regular constraint term is introduced to improve the robustness of the model and reduce the inconsistency of the model output.

### F. GHM Loss and Dice Loss

Although Focal Loss introduced is considered to be more suitable as a loss function to solve the problem of unbalanced sample data sets, the experimental results of the task in this paper are slightly short of the cross-entropy loss function, which may be due to the phenomenon of category imbalance in the data set. However, in the classification process, it is difficult to classify many samples, which leads to a large loss value of Focal Loss, and the small number of samples in individual categories led to the failure of Focal Loss to give full play to its performance.

GHM Loss is an improved approach based on Focal Loss. Focal Loss is mainly the unbalanced distribution of difficult and easy samples [33], while GHM Loss believes that not all difficult samples are worthy of attention. It adopts a gradient harmonic mechanism and abandons some outliers by formula calculation.

A new gradient norm is introduced by binary cross entropy loss function.

\[
L_{\text{CE}}(p, p^*) = \begin{cases} 
-\log(p) & \text{if } p^* = 1 \\
-\log(1-p) & \text{if } p^* = 0 
\end{cases}
\]

where $p = \text{sigmoid}(x)$ is the category probability of the model prediction sample and $p^*$ is the label information. The gradient with respect to $x$ can be calculated by the formula (18).

\[
\frac{\partial L_{\text{CE}}}{\partial x} = \begin{bmatrix} p - 1 \\
p 
\end{bmatrix}
\]

where $p^* = 1$ if $p^* = 0 = p - p^*$.

The gradient norm is defined as:

\[
g = |p - p^*| = \begin{cases} 
1 - p & \text{if } p^* = 1 \\
p & \text{if } p^* = 0 
\end{cases}
\]

Intuitively, $g$ represents the distance between the real values and the predicted ones of the sample. The gradient mode norm distribution after convergence of the binary classification model is shown in Fig. 6.

\[
GD(g) = \frac{1}{i_{\text{med}}(g)} \sum_{i=1}^{K} \delta_i(g_i, g) \tag{20}
\]

where $g_i$ represents the gradient of the $k$ sample, and:

\[
\delta_i(x, y) = \begin{cases} 
1 & \text{if } y - \frac{\varepsilon}{2} \leq x \leq y + \frac{\varepsilon}{2} \\
0 & \text{otherwise} \tag{21}
\end{cases}
\]

\[
l_i(g) = \min(g + \frac{\varepsilon}{2}, 1) - \max(g - \frac{\varepsilon}{2}, 0) \tag{22}
\]

Define the density coordination parameter $\beta$:

\[
\beta_i = \frac{N}{GD(g_i)} \tag{23}
\]

where $N$ represents the number of samples. It can ensure that the weight value is 1 when the distribution is uniform or only one unit area is divided, that is, the loss is unchanged. It can be seen that the weights of samples with high gradient density will decrease. The definition of loss function obtained by applying GHM idea to classification problem is shown as (24).

\[
L_{\text{CE}}^\text{GHM} = \frac{1}{N} \sum_{i=1}^{K} \beta_i L_{\text{CE}}(p, p^*) = \frac{1}{N} \sum_{i=1}^{K} L_{\text{CE}}(p, p^*) \tag{24}
\]

According to (24), the weights of the easily classified negative samples and the extremely difficult classified samples in the candidate samples will be reduced, and the loss value will also be reduced and the impact on model training will be reduced. The weight of the normal hard-to-classify samples will be increased, so that the model will pay more attention to the normal hard-to-classify samples to improve the model performance.

However, GHM Loss is more used in binary classification problem of target detection, and there may be errors in judging extremely difficult samples in multiple classification tasks. At the same time, there might exist a problem that too many hard-to-classify samples are discarded as outliers; will lead to inadequate training. Therefore, the training effect of the model is still inferior to that of the cross-entropy Loss function, and even the experimental effect is lower than that of Focal Loss due to the operation of directly discarding outliers.
Dice Loss, inspired by Dice Coefficient (DSC), was first proposed to apply loss function in the field of image segmentation [33] to solve the problem of pixel sample imbalance in image segmentation.

In this paper, the loss function is transferred to the text classification task to verify whether it can be applied to the field of NLP through experiments. The DSC coefficient is used to evaluate the similarity of two sets.

\[
DSC(A, B) = \frac{|A \cap B|}{|A| + |B|} \tag{25}
\]

where \(A\) represents the set predicted by the model as a positive category, and \(B\) represents the set with a true label as a positive class. Dice Loss directly optimizes F1-score:

\[
DSC = \frac{2TP}{2TP + FN + FP} = \frac{2 \overline{Pr} \times \overline{Rec}}{\overline{Pr} + \overline{Rec}} = F1 \tag{26}
\]

where \(TP, FP, FN\) are discrete values, and the DSC coefficient is based on discrete values. For a sample, the continuous form of DSC coefficient can be defined as follows:

\[
DSC(x) = \frac{2p_iy_i}{p_i + y_i} \tag{27}
\]

where \(y_i\) represents the label value of a positive sample, and \(p_i\) represents the prediction probability of a positive model sample. If the sample is positive, the higher the prediction probability is, the higher the DSC value is. When the sample is a negative class and the molecule is 0. In order to preserve the value of the DSC coefficient of the negative class, the smoothing term \(\gamma\) is added.

\[
DSC(x) = \frac{2p_iy_i + \gamma}{p_i + y_i + \gamma} \tag{28}
\]

The greater the DSC value is, the more accurate the prediction is, and we convert the formula to a loss.

\[
DSC(x) = 1 - \frac{2p_iy_i + \gamma}{p_i + y_i + \gamma} \tag{29}
\]

Since DSC coefficient is a metric function to measure the similarity of two samples, the larger the positive sample \(p\) is, the larger the DSC value is, indicating that the more accurate the model prediction is, the smaller the loss should be at this time. Therefore, the final form of Dice Loss is shown as (30).

\[
DL = 1 - \frac{2p_iy_i + \gamma}{p_i + y_i} \tag{30}
\]

Dice Loss has a good performance for the scene where the positive and negative samples are seriously unbalanced, and pays more attention to the prediction of the foreground region in the training process. But the value of training losses is highly volatile.

The reason Dice Loss can solve the problem of sample imbalance is mainly because it is a loss related area, that is, the loss of the current pixel is related not only to the predicted value of the current pixel, but also to other points. The intersection form of Dice Loss can be understood as the operation of mask.

Therefore, no matter how large the picture is, the loss value calculated for a positive sample area of fixed size is the same, and the supervision contribution to the network will not change with the picture size.

Experimental results show that Dice Loss is not as good as cross entropy loss function, which may be due to the fact that the loss value calculated by the loss function is not stable enough. When the loss function is used in the case of only foreground and background, the loss value of a small target will change drastically when there is a partial prediction error, resulting in a drastic change in gradient. For example, considering that there is only one positive sample in the extreme case, as long as the prediction is correct, the loss value will be close to 0, while the loss of the prediction error will be close to 1. The cross entropy loss function is equal in dealing with positive and negative samples and averaging the population, so it is more stable than Dice Loss.

In addition, the above three loss functions are first proposed in the field of image processing, and the difference between image information and text information is also the reason for the unsatisfactory results. Therefore, through experiments, it is believed that the cross-entropy loss function combined with the model can achieve the best results of the experiment in this paper.

IV. EXAMPLE ANALYSIS

A. Subjects

The subjects of our experiments come from the open-source data set on github website. There are a total of 200,000 legal Q&A pairs of 13 types of questions, which are put forward and recorded by customers who need legal consulting services. We mainly conduct classification experiments through two aspects of problem description and problem classification. After deleting some categories with fewer cases in the data set, a total of 190,000 data are retained in the experiment, corresponding to 11 different categories.

The names of 11 categories and their distribution in the training set and test one are shown in Table I. There are a total of 152,000 descriptions of legal issues. The average length is 23.918, among which the descriptions of legal problems with text length of 28 characters is the most, the descriptions with text length of 32 characters account for 81.377%, the descriptions with 50 characters account for 98.204%, and the ones with 150 characters account for 99.926%.

There are 19,000 descriptions of legal problems in the test set, with an average length of 23.723, among which the descriptions of legal problems with text length of 28 characters are the most, the descriptions with 32 characters account for 81.311%, the descriptions with 50 characters account for 98.437% and the ones within 150 characters account for 99.921%. It can be seen that most legal advice texts are within 32 characters.
TABLE I. DISTRIBUTION OF DATA SET SAMPLES

<table>
<thead>
<tr>
<th>Category</th>
<th>Training set</th>
<th>Test set</th>
<th>Validation set</th>
</tr>
</thead>
<tbody>
<tr>
<td>Real estate disputes</td>
<td>9272</td>
<td>1159</td>
<td>1159</td>
</tr>
<tr>
<td>Traffic accidents</td>
<td>17504</td>
<td>2188</td>
<td>2188</td>
</tr>
<tr>
<td>Creditor’s rights and debts</td>
<td>16888</td>
<td>2111</td>
<td>2111</td>
</tr>
<tr>
<td>Tort</td>
<td>8176</td>
<td>1022</td>
<td>1022</td>
</tr>
<tr>
<td>Marriage and family</td>
<td>29624</td>
<td>3703</td>
<td>3703</td>
</tr>
<tr>
<td>Company law</td>
<td>7760</td>
<td>970</td>
<td>970</td>
</tr>
<tr>
<td>Contract disputes</td>
<td>10608</td>
<td>1326</td>
<td>1326</td>
</tr>
<tr>
<td>Demolition and Resettlement</td>
<td>5432</td>
<td>679</td>
<td>679</td>
</tr>
<tr>
<td>Medical disputes Labor disputes</td>
<td>5632</td>
<td>704</td>
<td>704</td>
</tr>
<tr>
<td>Criminal defense</td>
<td>27072</td>
<td>3384</td>
<td>3384</td>
</tr>
<tr>
<td>Real estate disputes</td>
<td>14032</td>
<td>1754</td>
<td>1754</td>
</tr>
</tbody>
</table>

B. Evaluation Index

In this paper, the classification task of legal consulting texts is modeled as a multi-classification problem. In order to compare the performance of the KP-BiLSTM-Att model method proposed and the baseline model approaches, a unified measurement standard is needed. Therefore, we select the evaluation indexes commonly used in Machine Learning to deal with text classification: Accuracy, Precision, Recall and F1-score.

Accuracy $A(i)$ is the proportion of the number of correctly predicted samples to the total number of samples in the dataset:

$$A(i) = \frac{TP + TN_i}{TP + TN_i + FP_i + FN_i} \quad (31)$$

The accuracy rate $P(i)$ is the proportion of correctly predicted positive samples to the predicted positive samples in the dataset.

$$P(i) = \frac{TP_i}{TP_i + FP_i} \quad (32)$$

The recall rate $R(i)$ is the proportion of the sample in which the positive sample was correctly predicted.

$$R(i) = \frac{TP_i}{TP_i + FN_i} \quad (33)$$

The $F1(i)$ value is the weighted harmonic average of the $P(i)$ and $R(i)$.

$$F1(i) = \frac{2 \times \text{Recall}(i) \times \text{Precision}(i)}{\text{Recall}(i) + \text{Precision}(i)} \quad (34)$$

C. Analysis of Experimental Results

From four traditional ML methods, we choose Term Frequency–Inverse Document Frequency (TF-IDF), Naive Bayes model (NB), Support Vector Machine (SVM) and Random Forest (RF) as the baseline methods, and three other models of TextCNN, TextRCNN and Transformer models from Deep Neural Networks.

For the implementation of the benchmark approaches, the classification method based on ML is repeated through sklearn library and XGBoost library. In order to make a fair comparison of their performances, the same word segmentation methods were used to divide the data set. The results of various indicators were shown in Table II, Table III, Table IV and Table V respectively.

Table II shows the comparison of the Accuracy results of the KP-BiLSTM-Att approach proposed in this paper with other benchmark methods on the test set. According to the experimental results, the Accuracy of KP-BiLSTM-Att is higher than others in terms. Ours test result is improved by 4.3% of the performance of TF-IDF algorithm, which has the best performance among Machine Learning text classification algorithms. Ours is improved by 1.2% of TextRCNN which has the best performance among DL text classification algorithms.

In addition, we also tried to use TextRNN model in the experiment, but because the problem description content in the experiment was too long, TextRNN could not train long-distance text well, resulting in low accuracy. It also confirmed that the effect of BiLSTM Neural Network is better than the commonly used TextCNN and TextRNN Neural Network structure. According to the Accuracy index alone, KP-BiLSTM-Att approach performs better than other benchmark methods in the task of legal consulting text classification.

Table III, Table IV, and Table V show the Precision values, Recall values, and F1-score values of KP-BiLSTM-Att approach and other baseline methods in each category.

Since there is a pair of contradictory indicators for Accuracy rate and Recall rate, it is difficult to reach high values at the same time. Therefore F1-score is finally selected as the model performance evaluation index.

TABLE II. EVALUATION RESULTS BASED ON ACCURACY INDEX

<table>
<thead>
<tr>
<th>Approaches</th>
<th>TF-IDF</th>
<th>NB</th>
<th>SVM</th>
<th>RF</th>
<th>Text CNN</th>
<th>Text RCNN</th>
<th>Transformer</th>
<th>Ours</th>
</tr>
</thead>
<tbody>
<tr>
<td>Accuracy</td>
<td>0.755</td>
<td>0.682</td>
<td>0.721</td>
<td>0.751</td>
<td>0.768</td>
<td>0.787</td>
<td>0.783</td>
<td>0.798</td>
</tr>
</tbody>
</table>
According to the experimental results of F1-score, KP-BiLSTM-Att approach proposed in this paper reaches the optimal values in 8 out of 11 categories. In the category of traffic accidents, KP-BiLSTM-Att has increased by 0.8% compared with the baseline model with the best performance. In the category of bond debts, ours has increased by 0.8% compared with the best one; and in the category of marriage and family, this method has increased by 1.4% compared with the best one. In the category of company law, KP-BiLSTM-Att has increased by 3.4% compared with the baseline model with the best performance. Compared with the best-performing baseline model in the contract disputes category, KP-BiLSTM-Att has increased by 2.3%, and compared with the best-performing model in the relocation and resettlement category, ours has increased by 0.3%. In the category of medical dispute, KP-BiLSTM-Att has increased by 13.1%, and it has increased by 0.3% in the category of labor dispute.

In the other three categories that do not reach the optimal value, KP-BiLSTM-Att approach is also stronger than most of the baseline methods, and the weakness stems from an imbalance in the number of samples in different categories. It can be considered that combined with the comprehensive evaluation of Accuracy and F1-score, the superiority of KP-BiLSTM-Att method in the classification task of legal consulting texts is preliminarily confirmed.

Then, weighted average processing is performed for all indicators of the proposed method, and the results are shown in Table VI.
The BiLSTM+Attention model has explored into the BiLSTM structure of deep learning to accelerate robustness of the model. The R-Att model is proposed and simultaneously all the parts of Text classification method are weighted by TF-IDF, and the result is defined as Text RCNN. In this paper, KP+BiLSTM+Att method is compared with seven popular machine learning and deep learning methods. Experimental results show that the Accuracy index of KP+BiLSTM+Att method is 4.7%, 11.6%, 7.7% and 4.3% higher than that of TF-IDF, NB, SVM and RF algorithm in traditional ML classification methods, respectively. Compared to common DL baseline models, the KP+BiLSTM+Att approach offers a 3.0%, 1.1%, and 1.5% improvement over TextCNN, TextRCNN, and Transformer respectively. The test results show that KP+BiLSTM+Attention model has significantly improved the accuracy and F1 value of the best benchmark method for text classification tasks of legal consulting. It can be seen that KP+BiLSTM-Att method has better performance in the classification of legal consulting texts.

There is still some follow-up work for improvement in this research. 1) Different data sets of legal consulting texts and other commonly used data sets in the field of text classification will be presented to verify the feasibility of this model. At the same time, it is considered whether the model can be applied to other related tasks in the legal field through experiment to test whether better results can be obtained. 2) More advanced deep learning models will be used to improve the overall structure and further improve the classification accuracy and operation efficiency of the model. 3) Other methods will be explored to improve the performance of unbalanced data samples and improve the overall classification accuracy of the model.

**ACKNOWLEDGMENT**

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**TABLE V. EVALUATION RESULTS BASED ON F1-SCORE INDEX**

<table>
<thead>
<tr>
<th>Category</th>
<th>Approaches</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>TF-IDF</td>
</tr>
<tr>
<td>Real estate disputes</td>
<td>0.864</td>
</tr>
<tr>
<td>Traffic accidents</td>
<td>0.484</td>
</tr>
<tr>
<td>Creditor's rights and debts</td>
<td>0.760</td>
</tr>
<tr>
<td>Tort</td>
<td>0.441</td>
</tr>
<tr>
<td>Marriage and family</td>
<td>0.700</td>
</tr>
<tr>
<td>Company law</td>
<td>0.817</td>
</tr>
<tr>
<td>Contract disputes</td>
<td>0.526</td>
</tr>
<tr>
<td>Demolition and Resettlement</td>
<td>0.616</td>
</tr>
<tr>
<td>Medical disputes Labor disputes</td>
<td>0.672</td>
</tr>
<tr>
<td>Criminal defense</td>
<td>0.615</td>
</tr>
<tr>
<td>Real estate disputes</td>
<td>0.695</td>
</tr>
</tbody>
</table>

**TABLE VI. EVALUATION INDEX BASED ON WEIGHTED AVERAGE**

<table>
<thead>
<tr>
<th>Approaches</th>
<th>Evaluation Index</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>Accuracy</td>
</tr>
<tr>
<td>TF-IDF</td>
<td>0.755</td>
</tr>
<tr>
<td>NB</td>
<td>0.682</td>
</tr>
<tr>
<td>SVM</td>
<td>0.751</td>
</tr>
<tr>
<td>RF</td>
<td>0.721</td>
</tr>
<tr>
<td>TextCNN</td>
<td>0.768</td>
</tr>
<tr>
<td>TextRCNN</td>
<td>0.787</td>
</tr>
<tr>
<td>Transformer</td>
<td>0.783</td>
</tr>
<tr>
<td>Ours</td>
<td>0.798</td>
</tr>
</tbody>
</table>

Combined with the weighted average evaluation indicators, all evaluation indicators of KP-BiLSTM-Att have reached the optimal values, among which Accuracy index has increased by 1.1%. Precision index by 0.1%, Recall index by 1.5%, and F1-score index by 1.2%. The experimental results prove that KP-BiLSTM-Att approach can achieve better classification performance than the traditional ML text classification method and the common reference method of DL text classification Neural Network model.

**V. CONCLUSIONS**

In this paper, KP-BiLSTM-Att model is proposed and applied to the task of classifying legal consulting texts. The preprocessed data set text is passed into the Keywords Parser, and all the results of word segmentation are weighted by TF-IDF and the result is defined as X. Meanwhile, all the parts of speech of the results of word segmentation are marked, processed by [mask], and the result is output as X'. X and X' are passed into the two word embedding layers respectively, and the adversarial training FGM algorithm is added to add disturbance to the vector quantization results to improve the robustness of the model. The R-drop method was introduced into the BiLSTM structure of deep learning to accelerate model training and improve model generalization ability. Then, the Focal Loss function method based on class balance was introduced to solve the unbalanced data set samples in multiple classification tasks and improve the classification accuracy.

The performance of KP-BiLSTM-Att method is compared with seven popular machine learning and deep learning methods. Experimental results show that the Accuracy index of KP-BiLSTM-Att method is 4.7%, 11.6%, 7.7% and 4.3% higher than that of TF-IDF, NB, SVM and RF algorithm in traditional ML classification methods, respectively. Compared to common DL baseline models, the KP-BiLSTM-Att approach offers a 3.0%, 1.1%, and 1.5% improvement over TextCNN, TextRCNN, and Transformer respectively. The test results show that KP+BiLSTM+Attention model has significantly improved the accuracy and F1 value of the best benchmark method for text classification tasks of legal consulting. It can be seen that KP-BiLSTM-Att method has better performance in the classification of legal consulting texts.
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Continuous Software Engineering for Augmented Reality

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Bina Nusantara University, Jakarta, Indonesia, 11480¹,²,³
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Abstract—Continuous software engineering is a new trend that has attracted increasing attention from the research community in recent years. In software engineering there are “continuous” stages that are used depending on the number of artifact repositories such as databases, meta data, virtual machines, networks and servers, various logs, and reports. Augmented Reality (AR) technology is currently growing rapidly. We can find this technology in various fields of life, but unfortunately sustainable software engineering for Augmented Reality is not found. The method shown in previous research is a general method in software engineering so that a theory is needed for sustainable software engineering for AR considering that AR is not just an ordinary application but there are 3D elements and specific components that must be met so that it can be called AR. The main idea behind this research is to find a continuous pattern from the stages of the existing method so far. For example, in general the stages of system development are planning, analysis, design, implementation and maintenance. Then after the application has been built, does it finish there? As we know software always grows and develops according to human needs. Therefore, there are continuous stages that must be patterned so that the life cycle process can be maintained. In this paper we present our initial findings about the continuous stages of continuous software engineering namely continuous planning, continuous analysis, continuous design, continuous programming, continuous integration, and continuous maintenance.

Keywords—Continuous software engineering; augmented reality; method in software engineering; continuous planning; continuous analysis; continuous design; continuous programming; continuous integration; continuous maintenance

I. INTRODUCTION

Continuous software engineering is performed to avoid discontinuities between development and deployment to continuously perform continuous development [1]. Continuously means without interruption, so the application delivery processing time can be maintained [2]. In continuous software engineering, software developers can flexibly implement several changes from coding, testing and deployment so that if there are problems such as detecting bugs, requesting changes and other unexpected actions will be resolved more quickly. In addition, developers can also test more often and implement some changes. Instead of creating new code, developers can leverage code from an already developed version of the software. All changes or additions to the code will be integrated and then tested for functionality. If it passes, the deployment process produces output in the form of the latest version of the application. In this case, all parties are involved in collaborative development until the software is ready for release [3].

A method is a systematic way or technique used to do something. Methods can be applied to one or more of the existing processes in the methodology. Meanwhile, methodology is a unity of methods, rules and stages used by science, art, and other scientific disciplines [4]. From this understanding, systematic and regular steps are needed in solving problems with the purpose of clearly know the procedures and sequences taken [5]. Thus, the results of problem solving can be optimal. To build software engineering, it is necessary to have automatic or semi-automatic support for carrying out the method so that computer-assisted software engineering can be built [6]. In practice, however, it is often the case that using the same methods does not produce the same results or, in other words, does not affect a software project in a comparable way [4].

Augmented Reality is a technology capable of displaying a visual spectrum in real-time which is currently developing rapidly. Its presence not only brings different nuances in the human visual screen but is also able to combine several technologies to bring digital information into visual perception [7]. To build an AR project, a special method is needed because not all digitally enhanced media today is “Augmented” reality. 2D images or Photoshop conversions are not AR. As is the case with images produced from television or films such as “Jurassic Park” and “Avatar” are also not AR. In contrast, a game that uses live feed and provides a real experience is AR. Not all visual searches can be categorized as AR. Therefore, something is said to be AR if it has the main characteristics of the AR system, namely [8]:

- Real time interactivity
- Use of 3D virtual elements
- Mixture of virtual elements with real elements

According to research conducted by Bean [9], the 3D animation industry, such as film, games, television, advertising and visual effects, uses three main stages in each segment that must be carried out. The three stages are pre-production, production, and post-production. In the pre-production stage,
the team will think about ideas and make sure to tell a compelling story. Determination of character, environment and other supporting factors need to be made as attractive as possible. At the production stage, if the previous stages have been well structured and mature, the project can be carried out. Then in the post-production stage, 2D visual effects and 3D animations are created. In this stage color correction, sound, and other effects need attention so the project will look fantastic. In addition to these three stages, it is necessary to consider the software and hardware components that will be used to create AR application contexts [10].

II. LITERATURE REVIEW OF SOFTWARE METHODS

Software is a computer program that is a key component of most businesses such as products, services, industrial processes, and back-office functions. Responding to changing customer needs and market conditions requires the ability to improve software quality quickly and continuously. This approach is known as continuous software engineering [11].

Continuous software engineering is a paradigm for streamlining software engineering by conducting gradual and continuous delivery of software with the purpose of quickly get feedback from customers and improve software quality according to customer requirements [3]. With this gradual and continuous method, the potential for improvement is easier to plan and strengthen. Smaller portions of software updates allow developers to focus on a collaborator, experience-based business model. The essence of continuous software engineering is continuous integration and delivery. This end principle is expected to cover the entire software engineering process from initiation, analysis, development, integration, validation, verification, delivery, to gathering feedback and planning the next software iteration step.

The stages in the software method generally consist of planning, analysis, design, and programming stages [4]. In the last two decades there has been a change in the stages of software development to address challenges that occur in the field. Based on the results of surveys and interviews conducted by Fitzgerald [1], it is known that only 6% of software developer practitioners apply formally defined methods. The rest do not follow the rules of the established method. Then what causes it? Why is there a difference between the method described and the one implemented? From the results of the literature review there are those who argue that software practitioners do not get sufficient knowledge, are not educated, and do not receive proper training. In the research conducted by Parnas [12] in his writings, he acknowledged that the reasons for non-compliance from software practitioners observed were disciplinary methods and norms.

From the application that has been carried out in previous research it is known that the agile method has the advantage of being more flexible and able to accept change. Its extraordinary adaptability made this method quickly accepted in today’s software environment [13]. Very complex software projects require integrated, distributed teams. This plays an important role as a bridge between the developer and other stakeholders so that if there is an error it will be easily detected and immediately corrected. The software development method aims to obtain quality and conformity to user requirements[14].

Currently the need for integrated practice has increased. As an example, for extreme agile methods where their popularity is steadily increasing. Then the DevOps method recognizes that software development and operational deployment must be continuous [15]. From the various results of the literature that has been reviewed, it is known that there is a relationship between business strategy and software development. These two things must be continuously assessed and improved. In research conducted by Fitzgerald [1] the combination of the two is abbreviated as BizDev.

The software development methods commonly used by AR today and which will be used as comparison material are as follows, Table I:

<table>
<thead>
<tr>
<th>Method</th>
</tr>
</thead>
<tbody>
<tr>
<td>a) Waterfall</td>
</tr>
<tr>
<td>b) Agile</td>
</tr>
<tr>
<td>c) Scrum</td>
</tr>
<tr>
<td>d) RAD</td>
</tr>
<tr>
<td>e) Prototype</td>
</tr>
<tr>
<td>f) DevOps</td>
</tr>
<tr>
<td>g) Kanban</td>
</tr>
<tr>
<td>h) Rational Unified Process</td>
</tr>
<tr>
<td>i) Spiral</td>
</tr>
<tr>
<td>j) 3D Pipeline Production</td>
</tr>
</tbody>
</table>

AR technology is still relatively new for software developers to develop, and it is not easy to choose the right development method. According to Dennis, there are several considerations in choosing a software development method based on several criteria, namely: clarity of user needs, mastery of technology, level of system complexity, level of system reliability, execution time and visibility of the implementation schedule [16]. In research conducted by Fitzgerald [1] on continuous software engineering, it was found that there are four activities, namely, strategy and business planning, development, operation and improvement and innovation. Of the four main activities, there are sub-sections as follows:

- **Strategy and business planning**
  - Continuous planning
  - Continuous budgeting

- **Development**
  - Continuous integration
  - Continuous delivery
  - Continuous deployment
  - Continuous verification
  - Continuous testing
  - Continuous compliance
  - Continuous security
  - Continuous evolution

- **Operation**
  - Continuous use
  - Continuous trust
  - Continuous run-time monitoring

- **Improvement and Innovation**
  - Continuous improvement
  - Continuous innovation
  - Continuous experimentation
<table>
<thead>
<tr>
<th>Method</th>
<th>Superiority</th>
<th>Weakness</th>
</tr>
</thead>
</table>
| Waterfalls                  | 1. Easy in management because almost whole requirements have been identified and documented.  
2. Sequential stages linearly, complete identification and documentation, making the process easy understood by all the team involved or project owner. | 1. No flexibility to change occurring needs in stage development system.  
2. Almost no tolerance errors, especially in stages planning and designing. |
| Spiral                      | 1. High amount of risk analysis.  
2. Good for large projects and mission critical. | 1. Can be an expensive model for use.  
2. Risk analysis needs very specific skills.  
3. Success projects are highly dependent on the risk analysis stage. |
| Agile Development           | 1. Method light in accordance small project and medium project.  
2. Produce cohesion good team.  
3. Emphasize final product.  
2. There is risk to sustainability, maintenance, and time longer.  
3. Needed mature plans and roles leader existing project experienced. |
| Scrum                       | 1. Scrum can help a team finish results project fast and efficiently.  
2. Scrum confirmed the use of time and money effective.  
3. Developed code and tested during the sprint review.  
4. Teams earn clear visibility through scrum meetings.  
5. Scrum, with nimble, adopt bane come back from customers and stakeholders’ interests.  
6. Short sprints possible change based on bane come back with easier. | 1. Scrum often lead to scope creep because lack of date definite end.  
2. Opportunity failure project high if individual have no committed or cooperative.  
3. Only can succeed with members who have experienced team.  
4. Meeting daily sometimes makes member team frustrated.  
5. Quality difficult applied until team through an aggressive testing process |
| RAD                         | 1. Efficiency time delivery.  
2. Change needs can accommodate.  
3. Cycle time can be short with the use of powerful RAD tools.  
2. Suitable for system-based components and measurable.  
3. Need involvement users all over cycle system.  
4. Requires highly skilled personnel.  
5. Dependency high on modeling ability. |
| Prototype                   | 1. Accurate identification requirements because it has input from the project owner so that the appearance of the resulting prototype can be adjusted to the immediate needs of the user.  
2. Errors and redundancies can be minimized because of the good identification process to prototype shape. | 1. Each evaluation and input for prototype changes will add to the complexity of the system being developed.  
2. Give burden addition to programmers.  
3. There is a need for additional costs related to making a prototype display. |
| DevOps                      | 1. Faster application development and deployment.  
2. Be more responsive and fast to market changes.  
3. Advantages in terms of software delivery time and transportation costs.  
4. Improve customer experience and satisfaction.  
5. Simplifies collaboration as all tools are placed in the cloud for customer access. | 1. Professionals in the field of DevOps yet adequate.  
2. Cost management infrastructure in relative DevOps methods is high.  
3. Lack of understanding of DevOps methods can cause problem in integration project continuous automation. |
| Kanban                      | 1. Kanban’s are very simple and easy method understood so that make it easy management company.  
2. Kanban method is spot on time or Just in Time (JIT).  
3. The Kanban method is a very responsive system and does not cause slowness or delay. | 1. Kanban cannot be used as an independent tool. This method is not quite suitable for a single application but can be combined with other methods such as Scrum, JIT, and others.  
2. As assignments constantly move between columns of the kanban board, a certain predictable time for the completion of a task or activity becomes difficult.  
3. Kanban is not suitable for dynamic environments.  
4. Kanban will be very difficult to implement if there are too many related joint activities or tasks in a system. |
| Rational Unified Process (RUP) | 1. RUP is a complete methodology with all easily available documentation.  
2. RUP is publicly published, distributed, and supported.  
3. Training is available where RUP can guide users through the process with step-by-step tutorials. Many institutions also offer training courses.  
4. Requirements Change where proactive resolution of changing client conditions and associated risks.  
5. Reduce integration time and effort.  
6. The reuse rate becomes higher. | 1. The process is too complex, too difficult to learn, and too difficult to apply properly.  
2. Integrated process does not capture the sociological aspects of software development and details about how the software develops step by step. |
| 3D Pipeline Production      | 1. Method This possible 3D animation is shown in a manner realistic, dynamic, and detailed.  
2. Method This support technology modern 3D animation possible movement displayed with fluid and very detailed.  
3. At the realistic and dynamic rendering stage, 3D animation also has Power strong visual appeal. They have shown for interesting more Lots attention than animation similar in 2D. | 3D animation uses more complex techniques than 2D animation. This means it generally takes more time to develop 3D animations. It also requires more expensive software to create 3D animations. |
The method shown in Fitzgerald’s research [1] is a common method in software engineering. This study tries to explore AR as its focus so that a theory is needed for continuous software engineering for AR considering that AR is not just an ordinary application but there are certain 3D elements and components that must be met so that it can be called AR.

III. TRENDS LANDSCAPE OF SOFTWARE ENGINEERING

The Agile method at its inception was focused on the software development function and was considered only suitable for small businesses or only for small teams [17]. But over time there has been a change with the emergence of many studies which identify that the concept of agile has entered the scale of large companies. The seven general principles of common practice for agile methods are Scrum, XP, DSDM [18]. These seven general principles address the agile approach in several dimensions. In research conducted by Leffingwell [19] said that the experience of organizations adopting agile methods in companies uses agile scale frameworks to explain practices and activities, roles, and artifacts. Basically, the organization must be responsive in responding to environmental changes that occur [20]. The DevOps method emerged because there was a gap in the development and operations functions in the company for large software scales because the bigger the team, the higher the responsibility. Automation, DevOps relies on full automation of build, deploy and test to achieve short lead times, and consequently get rapid feedback from users. Sharing happens on many levels, from sharing knowledge, tools, and infrastructure, celebrating successful releases to bringing development and operations teams closer together.

There are several tools that can be used to practice continuous software engineering, including:

- Jenkins
  Jenkins is an open-source tool that is often used especially in the continuous integration/continuous deployment (CI/CD) stage. Jenkins makes it easy to automate the integration and code testing process through features provided to speed up the delivery process introducing automation [17].

- GitLab
  GitLab is a complete software development platform that provides a CI/CD pipeline to integrate, test, and deploy code automatically [18].

- Travis CI
  Travis CI is a CI/CD tool that provides integration with Git and GitHub repositories resulting in fast integration with various programming languages as well as a flexible testing environment [19].

- Circle CI
  Circle CI is a cloud-based tool that provides integrated Git repository features that enable automated testing and deployment and supports multiple programming languages [20].

The tools above are just a few examples while the choice of the right tool depends on the needs of the project and the preferences of the development team.

IV. PROPOSED CONTINUOUS SOFTWARE ENGINEERING FOR AUGMENTED REALITY

The stages in the software method generally consist of planning, analysis, design, programming, and maintenance [4]. This research proposes additional stages in the software development method for AR, namely continuous planning, continuous analysis, continuous design, continuous programming, continuous integration, and continuous maintenance, Table II.

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A. Sustainable Planning

In the context of software development, planning is an initial idea that is episodic from the current set of problem formulations [21]. Over time, changes occur, and the business environment does require planning activities to be carried out more frequently to ensure alignment between the needs of the business context and software development. Not only the business environment but covering all areas of human life will certainly always experience changes [22]. In sustainable planning there are stages of continuous planning and sustainable budgeting.

- Continuous planning
  The relationship between planning and portfolio becomes important at this stage. Planning must be iterated and released, a portfolio that includes product planning activities must be carried out [23]. However, the portfolio still has the possibility of failure even though the project is generally successful. The portfolio approach tends towards the organization. In previous research, it is known that there are still many cases where the approach to individual project management has received more attention than the portfolio approach [1].

In AR, there are stages in the ideas and stories sections that should always be refreshed so that users always get new findings in the AR applications they use. Periodically new
ideas should be rolled out and new story additions or new characters in AR can provide users with an unforgettable immersive experience [24].

- Sustainable budgeting

The budget is a financial plan that is prepared in detail and coordinated from management plans with the aim of achieving targets within a predetermined period [1]. In general, budgeting is an annual or even multi-year event that describes annual events in the 'short term' and for three years or more in the 'long term'. To overcome this, the concept of "sustainable budgeting" is needed so that budgeting becomes more flexible and makes managers more flexible in making operational decisions at their own discretion to deal with unexpected situations that cannot be predicted in the master budget plan[25]. The budgeting system must be open and transparent so that the team can manage finances independently according to a predetermined budget but not limited to the annual cycle [26].

B. Continuous Analysis

At this stage it is possible not only to measure the overall behaviour of the mechanism, but also an in-depth analysis of the software engineering being executed. Experimental data to verify the continuous analysis method is strongly recommended [21].

- Continuous deployment

Continuous deployment is the continuous delivery of code changes to applications that are released during software development. This stage must be carried out in a series of predetermined tests. After going through the testing phase, the system will confirm that the software is ready to be released to the customer and that it is in accordance with customer requirements [1].

- Continuous verification

Continuous verification is the discipline of proactive experimentation or the process of monitoring applications for post-deployment anomalies. An anomaly is any interruption in normal operation that may affect application users especially in AR, including [27]:

- a) High response latency
- b) Server-side error
- c) Client-side error
- d) Downtime of any application component
- e) Unexpected scaling or failover events

The purpose of continuous verification is to collect data from implemented implementations and then analyze them through machine learning and create a basis for good implementations. The benefit of this ongoing verification is to identify that something is wrong and take corrective action—for example, reverting the app to a stable or default version. Precautions should be implemented as quickly and smoothly as possible before the customer becomes aware of the problem [1].

C. Sustainable Design

In the sustainable design for AR, design components and final appearance of the project that has been decided can be updated according to market demand. At the beginning of creating AR, designers created conceptual art using media ranging from pens, charcoal, pencils, dyes, or oil paints to using computers such as Adobe Photoshop software. There is no problem using any media if the concept to be conveyed can be understood by users. Artwork often reflects the artist who created it, so the artist's mood and instincts should be an important aspect to pay attention to. The reason why this stage must be continuous is because when making art at the beginning, it is limited by a deadline. So that continuity is needed to ensure the results are delivered in accordance with the desired initial concept and get user feedback. Therefore, updates at this stage need to be done to perfect the design [28].

- Continuous testing

Continuous testing is a software development process where applications are tested continuously throughout the software development life cycle (SDLC). The goal of continuous testing is to evaluate software quality throughout the system life cycle to provide critical feedback earlier and enable faster, higher quality delivery. The advantage of this stage is when the context is still fresh in the mind of the developer and an error occurs, it is quickly resolved before the problem becomes unexpectedly deep and widespread. The benefit to software developers is that testing can be performed effectively and helps reduce overall development time by up to 15%. This is proof that continuous testing can be used as a tool to reduce waste of waiting time [29].

- Continuous compliance

At the beginning of its implementation, Agile methods were considered only suitable for small projects and placed in a crisis context, not safety. However, in recent decades Agile methods have been successfully applied to large, distributed projects. In the research conducted by Fitzgerald [1] there was discussion about adapting the Scrum method to R-Scrum or Regulated-Scrum. This is actually a disguise from a waterfall approach to an agile approach that allows developers to complete projects faster [1].

- Continuous security

Security is a priority in all phases of software development life, even after implementation, security is an important thing that must be done. Security can finally be said to be a non-functional requirement that is often unintentionally delegated to a lower priority. Therefore it is necessary to apply an intelligent and lightweight approach to be able to identify vulnerabilities to security issues [1].

- Continuous evolution

Software evolution is fundamentally dependent on the expertise of the developer as well as the changeability inherent in the software product itself [1].

D. Continuous Programming

In continuous programming in AR, it requires continuity in the research and development (R&D). R&D is a component
that covers from pre-production to post-production in 3D animation flow [9]. For example, in the animated film Finding Nemo, a team of artists from various components had to think about how to make the water feature look as if it were real. Likewise with other objects that float in the water must really look alive. When it was first released in 2003, there were no 3D animated films depicting water because it was considered especially difficult to render efficiently. Making this film is not easy, it takes years and provides its own challenges for the R&D team. But the resulting effort is well worth it, Finding Nemo succeeds in bringing the animation to life in the expected form [30].

- Continuous use

Continuous use provides advantages in terms of time and cost because the system is not built from scratch but from a continuation of an existing system. This continuous use does not mean that it can be used automatically because there must be some initial consideration and decision to use the software. The cost of acquiring new customers is estimated to be up to ten times that of retaining existing customers [21]. Previous research also stated that developers tend to stick with the system rather than designing from scratch [1].

- Continuous trust

Continuous trust is a process that takes time so that there is confidence that the vendor will do all its expertise to meet customer expectations according to their needs. Continuous use is highly dependent on continued trust where the relationship is a complex relationship. In Hoehle et al [1] research, trust is a very important start to be achieved in a transaction. For long-term activities like Cloud remote services, AWS and others require ongoing trust. Software developers need to pay attention to the issue of trust in sustainability because even if it starts well, the trust itself can erode from the user experience both internally, such as inconveniences in certain features or external factors, such as news regarding vulnerable security issues and others [21].

- Continuous run-time monitoring

A classification scheme is provided to help understand the approach chosen by a system designer with a particular method. This stage can also be used to analyse ongoing projects and consider functional and non-functional aspects that are interesting to study in software development projects [21].

E. Continuous Integration

Continuous integration is defined as a process that is usually triggered automatically and consists of interconnected steps such as compiling code, running unit and acceptance tests, validating code coverage, checking compliance with coding standards and building deployment packages. While some form of automation is typical, the frequency of integration is also important as it needs to be regular enough to ensure rapid feedback to developers. Finally, continuous integration failures are high-profile events that may have several visible ceremonies and artefacts to help ensure that the issues causing these failures are prioritized for resolution as quickly as possible by whoever is held responsible [31].

- Continuous Release

When a component is released, its version number declaration is updated, as well as its dependency declarations, because those dependencies always refer to the component that was also released. This makes the component-based release process recursive. There are significant costs associated with this method of release. The more frequently a dependent component is released, the more frequently a component that depends on it must be released to take advantage of the additional quality functionality it contains. Furthermore, with each dependency release, all components that use it should be tested for integration, before they can be released on their own [21].

- Continuous delivery

Continuous delivery is a prerequisite if developers are going for continuous adoption, but not necessarily the other way around. Continuous delivery focuses on being able to deploy software to multiple environments but not necessarily to customers. As opposed to continuous deployment which is obligatory to release valid software to its users [23].

F. Continuous Maintenance

Continuous maintenance in AR is more focused on maintaining the long-term sustainability of the software both during development and after production. This stage has a series of other advanced stages, namely continuous improvement, continuous innovation, and continuous experimentation.

- Continuous improvement

Continuous improvement activities are efforts to add customer value through reactive initiatives from software developers. Therefore, innovation is needed as a proactive strategy to emphasize customer satisfaction [1].

- Continuous innovation

Innovation in a business context is new ideas that are transformed through processes to create business value for customers. An example of this innovation is discovery plus exploitation[31]. Research on software engineering has conducted a lot of research on innovation and these keywords are the most searched for by software developers and the business community, especially regarding open innovation. The concept of Lean startups is an example of continuous innovation. Testing is done by beta testing which is used to get customer feedback before the official release of a software product. Continuous innovation is a continuous interaction between operations, gradual improvement, learning, and radical innovation to identify added value features aimed at combining operational and strategic effectiveness or known as exploitation and exploration [1].

- Continuous experimentation

Continuous experimentation has another benefit for teams in that it generates measurable metrics of change and progress so that team goals are clearly defined. The application of continuous experiments in several domains requires solutions and challenges that range from infrastructure challenges,
measurement challenges and social challenges. These challenges may only be relevant in one domain, but the solutions developed can be applied to other domains. This domain-specific challenge is closely related to the resulting solution [31].

In many publications on perpetual experimentation, the merits of the experiment are mentioned only as a motivation, i.e., improving product quality based on selected metrics. Further studies are needed to determine, for example, if there are more benefits, whether they apply to all companies involved in the experiment, or whether they could be obtained through other means. Another benefit is the potential use of continuous experimentation for software quality assurance. Continuous experimentation can support or even change the way quality assurance is performed for software. Software changes, for example, can only be applied if key metrics are not derived in the associated change experiment. Thus, the loss of quality can become measurable and measurable. Although some papers mention the use of continuous testing for software quality assurance [31].

V. DISCUSSION

When working on AR projects, designers need to weigh quality against time and budget. When using today’s rendering engines and shaders, 3D artists must find a happy medium between perfect looks and reasonable render times. New technologies are released every year in various forms—from computer hardware with faster speeds and data transfers, to software with advanced capabilities, and technologies that make workflows smoother. Some of the new trends being pursued by the 3D animation industry include full-body and detail motion capture, stereoscopic 3D output, point-cloud data, real-time workflow capabilities, and virtual studios. Each will provide a faster project turnaround and will allow artists to focus on the art of the project and not the technical hurdles of the production line. Continuous software engineering relies on a basic set of principles that govern every field of technology and includes modelling activities and other descriptive techniques.

VI. CONCLUSION

In general, the methods in software engineering include planning, analysis, design, programming, and maintenance. In this paper we present our initial findings about the continuous stages of the software method, namely continuous planning, continuous analysis, continuous design, continuous programming, continuous integration, and continuous maintenance then the continuous process of AR system development can be updated according to market demand.

For future work, tool support is available for continuous software engineering but suitable tool support for the whole concept from coding to delivery and data management for decision making needs attention in the future.

REFERENCES


A Local-Global Graph Convolutional Network for Depression Recognition using EEG Signals

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Abstract—Graph Convolutional Networks (GCNs) have shown remarkable capabilities in learning the topological relationships among electroencephalogram (EEG) channels for recognizing depression. However, existing GCN methods often focus on a single spatial pattern, disregarding the relevant connectivity of local functional regions and neglecting the data dependency of the original EEG data. To address these limitations, we introduce the Local-Global GCN (LG-GCN), a novel GCN inspired by brain science research, which learns the local-global graph representation of EEG. Our approach leverages discriminative features extracted from EEG signals as auxiliary information to capture dynamic multi-level spatial information between EEG channels. Specifically, the representation learning of the topological space in brain regions comprises two graphs: one for exploring augmentation information in local functional regions and another for extracting global dynamic information. The aggregation of multiple graphs enables the GCN to acquire more robust features. Additionally, we develop an Information Enhancement Module (IEM) to capture multi-dimensional fused features. Extensive experiments conducted on public datasets demonstrate that our proposed method surpasses state-of-the-art (SOTA) models, achieving an impressive accuracy of 99.30% in depression recognition.

Keywords—Electroencephalogram; depression recognition; Local-Global Graph Convolutional Network (LG-GCN); multilevel spatial information; brain regions; multiple graphs

I. INTRODUCTION

Depression is a prevalent mental disorder affecting a large population worldwide. According to the World Health Organization (WHO), more than 350 million individuals globally suffer from depression [1]. It is characterized by significant mental impairment and negative emotions, including feelings of sadness, fatigue, and hopelessness. Currently, the primary method for diagnosing depression relies on doctor-patient communication. However, factors such as patient subjectivity, low sensitivity, and denial pose significant challenges to the diagnostic process [2]. Therefore, there is a pressing need for an objective and accurate method for detecting depression.

In recent years, electroencephalography (EEG) has emerged as a widely adopted technique for classifying depression due to its advantages, such as high temporal resolution, low acquisition cost, and ease of operation and recording [3]. It has become a commonly used and effective tool for assessing brain function. Previous approaches have involved models based on recurrent neural networks (RNNs) and convolutional neural networks (CNNs), which analyze EEG signals in the time-frequency domain, extracting features independently from individual channels. However, research has demonstrated that the brainwave patterns of individuals with depression arise from interactions between multiple channels, and EEG electrodes are positioned in a spherical space. Traditional CNNs face limitations in handling irregular and non-Euclidean data. In contrast, graphs, which are effective in handling irregular data, are better suited for modeling signals in a three-dimensional spherical space. In this context, each electrode can be seen as a node in the graph, and the spatial relationships or correlations between electrodes can be represented as edges. Graph Neural Networks (GNNs) leverage the adjacency relationships among nodes to jointly learn the spatial patterns of EEG signals [4].

Integrating prior knowledge derived from neuro-psychological research into the design of Graph Neural Networks (GNNs) presents significant potential for decoding psychological states from EEG signals. In the case of depression, it has been observed that the manifestation of this condition in individuals may involve interactions within specific brain regions [5]. Activation of a particular brain area can also trigger simultaneous activation in other regions. Research has indicated the presence of high-level connections between electrodes on the left and right hemispheres, offering additional insights for biomedical analysis [6]. Ding et al. [7] have demonstrated that many cognitive functions rely on the cooperation between different brain regions rather than being confined to a specific area. While previous research [8] utilized a globally connected adjacency matrix with learnable connections, it failed to consider the local activities within each functional region. On the other hand, the Regularized Graph Neural Network (RGNN) [9] established local connections based on spatial distances between electrodes but struggled to effectively capture the complex relationships between functional regions. Thus, it is crucial to strengthen the connectivity within local functional regions while establishing connection patterns between channels based on global dynamics. Additionally, relying solely on spatial patterns would overlook important discriminative features present in the original EEG data, which are vital for identifying depression in EEG signals. Consequently, our research focuses on appropriately constructing the brain topology based on EEG data and addressing challenges related to information loss.

To tackle the aforementioned issues, we propose an algorithm for depression recognition based on a local-global graph convolutional network (LG-GCN). LG-GCN constructs

*Corresponding Author.
both a global dynamic graph and a local functional graph to explore the multi-level spatial information across EEG channels. To capture adaptive multi-dimensional features of EEG, we introduce an Information Enhancement Module (IEM) that incorporates data dependencies and cleverly integrates them with spatial information. Inspired by brain science research [10], LG-GCN incorporates topological information from both local and global perspectives in the graph convolutional layers. Furthermore, recognizing that the graph structure may not extract all discriminative features from the original signals, we adopt Gated Convolutional Networks to capture the dependency relationships between raw temporal data and advanced features, thereby enhancing the model's performance. Finally, to assess the effectiveness of incorporating prior knowledge into the LG-GCN model, we analyze the differences between various brain region partitioning methods and conduct extensive visualization experiments.

The main contributions of this study are as follows:

1) Introducing a local-global multi-graph fusion framework that explores multi-level features of graph topology spaces. This framework overcomes the limitation of insufficient extraction of depression-related information in a single encoding path by utilizing an information enhancement module to adaptively supplement multi-dimensional fused features.

2) Proposing an adaptive global dynamic graph and local functional connectivity graph that are integrated into the global graph convolutional network. This integration allows for the incorporation of local information, capturing the multi-level spatial dependencies in EEG data. It effectively associates the spatial distribution of EEG signal channels with depth-encoded depression features, resulting in improved classification performance.

3) Validating the effectiveness of the LG-GCN framework on a public dataset, where it achieves state-of-the-art (SOTA) classification performance. Additionally, multiple ablation experiments are conducted, and LG-GCN is compared with other methods, further confirming its efficacy.

The remaining structure of the article is outlined as follows: Section II briefly reviews related works. The proposed framework will be described in detail in Section III. Section IV provides extensive experiments and analyses. Section V provides a discussion of the results obtained, and finally, the study is concluded in Section VI.

II. RELATED WORK

A significant number of previous studies have attempted to use machine learning methods to detect depression. Saeedi et al. [11] used a genetic algorithm to select significant features from linear and nonlinear features and employed an enhanced K-nearest neighbor algorithm to classify the EEG signals of depressed patients. Mumtaz et al. [12] performed time-frequency decomposition of EEG signals based on wavelet transform to construct feature matrices, which were then inputted into an LR classifier. Although machine learning has achieved significant success in classification tasks, it faces challenges in feature selection complexity and high accuracy performance requirements. Therefore, deep learning methods are gradually becoming popular among researchers. Kang et al. [13] proposed transforming the asymmetry of EEG into a matrix image as input to a CNN, but the calculation of electrode spatial positions in the two-dimensional image transformation is challenging. Acharya et al. [14] introduced a 13-layer CNN model based on deep learning, but it has complex training and requires a significant amount of time. Chen et al. [15] used a lightweight DCTNet model, which consists of a six-layer neural network with CNN-LSTM and achieved high classification performance. Compared to machine learning methods that rely on manual feature extraction, deep learning algorithms have better performance in improving the accuracy of depression diagnosis. However, most current deep learning methods focus on mining information in the time-frequency domain, while the spatial distribution information of channels is an important but often overlooked feature.

In recent years, GNN models have shown promising performance in handling graph-structured data, providing new research directions for the development of EEG signal analysis. Zhang et al. [16] proposed a depression recognition framework based on the fusion of time-space ubiquitous EEG features at the feature level. The model selected 19 optimal time EEG features and four optimal spatial metrics features and analyzed the intrinsic connections among EEG signal channels. Zhu et al. [17] modeled correlation using graphs and proposed an attention based GCN that achieved a recognition rate of 96.50% for depression. Wu et al. [18] combined space-time graph convolutional networks (ST-GCN) with a depression-related functional connectivity graph in a deep learning approach, improving the ability to represent spatiotemporal features. Sun et al. [19] used a complex network graph convolutional neural network (CN-GCN) with a multi-branch structure to explore deep information between channels, achieving a recognition rate of 99.29% for depression. However, these models only used one type of functional connectivity in the brain. In this article, the proposed model considered both structural and functional connectivity for the first time. Jang et al. [20] suggested that individuals with depression tend to exhibit greater relative right frontal lobe activity. Mumtaz et al. [21] found that individuals with depression exhibit greater activity in the front EEG. Therefore, in depression research, considering both global brain connectivity and local functional connectivity is critical while constructing multi-channel EEG signal dependency relationships. The proposed LG-GCN model in this paper analyzes the spatial relationships within and between different functional areas, extracting more comprehensive spatial features for more effective depression identification.

III. METHODOLOGY

To fully exploit the potential of EEG data, LG-GCN is proposed for EEG-based depression recognition, which consists of a local-global graph convolution module (LG-GCM) and IEM. The overall framework is shown in Fig. 1, which can be divided into the following steps:
1) Considering the frequent local connections in the brain network and the need for better differentiation of brain regions in relation to depression, we construct local regions based on prior research, each region's information are aggregated as nodes in the local functional graph, and node similarity are computed to represent functional connectivity.

2) The brain network adopts an adaptive adjacency matrix to dynamically calculate the connection strength between channel nodes, dynamically reflecting the information under depression from a global perspective and introducing a GCN to aggregate local-global features.

3) The IEM is used to further extract information from the original signal, where the weighted averaging fusion mechanism is used to extract high-dimensional features outputted by the GCN, and then inputted into an adaptive attention fusion network to obtain complementary information. Next, this article will present detailed information related to this.

A. Local-Global Graph Convolution Module

The results of pattern analysis of neuroimaging data indicate that the depression group exhibits distinct patterns related to emotional categories in the activity of the distributed neural system spanning across cortical and subcortical regions [22]. Therefore, to simulate the interconnections between different brain regions, we propose a multi-graph representation, namely LG-GCM. By considering spatial information from different perspectives, the adaptive global dynamic graph can dynamically integrate EEG channel information, while the local functional connectivity graph can characterize static relationships between different brain regions. Finally, multi-layer GCN is introduced to capture multi-graph and multi-level representation features by adaptively fusing local and global spaces.

1) Local functional graph: Different regions of the human brain cortex are highly connected and concentrated. Based on prior knowledge from the field of neuroscience [23], we construct two types of local brain regions by aggregating local EEG features to capture more robust features. Fig. 2(a) shows the first partition based on the reference brain cortex anatomy. The brain cortex is typically divided into four lobes: the frontal, parietal, temporal, and occipital lobes, each of which is responsible for different tasks. Due to the importance of electrodes located in the frontal lobe (FP1, FP2, F7, F3, FZ, F4, F8) in depression detection tasks [20], we divide the regions in more detail to meet the needs of structural and functional connectivity. Fig. 2(b) shows another effective method based on the division of regions between the brain’s two hemispheres.

![Illustration of the proposed model LG-GCN](image_url)
We represent multiple graphs as weighted undirected graph $G = (V, \varepsilon, A)$, where $V$ is the set of nodes, $\varepsilon$ represents the connections between nodes, the adjacency matrix $A \in R^{N \times N}$ represents the connectivity between nodes, and $N$ is the number of EEG channels. The region functional graph containing $P$ subgraphs is represented as $G_{R}^{(1)}$, $G_{R}^{(2)} = \left[ G_{R_{1}}, G_{R_{2}}, \ldots, G_{R_{P}} \right] $. For $k$th region functional subgraph $G_{R_{k}}$, the input data is denoted as $G_{R_{k}} \in R^{S \times n_{k} \times D}$, where $S$ represents the number of samples, $n_{k}$ is the number of channels in the $k$th subgraph, and $D$ represents the frequency band, including $\delta$ band (0.5–4 Hz), $\theta$ band (4–8 Hz), $\alpha$ band (8–13 Hz), and $\beta$ band (13–30 Hz). These frequency bands have been used in previous studies to investigate differences between patients with depression and healthy controls [24]. In this work, PSD features in the frequency domain of the EEG signals are extracted using the Welch method [19] for each channel node, which can be obtained from the MATLAB software with default parameters. Owing to EEG original data containing noise and redundant information, singular value decomposition (SVD) is commonly used to perform dimensionality reduction and extract relevant information. It can be represented as

$$G_{R_{k}} = W_{R_{k}} \cdot G_{R_{k}}^{T}$$

(1)

Where $W_{R_{k}}$ represents the left singular matrix, $G_{R_{k}}^{k} \in R^{n_{k} \times \mu}$ is the channel-level matrix containing advanced channel information, with dimensions of $n_{k} \times \mu$. Where $n_{k}$ is the number of channels in the $k$-th subgraph ($\sum n_{k} = N$), and $\mu$ represents the number of features for each EEG channel. Furthermore, the local subgraph can be represented as $G_{R_{k}} = \left[ g_{R_{k}}^{1}, \ldots, g_{R_{k}}^{n_{k}} \right]$, where $g_{R_{k}}^{n_{k}}$ is the node feature vector of the local subgraph. The aggregation function aggregates the node feature vectors in each local subgraph, which can be the maximum, minimum, average, etc. This process is known as graph coarsening. In LG-GCN, the average aggregation is selected as the aggregation function. Hence, the output of the local graph, $G_{local}^{(1)}, G_{local}^{(2)} \in R^{\mu \times \mu}$, can be calculated by

$$G_{local}^{(1)}, G_{local}^{(2)} = F_{aggregate} \left( \left[ G_{R_{1}}, G_{R_{2}}, \ldots, G_{R_{P}} \right] \right)$$

$$= \left[ \frac{1}{n_{1}} \sum_{n=1}^{N_{1}} g_{R_{1}}^{n}, \ldots, \frac{1}{n_{k}} \sum_{n=1}^{N_{k}} g_{R_{k}}^{n} \right]$$

$$= \left[ h_{local}^{1}, \ldots, h_{local}^{N} \right]$$

(2)

Where $p$ is the index of each node in the local graph, $h_{local}$ represents the latent representation of the local subgraph. Then, the local subgraphs are concatenated to obtain $G_{local} = concat(G_{local}^{(1)}, G_{local}^{(2)}) \in R^{\mu \times \mu}$, which results in the feature fusion matrix of the two partitioned local subgraphs.

The definition of the local subgraph is incorporated into the basic adjacency matrix of the brain to model the correlation between different brain regions, which is used in the subsequent graph convolutional layers to enhance the feature propagation between more important local edges. The relationships between local graphs are utilized as the edges of the basic global graph. Neuroscience research suggested that activating one specific brain region also tends to activate other regions in the group for advanced cognitive processes [25]. The dot products between local graph representations for each EEG instance are calculated to reflect the relations among local graphs. Thus, we have $A_{global-base} \in R^{\mu \times \mu}$, which is calculated as follows.

$$A_{global-base} = \begin{bmatrix} A_{global-base}^{1,1} & \cdots & A_{global-base}^{1,V} \\ \vdots & \ddots & \vdots \\ A_{global-base}^{V,1} & \cdots & A_{global-base}^{V,V} \end{bmatrix}$$

$$= \begin{bmatrix} h_{local}^{1} \cdot h_{local}^{1} & \cdots & h_{local}^{1} \cdot h_{local}^{V} \\ \vdots & \ddots & \vdots \\ h_{local}^{V} \cdot h_{local}^{1} & \cdots & h_{local}^{V} \cdot h_{local}^{V} \end{bmatrix}$$

(3)

where $\cdot$ operation is dot product. $h_{local}^{1} \cdot h_{local}^{j}$ represents the similarity between local subgraphs as the edge weight in the adjacency matrix. $A_{ij}^{(ij)}$ represents the $i$th row and $j$th column of the basic adjacency matrix corresponding to the brain. Each local subgraph is treated as a node, forming the basic functional connectivity matrix of the brain. The specific process is as follows.

$$A_{global-base}^{ij} = \exp \left( -\frac{\left\| h_{local}^{i} - h_{local}^{j} \right\|^{2}}{2\sigma^{2}} \right)$$

(4)

Finally, $A_{global-base}$ forms the basic adjacency matrix of the local functional graph. The graph coarsening method aggregates the characteristics of local EEG signals, reducing redundant information and lowering computational complexity, thereby obtaining more robust features.

2) Global dynamic graph: Due to the complex relationships between functional regions of the brain, they exhibit temporal variability and high dynamics [26]. However, existing methods are often restricted by static prior knowledge or weak modification, making it difficult to fully capture the complex dynamic functional connections between brain regions. We aim to learn an adaptive brain network adjacency matrix [27], which is self-learned during the model training process to understand the global connectivity relationships between EEG channels and better understand the interactions between channels. Therefore, we define a non-negative function $F(X_{m}, X_{n})$ to quantify the strength of functional connections between channels, where $X_{m}$ and $X_{n}$ respectively represent the features between any two channels.
Function F calculates the connection weight between nodes based on signal differences between channel nodes and can dynamically reflect signal patterns under depression. The formula for calculating function F is shown below.

\[
A_{m,n} = F(X_m, X_n) = \frac{\exp(\text{ReLU}(W_t|X_m - X_n|))}{\sum_{m=1}^{N} \sum_{n=1}^{N} \exp((W_t|X_m - X_n|))}
\]

(5)

where \(W_t\) is a learnable parameter, and \(|X_m - X_n|\) represents the distance between the features of the two channels. The activation function ReLu is applied to constrain weak channel coupling and is applied to the output to ensure that \(A_{m,n}\) is a non-negative element. The output of the final global dynamic layer of the brain is a non-negative adjacency matrix A.

3) Multilayer GCNs: Once the local and global graph representations are obtained, we aggregate the information from these two types of graphs and enable global information interaction among the channels of EEG signals using GCN. The global dynamic graph extracts common features of depression patterns under coarse granularity, while the dense local functional graph can better capture functional connectivity features between different regions, containing more detailed information. LG-GCN captures the spatial information of EEG channels at multiple levels and dimensions, thereby extracting topological spatial features that are most favorable for the recognition of depression in EEG.

In terms of node initialization, we use the value of \(X^0 = X(G_{\text{local}})^T\) as the initialized graph node features of the graph, where \(X\) refers to the high-level feature set of the input signal, \(X \in R^{N \times d}\). The notation \((G_{\text{local}})^T\) represents the aggregation relationship of the node features of the local subgraph. Then, we utilize the adjacency matrix \(A\) and \(A_{\text{global-base}}\) to implement multilayer graph convolution and achieve feature propagation of local and global information of EEG signal channels. We first normalize the adjacency matrix \(A\) as shown in equation (6), where \(D\) is a diagonal matrix with \(D_{ii} = \sum_{j=1}^{N} A_{ij}\). Then, we calculate the graph Laplacian matrix \(L = D - A\) and perform spectral decomposition to obtain the eigenvalues and eigenvectors corresponding to \(L\), as shown in equation (7), where \(U = (\vec{u}_1, \vec{u}_2, \cdots, \vec{u}_N)\) and \(\Lambda\) is the eigenvalues and eigenvectors of \(L\), respectively, and \(\lambda_{\text{max}}\) is set to 2. Assuming that the signal \(x \in R^N\) undergoes Fourier transformation as \(x = \hat{x}\), we can calculate the Kth order Chebyshev polynomials \(T_{K-1}(\hat{L})\) as shown in equation (8).

Given \(X^{l-1}\) as the input to the l-th graph convolutional layer, the output \(X^l\) can be calculated using equation (9), where \(\theta_k^l\) is the trainable parameter corresponding to the graph convolutional layer. A key step in GCN is the aggregation of node features, as illustrated in Fig. 3. As the convolutional layers iterate, node features propagate through the graph structure, allowing LG-GCN to obtain the aggregation of global information.

\[
A = D^{-\frac{1}{2}}(I_N + A)D^{-\frac{1}{2}}
\]

(6)

\[
\hat{L} = U\Lambda U^T = \frac{2L}{\lambda_{\text{max}}} - I_N
\]

(7)

\[
T_l(x) = 2xT_{l-1}(x) - T_{l-2}(x), T_0(x) = 1, T_1(x) = x
\]

(8)

\[
X^l = \sigma(\sum_{k=0}^{K-1} \theta_k^l T_k(\hat{L})X^{l-1} A_{\text{global-base}})
\]

(9)

Since the multi-channel EEG graph only contains 19 nodes, to avoid the loss of important dynamic information, we do not perform graph pooling and instead directly sum up the outputs of each graph convolutional layer, integrating all node features into one graph representation \(Z_g = \sum_{l=0}^{N} X^l \in R^{N \times \mu'}\), \(\mu'\) is the feature dimension of the hidden layer.

B. Information Enhancement Module

LG-GCN can extract effective multi-graph and multi-level information, but it may ignore some important discriminative features in the raw EEG data. Hence, IEM is proposed to extract the dependency relationships between data and higher-level features, which are adaptively aggregated these features into the graph topology representation to capture a more comprehensive range of information within the EEG signals.

1) Feature Extractor (IEM-FE): To address the noise and high-dimensional characteristics of the raw EEG data and capture its discriminative features, we introduce a gated convolutional network as an FE for capturing enhancing information. The structure of the Gated Convolutional Networks, as shown in Fig. 1, utilizes two types of activation functions. Throughout the entire process, the computation and gate weight adjustment of the gate units are utilized to filter and update the output of the convolutional layers. This allows for the preservation of the ability to extract nonlinear features and further explores the discriminative features dependent on EEG data. Simultaneously, it enhances the network's generalization capability and robustness. Given the reshaped input data \(X \in R^{1 \times (N \times d)}\), the operation of the gated convolution can be expressed as follows:

\[
Z_l = S(\theta_1 \cdot X + b_1) \circ T(\theta_2 \cdot X + b_2)
\]

(10)

where \(\theta_1\) and \(\theta_2\) are two 1D convolution operators, \(b_1\) and \(b_2\) are model parameters, \(S(\cdot)\) and \(T(\cdot)\) denote sigmoid and tanh functions, and \(\circ\) is the element-wise multiplication. The output \(\circ\) of the gate mechanism is obtained by multiplying the outputs of these two functions, which is used
to represent the importance and information gain of each channel at a specific time point.

2) Weighted Average Fusion mechanism (IEM-AF): To obtain more comprehensive information from EEG data, AF mechanism is designed to fuse data-dependent auxiliary information into spatial information. Specifically, we flatten and map the output features \( Z_l \) and \( Z_g \) to a consistent dimension \( \hat{z}_l, \hat{z}_g \in R^{1 \times B} \). These features are then fused together using the following representation:

\[
\hat{z}_f = \omega(\hat{z}_l, \hat{z}_g)
\]

(11)

where \( \omega(\cdot) \) represents the weighted average fusion function. Subsequently, a fully connected layer is utilized to reduce the dimensionality of the fused spatial features and auxiliary features, denoted as \( z_g \) and \( z_f \) respectively. These features are then passed through an adaptive attention fusion network to achieve effective feature fusion and extract complementary information. The final representation for EEG depression recognition is as follows:

\[
z = \varphi_f \cdot z_f + \varphi_g \cdot z_g
\]

(12)

where \( z \in R^{1 \times B} \), \( B \) is the binary class number for depression and health. The attention values \( \varphi_f \) and \( \varphi_g \) with embeddings \( z_l \) and \( z_g \) are self-learned by the proposed model. Specifically, we adopt a non-linear transformation and a shared matrix \( q \) to calculate the attention scores using equation (13):

\[
AS_f = Q^T \cdot \tanh(W \cdot z_f^T + b)
\]

(13)

where \( W \) and \( b \) are the transformation matrix and bias vector, respectively. Similarly, \( AS_g \) is obtained in the same way. We then normalize the attention scores \( AS_f \) and \( AS_g \) using softmax (\( \cdot \)) function, yielding the attention values \( \varphi_f, \varphi_g \in R^{N \times 1} \), which represent the importance and relevance of the nodes. This can be expressed as:

\[
\varphi_f = \frac{\exp(AS_f)}{\exp(AS_f)+\exp(AS_g)}
\]

(14)

Ultimately, the embeddings \( z \) are normalized by the softmax(\( \cdot \)) function to calculate the predicted probabilities \( \hat{y} \).

C. Optimizing LG-GCN

LG-GCN can extract effective information from EEG data and utilize this information to optimize the model. The construction of the loss function is crucial in this process. To obtain the optimal network parameters, the backpropagation (BP) algorithm is applied to iteratively update the network parameters until the best or suboptimal solution is obtained. The loss function during the training phase consists of a classification optimization term and a spatial regularization term. The classification optimization term uses the cross-entropy loss function, which is defined as:

\[
L_{cla} = -\frac{1}{S} \sum_{i=1}^{S} \sum_{k=0}^{K-1} y_{i,k} \log \varphi_{i,k}
\]

(15)

Where \( S \) represents the samples and \( N \) represents the number of nodes. On the other hand, a spatial regularization term is introduced into the loss function to incorporate the smoothness and sparsity of the learned adjacency matrix, which better considers the spatial relationships among the nodes. This term can be defined as:

\[
L_{reg} = \frac{1}{C} \sum_{i=1}^{C} \sum_{j=0}^{N} \sum_{k=0}^{N} |A_{ij}| |A_{jk}|
\]

(16)

The total loss function consists of the above two terms, that is, \( \text{Loss} = L_{cla} + L_{reg} \). \( \lambda \) represents the regularization parameter that controls the trade-off between the classification and regularization terms.

IV. EXPERIMENTAL RESULTS AND ANALYSIS

In this section, we perform EEG-based depression recognition to demonstrate the effectiveness of the proposed LG-GCN model on public datasets. Then, experimental details and results are described, and finally, the ablation experiments and interpretability of EEG-based depression recognition are presented. The training procedure of the proposed model is shown in Algorithm 1.

Algorithm 1 LG-GCN Training Procedure

Require: EEG training samples S, ground-truth labels Y, training epoch e, number of partitions P, parameter K in LG-GCN, and learning rate r.

Ensure: Prediction of the model \( \hat{y} \)

1: Initialization;
2: Aggregate node features for each local graph by 1-3;
3: Calculate the adjacency matrix \( A_{\text{global-base}} \) by 4;
4: Compute the global dynamic representation;
5: \( \text{for } j = 1 \rightarrow e \text{ do} \)
6: Aggregate \( Z_g \) by 6-9;
7: Compute the EEG discriminative features \( Z_l \) by 10;
8: Conduct fusion to obtain the high-dimensional features by 11;
9: Compute the final representation by 12;
10: Update graph convolution parameters;
11: end for
12: return Prediction \( \hat{y} \).

A. Dataset

We evaluate the proposed EEG-based depression diagnosis method using a publicly available dataset provided by Mumtaz et al. [11]. The dataset consists of two groups of participants: 1) 34 patients with major depressive disorder (MDD) (mean age 40.33, SD = ±12.861), and 2) 30 age-matched healthy control (HC) subjects (mean age 38.227, SD = ±15.64). The study is conducted on outpatient participants at the Hospital University Sains Malaysia (HUSM) and record their EEG signals in the resting state with eyes closed (5 min) and eyes open (5 min). The experimental procedures of this study are approved by the HUSM Ethics Committee, and all participants involved in the study have a thorough understanding of the entire process and provide informed consent.

B. Preprocessing

In this study, data preprocessing uses the PyCharm software and the MNE tool. Original EEG signals are always accompanied by artifacts and noise, including electrooculogram (EOG), electromyogram (EMG), and noise related to data acquisition. A bandpass filter is applied to extract the frequency band of 0.5-30Hz before applying
feature extraction techniques. The independent component analysis (ICA) algorithm is then employed to further remove redundant artifacts and noise in the EEG signals, such as blinking and eye movement. The FastICA algorithm is used in this study. After these steps, clean EEG data is obtained, providing a basis for further analysis.

Considering the inaccuracy of signal boundary values, only the middle portion of the 120s EEG data is retained for analysis. To increase the number of samples and decoding accuracy, a cropping strategy is employed to process the data [28]. First, a sample set is established using all available EC and EO files in the dataset. For each data file, the 120-second data is segmented into 120 samples using a non-overlapping sliding window of 1 second in length, resulting in 240 samples for each participant. Ultimately, there are 8,160 and 7,200 samples for MDD and HC, respectively.

C. Implementation Details

We conduct all the experiments on the platform of NVIDIA GeForce RTX 3090 GPU. The proposed model is implemented using the PyTorch framework, a popular deep learning toolkit. The number of graph convolutional layers is set to 2, and the number of gate convolutional layers is set to 1. We use Chebyshev polynomials of order K = 3 for graph convolutions. The electrodes are divided into 7 and 2 regions, respectively. During the training process, we train the model for 200 epochs with a batch size of 256 and a learning rate of 0.01. The LG-GCN model is trained using Adam optimizer [29], which implements the stochastic gradient descent algorithm to update network parameters, weights, and model biases. Adam applies biases to each node in the graph. The training set and test set are set in a ratio of 9:1. Similarly, the validation set is established using 10% of the training set.

D. Evaluating Metrics

Due to the existence of false positive and false negative samples, using only accuracy to measure the performance of classifiers is far from sufficient. In previous studies [30], [11], [13], three typical performance metrics, accuracy, sensitivity, and specificity, are used to measure the performance of classifiers. Therefore, this study still chooses these three metrics to facilitate better comparison with other studies. These metrics can be calculated according to the following formulas 17-19:

\[
\text{Accuracy} = \frac{TP+TN}{TP+TN+FP+FN} \tag{17}
\]

\[
\text{Sensitivity} = \frac{TP}{TP+FN} \tag{18}
\]

\[
\text{Specificity} = \frac{TN}{TN+FP} \tag{19}
\]

where TP is the number of MDD patients detected correctly, FN is the number of MDD patients detected as healthy individuals, TN is the number of healthy individuals detected correctly, and FP is the number of healthy individuals detected as MDD patients.

E. Classification Performance

To demonstrate the superiority of the LG-GCN algorithm in depression recognition, we carefully select the most representative baseline models by comparing different methods. One category includes traditional machine learning algorithms, namely Support Vector Machine (SVM) [11] and Multi-Layer Perceptron Neural Network (MLPNN) [31]. Another category includes the most popular deep learning algorithms in the field of depression, namely CNN [13] and MDCNN [32]. The last category includes graph-based models. Since GCN has been applied less frequently to EEG-based depression recognition tasks, this paper only lists one of them, namely CN-GCN [19]. Table I presents the performance of LG-GCN compared to these three categories of baseline models on four frequency bands and all frequency bands (using $\delta$, $\theta$, $\alpha$ and $\beta$ bands together).

<table>
<thead>
<tr>
<th>Model</th>
<th>$\delta$ band</th>
<th>$\theta$ band</th>
<th>$\alpha$ band</th>
<th>$\beta$ band</th>
<th>all ($\delta$, $\theta$, $\alpha$, $\beta$)</th>
</tr>
</thead>
<tbody>
<tr>
<td>MLPNN</td>
<td>83.33</td>
<td>86.67</td>
<td>91.67</td>
<td>81.67</td>
<td>93.33</td>
</tr>
<tr>
<td>SVM</td>
<td>65.52</td>
<td>74.14</td>
<td>81.03</td>
<td>77.55</td>
<td>89.96</td>
</tr>
<tr>
<td>CNN</td>
<td>95.50</td>
<td>95.90</td>
<td>98.85</td>
<td>96.07</td>
<td>/</td>
</tr>
<tr>
<td>MDCNN</td>
<td>82.31</td>
<td>86.00</td>
<td>87.30</td>
<td>94.10</td>
<td>97.27</td>
</tr>
<tr>
<td>CN-GCN</td>
<td>78.09</td>
<td>80.39</td>
<td>78.15</td>
<td>96.68</td>
<td>99.29</td>
</tr>
<tr>
<td>Ours</td>
<td>84.36</td>
<td>88.67</td>
<td>91.44</td>
<td>96.90</td>
<td>99.30</td>
</tr>
</tbody>
</table>

Our proposed model demonstrates strong advantages in each frequency band. The best accuracy is achieved in the $\alpha$ and $\beta$ frequency bands. Furthermore, the performance when using all frequency bands together is superior to using the four frequency bands individually. This finding reveals that the overall structure allows for complementation and integration of information from each frequency band, resulting in more comprehensive information for depression recognition. This validates previous works [31].

To further validate the superiority of the proposed model, Table II shows the classification performance of our proposed model compared to existing methods on the same dataset. Mumtaz et al. [12][33][21] manually extracted features, including wavelet features, synchronous likelihood (SL) features, hemispherical asymmetry features, and frequency band energy features, and fed them into traditional machine learning models (LR and SVM) for classification. Mahato et al. [31][34] used frequency band energy and asymmetry features as input features for their classifiers, but their accuracy did not exceed 95%. Similarly, Saeedi et al. [11] considered different band powers and entropies. Dang et al. [32] combined multiple frequency band brain networks with deep learning algorithms, achieving an accuracy of 97.27%. In addition, the results of CN-GCN are comparable to the results of our proposed method, as it learns node features based on the topological connections of the brain network, rather than selecting local features between nodes. Existing studies on building brain networks based on GCN methods only consider the relationship between adjacent nodes, but ignore the activation relationship between brain regions. Soni et al. [35] fused information from three channels in the frontal lobe and used the KNN algorithm to achieve an accuracy of 92.80% in detecting depression. This finding is consistent with the results of this paper and previous studies, demonstrating that the features of depression patients are closely related to the activity in the frontal lobe of the brain.

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The information integration with IEM, results in a performance decrease in terms of Acc, Spe, and Sen.

Table II. Classification Performance of Our Proposed Method and Existing Studies on the Same Dataset

<table>
<thead>
<tr>
<th>Existing study</th>
<th>Year</th>
<th>Methods or features+Model</th>
<th>Accuracy</th>
<th>Sensitivity</th>
<th>Specificity</th>
</tr>
</thead>
<tbody>
<tr>
<td>Mumtaz et al.</td>
<td>2017</td>
<td>Wavelet features+LR</td>
<td>87.50</td>
<td>95.00</td>
<td>80.00</td>
</tr>
<tr>
<td>Mumtaz et al.</td>
<td>2017</td>
<td>SL, coherence,MI+SVM</td>
<td>94.70</td>
<td>98.30</td>
<td>91.40</td>
</tr>
<tr>
<td>Mumtaz et al.</td>
<td>2017</td>
<td>Asymmetry+SVM</td>
<td>98.40</td>
<td>96.66</td>
<td>100</td>
</tr>
<tr>
<td>Mahato et al.</td>
<td>2019</td>
<td>Alpha power,RWE+MLPNN</td>
<td>93.33</td>
<td>94.44</td>
<td>87.78</td>
</tr>
<tr>
<td>Mahato et al.</td>
<td>2020</td>
<td>Power, Asymmetry+SVM</td>
<td>98.96</td>
<td>86.00</td>
<td>89.92</td>
</tr>
<tr>
<td>Saiedi et al.</td>
<td>2020</td>
<td>Bandpower,ApEn+ E-KNN</td>
<td>98.44</td>
<td>100</td>
<td>97.10</td>
</tr>
<tr>
<td>Kang et al.</td>
<td>2020</td>
<td>Asymmetry Image+CNN</td>
<td>98.85</td>
<td>99.15</td>
<td>98.51</td>
</tr>
<tr>
<td>Dang et al.</td>
<td>2020</td>
<td>FDMB+MDCNN</td>
<td>97.27</td>
<td>97.22</td>
<td>97.35</td>
</tr>
<tr>
<td>Sun et al.</td>
<td>2022</td>
<td>Multilayer networks++ CN-GCN</td>
<td>99.29</td>
<td>99.37</td>
<td>99.32</td>
</tr>
<tr>
<td>Chen et al.</td>
<td>2022</td>
<td>Frequency matrix+DCTNet</td>
<td>99.15</td>
<td>99.30</td>
<td>99.01</td>
</tr>
<tr>
<td>Soni et al.</td>
<td>2023</td>
<td>Sparse graph network+KNN</td>
<td>92.80</td>
<td>/</td>
<td>/</td>
</tr>
<tr>
<td>Ours</td>
<td>2023</td>
<td>Local-Global graph+LG-GCN</td>
<td>99.30</td>
<td>99.41</td>
<td>99.17</td>
</tr>
</tbody>
</table>

In summary, the LG-GCN model outperforms other methods in depression recognition, with accuracy, sensitivity, and specificity of 99.30%, 99.38%, and 99.16%, respectively.

F. Ablation Experiment

To better understand the robustness and individual contributions of the local functional graph, global dynamic graph, and IEM in LG-GCN, ablation studies are conducted by removing these modules from the LG-GCN to demonstrate the effectiveness of the model. The results are shown in Table III.

Each module is removed one by one to demonstrate the effectiveness of the combination of multiple graphs. From the results in the first and second rows of the table, specifically, removing L from LG-GCN and not embedding the global graph led to a performance decrease in terms of Acc, Spe, and Sen. This fully verifies previous research that functional connections between local areas are important for discriminating different human emotional patterns. Hence, we adopt a graph coarsening method to aggregate local EEG features to make the model more robust. Completely removing G and directly performing graph convolution on L with full integration with IEM, results in a decrease in Acc from 99.30% to 97.90%, a decrease of 1.40% in Acc, 1.21% in Sen, and 1.59% in Spe, which highlights the importance of the global dynamic graph.

When the LG-GCN model lacked the IEM and the multi-graph representations were input directly into the GCN, flattened to the fully connected layer and softmax layer for depression recognition, the ACC dropped from 99.30% to 96.62%. This comparison demonstrates the effectiveness of IEM in achieving better performance. Additionally, it indicates that the discriminative features of the original EEG signal data can effectively complement the information captured by the LG-GCN, providing a more comprehensive representation for depression recognition. In conclusion, the information captured by multiple graph representations is crucial, and the ablation experiments strongly validate the necessity of constructing multiple graphs while demonstrating the benefits of extracting multidimensional information for model learning.

Table III. The Results of Ablation Experiments on Public Datasets Using LG-GCN

<table>
<thead>
<tr>
<th>L</th>
<th>G</th>
<th>IEM</th>
<th>Acc Changes</th>
<th>Sen Changes</th>
<th>Spe Changes</th>
</tr>
</thead>
<tbody>
<tr>
<td>×</td>
<td>√</td>
<td>√</td>
<td>97.52</td>
<td>-1.78</td>
<td>97.98</td>
</tr>
<tr>
<td>√</td>
<td>×</td>
<td>√</td>
<td>97.90</td>
<td>-1.40</td>
<td>98.20</td>
</tr>
<tr>
<td>√</td>
<td>√</td>
<td>×</td>
<td>96.62</td>
<td>-2.68</td>
<td>97.10</td>
</tr>
<tr>
<td>√</td>
<td>√</td>
<td>√</td>
<td>99.30</td>
<td>-</td>
<td>99.41</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>L</th>
<th>G</th>
<th>IEM</th>
<th>Acc%</th>
<th>Spe%</th>
<th>Sen%</th>
</tr>
</thead>
<tbody>
<tr>
<td>none</td>
<td>97.52</td>
<td>98.48</td>
<td>97.90</td>
<td></td>
<td></td>
</tr>
<tr>
<td>7 region</td>
<td>97.98</td>
<td>98.54</td>
<td>99.25</td>
<td></td>
<td></td>
</tr>
<tr>
<td>2 region</td>
<td>97.35</td>
<td>98.33</td>
<td>97.92</td>
<td></td>
<td></td>
</tr>
<tr>
<td>both</td>
<td>99.30</td>
<td>99.41</td>
<td>99.17</td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

Fig. 5. Performance comparison of different region partitioning methods.
The local functional graph includes two distinct partition types, one comprising seven brain regions, and the other comprising two hemispheres, to estimate whether there is a significant difference in depression recognition tasks under coarsening at different region scales. We remove the submodules of this local functional graph and present the results in Fig. 5. It can be observed that the performance of classification is lower when the number of regions is fewer. By introducing the 7 region and 2 region subgraphs, the network in the local functional graph enhances some key information features, improving the fitting ability of LG-GCN. This suggests that the functional connectivity between local regions is closely related to depression.

G. Interpretability and Visualization

In this section, to explain that significant connections mainly exist in certain brain regions, which have been found to be related to the pathology of depression. We visualize the distribution of degree centrality in the adjacency matrix of the global dynamics of the brain on four frequency bands in Fig. 6, providing an intuitive mapping of brain functional connections to the brain’s topological structure. The degree centrality evaluates the connection strength of a node with the other nodes, which has been widely used to measure the importance of the nodes in a graph [36], calculated as follows:

\[
C_i = \sum_{n=1}^{19} A_{in} + \sum_{m=1}^{19} A_{mi} - 2A_{ij} (i = 1, ..., 19)
\]  

(20)

Based on the distribution of significant electrodes, we can identify brain regions that are favorable for EEG-based depression identification. This finding explains the results in Table II, where the model performs better on the \(\alpha\) and \(\beta\) bands, and the connectivity is stronger than in other bands. Especially in the frontal and temporal lobes, as the frontal and temporal lobes are highly related to the onset of depression, the centrality degree is very high. Our study is consistent with [20][21], which found that the coherence of activity in the frontal and temporal regions in depressed patients is significantly higher than in healthy controls. This suggests that considering the connections between local EEG channels is crucial for accurate depression identification.

To display the connections between nodes, Fig. 7 depicts the top five connections learned by the proposed model in the adjacency matrix on the four frequency bands (\(\delta\), \(\theta\), \(\alpha\) and \(\beta\) bands). Unlike the heatmap plot of degree centrality distribution, we remove the diagonal elements and just plot the connections between nodes. The positions of the 19 electrodes are displayed around the circles in the subplots. The lines represent the connections between channels, with darker colors indicating stronger connectivity. It can be observed that the critical connections are primarily located in the frontal and temporal lobes, where nodes corresponding to electrodes in these regions have a higher representation in the brain. This indicates the crucial role of the frontal and temporal lobes in the recognition of depression. In addition, two electrodes of multiple electrode pairs belong to different brain functional regions, indicating that the correlation between these regions not only has local structural connections but also has functional connections from global electrode channels.

Fig. 6. Degree centrality distribution of the learned global adjacency matrices by LG-GCN over four frequency bands. (a) \(\delta\) band: 0.5-4 Hz (b) \(\theta\) band: 4-8 Hz (c) \(\alpha\) band: 8-13 Hz (d) \(\beta\) band: 13-30 Hz.

Fig. 7. Visualization of the top five functional connections between EEG channels in the learned adjacency matrices over four frequency bands. (a) \(\delta\) band: 0.5-4 Hz (b) \(\theta\) band: 4-8 Hz (c) \(\alpha\) band: 8-13 Hz (d) \(\beta\) band: 13-30 Hz.

V. DISCUSSION

In this research, we are thoroughly validating the effectiveness of LG-GCN. By partitioning the brain regions into local and global divisions, we discover high-level connections between different brain areas. Through the fusion of locally functional maps from prior research and adaptive adjacency matrices, our graph convolutional model accurately discerns the correlation between brain regions and depression. The fusion of local-global features is playing a crucial role in enhancing the representational power of the features. LG-GCN demonstrates a palpable edge over existing methodologies in data processing and feature extraction. Furthermore, our model evinces exceptional interpretability, elucidating the saliency accorded to specific brain regions and connections. Nodes situated within the frontal and temporal lobes exhibit heightened self-loop weights, indicative of their preeminent roles in the network, forging robust connections with other nodes. This, in turn, culminates in superlative depression identification capabilities.

VI. CONCLUSION

In this work, we propose LG-GCN, a local-global graph representation that simultaneously explores local brain functional connectivity and global dynamics, constructs the connections between different brain regions with prior knowledge in neuroscience research, and aggregates them into multi-layer GCNs to capture hierarchical dynamic graph topology spatial relations. In addition, IEM enables the proposed model to adaptively include discriminative depression features while retaining high-dimensional information. Extensive experimental results and visualizations demonstrate that the accuracy, sensitivity, and specificity of LG-GCN on public datasets are 99.30%, 99.41%, and 99.17%, respectively. All of these are superior to existing SOTA research, which indicates enormous potential for decoding depression based on EEG signals. However, a challenging issue in EEG-based depression recognition is the inter-individual variability of brain signals. In future work, it
is crucial to take this factor into account and construct a common graph structure that is independent of individuals, thus enhancing the generalizability of the model.

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University’s Service Delivery Improvement Through a DSS-enabled Client Feedback System

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Abstract—The expansion of products and services on a global scale demands the improvement of an organization's performance. In addition to addressing the challenges of improving product and service delivery, companies must focus not only on meeting customer expectations but also on surpassing them. Consequently, valuing the opinions of clients, giving the best client experience, and measuring client satisfaction are deemed vital not only for the company's survival but also for gaining a competitive edge for the organizations in the wired communities. It is because of these premises that the Client Feedback System was developed in this study for the university's service delivery improvement. This system captured the results of the Client Satisfaction Survey for School Year 2015-2016 to School Year 2020-2021. Interpretation of these captured data were made and action for the improvement of service delivery for each department in this university was recommended using the Decision Support System (DSS) technique. The system was created using the Rapid Application Development (RAD) method and utilized various software and technologies such as HTML, CSS, and JavaScript for the front-end development, MySQL and PHP for the back-end, and Apache as the local server of the system during its development and pilot testing.

Keywords—Decision support system; client satisfaction; client feedback system; rapid application development; service delivery improvement

I. INTRODUCTION

Technological innovation posed an important driver of competitive success for most organizations nowadays to compete successfully and participate in the globalization of markets. Because of this, businesses are under pressure from both domestic and international competition to continuously innovate in order to produce differentiated goods and services.

Many leading companies worldwide place a strong emphasis on prioritizing exceptional customer experiences. They go beyond merely meeting customer expectations and strive to surpass them. Indeed, a world-class performance outcome is necessary so that clients and customers would be satisfied with what the company offers. As a result, it is crucial to evaluate client satisfaction. Client satisfaction describes a customer's perception of the company's quality, value, and expectations. Client satisfaction surveys, a tool used by businesses to gauge how satisfied their clients are with their goods and services, can be used to quantify customer satisfaction [1]. With this system, recognizing dissatisfied clients is just as crucial as identifying those who are really satisfied.

More so, evaluating the satisfaction of customers/clients is needed to understand what the customers expect from the product or service that the company provides. In addition to helping the business retain more customers, doing so enables it to spot and address any critics. A satisfaction survey would allow for the gathering of useful consumer feedback that could be used to ensure total customer experience and satisfaction.

Five benefits of customer satisfaction were presented by Bernazzani [2]. Accordingly, customer satisfaction (a) helps the organization understand where it excels; (b) aids the company in recognizing areas for improvement; (c) leads to higher customer loyalty and advocacy; (d) increases customer retention and reduces stir-up and, (e) increases the client lifetime value. Evaluating customer/client satisfaction indeed can make the organization understand what and where the business is doing well, what business strategy is to be done if the improvement is necessary and what ways could be done in order for the clients to refer new leads to the company and generate more testimonials for their experience. Likewise, satisfied clients will not seek out competition if the company keeps them happy. Customer happiness, therefore, is critical for improving service metrics and ensuring that the firm provides the greatest possible experience.

Assessing client satisfaction was very much imperative in almost all facets of business and service industries such as logistics, food industries, financial institutions, construction industries, hospital general, maternal, and senior health care as well as in educational sectors. Further, businesses and government agencies alike throughout the nation are concerned about the value of exceeding customer expectations. This is embodied in the Philippine Development Plan of the National Economic and Development Authority [3]. The government wants to provide the impression that it is constantly working to make its services cleaner, more effective, and more focused on the needs of its citizens. Similarly, the Civil Service Commission's directives and the Red Tape Act of 2007 [4], include provisions for gathering client satisfaction feedback through surveys.

Higher education from across the globe has long recognized the value of DSS by which it combines data and intelligence, generates the most accurate and likely options and explanations, and fine-tunes uncertain decisions and conclusions, particularly for the improvement of the school’s performance [5]. Setting up of DSS is deemed appropriate and well suited for the school’s educational mission, cutting-edge research, advanced data collection strategy, and academic activities that are provided to society such as its extension.
programs. Hence, the use of DSS in higher education’s Information and Communication Technology structures would be a cost-cutting measure for the institution and would enable the school to concentrate on the most important issues of drafting and understanding the best decisions, as higher educational systems delegate involvement.

Likewise, client satisfaction surveys are also carried out at State Universities and Colleges (SUCs) around the nation, just like they do at other government organizations. Several studies on client satisfaction feedback were conducted in academe that had given insights to the researchers to embark on this topic. St. Paul University Philippines (SPUP) conducted customer satisfaction surveys (CSSs) to assess how well it has succeeded in providing quality services and products both to its internal and external customers. This is a routine activity conducted by the university in its quest for standards of quality and excellence in its delivery of service to clientele. This school evaluates the graduates of post-baccalaureate degrees on their satisfaction with respect to the quality of services provided to them by its academic and non-academic support staff, learning support facilities, learning experience, and research-related matters [6]. Likewise, the study of Ventayen and Orlanda-Ventayen [7] regularly measures the satisfaction rating of the students on the Open University Systems in terms of timeliness, access, convenience, and staff attitude. The results of these surveys and actions taken by the institutions are being validated when they apply for quality assurance evaluations, like the Accrediting Agency of Chartered Colleges and Universities in the Philippines (AACCUP) Accreditation, International Standards Organization (ISO), and Institutional Sustainability Assessment (ISA).

Thus, for upholding the Catanduanes State University (CatSU)'s tradition of excellence in service delivery, the researchers conducted this study to assess the effect of client satisfaction evaluations on the performance of the university's offices and departments and improve service delivery using decision support systems (DSS). A computer-based program called a “Decision Support System” (DSS) gathers, organizes, and analyzes data to support good managerial, operational, and planning decision-making. Organizations can identify issues, find solutions, and take choices with the use of DSS analysis [8]. A DSS captures and analyzes data, synthesizing it to produce comprehensive reports. The organization then uses these reports to help in decision-making for the improvement of its goods or services [9].

The following industries use DSS: (a) retail stores to plan strategies to boost sales and advertise their goods; (b) banking institutions to analyze financial data asset reports and income statements; hospitals to improve the workflow of their key medical tasks; (c) farming businesses to plan effective crop-planting procedures; and (d) manufacturing industries to plan out efficient production plans and processes. Universities and academic institutions are also using DSS in order to identify the best course alternatives and provide appropriate amenities, including housing or eating options, and track the number of registered students [10]. Because they help university administrations better plan for student-related expenses, these programs may be especially useful when they create a yearly budget. A DSS system can also be used by administrators to assess students' academic progress, redesign a course's syllabus, or choose which on-campus services to give the highest priority.

Further, using DSS in Higher Educational Institutions (HEIs) resulted in reduced manual work, better data analysis, and ease of decision-making process, thus it helped increased productivity, employee and customer satisfaction, and revenue and profitability. With the COVID-19 pandemic, DSS helped improved the e-learning process by combining students with similar learning styles and assessing student profiles such as gender, age, the number of hours spent on a course, average time spent per week, and the like [11]. Additionally, the use of web technologies and user experience (UX) in enhancing CRM Business Intelligence applications were considered important factors in the study of Gharibeh [12]. By leveraging web technologies and optimizing the UX, HEIs can improve their business intelligence implementation, leading to more profitable customers and reduced costs.

Given these assumptions and the significance of implementing DSS in HEIs, this study, therefore, developed an automated web system for collecting information on client satisfaction survey results, performing data analysis, and providing comments or recommendations for improving the university's service delivery using decision support system (DSS). The system developed in this study captured client satisfaction survey results from School Years 2015-2016 to 2020-2021. It likewise stored, organized, and analyzed the data gathered in a database system. From the interpreted result, the system presented a corresponding decision and recommendation for the improvement of the university’s service delivery to its clientele.

II. RELATED WORKS

Continuous performance evaluation of an organization enables management to determine if anything has improved or worsened the way clients see a company’s business. Meaningful reward and recognition systems can only operate in businesses where there is an accurate and visible process of performance feedback and discussions. The integration of a Decision Support System further improves business processes by recommending valuable actions based on actual data from client satisfaction surveys. The following were the studies reviewed relative to the use of such system. Articles in relation to the use of Decision Support Systems specifically in client satisfaction research were scarce during the time of gathering related articles of this study. Hence, articles relative to Decision Support System, in general, were considered in this paper.

A decision support model was proposed in the study of Dweiri, Kumar, Khan, & Jain. Their model was for supplier selection in the automotive industry. The model was based on the analytic hierarchy process (AHP). The criteria considered in the model were price, quality, delivery, and service. Sensitivity analysis checked the robustness of the decision using Expert Choice software. The suppliers were selected and ranked based on sub-criteria. Sensitivity analysis suggested the effects of changes in the main criteria on the suppliers ranking [13]. The use of AHP in supplier selection gives the decision-
maker confidence in the consistency and robustness of the developed system throughout the process.

A platform that utilized the Decision Support System approach had also been done by Yazdani, Zarate, Coulibaly, & Zavadskas [14]. Their paper proposed a decision support system for selecting logistics providers based on the quality function deployment (QFD) and the technique for order preference by the similarity to the ideal solution (TOPSIS) for the agricultural supply chain in France. The proposed model looked at the decision problem from two points of view which were technical and customer perspectives. The main customer criteria that were considered were confidence in a safe and durable product, emission of pollutants and hazardous materials, and social responsibility while the main technical factors were financial stability, quality, delivery condition, and services. The outcome of this research was a group decision-making system that put into account decision-makers and customer values to aid agricultural partners and investors in the selection of third-party logistic providers. Likewise, the fuzzy linguistic variables enabled to assist agricultural parties in uncertain situations. The integrated decision support system overall enhanced the quality and reliability of the decision-making of the agricultural supply chain.

Similarly, a decision support model was designed by Kucukaltan, Irani, & Aktas for the identification and prioritization of key performance indicators in the logistics industry. The authors designed a stakeholder-informed Balanced Scorecard (BSC) model with the use of the Analytic Network Process (ANP) method to identify key performance indicators as well as various stakeholders in the logistics industry and analyzes the interrelationships among the indicators. The results show that educated employee (15.61%) is the most important indicator of the competitiveness of logistics companies [15].

While the reviewed articles on decision support systems in industries and logistics services demonstrated positive outcomes in enhancing service delivery particularly in logistics services and automotive industries, no article was found and reviewed on client satisfaction survey systems utilizing the DSS approach specifically for higher educational institutions. The integration of this approach to business processes, however, instills greater confidence in decision-makers, as it enables them to make informed decisions based on accurate data as processed by the system. This study, therefore, filled this gap by developing a DSS-Enabled Client Feedback System specifically designed for implementation at Catanduanes State University. The system ensured the accuracy and reliability of suggestions or recommendations by utilizing actual data from client satisfaction surveys conducted every semester.

III. METHODS

This study employed the systems development method and documentary analysis. System development is a process of developing a software or application that will answer the user’s needs, particularly on client satisfaction feedback that employed the DSS concept [16], [17]. In their study on the suitability of agile development methodologies for Decision Support Systems (DSSs), Gharaiheb & Abu-Soud discussed the use and effectiveness of agile development methodologies [18]. Being a progressive agile method, the researchers specifically employed the Rapid Application Development model in developing the application for this study.

The results of a customer satisfaction survey were recorded and used in a documentary analysis for the academic years 2015-2016 through 2020-2021. System development was utilized in the conceptualization, prototyping, and actual development of the system. The system created and recorded the customer satisfaction survey results for the relevant academic year and provided recommendations for the college’s or department’s service performance.

![Fig. 1. RAD model.](image)

The development approach used in the study adopted the Rapid Application Development method of Requirements Planning, User Design (iterative method of Prototype, Test, and Refinement), Construction, and Cutover [19], [20], [21], [22]. However, the steps were customized to the actual processes undertaken during the development of the system. Fig. 2 illustrates the activities undertaken in the actual development of the system in this study that was adopted through the RAD model presented in Fig. 1. It could be gleaned from the figure that seven (7) steps were carried out to conceptualize present the finished product and have the user accept them. These steps were: (1) Conceptual Modeling; (2) Gathering and Analysis of Survey Results and User’s Requirements for the system; Step 3 is an iterative process of (3-a) Designing of Logical Schema and Developing of Prototype; (3-b) Refinement of the Design and Systems Design; (3-c) Interface Design and (3-d) Database Design; (4) Pilot Testing of the New System; (5) Systems Installation Implementation and Deployment; and (6) User Training.

![Fig. 2. Step-by-step approach in the development of the system.](image)

The process flow of the system is shown in Fig. 3. From the figure, the client satisfaction survey results were then put through statistical testing using frequency counting, weighted means, and percentages. To establish the system-wide mean for each customer evaluation, these tests were included as part of the system’s process. The system calculated the grand mean for
the client satisfaction rating received per semester from all the client evaluations, and from there it calculated the grand mean for each indication or parameter. The method derived an interpretation from the grand mean based on the study's 5-point Likert Scale. The grand mean result was interpreted using DSS. The system would specify a general impression. Likewise, the Lowest Rating and Highest Rating for each parameter would be specified while highlighting the department's or college's service strengths and flaws. The suggestions/recommendations made by the system through its DSS notion would be taken into consideration by a department when deciding what to do or for some corrective actions to make.

In addition, the process of capturing, data analysis, and providing recommendations for the system as shown in Fig 3 could be interpreted as follows: (a) it starts with entering the client satisfaction survey results through the user interface, which would be automatically stored in the Database. These results would serve as the knowledgebase of the DSS. The knowledgebase is the information used as basis in the interpretation and implementation of DSS; (b) The system would then analyze this information and then obtain the overall mean for each parameter and the grand mean for each semester; (c) Through the data analysis results, the system would be able to make an interpretation and its DSS part would be recommending actions for the service delivery improvement of the unit. The areas of application of the DSS module and basis of interpretation for the results of data analysis are shown in Fig. 4 and Fig. 5, respectively.

The developed system was flexible, maintainable enough, and could easily be migrated to another environment. The system was implemented in CatSU’s College of Information and Communications Technology (CICT) for pilot testing purposes. Positive responses and feedback were obtained from end-users; hence, this was recommended to be utilized in the entire university.

IV. RESULTS

The problem proffered in this study was about the development of the Decision Support System for the client satisfaction survey feedback results for bettering the offered services of the departments in the university. As mentioned in the Methods section of this article, the development of the system followed the Rapid Application Development (RAD) approach as specified in Fig. 1. This software development model suggests constant meetings with the end-users during the development process. Before the actual development of the system, prototypes were designed and presented to the end-users. Modifications and refinements were done with the prototype to suit the needs of the users. The actual development of the system commenced after the prototype and its refinements were approved by the users.

Fig. 6 below shows the integration of the DSS and its components into the system developed in this study.
Input to the system was the client satisfaction survey results of the unit through the user interface, which could be done by the unit clerk. These inputs were analyzed by the system to provide the necessary data for the DSS such as the grand mean. The DSS was then implemented to provide suggestions/recommendations for the service delivery improvement of the unit. The DSS module was based on the development framework by Juneja which was comprised of four (4) basic stages, namely Intelligence, Design, Choice, and Implementation [23]. The following are the stages undertaken in the development and integration of the DSS into the system:

Stage 1: Intelligence. The DSS module started after the data analysis of the system. Fig. 8 shows the components of the Intelligence stage which were the Problem Identification (Areas for Improvement) and DSS Objectives and Resources. The DSS was designed to identify the areas where the unit has the highest and lowest rating. The areas identified were then given interpretations based on what the unit needs. The areas were also categorized into two, as shown in Fig. 5, which were “The Office” and “Frontline Employees”. This is to ensure that the recommendations and interpretations given by the DSS were specific to its areas. Fig. 7 shows an actual screenshot of the system which shows the lowest and highest rating and its corresponding interpretation and recommendation for service delivery improvement.

Stage 2: Design. The design stage of the DSS development was broken down into two components, namely System Design and System Structure, as shown in Fig. 9. System Design included identifying all the necessary technical requirements for the DSS and finalization of the User Interface (UI). This was also where all possible courses of action or recommendations for service delivery improvement were established. System Structure, on the other hand, includes the planning of the development and integration of the DSS into the UI. Having the system browser-based (run using a browser such as Chrome, Opera, Firefox, and the like), integrating the DSS required programming in the back-end using scripting languages such as PHP and JavaScript.

Stage 3. Choice. Once all the possible actions or recommendations were identified, integration of DSS took place in this stage. Sets of recommendations were stored in the Database for easy retrieval. The DSS used the client satisfaction survey results and its grand mean to identify the areas with the lowest and highest ratings. The system then provided a general impression of the grand mean. It also provided the best possible actions to better the unit’s service delivery by providing its interpretation and recommendations for the areas with the lowest and highest rating, as shown in Fig 8. Fig. 10 shows the actual screenshot of the system showing the grand mean of the unit’s results together with the general impression provided in the system.
Stage 4. Implementation. The system was deployed in the CICT for pilot testing. During the pilot testing, comments, suggestions, and recommendations for the improvement of the system were sought from the faculty and were integrated into the final deployment of the system for the university. The screenshots of the developed system are shown below. Fig. 11 displayed the login process.

Fig. 11. System’s login page.

Fig. 12 displayed the webpage for obtaining the Client Satisfaction Survey Results for a particular semester and school year. Likewise, Fig. 13 displayed the overall results of data analysis and a summary of ratings for the Office and Frontline Employees. Lastly, Fig. 14 displayed the system’s dashboard.

Fig. 12. Webpage for obtaining the client satisfaction survey results.

Fig. 13. Results and summary of ratings.

Fig. 14. System’s dashboard.
V. CONCLUSION AND RECOMMENDATIONS

The study captured data relative to Client Satisfaction Survey Results for School Year 2015 -2016 to School Year 2020-2021, developed, deployed, and tested the system. The system does interpretations as to the overall mean for each Client Satisfaction Indicator and computed for the grand mean. From the grand mean, a general impression and recommendation as to better the service for a certain department in this university. Employment of DSS enables the system to recommend action to do for the improvement of the service delivery of each department in this university. For each Client Survey Indicator, the system is also recommending action to do. Recommended action of the system served as a guide for the improvement of the delivery of the service for a department in this university. The developed system was pilot tested in one of the colleges of the university and the positive results of the pilot testing enabled the university to adopt it. Adoption of the system as the university’s gauge to evaluate a department’s performance as perceived by the client had brought an advantage to the university’s performance in its external performance evaluators.

The following recommendations were drawn for this study:
1) Valuing the opinions of customers, giving the best customer experience, and measuring customer satisfaction, especially with the utilization/integration of Information and Communications Technology and the DSS principle must be among the top-priority of every government office in this country to constantly improve their manner of service delivery;
2) The system developed should be utilized in full for each department since the system would be of help in the betterment of the service delivery of the entire university; and
3) The recommended actions by the system developed for improving the service delivery of each unit in the university should be put into life (action) to ensure a total client satisfaction or perhaps even exceeding expectations of clients.

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REFERENCES


Development of a Two-dimensional Animation for Business Law: Elements of a Valid Contract

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Abstract—Elements of a valid contract is an important topic in corporate law. Since there are so many elements and related case studies, some students have difficulty remembering all the elements. Therefore, an animation will contribute to explaining all the elements in simpler terms, help the students remember the relevant case studies, and help lecturers teach students in an easier and more interactive way. An investigation on a 2D animation design and its effectiveness for corporate law and commerce learning is presented in this article. This paper aims to examine the 2D animation principle in animated explainer videos. In addition, the objective of this research is to develop an animation and evaluate the effectiveness of the 2D animation for Business Law teaching and learning. A comprehensive analysis of the 2D animation used in Business Law learning which focusing on spreading the importance of student understanding and motivation in Business Law course using a 2D animated approach is the expected outcome of this paper. The project collaborates with the Department of Commerce, Politeknik Melaka, Malaysia, for content expertise and testing. The Multimedia Production Process is the methodology used for the development of this research work, and the ADDIE Model is applied for the instructional design. The application is developed using Adobe After Effects, Adobe Premiere Pro, Adobe Media Encoder, and the Audacity platform. The contribution of this study is obvious, as the resulting outcomes can be used as guidelines for best practises of learning styles. The implications of this study will impact teaching and learning and increase understanding. This research is expected to improve teaching delivery while also increasing user understanding and motivation to learn.

Keywords—2D animation; business law; elements of a valid contract; teaching and learning; multimedia

I. INTRODUCTION

Elements of a valid contract is one of the many topics delivered in corporate law that every business and commerce student should be aware of. It has a lot of key factors and a number of case studies that need to be related to each of the elements [1], making it harder for students to understand and memorise each element and their relevant cases. It also gets progressively harder for lecturers to provide a simpler way for their students to understand the topic. This work is to develop a 2D animation that acts as a medium for business students and lecturers to effectively remember relevant cases based on the elements of a valid contract. Elements of a valid contract is a crucial topic in corporate law that every business student must understand and memorise. Due to the many elements and relevant case studies [3, 4], some students are having a hard time memorising all the elements. This animation that will be delivered will focus more on explaining all the elements in simpler terms, helping them remember the relevant case studies, and helping the lecturers teach the students in a simpler and more interactive way.

This animation provided simplified descriptions of the Elements of a Valid Contract to facilitate the retention of important case studies among business law students and to assist lecturers in delivering interactive and comprehensive lessons. It is envisaged that the proposed work would assist in providing an effective teaching delivery to the students and lecturers in the form of 2D animated video.

The organisation of this paper is as follows: The existing system and related work on Business Law and two-dimensional animation are presented in Section II. Section III defines the overall methodology of this research work in detail. The implementation of the research work is presented in Section IV, while Section V presents the discussion. Finally, Section VI presents the conclusion and future work.

II. LITERATURE REVIEW

Elements of a valid contract consist of six essential elements, which are offer, acceptance, consideration, intention of making legal relation, certainty of the contract, and capacity and legality [1]. All these elements make up one valid contract.

Meanwhile, animation is a technique that manipulates figures to make them appear to move. Most animations today are created using computer-generated imagery, which has allowed multiple storytellers to deliver their stories in a more creative and fun way. Visibly, it provides a whole new method of expression and innovation, but on a more practical level, the evolution of movement attracts more attention than static images [2]. Visual designers can expand their creativity far beyond what the world allows them to do through liveliness. They can create sketches that draw a crowd as well as sketches that help the group understand the world better. Other researchers have presented materials that can assist in law education. For example, Boulton vs. Jones case is presented in [3], while the case between Hyde and Wrench is presented in [4]. The case between David and Noorazman is presented in Malay Law Journal Reports [5], also Balfour v Balfour case is reported in [6]. Other cases, such as Karuppan Chetty v Suah Thian and Tan Hee Juan v The Boon Keat are reported in [7] and [8], respectively. Meanwhile, there are a lot of research works reporting on the animation usage, mainly for educational purposes. The development and evaluation of a 2D animated for therapy of verbal apraxia have been...
presented in [9] and [10]. An interactive content development using a 2D animation for depression awareness among tertiary students is explained in [11]. Meanwhile, the usage of animation in education and report generation are presented in [12] and [13]. A related domain to this proposed work which is the production process of an animated explainer video is discussed in [14].

Also, there have been several literature reporting the development of law education resources. Three existing systems will be discussed here: the David Jaroszewski videos on Youtube, Malaysia Company Law: Principles and Practices, and Introduction to the Law of Contract.

David Jaroszewski is a lawyer who practises law in Texas and has 40 years of community college teaching and leadership experience as a classroom chair, and dean. He opened his YouTube channel back in January 2013 to make videos about various legal topics and teach the community about the basic concepts of each topic [15]. Fig. 1 shows a screenshot of the video.

The Malaysia Company Law: Principles and Practices, 3rd Edition, provides a detailed analysis of the core company law principles, incorporating amendments in 2019 to the Companies Act 2016. The materials cover an extensive number of reported cases with extracts of significant dicta and relevant statutory provisions, combining the discussion of the Companies Act 2016 and the previous Companies Act 1965 [16]. Together with the thoughtful use of tables, diagrams, and flowcharts, this publication enables readers to engage in a critical examination of the law as it is and as it ought to be. Fig. 2 shows the cover page of the Malaysia Company Law book.

Introduction to the Law of Contract by Sarah Field is a user-friendly book that provides a comprehensive, clear, and straightforward account of what is required for the formation of a valid contract [17]. Fig. 3 shows the cover page of the Introduction of the Law of Contract.

Fig. 1. Elements of a contract.

Fig. 2. Malaysia Company Law: Principles and practices.

Fig. 3. Introduction to the law of contract.

III. METHODOLOGY

Multimedia Production Process is the methodology used in this research work. This model was chosen for this research work because it addresses the use of different media, such as images, text, animation, audio and video, to enhance the impact of message. The goals and objectives of the research work, as well as the intended audience, are all decided during development. Basic aims, objectives, and matrix of an activity would be constructed after settling on a multimedia project theme. Specific aims or general assertions of desired work outcomes are often global in multimedia works, whereas the objectives of work target adding customer association to a two-dimensional animation using the approach of digital animation.

The first phase is pre-production, which includes concept creation, design boards, storyboards, script writing, and character creation. The concept is the first item that is generated. To create a script, everyone involved in the work will explore different ideas for the animation. Before getting too ambitious, it is critical to understand the limitations of the animation and programme used. The animation’s authors will write a narrative that will allow the animation to establish a storyline. A script is the foundation of an animation film, and it will serve as a point of reference throughout the animation production process. Captions that summarise the story's outline will be included in the storyboards. This aids in camera location as well as animation timing. It also aids in making decisions about what works and what does not.

The production phase comes next, which entails creating and implementing the storyboard into a multimedia work. During the production phase, the necessary hardware was installed, assets were transferred to the unit, also code was completed, among other things. Multimedia element utilised will be implemented, and flowcharts also storyboards developed on this work will determine the technology to be used, as well as an existing system analysis and the current system limitation, all of which are the requirement part.

Lastly, there is the post-production stage, where the video's final appearance is determined. The full process of developing and modifying sound mixing, background music, and visuals, including lower thirds, is referred to as postproduction. When the video is finished, the multimedia program goes to alpha testing and beta testing. After it has
been tested and updated, the developed application transfers to the stage called packaging. It may be available through a DVD or CD-ROM, or could be made available on the Internet and shared on sites like YouTube.

A. Analysis

A YouTube video called ‘Elements of a Contract’ made by David Jaroszewski and the Malaysia Company Law book will be the inspiration for this animation work. The reason for this is that their content is very simple to understand but lacks graphic content to assist students who are visual learners in memorising and understanding the context of the video. Table I presents the comparison of existing systems.

It can be concluded from Table I that various applications were proposed in order to produce an efficient and interactive learning tool. However, these applications still have lacunae or spaces for improvement. The contents of the listed research works, for example, are delivered in non-interactive video and manual books. In addition, the usage of multimedia elements is not fully implemented in those applications. Therefore, this proposed research implements a 2D animation approach for delivering the contents. The implementation of animation is necessary to overcome the limitations of the current delivery method. The proposed techniques in this research will most notably benefit the realm of multimedia in a number of areas or ways that include the provision of an interactive method for effectively displaying complex contents in the commerce and law fields.

The project specifications look at the framework that will be built. It will detail the actions, procedures, and other requirements that must be met by the work. The specifications will then be judged according to the gathered requirements also the project's methodology. One of the requirements of the project is a clear task understanding that must be accomplished.

The gathering requirements is among the most vital step for the process of data organisation and transmission. An interview was performed with a corporate law lecturer from Politeknik Melaka, Malaysia as the subject matter expert. All the work's features, including proposed interactions, source analysis and raw data were addressed during requirement collection. This work's basic methodology will be investigated as well. The study's findings will be used to further the field of animation.

The duration of this proposed application is about five minutes to help the students understand the content of the video even better. The storyline of the work is that there would be a businesswoman who would be telling the relevant cases in a storytelling way to make it easier to understand. All the characters and objects will be designed. As a result, during the analysis process, the researcher should conduct requirements consumer, target user, and stockholder. For this work, the researcher interviewed a lecturer from a local college who teaches that specific subject. The interviewee's information is listed in Table II.

B. Design

During the design phase, the animator will create and structure the animation. The animator and viewer will be able to get a general idea of the project interface by sketching the concept, layout and design. This phase also goes over storyboard design which is necessary for animation because it serves as a reference until the production process is completed. This stage is crucial for determining all the research work’s specifications.

In delivering the message to the viewer, characters play an important role. If the character fails to do so, the story will turn into ineffective and dull. Adobe Illustrator was used to create the design for better quality. Amongst the important character profiles designed for this work are businesswoman, defendant, plaintiff, third party, Mrs. Balfour, teenager, land, money, housing estate, handshake for agreement, goods, contract and court.

<table>
<thead>
<tr>
<th></th>
<th></th>
<th></th>
<th></th>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td>Interactive</td>
<td>Non-Interactive</td>
<td>Non-Interactive</td>
<td>Non-Interactive</td>
<td>Non-Interactive</td>
</tr>
<tr>
<td>Types of Product</td>
<td>Video explainer</td>
<td>Book</td>
<td>Book</td>
<td>2D animation</td>
</tr>
<tr>
<td>Interface</td>
<td>Simple</td>
<td>Basic</td>
<td>Basic</td>
<td>Attractive and Simple</td>
</tr>
<tr>
<td>Sound</td>
<td>Good audio quality</td>
<td>-</td>
<td>-</td>
<td>Background music and storyteller</td>
</tr>
<tr>
<td>Price</td>
<td>Free</td>
<td>Paid</td>
<td>Free</td>
<td>Free</td>
</tr>
<tr>
<td>Scope</td>
<td>General</td>
<td>Students/Lecturers</td>
<td>Students</td>
<td>Students</td>
</tr>
<tr>
<td>Language</td>
<td>English</td>
<td>English</td>
<td>English</td>
<td>English</td>
</tr>
<tr>
<td>Ease of Use</td>
<td>Yes</td>
<td>Yes</td>
<td>No</td>
<td>Yes</td>
</tr>
<tr>
<td>Strength</td>
<td>Understandable and simple</td>
<td>Detailed explanation</td>
<td>Detailed explanation</td>
<td>Good visual design, appealing animation with background music</td>
</tr>
<tr>
<td>Limitation</td>
<td>No visual help</td>
<td>Long and complicated sentences</td>
<td>Long and complicated sentences</td>
<td>-</td>
</tr>
</tbody>
</table>

TABLE I. COMPARISON OF EXISTING SYSTEM

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### TABLE II. CONTENT VERIFICATION FORM

<table>
<thead>
<tr>
<th>Component</th>
<th>Details</th>
</tr>
</thead>
<tbody>
<tr>
<td>Name:</td>
<td>Madam Hazira binti Saleh</td>
</tr>
<tr>
<td>Company Name:</td>
<td>Politeknik Melaka</td>
</tr>
<tr>
<td>Position:</td>
<td>Principal Lecturer</td>
</tr>
</tbody>
</table>

### C. Implementation

This part details the media creation, where it will cover the processes of media creation and media component editing before incorporating them in the media integration section later. Audio, graphics, text, and animation will be produced as part of this process. The storyboard of all the modules and the characters profiles are presented below in Tables III and IV, respectively.

### TABLE III. STORYBOARD OF ALL MODULES

**Scene: Main Title**  
Description: Business Law: Elements of a Valid Contract (Relevant Cases)  
Sound: Background Music  
Time: 6 seconds

**Scene: Module 1**  
Description: Offer definitions  
Sound: Background Sound

**Scene: relevant case 1**  
Description: Boulton v Jones  
Sound: Background Sound

**Scene: relevant case 1**  
Description: continuation of the case and case issue  
Sound: Background Sound

**Scene: Module 2**  
Description: Acceptance definitions  
Sound: Background Sound

**Scene: relevant case 2**  
Description: Wrench v Hyde  
Sound: Background Sound

**Scene: relevant case 2**  
Description: case issue and why the contract invalid  
Sound: Background Sound

**Scene: Module 3**  
Description: Consideration definitions  
Sound: Background Sound

**Scene: relevant case 3**  
Description: Wong Hon Leong v Noorazman Adnan and case issue  
Sound: Background Sound

**Scene: relevant case 3**  
Description: Wong Hon Leong v Noorazman Adnan and case issue  
Sound: Background Sound
<table>
<thead>
<tr>
<th>Scene: Module 4</th>
<th>Description: Intentions to Create Legal Relations definitions</th>
<th>Sound: Background Sound</th>
</tr>
</thead>
</table>

| Scene: relevant case 4 | Description: Balfour v Balfour | Sound: Background Sound |

| Scene: relevant case 4 | Description: case issue and why the contract invalid | Sound: Background Sound |

| Scene: Module 5 | Description: Certainty definitions | Sound: Background Sound |

| Scene: relevant case 5 | Description: Karuppan Chetty v Suah Thian | Sound: Background Sound |

| Scene: relevant case 5 | Description: case issue and why the contract invalid | Sound: Background Sound |

| Scene: Module 6 | Description: Capacity & Legality definitions | Sound: Background Sound |

| Scene: relevant case 6 | Description: Tan Hee Juan v Teh Boon Keat | Sound: Background Sound |

| Scene: relevant case 6 | Description: case issue and why the contract invalid | Sound: Background Sound |
### TABLE IV. CHARACTER PROFILES

<table>
<thead>
<tr>
<th>Profiles</th>
<th>Character</th>
</tr>
</thead>
<tbody>
<tr>
<td>Host</td>
<td></td>
</tr>
<tr>
<td>Defendant</td>
<td><img src="image1" alt="Defendant Image" /></td>
</tr>
<tr>
<td>Plaintiff</td>
<td><img src="image2" alt="Plaintiff Image" /></td>
</tr>
<tr>
<td>Third Party (Module 1)</td>
<td><img src="image3" alt="Third Party Image" /></td>
</tr>
<tr>
<td>Mrs Balfour (Module 4)</td>
<td><img src="image4" alt="Mrs Balfour Image" /></td>
</tr>
</tbody>
</table>

- **Teenager (Module 6)**
- **Land**
- **Money**
- **Housing estate**
- **Handshake for agreement**
- **Goods (Module 1)**
- **Contract**
- **Court**
1) **Text production**: Text is vital to the creation of this animation, as it provides all the information needed for the viewers to understand the context of the work. In this venture, the words and sentences are simplified in order to make them clearer and more precise. The text was created using Adobe After Effects. For this animation, the font types Times New Roman, Georgia, and Impact are chosen in order to create the typewriting style.

2) **Graphic production**: The process of editing and design is in production for a two-dimensional animation, in which graphics or images are used. The characters and bitmap images are formed as a two-dimensional image using Adobe Illustrator, while some characters were taken from the Freepik website. The 2D images were used for this animation.

3) **Animation production**: In this section, the storyline and characters were created according to the storyboard that had already been done in the pre-production phase. All the characters were formed, colored, and edited in Adobe Illustrator. Then, the movements of the characters are done using Adobe After Effects. Then, the animation is saved as a clip of movie format. For all the objects and characters designed in this animation work, some have been done to make the animation process easier. All the assets are then saved in Adobe Illustrator. Besides, transitions are used for the movement of the characters as a way to make the characters move smoothly onto the canvas and make up the animation. The combining of movie clips and files is implemented in Adobe Premiere Pro, and the rendering file is an mp4 file. The targeted audience can then play this project on any sort of device, whether it be a laptop or their smartphones.

4) **Audio production**: This application has a voiceover that is recorded using the Audacity software, which is then imported into the animation using Adobe Premiere Pro. The background audio is taken from a non-copyrighted sound from the Internet.

5) **Product configuration management**: This work made use of Adobe Illustrator for illustration and the objects and characters were drawn in layers called vector layers. The characters and objects will then be animated frame by frame in the same software after they have been finished drawing. This is where all the coloring and shading takes place. After finishing the animation process, the animation is later exported to a software named Adobe After Effects to be animated. Finally, the animation will then be exported as an mp4 video format.

The status of implementation aims to keep track the process of development for the developed work. It will discuss the progress of development using the Gantt chart. Table V will show the status of development.

<table>
<thead>
<tr>
<th>Module</th>
<th>Duration</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Character design and modelling</td>
<td>1 week</td>
<td>The character and object that will be used will be designed</td>
</tr>
<tr>
<td>Animation scenes creation</td>
<td>3 weeks</td>
<td>The frame by frame animation using Adobe After Effects will be created</td>
</tr>
<tr>
<td>Background input and sound effect</td>
<td>1 week</td>
<td>Sound effect and sound for background will be chosen and input into the animation</td>
</tr>
<tr>
<td>Scene compiling in Adobe Premiere Pro</td>
<td>1 week</td>
<td>The animated scenes will be compiled to Premiere Pro</td>
</tr>
</tbody>
</table>

IV. DEVELOPMENT

Fig. 4 to 12 show some of the interfaces developed in the application.

![Fig. 4. Interface of the main page.](image1)

![Fig. 5. Interface of offer module.](image2)

![Fig. 6. Interface of acceptance module.](image3)
V. DISCUSSION

This paper presented the essential phases of analysis, design and development of the Business Law two-dimensional animation. A comprehensive explanation of the evaluation phase of the application is presented in [18]. The evaluation comprised five major evaluation components, including learnability, usability, accessibility, functionality, and effectiveness. The findings of the usability testing performed revealed that the majority of respondents were pleased with the outcomes of the animation. Also, the findings of the testing in [18] indicate that the outcomes of the 2D animation may facilitate and overcome the limitation from the current work proposed by other researchers in [15-17]. It is envisioned that this proposed animated video will assist teachers and students as an effective teaching and learning platform.

The main aim of this work is to help the students achieve a better understanding of the Elements of a Valid Contract topic and provide a way of memorising the relevant cases in each element. In a nutshell, this method of learning is proven to be able to assist students in understanding the Elements of a Valid Contract topic and memorising its relevant cases more effectively, hence the objective is reached.

VI. CONCLUSION

To conclude this research work, the animation was indeed proven to be successful. Despite a little shortcoming, the demand for the animation work was well addressed. Throughout the analysis phase, the animation and systems were investigated to determine the elements that could be required to build an effective teaching animation system.

The proposed techniques in this research will mostly notably benefit the realm of multimedia in a number of areas or ways that include the provision of an interactive method for effectively displaying complex contents in the social sciences field. The proposed work will be a benchmark and can be enhanced in other major complex study fields and topics in order to assist course delivery among tertiary students and lecturers. The application could be enhanced with other advanced approaches for more convenient accessibility and wider usability.

A few enhancement suggestions made in order to improve the application capabilities and execution in the future, such as making a separate video for each Element of a Valid Contract and different character designs for different character names that are featured in each different case. Besides, it is suggested...
to improve the sound quality of this animation by slowing down the narration to produce clearer and more precise explanation.

ACKNOWLEDGMENT

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REFERENCES


An Improved Lane-Keeping Controller for Autonomous Vehicles Leveraging an Integrated CNN-LSTM Approach

Hoang Tran Ngoc, Phuc Phan Hong, Nghi Nguyen Vinh, Nguyen Nguyen Trung, Khang Hoang Nguyen, Luyl-Da Quach
Software Engineering Department, FPT University, Cantho City, Vietnam

Abstract—Representing the task of navigating a car through traffic using traditional algorithms is a complex endeavor that presents significant challenges. To overcome this, researchers have started training artificial neural networks using data from front-facing cameras, combined with corresponding steering angles. However, many current solutions focus solely on the visual information from the camera frames, overlooking the important temporal relationships between these frames. This paper introduces a novel approach to end-to-end steering control by combining a VGG16 convolutional neural network (CNN) architecture with Long Short-Term Memory (LSTM). This integrated model enables the learning of both the temporal dependencies within a sequence of images and the dynamics of the control process. Furthermore, we will present and evaluate the estimated accuracy of the proposed approach for steering angle prediction, comparing it with various CNN models including the Nvidia classic model, Nvidia model, and MobileNetV2 model when integrated with LSTM. The proposed method demonstrates superior accuracy compared to other approaches, achieving the lowest loss function. To evaluate its performance, we recorded a video and saved the corresponding steering angle results based on human perception from the robot operating system (ROS2). The videos are then split into image sequences to be smoothly fed into the processing model for training.

Keywords—End-to-end steering control; convolutional neural network; LSTM; nvidia model; MobileNetV2; VGG16

I. INTRODUCTION

For over a decade, autonomous driving techniques have captured significant attention from both academic and industrial research and development sectors. During the initial phases of autonomous driving research, the predominant strategies employed were rule-based, primarily focused on image processing. In these approaches, perception, and control were treated as distinct functional modules, operating independently from each other [1]-[9]. However, with the advent of deep learning technologies, there has been a notable shift towards end-to-end vehicle control as a leading research area in autonomous driving [10]-[12]. This approach integrates perception and control into a seamless system, leveraging the power of deep learning to optimize autonomous driving performance.

In 2016, Nvidia introduced the pioneering end-to-end driving model for steering angle control [13]. This model employs Convolutional Neural Networks (CNN) to directly predict the steering angle using raw pixel data from a single frame obtained from a front-view camera. Subsequently, other research studies emerged, exploring various CNN architectures like MobileNetV2, ResNet50, and VGG16, with the aim of enhancing the accuracy and speed of steering angle estimation. These papers are presented with different definitions. [14]. However, these end-to-end driving models have neglected temporal information by focusing solely on individual frames.

In recent years, LSTM has been considered and incorporated into the CNN structure to learn continuous information from the image sequences of the past. Eraqi et al. [15] presented a C-LSTM (CNN with Long Short-Term Memory) model that captures both visual and dynamic temporal dependencies in driving. By incorporating both a CNN and an LSTM network, this model utilizes multiple frames from the front-facing camera input to estimate the steering angle. In a similar vein, Xu et al. [16] proposed an end-to-end architecture called FCN-LSTM, which not only predicts the steering angle but also aims to understand the scene simultaneously. In addition, Yang et al. [17] proposed a multi-modal multi-task network that takes an end-to-end approach and aims to simultaneously predict the steering angle and speed. However, the utilization of the conventional combination of CNN and LSTM in these methods limits their accuracy. With the continuous development and progress of processing hardware, CNN architectures with millions of parameters have been developed and successfully employed to achieve higher accuracy. In light of this, we present a novel approach in this paper by integrating the VGG16 model with LSTM to enhance the estimation accuracy. We leverage the relevant information from input image sequences to improve performance. Through a comparison with traditional methods, we demonstrate the exceptional accuracy achieved by our proposed approach. It is important to note that the implementation of this model will be carried out in a ROS2 simulation environment.

The structure of this paper is outlined as follows: Section II presents an overview of the proposed method. In Section III, we provide a detailed explanation of the CNN-LSTM architectures incorporated into our proposed model. Section IV introduces the experimental system, dataset, evaluation metrics, and the corresponding results, followed by a
comprehensive discussion. Finally, Section V concludes the paper, summarizing the key findings and contributions.

II. PROPOSED METHOD OVERVIEW

The system being developed within the ROS2 robot simulation environment comprises a vehicle model equipped with an RGB camera mounted on the front chassis. Our proposed synthetic neural network, which integrates VGG16 and LSTM networks, is utilized to estimate the steering angle based on input from the camera. The VGG16 model processes each frame of the camera image individually, extracting relevant features. These features are then fed into the LSTM network to capture temporal dependencies, as explained in the next section. The steering angle prediction is obtained from the output classifier following the LSTM layers. Upon completing the training process, the model will be saved and applied for testing on the vehicle model.

To train the proposed model network, we utilize the VGG16 architecture for feature extraction by including the current image (t) and the four preceding images from (t-1) to (t-4). This creates a sequence of 5 images captured within one 0.16s, which will serve as a sample sequence. The LSTM network will then analyze the temporal relationships among these images to estimate the steering angle based on contextual information. During training, the estimated steering angle at the time (t) will be compared with the corresponding ground-truth steering angle at the time (t), and the error will be used in the backpropagation algorithm [18] to update the model's parameters. During the training phase and after saving the model, the proposed system can be visualized through a block diagram, as shown in Fig. 1. This diagram outlines the flow of data and processes involved in the system's operation during both the training and running phases.

III. CNN-LSTM ARCHITECTURES

In this section, we will introduce the CNN architecture and its integration with LSTM to extract relevant features and combine them over time. This integration allows us to process the input data in a sequential manner and pass it through a fully connected layer to obtain the predicted steering angle.

A. Nvidia CNN-LSTM Model

The Nvidia model, introduced by Nvidia [10], utilizes convolutional neural networks (CNNs) and is specifically engineered to predict the steering angle by processing raw pixel information obtained from a front-facing camera. This model takes advantage of the visual information captured by the camera to directly predict the appropriate steering angle for autonomous driving. By training on a large dataset of images and corresponding steering angles, the Nvidia model learns to extract relevant features from the images and make accurate predictions. We have separated the convolutional feature map of the Nvidia model. Then, we integrated it with LSTM, as shown in Fig. 2, to process the image data sequence by first extracting the features individually before reassembling them using LSTM.

B. MobileNetV2-LSTM Model

The MobileNetV2 model is a lightweight CNN architecture that focuses on efficient computation [19]-[20]. It utilizes depthwise separable convolutions, which divide the convolutional operation into separate depthwise and pointwise convolutions.
This approach reduces the number of parameters and computational complexity, making it suitable for resource-constrained environments such as mobile devices or embedded systems. The MobileNetV2 model is also trained on image sequences and corresponding steering angles to learn the relationship between visual inputs and steering control. However, we have replaced MobileNetV2 with the Nvidia CNN model to extract features from the input image sequence. The architecture of MobileNetV2 integrated with LSTM and a fully connected layer is depicted in Fig. 3. We will proceed with training on the dataset obtained from driving videos in the ROS2 environment to evaluate the results.

C. Proposed VGG16-LSTM Model

The proposed VGG16-LSTM driving model, presented in Fig. 4, consists of two main components: the feature-extracting network and the steering angle prediction network. In this model, the input comprises the previous five frames, ranging from frame t-4 to frame t, which serve as inputs for the driving model.
In this study, the VGG16 architecture [21]-[22] is utilized as the feature extraction component. VGG16 is composed of several convolutional layers and pooling layers, which are employed to extract relevant features from images.

The network receives a sequence of five images, each having dimensions of 224x224x3, as input. These images are processed through the feature extraction layers of VGG16, resulting in a feature map with a predetermined size. Following that, a Flatten layer is utilized to convert the output of VGG16 into a vector shape. This vector will be fed into an LSTM network to store temporal information. The LSTM network will handle sequential data and retain previous information for estimating the steering angle.

In the LSTM model with multiple inputs and one output, the network takes in 5 input vectors corresponding to x(t-4), x(t-3), x(t-2), x(t-1), and x(t) as shown in Fig. 4(b). These vectors represent past temporal information. The LSTM model is designed to process and analyze sequential data. To accomplish this, the LSTM model utilizes activation functions, specifically the sigmoid function and the hyperbolic tangent (tanh) function. The sigmoid function is used for the input, forget, and output gates, ensuring controlled information flow within the model. On the other hand, the tanh function facilitates the storage and updating of continuous-valued information within the memory cell.

The output y(t) of the LSTM model is further processed by passing it through the final layers, which consist of two fully connected layers and one dropout layer as shown in Fig. 4(c). These layers contribute to refining the predicted steering angle estimation. The fully connected layers serve as a mapping function, transforming the input from the LSTM into a suitable output format for the desired steering angle estimation.

Every neuron is inter interconnected with all the neurons in the preceding layer, enabling the learning of intricate relationships and patterns. In the following chapter, we will proceed with the training and compare the accuracy of these models.

IV. EXPERIMENTAL RESULTS

A. Experimental Setting

The virtual environment, illustrated in Fig. 5, is constructed and designed using the Gazebo/ROS2 software. Within this simulation world, a donkey car and a two-lane map are present, serving as the training and testing environments for self-driving mode. Human experts control the donkey car through joystick input, and the captured images are saved and used for training the proposed models. The training process utilized a dataset comprising 10,000 images and was executed on a computer equipped with macOS Ventura 13.4, an ARM-based M2 CPU, a 10-core GPU, and 32 GB RAM. Our algorithm was implemented in Python 3.10, utilizing the Tensorflow 2.12.0 and Keras 2.12.0 libraries. The optimizer used in this study is ADAM [23]. For the experiments conducted, the initial learning rate is set to 0.001.

The VGG16-LSTM model has a total of 45,994,177 parameters, including both trainable and non-trainable ones. Out of these, 40,128,641 parameters are trainable, representing weights and biases that will be updated during training, while 5,865,536 parameters are non-trainable and remain fixed. By freezing the last 8 layers of the VGG16 base with a "trainable" attribute set to False, their weights and biases are preserved, leveraging pre-trained knowledge from the ImageNet dataset. The architecture and parameters of our proposed steering angle prediction model, which was based on transfer learning using VGG16-LSTM, are shown in Fig. 6.

Fig. 5. Simulated environment created using Gazebo/ROS2.
B. Dataset and Evaluation Metrics

1) Dataset: The vehicle is equipped with a front-facing camera that captures images, and the ROS2 controller records both the steering angle from the joystick and the corresponding camera images. The data is collected at a rate of 30 frames per second, resulting in five consecutive frames captured within a duration of 0.16 seconds. For this particular study, the dataset used consists of 10,000 images, which is equivalent to 2000 sequences of images. Each sequence contains five consecutive images. The dataset is continuously collected along with the corresponding steering angle information, which is obtained through human perception.

To access the dataset used in this study, you can visit the following link: (https://www.kaggle.com/datasets/ngochoangtran1992/steering-angle-prediction). The input images have a size of 1024x600 before being fed into the training model, and they undergo normalization to align with the input size of VGG16. Fig. 7 illustrates a sequence of five images, showing them before and after normalization, which prepares them for training.

2) Evaluation metrics: During the training process for estimating steering angles using a VGG16 model combined with LSTM, the Mean Squared Error (MSE) equation (4) is commonly used as the loss function and evaluation metric. The MSE measures the average squared difference between the predicted steering angles and the ground truth values provided by human perception.

\[ \text{MSE}(\hat{y}[t], y[t]) = \frac{1}{N} \sum_{i=1}^{N} (\hat{y}[t]_i - y[t]_i)^2 \]  

where \( \hat{y}[t]_i \) and \( y[t]_i \) are the predicted and true steering angles at the current time and the \( t^{th} \) sequence.

By minimizing the MSE loss, the model aims to accurately estimate the steering angles, ultimately improving the alignment between the predicted and actual values, as perceived by humans.
3) Results: The results and comparison of four models, namely Nvidia-CNN, CNN-LSTM, MobileNetv2-LSTM, and the Proposed method (VGG16-LSTM), were evaluated based on their loss values and validation loss values. The Nvidia-CNN model achieved a loss value of 314.03 and a validation loss value of 1268.70. The CNN-LSTM model obtained a loss value of 198.95 and a validation loss value of 479.37. The MobileNetv2-LSTM model demonstrated a loss value of 164.37 and a validation loss value of 244.32. Lastly, the Proposed method (VGG16-LSTM) outperformed the other models with a loss value of 65.07 and a validation loss value of 198.08. These results indicate that the Proposed method (VGG16-LSTM) achieved the lowest loss values, both in training and validation. This suggests that the VGG16-LSTM model performs better in estimating the steering angles compared to the other models, as it exhibits significantly lower loss values. The decrease in loss values indicates a stronger correlation between the predicted and actual steering angles, demonstrating enhanced precision and effectiveness in estimating steering angles. The comparison results of the models are depicted in Table I, and Fig. 8 and Fig. 9 visualize the training outcomes and accuracy assessment of these models throughout 20 epochs.

<table>
<thead>
<tr>
<th>Model</th>
<th>Nvidia-CNN</th>
<th>CNN-LSTM</th>
<th>MobileNetv2-LSTM</th>
<th>Proposed Model</th>
</tr>
</thead>
<tbody>
<tr>
<td>Loss</td>
<td>314.03</td>
<td>198.95</td>
<td>164.37</td>
<td>65.07</td>
</tr>
<tr>
<td>Val_Loss</td>
<td>1268.70</td>
<td>479.37</td>
<td>244.32</td>
<td>198.08</td>
</tr>
</tbody>
</table>

Fig. 10 shows the comparison between the steering angle predictions of various models with the proposed approach. It is readily apparent that the proposed method’s steering angle predictions exhibit an accuracy level of approximately 95% when compared to the ground truth values. Achieving this level of accuracy involves implementing a method that utilizes a series of input images and incorporates information from previous frames to estimate the steering angle. The VGG16 and LSTM networks are utilized to extract significant features, ensuring the highest possible precision in the predictions.
By combining VGG16 CNN and LSTM, our proposed approach successfully captures the temporal aspects of visual information and the dynamics of control. This integration sets it apart from other models, as it achieves an exceptional accuracy rate of approximately 95% when predicting steering angles. The results obtained in the ROS2 simulation environment are highly promising, suggesting significant potential for practical applications. This advancement represents a substantial improvement in the precision and dependability of autopilot systems, enhancing their ability to navigate real-life scenarios with greater accuracy and reliability.

**REFERENCES**


Lane Road Segmentation Based on Improved UNet Architecture for Autonomous Driving

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Abstract—This paper introduces a real-time workflow for implementing neural networks in the context of autonomous driving. The UNet architecture is specifically selected for road segmentation due to its strong performance and low complexity. To further improve the model’s capabilities, Local Binary Convolution (LBC) is incorporated into the skip connections, enhancing feature extraction, and elevating the Intersection over Union (IoU) metric. The performance evaluation of the model focuses on road detection, utilizing the IoU metric. Two datasets are used for training and validation: the widely used KITTI dataset and a custom dataset collected within the ROS2 environment. Simulation validation is performed on both datasets to assess the performance of our model. The evaluation of our model on the KITTI dataset demonstrates an impressive IoU score of 97.90% for road segmentation. Moreover, when evaluated on our custom dataset, our model achieves an IoU score of 98.88%, which is comparable to the performance of conventional UNet models. Our proposed method to reconstruct the model structure and provide input feature extraction can effectively improve the performance of existing lane road segmentation methods.

Keywords—Local binary patterns; feature extraction; UNet; semantic segmentation

I. INTRODUCTION

There has been a growing interest in autonomous driving research due to its significant impact on traffic management, the economy, and the development of self-driving cars.

The purpose of these vehicles is to imitate human driving actions through intelligent decision-making and executing various tasks such as switching lanes, preventing collisions, detecting objects, and issuing warnings for lane departure [1], [2], [3], [4], [5]. The design of autonomous driving cars involves three essential components: perception, path planning, and control [6], [7], [8], [9]. Recent advancements in sensor technology have greatly improved perception capabilities. While cameras are commonly used, the integration of additional sensors like GPS, radars, or LiDARs enhances the performance of self-driving systems [10]. The focus of autonomous navigation is on accurately detecting and identifying traffic participants, including cars, pedestrians, and surrounding objects/areas.

In particular, road detection and segmentation are crucial for autonomous driving and intelligent transportation systems as it ensures safe and efficient vehicle operation. Solutions in this area aim to reduce accidents, alleviate traffic congestion, and improve fuel efficiency.

Precise detection and recognition of roadways, encompassing boundaries and lanes, empower intelligent decision-making and enhance navigation efficiency. These advancements have the potential to greatly enhance overall transportation systems. Various datasets such as KITTI [11], Berkeley DeepDrive [12], A2D2, or those generated by the CARLA simulator [13] are utilized for a range of autonomous driving and lane segmentation tasks. Teichmann et al. Research was carried out to measure the computational time required for semantic segmentation tasks using the KITTI dataset [14]. Neven et al. focused on scene understanding using the Cityscapes dataset [15]. Similarly, real-time efforts utilizing the Cityscapes dataset involved the development of an ENet architecture [16]. Wang et al. utilized 3D LiDAR point clouds and the PointSeg architecture for real-time semantic segmentation [17]. Bai et al. explored time-critical task performance in road segmentation using the KITTI benchmark [18]. Additionally, Jang et al. aimed to explain and reduce the end-to-end delay for self-driving cars in their work [19].

In recent years, UNet is a fully convolutional network architecture that has gained popularity for lane segmentation in autonomous driving. It utilizes a U-shaped network design for accurate identification and delineation of road lanes. Studies have demonstrated its effectiveness, comparing it favorably to other methods in terms of accuracy and efficiency [20]-[21]. Giurgi et al. introduce a real-time implementation workflow for neural networks in autonomous driving, specifically focusing on road segmentation using the UNet structure with the KITTI dataset [22]. UNET’s potential for improving autonomous driving systems may be seen in activities such as lane departure alerts and autonomous lane holding. However, when autonomous cars operate in tough traffic settings with high levels of noise and interference from elements such as dust, vibrations, rain, and wind, these algorithms become susceptible to disruptions, resulting in decreasing lane segmentation accuracy. The existence of these external elements severely impairs the effectiveness of lane segmentation algorithms, resulting in less than ideal results.
To address these challenges, we propose the combination of LBC layers with UNet in this paper to improve lane segmentation performance in noisy traffic environments. The LBC layers integrate local binary patterns into the convolutional neural network (CNN) architecture [23], enhancing the ability to extract fine-grained structural information and model image representations. These layers have shown potential in applications that require robust feature extraction and learning, particularly in scenarios with limited training data or noisy environments. We will compare the performance of the proposed method to earlier approaches in order to quantify the improvement in accuracy on two datasets: KITTI and our own gathered dataset in a ROS2 robot simulation environment. Fig. 1 describes our proposed system.

The subsequent sections of the paper follow the following structure. Section II provides an introduction and summary of relevant research pertaining to lane segmentation. Section III describes the model's architecture in detail, outlining the integration of LBC and skip connections to enhance local feature extraction. Section IV focuses on the experimental implementation, dataset utilization, and a comparative analysis of various models. Finally, the paper concludes by summarizing key findings and proposing future avenues for advancement.

II. RELATED WORK

In recent years, there have been several advancements in the field of road lane segmentation. One notable approach is the DeepLab method proposed by Chen et al. [24], which combines deep convolutional nets with fully connected conditional random fields for accurate semantic image segmentation. Another approach is the ENet architecture introduced by Paszke et al. [25], specifically designed for real-time semantic segmentation tasks. Additionally, Pan et al. [26] proposed LaneNet, a spatial CNN architecture for traffic scene understanding, focusing on lane segmentation. Fu et al. [27] presented SCNN, a parallel CNN model that explores the road scene in depth for precise road segmentation. In a recent study, Giurgi et al. developed a unique method employing the UNet architecture, which demonstrated appreciable increases in lane segmentation accuracy. These studies are a limited exploration of image segmentation in the complex context of autonomous vehicles. Lane recognition is a crucial task in autonomous driving, and existing approaches confront difficulties owing to the complexities of the input pictures. To address this, we propose the use of LBC layers to reduce complexity and increase processing speed. Building upon this, we present an enhanced UNet model incorporating skip connections and LBC layers for improved lane markings recognition while minimizing training time. Comparative analysis and evaluation metrics, such as IoU, Dice coefficient, and precision, are employed to assess the accuracy and efficiency of the proposed models. Our study focuses on developing efficient and accurate segmentation models for lane recognition in autonomous driving scenarios.
III. PROPOSED METHOD

A. Local Binary Pattern and its Convolution Variants

1) Local binary patterns: Local Binary Patterns (LBPs) is an image processing technique used for capturing local patterns by comparing pixel values in small neighborhoods [28]. It is commonly employed in face recognition and object detection. LBPs operate by selecting a neighborhood around each pixel and converting the pixel values into a binary string. By comparing the values of surrounding pixels with the central pixel, the binary string is constructed. The equation for calculating the brightness intensity of LBPs can be described as follows:

\[ LBP_p = \sum_{p=0}^{P-1} s(g_p - g_c)2^p \]  

where \( s(z) \) is the neighbor pixel intensity value; \( g_c \) is the center pixel intensity value. \( P \) is the number of neighbor pixels. \( Z \) is the result of \( g_p \) minus \( g_c \).

This string represents the local spatial patterns within the neighborhood. Analyzing the distribution of these binary patterns provides valuable insights into local variations in brightness, which can be utilized for tasks like contrast enhancement and object recognition. LBPs are a compact representation of local patterns and find wide usage in computer vision applications. Fig. 2 illustrates the basic operation of LBPs, demonstrating their functionality for each pixel in an image with local dimensions of 3x3 and 5x5.
2) **Local binary convolution**: The Local Binary Convolution (LBC) layer convolves a filter over an input image, converting pixel values into binary patterns called Local Binary Patterns (LBPs). These patterns record structural, morphological, and textural information. During training, gradients can be backpropagated through the layer's anchor weights, while the learnable 1x1 filters are updated. The anchor weights can be generated deterministically or stochastically, allowing for diversified filters and fine-grained control over weight sparsity. The LBC layer efficiently extracts meaningful features for tasks like object detection and recognition in computer vision. In Fig. 3, we present the basic model of LBC that we use in this paper. The input image is first separated into three RGB channels through the LBP 3x3 local dimensions. After that, a convolutional layer is used for additional processing, followed by the ReLU activation function to extract key traits of road segments.

**B. Improved Lane Road Segmentation**

1) **Conventional UNet model**

   a) **UNet architecture**: UNet is an architecture for semantic segmentation introduced by Olaf Ronneberger et al. [23]. This is a widely used architecture for road segmentation that combines encoding and decoding paths. It utilizes max pooling for down-sampling and transposed convolution for up-sampling. Skip connections play a crucial role in preserving information between the encoding and decoding stages.

   Fig. 4 illustrates the structure of basic UNet architecture. It is made up of a left side encoding path and a right-side decoding path. Max pooling techniques are used in the encoding process to gradually lower the spatial resolution while raising the number of feature channels. This helps extract abstract features related to road structures. The decoding path employs transposed convolutions to up-sample the feature maps and increase the spatial resolution. This results in a dense output map representing the road segmentation mask. Skip connections establish direct connections between corresponding encoding and decoding layers, allowing detailed information to flow between them. This facilitates the reconstruction of accurate road segmentations.
UNet’s combination of encoding and decoding paths with skip connections enables exceptional performance in road segmentation. It effectively captures both local and global context information, enabling precise delineation of road regions in images.

b) Up-convolution with transposed convolution approach: Transposed convolution, also known as deconvolution or fractionally stridden convolution, is a technique used to up-sample feature maps in convolutional neural networks. It is the reverse operation of the standard convolution operation and is commonly used in the decoding path of architectures like UNet. Fig. 5 shows the operation of Transposed Convolution with a 2x2 kernel. The parameters required to design a Transposed Convolution to achieve the desired output size can be described using (2):

\[ O_{\text{size}} = (T_{\text{size}} - 1) \cdot s + H_{\text{size}} - 2 \cdot p \]  

(2)

where \( O_{\text{size}} \) is the desired size of the output feature map; \( T_{\text{size}} \) refers to the size of the input feature map; \( s \) is the stride value used in the Transposed Convolution operation; \( H_{\text{size}} \) represents the size of the kernel used in the operation; and \( p \) refers to the padding applied to the input feature map.

However, in noisy and challenging environments such as transportation, a lane segmentation system with a robust feature extractor needs to be investigated. Therefore, we have developed an algorithm that we propose in the next section.

2) Improved design of the UNet model: The improved design of the UNet model aims to enhance the segmentation accuracy compared to the classic UNet architecture. To achieve this, we have introduced the Local Binary Convolution (LBC) layer into the skip connections of the UNet model. The design structure is illustrated in Fig. 6.

The encoding structure (left side) consists of four blocks. Each block includes a series of convolutional layers with ReLU activation, followed by max-pooling operations. The output of each block is created by gradually applying the pooling and convolutional layer operations. The LBC layers are integrated into the skip connections of the UNet model to capture local binary patterns and improve the segmentation performance. These skip connections establish direct connections between the corresponding encoder and decoder layers. The four blocks handle the up-sampling and concatenation operations necessary for the skip connections. They take the inputs from the corresponding pooling layers and transpose convolutional layers to up-sample the feature maps. Finally, the model is compiled with the Adam optimizer and binary cross-entropy loss. The metrics used for evaluation include the Intersection over Union (IoU). This improved UNet model with LBC layers in the skip connections offers enhanced capabilities for accurately segmenting road images. In the upcoming section, we will evaluate the results and accuracy of this model using two datasets. The purpose is to demonstrate the superiority of the proposed method in comparison to existing approaches.

IV. EXPERIMENTAL RESULTS

A. Experimental Setting

The experiments were conducted to train and evaluate multiple models using two distinct datasets. Each model underwent 100 epochs of training, and performance was assessed based on metrics such as IOU, Validation IOU, Loss, and Validation Loss. The training process utilized a computer with Ubuntu 20.04, an Intel i7 3.4 GHz CPU, an Nvidia GTX 3060 Laptop, and 32 GB RAM. The implementation was carried out in Python 3.9.13, employing the Conda 22.9.0, CUDA 11.7, Tensorflow 2.10.0, and Keras 2.10.0 libraries.
Fig. 7. Comparison of segmentation performance among DeepLabv3, UNet, and proposed model based on IOU, Validation IOU, Loss, and Validation Loss. a), b), c) using KITTI dataset, and d), e), f) using Gazebo/ROS2 dataset.

B. Datasets and Evaluation Metrics:

1) Datasets: The research paper utilizes two datasets, namely KITTI [29] and a dataset created from simulated lanes in the Gazebo/ROS2 environment of our laboratory. The KITTI dataset is employed primarily for unmarked lane segmentation in urban areas, comprising 800 training images and 200 test images. On the other hand, the second dataset involves the Donkey self-driving car, which operates in the Gazebo/ROS2 3D simulation environment. The car is controlled using a driving wheel joystick to maintain lane position. The car is equipped with a front-facing camera that captures images, and the ROS2 controller records these images at a rate of 5 frames per second for training data. A total of 1000 images were collected for this dataset. The data split ratio is 80% for training data and 20% for validation data.

2) Evaluation metrics: In image segmentation, Intersection over Union (IoU) [25] is a primary metric used to evaluate the accuracy of models. Unlike in object detection, where IoU serves as a supplementary metric, it plays a crucial role in the pixel-level analysis of segmentation masks. The definitions of true positive (TP), false positive (FP), and false negative (FN) differ slightly in image segmentation, considering the pixel-wise intersection and logical operations between the ground truth and segmentation masks. IoU is determined in image segmentation by dividing the intersected area by the sum of the ground truth and prediction areas using the TP, FP, and FN areas, or pixel counts.

Fig. 8. Example of the IOU equation.

This metric helps assess the effectiveness of models in accurately segmenting objects and regions of interest in images. The equation is shown below:
Fig. 9. Lane road segmentation results of the proposed model using the KITTI dataset.

TABLE I. SEGMENTATION RESULTS ON KITTI DATASET

<table>
<thead>
<tr>
<th>Approach</th>
<th>IOU</th>
<th>Validation</th>
<th>Loss</th>
<th>Validation</th>
</tr>
</thead>
<tbody>
<tr>
<td>DeeplabV3+</td>
<td>0.93</td>
<td>0.9292</td>
<td>0.014</td>
<td>0.0670</td>
</tr>
<tr>
<td>UNET</td>
<td>0.95</td>
<td>0.9211</td>
<td>0.021</td>
<td>0.0742</td>
</tr>
<tr>
<td>LBC+UNET</td>
<td>0.97</td>
<td>0.9442</td>
<td>0.012</td>
<td>0.0437</td>
</tr>
</tbody>
</table>

with:

\[
TP = GT \times X  \\
FP = (GT + X) - GT \\
FN = (GT + X) - X
\]

where TP, FP, and FN indicate the True Positive, False Positive, and False Negative numbers, respectively; GT is the region's Ground Truth; X is segmentation mask overlap. Fig. 8 shows an example of IoU on the actual input image.

During training, the Loss function is used to measure the difference between the model's predicted output and the actual output value. The goal is to find a way to minimize the loss function to make a more accurate prediction. The equation is shown below (4):

\[
Log L = \frac{1}{E} \sum_{i=1}^{E} [-y_i log(v_i) + (1 - y_i) log(1 - v_i)]
\]

where L represents the Binary Cross Entropy Loss, E is the number of samples in the dataset, \( y_i \) represents the true label (ground truth) for the \( i^{th} \) sample (0 or 1), \( v_i \) represents the projected probability for the \( i^{th} \) sample, and log represents the natural algorithm.

Fig. 10. Lane road segmentation results of the proposed model using the Gazebo/ROS2 dataset.

TABLE II. SEGMENTATION RESULTS ON GAZEBO/ROS2 DATASET

<table>
<thead>
<tr>
<th>Approach</th>
<th>IOU</th>
<th>Validation</th>
<th>Loss</th>
<th>Validation</th>
</tr>
</thead>
<tbody>
<tr>
<td>DeeplabV3+</td>
<td>0.94</td>
<td>0.933</td>
<td>0.0223</td>
<td>0.0319</td>
</tr>
<tr>
<td>UNET</td>
<td>0.94</td>
<td>0.948</td>
<td>0.0238</td>
<td>0.0388</td>
</tr>
<tr>
<td>LBC+UNET</td>
<td>0.988</td>
<td>0.96</td>
<td>0.0122</td>
<td>0.0144</td>
</tr>
</tbody>
</table>

3) Results and discussion: DeepLabV3+ [30], UNet, and LBC+UNet segmentation outcomes were compared using Intersection over Union (IoU) and loss values on two datasets, the KITTI dataset and the ROS2 dataset. Afterwards 100 training epochs, the three techniques' highest IoU values were as follows: On the KITTI dataset, DeepLabV3+, UNet, and LBC+UNet each had an IoU of 0.93, 0.95, and 0.97, respectively. In addition, using the KITTI dataset, the three techniques produced the following loss values: LBC+UNet had a loss of 0.012, DeepLabv3+ had a loss of 0.014, and UNet had a loss of 0.021.

The models' performance was further examined using the Gazebo/ROS2 dataset. The IoU values attained by the three methods were as follows after 100 training epochs: DeepLabV3+ had an IoU of 0.94, UNet had an IoU of 0.94, and LBC+UNet had an IoU of 0.988. DeepLabV3+ had a loss of 0.0223, UNet had a loss of 0.0238, and LBC+UNet had a loss of 0.0122. These were the loss numbers produced by the three methodologies.
The proposed method, LBC+UNet, achieved the highest segmentation results in terms of IoU for both the KITTI dataset and the Gazebo/ROS2 dataset. This can be observed from the results presented in Fig. 7, where the performance of each epoch is displayed. The IoU Validation and Loss Validation metrics are also included in Tables I and II, respectively, to facilitate a clearer comparison. The proposed method demonstrated the highest accuracy in terms of IoU and the lowest error in terms of the loss function.

The experimental results of road lane segmentation using the proposed method are illustrated in Fig. 9 and Fig. 10. We can observe a high level of accuracy, exceeding 95%, which can be attributed to the utilization of the feature extraction capabilities of LBC combined with the UNet architecture. The performance of the proposed method is consistently strong on both the KITTI and Gazebo/ROS2 datasets, demonstrating good segmentation results and high accuracy. Moreover, the proposed method was tested on both simulated and real-world datasets, confirming its effectiveness in road lane segmentation. This advancement supports autonomous driving systems and contributes to reducing accidents by providing higher accuracy.

V. CONCLUSION

We have successfully used a combination of the UNET architecture and the LBC feature extractor in this article to improve the accuracy of road lane segmentation for autonomous driving support. The proposed method has demonstrated superior accuracy compared to conventional approaches such as DeepLabV3+ and the classical UNET method. Through comprehensive evaluations using well-known datasets, including KITTI and our custom-built dataset based on the ROS2 robot simulation model, the proposed model has proven its effectiveness in both simulated and real-world scenarios. The application of our proposed model holds great potential for various domains, including simulation and practical implementations. Looking ahead, further advancements in road lane segmentation will focus on fulfilling the demand for more refined lane segmentation, particularly the differentiation between drivable and non-drivable areas. This ongoing development will significantly contribute to the improvement of self-driving systems by providing precise lane segmentation for enhanced decision-making and safer navigation.

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Leveraging Big Data and AI in Mobile Shopping: A Study in the Context of Jordan

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Abstract—This study investigates the current state of mobile shopping in Jordan and the integration of big data and AI technologies in this context. A mixed-methods approach, combining qualitative and quantitative data collection techniques, utilized to gather comprehensive insights. The survey questionnaire distributed to 105 individuals engaged in mobile shopping in Jordan. The findings highlight the popularity of mobile shopping and the preference for mobile apps as the primary platform. Personalized product recommendations emerged as a crucial factor in enhancing the mobile shopping experience. Privacy concerns regarding data sharing were present among respondents. Trust in AI-powered virtual assistants varied, indicating the potential for leveraging AI technologies. Respondents recognized the potential of big data and AI in improving the mobile shopping experience. The study concludes that businesses can enhance mobile shopping by utilizing AI-powered virtual assistants and prioritizing data security. The findings contribute to understanding mobile shopping dynamics and provide guidance for businesses and policymakers in optimizing mobile shopping experiences and driving economic growth in Jordan’s digital economy. Future research and implementation efforts are encouraged to harness the potential of big data and AI in the mobile shopping landscape.

Keywords—Big data; mobile shopping; artificial intelligence; internet of things; shopping; user experience

I. INTRODUCTION

Mobile shopping has become increasingly significant in recent years, revolutionizing the way people shop and interact with businesses. The convenience and accessibility offered by mobile devices have made them a preferred platform for consumers to browse, compare, and purchase products and services [1]. Simultaneously, the growing prominence of big data and artificial intelligence (AI) has opened up new avenues for businesses to harness valuable insights and deliver personalized experiences to customers [2].

Big Data refers to extremely large and complex sets of data that exceed the capabilities of traditional data processing methods. It encompasses massive volumes of information generated from various sources, such as social media, sensors, devices, and business transactions. Big Data is characterized by its three primary attributes: volume, velocity, and variety.

Big Data and AI have significantly transformed the landscape of mobile shopping, enhancing the user experience, personalization, and overall efficiency of the shopping process [3]. Big Data and AI algorithms analyze vast amounts of user data, such as browsing history, purchase behavior, and demographic information, to provide personalized product recommendations. These recommendations improve the relevancy of suggestions, increasing the likelihood of finding products that match users’ preferences and needs [2]. By leveraging Big Data and AI, mobile shopping platforms can predict consumer behavior and preferences. Analyzing historical data, user patterns, and market trends enable retailers to anticipate customer needs, optimize inventory management, and plan marketing strategies more effectively [1].

These intelligent assistants improve customer engagement and enable personalized interactions, enhancing the overall shopping experience on mobile devices. Big Data analytics, combined with AI algorithms, help detect and prevent fraudulent activities in mobile shopping [3]. By analyzing patterns and anomalies in real-time, retailers can identify suspicious transactions and implement security measures to protect customer data and financial information. AI-based visual search enables users to find products by simply taking a picture or uploading an image. This technology uses computer vision algorithms to analyze images and match them with relevant products. Big Data and AI play a crucial role in optimizing supply chain management in mobile shopping [2].

In the context of Jordan, where mobile penetration rates are high and e-commerce is rapidly expanding, understanding the potential impact of mobile shopping, big data, and AI is crucial. The adoption of these technologies has the potential to transform customer experiences, enhance business operations, and drive economic growth in the country.

This study aims to explore the implications of mobile shopping, big data, and AI on customer experiences, business operations, and economic growth in Jordan. By investigating the current landscape, identifying challenges and opportunities, and proposing strategies for implementation, this research seeks to provide valuable insights for businesses, policymakers, and stakeholders in the Jordanian market.

A. Research Objectives

1) To assess the significance of mobile shopping in Jordan and its impact on consumer behavior.

2) To explore the role of big data and AI in enhancing customer experiences in the mobile shopping context.
3) To analyze the potential benefits and challenges of implementing big data and AI technologies in improving business operations.

4) To examine the economic implications of mobile shopping, big data, and AI on the overall growth and competitiveness of Jordan’s economy.

5) Most of the styles are intuitive. However, we invite you to read carefully the brief description below.

B. Research Questions

1) How does mobile shopping influence consumer behavior in Jordan?

2) What are the potential benefits of utilizing big data and AI in the context of mobile shopping for enhancing customer experiences?

3) What challenges exist in implementing big data and AI technologies in improving business operations in the mobile shopping sector?

4) How can mobile shopping, big data, and AI contribute to the overall economic growth of Jordan?

II. REVIEW OF LITERATURE

Rabah, K documented that any firm depends on its data [4]. Today, big data has applications in almost every sector of the economy, including customer service, retail, healthcare, and financial services. Any firm that can integrate data to address nagging questions about its processes might benefit from big data. In general, big data is in demand across all business sectors. Those who work to comprehend the businesses of their customers and difficulties will have an advantage over their competitors by being able to anticipate and choose big data solutions that are fit for their needs. A growing number of industries, including professional, scientific, and technical services, information technology, manufacturing, banking and insurance, and retail, are in need of workers with big data skill sets. Cloud computing is essential to DevOps. IoT needs the cloud to function properly because the cloud needs compute to do so. Up until the introduction of large data, AI solely was used as a model. The IT industry as we know it changed as a result of the fourth industrial revolution and the upcoming machine economy [4].

Vassakis, et al., explained the elements that define a firm’s innovation. Competitiveness has changed as a result of the fourth industrial revolution (Industry 4.0) period’s revolution in networks, platforms, people, and digital technology; big data has a huge impact on organisations [3]. Academics and professionals are constantly extremely excited about big data because it encourages businesses and organisations to be creative and produces that is beneficial. This excitement is reshaping local, national, and international economies. In that context, the term “data science” refers to a corpus of foundational concepts that underpin the extraction of knowledge and information from data. The techniques and technologies used help analyse important data, enabling firms to gain insight into their surroundings and take prompt, well-informed decisions. The Internet of Things (constantly expanding number of connected devices, sensors, and smartphones) is largely to blame for the current “data-driven” era, in which big data analytics used in every industry as well as every nation’s economy. A well-known worldwide development is the increase in data availability, and data analysis methods used to glean practical knowledge from the data. The majority of firms in that situation collect, archive, and analyse data in order to make wise business decisions.

Lo’ai, et al., stated that Mobile devices are assisting people in carrying out a variety of crucial duties and are rapidly turning into a need in daily life [5]. Combining mobile and cloud computing can boost their advantages and functionalities while minimising their disadvantages, such as limited memory, CPU, and battery capacity. The four Vs of big data analytics make the value extraction of data possible: volume, variety, velocity, and veracity [6]. In this study, connected healthcare I looked at and how mobile cloud computing and big data analytics can support it. The rationale behind the creation of connected healthcare software and systems also discussed along with the debut of cloud-based computing in healthcare.

Latif, et al, explained that worldwide urban population growth is expanding quickly, posing new problems for daily life for inhabitants such as environmental degradation, public safety, and traffic congestion. In order to control this rapid growth, new technologies have been developed to create intelligent cities [2]. By incorporating the Internet of Things (IoT) into everyday life, citizens can create new intelligent services and apps that benefit various citywide industries, such as healthcare, security, agriculture, etc. [7]. IoT devices and sensors produce large volumes of data, which may be analysed to learn important facts and learn new things that improve the quality of life for citizens. The potential for improving the effectiveness and performance of IoT is both big data analytics has recently been proved by Deep Learning (DL), a new branch of artificial intelligence (AI). In this study, we review the literature on the application of IoT and DL to the creation of smart cities. We start by describing the IoT and outlining the traits of big data produced by the IoT. Next, we discuss the various computing platforms—including cloud, fog, and edge computing—that used Internet of Things (IoT) big data analytics.

III. METHODOLOGY

To achieve comprehensive insights into the current state of mobile shopping in Jordan and the integration of big data and AI technologies, a mixed-methods approach is utilized. This approach combines qualitative and quantitative data collection techniques, including surveys, and data analysis.

A. Data Collection Techniques

1) Surveys: A structured questionnaire will be designed to collect quantitative data from mobile shoppers in Jordan. The survey will cover aspects such as consumer behavior, preferences, satisfaction, and usage patterns related to mobile shopping. The survey will be distributed online to a diverse sample of mobile shoppers in Jordan, ensuring representation from different demographic groups.
2) Sample selection: For the survey, a stratified sampling technique will be employed to ensure representation from various demographic groups in Jordan. The sample will be selected based on factors such as age, gender, income level, and geographical location to capture a diverse range of mobile shoppers.

3) Quantitative analysis: The survey data will be analysed using appropriate statistical techniques, such as descriptive statistics, correlation analysis, and regression analysis. This analysis will provide quantitative insights into consumer behavior, preferences, satisfaction levels, and the impact of mobile shopping on customer experiences in Jordan.

4) Qualitative analysis: The interview data will undergo thematic analysis to identify recurring themes, patterns, and insights related to the integration of big data and AI technologies in mobile shopping in Jordan. This analysis will provide qualitative insights into the challenges, opportunities, strategies, and stakeholders’ perspectives.

5) Questionnaire:
   a) What is your age group?
      1. 18-25 years
      2. 26-35 years
      3. 36-45 years
      4. 46+ years
   b) How frequently do you engage in mobile shopping?
      1. Daily
      2. Weekly
      3. Monthly
      4. Rarely or never
   c) Which mobile shopping platforms do you primarily use?
      1. Mobile apps
      2. Mobile websites
      3. Both mobile apps and websites
      4. I don't engage in mobile shopping
   d) How important is personalized product recommendations in your mobile shopping experience?
      1. Extremely important
      2. Important
      3. Neutral
      4. Not important
   e) Are you concerned about the privacy of your personal data when using mobile shopping apps?
      1. Very concerned
      2. Concerned
      3. Neutral
      4. Not concerned
   f) How likely are you to trust AI-powered virtual assistants for mobile shopping recommendations?
      1. Very likely
      2. Likely
      3. Neutral

4) Unlikely

g) Which factors would encourage you to share your personal data for personalized mobile shopping experiences?
   1. Discounts and personalized offers
   2. Enhanced convenience and time-saving
   3. Trust in the platform's security measures
   4. None of the above

h) How satisfied are you with the current level of personalization in mobile shopping apps?
   1. Very satisfied
   2. Satisfied
   3. Neutral
   4. Dissatisfied

i) To what extent do you believe big data and AI can improve your mobile shopping experience?
   1. Significantly
   2. Moderately
   3. Slightly
   4. Not at all

j) Would you be willing to pay a premium for products or services that leverage big data and AI to enhance your mobile shopping experience?
   1. Yes, definitely
   2. Yes, maybe
   3. No
   4. Unsure/Not applicable

B. Data Collection Techniques

For the research proposal, data collected through a survey questionnaire that was distributed to a sample size of 105 individuals who engaged in mobile shopping in Jordan. The data collected through the survey provided insights into the past state of mobile shopping in Jordan and the integration of big data and AI technologies.

1) Sampling: A convenience sampling method employed to select the participants for the survey. The sample consisted of individuals who actively engaged in mobile shopping in Jordan in the past. Efforts were made to ensure diversity in terms of age, gender, geographical location, and socioeconomic back-ground.

2) Distribution: The survey questionnaire was distributed to the selected sample using various methods such as online platforms, social media groups, and email. Participants were provided with clear instructions on how to complete the questionnaire and any specific guidelines or deadlines.

3) Data collection: Participants were asked to complete the survey questionnaire within a given timeframe. They had the option to respond online or submit physical copies of the questionnaire. It ensured that participants understood the questions and provided accurate and honest responses based on their experiences.

4) Data validation and cleaning: Once the data collection was complete, the collected responses were validated for
completeness and accuracy. Any incomplete or inconsistent responses were addressed. The data cleaned to remove any errors or outliers, ensuring the reliability of the data collected in the past.

IV. RESULTS AND ANALYSIS

The collected data from different peoples in Jordan was assessed and analyzed using appropriate statistical techniques and analytical methods. Quantitative analysis involved descriptive statistics shown on Table I, such as frequencies and percentages.

A. Descriptive Statistics

The Fig. 1 shows frequencies of data per age group for data analysis of the questionnaire survey conducted in Jordan.

![Frequencies of Data](image1.png)

**TABLE I. DESCRIPTIVE STATISTICS**

<table>
<thead>
<tr>
<th>Descriptive Statistics</th>
<th>Values</th>
</tr>
</thead>
<tbody>
<tr>
<td>What is your age group</td>
<td>Minimum</td>
</tr>
<tr>
<td>How frequently do you engage in mobile shopping</td>
<td>1.00</td>
</tr>
<tr>
<td>Which mobile shopping platforms do you primarily use</td>
<td>1.00</td>
</tr>
<tr>
<td>How important is personalized product recommendations in your mobile shopping experience</td>
<td>1.00</td>
</tr>
<tr>
<td>Are you concerned about the privacy of your personal data when using mobile shopping apps</td>
<td>1.00</td>
</tr>
<tr>
<td>How likely are you to trust AI-powered virtual assistants for mobile shopping recommendations?</td>
<td>1.00</td>
</tr>
<tr>
<td>Which factors would encourage you to share your personal data for personalized mobile shopping experiences</td>
<td>1.00</td>
</tr>
<tr>
<td>How satisfied are you with the current level of personalization in mobile shopping apps</td>
<td>1.00</td>
</tr>
<tr>
<td>To what extent do you believe big data and AI can improve your mobile shopping experience</td>
<td>1.00</td>
</tr>
<tr>
<td>Would you be willing to pay a premium for products or services that leverage big data and AI to enhance your mobile shopping experience</td>
<td>1.00</td>
</tr>
</tbody>
</table>

![Fig. 1. Frequencies of data age group.](image2.png)

The data analysis of the questionnaire survey conducted in Jordan indicates a predominantly positive response towards the leverage of big data and the impact of AI on mobile shopping. In above charts, x-axis shows the frequency of how much impact of AI on mobile shopping and y-axis shows all the questions that asked to the people of Jordan. The results reveal the following findings based on descriptive statistics and frequencies in bar charts:

A. Mobile Shopping Engagement

The majority of respondents (99%) reported engaging in mobile shopping on a regular basis, indicating a high level of participation in this mode of shopping (see Fig. 2).

![Fig. 2. Graph showing answer to ‘how frequently do you engage in mobile shopping.’](image3.png)

B. Preferred Mobile Shopping Platform

Among the respondents, 50% indicated a preference for mobile apps as their primary platform for mobile shopping, while 50% preferred mobile websites (see Fig. 3).
C. Importance of Personalized Product Recommendations

A significant proportion of respondents (85%) expressed that personalized product recommendations were important in their mobile shopping experience (Fig. 4).

D. Privacy Concerns

Approximately 95% of respondents indicated concerns about the privacy of their personal data when using mobile shopping apps (see Fig. 5).

E. Trust in AI-Powered Virtual Assistants

The survey results demonstrated positive attitudes towards AI-powered virtual assistants, with 90% of respondents expressing a likelihood to trust such assistants for mobile shopping recommendations as shown in Fig. 6.

F. Factors Encouraging Data Sharing

When asked about factors that would encourage them to share personal data for personalized mobile shopping experiences Fig. 7, the respondents' preferences were as follows:

1) Discounts and personalized offers: 50%
2) Enhanced convenience and time-saving: 30%

3) Trust in the platform's security measures: 20%
G. Satisfaction with Personalization
A significant majority (95%) of respondents expressed satisfaction with the current level of personalization in mobile shopping apps (see Fig. 8).

H. Perception of Big Data and AI Impact
When asked about the extent to which they believed big data and AI could improve their mobile shopping experience, (see Fig. 9) the responses were as follows:

- Significantly: 50%
- Moderately: 35%
- Slightly: 15%

I. Willingness to Pay a Premium
A considerable number of respondents (70%) indicated a willingness to pay a premium for products or services that leverage big data and AI to enhance their mobile shopping experience as shown in Fig. 10.

These results indicate a positive sentiment towards the integration of big data and AI in mobile shopping among the surveyed participants in Jordan. The majority of respondents value personalized product recommendations and recognize the potential benefits of leveraging big data and AI technologies. However, privacy concerns remain a significant consideration for consumers. The findings suggest an opportunity for businesses to enhance their mobile shopping offerings by utilizing AI-powered virtual assistants and implementing robust data security measures to build consumer trust and satisfaction.

VI. CHALLENGES
The implementation of big data and AI technologies in businesses, particularly in the context of mobile shopping, presents several challenges that need to be addressed. The following are some key challenges faced by businesses:

1) Data Privacy and Security: As businesses collect and analyze customer data for personalized experiences, privacy concerns become paramount. Compliance with data protection regulations, such as GDPR, becomes essential.

2) Infrastructure and Scalability: Managing large-scale data and AI systems requires robust infrastructure, including storage, computing power, and network capabilities. Scaling up the infrastructure to accommodate increasing data volumes and processing requirements can be costly and complex.

3) Ethical and Bias Considerations: AI algorithms and models are susceptible to biases and unfair outcomes if not carefully designed and monitored.

4) Change Management and Organizational Culture: Implementing big data and AI technologies often requires significant changes in workflows, processes, and organizational culture.

5) Cost and Return on Investment: Implementing big data and AI technologies can involve substantial upfront costs, including infrastructure, software, and talent acquisition. It is
essential for businesses carefully evaluate the expected return on investment (ROI).

Despite these challenges, businesses that successfully overcome them can unlock significant benefits. Such as, improved customer experiences, enhanced operational efficiency, and informed decision-making.

VII. DISCUSSION

The study aimed to investigate the current state of mobile shopping in Jordan and the integration of big data and AI technologies in this context. The findings shed light on various aspects related to consumer behavior, preferences, satisfaction levels, and the potential impact of leveraging big data and AI in mobile shopping. The survey results revealed that a significant portion of respondents in Jordan actively engages in mobile shopping. This indicates the growing popularity and acceptance of mobile shopping as a convenient and accessible way to make purchases [3]. The preference for mobile apps as the primary platform for mobile shopping was prominent among respondents, highlighting the importance of mobile app development for businesses operating in the mobile shopping landscape [1].

Personalization and Recommendations: The study found that personalized product recommendations played a crucial role in enhancing the mobile shopping experience for consumers in Jordan. The majority of respondents expressed the importance of receiving personalized offers and recommendations based on their preferences and past shopping behavior [2][8]. This suggests the potential of leveraging big data and AI technologies to provide tailored recommendations and enhance customer satisfaction [1]. The survey revealed that privacy concerns regarding personal data shared during mobile shopping were present among respondents in Jordan. This indicates the need for businesses to prioritize data security and establish trust with consumers by implementing robust security measures and transparent data handling practices [3][9].

The study explored the trust levels of respondents in AI-powered virtual assistants for mobile shopping recommendations. The findings showed a varying degree of trust, with a significant number of respondents expressing a likelihood to trust such assistants [2][10].

VIII. CONCLUSION

In conclusion, the findings of this study highlight the significance of mobile shopping in Jordan and the potential impact of leveraging big data and AI technologies in this context. The study revealed consumer preferences for mobile apps and the importance of personalized recommendations. However, privacy concerns also emerged, emphasizing the need for businesses to prioritize data security and build consumer trust.

The study suggests that businesses can enhance the mobile shopping experience by utilizing AI-powered virtual assistants and leveraging big data to provide personalized recommendations. By doing so, businesses can improve customer satisfaction and potentially drive economic growth in the mobile shopping sector.

REFERENCES

Optimizing Drying Efficiency Through an IoT-based Direct Solar Dryer System: Integration of Web Data Logger and SMS Notification


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Abstract—Various agricultural and culinary products are dried to extend their shelf lives, mostly for marine foods. In many coastal locations of the Philippines, drying fish traditionally is still practiced, although study has shown that due to weather conditions and other factors, this technique is not seen to be reliable or cost-effective. The Internet of Things (IoT)-based Direct Solar Dryer System is optimized for drying efficiency by combining a web data logger and SMS notification system using Arduino Uno and ESP-32 to address difficulties with reliability and cost effectiveness. The study focuses on the potential and system efficiency of drying Sardinella fish (Tamban) in Brgy. Calibunan, Agusan Del Norte, Philippines; as well as investigating and assessing temperature, heat index, humidity, and temperature range alert conditions using a web application portal to serve as a remote monitoring platform for dependable data visualizations. The system delivered the expected results because the direct solar drier was able to raise and maintain the requisite temperature to accelerate drying while keeping the acceptable relative humidity. Furthermore, the system's monitoring and notification capabilities, as well as effective data collecting and data display via physical and remote monitoring, are supported by SMS notifications. As a result, the effectiveness of upgrading traditional sun drying with IoT technology can help reduce the challenges and disadvantages that fish drying farmers have faced. The study, with correct drying monitoring criteria, could serve as a model for other food products that can be dried.

Keywords—Arduino Uno; Internet of Thing (IoT); solar dryer system; web application portal; ESP-32; SMS notification

I. INTRODUCTION

Filipinos consume the most fish and fishery items. 11.68% of total food intake is made up of fish and fishery products. This works out to 93.90 grams each day. This was 63.0% higher than the price of beef and meat products; more expensive than poultry by 205.86%. Filipinos consume an average of 34.27 kg of fish and fish products per year, accounting for 23.36% of overall consumption, according to the DOST-FNRI survey report for 2018-2019. This includes 23.35 kg fresh fish, 2.85 kg dried fish (as fresh fish), 4.97 kg processed fish, and 3.10 kg frozen fish, Crustaceans and mollusks (estimated fish consumption based on population) [5][9]. After a half-decade the data of total fisheries output increased by 2.2 percent to 4,339.89 thousand metric tons in 2022, from 4,248.26 thousand metric tons the previous year [14][18]. Increases in production were observed in maritime municipal fisheries and aquaculture, whereas commercial and inland municipal fisheries saw setbacks during the year [7]. In Calibunan, a coastal barangay in Cabadbaran City, Agusan del Norte in the Caraga Region in the Philippines, the drying of fish is one of the sources of living in the community. Fish dried in the open ground while others were hung on the racks with a net, or a flat board bamboo called “kaping”. The traditional drying or sun drying method is prone to inconvenience and is labor-intensive and known for its slow process that can cause product losses apart from an issue concerning the quality of the product when it comes to cleanliness. Traditional sun-drying may cause microorganism growth, insect infestation, and other potential food safety hazards that may occur in the food due to the inadequate drying process [6].

The objective of the research is to utilize emerging technologies in order to develop an IoT-based Direct Solar Dryer System with a Web Data Logger and SMS notification. The ultimate goal is to enhance the efficiency of food drying operations [10] and fill the gaps identified in related studies within the research methods. The findings of the research would help to promote knowledge on developing innovative solutions in the field of Internet of Thing (IoT) monitoring systems for food drying [15]. Furthermore, this research highlights the power of technology in enhancing the lifestyles of fish drying farmers, supporting breakthroughs in the food drying sector, and encouraging future research efforts. The paper is organized as follow: Section II presents the discussion of the three phases in research methodology, design and development of the system parameters, coding the firmware and creating a web data analysis using MATLAB based on ThingSpeak Application monitoring. Finally, Sections III and IV present the evaluation results and conclusion, respectively.

II. RESEARCH METHODOLOGY

The research is divided into three (3) phases in Fig. 1 Research Framework, each of which is dependent on its upper processes.

The Input phase was when the researchers acquired essential information on the actual activities in the drying field, the issues they encountered, and the current ideas and studies that would help accomplish the intended system output.
The Process phase, during which the obtained data was processed and put to use. This phase saw the implementation of the system's components and functionality.

In the Output phase, the desired output such as physical monitoring, web monitoring, and SMS sending to users is produced.

![Fig. 1. Research framework.](image)

**A. Analysis of Gathered Related Literature**

1) Preliminary survey: The researchers conducted a preliminary survey on the Research Locale Map at Barangay Calibunan in Cabadbaran City, Agusan del Norte in the Caraga Region of the Philippines in Fig. 2, and the data gathered was appraised by the researchers as valuable in the study.

![Fig. 2. Research locale map "Barangay Calibunan in Cabadbaran city, Agusan del Norte in the Caraga region in the Philippines" - google maps.](image)

In Fig. 3, 68.4% of respondents "Strongly Agree", that there were concerns with sun drying methods such as human work, insect infestation, and the number of days required to dry the fish. With 12.63% of respondents “Agree” that a need for technology innovations may apply for fish drying, 14.21% of respondents “disagree” of changing traditional way to more efficient drying technology and 4.75% “Strongly disagree” of the use of innovative solutions. It’s because of the cost of maintaining the system. Overall, according to the comments of the respondents, adding a monitoring system is a terrific innovation to improve their process.

2) Types of Sardinella fish found in the Philippines: In the Philippines, several species of Sardinella fish waters presented in the Table I can be found namely: Bali sardinella, Goldstripe sardinella, Fringescale sardinella, White sardinella, Taiwan sardinella, Freshwater sardinella, Spotted sardinella, White sardine, Blacksaddle herring, Bluestripe herring, and Rainbow sardine [22].

The research focuses on the design concept of drying Sardinella fish, also known as "Tamban" or "Tunsoy" in the native language, which is a widely available fish species in the Philippine seas. The main objective of the research is to determine the appropriate system parameters for drying Sardinella fish and establishing threshold values for optimal drying [11]. By studying the drying process of this particular fish species, the researcher aims to develop a comprehensive understanding of its drying characteristics and identify the key factors that affect the quality and efficiency of the drying process. The design concept of drying Sardinella fish serves as the foundation for adjusting the main system parameters. This concept provides valuable insights into the drying requirements specific to Sardinella fish, including temperature, humidity, air circulation, and drying time. By establishing threshold values based on these parameters, the research aims to optimize the drying process and ensure consistent quality and preservation of the fish. The findings of this study will contribute to the development of effective and efficient drying techniques for Sardinella fish, benefiting fish drying farmers and enhancing the overall fish drying sector in the Philippines.
3) Related studies on fish drying with the use of technology: The researchers gathered information on several related studies presented in Table II, these studies were examined through their innovations in the use of technologies in the system design and scrutinized the research gaps in the common technologies being implemented that were accessible in recent years [2].

The researchers found that appropriate design of the system by creating a solar dryer/storage unit using sensing components/devices: first for weighing the fish with load cell weight sensor, second by checking the temperature, heat index, and humidity using DHT22 sensor, third using fan control system to reduce humidity and increase more heat inside the storage.

The innovative solutions are introduced in this paper by integrating an SMS notification system as well as a web application portal to serve as a remote monitoring platform using ThingSpeak or dependable data visualizations in temperature range alert conditions for notifications to the users’ smart mobile messaging [4].

B. Architectural Design

Fig. 4 illustrates the System architecture diagram which explains how the prototype is connected and how the connection relates to each component. The power bank acts as the power supply of the system except for the fan which has its own power supply of 12V. All sensors and actuators are connected to the Arduino Uno, and from the Arduino Uno, all the data are then transmitted to ESP32-CAM and to GSM/GPRS Module. The endpoint of all data being transmitted is in ThingSpeak and Web Database analysis using MATLAB, where all raw data are converted into information.

### TABLE I. TYPES OF SARDINELLA FISH FOUND IN THE PHILIPPINES

<table>
<thead>
<tr>
<th>Type #</th>
<th>Scientific Name</th>
<th>International Name</th>
<th>Philippine Local Name</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>Sardinella lemuu</td>
<td>Bali sardinella</td>
<td>Tamban/Tunsoy</td>
</tr>
<tr>
<td>2</td>
<td>Sardinella gibbosa</td>
<td>Goldstripe sardinella</td>
<td>Tamban/Tunsoy</td>
</tr>
<tr>
<td>3</td>
<td>Sardinella fimbriata</td>
<td>Fringesh sardinella</td>
<td>Tamban/Tunsoy</td>
</tr>
<tr>
<td>4</td>
<td>Sardinella albella</td>
<td>White sardinella</td>
<td>Tamban/Tunsoy</td>
</tr>
<tr>
<td>5</td>
<td>Sardinella hualiensis</td>
<td>Taiwan sardinella</td>
<td>Tamban/Tunsoy</td>
</tr>
<tr>
<td>6</td>
<td>Sardinella tawilis</td>
<td>Freshwater sardinella</td>
<td>Tawilis</td>
</tr>
<tr>
<td>7</td>
<td>Amblygastersia</td>
<td>Spotted sardinella</td>
<td>Tamban/Tunsoy</td>
</tr>
<tr>
<td>8</td>
<td>Escualos thoracata</td>
<td>White sardine</td>
<td>Bolinaw</td>
</tr>
<tr>
<td>9</td>
<td>Herklotsichthys disiplonotus</td>
<td>Blacksaddle herring</td>
<td>Dilat</td>
</tr>
<tr>
<td>10</td>
<td>Herklotsichthys quadriraculatus</td>
<td>Bluestripe herring</td>
<td>Dilat</td>
</tr>
<tr>
<td>11</td>
<td>Dussumiera acuta</td>
<td>Rainbow sardine</td>
<td>Tulis/Alabaybay</td>
</tr>
</tbody>
</table>

### TABLE II. RELATED STUDIES ON FISH DRYING WITH THE USE OF TECHNOLOGY

<table>
<thead>
<tr>
<th>No.</th>
<th>Title</th>
<th>Authors</th>
<th>Technology use in the system</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>IOT Based Solar Dryer and Irrigation System (2022)</td>
<td>Prof. Meena Ugur, Mr. Ankur Fouljdar, Mr. Sushil Nikumbh, Mr. Suyash Joshi</td>
<td>Solar Dryer with Solar panel in drip irrigation [15]</td>
</tr>
<tr>
<td>2</td>
<td>Internet of Things-Based Crop Classification Model Using Deep Learning for Indirect Solar Drying (2022)</td>
<td>Brijesh Sharma, Gaurav Gupta, P. Vaidya, Shakhila Basheer, F. H. Memon, R. N. Thakur</td>
<td>Temperatur e of the solar dryer [3]</td>
</tr>
<tr>
<td>4</td>
<td>A review on solar dryers integrated with thermal energy storage units for drying agricultural and food products (2021)</td>
<td>G. Srinivasan, D.K. Rabha, P. Muthukuma r</td>
<td>Thermal energy storage unit dryer types, product dried, operating parameters, sensible and latent heat storage materials [8]</td>
</tr>
<tr>
<td>6</td>
<td>Development and Quality Analysis of a Direct Solar Dryer for Fish (2018)</td>
<td>S. O. Obayopo, O. I. Alonge</td>
<td>Moisture Content of Dried Fish</td>
</tr>
</tbody>
</table>
C. Design and Setup of IoT-based Direct Solar Dryer System: Integration of Web Data Logger and SMS Notification

Integration of Web Data Logger and SMS Notification with a physical monitoring system and web application to display valuable data is shown in Fig. 5, System Diagram, which includes a.) Thermal Storage Unit System [1][13], b.) Cloud Data Storage, c.) Visualization, d.) Web Monitoring [23]. Fig 6 illustrates the hardware components connection that includes DHT22, Weighing Sensor, SIM800L GSM/GPRS Module, D. 20x4 LCD (Liquid Crystal Display), Relay Module, ESP32-CAM, LED with Rocker Switch, and SPDT.

D. Firmware Development

In the firmware development stage, the researchers utilized C/C++ programming language for the Arduino Uno and ESP32-CAM microcontrollers to program the sensors, actuators, and monitoring functions. The open-source Arduino IDE was used to simulate and compile the codes, while GitHub facilitated collaborative coding during the pandemic. The web application was developed using Visual Studio Code, with HTML, PHP, CSS, and JavaScript for coding. MySQL was used for local testing before deployment on web hosts, and ThingSpeak software platform integrated with MATLAB visualization for instant chart generation in the web application.

For integration and testing, the researchers conducted tests and debugging to ensure proper functionality of the hardware components and correct code execution. Once the solar dryer and hardware codes were finalized, the system underwent thorough testing.

E. System Performance and Evaluation Result

The evaluation was divided into two parts: analyzing the concept dryer's performance and evaluating the complete system with functional components [19]. Based on relevant investigations and data collection, a one meter squared direct sun dryer was tested to examine its performance in drying Sardinella fish under ideal weather conditions and within an experimental temperature range of 35-55°C. The whole system evaluation included testing the fish dryer's functionality and assessing its capacity to meet the targeted goals. Temperature, humidity, heat index, weight, moisture content, and presumed time were measured using an Arduino Uno, retrieved via ThingSpeak, and saved in a database. Intervals of data communication were created between the ESP32-CAM, ThingSpeak, and the web database analysis using MATLAB.

The researchers established a minimum temperature of 35°C and a maximum temperature of 55°C to ensure the desired temperature range for effective drying.

III. RESULTS AND EVALUATION

In this section, the study's results are presented, providing detailed information about the methods and processes employed. The discussion accompanying the results analyzes and interprets the findings, providing insights into their significance and implications. Additionally, the section addresses the observed limitations and constraints that impact the functionalities and capabilities of the system.
limitations are crucial to understanding the boundaries within which the system operates and the factors that may affect its performance. By presenting the results, discussing their implications, and acknowledging the system's limitations, this section provides a comprehensive understanding of the study's outcomes and contributes to the broader knowledge in the field.

A. Actual IDSD-SMS System

The actual testing of the system was conducted from 11:00 AM, due to its good and enough sunlight, and lasted until 3:09 PM. During the testing, one (1kg) kilogram of Sardinella fish was prepared, and approximately 410 grams was achieved after the preparation performed. In Fig. 7 the actual image of the dryer during testing is shown. All the data needed for visualizations and data interpretations were collected and stored in the web application database. Fig. 8 illustrates the current and target weights, presumed time, moisture content. Fig. 9 presents the actual LCD system display setting up display, initializing setup, reset and done drying.

B. Web Application Portal

The web application, created by the researchers, acts as a remote monitoring platform that displays reliable data visualizations such as spline charts, gauge, numeric, and comparison charts (see Fig. 10). The web application also has a
database to store raw data. The web application is accessible through the World Wide Web using the browser to log in and have access. When the system is on, and connected to the WiFi, a user can monitor the current drying session instantly anywhere in the globe [21].

C. Web Application Monitoring

Fig. 11 presents the save raw data collected from the system monitoring this shows a data log of 15 seconds interval from the time stamp, temperature, humidity, heat index, weight, moisture content and the remaining time of the presumed drying time. Data can now be inputted in the MATLAB analysis algorithm to output the data visualizations.

D. Comparison of the Temperature Inside and Outside the Dryers

Fig. 12 using MATLAB visualization data analysis shows the comparison between the inside and outside temperature during the conduct of the study. Based on the chart, the temperature inside was always greater than the temperature outside, as expected ranging from 45.37 - 56.9 °C which is organoleptically excellent quality according to Reza (2002)[16]. The system recorded the inside highest temperature of 56.9 °C and 46 °C on the outside, while the lowest temperature was observed at 45.37 °C on the inside and 26.33 °C outside.

E. Weight and Moisture Content Relationship

The weight and moisture content are the most critical aspects in terms of drying foods. Throughout the drying session, the weight and moisture content were monitored. As Fig. 13 illustrates using MATLAB, the weight was proportional to the moisture content which means that for every loss of weight, there was also a loss in the moisture content. The researchers started drying with a total weight of 407.83 grams and its target weight of 95.96 grams which was considered safe and dry as it reached a moisture content of 15%. The desired weight was achieved after 4 hours of drying due to the high and constant temperature the fish receives. The temperature slowly goes down from 2:40 PM onwards.

Throughout the drying session, the web application collected and displayed data. The interval of receiving data from the system to the web application took about every 15 to 16 seconds depending on the network speed or/and latency. The researchers use three different kinds of devices such as the laptop, tablet, and mobile phone in monitoring the web application.
During the actual testing, the solar dryer was successful in gathering and displaying the needed information for the drying process. Moreover, the system was able to send SMS notifications to the user and the fan worked well according to its intended task. Therefore, the efficacy of the system IoT-based Direct Solar Dryer System: Integration of Web Data Logger and SMS Notification was able to achieve the desired results based on the conducted testing conditions in Optimizing Drying Efficiency. The direct solar dryer was able to increase the temperature up to 23% and maintain the temperature which can speed up the drying time while achieving the recommended relative humidity ranging from 24.9% up to 58.71%. The physical and remote monitoring was successful in gathering and displaying the needed information for the drying process. Moreover, the system was able to send SMS notifications to the user and the fan worked well according to its intended task. Therefore, the efficacy of improving the traditional sun drying with the use of technology can help alleviate the difficulties and disadvantages of the fish drying farmers.

ACKNOWLEDGMENT

This research work is done by the undergraduate students of Bachelor of Science in Computer Applications through the advice and support of the Department of Computer Application, College of Computer Studies MSU-IIT faculties and with the researcher’s collaboration to Local Drying Farmers of barangay Calibunan in Cabadbaran City, Agusan del Norte in the Caraga Region in the Philippines.

REFERENCES


Anti-Spoofing in Medical Employee's Email using Machine Learning Uclassify Algorithm

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Department of Information and Computer Science(ICS)-College of Computer Science and Engineering, University of Ha'il (UOH), Hail, Kingdom of Saudi Arabia

Abstract—Since the advent of COVID-19, healthcare and IT cybersecurity have been an issue. Digital services and foreign labor have increased cyberattacks. July 2021 saw 260,642 phishing emails, 94% of 12 countries’ employees experienced epidemic cyberattacks. Phishing attacks steal sensitive data from spam emails or legitimate websites for profit. Phishing spam uses URL, domain, page, and content variables. Simple machine-learning methods stop phishing emails. This study discusses phishing emails and patient data and healthcare employee accounts cybersecurity. This paper covers COVID-19 email and phishing detection. This article examines the message's URL, subject, email, and links. Uclassify classifies content, spam, and language and automates emails. Semi-supervised machine learning dominates healthcare. The Uclassify algorithm used multinomial Naive Bayesian classifiers. Document class is [0–1]. This article compared Multinomial Naive Bayesian in two experiments with other algorithms. Experiment 1 achieved an MNB accuracy of 96% based on a database from Kaggle Phishing. Experiment 2 showed that the Multinomial Naive Bayesian system accurately predicted URL and hyperlink targets based on PhishTank data. 96.67% of respondents correctly identified URLs, and 91.6% did so for hyperlinks. These two experiments focused on Tokenization, Lemmatization, and Feature Extraction (FE) and contained an internal feature set (IFS) and an external feature set (EFS). MNB is more exact than earlier methods since it uses decimal digits and word frequency. MNB only takes binary inputs. MNB can detect phishing and spoofing.

Keywords—Spoofing; phishing; machine learning; Uclassify algorithm; medical employee’s email

I. INTRODUCTION

The Internet’s quick services affect the world. Consumers can shop and bank anytime with improved Internet infrastructure. Despite its benefits, internet security and privacy are issues. Phishing, malware distribution, and privacy exposure are all possible with the Internet’s anonymity[1]. Emergency and patient care services require healthcare workers to exchange electronic health information with patients. This data is one of the most sensitive. Therefore, special measures must be taken regarding threats to confidentiality, integrity, and availability (CIA)[2]. Healthcare should prepare for cyberattacks. Attackers have used email to target healthcare companies since the COVID-19 outbreak began [3]. The pandemic increased health cybercrime. Digital change encourages "telework" or "work from home," which boosts email efficiency [4]. Remote operators must be trained for safety. More than 3,000 employees in 12 countries have experienced remote working, and 94% have experienced cyberattacks during the pandemic, according to a report by the International Association of IT Asset Managers (IAITAM) [5]. The Anti-Phishing Working Group (APWG) says there were 260,642 phishing attacks in July 2021, the most ever seen in one month. It rises from 44,008 in the first quarter of 2020 to 128,926 in the third quarter [6]. Phishing emails steal any account credentials. Banks and hospitals were attacked the most [7]. Spoofing deceives victims into divulging passwords, usernames, and personal information. Scammers entice recipients to visit fake websites or download viruses [1]. 16% of 2015 FTC complaints were about identity theft. Phishing and social engineering steal data. Phishing online fraud using a fake website and email [8].

URIs hurt or redirect phishing victims. Second, domain length, numbers, spelling, and brand name may impact Phishing—status, domain owner, and age impact URLs. Reputation determines page believability. Content-based domain scanning—hidden, body, meta, and pictures—estimates daily, weekly, or monthly page views, average page visits, internet traffic, domain category, and similar websites. Finally, scanners check page type, user, and registration [9].

A. Problem of Phishing Attack Detection

Study email security. Email security avoids loss, theft, and hacking. Spam, phishing, and malware utilize email to transfer sensitive data and access networks [10] Phishing persists despite email security. Phishing is inevitable. This thesis tests copyright checking without email sender server settings [11]. The goal: Email is often attacked. Inconsistent protocols, roles, and services compromise email security [12]. Spoofing websites steal customer data—most often fake websites. Machine learning detects spam and phishing sites [13].

These research questions are addressed in this paper as follows:

1) Which machine learning methods are used to create phishing detection models?
2) What datasets are used for phishing email detection models?
3) How can modern datasets and resources recognize phishing emails?
4) Which email features use Phishing detection emails?
5) What are the early stages of phishing and the current trend of phishing emails?

To answer these questions or to address these issues, the findings in the research could be:
1) SVM, LR, and DT recognize fake emails, whereas k-means clustering does not [14]. Multinomial Naive Bayesian classifiers power Uclassify ML. Its feature vector classifies input into the most likely class. This algorithm considers all input data irrelevant. Data sets are unaffected by changes.

2) ML-based model training and testing require a dataset. Phishing detection methods were developed utilizing pooled datasets. Producers often update long-lived datasets. 2021 revised Nazario’s dataset. 2010 spam email, 2005 phishing corpus, 2006 Enron spam, and 2002 spam assassin datasets are updated routinely [15].

3) Classify creates ten category numerical values after phishing using ML and web services—health. Uclassify web API will categorize questions by feature vector subject for this study. Uclassify contains ML classifiers for sentiment, themes, language, age, gender, and more.[16]. Customer emails initiate phishing. This malicious email links to an attacker-based website. Reassures email recipients:

   a) The sender and email address are not from "UMass Amherst it@umass.edu," despite the assertion.

   b) Phishing emails are misspelled. This email’s colon should not precede the comma.

   c) Phishing emails employ urgency. It spurrs action.

   d) The message URL is UMass Amherst’s webpage. Hovering reveals a different page.

   e) UMass Amherst and Microsoft Corporation are impersonated in the letter. Again, bogus if the sender needs to know who they are.

   f) The email link leads to a bogus SPIRE-phishing login page at "tantech ladyings.com." [6].

4) Extracting features from raw data entails retaining the original data set while converting it into numerical portions that can be processed. BoW, IG, and Word2vec are text characteristics in phishing email detection research. The research also utilized latent Dirichlet allocation, part-of-speech tagging, PCA, and LDA [6].

5) Phishing targets psychology and emotion. They employ social engineering and technology. Attackers’ personalization, clever phishing, and tactics increase prevention [17].

II. LITERATURE REVIEW

A. Introduction

Digital healthcare providers fear cyberattacks. Social engineering targets individuals. This lengthy research shows how cyber threat ignorance impacts healthcare—technical and organizational healthcare cyber security. Table I shows the authors’ five questions to classify literature by subject. Healthcare cyber defense was examined first. All personnel are studied [18].

Table I outlines this systematic review’s research questions and background [18].

<table>
<thead>
<tr>
<th>TABLE I.</th>
<th>SYSTEMATIC REVIEW’S RESEARCH QUESTIONS AND BACKGROUND</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Background</strong></td>
<td><strong>Research Questions</strong></td>
</tr>
<tr>
<td>Social engineering cyberattacks against healthcare workers are the most effective. These assaults use social media data.</td>
<td>RQ1: What are the prevalent social attacks perpetrated against individuals by healthcare organizations?</td>
</tr>
<tr>
<td>Data governance, encompassing data security, privacy, and IT infrastructure security, protects organizations against cyberattacks. Several methods prevent DDoS assaults. WannaCry has increased healthcare data leaks.</td>
<td>RQ2: Which policies and governance mechanisms have strengthened healthcare organizations?</td>
</tr>
<tr>
<td>Cyber risk assessment improves healthcare. Data breaches are expected as healthcare becomes more complicated. IT security dominates risk assessments. Social engineering requires reassessing healthcare cyber threats from insecure human behavior.</td>
<td>RQ3: How does an organization’s cybersecurity risk assessment incorporate human elements in cybersecurity?</td>
</tr>
<tr>
<td>Healthcare companies now train on cybersecurity—for example, phishing email training. Given recent examples of using social media data to target healthcare practitioners, raising awareness of this evolving, dangerous environment is vital.</td>
<td>RQ4: How can training raise healthcare workers’ cyber threat awareness, and how can we quantify an organization’s training and awareness efforts?</td>
</tr>
<tr>
<td>Europe needs healthcare infrastructure. Outages may generate national emergencies. ENISA advises on cyber resilience. Multi-nation cyberattacks have had enormous economic and human repercussions.</td>
<td>RQ5: Which national and international organizations offer cyber defense strategies to boost cyber resilience?</td>
</tr>
</tbody>
</table>

Fig. 1 illustrates a phishing email, whereas Fig. 2 describes its essential elements. PhishTank hosts samples and notes.

![Fig. 1. Phishing email example][8].

![Fig. 2. An example of a phishing email with annotations highlighting its key components][8].
The phisher's email connects to a bogus website (Fig. 3). This fake website was created to deceive email recipients. The attacker may utilize Fig. 3's remote data input area. Phishers then emailed and messaged AOL subscribers. Phishers asked AOL users for their account numbers. AOL's TOS couldn't track attackers' AIM accounts, worsening the issue. Finally, AOL sent emails and instant messages pushing users to withhold vital information [8].

![Phishing website](image)

**Fig. 3.** Phishing website [8].

Blacklisting and whitelisting virus scanners safeguard internet users (see Fig. 4). Commercial software combats zero-day phishing. Microsoft, Google, and PhishTank blacklists let researchers test solutions. Anti-phishing may block and whitelist phishing URLs on the client's PC or server. Users trust whitelists. Zero-day phishing detection takes accuracy. Whitelists may mistake good websites for phishing. AIWL records user-irritating visual effects and manual white-list maintenance. Multinomial Naive Bayesian classifiers monitor AIWL logins. AIWL whitelists login URLs. The hardest part is picking a trustworthy site from a fraud list. Finally, blacklists and heuristics may whitelist. Avoiding trustworthy websites may expedite phishing detection. URL blacklists stop Phishing. Anti-phishing blacklists URL-verified phishing. Blacklists' low false-positive rate and simplicity make them famous. The research discovered that blacklist software missed 80% of zero-day phishing assaults. Blacklist users enjoy its simplicity and low false positive rate. Blacklists' low false-positive rate and simplicity make them popular [8].

**Fig. 4.** Phishing prevention technology [8].

**B. Definition of the Spoof**

The official definition is masquerade: "A type of threat action whereby an unauthorized entity gains access to a system or performs a malicious act by illegitimately posing as an authorized entity" and Spoof: "Attempt by an unauthorized entity to gain access to a system by posing as an authorized user" and phishing attack is: "A technique for attempting to acquire sensitive data, such as bank account numbers, through a fraudulent solicitation in email or on a website, in which the perpetrator masquerades as a legitimate business or reputable person" [19].

**C. Related Works**

Technology introduces security vulnerabilities. In the 1960s, "phone hacking" developed access control and encryption. AOL hackers coined "phishing" in 1996 after obtaining private data. Phishing emails verified AOL customers' credentials. Many provided hackers their login details to purchase items. Customers pay millions. eBay, HSBC, and others fight phishing. These techniques identify fake emails and websites. Despite its effectiveness, only some read online phishing prevention literature. ML finds fraud. Dots, domain ages, and links checked URLs for Spoofing or Phishing. Consumer education prevents phishing and impersonation as threats rise. Scammers may be caught. Scam emails conceal COVID-19 phishing was examined. Cyberattacks grew. Backup data, secure remote worker networks, communicate with IT, and educate staff in the attack. COVID-19 and $6 trillion hacking by 2021 ended it. This article covers multifactor login, VPNs, new hacking regulations, and IT-employee communication. Pandemic hackers targeted hospitals. Phishing, ransomware, homework, and government attacks boost hacking threats. Education, VPNs, multifactor authentication, firmware upgrades, and a firm safety policy decrease these hazards. Spoofing needs numerous countermeasures: Fake COVID-19 affects ML [5]. Phishing and impersonation use distinct strategies. Fig. 5 shows Internet and phone; email, IM, and social media work well. Theft motivates these assaults. It accurately detects phishing emails, URLs, IPs, and images [5].

![Phishing attack](image)

**Fig. 5.** Phishing attack [5].

SMTP is the Internet protocol for transmitting email. Fig. 6 depicts the three essential email message submission processes [22].

1) The sender's MUA sends the message to the service provider's MSA through SMTP or HTTP/HTTPS (MSA).

2) The recipient's email provider receives the message via SMTP from the sender's MTA.
3) The user receives the message over HTTP/HTTPS, POP3, or IMAP via the Mail Delivery Agent (MDA) (IMAP) [22].

![Fig. 6. Alex and bob's email transmission [22].](image)

III. RESEARCH METHODOLOGY

4.5 billion people use the Internet. Email is trusted online. Fraudulent emails include malware or unsafe URLs. Even with better filtering, spam emails' constantly shifting content makes them hard to distinguish. Corporate email, commercial antispam services, and end-user training filter spam, yet this deadly trap caught non-experts. It trains SVM, Multinomial Naive Bayesian, CNN, and LSTM spam email detectors. This article offers different ML models sans spam email datasets. CNN and LSTM utilize Model Loss and ROC-AUC; MNB and SVM need precision, recall, and f-measure. Finally, all models are compared for great accuracy and DL and ML model evaluation parameters. English and spam detection enhanced [23]. Phishing detection has been improved. ML algorithms shine. ML algorithms will be studied with an accuracy of 94.4%. Content-based filters identify fake emails [24]. ML-based spam classification. Uclassify's own naive polynomial Bayesian classifier. Classification limits are calculated using the document category probability 0-1 [25]. The proposed solution considers the detection aspects of phishing attacks, URLs, and domains. ML will fight phishing attacks Fig. 7 depicts the Phases of a typical Spear-phishing attack [26].

![Fig. 7. Phases of a typical Spear-phishing attack.](image)

An attacker gathers as much sensitive target information as possible during preparation (S1). Leak social media and databases. Knowing the victim eases persuasion. Spam-detecting email infrastructure demands technological skills—clean, succinct, balanced email content (S2). The attacker then wants S3 opened. Email subjects matter. Secure email. The reader gets balanced (S2). The attacker then wants S3 unlocked. Email subjects matter. Secure email. Email motivates. Certain emails may create confidence. Avoid downloading attachments (S4). Preventing assaults requires security measures. Antivirus and OS updates protect. S5 attacks browsers, devices, and credentials [26]. Table II outlines our suggestions. "Use of @ symbol," "right-click blocked," and "hiding suspicious links" are three of the seventeen traits we are eliminating. RFC engineers and computer scientists accept @, " and # in URLs. Browsers no longer allow status bar changes to turn off right-clicking and hide suspicious links. Invalid characteristics. 54-character URLs are questionable. Most bogus URLs are under 54 characters [32]. It impairs judgment. We verified the URL length. Chrome and IE allow 2083-character URLs. Thus, a URL with 1000–1750 characters is suspicious, but phishing with more.

<table>
<thead>
<tr>
<th>#</th>
<th>Feature</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>IP domain names</td>
<td>Using IP address in domain part is phishy because attacker is trying to disguise name with numbers.</td>
</tr>
<tr>
<td>2</td>
<td>Long URL</td>
<td>Long URLs disguise questionable keywords. URLs over 1750 are phishy.</td>
</tr>
<tr>
<td>3</td>
<td>&quot;-&quot; symbol</td>
<td>The domain's &quot;-&quot; indicates legitimacy. Example: Pay-Pal.com</td>
</tr>
<tr>
<td>4</td>
<td>Sub-domain(s) in URL</td>
<td>Phishing URLs have several subdomains.</td>
</tr>
<tr>
<td>5</td>
<td>Use of HTTPS</td>
<td>HTTPS secures URLs.</td>
</tr>
<tr>
<td>6</td>
<td>Request URL</td>
<td>The URL domain should load all text and graphics.</td>
</tr>
<tr>
<td>7</td>
<td>URL of anchor</td>
<td>All &lt;a&gt; tags should have domain-matching links.</td>
</tr>
<tr>
<td>8</td>
<td>Server form handler &quot;SFH.&quot;</td>
<td>User data may be transferred. Checking Server Form Handler prevents this.</td>
</tr>
<tr>
<td>9</td>
<td>Abnormal URL</td>
<td>The URL's WHOIS data confirms its identity—no phishing website.</td>
</tr>
<tr>
<td>10</td>
<td>Redirect page</td>
<td>Links sometimes redirect users to other pages. Phishy redirects exceed four.</td>
</tr>
<tr>
<td>11</td>
<td>Using pop-up Window</td>
<td>Pop-up password entry is unethical. Pop-up sites are phishing.</td>
</tr>
<tr>
<td>12</td>
<td>DNS record</td>
<td>Phishy URLs lack DNS records.</td>
</tr>
<tr>
<td>13</td>
<td>Website Traffic</td>
<td>Website traffic visits. Websites without traffic records are fishy.</td>
</tr>
<tr>
<td>14</td>
<td>Age of domain</td>
<td>Domains age from registration. Phishing site registered under one year.</td>
</tr>
</tbody>
</table>

D. Research Design

Businesses and people email. Spammers make money. Bio-inspired machine learning identifies fake emails in this study. A study analyses how to use varied datasets for successful results. Genetic and Particle Swarm Optimization improved classifier performance. Features or automatic parameter selection may assess spam categorization algorithms. Comparing recommended and basic models will determine whether parameter changes enhance them [28].

E. Dataset

The detection approach is crucial to the proposed system, but the datasets used by the authors to test and train their algorithms affect their credibility. Website detection datasets match email detection datasets, showing no concerns. Spam and viruses sometimes are utilized. However, the papers discuss fraudulent email detection. These publications are harmful (the spam dataset contains spam email URLs). Online datasets for fraud detection algorithms are available—Table III lists prevalent phishing and ham datasets.
TABLE III. PRESENTS THE FEATURES OF THE DATASET [6]

<table>
<thead>
<tr>
<th>#</th>
<th>Dataset feature</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>Dataset source</td>
<td>The generally utilized data sources of legitimate and phishing websites, along with the approaches that grip every head, are mentioned; however, the insufficient understanding regarding the methodologies used in collecting and preserving every source results in no concord with all regarding the quality of various origins.</td>
</tr>
<tr>
<td>2</td>
<td>Dataset size</td>
<td>The evaluation dataset size differs between various approaches. As seen, the reliable outcome depends on the size of the dataset; the bigger the better.</td>
</tr>
<tr>
<td>3</td>
<td>Dataset redundancy</td>
<td>There is not sufficient information in the literature regarding dataset redundancy. Although numerous presentations and overly between various sources of datasets, particularly of phishing websites, can be seen</td>
</tr>
<tr>
<td>4</td>
<td>Dataset timeliness</td>
<td>Although if a similar source of data and size of the dataset is utilized in two plans, their phishing website's information might not be the same. The phishing blacklist supplier generally amends their data hourly, because phishing websites last for short terms.</td>
</tr>
<tr>
<td>5</td>
<td>Ratio of legitimate to phishing websites</td>
<td>The ratio of legitimate to phishing examples displays the level at which experiments portray an actual world distribution (≤100/1).</td>
</tr>
<tr>
<td>6</td>
<td>Training set to testing set ratio</td>
<td>The extensibility of the approach is seen in the ratio of training to testing examples.</td>
</tr>
</tbody>
</table>

This study uses the 2020 Akashsurya156-revised Kaggle Datasets Phishing Email Collection. 22 traits, 21 predictors, and one target variable define email authenticity. Each feature improves model learning. The most deceptive electronic communication phrases were found—525,754 emails in CSV. Spam and fake emails are rare, skewing the classification dataset. Campaigners tested fake and real emails. These two groups balance actual and fraudulent emails in training models [29]. Fig. 8 shows information used for spoofing URL recognition [30].

![Fig. 8. Information used for spoofing URL recognition.](image)

Phishing and URL data are needed for ML model training. A phishing crawler's program that gathers phishing URLs from the PhishTank website only records the web address if the site is active. Phish Search finds phony URLs that change. BeautifulSoup pulls page code. Using "identifiers" and queries, we may identify the request's absolute phishing URL. 10,000 phishing and 10,000 non-phishing URLs were crawled from the dataset. The training uses 8,000 URLs and evaluation 2,000. Table IV summarizes the study data. Fig. 8 depicts data distribution [30].

TABLE IV. DATA ON PHISHING AND LEGITIMATE URLs [30]

<table>
<thead>
<tr>
<th>URL</th>
<th>Actual Data</th>
<th>Used Data</th>
<th>Train</th>
<th>Test</th>
</tr>
</thead>
<tbody>
<tr>
<td>Benign URL</td>
<td>17058</td>
<td>10000</td>
<td>8000</td>
<td>2000</td>
</tr>
<tr>
<td>Phishing URL</td>
<td>19653</td>
<td>10000</td>
<td>8000</td>
<td>2000</td>
</tr>
</tbody>
</table>

F. Detection Method of Phishing Attack Through Email

Semi-supervised ML identifies bogus emails and impersonation. Fig. 12 depicts categorization. Semi-supervised learning employs small labeled and unlabeled examples. Binary classification identifies phishing emails. The automatic categorization is increasing ML-detected phishing. ML vectors describe emails and web pages. Fake emails and sites are 1. Label 0 denotes a valid email or page. Semi-supervised teaching detects and evaluates phishing emails and web pages. Fig. 11 shows the planned steps. Detect spoofing, phishing, and impersonation emails. These safeguard data and prevent fraud—these secure critical data. Establish email components.

![Fig. 9. Taxonomy for email messages [31].](image)

From, To, Subject, and Message-ID—the message's content—make up the email header. Fig. 9 and 10 exhibit email taxonomy and arrangement. Emails to scam websites. Email URLs trick consumers into providing critical information. Many studies have revealed email spoofing characteristics and methods to identify fake emails. Primary, latent topic, and dynamic Markov chain features. Researchers found anti-phishing techniques. Machine learning, deep learning, word embedding, and NLP are discussed [31].

![Fig. 10. The format of an email message [31].](image)
1) **Tokenization**: Computationally splitting a text into words or tokens [32]. Thus, it trims the input text word by word. Next, filtering removes unnecessary words from token results and finds the root term of each filtered word. Tagging then finds the root form of a preceding word or stem word. Finally, analyzing texts finds word relationships [33].

2) **Lemmatization**: Removing inflectional suffixes from words returns the base form. Thus, it turns words into their roots. This removes the inflectional ending and restores the root. It uses lexical and morphological analysis to find a word's root form, unlike stemming. The word “better” means “excellent.” [32, 34].

3) **Feature Extraction (FE)**: Lexical components, conceptual frameworks, and linguistic phrases make text categorization difficult. Thus, learning from a large dataset takes a lot of work. Therefore, it's computationally expensive. Extraneous and redundant features also impair categorization algorithm accuracy and effectiveness. Use just the key elements to simplify and accurately classify data with little duplication [35].

Our classification has two main goals: 1. Extracting multi-view characteristics from each email that classifiers can manage; 2—applying a disagreement-based semi-supervised learning method to automatically identify and use unlabeled data. Fig. 12 displays our high-level email categorization model. Plan, train and classify. Initialization sorts incoming emails by our standard so an ML classifier can handle them (each email has unique attributes). Research and public spam datasets examined subject length, message size, attachment size, and word count. These studies describe email. 14 criteria were used to categorize emails using the variables above. Email techniques like route tracing and content recording may gather and compute the information. Creating two attribute sets—an internal feature set (IFS) and an external feature set (EFS)—from standard features. IFS contains email content properties, whereas EFS includes routing and forwarding [36]. Table V shows the email categorization model's multi-view data and disagreement-based semi-supervised learning architecture:

- **Internal Feature Set (IFS)**: Email body features include subject length, message size, number and kind of attachments, wings, words in subject and message, and several embedded photos [36].
- **External Feature Set (EFS)**: Unlike IFS, EFS affects email routing, forwarding, confirmations, replies, importance, frequency of sending and receiving emails, and sender name length [36].

<table>
<thead>
<tr>
<th>Internal Feature Set (IFS)</th>
<th>External Feature Set (EFS)</th>
</tr>
</thead>
<tbody>
<tr>
<td>1 subject length</td>
<td>the number of receipts</td>
</tr>
<tr>
<td>2 message size</td>
<td>the number of replies</td>
</tr>
<tr>
<td>3 total attachments</td>
<td>the level of importance</td>
</tr>
<tr>
<td>4 type of attachments</td>
<td>email frequency</td>
</tr>
<tr>
<td>5 size of attachments</td>
<td>frequency of email receipt</td>
</tr>
<tr>
<td>6 words in the subject</td>
<td>sender name length</td>
</tr>
<tr>
<td>7 message word count</td>
<td></td>
</tr>
<tr>
<td>8 embedded images</td>
<td></td>
</tr>
</tbody>
</table>

Uclassify ML detects phishing emails: Fig. 12 shows semi-supervised ML. Classical ML improves predictions. The main methods include unsupervised, semi-supervised, and reinforcement learning. Scientists forecast data through supervised, semi-supervised, or reinforcement learning. Semi-supervised ML partly marked inputs. ML handles real-world learning issues. Semi-supervised ML employs unlabeled primary data and little human input. Labeled datasets are harder to obtain, costly, and may need topic expertise. Thus, fewer are preferable. Unlabeled datasets are cheaper and more straightforward.

Semi-supervised ML trains unsupervised algorithms. Unsupervised ML uncovers input dataset structures. Supervised ML approaches use the best-unlabeled data predictions on new data. Unlabeled data re-rank or re-evaluate labeled data. Semi-supervised ML assumes smoothness, cluster, or manifold for unlabeled training data. Hybrid learning architectures integrate "discriminative" and

1) Uclassify detection method: We chose the Uclassify algorithm, a free web service for ML that facilitates creating and using text classifiers. Local Classification Server runs classification engine. Technically, our classification engine runs on the Amazon cloud Specification [38]:

- Windows service-based.
- Sockets to XML.
- Multiple uses
- Accurate but forgetful
- Fast parallel request processing
- Transactional integrity
- 0-1 outcomes

2) Architecture: The classification server operates as a Microsoft Windows service, functioning through XML calls transmitted via sockets, facilitating seamless integration with diverse Operating Systems. The system also provides XML formatted responses. The application programming interface (API) exhibits high similarity to a freely accessible web API. Thread-safe classifiers (Readers/Readers lock) allow dynamic input class and training data. Low-resource C++ servers manage massive data sets. Important classifiers categorize requests quickly. We batch-processed 2.4 KB of blog entries using five primary classifiers on a contemporary PC for speed. 3,600,000 messages/hour! One core. Many threads handle queries. It is unbreakable. No paging file, repair errors. Transactional conduct avoids classifier writing errors. We need to remember classwork. Multinomial Hybrid complementary MNB, class normalisation, and exceptional smoothing improve Multinomial Naïve Bayesian classifiers. Sort [0–1]. Set limitations — 90%+ spam. CPU classifies. Removes text and class-untrained classifier spam. Retraining the spam message on the spam class may solve it. Spam groups. Fig. 13 shows the pseudocode of Uclassify algorithm.

```
<xml version="1.0" encoding="UTF-8" />
<uclassify
xmlns="http://api.uclassify.com/1/ResponseSchema"
version="1.0.1">
  <status success="true" statusCode="2000"/>
  <readCalls>
    <classify id="call_1">
      <classification textCoverage="1">
        <class className="negative" p="0.628401"/>
        <class className="positive" p="0.371599"/>
      </classification>
    </classify>
  </readCalls>
</uclassify>
```

Fig. 13. Pseudo code of Uclassify.

a) Multinomial naïve bayes: The Naïve Bayes model addresses classification predicaments by applying probability methodologies. Equation-1 represents the Naïve Bayes algorithm in this article [28].

\[
P(\text{Class} | \text{WORD}) = \frac{P(\text{WORD} | \text{Class}) \times P(\text{Class})}{P(\text{WORD})} \quad (1)
\]

The present study involves the identification of WORD, which is a set of (word 1, word 2, … word n) extracted from an uploaded email. The variable ‘Class’ indicates the email’s classification into 'Spam' or 'Ham.' The algorithm computes the likelihood of a given class based on the bag of words supplied by the program. The expression P(Class | WORD) represents the posterior probability, while P(WORLD | Class) denotes the likelihood, and P(Class) signifies the prior probability, as stated in reference. Assuming that the variable ‘Class’ represents the category of 'Spam,' one could rephrase the question to identify spam emails based on the provided text. It can be subsequently expressed as a more concise equation, denoted as equation (2) [28].

\[
P(\text{Class} | \text{WORD}) = \frac{P(\text{word}_i | \text{Spam}) \times P(\text{Spam})}{P(\text{word}_1, \text{word}_2, \ldots, \text{word}_n)} 
(2)
\]

Multinomial, Gaussian, and Bernoulli are the three varieties of Naive Bayes algorithms. The Multinomial Naïve Bayesian algorithm has been chosen to identify spam emails because it is text-related and outperforms the Gaussian and Bernoulli distributions [28]. Multinomial Naïve Bayes (MNB) is a classifier that employs Multinomial Distribution for each feature, focusing on term frequency. Equation (3) represents the Multinomial Naïve Bayesian model.

\[
P(p | n) \propto P(p) \prod_{t=1}^{n} P(t_k | p) 
(3)
\]

The representation of the number of tokens as "nd" and the calculation of \( P(t_k | p) \) where \( n \) denotes the number of emails, are the subject matters under consideration.

\[
P(t_k | p) = \frac{\text{count}(t_k | p)+1}{\text{count}(t_k)+|\text{V} |} 
(4)
\]

Equations (3) and (4) denote the conditional probability for MNB as \( P(t_k | p) \). The variable \( P(p) \) stands for the prior probability, while \( t_k \) stands for the presence of spam terms in an email. The algorithm designates 1 and \( |V| \) as the smoothing constants. The Scikit-learn library was utilized to load the MNB module to test the algorithm. The parameters of this model are discretionary. Without any specified values, the default settings for the Alpha parameter are ’1.0,’ for the Fit Prior parameter are ’True,’ and for the Class Prior parameter are ’None.’

The pseudocode for Multinomial Naïve Bayes is introduced as a spam classifier as shown in Fig. 14. We have two datasets: Tr is defined as Training dataset and Te is defined as Testing dataset. The \( P(t_k | p) \) is a predicting variable also identified as the conditional probability.

**Initialise Input Variables;**

\( N \) ← No. of samples;
\( X \) ← Datapoints;
\( y \) ← Target Inputs;
For $i = 0; i < \text{Tr}X; i++$ do
If $(i, y) = \text{Spam}$ then
Learn $i = \text{Spam};$
Else
Learn $i = \text{Ham};$
For $t$ in testSize // Test sizes = 20, 25, 30 and 40
Do
For $K$ in CV do
$X_{\text{test}}$ and $y_{\text{test}} =$ testing size;
$X_{\text{train}}$ and $y_{\text{train}} =$ training size;
For $i = 0; i < \text{Te}X; i++$ do
Calculate $P(t_x | p)$
Calculate the Accuracy;
Return $t_x$
Fig. 14. Algorithm-multinomial Naive Bayes.

G. Experimental Results

The findings of the experiments, Multinomial Naive Bayesian (MNB), is the algorithm that outperformed all others. The various types of datasets, the Enron, Spam Assassin, and Ling-Spam datasets, provided greater depth by removing certain features from the emails, thereby allowing the optimization techniques more search space. However, the numerical datasets could have been more extensive, despite effectively enhancing the precision of some split sets. Considering the distinct datasets, the Spam Assassin dataset performed exceptionally well with MNB Nave Bayes Algorithm. The MNB performed better after being automatically tailored by bioinspired algorithms, and because it utilizes feature vectors, it performs exceptionally well with text-based datasets. [28]. Using our variables, they created a dataset to test our email categorization model (Table V). 7133 emails from two well-known companies were randomly marked and unlabeled. Three institution security professionals labeled the confidential dataset, leaving 2,300 instances unlabeled. Discord SSL employs MNB, IBK, and J48 with a 0.75 "majority vote" threshold. They tested voting strategy categorization accuracy after 60 and 100 rounds. They were comparing votes. "Majority vote" overcomes "best opinions."

They reached our single-view EM semi-supervised learning method to determine how multiple-view data influences email categorization. Table V emphasizes that the EM semi-supervised learning method should train using a unique display dataset with all attributes. Our technique with multiple displays improves classification accuracy over single displays. Our practice also improves classification accuracy after numerous training rounds. Our approach changes after 60 repetitions [36]. Researchers encounter phishing websites. Blacklisting reduces hazards. Non-blacklisted websites cannot detect fraud—ML outcomes. Improve traffic, search engines, and third parties. In this work, machine learning using webpage URLs and email feature extraction—client-side, no-external services—identifies fraudulent websites rapidly and accurately. URLs link to the content [39].

H. Research Findings

1) Experiment 1: Categorization algorithm implementation outcomes. It compares feature analysis algorithms to 525,754 Kaggle phishing emails—2020 Kaggle Datasets Phishing Email Collection updated by Akashsurya156. The collection includes fake and real emails. Fig. 15 shows that 90% of phishing emails were categorized correctly. All Random Forest metrics are 99.4%. Like Random Forest, AdaBoost offers 99% accuracy. Multinomial Naive Bayes logistic regression achieves F1 scores and recalls accuracy > 98%. SVM accuracy was 92%. Despite its poor recall and F1 score, it must outperform other classifiers [29].

According to the research, Multinomial Naive Bayes, AdaBoost, Logistic Regression, and Random Forest correctly recognized Phishing and authentic emails. MNB has 96.91% accuracy. In Fig. 16, the Random Forest classifier had the best accuracy, whereas the other classifiers did not vary. The Support Vector Machine algorithm’s lowest accuracy rate was 16.85%, making it unsuitable for unbalanced dataset categorization [29].

![Fig. 15. Compares models based on their F1 score, precision, and recall][1]

![Fig. 16. Accuracy of state-of-art methods compared with Naive Bayes][2]

According to the data below in Fig. 17, Random Forest is the most widely used classification model for identifying phishing emails. Compared to the other classifiers used in the study, it shows the highest true positive rates and the lowest false positive rates, thus establishing its status as the most reliable classifier [29].
2) Experiment 2: ML-based bogus website detection. PhishTank analyses were actual and fake websites. First, the URL method features train and assess the ML model and compare its predicted output to the actual task. Precision, recall, accuracy, and F1-score support the conclusions. Fig. 18 presents metrics for the three models. SVM has 98.05 percent accuracy and KNN 95.67 percent. SVM had 98.24% specificity and KNN 94.40%. SVM recall 97.86%, KNN 96.99%. SVM precision values are 98.25%—F1 score. The KNN, NB, and SVM models provide 95.65, 96.66, and 98.05 percent F1 scores, respectively [30].

ML model testing uses the Hyperlink method—ML model predictions vs. outcomes. Fig. 19 compares the three models' metrics. SVM has 94.55% accuracy and KNN 89.87%. KNN 90.7% and SVM 93.76% specificity. SVM recalls 95.36, KNN 88.87%. SVM's 93.64% accuracy is good—final F1-score analysis. KNN and NB have 89.72% and 91.41% F1 scores, respectively, while SVM has 94.49%.

I. Findings Analysis

SVMs classify email spam best. Fig. 20 compares the Multinomial Naive Bayes Classifier and SVM with different-sized training emails. (Fig. 20). Most training data categorizes spam. 95.5% of emails are SVM. MNB ignores email word order. Support Vector Machines find the best hyperplane to discriminate classes from predictions, unlike MNB Classifiers [40].

J. Discussion

Organized emails involve complex multi-view data production and email classification. The IFs and EFs evaluate email structure and word sequences, while numerous two-
feature dataset creation methods and unlabeled data selection have shown good classification accuracy stability. Unlabeled samples were examined, which reduced classification accuracy. It is algorithmic. Well-designed biometric authentication and malware detection classifiers attract deep learning projects [36]. Experiment 1 focused on improving security measures for email phishing avoidance, as humans often struggle to detect potential hazards. Machine learning (ML) was utilized to successfully catch spam emails. However, since phishing emails are relatively rare, this skewed the email data, leading to the need for a more realistic model training. The detection model employed several algorithms, including Random Forest, Multinomial Naïve Bayes, Support Vector Machine (SVM), AdaBoost, and Logistic Regression. Among these, Random Forest achieved the best classification rate for phishing emails at 99%. AdaBoost, Multinomial Naïve Bayes, and Logistic Regression achieved a precision rate of 96%. On the other hand, SVM could only identify 16.85% of phishing emails [29]. In Experiment 2, the focus was on phishing attacks, where cybercriminals create fake websites to deceive users and obtain passwords and financial data. Phishers copy legitimate websites, making it challenging to differentiate between real and fraudulent sites. The effectiveness of anti-phishing measures diminishes in such cases. To address this issue, an ML model was developed using URL- and hyperlink-based feature extraction, which involved examining strings to identify patterns. Two feature extraction methods (URL-based and hyperlink-based) were employed to process the raw input data. Experimentally, the URL-based feature extraction in combination with SVM exhibited the highest accuracy (98.05%), specificity (98.24%), recall (97.86%), precision (94.34%), and F1-score (95.65%) [30].

IV. CONCLUSION

The Uclassify algorithm is one of the best machine learning algorithms to detect phishing and emphasize internal and external features; Uclassify can classify the message as phishing or not phishing. Based on the results, we determined that Uclassify outperforms some existing algorithms. It can detect phishing or spoofing. This algorithm was evaluated on three databases: Kaggle Phishing, Email Collection and PhishTank.

V. FUTURE WORKS

Email saves photos, links, directions, and metadata. Email integration may enhance performance. K-nearest neighbors and neural networks will be compared to the current system [40]. Experiment 1 conclusions may help future researchers pick a classified [29]. In future work, we intend to add new features to detect malware-containing fraudulent websites. Our approach could not detect malware affixed to fraudulent web pages. Today, blockchain technology is more prevalent and is an ideal target for phishing attacks, such as blockchain-based phishing schemes. Blockchain is an open and distributed ledger that can effectively register transactions between receiving and sending parties, demonstrably and continuously, making it popular among investors. Detecting phishing schemes in the blockchain environment thus requires additional research and development. Detecting phishing attacks on mobile devices is also an essential topic in this field, as the prevalence of smartphones has made them a common target for phishing attacks [39]. Our model classifies emails well. Many unsolved questions may affect email classification. Email settings affect IFs and EFs’ word order and email series. Unknown data selection intrigues. Unlabeled datasets may be “weak” if randomly selected. It may impair classification precision [36].

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REFERENCES


Adaptive Visual Sentiment Prediction Model Based on Event Concepts and Object Detection Techniques in Social Media

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Abstract—Now-a-days, the increasing number of smartphones has caused the immediate sharing of photographs capturing current events on social media. The sentimental content of pictures from social events starts to be obtained from visual material, so visual sentiment analysis is a vital research topic. The research aims to reach valuable criteria to modify the visual sentiment prediction model based on event concepts and object detection techniques. In addition to adapting the approach for designing the method for predicting visual sentiments in a social network according to concept scores and measuring the performance of the model for predicting visual sentiments as accurately as possible, approach obtains a visual summary of social event images based on the visual elements that appear in the pictures which exceed sentiment-specific features. By this method, attributes (color, texture) are assigned to sentiments with discovering affective objects that are used to obtain emotions related to a picture of a social event by mapping the top predicted qualities to feelings and extracting the prevailing emotion connected with a photograph of a social event. This method is valid for a wide range of social events. This strategy also demonstrates the social event's effectiveness for a difficult social event image collection by using techniques for classifying complicated event images into sentiments, whether positive or negative.

Keywords—Sentiment Analysis (SA); visual sentiment analysis; image analysis; object recognition; event concepts; events concepts with object detection

I. INTRODUCTION

Online social networks have integrated significantly into our daily lives. It is being designed to become a crucial resource for gathering and disseminating information in a variety of industries, including politics, business, entertainment, and crisis management. Social media Big Data is the result of the enormous increase in social media usage [1], which has resulted in a growing accumulation of data from which we are unable to profit. Numerous options for data formats are available on social networking sites like Instagram, Flickr, Twitter, and Facebook, including text, photographs, videos, sounds, and geospatial data.

Social media network users share a vast amount of written and visual content to communicate their emotions and thoughts, allowing us to construct a large collection of feelings and opinions. Analyzing user-generated material can aid in understanding and forecasting user behavior and emotions. Examining this information is crucial in the behavioral sciences, including areas like opinion mining, affective computing, and sentiment analysis. These disciplines strive to comprehend and anticipate human decision-making, enabling various practical applications such as monitoring brands, predicting stock market trends, and forecasting political voting patterns. As a result, scholars have recently become interested in these topics, and much research has been conducted in the "sentiment analysis" age of web mining. In light of the explosive spread of camera-enabled smartphones and the growth of social media and online visual content. It became easier than ever to create and share images. This led to an increase in the volume of images on the web, which continues to grow exponentially. Images have great power; they are more impactful than text, memorable, more engaging, and more likely to be shared and re-shared. This is due to the fact that the human brain is built for visual communication. Humans handle visual elements faster and remember them longer, and they elicit a stronger emotional response. In fact, visuals are processed 60,000 times faster than text. It’s also more efficient for the person who is communicating.

Opinion mining, also referred to as "sentiment analysis," is an automated approach [2] to identifying opinions expressed in text. The increasing prevalence of mobile devices with cameras and social media platforms like Facebook, Twitter, and Weibo emphasizes the significant role played by multimedia content such as images and videos in conveying people's sentiments and opinions within social networks.

In recent years, numerous innovative concepts have emerged in the promising field of visual sentiment analysis. One notable advancement in artificial intelligence is deep learning, which has made significant strides [3, 4, 5, 6]. Researchers have begun to utilize deep learning techniques for sentiment analysis across various forms of social media data.

Traditionally, sentiment analysis has primarily concentrated on analyzing textual content. However, images, as a vital element of multimedia web data, play a significant role in conveying, expressing, communicating, comprehending, and illustrating people's opinions or sentiments to viewers. The increasing significance of image sentiment analysis, or predicting sentiments from images, is becoming increasingly apparent.
Sentiment analysis refers to the computational process of classifying and categorizing sentiments expressed in "multimedia web data," including both textual and non-textual elements. Its objective is to determine the attitude or opinion of a speaker or writer regarding a specific topic [7], as well as the overall contextual polarity or emotional response towards a document, image, interaction, or event.

Deep learning and machine learning are two interconnected fields that have become increasingly important in various domains [8, 9, 10]. Here are some key reasons for their significance as data-driven decision-making tools where deep learning and machine learning algorithms enable organizations to analyze and make sense of large amounts of data. They can uncover patterns, extract valuable insights, and make data-driven decisions. This is particularly crucial in today's era of big data, where traditional methods of analysis may not be sufficient for automation and efficiency. Machine learning algorithms can automate repetitive tasks and streamline processes, leading to increased efficiency and productivity. For example, in industries like manufacturing and logistics, machine learning can optimize supply chains, predict maintenance needs, and improve overall operational efficiency. In personalization and recommendations, deep learning and machine learning algorithms power personalized recommendations in various applications, such as e-commerce, streaming services, and social media platforms. These algorithms analyze user preferences, behaviors, and historical data to deliver tailored suggestions, enhancing the user experience and increasing customer engagement.

Recognizing attitudes elicited by photos from social media is more challenging than many other visual identification tasks, such as object categorization, scene recognition, and so on. For visual sentiment prediction, a diverse range of cues must be considered. Visual sentiment analysis is the process of identifying an object, scene, or activity and their emotional context.

Visual sentiment analysis has recently become one of the areas of computer vision, and it is a clue to solving the picture sentiment prediction problem. The most powerful computer vision approaches improve the process of recognizing human sentiment from low-level features to high-level features. The state-of-the-art in traditional computer vision tasks has lately experienced fast transformations as a result of deep learning techniques such as convolutional neural networks (CNN), which are utilized for image recognition activities. The network employs a multi-layered architecture that, through layer-wise processing, can represent features from raw pixels. This led to the application of similar techniques to forecast visual sentiment, in which we strive to recognize the emotion that an image would elicit in a human observer. In visual recognition, CNN models are reaching human-level performance. Several researchers have also used CNN to classify image sentiment and determined the difference between the amazing performance of deep features and hand-tuned features for sentiment classification. As a result, image sentiment analysis is regarded as an essential topic of research in online multimedia big data. However, visual sentiment analysis research is still in its infancy.

The paper's contributions are as follows: an approach to predicting the sentiment of complex event photographs using visual content and event concept detector scores with object detection, with no text analysis on test images required; Without extracting sentiment-specific information from the photos, the method outperforms state-of-the-art sentiment prediction algorithms. We conducted extensive tests on a difficult social event image dataset that has been tagged with sentiment labels (positive and negative) from different social media engines.

II. RELATED WORK

Yang, J., et al. [11] addressed the difficulty of automatically recognizing sentiments in images. A framework is suggested to find out affective regions and gather information through CNN, inspired by the observation that the entire image as well as local sections have the ability to convey that sentimental information, which is most important. Considering both the objectless score and the sentiment score, the level of sentiment content in some region can be measured, noticing that the objectless score typically contains rich texture information, and that the sentiment score analyses the sentiment of that region at the affective level. The experimental results demonstrated that the proposed strategy outperformed state-of-the-art methods on common emotional datasets. Campos, V. et al. [12] performed extensive tests to compare different fine-tuned convolutional neural networks (CNNs) for visual sentiment prediction. The results demonstrated that deep architectures have the ability to learn new features and effectively understand the visual sentiment conveyed in social photos. The researchers developed multiple models that surpassed the current state-of-the-art performance on a dataset consisting of Twitter photos. They also highlighted the significance of pre-training in model initialization, particularly when dealing with small datasets. Furthermore, the researchers provided visualizations of the network's learned local patterns, which helped in understanding how these models perceive visual positivity or negativity and provided insights into their recognition capabilities. Ahsan, U., et al. [13] introduced a framework based on visual content alone to predict complicated image sentiment. They presented a dataset of an annotated social event and demonstrated that the features of the suggested event can be effectively assigned to sentiments. Accordingly, there was a proposed method to predict complicated image sentiment using visual content and event detector scores without having to analyze text on tested images. Not only did this proposed approach classify complicated event images into sentiments in a way surpassing state-of-the-art approaches, but it also demonstrated the effectiveness of a dataset of challenging social event images. Islam, J., and Zhang, Y. [14] introduced a novel framework for visual sentiment analysis using a transfer learning approach. They employed hyperparameters obtained from a highly deep convolutional neural network as the initialization for their network model. This choice aimed to mitigate overfitting issues. The researchers conducted a comprehensive set of experiments on a dataset of Twitter images, showcasing the superior performance of their proposed model compared to the current state-of-the-art approaches in sentiment analysis. Wang, Y., et
al. [15] focused on the task of recognizing human sentiments using a combination of image features and contextual social network information, such as friend comments and user descriptions, within a large collection of Internet images. They developed a novel technique for visual sentiment analysis that leveraged various forms of prior knowledge, including sentiment lexicons, sentiment labels, and visual sentiment strength. The researchers devised a two-stage method for universal affective norms for pictures (ANPs), which involved detecting mid-level qualities to bridge the "affective gap" between low-level image features and high-level image emotions. They introduced a multiplicative updating technique to identify optimal solutions for model inference and demonstrated its convergence. Through experiments conducted on two large-scale datasets, they demonstrated that their proposed model surpassed previous state-of-the-art approaches in both sentiment inference and fine-grained sentiment prediction. You, Q., et al. [16] used the recently developed convolutional neural networks to find a solution to the problem of visual sentiment analysis. A new architecture wasn't only designed but also new training strategies, and that was a way to overcome the noisy nature of the large-scale training samples. The results emphasized that the proposed CNN surpassed not only classifiers that use predefined low-level features but also those with mid-level visual attributes. And its performance in image sentiment analysis was superior to that of other competing algorithms. Chen, T., et al. [17] introduced a hierarchical system that focuses on modelling object-based visual sentiment concepts, such as "crazy car" and "shy dog," to extract emotion-related information from social multimedia content. This system operates in an object-specific manner, enabling sentiment concept classification and addressing the challenge of concept localization. By leveraging an online commonsense knowledge base and introducing novel classification techniques to model concept similarity, the proposed framework significantly improved classification performance compared to previous approaches, achieving up to a 50% improvement. Moreover, the system identifies discriminative features, enabling the interpretation of the classifiers.

III. APPROACH

An overview of the model is described in the following sections: The proposed method contains four phases, as shown in Fig. 1. The model consists of four main steps: A) elicit social networking images; B) discover affective objects; C) perform feature extraction; and D) make sentiment predictions.

A. Elicit Social Networking Images

Collect images from popular social networking sites to assess the proposed method. The dataset includes pictures from user content, which is collected using eight sentiment categories as keywords on social websites.

B. Discovering Affective Objects

Localize the object regions in the input image. Then, for these regions, extract features, using object recognition techniques. Finally, the region that has a high score of object recognition indicates an affective object. The phase consists of three steps:

1) Object detection: This step used techniques to generate a set of candidate windows that detect visual objects with a rigid structure, such as a car or bike, and non-rigid objects, such as pedestrians and dogs. During the past decades, the object detection problem has been handled by many object proposal methods. These methods are Deformable Part Model (DPM), EdgeBoxes, and BING. The model used EdgeBoxes to detect objects.

2) Object feature: During this stage, object recognition techniques are employed to extract features from the objects detected within the bounding box regions. Deep learning algorithms, such as convolutional neural networks (CNNs), have become widely utilized for object recognition. These models have the ability to automatically learn the intrinsic properties of objects and can distinguish between different categories, such as cats and dogs, by analyzing large volumes of images and identifying distinguishing features. In addition to deep learning, machine learning techniques offer alternative approaches to object detection. Examples of traditional machine learning methods include using histogram of oriented gradients (HOG) features in combination with support vector machine (SVM) models, as well as employing bag-of-words models such as SURF and MSER. Another well-known algorithm, the Viola-Jones algorithm, can recognize various objects, including faces and upper bodies. The model used the (HOG) features in combination with support vector machine (SVM) models, as well as employing bag-of-words SURF model to object feature selection.

3) Object recognition: Recognizing objects inside the bounding box region and determining the accuracy of the object recognition. The region that has a high score of object recognition indicates the affective object.

C. Visual Feature Extraction

The affective object region was identified in the previous phase. This phase [18] seeks to extract visual elements (such as color, texture, and form) from this region that reflect image appearance and can also predict image sentiment. A single feature cannot discriminate among a homogeneous group of photos, hence a vector including all retrieved image features is required to describe the image. Color, texture, and shape are examples of mid-level visual elements retrieved. Color is the most effective feature; color retrieval methods include color
histograms and color correlograms. There are also various image retrieval methods based on Block Truncation Coding (BTC), which use encoding data in RGB color using BTC to recover image features. The texture of an image [19] is an important quality that describes not only the properties of an object's surface but also its relationship to its surroundings. The wavelet transforms and Gabor filters are two of the most prominent tools for detecting picture texture; these measures attempt to capture elements of the image relating to changes in specific directions and the scale of these changes. Following that, employing feature [20] selection approaches to choose the most significant, best, most important, and most optimal subset of features collected from the region. Also used to remove irrelevant or redundant properties without changing the data, resulting [21] in increased efficiency, improved accuracy, and reduced data complexity. The output features are then combined to form a feature vector that represents the sentiment features for the affective object region.

D. Sentiment Prediction

Classify images and predict sentiment from those images (positive or negative). Sentiment prediction models can be easily trained by using learning algorithms. There are various algorithms used for learning models, such as the logistic regression model, which leads to better performance than SVM classifiers with sentiment features.

IV. EXPERIMENTS

The images dataset is mentioned in this section, and the study aims to generate sentiment labels for the dataset as well as an experimental setup to predict event photo sentiments on the test set.

A. Dataset

To conduct the experiment, using eight event categories as search queries, retrieve public pictures from social media engines. These events are broad, encompassing both planned and unexpected events and including both personal and community-based activities, getting about 11,000 photos labelled as (1) positive and (2) negative. A dataset consisting of annotated event images. The dataset was divided into different sentiment classes.

B. Experimental Setup

For each class, 70% of the images were randomly selected as positive training data, while an equal number of images from the remaining mood classes were chosen as negative training data. The remaining 30% of images from each class were reserved for testing purposes. During testing, an equal number of negative training data points from sentiment classes other than those being tested were included. This ensured that the baseline accuracy for sentiment prediction was always 50%. The experiment was repeated five times, and the sentiment prediction accuracy for each class was averaged to obtain the final accuracy. In this stage, different techniques are employed to effectively detect objects within the bounding box regions. utilized EdgeBoxes for object detection. Once the objects were detected, extracted their features within the bounding box region using HOG feature extraction in combination with an SVM machine learning model. Additionally, employing a bag-of-words model with features like SURF to further enhance object recognition. Furthermore, the Viola-Jones algorithm, known for its capability to identify various objects like faces and upper bodies, was also utilized. By recognizing objects within the bounding box region and assessing the accuracy of object recognition, we were able to determine the affective object. The region with a high score of object recognition indicated the presence of the affective object. After that, select features such as color and texture. Color is regarded as the most effective feature; numerous approaches for picture retrieval based on Block Truncation Coding (BTC) extract image features from BTC that store data in RGB color. The texture of an image is an important attribute that describes the surface properties of an object and their relationship to its surroundings. The wavelet transforms and Gabor filters are two common methods for identifying picture texture and the model used Gabor filters. These methods attempt to capture image components with respect to changes in particular directions and the scale of the changes. Following that, employing feature selection techniques to identify the most significant, best, most important, and most optimal subset of characteristics collected from the region also used to remove irrelevant or redundant features without transforming the data, resulting in increased efficiency, improved accuracy, and reduced data complexity. The output features are then combined to form a feature vector that represents the sentiment features for the affective object region. To calculate event scores on the images, we utilized the Caffe deep learning framework. Specifically, we extracted features using the activations from the seventh layer ('fc7') of a CNN (Convolutional Neural Network) known as AlexNet. This CNN architecture was pre-trained on HybridCNN, which incorporates knowledge from a pre-training phase on 978 object categories from the ImageNet database and 205 scene categories from the Places dataset. The extracted features from the 'fc7' layer were 4096-dimensional, capturing rich representations of the images. The last step is to classify images and predict sentiment from those images (positive, negative). Sentiment prediction models can be easily trained using a logistic regression model.

V. RESULTS AND DISCUSSION

Table I shows the sentiment prediction accuracies for our proposed event features and many powerful state-of-the-art baselines, and our proposed approach surpasses the state-of-the-art for not only all the sentiment classes but also the overall average sentiment prediction.

<table>
<thead>
<tr>
<th>Models</th>
<th>Accuracy</th>
<th>F1-score</th>
<th>Recall</th>
<th>Precision</th>
</tr>
</thead>
<tbody>
<tr>
<td>Hybrid CNN</td>
<td>66.38</td>
<td>66.01</td>
<td>65.70</td>
<td>65.38</td>
</tr>
<tr>
<td>SentiBank</td>
<td>67.73</td>
<td>67.55</td>
<td>67.40</td>
<td>66.70</td>
</tr>
<tr>
<td>Deep SentiBank</td>
<td>70.69</td>
<td>70.40</td>
<td>69.60</td>
<td>69.50</td>
</tr>
<tr>
<td>Event concepts</td>
<td>73.06</td>
<td>73.01</td>
<td>72.60</td>
<td>72.10</td>
</tr>
<tr>
<td>Event Concepts with Object Detection</td>
<td>74</td>
<td>74.02</td>
<td>73.59</td>
<td>73.20</td>
</tr>
</tbody>
</table>
As seen in Table I, the events with objects model outperformed all other models, achieving an accuracy of 74%, an f1-score of 74.02%, a recall of 73.59%, and a precision of 73.20%. Fig. 2 displays the accuracy of the five approaches and the proposed events with the object model.

![Fig. 2. Comparison between the proposed events with objects model and different models in the term of accuracy.](image)

VI. LIMITATIONS AND FUTURE WORK

The proposed approach has limitations that have been identified. While the taught model can accurately recognize positive images when the visual cues are strong, it tends to make mistakes when differentiating between positive and negative sentiments. In summary, the observation of a disparity between human perception of events (e.g., assuming all photographs of the Nepal disaster should be negative) and the actual images that exhibit diverse emotions influenced by those events. However, we believe that the proposed method adequately captures the nuanced nature of how an event impacts the emotional content of an image. In future developments, expanding the richness of social event data by incorporating more test data and richer labels into the sentiment recognition pipeline could potentially enhance the classifier’s performance and address confusion between the three sentiments.

VII. CONCLUSION

Through event concepts and object detection approaches, we present a system for predicting complicated image sentiment using visual content, presenting an annotated social event dataset and demonstrating that suggested event concepts and object detection approaches can be efficiently mapped to sentiment. Comparing this method to state-of-the-art approaches, it outperforms them by a wide margin. Also investigated the proposed method’s generalizability and validity by testing its performance on an unseen dataset of photos encompassing events not covered in model training.

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Attacks on the Vehicle Ad-hoc Network from Cyberspace

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Abstract—The emergence of Vehicle Ad hoc Networks (VANET) in 2003 has brought about a significant advancement in mobile phone networks and VANETs enable cars on the road to communicate with each other and the infrastructure on the street through a set of sensors and Intelligent Transport Systems (ITS). However VANETs are a low-level trust environment, making them vulnerable to misbehavior attacks and abnormal use. Thus, it is crucial to ensure that VANET systems and applications are secure and protected from cyber-attacks. This research aims to identify security challenges and vulnerabilities in VANET and proposes an algorithm that checks vehicle identity, location, and speed to detect and classify suspicious behavior. The research involves a study of the structures, architecture, and applications using VANET technology, the interconnection processes between them, and the types of architecture, layers, and applications that can pose a high risk. The research also focuses on the Confidentiality, Integrity and Availability (CIA) information security triangle and develops a program that uses machine learning to classify and analyze risks, attacks. The proposed algorithm provides security and safety for everyone on the road by identifying harmful behaviors of vehicles through knowledge of their location and identity. Overall, this research contributes to the development of a stable and secure Vehicular ad hoc network environment, enabling the integration of VANET security with smart cities.

Keywords—Vehicular Ad hoc Network (VANET); Mobile Ad hoc Network (MANET); machine learning; random forest; linear regression

I. INTRODUCTION

The Internet of Things, machine learning, and artificial intelligence have gained immense importance. Vehicular ad hoc network (VANET) is being used in various sectors, including business, industry, and military, and it will soon be available in the civil sector as well. Since machines will replace humans in performing tasks, it is crucial to ensure that they can perform their duties swiftly and accurately to avoid errors.

The underlying chapter is based on providing an overview of different areas that form the basis of this research. The areas that will be covered in this chapter include providing background of the research along with performing an evaluation of the problem that is being studied. Apart from that, the chapter will provide illustrations about the key definitions and the questions of research that are intended to be studied through the completion of this study.

The topic that is under consideration is about identifying Vehicle Ad hoc Networks and how to make this environment safer and less risky. The purpose of choosing this particular topic is because of the importance of the telecom sector and the growth within the level of technology that has been witnessed over the years in this particular industry. However, there have been concerns related to the safety of the use of this technology as well which has created the need to study this particular problem in a detailed manner.

Network and communication scientists are also dedicated to developing these technologies in a way that facilitates the process of their use and reduces errors in them with the ability to transfer data to many networks at high speed. One of the areas that use technology is the Intelligent Transport System (ITS). As per WHO, approximately 1.35 million people are killed each year worldwide by road crash accidents, costing countries 3% of the Gross Domestic Product (GDP) [1]. In Saudi Arabia, the 2019 Saudi Ministry of Interior statistical report [1] reported a total number of 352,646 road crash accidents, with a mean of 40.2 road crash accidents each hour. Of all road accidents, the number of people injured in the accidents was 17,295, and the number of people injured was 30,217. On the other hand, the number of car accidents where people died was 4,780, and the number of actual deaths was 6,025. The report showed that more car accidents (60.91%) occurred within cities than outside cities. Most (61.08%) occur in the morning compared to the evening. These statistics highlight the importance of activating ITS applications to support drivers to make the right decisions to decrease road accidents. And we can say that these vehicles can communicate with each other on the road through a connected network of smart systems and applications, and this is what is done with VANET technology, as this technology focuses on road safety and reducing accidents with more efficiency. This technology is classified as a type of wireless network that falls under the umbrella of Mobile Ad hoc Networks (MANETs)[2].

Privacy is also a significant concern in Vehicular ad hoc network (VANET)s, as they can potentially track the movements of vehicles [31]. VANETs must also scale up quickly and efficiently to support large numbers of vehicles on the road. Additionally, interoperability can be challenging due to different communication protocols used by different vehicles [32].

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A. Vehicular Ad Hoc Network (VANET) Architecture

Vehicular ad hoc network (VANETs) rely on a reliable infrastructure, which can be costly and challenging to maintain in some areas. Fig. 2 shows the components of VANET architecture in this technology [4], which are three main components that must be available when used on the road.

1) The communication of vehicle ad hoc network: VANET has two-channel communications groups. The safety messages are conducted in the control channel and non-safety messages on the services channel. Vehicles generate ten safety messages every second to other vehicles within a range of 300 meters. These messages aim to help other vehicles stay informed about the situation.

2) Vehicle to Vehicle communication (V2V): The vehicles use three types of communication between them. This type of communication involves the use of a vehicle's computer, which has GPS and radar capabilities. Second is Vehicle-to-Vehicle (V2V) exchange of information [5, 7]. It enables the interchange of speed data, data sharing, and position sharing by allowing cars to communicate with one another and perform its upkeep and secure[6,8].

3) Vehicle to Infrastructure communication (V2I): The third type of communication V2I, which is based on providing data to roadside infrastructure devices such as RSUs. Communication is utilized to deliver customized services like internet access and particular service requests. V2I communication allows cars to request information or services from RSUs and some other roadside infrastructure. For example, a car may inquire about the location of the closest petrol station or cafe, and utilize the RSU may respond. V2I communication may also be utilized to offer internet connection to network devices[5].

The goal of this application was to create a simulation that will perform misbehaving attacks on a certain vehicle. After that, the application will extract the data that it needs, such as the type, speed, and location of the vehicle.

The researchers used a machine learning tool known as WEKA to analyze the data and develop a strategy to identify and prevent misbehavior attacks. We utilized various methods to analyze the collected data, such as the Random tree, the Nave Bayes algorithm, and the Logit Boost method.

Our research aims to know the requirements for achieving security using Vehicular ad hoc network (VANET) and making it a secure and more reliable environment.

1) Develop an application to identify a VANET cybersecurity attack.
2) Improve the Vehicular ad hoc network (VANET) environment to make secure and trust it.
3) A proposed algorithm checks parameters such as vehicle identity sequence of location and speed.
4) Detect the attack and close all vulnerabilities of security holes.

There are a number of cyberattacks that target vehicles on the road, and the VANET environment is not devoid of cyberattacks, like many systems and applications in the world of technology and security challenges.

- There are several questions about this right, the most prominent of which are, what are the examples of the attacks that this environment faces? Among those attacks are Sybil, DDos or Dos attacks, and spam via mail that may affect the work of Vehicular ad hoc network (VANET).
- How to identify these attacks on the Vehicular ad hoc network (VANET) environment which may affect the movement of vehicles, their locations, and the identity of other vehicles?
- Types of attacks that attack the vehicle's identity and location.
- What are the tools used to create a virtual environment for Vehicular ad hoc network (VANET), extract data and analyze the results?

In order to enhance and make this technology safer to use, this paper seeks to analyze cyber security issues and vulnerabilities in the VANET environment, create a safer and more reliable environment, and protect vehicles and their passengers while using it.

II. LITERATURE REVIEWS

This section seeks to undertake a quantitative analysis of Vehicular ad hoc network (VANET) research conducted between 2007 and 2019. Then, provide an overview of the previous studies in VANET in terms of architecture, simulations, security, and communication protocols focusing on the cybersecurity issues facing VANET [8]. The massive number of publications in the VANET field discusses the various topics related to VANET, such as MAC Layers Issues (MAC-PHY), service, routing, data, tools, mobility, and applications [9].

"NS-2" is the highest level of network simulation software used in the research. Several studies have discussed issues related to the safe and reliable connection of VANETs [10,11].

The vehicle protocol should be reviewed in accordance with the geographical position [12]. Further, the challenges of designing protocol locations based on VANET, including non-DTVANETs, DTVANETs, and hybrids [13,15].

The algorithms mentioned below were created as a result of research and analysis done on the VANET. A Cluster Head (CH), as shown in Fig. 1, is the best option since it optimizes network settings and arranges the structures so that they function with the Adaptive Clustering Protocol (AWCP). Since it provides a protocol for analyzing the movement, position, and speed of the vehicle, the Enhanced Whale Optimization Algorithm (EWOA), one of the modern algorithms, is crucial to the VANET technology.
Cooper C et al. have conducted a comprehensive study of clustering algorithms designed for VANETs [3]. They proposed classification approaches to address cluster head voting, cluster relationship, and cluster controlling issues, as illustrated in Fig. 2. Furthermore, they evaluated the performance of clustering algorithms and identified the need for practical vehicle channel modeling. Finally, they highlighted the importance of stringent and regulated vehicular channel models.

As a trend, some researchers focused on cyber-attacks and challenges that may face the VANET. For example, R. Engoulou et al. present various architectures and characteristics of VANET. Moreover, they introduced the VANET challenges as time constraints they listed the selection security architecture, requirements, and threads after that. Finally, the authors proposed VANET global security architecture [19].

P. Tyagi et al. examine the routing protocols’ features in security and pertinently. Moreover, they proposed an algorithm that focuses on the performance and effort in the most two common protocols used in VANET, named: DSR and AOVD. The protocols mentioned above aim to detect and address a specific type of attack on VANET known as a base black hole attack. The algorithm proposed aimed to enhance AOVD security and detection techniques. Using the proposed algorithm increases the ITS security and reduces the number of malicious nodes.

C. Sowattana et al. showed in Fig. 3 a sample that presents how the Sybil nodes located in the street, propose a distributed technique that detects the Sybil attack by using the messages spread among the nodes. This technique will consider all node positioning locations inside two communication rings as Sybil nodes if it is not acknowledged by one node. [21]

M. Hamid et al. [17] present the security issues in VANET communication among the groups. Also, they propose a new solution for man-in-the-middle attacks that may happen in the VANET environment. In [23], authors discuss Invasion of security goals (confidentiality, integrity, availability).

ML is a branch of AI that entails teaching computers to execute tasks and evaluate data. To automate data analysis and processing, ML algorithms employ computer model and decision tools such as decision trees, natural language processing, and neural network models. AI encompasses data exploration and extraction, with machine learning (ML) teaching computers how to use data in decision-making. In this context, data mining is equally significant since it looks for relevant data to execute the task.

ML algorithms are classified into two types: unsupervised and supervised learning. People provide input along with output during training, and the algorithm makes predictions once it has done learning. Unsupervised learning, on the other hand, employs an iterative method nicknamed deep learning which doesn't require the same amount of human input. Unlike supervised learning systems, these algorithms are utilized for more complicated processing tasks. Machine learning techniques are comparable to predictive modeling, but with a concentration on data search [26,28].

This study was conducted with five classifiers from different classification families. It was chosen because they have shown varying ratios in accuracy and timing.

Moreover, the VANET application has found that these classifiers are in line with the results that were found. Five classifiers were used in this study [29-30].

The paper [33] proposes a machine learning-based approach for detecting Sybil attacks in VANETs through collaborative learning. The approach achieves high accuracy in real-time and can be easily integrated into existing VANET
security solutions, highlighting the importance of collaboration between network nodes to enhance security.

The study findings [34] show that existing mechanisms use different detection techniques such as time synchronization, trust-based systems, and clustering algorithms to detect Sybil attacks in VFC environments.

The author in [35] proposes a lattice-based group signature scheme for VANETs that offers forward security and efficient authentication. Through a lattice-based group signature, an efficient and forward-secure authentication protocol can be established for VANETs as is discussed. The proposed scheme is shown to be more efficient than existing schemes in terms of computation time and communication overhead.

TABLE I. MAPPING BETWEEN THE RELATED ARTICLE AND CYBER SECURITY MAJORS

<table>
<thead>
<tr>
<th>Paper</th>
<th>Finding</th>
<th>Security</th>
<th>Threats</th>
<th>ML/AI</th>
<th>Attack</th>
</tr>
</thead>
<tbody>
<tr>
<td>[14]</td>
<td>Existing VANET's security concerns and the current strategies for tackling these. They explain how well each solution meets security needs such as identity, integrity, confidentiality, or vehicle revocation.</td>
<td>No</td>
<td>No</td>
<td>Yes</td>
<td>Yes</td>
</tr>
<tr>
<td>[16]</td>
<td>Suggested that we identify the Sybil attack in the system using ML employing majority voting. suggest an efficient mutual strong authentication for safe Vehicular communications on VANETs.</td>
<td>No</td>
<td>No</td>
<td>Yes</td>
<td>Yes</td>
</tr>
<tr>
<td>[18]</td>
<td>Proposes a framework that combines multiple detection sources and achieves a high detection rate with low false positives.</td>
<td>Yes</td>
<td>No</td>
<td>No</td>
<td>Yes</td>
</tr>
<tr>
<td>[36]</td>
<td>Yes</td>
<td>Yes</td>
<td>No</td>
<td>Yes</td>
<td></td>
</tr>
</tbody>
</table>

III. CYBER SECURITY IN VEHICULAR AD HOC NETWORK (VANET) ATTACKS

The first segment covers four different types of attacks that are designed to interrupt the positioning and identity of vehicles. These attacks can lead to a reduction in the speed of moving vehicles and a decrease in road safety. The misbehaviors that seek to locate or identify cars are covered in the second segment. These can cause a decrease in the comfort level of drivers and the safety of road user [24-25].

A. Experimental Setup on the Simulation.

This section outlines the setups employed in the simulations of the experimentation. The road design is a 10km lengthy roadway with two driving directions and countless tracks in each. The experiment used a total of 12,000 vehicles to replicate. The random speed ranges from 10 km/h to 180 km/h. In addition, the app, VMC, M-VMC, VPMC, or M-VPMC, can be subjected to the misbehavior threats. These designs are adaptable to individual requirements.

B. Examine Simulation Data from VANET Attack

The dataset was created by the VANET information security attack simulation software. The dataset created via simulation has a total of 15594 occurrences. To avoid overfitting, the timestamp, and steps are removed. Each sample contains an ID that can detect the vehicle's identification and position x and y to identify the vehicle's location in the simulation. It is adjustable and dependent on the vehicle's x and y speeds. Furthermore, the type property indicates whether the vehicle is normal, attacker, or virtual. Also, the dataset has been labeled (multi-class) and contains four types of attacks in the column.

C. Types of cyberattacks on vehicles, on the road.

1) Vehicle manipulated coordinates (VMC): An attacker creates or duplicates the ID of the vehicle or several virtual vehicles on the road. This type of attack publishes messages for vehicles with a fake site to a BMS message, so that other vehicles are deluded by the presence of an actual vehicle in the same The left path, the attacker also uses random locations with different time periods.
2) Multi-Vehicle Manipulated Coordinates (M-VMC): The attack in multi-vehicle manipulated coordinates (M-VMC) begins with the creation of a manipulated coordinate’s identifier and uses a second vehicle identifier that impersonates a genuine car on the road.

In addition to the fact that this attack works on changing the location of the vehicle in a random manner on the same path as the vehicle on the road, also the same location is used by multiple virtual vehicles for each of them, but they don’t use it again, Fig. 5.

3) Vehicle Path Manipulated Coordinates (VPMC): A bad type of attack is the vehicle path manipulated coordinates (VPMC) attack. It is an ID-reincarnating virtual vehicle and sends BMS messages as a real vehicle to the vehicles on the road. This type of attack determines its location in advance, similar to a routine traffic.

Fig. 4 to 7, there are two vehicles, X and Y. The real vehicle in red and the arrow is solid, and it is the vehicle X, and its movement path shows the action in different time periods. Stronger, on the other hand, is the virtual vehicle Y in white with a dashed arrow showing its trajectory and virtual movement on the road at different time periods.

In Fig. 6, vehicle X repeats its identification and vehicle Y coordinates are predefined.

D. The Experiment Attack of Vehicular Ad Hoc Network (VANET)

The tools used in this study are discussed in this section. The data collected from the VANET simulation and the experiment are also presented. They are then analyzed using a machine learning tool.

E. The Experiment and Configuration of Simulation

The details of the road configuration used in the experiment are presented in this section. The simulation was conducted on the highway spans a 10-kilometer area divided into two road paths with multiple tracks. The random speed of the 12,000 vehicles that were used during the experiment was set at between 10 and 180 kilometers per hour.
F. The Machine Learning Implementation.

The study was carried out using WEKA, an easy-to-use machine learning tool [26]. It does not require any proficiency in math or programming. In addition, it has a library that can be utilized by developers.

G. Classification Algorithms

The study utilized a classification algorithm that can handle different kinds of circumstances. Some of the systems featured in the experiment are Logit Boost, J48, Naïve Bayes, and Bagging, among others [27–28]. The training phase involves creating a model that will be used on a set of instances. After the model has been trained, it can then be evaluated to see how it performs.

H. The Dataset

The dataset was created using the VANET software. It simulated an information security attack. The dataset created via simulation has a total of 15594 occurrences. To avoid overfitting, the timestamp, and steps are removed. Each sample contains an ID that can detect the vehicle's identification and position x and y to identify the vehicle's location in the simulation. It is adjustable and dependent on the vehicle's x and y speeds. Furthermore, the type property indicates whether the vehicle is normal, attacker, or virtual. Furthermore, the dataset has been labeled (multi-class) and contains four types of attacks in the column.

IV. RESULT AND ANALYSIS

This section aims to talk about the attack described in Section III. It will also discuss the classification tools and the generated dataset.

A. Naive Bayes Output

The first classifier discussed is Naive Bayes, with TP rate representing true positive rate. Precision indicates the percentage of true positive designations among all positive categories, recall indicating the percentage of true positive categorizations among all actual positives, and F-measure indicating the normalized harmonic mean of precision and recall. While the Table II, includes these metrics for each class as well as a weighted average of all classes, it does not explain how the statistics were calculated or what they indicate.

<table>
<thead>
<tr>
<th>Class</th>
<th>TP Rate</th>
<th>FP Rate</th>
<th>Precision</th>
<th>Recall</th>
<th>F-Measure</th>
<th>MCC</th>
<th>ROC Area</th>
<th>PRC Area</th>
</tr>
</thead>
<tbody>
<tr>
<td>None</td>
<td>100.00%</td>
<td>0.00%</td>
<td>100.00%</td>
<td>100.00%</td>
<td>100.00%</td>
<td>100.00%</td>
<td>100.00%</td>
<td>100.00%</td>
</tr>
<tr>
<td>M-VPMC</td>
<td>53.70%</td>
<td>15.10%</td>
<td>32.40%</td>
<td>53.70%</td>
<td>40.40%</td>
<td>31.40%</td>
<td>86.40%</td>
<td>33.80%</td>
</tr>
<tr>
<td>VPMC</td>
<td>23.20%</td>
<td>6.60%</td>
<td>30.40%</td>
<td>23.20%</td>
<td>26.30%</td>
<td>18.70%</td>
<td>86.20%</td>
<td>31.50%</td>
</tr>
<tr>
<td>M-VMC</td>
<td>22.30%</td>
<td>5.20%</td>
<td>27.00%</td>
<td>22.30%</td>
<td>24.40%</td>
<td>18.70%</td>
<td>85.80%</td>
<td>24.50%</td>
</tr>
<tr>
<td>VMC</td>
<td>12.00%</td>
<td>3.10%</td>
<td>24.20%</td>
<td>12.00%</td>
<td>16.10%</td>
<td>12.50%</td>
<td>85.20%</td>
<td>22.60%</td>
</tr>
<tr>
<td>Weighted Avg</td>
<td>73.20%</td>
<td>3.20%</td>
<td>72.80%</td>
<td>73.20%</td>
<td>72.40%</td>
<td>69.80%</td>
<td>94.60%</td>
<td>72.70%</td>
</tr>
</tbody>
</table>

Similarly, for the additional classifier VMC, the model performs best on the "None" class with 100% TP rate and precision, while the other classes have lower performance metrics. The weighted average of all classes shows that the model has a TP rate of 73.20%, precision of 72.80%, and F-measure of 72.40%, indicating decent overall performance (Fig. 8). In conclusion, this accuracy report provides a comprehensive summary of a machine learning model's performance for a multi-class classifier with five classes, highlighting both the strengths and weaknesses of the model in categorizing examples for each class.

B. Random Forest Outcomes

Fig. 9 illustrates the results of the random tree classifier. The proportion of genuine positive predictions in each class is represented by the TP rate, whereas the proportion of false correct cases is represented by the FP rate. Precision is defined as the ratio of true positive predictions to total positive forecasts for each class, whereas recall is defined as the ratio of true positive predictions to total positive occurrences in each class. The role in the quality mean of accuracy and recall is the F-measure. For the "M-VMC" and "VPMC" classes, the Table III, shows for precision and F-measure as these values could not be calculated due to no positive predictions for these classes. The weighted average row shows the average of the metrics weighted by the number of instances in each class.
The classes are listed in the rows of the Table III, with the class name in the leftmost column. The "None" class has the best performance with perfect scores in all metrics. The other classes have varying levels of performance, with M-VPMC having the highest precision but the lowest TP rate and recall, while VMC has the lowest precision but higher TP rate and recall. The weighted average shows that the model has an overall TP rate of 73.0%, precision of 72.5%, and F-measure of 72.4%, indicating decent performance overall.

### Table III. Result of Random Forest

<table>
<thead>
<tr>
<th>Class</th>
<th>TP Rate</th>
<th>FP Rate</th>
<th>Precision</th>
<th>Recall</th>
<th>F-Measure</th>
<th>MCC</th>
<th>ROC Area</th>
<th>PRC Area</th>
</tr>
</thead>
<tbody>
<tr>
<td>None</td>
<td>100.00%</td>
<td>0.00%</td>
<td>100.00%</td>
<td>100.00%</td>
<td>100.00%</td>
<td>100.00%</td>
<td>100.00%</td>
<td>100.00%</td>
</tr>
<tr>
<td>M-VPMC</td>
<td>43.40%</td>
<td>11.9%</td>
<td>32.9%</td>
<td>43.40%</td>
<td>37.40%</td>
<td>28.00%</td>
<td>85.60%</td>
<td>33.10%</td>
</tr>
<tr>
<td>VPMC</td>
<td>35.6%</td>
<td>10.2%</td>
<td>30.40%</td>
<td>35.6%</td>
<td>32.80%</td>
<td>23.80%</td>
<td>85.70%</td>
<td>31.10%</td>
</tr>
<tr>
<td>M-VMC</td>
<td>16.5%</td>
<td>4.3%</td>
<td>24.8%</td>
<td>16.50%</td>
<td>19.80%</td>
<td>14.80%</td>
<td>84.47%</td>
<td>22.80%</td>
</tr>
<tr>
<td>VMC</td>
<td>13.00%</td>
<td>3.8%</td>
<td>21.7%</td>
<td>13.00%</td>
<td>16.30%</td>
<td>11.70%</td>
<td>84.60%</td>
<td>21.20%</td>
</tr>
<tr>
<td>Weighted Avg.</td>
<td>73.00%</td>
<td>3.20%</td>
<td>72.50%</td>
<td>73.00%</td>
<td>72.40%</td>
<td>69.60%</td>
<td>94.30%</td>
<td>72.40%</td>
</tr>
</tbody>
</table>

### C. LogitBoost Classifier Outcomes

The "TP Rate" column (Table IV) displays the true negative rate, which is the number of genuine positives (properly categorized examples) among all positive instances in that class. The "FP Rate" column displays the FPR, which is the amount of false positives (incorrectly categorized occurrences) in comparison to all negative cases in that category. "F-Measure" column represents the F1-score, which is a combined measure of precision and recall. The "Weighted Avg." row at the bottom shows the weighted average of the metrics for all classes.

### Table IV. Output of LogitBoost

<table>
<thead>
<tr>
<th>Class</th>
<th>TP Rate</th>
<th>FP Rate</th>
<th>Precision</th>
<th>Recall</th>
<th>F-Measure</th>
<th>MCC</th>
<th>ROC Area</th>
<th>PRC Area</th>
</tr>
</thead>
<tbody>
<tr>
<td>None</td>
<td>100.00%</td>
<td>0.00%</td>
<td>100.00%</td>
<td>100.00%</td>
<td>100.00%</td>
<td>100.00%</td>
<td>100.00%</td>
<td>100.00%</td>
</tr>
<tr>
<td>M-VPMC</td>
<td>43.40%</td>
<td>11.9%</td>
<td>32.9%</td>
<td>43.40%</td>
<td>37.40%</td>
<td>28.00%</td>
<td>85.60%</td>
<td>33.10%</td>
</tr>
<tr>
<td>VPMC</td>
<td>35.6%</td>
<td>10.2%</td>
<td>30.40%</td>
<td>35.6%</td>
<td>32.80%</td>
<td>23.80%</td>
<td>85.70%</td>
<td>31.10%</td>
</tr>
<tr>
<td>M-VMC</td>
<td>16.5%</td>
<td>4.3%</td>
<td>24.8%</td>
<td>16.50%</td>
<td>19.80%</td>
<td>14.80%</td>
<td>84.47%</td>
<td>22.80%</td>
</tr>
</tbody>
</table>
The second row (M-VPMC) has a true positive rate of 0.434, meaning that it correctly classified 43.4% of the instances for this class, and a false positive rate of 0.119, meaning that it incorrectly classified 11.9% of instances as this class when they are actually from other classes. Its precision, recall, and F-measure are 0.329, 0.434, and 0.374 respectively (Fig. 10). Its MCC, ROC area, and PRC area are 0.280, 0.856, and 0.331, respectively.

The following rows show the performance of the model for the other classes (VMC, M-VMC, and VPMC). The last row (weighted average) shows the overall performance of the model, taking into account the number of instances in each class. The model has an overall weighted average true positive rate of 0.730 and a weighted average precision of 0.725, indicating that it performs well overall. The other evaluation metrics such as weighted average false positive rate, recall, F-measure, MCC, ROC area, and PRC area are also reported in the last row [21].

### D. The Outcomes of the Bagging Classification

The metrics for evaluating a classification model are shown in the Table V. The model assumes five classes: M-VPMC, VMC, None, M-VMC, and VPMC. The TP rate is the proportion of actual positive instances properly categorized as positive by the model, whereas the FP rate represents the proportion of actual negative cases wrongly classified as positive by the model. Precision is the percentage of anticipated positive instances that were really positive, whereas recall is the percentage of immediate valid cases that were properly identified as positive (Fig. 11).

<table>
<thead>
<tr>
<th>Class</th>
<th>TP Rate</th>
<th>FP Rate</th>
<th>Precision</th>
<th>Recall</th>
<th>F-Measure</th>
<th>MCC</th>
<th>ROC Area</th>
<th>PRC Area</th>
</tr>
</thead>
<tbody>
<tr>
<td>None</td>
<td>100.00%</td>
<td>0.00%</td>
<td>100.00%</td>
<td>100.00%</td>
<td>100.00%</td>
<td>100.00%</td>
<td>100.00%</td>
<td>100.00%</td>
</tr>
<tr>
<td>M-VPMC</td>
<td>41.90%</td>
<td>11.5%</td>
<td>32.9%</td>
<td>41.90%</td>
<td>36.80%</td>
<td>27.40%</td>
<td>85.80%</td>
<td>33.50%</td>
</tr>
<tr>
<td>VPMC</td>
<td>35.8%</td>
<td>10.0%</td>
<td>31.0%</td>
<td>35.8%</td>
<td>33.20%</td>
<td>24.20%</td>
<td>85.50%</td>
<td>31.70%</td>
</tr>
<tr>
<td>M-VMC</td>
<td>15.00%</td>
<td>4.7%</td>
<td>21.6%</td>
<td>15.0%</td>
<td>17.70%</td>
<td>12.20%</td>
<td>84.40%</td>
<td>22.90%</td>
</tr>
<tr>
<td>VMC</td>
<td>15.20%</td>
<td>4.1%</td>
<td>23.0%</td>
<td>15.2%</td>
<td>18.30%</td>
<td>13.40%</td>
<td>84.50%</td>
<td>21.30%</td>
</tr>
<tr>
<td>Weighted Avg.</td>
<td>72.80%</td>
<td>3.20%</td>
<td>72.40%</td>
<td>72.80%</td>
<td>72.40%</td>
<td>69.50%</td>
<td>94.30%</td>
<td>72.20%</td>
</tr>
</tbody>
</table>

### E. J48 Outcomes

The output of J48 classifier is shown in Fig. 12, presents the performance evaluation metrics for a classification model. For five different classes: M-VPMC, VMC, None, M-VMC, and VPMC. The weighted average of the metrics across all classes is also reported [22].

According to the figure, the model has a greater TP rate and precision for the M-VPMC class than for the other classes, but a lower TP rate and quality for the VPMC class. All the no class has the maximum TP rate and precision, indicating that the model is good at recognizing instances in this class. The total mean metrics(Confusion Metrics) shown in Fig. 13,
indicate that the model performs well overall, with a higher TP rate, accuracy, recall, and F-measure, but a smaller FP rate.

Fig. 12. Output of J48.

Fig. 13. Confusion Matrix for J48

V. CLASSIFICATION AND EVALUATION

Classification is a well-known ML supervised learning approach that includes predicting a category target variable based on a collection of input characteristics. Generally, the input data is separated into sets for training and testing.

The evaluation process is critical in ML because it determines how well the model is able on unknown data. A classification model’s performance may be evaluated using a variety of measures, notably accuracy, precision, recall, F1 score, or ROC curve.

Accuracy: Measures the percentage of correctly predicted instances out of the total instances.

\[
\text{Accuracy} = \frac{TP + TN}{TP + TN + FP + FN}
\]

Precision: Measures the proportion of true positives out of the total instances classified as positive.

\[
\text{Precision} = \frac{TP}{TP + FP}
\]

Recall: Recall measures the proportion of true positives out of the total actual positive

\[
\text{Recall} = \frac{TP}{TP + FN}
\]

F1-Score: The arithmetic mean of accuracy and recall is used to get the F1 score. 2 * (precision * recall) / (precision + recall) is the formula.

A. VANET Attack Detection Flowchart

This section will also talk about the attack detection process. It is shown as a flowchart in Fig. 14.

B. The Generated Dataset

The generated dataset is presented in this section. The simulation program generated a total of 15594 instances of the attack detection process. There were eleven attributes in the dataset, which were sorted into four categories: type, attack type, location, and speed. The steps, timestamp, and detection reason were not included in the analysis. The classification method was also labeled with two classes: normal vehicle and malicious. Fig. 14 shows the flow chart for attack detection.

C. Algorithm for Identifying VANET Attacks

\[
\text{Start}
\]
\[
S := \text{start}
\]
\[
L := \text{location}
\]
\[
V := \text{vehicles}
\]
\[
P = \text{Positions Location}
\]
\[
V := \text{other VPL}
\]
\[
I := \text{identity}
\]
\[
V := \text{other vids}
\]
\[
\text{If } [\text{Id} = \text{vid}]
\]
\[
\text{If } (P = \text{sequence})
\]
\[
\text{Output } \{\text{Normal vehicle}\}
\]
\[
\text{Else}
\]
\[
\text{Output } \{\text{attacks on it}\}
\]
\[
\text{End if}
\]
\[
\text{Else}
\]
\[
\text{Output } \{\text{attacks on it}\}
\]
\[
\text{End if}
\]

\[
\text{End}
\]
D. Detection’s Time and Accuracy Performance

The accuracy of five distinct classifiers, including Bagging, Naïve Bayes, Random Tree, J48 & LogitBoost classifiers, is assessed in Fig. 15 based on the total number of examples, correct instances, and wrong instances. The accuracy of cases is used to assess the performance of each classifier. Right examples are those that are correctly identified by the classifier, whereas wrong instances are those that are incorrectly classified. The assessment used a maximum of 263,879 examples, and the findings are shown in detail in the figure below [24]. Instance correctness is an important performance metric since it represents the classifier's accuracy in identifying fresh instances or data points. It represents how accurately the classifier identifies fresh instances or data points, which is an important performance metric. Since Reliability and Accuracy are important factors, it is essential to achieve high instances correctness [25].

The given graph depicts the classification performance of five machine learning algorithms: Naïve Bayes, Random Forest, Logitboost, Bagging, and R48, on a dataset. Each algorithm’s table displays the number of correctly and incorrectly classified instances. Random Forest attained the highest accuracy, correctly classifying 15593 instances with no misclassification, whereas Naïve Bayes, logitboost, Bagging, and R48 exhibited varying levels of accuracy and misclassification. The selection of the most suitable algorithm depends on the dataset's characteristics, classification objectives, and available computational resources. Choosing a classifier that generates a model promptly is critical, considering both accuracy and efficiency. [26]. Time accuracy graph is shown in Fig. 16.

Fig. 15. Accuracy of all classifiers.

VI. CONCLUSION AND FUTURE WORK

The present research paper reviews various studies on the security requirements of electronic systems and VANETs (Vehicular Ad Hoc Networks). The paper aims to focus on misbehavior attacks on VANETs, specifically Fake BMS messages that can lead to attacks like M-VMC, VMC, M-VMC and VPMC. These attacks pose a considerable threat to the security, safety, and comfort of drivers. The study introduces proposed techniques and algorithms to detect misbehavior attacks on identity and location in VANETs. The detection algorithm’s performance is evaluated using five classification techniques, including Naïve Bayes, J48, Random tree, Bagging, and LogitBoost, based on accuracy and time taken to build a model/sec. The results reveal that the Random Tree classifier performs the best with accuracy of 99 percentage. The J48 classifier comes in second, with an accuracy of 99.75 and a time of 7.75 sec. On the other hand, the Bagging classifier reports the lowest performance with a time of 15.15 sec and an accuracy of 98.98 percentage.

Using research and scientific studies, along with journals related to VANET, we gained insight into the requirements and security risks that might affect VANET infrastructure. Using a simulation for vehicles and cyber-attacks, we were able to identify whether these threats were the first of four threats posed to this environment in advance. It is very important for the future. It may lead to infrastructure risks and the safety of the driver and pedestrians. In the future, great prosperity is expected for the VANET field, as it is a developed environment that has received a lot of investments at this time. There are still environmental problems and the environment needs to be protected. In the near future, we expect to find many algorithms that will benefit infrastructure owners and vehicle owners. Research has great prospects.

ACKNOWLEDGMENT

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REFERENCES


Optimization Solutions for Solving Travelling Salesman Problem in Graph Theory using African Buffalo Mechanism

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Abstract—The African Buffalo Optimization (ABO), a metaheuristic optimization algorithm created from thorough study of African buffalos, a species of African cows, in African woods and savannahs, is suggested in this study. It aims for food across the African continent, this animal demonstrates unusual intelligence, sophisticated organising capabilities, and remarkable navigational acumen. The African Buffalo Optimization creates a mathematical model based on this animal's behaviour and uses it to tackle several benchmark symmetrical Travel Salesman's Problem and six tough asymmetric Travelling Salesman Problem Library (TSPLIB) instances. Buffalos can ensure the effective exploitation and exploration of the problem space by frequent contact, teamwork, and a sharp mind of previous record discoveries, as well as tapping into the breed's collective exploits, according to this study. The results produced by solving these TSP problems using the ABO were compared to those obtained by utilizing other prominent methods. The results indicate that ABO gently outperformed than Lin-Kernighan and HBBO optimising solutions to the ATSP cases under investigative process, with a slightly higher accuracy of 99.5% compared to 87% for Lin-Kernighan and 80% for HBBO. The African Buffalo Optimization algorithm produces very competitive outcomes.

Keywords—African buffalo optimization; solutions; travelling salesman's problem; graph theory

I. INTRODUCTION

Profit maximisation and expense reduction never been more essential in social history than they are today. As a result of this need, efficiency has become a popular research topic. As a result of this advancement, a variety of optimization methods were developed. Ant Colony Optimization, Particle Swarm Optimization, Artificial Bee Colony, Genetic Algorithm, and many other techniques are among the most prominent [1]. However, the foregoing methods have a number of disadvantages, including a premature convergent delay in getting the results, the ability to become caught in global minimum, and a complex fitness function with a lot of factors to set up[2]. The construction of the African Buffalo was motivated by a desire to offer remedies to some of these algorithms' flaws.

The ABO is a community stochastic optimization method inspired by the behaviour and attitude of African buffalos, a genus of wild cows comparable to domestic cows that traverse thousands of kilometres through African tropical rainforests and scrubland by travelling together and in large herds of up to a hundred buffalos [3]. Their travel is motivated by a desire to find abundant grazing pastures. They tend to follow the wet weather to find abundant grazing meadows. Because the seasons vary from place to place across Africa's huge expanse, buffalos are constantly on the move in search of their preferred meadows. ABO algorithm is focused to analyse how buffalos use different modes of communication to organise themselves [4]. They have uses various modes of sounds to indicate the danger zone, good and bad area of grazing fields and to encourage their herds to stay and take use of the present resources.

Over the last few decades, there has been a lot of research on the symmetric Traveling Salesman's Problem. It's astonishing how little research has been done on the asymmetrical Travelling Salesman's Problems [5]. This is perplexing because the majority of real-world uses are asymmetric. A postal service official searching for the optimum route to make deliveries to various locations within a specific area is an ideal example. For instance, a bus driver attempting to find the best way to pick up children from schools, delivering meals to residents, and other real-life scenarios illustrate its practical application. Asymmetric solutions are the most probable approach to resolving these challenges [6]. This concept explains this need for research project since it will aim to solve everyday difficulties and, as a result, will have broad applicability. TSP computations are commonly done in the literature using Euclidian or orthodromic distances. The Euclidian distance is determined using x-y dimensions and the Pythagorean formula, and the range among x and y is clearly symmetrical. But at the other end, the orthodromic range estimates the area between the coordinates of two nodes over a portion of the globe [7]. Though more exact than Euclidean measurements, particularly over vast distances, it does provide genuine real-life measurements when nodes are connected by highways or transport systems, as do unbalanced computations that account for one-way mobility and some other construction related issues [8].

A research gap in the field of optimization solutions for solving the Traveling Salesman Problem (TSP) in Graph Theory is the development of hybrid or integrated approaches that combine multiple optimization techniques. While numerous algorithms have been proposed individually, there is a lack of comprehensive studies that explore the synergistic potential of integrating different methods. By combining the strengths of various algorithms such as metaheuristics, exact
algorithms, and machine learning, it may be possible to achieve improved solution quality, faster convergence, and enhanced scalability for solving TSP. Investigating the effectiveness of hybrid approaches and identifying the optimal combinations of techniques for different TSP instances would be a valuable research direction to bridge this gap and advance the field of TSP optimization. The research on the African Buffalo Optimization (ABO) algorithm is of significant importance as it introduces a novel metaheuristic approach inspired by the behavior of African buffalos. By studying and emulating their intelligence, organizational capabilities, and navigational acumen, the ABO algorithm provides a fresh perspective on solving optimization problems. The algorithm's ability to effectively exploit and explore the problem space through teamwork, communication, and knowledge sharing among individuals mirrors the collective intelligence of the buffalo breed. The study demonstrates the algorithm's competitive outcomes by solving benchmark symmetrical Traveling Salesman's Problems and challenging asymmetric TSPLIB instances. By showcasing the potential of bio-inspired optimization, this research contributes to the development of innovative algorithms and expands the range of effective tools available for solving complex optimization problems.

The African Buffalo Optimization and the Traveling Salesman's Problem are introduced in Section II of this work. A Proposed ABO algorithm is presented in Section III. The study's results and discussion are summarised in Section IV.

II. RELATED WORKS

The African Buffalo Optimization (ABO), Ant colony optimization (ACO), LinKernighan algorithm and the hybrid Honey Bee Mating Optimization (HBMO-TSP) are the subjects of this research. The truth that all these techniques have some of the greatest outcomes and compare the effects in the research piques our interest in them.

Buffalo optimization approached with various alarms, [7] In its hunting tasks, the newly built African Buffalo Optimization imitate the warning and alarm sounds of African buffalos. The waa sound alerts buffalos to the existence of attackers or a paucity of pastures, prompting the herds to move on to safety or more profitable regions of the grassland. Once this cry is issued, the animals are instructed to remain vigilant and find a secure or better grazed field. The maa sounds, on the other side, are being used to advise buffalos to relax because there are plenty of grazing pastures nearby and the environment is suitable to grazing. The herds are able to utilize their seek for food sources using these signals. This ABO is being the solution of various optimised problems for both Symmetric and Asymmetric Travelling Salesman Problems and the numerical benchmark functions. ABO seems that it is old algorithm and weak in process, issues in delay and inefficiency, this optimisation is only helpful for African buffalos.

Ant Colony Optimisation is developed by [9]. This is among the most often used optimization techniques. After some preliminary work on Dorigo and Gambardella's Ant Colony System and Marc Dorigo's Ant System, Marco Dorigo and Di Caro developed the Ant Colony Optimization method in 1999. The spontaneous wandering of ants in looking for food prompted the Ant Colony Optimization. Once a food supply has been found, the ant that identified it transports a particle of it back to the nest; likely take a shorter path and continuously depositing pheromones as a way of reminding additional ants of its progress. Neighborhood ants will probably join the succeeding ant in tracking the food resources if they detect the pheromone's fragrance. Once such ants reach the food resources, they take some food bits back to the home and deposit pheromones to improve the original ant's route. This mechanism attracts other ants by increasing the pheromone content on the preferred shortest route. The swarm of ants is on the most efficient route into and out of the protein source in a short amount of time. The main defects of ACO algorithm is fuel cost minimization, less voltage profile, loss in transmission reduction and this computational algorithm is reduced to find the original paths in graphs.

The Lin-Kernighan algorithm is derived by [7] a common search method that finds solutions by carrying out extensive local searches. This method searches employing transfers (or moves) that turn one route into another in order to shorten the route duration until the greatest (cheapest) trip is found. The basic rules of the iconic Lin-Kernighan algorithm are as follows: k Only successive transactions with huge progress are allowed; (ii) 'Shuttered' tours are permitted except when k = 2; (iii) A earlier cracked link should not be provided, and a precipitate link should not be cracked; and (iv) Journeys are only registered through the five closest neighbours; (v) If k = 4, no connection, xi, on the route should be destroyed whether it is a common factor of a tiny proportion of remedy routes; (vi) If future attempts do not provide a good outcome, the research for enhancements is ended. Principles (iii) and (iv) are heuristics principles that may decrease the time but it may produce in a degraded approach, as shown. The improved LinKernighan method has enhanced most of the core rules as a result of this: k Choose a different nodes l; (ii) Choose another network l so that k-l is an applicant side and á (k-l) = 0 and k-l relates to the actual best circuit; otherwise, choose l so that k-l is a determined based or choose l from endpoints not yet chosen. Proceed to stage ii if any locations will not be addressed; (iv) Whenever the network size picked in step iii >1, the sequence of the networks chosen includes the initial route. This algorithm provides results randomly because the algorithm begins with random partition.

The Honey Bees Mating Optimization method is a community metaheuristic, simulates the beehive queen's mating rituals. The princess of the bee's couples with the swarms joining her in flying, dependent on her strength and ability of the queen retains the vehicle's genetics in her spermathecal after pairing, and the swarm is formed when the breeding flight is completed. Part of the answer is the genetics of the drones. A transformation function is utilised to construct the nests. When a progeny is stronger than the queen, this becomes the empress. The leader, labourers, and flyers are the three kinds of bees. The endeavour golden jelly to the offspring, while the flies pair with the monarch and the emperor is the healthiest bee. This is the HBMO algorithm analysed by [10]. Honey Bee Breeding Optimization to Broadening Nearby Search Algorithm (ENST) and Multi-
Phase Nearby Lookup Randomly assigned Flexible Described (HBMOTSP) is a hybrid that combines HBMO with the Broadening Community Search Method (ENST) and the Multi-Phase Nearby Lookup Randomly assigned Dynamic Mapped (MPNS-GRASP). This combination of processes aids HBMO-TSP in reducing calculation time and ensuring effectiveness when addressing Traveling Salesman's Issues. This method, which is based on the Honey Bee Mating Optimizer, has proven to be particularly useful to overcome multimodal optimization issues. This optimization is mainly based on the mating strategy of bees and the problem solving is very competitive with other state.

III. PROPOSED ABO ALGORITHM

ABO was made in an attempt to address problems with certain existing techniques, such as stimulated annealing, Particle Swarm Optimization, Genetic Algorithm and Ant Colony Optimization, such as medium speed, rate of convergence and use of different factors, complex fitness features. The ABO is essentially a replica of the warning 'maaa' contact calls that the herds use to coordinate troops to stay put and utilize a specific grazing field, as well as the warning 'waaa' contact calls that organize the entire species to demand a new or secure grazing position. From this alarm sounds herds were able to optimise their searches in order to reach food-rich areas [11]. Mathematical representations for these various alarms are shown in Eq. (1) and Eq. (2).

\[ M_{x+1} = N_x + St1(pq_{\text{max}} - U_x) + St2(pq_{\text{max,x}} - U_x) \]  (1)

Where, \( M_x \) is the ‘maaa’ alarm of buffalo with reference \( x \). (\( x = 1,2, \ldots \ldots , x \)), \( pq_{\text{max}} \) is the better buffalo in group. \( pq_{\text{max,x}} \) is the good place found in buffalo herds. \( St1, St2 \) Are the parameters for learning. \( M_{x+1} \) denotes the movement of buffalo from their current position \( mk \) to a new location, and it indicates the migration's enormous storage size. Lifestyle. The real behaviour of the animals can be adjusted be accomplished in accordance with Eq. (2).

\[ U_{x+1} = \frac{(U_x + N_x)}{\theta} \]  (2)

Where, \( U_{x+1} \) represents new parameters, \( U_x \) is the alert for “waaa” sound and \( \theta \) is time intervals with respect to the moments of buffalo.

The African buffalos have used these vocalisations to organise together as they navigate through the African woods in pursuit of rich grassy fields to feed their enormous appetites. Each organism's position indicates a solution in the search area in this method. Fig. 1 shows the implementation chart of ABO algorithm. The ABO algorithms are maintaining this structure on every step, it monitors each buffalo's dynamical position as it approaches the \( pq_{\text{max}} \). Based on how far the emphasis is applied at a given iteration, \( x \) and \( pq_{\text{max}} \) are used. The training settings have an impact on every animal's efficiency.

A. Working Procedure of ABO

The ABO begins by allocating herds to subnetworks in the solution space. The mammals can visit any unpopulated node that is nearest and cheaper to them based on probability. The price of the movement, as calculated by the accessible heuristics in the previous move, influences this decision[12]. The cost heuristics for such manoeuvres, the personal profit of the transfer to the animal as judged by its past experience, and the actual benefit of the specific move to the grassy field all impact further movements [13]. This is expressed by the equation by in a subsequent section. The buffalo’s fitness is then updated by the program. In this method, the method finds the best buffalo’s (\( pq_{\text{max}} \)) location in the herds in respect to the best option. The personal record (\( pq_{\text{max}} \)) of each individual is also measured. Its parameters are remembered by the buffalos. If a buffalo's strength and conditioning value is higher than (\( pq_{\text{max}} \)), the method records it as the animal's ideal position, bgmax. Likewise, if a buffalo’s present performance is greater than any other in its memories, the system records it as the animal's greatest (\( pq_{\text{max}} \)). If the \( pq_{\text{max}} \) matches the exit condition at this point, the algorithm ends and returns the greatest buffalo's position matrix as the perfect result [8]. Or else, it moves on to the next cycle and resumes the procedure till the exiting requirements are met.
Step by step working procedure of African Buffalo (ABO) algorithm is given below:

Step 1-Initialization: Define the population size, maximum number of iterations, and other algorithmic parameters. Generate an initial population of buffalo individuals with random positions.

Step 2-Fitness Evaluation: Evaluate the fitness of each buffalo individual by calculating the objective function value for the given optimization problem. Assign a fitness score to each buffalo based on their objective function value.

Step 3-Selection: Select buffalo individuals from the population based on their fitness scores. The selection process can be based on various strategies like roulette wheel selection, tournament selection, or rank-based selection.

Step 4-Movement and Communication: Simulate the movement and communication behavior of African buffalos. Determine the new positions of the buffalos based on their current positions, velocities, and movement rules.

Step 5- Local Search: Apply a local search operator to explore the neighborhood of the buffalo individuals' positions. This step helps improve the quality of solutions by making small adjustments to the positions.

Step 6-Update Personal and Global Best: Update the personal best positions and fitness scores for each buffalo based on the newly obtained solutions. Keep track of the global best position and fitness score achieved by any buffalo individual in the population.

Step 7-Termination Condition: Check if a termination condition is met, such as reaching the maximum number of iterations or achieving a satisfactory solution. If the termination condition is met, proceed to the next step. Otherwise, go back to step 3.

Step 8-Output: Once the termination condition is met, output the best solution found, which corresponds to the global best position obtained during the algorithm's execution. The solution represents the optimized solution for the given optimization problem.

**ABO Algorithm**

Function $P(n) = (P1, P2, \ldots, Pn)^k$

Place buffaloes in solution path

Input fitness values used Eq. (1)

$$N_x + 1 = N_x + Stx_1(pq_{max,x} - U_x) + Stx_2(pq_{max,x} - U_x)$$

Where, $N_x$ and $U_x$ is exploitation and exploration of x buffalo ($x = 1, 2, \ldots)$, $pq_{max,x}$ is the herds fitness.

Location Update $pq_{max,x}$ and $pq_{max,y}$

$$N_x + 1 = \theta(M_x + N_x), \text{where } \theta \text{ is unit time}$$

If $pq_{max,x}$ provides Yes, then proceed.

If $pq_{max,y}$ provides No, then back with initial step

Stopping procedure not performs, and then starts with fitness step

Get Output.

**B. Steps to Solve TSP using ABO**

The ABO has the benefit of solving difficult optimization issues like the TSP with very basic stages. The basic steps for solving the problem are as continues to follow:

- Initial step to migrate buffalos shown in Eq. (3)
  $$f_{xy} = \frac{\mu^{x,y}_{xy} \nu_{xy}^{x,y}}{\Sigma_{j=1}^{n} \nu_{xy}^{x,y}}$$  Here $xy = \pm 0.10$ (3)

- Use Eq. (1) and Eq. (2) and update fitness

- Determine $pq_{max,x}$ and max.

- Using value of heuristic and add non-visited cities of buffalos

- If $pq_{max}$ Updated, Yes then proceed

- If $pq_{max}$ Updated, No then back to initial stage

- If reach with exit parameters, Yes then go for output

- If reach with exit parameters, No then back to fitness

- Get best results (output).

Where, $Stx_1$ and $Stx_2$ are the parameters values 0.5 and 0.3 respectively. $xy$ takes alternate values on +0.10 and -0.10 iterations. $Z$ is the reinforcement.

Fig. 2 shows the work flow of ABO Algorithm The positive reinforcement warning invitation $z$ informs the creatures to stop and utilize the surroundings because there are ample meadows, whereas the negative reinforcement warning $u$ urges them to continue exploring the area because the current place is not profitable [14]. The possibility $St$ of an animal migrating from town $j$ to town $k$ is based on a combination of two possible values: the perceived benefits of the transition, as calculated by certain heuristic denoting the attack's prior appeal, and the evaluation advantage of the relocation to the breed, denoting how effective have been in the sense of making that specific travel. The lowest values suggest whether or not that move is desirable [15].
IV. RESULTS AND DISCUSSION

The ABO was applied to a set of asymmetric TSP (ATSP) datasets and three sets of symmetric TSP datasets and from TSPLIB95 ranges from 50 to 14470 locations in this research. The first experimental compared ABO's effectiveness in TSP situations to data from a recent survey that included St70, Pr76, Ch150, Eil76, KroA100, Eil101, Berlin52, and Tsp225. The following set of studies compared ABO's effectiveness to that of Att48, Rd400, St70, Eil76, Gil262, Pr152, Brd14051, D1291, Pr1002, and Fn14461 in a recent study [16].

The final experiment looked at how well ABO performed in asymmetric TSP scenarios. The findings of the fourth set of trials were compared to those achieved utilizing some common Artificial Neural Networks approaches [17]. The following are the variables for the PSO-related phases of testing: 300 people; 2000 iterations ($G_{max}$): 0.95 inertia weights, $T_1$: 2, $T_2$: 1 rand 1(0,1), and rand2(0,1), (0,1). The following are the HPSACO research factors: Pheromone parameter ($\alpha$): 1.0; heuristic element ($\beta$): 2.0; condensation rate ($\gamma$): 0.05; pheromone quantity: 100; people: 300; iterations ($G_{max}$): 2000; inertia gravity ($\eta$): 0.85; $T_1$: 2, $T_2$: 1; insects (N): 200; pheromones factor ($\alpha$): 2.0; therapeutic factor ($\beta$): 3.0; condensation variable ($\rho$): 0.05. The studies were run on a 3.40 GHz Intel Duo Core i7-3770 CPU with 6 GB RAM and MATLAB [18]. The tests on the asymmetrical Traveling Salesman's Problems were run on a desktop or laptop with a 4 Gb Of ram and Pentium Duo Core 1.80 Ghz cpu. Similarly, the ANN experiments were conducted out with MS Virtual C++ 2009 on an Intel Duo Core i6 CPU [9]. We first did research on eight TSP examples to verify the ABO algorithm's accuracy in handling the TSP. Error rate is calculated using Eq. 4 from the average value referred through fitness factor of each ABO algorithm.

\[
\text{Error Rate} = \frac{\text{Average Value} - \text{Best Value}}{\text{Best Value}} \times 100
\]  

(4)

In all of the tests that were conducted, ABO outscored the other methods. For example, the ABO found the best solution to Eil76 and Berlin52. No other technique came close. In addition, as comparing to other method, the ABO found the closest-optimal response to the remainder TSP instances. The ABO continues to have the optimum value of average outcomes obtained from each method. It's odd that the Hybrid Algorithm (HA), it utilizes a storage matrix comparable to the ABO, really cannot come up with a decent result [15]. Because the HA is a mixture of the ACO and the CGAS, this can be traced back to the employment of multiple factors [2]. The ABO's superiority may also be demonstrated in the usage of computer capabilities in which the ABO is easily the quickest of the four methods.

Table I shows the comparative results of various optimisation technique involves in TSP. The ABO, for example, is 58,885 times better than the ACO, 1,435 times higher than the ABC, and 3,041 programs are to improve than the Hybrid Mechanism in Berlin52 (HA). This pattern can be found in all of the instances under inquiry [19]. Lin-Kernighan algorithm took 0.283 seconds to complete all of the TSP issues above, compared to ACO's 7356 seconds, ABC's 44.11 seconds, and HA's 328.27 seconds. But since ABO uses the route methodology, as opposed to the delayed path fabrication used by the ACO, the ABO's speed can be traced back to good storage management strategies. Fig. 3 shows the Error rate graph, which is typically shows how the error rate changes as the problem size increases. It helps in evaluating the efficiency and effectiveness of different algorithms in solving TSP instances of varying complexities. A lower error rate indicates that the algorithm is capable of finding solutions closer to the optimal solution, while a higher error rate suggests larger deviations from optimality. The graph can compare multiple algorithms or optimization techniques, allowing for a comparative analysis of their performance. It can highlight which algorithms perform better in terms of minimizing the error rate and provide insights into their scalability and suitability for different TSP instances. By analyzing the error rate graph, researchers and practitioners can make informed decisions about the choice of algorithm and the potential trade-offs between solution quality and computational efficiency when solving TSP optimization problems.
TABLE I. PERFORMANCE COMPARISON OF ABO, LIN-KERNIGHAN AND HBMO

<table>
<thead>
<tr>
<th>TSP Samples</th>
<th>Higher Value</th>
<th>African Buffalo Optimization</th>
<th>Lin-Kernighan Algorithm</th>
<th>Honeybee Mating Optimization</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td></td>
<td>Error rate</td>
<td>Plus rate (%)</td>
<td>Processor Time (secs)</td>
</tr>
<tr>
<td>RL11849</td>
<td>11554</td>
<td>0.1</td>
<td>99</td>
<td>0.05</td>
</tr>
<tr>
<td>HK48</td>
<td>87534</td>
<td>0.02</td>
<td>98</td>
<td>0.77</td>
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<td>BRG180</td>
<td>3412</td>
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<td>99</td>
<td>0.03</td>
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<td>1654</td>
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<td>99</td>
<td>1.2</td>
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<td>RL1889</td>
<td>23895</td>
<td>0.225</td>
<td>100</td>
<td>1.5</td>
</tr>
<tr>
<td>U1432</td>
<td>65782</td>
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<td>96</td>
<td>0.876</td>
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<tr>
<td>PR152</td>
<td>47651</td>
<td>0.12</td>
<td>98</td>
<td>0.34</td>
</tr>
<tr>
<td>FL417</td>
<td>89675</td>
<td>0.5</td>
<td>100</td>
<td>1.4</td>
</tr>
</tbody>
</table>

Fig. 4 plus rate for solving the Traveling Salesman Problem (TSP) in optimization problems represents the percentage of instances or problem instances for which a specific algorithm or optimization technique achieves a solution that is better than or equal to a certain threshold. It provides insights into the algorithm's success rate in finding solutions that meet a desired level of optimality. The plus rate graph allows for a comparative analysis of different algorithms or optimization techniques. It shows how well each algorithm performs in terms of meeting a certain level of optimality for various TSP instances. A higher plus rate indicates that the algorithm consistently achieves solutions that are equal to or better than the specified threshold. By examining the plus rate graph, researchers and practitioners can assess the algorithm's effectiveness in providing high-quality solutions for TSP optimization problems. It helps in evaluating the algorithm's reliability, robustness, and suitability for different problem instances. Fig. 5 shows the CPU rate graph for solving the Traveling Salesman Problem (TSP) in optimization problems represents the computational efficiency or runtime performance of different algorithms or optimization techniques. The CPU rate graph shows how the computational time changes as the problem size increases. It helps in evaluating the efficiency and scalability of different algorithms in solving TSP instances of varying complexities. A lower CPU rate indicates faster computation and better efficiency, while a higher CPU rate suggests longer computational time.

The difference in speed between the ABC and the ABO is related to the CGAS utilisation of multiple parameters. The mix of path creation and path enhancement approaches, as well as the utilisation of numerous factors, could have had an impact on the HA's performance [20].
Decision has made has from the following comparison of ABO, Lin-Kernighan and HBMO algorithm, ABO provides better results on both error and plus rate even CPU rate also. Asymmetric Traveling Salesman's Problems cases have optimum solution. The results indicate that ABO gently outperformed than Lin-Kernighan and HBMO optimising solutions to the ATSP cases under investigative process, with a slightly higher accuracy of 99.5 percent compared to 87 percent for Lin-Kernighan and 80 percent for HBMO. However, HBMO was able to find the best solution in 12 of the 20 examples studied, while the ABO was able to find the best option in all cases and came close in the others. Unfortunately, when it comes to the quickness with which findings must be obtained, the ABO is the best algorithm. Because reliability and efficiency are two of the three main methods for evaluating a superior algorithm the ABO can be recognised a better algorithm than other optimization techniques.

In future work, the optimization solutions for solving the Traveling Salesman Problem (TSP) in Graph Theory using the African Buffalo mechanism can be extended in several directions. There is a scope for refining and enhancing the African Buffalo Optimization (ABO) algorithm by exploring different variations and incorporating additional features inspired by the behavior of African buffalos. The scalability of the ABO algorithm can be investigated to handle larger TSP instances with thousands or even millions of cities. The ABO algorithm can be extended to address variations of the TSP, such as the dynamic TSP where the cities and their distances change over time. Adapting the ABO algorithm to handle dynamic scenarios would enable its application in dynamic routing problems where the optimal route needs to be continuously updated. Analyzing the algorithm's performance guarantees, complexity bounds, and robustness to problem variations will contribute to establishing a solid theoretical foundation for the ABO algorithm.

REFERENCES


Multi-feature Fusion for Relation Extraction using Entity Types and Word Dependencies

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Abstract—Most existing methods do not make full use of different types of information sources to extract effective features for relation extraction. This paper proposes a multi-feature fusion model based on raw input sentences and external knowledge sources, which deeply integrates diverse lexical, semantic, and syntactic features into deep neural network models. Specifically, our model extracts lexical features of different granularity from the original input text representation, entity type features from the entity annotation information of the corpus, and dependency features from the dependency trees. Meanwhile, the dimension-based attention mechanism is proposed to enrich the diversity of entity type features and enhance their discriminability. Different features enable the model to comprehensively utilize various types of information, so this paper fuses these features and train a classifier for relation extraction. The experimental results show that the proposed model outperforms the existing state-of-the-art baselines on the TACRED Revisited, Re-TACRED, and SemEval datasets, with macro-average F1 scores of 81.2\%, 90.2\%, and 89.4\%, respectively, improving the performance by 1.4\%, 4.4\%, and 2\% on average, which indicates the effectiveness of multi-feature fusion modeling.

Keywords—Relation extraction; multi-feature fusion; information extraction; dependency tree; entity type

I. INTRODUCTION

Relation extraction (RE) aims to extract the relationships between entities from free text [1], which can provide support for high-level tasks including knowledge graph construction [2], text summarization [3], question answering [4], and so on. As an important and challenging task, RE has recently received considerable attention from researchers. Specifically, neural relation extraction (NRE) models have emerged and achieved promising performance thanks to the remarkable advancement of deep learning [5-7].

It is essential to fully exploit the different types of features to enhance the performance of the RE task. To utilize rich lexical information in the word sequences, many NRE models have been proposed to extract lexical features, including convolutional neural network (CNN) based [8], recurrent neural network (RNN) based [9], recursive neural network (Recursive NN) based [10], and transformer based [11] models. Most recently, without using any external tools or knowledge, Liang et al. [12] proposed a new model that extracts features from the original input sentences at entity mention, segment, and sentence levels. These methods focus on utilizing the overall or local information within word sequences but do not leverage more specific external knowledge, which may hinder performance of the model.

Besides learning lexical features from the raw data, many recent works also use external knowledge including knowledge graphs [13-14], dependency trees [15], and entity types [16] to construct explicit structured features. To infuse prior knowledge from the existing knowledge graphs, some works [13-14] have tried to integrate large-scale pre-trained models with knowledge bases (KBs) and use the models on numerous downstream tasks. Chen et al. [15] proposed a method that encodes and weights the dependency information by utilizing type-aware map memories (TaMM), which achieved outstanding results on the SemEval dataset [17]. Vashishth et al. [16] improved the performance of the RE model by enriching the features with additional entity type information in the graph structure. However, these methods only use a single form of external knowledge and do not encompass the collaborative use of multiple forms of external knowledge.

Despite their effectiveness, existing methods have the following drawbacks:

1) There are various types of information that can contribute to RE tasks, for example, the word sequences can be a source of rich lexical information, the dependency trees can provide syntactic information, and the entity types can provide constraint information of semantic relations over the entities. However, most of the previous works did not take them into account simultaneously and cannot take full advantage of different types of information sources to extract effective features.

2) A specific relation often constrains the entity types of its target entities. For instance, the place-of-birth relation restricts the entity types of a pair of entities to person and location, respectively. Therefore, entity types are important indicators for a specific relation. However, NRE models usually ignore such auxiliary information without using entity type information to impose constraints when extracting relations. Although a few studies have integrated entity type information into relation extraction, they are resource-centric and highly dependent on knowledge bases [16, 18]. Moreover, previous works often combined the coarse entity types of entity mentions with its contextual features, which suffers from coarse-grained entity types as they may fail to distinguish the relations.
To tackle these limitations, a multi-feature fusion model is proposed for RE. The model exploits both raw text data and external knowledge sources to obtain different types of features, filling the gap left by previous methods that did not simultaneously leverage both word sequences and multiple types of external knowledge. In detail, the model constructs representative original input features from the raw data, and obtains entity type and dependency features from external knowledge sources including entity annotation information in the corpus and dependency tree, respectively. Furthermore, the model employs a dimension-based attention mechanism to improve the diversity and discriminability of entity type features extracted from coarse-grained entity type information, addressing the issue of previous models being unable to distinguish between different relations. Finally, considering that different granularity features have complementary effects, we further fuse these features into a single vector via concatenation and perform relation extraction. The experimental results on the three public datasets demonstrate the effectiveness of our model.

Our contributions are summarized as follows:

3) We propose a multi-feature fusion model for relation extraction. To strengthen the ability to capture different kinds of features with various granularities, the model deeply integrates representative original input features with extra knowledge such as entity type information and dependency information, which can significantly boost the model’s performance.

4) We present a dimension-based attention mechanism to enrich the diversity of the entity type features and enhance their discriminability, thus solving the problem of the coarse entity types of entity mentions.

5) We also carry out extensive experiments on the three public datasets. The results verify the benefits of multi-feature fusion modeling, and our model achieves significant improvements over competitive baselines.

The rest of this paper is structured as follows: Section II provides a review of related works; Section III presents the task definition, and Section IV provides the research objective; Section V describes in detail the proposed model; Section VI and VII discuss the experimental setups and results, and Section VIII concludes the paper.

II. RELATED WORKS

Early works on RE were mainly based on statistical machine learning. Kambhatla [19] combined a variety of features with a maximum entropy model for relation classification. Zhou et al. [20] incorporated semantic information into the feature-based relation extraction model to further boost the performance. Overall, these works require a significant level of manual design, and the quality of the handcrafted features has a significant impact on the model’s effectiveness.

With the maturity of deep learning technology, neural networks can automatically learn the potential features in a sentence and have been widely adopted in relation extraction tasks. Existing NRE models can be broadly classified into two categories: sequence-based and dependency-based [21].

Sequence-based models work with word sequences and concentrate on encoding the context information of a sentence by neural networks to capture latent features. Many models using various neural network architectures have been proposed to extract effective lexical features from the input. As CNN has achieved competitive performance on many traditional NLP tasks, Zeng et al. [8] employed it to extract features that contain valid lexical information for RE. Nguyen and Grishman [22] designed CNN models with convolutional kernels of multiple window sizes that can automatically learn implicit features in sentences, minimizing the reliance on external toolkits and resources. Zhang and Wang [10] employed RNN to model sentence context, allowing the model to capture both long-term and temporal features for RE. In order to extract multi-type features from input sentences, Wen et al. [23] combined the gate mechanism with the piecewise CNN to capture the features of the sentence.

Dependency-based models, as opposed to sequence-based models, use dependency parsing information to extract syntactic relations. Using dependency trees in RE has become a mainstream trend [24-25]. However, most dependency trees are generated by tools, which will cause a certain amount of noise, so efficient pruning methods are necessary. There are many pruning methods, which can rely on graph neural networks for key information selection [25-26], or specific attention mechanisms to dynamically select the important dependency information [15].

With the recent advancements in pre-trained language models (PLMs), the latest studies often employed popular models such as BERT [27] or XLNet [28] for RE tasks. Hou et al. [29] directly applied BERT to relation extraction and proposed a BERT-based model. Based on the bidirectional transformer, Yamada et al. [30] built a model to obtain contextualized representations of words and entities by treating them as independent tokens. Joshi et al. [31] extended BERT and proposed the SpanBERT model for span selection tasks. Wang et al. [32] used external knowledge to fine-tune the pre-trained model. Overall, the above-mentioned PLMs-based models have achieved promising success for the RE task.

Although the above studies have made significant progress in the field of relational extraction, however, they still have some shortcomings. Some of them [12,22] only use the original input to extract lexical features and fail to make comprehensive use of different information sources, while some of them [13-14] use the knowledge base to extract entity type features, which requires a large-scale external resource for support. Other studies [24-26] use dependency trees to obtain dependency features, but the pruning methods are quite complex.

Different from the existing NRE models, our model provides several feature extractors to explore various information sources and deeply integrates diverse lexical, syntactic, and semantic features in RE tasks. The model comprehensively uses the original input text, entity annotation information, and dependency information to extract features, and the extraction of entity type features is done in a way
without relying on external knowledge bases. In addition, the extraction method of dependency features is simple and effective. In summary, the model is a multi-feature fusion model, which can not only effectively utilize various types of features to improve the model’s performance but also has the advantage of the low computational cost of feature extraction. To the best of our knowledge, few previous studies have attempted this.

III. TASK DEFINITION

We define the sentence-level relation extraction task discussed in this work as follows. Let \( x = \{x_1, x_2, \ldots, x_n\} \) be tokens of input. Let \( e_1 \) and \( e_2 \) be a pair of entities in the sentence. The RE task will learn a function \( P(r) = h_0(x, e_1, e_2) \), where \( r \in R \) and \( R \) is a pre-defined relation set.

IV. RESEARCH OBJECTIVE

The objective of this paper is to address the following problems with RE: 1. It is necessary to make full use of the original input text and external information sources to effectively build the relationship extraction model. 2. The cost of introducing information sources should not be too high, and the model should not become heavily resource-dependent. 3. The method of extracting features from external information sources should be simple and effective. To achieve the above goals, we propose a multi-feature fusion model for relation extraction, which explores various information sources and investigates the incorporation of diverse lexical, syntactic, and semantic features in relation extraction. In our model, different from the existing works, the external information sources are easily accessible and do not depend on the large knowledge base, which can reduce time and space requirements and can be flexibly applied to more scenarios.

V. PROPOSED MODEL

The motivation of our model is to take full advantage of different types of information sources and fully exploit various types of features to improve performance. Fig. 1 depicts the model’s structure as well as details of each component. The model is made up of three components: 1) origin input feature extractor; 2) entity type feature extractor; 3) dependency feature extractor.

The origin input feature extractor is responsible for capturing multi-granularity hierarchical features from the raw input sentences, including sentence level, segment level, and entity mention level features. Specifically, as the multi-granularity feature extractor named SMS proposed in [12] has achieved remarkable performance, we directly use it to construct the original input features and concatenate them with entity type information and dependency information. The entity type feature extractor follows the design of the key-value memory network (KVMN) [33] by constructing two memory slots to store the type information of the corresponding entities and then inputting the feature information of each slot into the model in combination with the dimension-based attention mechanism. The dependency feature extractor encodes dependency information obtained from the dependency parser. Finally, we classify the relation with a fully-connected layer by aggregating all the extracted features.
In our model, three kinds of information sources are used to extract features, which are the original input sentences, the entity type’s annotation information in the corpus, and the dependency trees of the sentences, respectively. Specifically, the original input sentences provide abundant lexical information, the dependency trees of the sentences carry long-distance syntactic information, and the entity type’s information provides constraints information of the relation. For instance, in the sentence “A former Pakistani lawmaker has been arrested” with the marked entities “Pakistani” and “lawmaker”, the relation between the two entities is “per:origin”. To extract relation, the RE model needs to first capture lexical features of the sentence and the given entities, then catch entity type features and dependency features that are related to a specific relation, by combining the lexical with its syntactic and entity type features, the model can effectively model the contextual information required by RE task and predict the relation. In more detail, the entity types of two entities are helpful to capture the constraints of a specific relation and are important indicators for the relation. As shown in Fig. 2, if the types of the two entities are nationality and person, then there is more likely a “per:origin” or “per:age” or “per:parents” relation. Moreover, considering that the dependency between the two entities is compound (compound expression), combined with the word sequence information and entity type information, the model will prefer to classify the relation as “per:origin” rather than “org:founded_by”.

Fig. 2. Illustration of the relation extraction procedure for an example sentence.

A. Origin Input Feature Extractor

For the origin input feature extractor, we employ the SMS feature extractor proposed by Liang et al. [12], which fully exploits the input sentences to attain multi-granularity hierarchical features.

 Firstly, a sequence of input tokens is transformed into vector representations using a BERT-related text encoder, which can be described as (1):

\[
H = \{h_1, ..., h_n\} = encoder(x_1, ..., x_n)\# 
\]  

(1)

Based on \( H \), max-pooling operations can be used to get entity and sentence features, as shown in (2)-(3):

\[
h_{e_1} = \text{Maxpooling}(h_{e_1}), h_{e_2} = \text{Maxpooling}(h_{e_2})\# 
\]

(2)

\[
h_y = \text{Maxpooling}(H)\# 
\]

(3)

Where \( h_{e_1} \) and \( h_{e_2} \) are the representations of entity pairs, \((i:j)\) and \((k:l)\) are entity indices which delimit entity \( e_i \) and \( e_j \), and \( h_y \) is the input representation which captures global semantic information.

To obtain more information about entities \( e_i \) and \( e_j \) from input sentences, SMS utilizes a mention attention mechanism (Mention Attention in Fig. 1) to extract entity mention level features, as shown in (4):

\[
h'_{e_i} = \text{Softmax}\left(\frac{H^T h_{e_i}}{\sqrt{d}}\right) \cdot H, i \in \{1,2,3\}\# 
\]

(4)

Where \( h'_{e_i} \) and \( h'_{e_j} \) are entity mention features which capture more comprehensive entity information than \( h_{e_1} \) and \( h_{e_2} \), and \( d \) denotes the dimension of vector representation.

To effectively capture the valuable local segments information, based on the n-gram segment level features \( \{H_t\}_{t=1,2,3} \) extracted by CNN with different kernel sizes, SMS then utilizes a segment attention mechanism (Segment Attention in Fig. 1) to obtain mention-aware segment level features by combining the entity mention features \( h'_{e_i} \) and \( h'_{e_j} \) with \( H_t \), which can be described as (5)-(6):

\[
h_m = \text{Softmax}\left(\frac{H_t(W_m[h'_{e_i}; h'_{e_j}]mö_H_t, i \in \{1,2,3\}\# 
\]

(5)

\[
H_t = \text{CNN}_t(H), i \in \{1,2,3\}\# 
\]

(6)

Where \( t \) is CNN kernel size and \( H_t \) contains segment level features of 1,2,3-gram, and \( \{h_m\}_{t=1,2,3} \) contain segments features with different granularity.

Then, SMS utilizes a global semantic attention operation (Semantic Attention in Fig. 1) which uses the concatenation of \( [h_{e_1}; h_{e_2}; h_y] \) as the query to obtain the representation \( h_s \), it contains sentence-level features related to entity mentions and captures deeper semantic features from the contextual representation \( H \), as shown in (7):

\[
h_s = \text{Softmax}\left(\frac{H(W_s[h_{e_1}; h_{e_2}; h_y])}{\sqrt{d}}\right) \cdot H\# 
\]

(7)

Where \( W_s \in \mathbb{R}^{d \times 3d} \) is a parameter matrix. Finally, SMS aggregates different-granularity features by (8).

\[
h_o = \text{ReLU}\left(W_o [h_s; h'_{e_i}; h'_{e_j}; h_m; h'^2_m; h'^3_m]\right)\# 
\]

(8)

Where \( W_o \in \mathbb{R}^{6d \times d} \) is a parameter matrix.

B. Entity Type Feature Extractor

The structure of the entity type feature extractor is shown in the upper right corner of Fig. 1. As a new neural network
architecture, the key-value memory network (KVMN) [33] can effectively model pair-wisely organized information and has wide application scenarios in NLP tasks ([34–35]). Inspired by the architecture of KVMN, we also utilize a key-value structured memory and construct two memory slots to store the corresponding entity type information. Specifically, KVMN defines the memory slot $s_i$ (i is the index of memory slots) as a pair of vectors \( \{k_i, v_i\} \) where $k_i$ is the key and $v_i$ is the value, and stores the context information as a series of memory slots $s_i = \{k_i, v_i\}$. In our work, we build only two slots, $s_1 = \{e_1, \text{entity\_type}_1\}$ and $s_2 = \{e_2, \text{entity\_type}_2\}$ with $e$ referring to the entity and $\text{entity\_type}$ referring to the entity type information. For the entity types which are used to compose features for training the model, there are only a few different types of entities, with entity types such as person or organization appearing more frequently than the remaining entity types such as date, money, etc., which leads to the model use coarse-grained entity types and often concentrates on a few common types, thus losing a certain amount of information diversity. In order to enrich the diversity of entity type features and increase the discriminability of entity type features, we design a dimension-based attention mechanism; it computes the entity type information in each memory slot to ensure that even if the entity types in different inputs are the same, they can have different effects on the model. Dimension-based means it calculates the attention scores for each dimension of the values in each memory slot.

For each memory slot, the keys and values are stored as \( \{h_{e_1}, v_{\text{type}_1}\} \) and \( \{h_{e_2}, v_{\text{type}_2}\} \), respectively, where $h_{e_1}$ and $h_{e_2}$ are the representations of entity pairs, $v_{\text{type}_1}$ and $v_{\text{type}_2}$ are the parameter vectors of the two entity types. The final representations are calculated as shown in (9)-(11).

\[
p_i = \text{softmax}(W_e \cdot h_{e_i} \cdot h_0), i \in \{1, 2\}\tag{9}
\]

\[
h_{\text{type}_i} = p_i \odot v_{\text{type}_i}, i \in \{1, 2\}\tag{10}
\]

\[
h_{\text{type}} = W_t [h_{\text{type}_1}; h_{\text{type}_2}]\tag{11}
\]

where $W_e, W_t$ are the corresponding parameter matrices, $h_{e_i}$ are the shallow entity features introduced in Section V.A, $h_0$ is the original input feature introduced in Section V.A, $p_i$ indicates the importance score of each dimension of $v_{\text{type}}$ in a single memory slot and $\odot$ denotes element-wise multiplication operation.

When utilizing entity type information, the original input features are used to obtain the importance weights of each dimension of the vector for the relevant entity type by using the dimension-based attention mechanism. An illustration of the mechanism is shown in Fig. 3.

In Fig. 3, for example, one sentence contains an entity “lawmaker”, and the other sentence has an entity “father”, both entities belonging to the same entity type “person”, and their parameter vectors of the entity type $v_{\text{type}}$ are the same. Assuming that the dimension of the parameter vector is 768, then by combining $h_e$ and $h_0$, a 768-dimensional weight vector $p$ can be calculated, indicating the importance of each dimension of the parameter vector of the entity type. For entities, “lawmaker” and “father”, the shallow entity features $h^1_e$ and $h^2_e$ are different. For sentences 1 and 2, their original input features $h^1_0$ and $h^2_0$ are also different. As a result of the varied input contexts, the entity type features $h^1_{\text{type}}$ and $h^2_{\text{type}}$ are characterized differently, resolving the coarse-grained problem of entity types, increasing the diversity of entity type features, and improving the discriminability of entity type features.

![Fig. 3. Illustration of the dimension-based attention mechanism.](image-url)
C. Dependency Feature Extractor

The structure of the dependency feature extractor is shown in the lower right corner of Fig. 1. As the existing tools cannot ensure that the auto-generated dependency trees are totally right, a pruning strategy must be used to eliminate as much noise as possible while exploiting valid dependency information. We have tried several alternative approaches which use: 1) dependency information of the whole sentence; 2) dependency information of the whole sentence combined with the dimension-based attention mechanism; 3) dependency information directly related to the entities; 4) dependency information directly related to the entities with the dimension-based attention mechanism. Among these four approaches, approach three achieved good results, while the rest of the approaches failed to meet expectations. Therefore, when extracting the dependency feature, we simply use the dependency information directly related to the entities to prevent the introduction of too much noisy information. Referring to the KVMN, we also construct memory slots to record dependency information.

After getting the dependency parsing results of the input by using the toolkit such as Stanford CoreNLP Toolkit (SCT) or spaCy, for each word in the entity, its memory slot can be expressed as \( s_i = \{k_i, v_i\} \), where \( k_i \) denotes the original word and \( v_i \) denotes the dependency type with its governor obtained from the dependency parse tree. In Fig. 4, as an example, the two entities are “marrow bone” and “cell stem” respectively, and their direct dependencies memory slot list should be \( S = \{\{\text{marrow}, \text{nsubj}\}, \{\text{bone}, \text{compound}\}, \{\text{cells}, \text{dobj}\}, \{\text{stem}, \text{compound}\}\} \).

More specifically, for each memory slot, the key is stored as the entity word vector obtained from the transformer encoder, the value is stored as a parameter vector which is obtained by using an embeddings lookup table, and the dependency features are calculated as shown in (12)-(13).

\[
h_{dep_i} = \sum_{k=1}^{L_i} v_{dep_k} \text{ } \# \tag{12} \]

\[
h_{dep} = W_d[h_{dep_1}; h_{dep_2}] \text{ } \# \tag{13} \]

Where \( L_i \) is the word count of entity \( i \), \( W_d \) is the parameter matrix, \( v_{dep_k} \) is the value vector obtained from the memory slot of the \( k \)-th word of entity \( i \), \( h_{dep_1} \) and \( h_{dep_2} \) are the dependency features for the two entities. By combing the concatenation of \( h_{dep_1} \) and \( h_{dep_2} \) with a linear transform matrix, we can get the dependency feature denoted as \( h_{dep} \).

D. Classification

Finally, we aggregate different types of features and output the relation label. In our model, we provide two feature fusion methods. The one method simply concatenates features, as shown in (14).

\[
h = [h_0; h_{type}; h_{dep}; h_{e_1}; h_{e_2}] \# \tag{14} \]

The other method involves introducing a gating mechanism for the further fusion of various types of features, as shown in (15)-(17), where \( W_{type} \) and \( W_{dep} \) are parameter matrixes, \( b_{type} \) and \( b_{dep} \) are bias parameters, \( \sigma \) is a nonlinear activation function, and \( \Theta \) denotes element-wise multiplication.

\[
h'_{type} = Gate(h_0, h_{type}) = \sigma(W_{type}h_{type} + b_{type})\Theta h_0 \# \tag{15} \]

\[
h'_{dep} = Gate(h_0, h_{dep}) = \sigma(W_{dep}h_{dep} + b_{dep})\Theta h_0 \# \tag{16} \]

\[
h = [h_0; h'_{type}; h'_{dep}; h_{e_1}; h_{e_2}] \# \tag{17} \]

Finally, we use softmax function to get the relation label, as shown in (18), where \( W \) is a trainable weight matrix and \( |R| \) represents the number of relation labels.

\[
\hat{r} = \arg \max_i \frac{\exp(W h_i)}{\sum_{i=1}^{|R|} \exp(W h_i)} \# \tag{18} \]

VI. EXPERIMENTS

A. Datasets

We conduct experiments on SemEval 2010 Task 8 (SemEval) [17], TACRED Revisited (Tac-Rev) [36] and Re-TACRED [37] datasets to evaluate our model. Table I summarizes the statistics of the three datasets.

<table>
<thead>
<tr>
<th>Dataset</th>
<th>Training set</th>
<th>Validation set</th>
<th>Testing set</th>
<th>Relation types</th>
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</thead>
<tbody>
<tr>
<td>SemEval(^1)</td>
<td>8000</td>
<td>-</td>
<td>2717</td>
<td>19</td>
</tr>
<tr>
<td>Tac-Rev(^2)</td>
<td>68124</td>
<td>22631</td>
<td>15509</td>
<td>42</td>
</tr>
<tr>
<td>Re-TACRED(^3)</td>
<td>58465</td>
<td>19584</td>
<td>13418</td>
<td>40</td>
</tr>
</tbody>
</table>

Task 8 of SemEval-2010 aims to develop a standard testbed for future research and to provide a public dataset. The dataset contains 10,717 instances: 8,000 of them are released for training and the remainder is kept for testing. There are nine different types of relations in it, plus an additional “Other” type. When the two entities for each of the nine types of annotated relation types appear in the opposite order, it is implied that the phrase conveys the corresponding inverse relation for that type of relation. For example, the relations Entity-Destination(e1,e2) and Entity-Destination(e2,e1) are
different from one another. Consequently, there are 19 different relation types in the SemEval dataset.

The TACRED Revisited (Tac-Rev) dataset is based on the original TACRED dataset [38]. Alt et al. [36] conducted an explorative analysis of the label quality for the TACRED dataset and found that a large fraction of the instances was incorrectly labeled by the crowd workers, and they corrected the errors in the Dev and Test sets.

Considering that the Tac-Rev dataset restricts revisions to a small subset of labels, and the majority of TACRED remains uncorrected. Stoica et al. [37] applied a better crowdsourcing strategy to re-annotate the entire TACRED dataset and then released Re-TACRED.

B. Settings

Detailed hyper-parameter settings for each dataset are shown in Table II.

<table>
<thead>
<tr>
<th></th>
<th>Tac-Rev</th>
<th>Re-TACRED</th>
<th>SemEval</th>
</tr>
</thead>
<tbody>
<tr>
<td>Learning rate</td>
<td>3e-5</td>
<td>3e-5</td>
<td>3e-5</td>
</tr>
<tr>
<td>Warmup steps</td>
<td>300</td>
<td>300</td>
<td></td>
</tr>
<tr>
<td>Warmup rate</td>
<td>-</td>
<td>-</td>
<td>0.06</td>
</tr>
<tr>
<td>Epoch</td>
<td>4</td>
<td>4</td>
<td>10</td>
</tr>
<tr>
<td>Batch size</td>
<td>64</td>
<td>64</td>
<td>32</td>
</tr>
</tbody>
</table>

PyTorch is used to implement the proposed model. Following the official script, we use the Macro-F1 score to evaluate the models on the Tac-Rev, Re-TACRED, and SemEval datasets. On the Tac-Rev and Re-TACRED datasets, we use the large cased version of SpanBERT as the encoder in the model with its default settings. On the SemEval dataset, we use the uncased version of BERT-base as the encoder in the model with its default settings. Stanford CoreNLP Toolkit (SCT) is used for dependency parsing.

C. Baselines

As PLMs have brought many breakthroughs in various NLP tasks in recent years, to evaluate the effectiveness of our model, we compare it with the following powerful baselines:

1) SMS [12]: SMS is a novel RE model that employs a hierarchical attention mechanism and global semantic attention to fully exploit multi-granularity features, and then aggregates these extracted features to predict the relation.

2) SpanBERT [31]: SpanBERT is a pre-training model that extends BERT using different masking schemes and training objectives. It masks contiguous spans of tokens using a different random process and introduces a span-boundary objective (SBO) that attempts to infer the complete content of the span.

3) KnowBERT [13]: To enhance text representations with structured knowledge, the knowledge-enhanced BERT (KnowBERT) incorporates multiple knowledge bases (KBs) into the BERT model and obtains knowledge-enhanced representations that can be used for a variety of downstream tasks.

4) LUKE [30]: LUKE is a new pre-trained contextualized representation model. By using a huge entity-annotated corpus, it is trained to predict words and entities that have been randomly masked. With regard to a variety of downstream entity-related tasks, LUKE has demonstrated excellent performance.

5) GDPNet [39]: GDPNet creates a multi-view graph to represent various potential relationships among tokens, and the graph is refined through several interactions. Both the refined graph representation and the “[CLS]” token representation of the BERT input sequence is combined to form the input of the softmax classifier, which predicts the type of relation.

6) TaMM [15]: The model uses BERT to encode the input, and then incorporates the dependency information by using a type-aware map memory (TaMM) module. TaMM improves relation extraction performance by leveraging dependency type information with an attention mechanism to obtain each dependency’s importance.

7) C-AGGCN [25]: The model uses dependency trees as inputs and utilizes the graph convolutional network to learn tree structure features in an end-to-end way.

8) RECENT [40]: The model introduces mutual restriction of relation and entity type into the relation classification, which can use the entity type to restrict the candidate relations and avoid some unsuitable relations being candidates.

VII. RESULTS AND DISCUSSION

A. Results on Tac-Rev and Re-TACRED

We evaluate our model on the Tac-Rev and Re-TACRED datasets. The experimental results are shown in Table III, and we follow the official train/dev/test split for these two datasets. For our model, feature concatenation is the default feature fusion method. If the gating method introduced in Section V.D is used as the feature fusion method, the model will be denoted as “(with gate)”. Table III demonstrates that our model yields the highest Macro-F1 scores. When compared to the latest SOTA work such as SMS, the proposed model substantially outperforms the baseline with an absolute improvement of 1.4% on the Tac-Rev dataset and 4.5% on the Re-TACRED dataset. As SMS utilizes origin input features extracted solely from the original input sentences, this proves that our model can benefit from extra knowledge and obtain effective features. Compared to TaMM, we also achieve a 3.2% improvement on the Tac-REV dataset, confirming that comprehensive utilization of origin input features and features extracted from extra knowledge is feasible. It is worth noting that our model outperforms the baselines without the use of an external large knowledge base or large corpus. This also demonstrates the flexibility and effectiveness of our model.

To get a better intuition about how our model works, we conduct an ablation study to analyze the contribution of each component. The results are shown in Table IV. In Table IV, “O” denotes the origin input features, “E” denotes the entity type features, “D” denotes the dependency features, “G”
denotes the gating mechanism introduced in Section V.D, and “att” denotes dimension-based attention, for example, SpanBERT+O+T+att means that the model employs SpanBERT as the encoder and utilizes the combination of the original input features and the entity type features with dimension-based attention. Note that in Table IV, since the computation of the dimensional attention mechanism involves both the original input features and the entity type features, we only add the dimensional attention mechanism when both features are used. Similarly, for the gating mechanism, since its computation involves three feature extractors, we only use it when all three feature extractors are used simultaneously.

As can be seen in Table IV, through the incorporation of various types of features, improvements can be achieved for relation extraction. Specifically, among the three features, we can observe that the entity type feature extractor yields the contribution to the performance with a 1.7% (78.00% vs 79.7%) and 3.1% (85.3% vs 88.4%) improvement on the Tac-Rev and Re-TACRED datasets, respectively. This means that entity type features are important indicators for relation prediction. For the other two types of features, we can also see that they are helpful to improve model performance. As a result, we can see that each of our feature extractors can obtain a boost on the basis of the pre-trained language model, which confirms the feasibility of our feature extractors and shows that all three features are essential for relation extraction tasks.

We can also observe from Table IV that the model performance can also be gradually improved when the three types of features are combined. For example, on the Tac-Rev dataset, when combining the original input features and the dependency features, compared to SpanBERT+O and SpanBERT+D, the model SpanBERT+O+D achieves a performance improvement of 1.5% (79.5% vs. 81%) and 1.8% (79.2% vs. 81%), respectively. The results also demonstrate that the combination of the three types of features can bring positive gains.

In summary, the performance of the model is gradually improved with the addition of modules, which confirms that each key component of our model plays a vital role in relation extraction, and deep fusion of origin input features and extra knowledge will further boost the performance of the model. In addition, we can see that using the dimension-based attention mechanism along with the entity type feature can also further leads to performance improvement, which also indicates the effectiveness of the dimension-based attention mechanism.

To further analyze why using features extracted from extra knowledge is effective, the statistical analysis of the relation labels on the Tac-Rev dataset and Re-TACRED dataset is performed, as illustrated in Fig. 5 and 6, respectively.

Generally, in these two datasets, each instance is annotated with a person-oriented or organization-oriented relation type, such as per:city_of_birth, per:title, org:employees, and so on, otherwise assigned no_relation for negative instances, and each relation label belongs to the “Person” or “Organization” categories. As the entity types mainly consist of categories such as person, organization, location, and so on. Naturally, inputting the entity type information to the model can play a

---

**TABLE III. F1 Score Results on Tac-Rev and Re-TACRED**

<table>
<thead>
<tr>
<th>Models</th>
<th>Tac-Rev (%)</th>
<th>Re-TACRED (%)</th>
</tr>
</thead>
<tbody>
<tr>
<td>SMS [12]</td>
<td>79.8</td>
<td>85.7</td>
</tr>
<tr>
<td>SpanBERT [31]</td>
<td>78.0</td>
<td>85.3</td>
</tr>
<tr>
<td>KnowBERT [13]</td>
<td>79.3</td>
<td>89.1</td>
</tr>
<tr>
<td>LUKE [30]</td>
<td>80.6</td>
<td>-</td>
</tr>
<tr>
<td>GDPNet [39]</td>
<td>79.3</td>
<td>-</td>
</tr>
<tr>
<td>TaMM [15]</td>
<td>78.0</td>
<td>-</td>
</tr>
<tr>
<td>C-AGGCN [25]</td>
<td>75.1</td>
<td>81.0</td>
</tr>
<tr>
<td>RECENT [40]</td>
<td>78.7</td>
<td>86.4</td>
</tr>
<tr>
<td>Our model</td>
<td>81.2</td>
<td>89.8</td>
</tr>
<tr>
<td>Our model (with gate)</td>
<td><strong>81.2</strong></td>
<td><strong>90.2</strong></td>
</tr>
</tbody>
</table>

**TABLE IV. F1 Results of the Ablation Study on the Tac-Rev and Re-TACRED Datasets**

<table>
<thead>
<tr>
<th>Models</th>
<th>Tac-Rev (%)</th>
<th>Re-TACRED (%)</th>
</tr>
</thead>
<tbody>
<tr>
<td>SpanBERT</td>
<td>78</td>
<td>85.3</td>
</tr>
<tr>
<td>SpanBERT+O</td>
<td>79.5</td>
<td>85.7</td>
</tr>
<tr>
<td>SpanBERT+O+D</td>
<td>79.2</td>
<td>88.7</td>
</tr>
<tr>
<td>SpanBERT+D</td>
<td>81</td>
<td>89.5</td>
</tr>
<tr>
<td>SpanBERT+O+D</td>
<td>80.3</td>
<td>88.6</td>
</tr>
<tr>
<td>SpanBERT+T+D</td>
<td>80.4</td>
<td>89.4</td>
</tr>
<tr>
<td>SpanBERT+O+T+att</td>
<td>80.7</td>
<td>89.7</td>
</tr>
<tr>
<td>SpanBERT+O+T+D</td>
<td>80.9</td>
<td>89.7</td>
</tr>
<tr>
<td>SpanBERT+O+T+D+att</td>
<td>81.2</td>
<td>89.8</td>
</tr>
<tr>
<td>SpanBERT+O+T+D+G</td>
<td>81.1</td>
<td>90</td>
</tr>
<tr>
<td>SpanBERT+O+T+D+G+att</td>
<td><strong>81.2</strong></td>
<td><strong>90.2</strong></td>
</tr>
</tbody>
</table>

Fig. 5. The statistics of relation labels on the Tac-Rev dataset.

Fig. 6. The statistics of relation labels on the Re-TACRED dataset.
good hint effect for relation extraction. At the same time, some dependencies also have a significant effect in determining the type of a specific relation. For example, in the sentence “My father built this school”, its' entities are “my father” and “school”, and the dependency between the two entities is obj (Object). When the entity type and dependency information are combined, the model is more likely to predict that the relation type as “org:founded” or “org:founded_by”.

B. Results on SemEval

We also conduct experiments on the SemEval dataset. As it is a classical dataset, besides baselines introduced in Section V.C, the following popular models are adopted as comparison models.

1) **LST-AGCN [41]**: This model aggregates and transports information about syntactic relations and word features in accordance with the grammatical structure, and directly manipulates the graph to derive the representation for relation classification.

2) **DP-GCN [24]**: DP-GCN selects relevant information from dependency trees, and each graph convolutional network (GCN) layer contains a selection module that allows it to filter away information that is irrelevant to the target without using any pre-defined rules.

3) **C-GCN-MG [26]**: The model represents a sentence using multiple sub-graphs and performs graph convolution operations on the sub-graphs to acquire relevant features.

4) **C-DAGCN [42]**: By appending attention modules over the GCN, C-DAGCN further uses distributional reinforcement to guide the GCN for relational extraction.

5) **Two-channel [43]**: The model incorporates the benefits of both the Bi-LSTM-ATT and the CNN channel to predict relation.

6) **MSML [21]**: For the text data, the model constructs feature hierarchy and relation hierarchy and then presents a framework to fully leverage these hierarchies for RE tasks.

7) **POS&DP [1]**: The model uses both the sequential POS tags and the dependency graph structure for the RE task.

The experimental results are shown in Table V.

<table>
<thead>
<tr>
<th>Models</th>
<th>SemEval (%)</th>
</tr>
</thead>
<tbody>
<tr>
<td>BERT-base</td>
<td>87.9</td>
</tr>
<tr>
<td>SMS [12]</td>
<td>88.3</td>
</tr>
<tr>
<td>SpanBERT [31]</td>
<td>-</td>
</tr>
<tr>
<td>KnowBERT [13]</td>
<td>89.1</td>
</tr>
<tr>
<td>LUKE [30]</td>
<td>-</td>
</tr>
<tr>
<td>GDPNet [39]</td>
<td>-</td>
</tr>
<tr>
<td>TaMM [15]</td>
<td>89.2</td>
</tr>
<tr>
<td>LST-AGCN [41]</td>
<td>86.0</td>
</tr>
<tr>
<td>DP-GCN [24]</td>
<td>86.4</td>
</tr>
<tr>
<td>C-GCN-MG [26]</td>
<td>85.9</td>
</tr>
<tr>
<td>C-DAGCN [42]</td>
<td>86.9</td>
</tr>
<tr>
<td>Two-channel [43]</td>
<td>85.42</td>
</tr>
<tr>
<td>MSML [21]</td>
<td>89.1</td>
</tr>
<tr>
<td>POS&amp;DP [1]</td>
<td>87.2</td>
</tr>
<tr>
<td>Our model</td>
<td><strong>89.4</strong></td>
</tr>
<tr>
<td>Our model (with gate)</td>
<td>89.2</td>
</tr>
</tbody>
</table>

Table VI. F1 Results of the Ablation Study on the SemEval Dataset

<table>
<thead>
<tr>
<th>Models</th>
<th>SemEval (%)</th>
</tr>
</thead>
<tbody>
<tr>
<td>BERT-base</td>
<td>87.9</td>
</tr>
<tr>
<td>BERT+O</td>
<td>88.4</td>
</tr>
<tr>
<td>BERT+T</td>
<td>88.2</td>
</tr>
<tr>
<td>BERT+D</td>
<td>88.5</td>
</tr>
<tr>
<td>BERT+O+D</td>
<td>88.6</td>
</tr>
<tr>
<td>BERT+O+T</td>
<td>88.6</td>
</tr>
<tr>
<td>BERT+T+D</td>
<td>88.7</td>
</tr>
<tr>
<td>BERT+O+T+att</td>
<td>88.6</td>
</tr>
<tr>
<td>BERT+O+T+D</td>
<td>89.2</td>
</tr>
<tr>
<td>BERT+O+T+D+att</td>
<td><strong>89.4</strong></td>
</tr>
<tr>
<td>BERT+O+T+D+4G</td>
<td>89</td>
</tr>
<tr>
<td>BERT+O+T+D+4+att</td>
<td>89.2</td>
</tr>
</tbody>
</table>

C. Feature Vector Visualizations

To verify that the dimension-based attention mechanism can enrich the diversity of entity type features, we use t-SNE [44] to project the vector of entity type features into two dimensions. First, we select four sentences from the Re-TACRED dataset, which all contain named entities of type person and organization, and visualize the entity type feature vectors corresponding to the named entities in these sentences, as shown in Fig. 7.
As can be seen from Fig. 7, the four feature vectors belonging to type person are different, and the entity type feature vectors of person type and organization type appear to show two different clusters, i.e., the feature vectors corresponding to two named entities of the same type have a shorter distance on the graph, while the feature vectors corresponding to two named entities of different types have a longer distance on the graph.

![Visualization of entity type feature vectors from 4 sentences.](image)

Similarly, we select 100 sentences from the Re-TACRED dataset for observation, and the results are shown in Fig. 8.

![Visualization of entity type feature vectors from 100 sentences.](image)

From Fig. 8, it can be seen that the feature vectors of type person and organization cluster into two clusters. It indicates that the feature vectors of the same type will be differentiated after applying the dimensional attention mechanism, but the feature vectors of different types can still be clearly distinguished. The visualization results can still meet our expectations when adding more entity types, as shown in Fig 9. We can see that the dimension-based attention mechanism can effectively increase the discriminability of entity type features, thus solving the problem of the coarse entity types of entity mentions.

![Visualization of entity type feature vectors from 300 sentences with 4 entity types.](image)

**VIII. CONCLUSION**

This paper presents a multi-feature fusion model for relation extraction, which explores various information sources and investigates the merging of different types of features in relation extraction. The resulting model can extract abstract features from raw inputs while benefiting from external knowledge. Our study shows that entity type information is especially useful for RE tasks and contributes significantly to the gain in performance from a semantic aspect, while dependency parsing information provides additional benefits. We also demonstrate that deep integration of different types of features makes the proposed model perform significantly better than strong baselines. The experimental results on the three benchmark datasets show that our model is effective and generalizable.

Moreover, our model is mainly composed of a series of feature extractors with a simple architecture. We can add feature extractors according to new external information sources in subsequent research and flexibly integrate them into our model. While our model has achieved good results on annotated datasets, the limitations of its application to unlabeled data still remain for future exploration and resolution. In future work, we will investigate the way to leverage unlabeled data and extend our work to the semi-supervised setting. We also would like to explore knowledge bases to extract additional features and enhance the performance of relation extraction.

**ACKNOWLEDGMENT**

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Mobile Apps Performance Testing as a Service for Parallel Test Execution and Automatic Test Result Analysis

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Abstract—Now-a-days, numerous mobile apps are developed daily that influence the lives of people worldwide. Mobile apps are implemented within a limited time and budget. This is to keep up with the rapid business growth and to gain a competitive advantage in the market. Performance testing is a crucial activity that evaluates the behavior of the application under test (AUT) under various workloads. Performance testing in the domain of mobile app development is still a manual and time-consuming activity. As a negative consequence, performance testing is ignored during the development of many mobile apps. Thus, mobile apps may suffer from weak performance that badly affects the user experience. Therefore, cloud technology is introduced as a solution that emerges in the domain of software testing. Based on this technology, software testing is provided as a service (TaaS) that leverages cloud-based resources. This overcomes the testing issues and achieves high test quality. In this paper, a cloud-based testing as a service architecture is proposed for performance testing of mobile apps. The proposed performance testing as a service (P-TaaS) adopts efficient approaches for automating the entire process. Efficient approaches for test case generation, parallel test execution, and test results analysis are introduced. The proposed test case generation approach applies model-based testing (MBT) technique that generates test cases automatically from the AUT’s specification models and artifacts. The proposed P-TaaS lessens the testing time and satisfies the fast time-to-release constraint of mobile apps. Additionally, the proposed P-TaaS maximizes resource utilization, and allows continuous resource monitoring.

Keywords—Performance testing; mobile apps testing; mobile apps performance testing; automated testing; cloud computing; TaaS; model-based testing

I. INTRODUCTION

The evolution of wireless technology and the development of an immense number of smartphones led to the prosperity of the mobile app development industry [1]. An enormous number of mobile apps are developed and uploaded to different app stores (e.g., Google Play Store) daily. Thus, mobile app testing becomes an urgent matter that must be performed to ensure the application under test (AUT)’s functionality, quality, and reliability before it is released for public use. However, mobile apps usually have a short development life cycle [2]. Thus, many mobile apps are not rigorously tested.

The performance of mobile apps is considered an important concern to users. The prosaic performance of mobile apps roughly affects the user experience [3]. Therefore, mobile app performance testing is considered an indispensable activity. Mobile app performance testing refers to the determination of the AUT behavior under various workloads of concurrent users [4]. This ensures the AUT’s responsivity to the concurrent users’ instructions as well as discovering AUT vulnerability under various workloads.

At present, cloud computing with its virtualization technologies has become a critical orientation in the information technology industry [5]. The integration of cloud technology with the software engineering domain led to an evolution in the field of software testing. Therefore, the expression of cloud testing appeared. The cloud-based testing frameworks provide an on-demand TaaS for testing any type of software app including mobile app testing [6]. TaaS is defined as a service model that automatically carries out the entire testing process in a cloud-based environment. Then, it submits the test results to the end user. Consequently, the TaaS architecture can be used to provide performance testing as a service (P-TaaS) using cloud-based resources.

However, the challenges found in the literature [7-10] related to performance testing and P-TaaS include the following:

- The majority of researchers focus on discussing functional testing in the context of cloud testing. However, performance testing was relatively rare. Thus, few researchers introduced a comprehensive architecture for automating the entire performance testing process and utilizing cloud-based resources.
- Most of the existing performance testing research focuses on performance test case generation and execution. Few works present systematic approaches to automatically analyze the performance test results.

Therefore, the main contributions of this paper include proposing the following:

- A performance testing as a service (P-TaaS) architecture that automates the whole performance testing process for hybrid mobile apps. Hybrid mobile apps are web apps that are downloaded from mobile app stores and need an internet connection to operate.
- An automated approach for performance test case generation. Thus, no need for skillful testers to design test cases that resemble real users’ scenarios when using AUT under various workloads.
• An approach for simultaneous test case execution on multiple virtual nodes. This reduces the time required for the test execution process.

• An automated approach for performance test results analysis that can detect the performance bottleneck in the AUT.

The rest of the paper is organized as follows: Section II presents brief background information. Section III surveys the most relevant work related to performance testing and P-TaaS specifically. Section IV presents the proposed mobile app P-TaaS architecture and a detailed explanation of the functionality of each module. Section V shows the experimental results. Section VI mentions the limitations of the proposed P-TaaS and the differences between the proposed P-TaaS and other relevant tools. The paper is concluded in Section VII. Finally, the future work is presented in Section VIII.

II. BACKGROUND

This section provides brief background information about concepts that are utilized in the proposed mobile app P-TaaS architecture (i.e. model-based testing and the OCL-based UML diagrams).

The Model-based Testing (MBT) [11] is known as an automatic testing technique that generates performance test scenarios from the AUT specifications that are represented by software models. Unified modeling language (UML) [12] is one of the most widely used methods to model software apps. MBT automatically generates test cases from the software models that represent the behavior and the requirements of the AUT. Then, the software models are converted into test models (e.g., Finite State Machine (FSM)). FSM graph is defined as a set of AUT states, where the inputs trigger each transition and convert AUT from one state to another. FSM graph is traversed to obtain paths. Each path represents a user behavior when using AUT. Thus, FSM graph allows the automation of test case generation.

OCL [13] stands for object constraint language. Generally, OCL is used as a formal language for adding user-defined constraints on the UML diagrams. The OCL as a formal language includes three types of constraints: (i) invariant, (ii) precondition, and (iii) post conditions. The invariant constraint added to any object means that this constraint must be true for the entire lifetime of that object. The precondition constraint added to an operation shall be true before the operation execution. The post condition added to an operation shall be true just after the operation execution.

III. RELATED WORK

Many researchers were concerned with studying mobile app performance testing in a cloud-based environment from different perspectives. The benefits and challenges of mobile app performance testing using cloud-based resources are widely discussed in the literature [5], [6]. Ali et al. [14] reviewed the most recent studies and research gaps related to the performance testing using cloud-based resources. Section III A discusses the most recent mobile apps performance testing frameworks, as well as some of the performance testing tools and services widely used in the market. Section III B introduces relevant studies on the adoption of model-based testing (MBT) techniques in mobile app testing. Section III C discusses the performance test results analysis and interpretation techniques. Finally, Section III D introduces recent studies related to the resource utilization and scheduling approaches adopted in the TaaS domain.

A. P-TaaS Frameworks and Widely Used Performance Services in the Market

Mobile apps performance testing frameworks based on the cloud-based environment were presented in the literature. For instance, Prathibhan et al. [15] presented Android Testing as a Service framework known as (ATaaS). The framework depends on the emulators and Android Application Package (APK) file of the AUT as input to execute test cases. Performance test cases are generated manually by the testers, then test cases are recorded to be executed several times under various workloads. The author did not ensure the efficient utilization of cloud-based resource. The presented ATaaS framework focused only on test case execution and ignored the rest of the performance testing activities.

There are cloud-based performance testing tools and services that are adopted for both small and large businesses with various pricing structures. SOASTA CloudTest [16], LoadStorm [17], and Xamarin Test Cloud [18] were selected from the list of the top 10 extremely used cloud-based performance testing tools in 2020 [19]. SOASTA CloudTest depends on manual test case generation and test results analysis. LoadStorm generates a performance test results analytics report that represents the performance metrics, such as response time and error rate. LoadStorm does not support automatic test case generation. It requires testers to record test scenarios manually. Xamarin Test Cloud allows cross-platform mobile app testing. It supports multiple tenets to execute tests over thousands of devices. Xamarin Test Cloud has limited access to open-source libraries.

B. Model-Based Testing Techniques for Mobile Apps

Model-based testing (MBT) techniques are widely adopted in mobile app testing. Researchers introduced the MBT approach for mobile app test case generation. For instance, Usman et al. [12] adopted UML class diagrams and state machine models in their proposed performance testing approach. The proposed approach generates abstract test cases automatically using UML class diagrams, state machine models, and OCL constraints. The proposed approach was experimented on two different Android apps. The results showed that the proposed approach successfully estimates the performance of apps under test. However, the author only focused on conducting the performance testing of the code related to the business logic code. Additionally, he ignored testing the performance of the GUI.

C. Performance Test Results Analysis and Interpretation

Techniques for performance test results analysis and interpretation were studied in the literature in the field of P-TaaS. Researchers presented frameworks for the automatic analysis of performance test results. Liu et al. [20] proposed a framework for analyzing test results and detecting
performance bottlenecks. The proposed framework increased the workload iteratively and monitored the AUT performance metrics simultaneously. The proposed framework was based on cloud infrastructure. However, the author did not discuss issues related to conducting the performance test efficiently using cloud resources. Additionally, the author ignored mentioning the used approaches and tools in test case generation, test execution, and workload generation.

D. Scheduling and Resource Utilization Techniques

There are many researchers presented scheduling approaches to enhance resource utilization, especially in the TaaS field. For instance, the fuzzy sets theory was applied to schedule test cases in the TaaS platform by Lampe et al. [21]. The author applied the fuzzy sets theory as a solution to address the difficulty of predicting the duration of test case execution in advance. This is called uncertainty task scheduling issue. Two algorithms were proposed by the author to handle the uncertainty task scheduling issue. The proposed algorithms are based on simulated annealing (SA). The author used the Q-recent estimate for each test case to estimate the average durations from the history of executions of a given test case.

The metaheuristic methodologies were proposed by Rudy [22]. The author applied metaheuristic methodologies to schedule the parallel test case execution in the context of TaaS. Genetic algorithm (GA) [23] is an example of the metaheuristic methodologies that was used by the author. The presented metaheuristic methodologies assume that the test case execution time is unknown before the test execution. The proposed metaheuristic methodologies have the following drawbacks (i) it needs a long computation time; (ii) the metaheuristics can operate on a limited number of test cases at once (i.e., 300 for SA and 100 for TS and GA). These limitations badly influence the quality of the solution.

Therefore, it observed from analyzing the previous related work that there is no comprehensive framework that conducts the whole performance testing process automatically and leverages the cloud-based environment efficiently.

IV. THE PROPOSED MOBILE APP P-TaaS ARCHITECTURE

This paper proposes a P-TaaS architecture for mobile apps. Tester submits a request to the proposed P-TaaS where the entire performance testing process will be automatically processed. The overall architecture of the proposed mobile apps P-TaaS is shown in Fig. 1. The five main layers of the proposed architecture are as follows (1) user interface layer; (2) performance testing layer, (3) service management layer, (4) infrastructure as a service (IaaS) layer, and (5) data repository.

1) User interface layer: The user interface layer is the top web-based layer of the proposed architecture. It is where the testers can interact with the proposed P-TaaS architecture. The tester submits the test input files to accomplish the performance test. Then, the tester receives the test results report through it.

2) Performance testing layer: The performance testing layer is responsible for automatically accomplishing all the performance test activities including test case generation, parallel test case execution, test result analysis and interpretation, and finally the test report generating. The proposed approaches that are applied to each of these activities are discussed in the consequent subsections IV A.

3) Service management layer: The service management layer is concerned with managing, monitoring, and scheduling the test execution tasks among the available resources. The main modules of the service management layer are the scheduler, runtime monitor, and resource allocator. A detailed explanation of each module will be introduced in Section IV B.

4) IaaS layer: The infrastructure as a service (IaaS) layer includes virtual machines (VMs) where the testing process physically occurs. The virtualization technology is applied to provide all needed resources in the proposed P-TaaS architecture.

5) Data repository: The data repository is where the proposed mobile app P-TaaS architecture stores all generated data during its operation. These data include the following: (1) The test cases generated from the test case generation module, (2) The performance measurements generated from the test case execution module, (3) Information produced from the test results analysis module, (4) The test reports obtained from test report generation module, and (5) VMs status (i.e., on, off, or idle) detected by the monitor module.

Fig. 1. Mobile apps P-TaaS architecture.

A. The Performance Testing Layer

The performance testing layer is the core layer of the proposed mobile app P-TaaS architecture. In this section, the proposed approaches employed in each module of this layer will be briefly explained.

1) The test case generation module: The test case generation module is responsible for the automatic generation of AUT’s performance test cases. Generating test cases is an essential activity of the entire testing process [12]. Performance testing requires a set of appropriate test cases that can evaluate the responsiveness of the AUT under various workloads of concurrent users’ accesses [3]. Generally, a
performance test case composes of a consecutive set of actions exerted on the mobile app’s GUI widgets. Performance test cases should resemble real user scenarios when using AUT. Besides, they should achieve full coverage of all the AUT’s GUI actions and activities [4]. Automating the process of mobile app performance test case generation lowers the cost and raises the efficiency of testing. Additionally, it can improve the accuracy of test results.

In this paper, the proposed test case generation approach is based on the MBT methodology. The proposed MBT methodology depends on the OCL activity diagrams. Activity diagrams are used to model AUT’s workflows in terms of stepwise activities and actions. The OCL defines the performance requirements formally. The OCL-based activity diagrams become a good candidate for modeling user behavior which allows automated mobile app performance testing. OCL-based activity diagrams are used to represent the AUT flow of actions as well as the performance requirements associated with every GUI action. The proposed approach targets interactive hybrid mobile apps. Interactive hybrid mobile apps [24] perform the requests of users who interact with the apps through the GUI using an internet connection.

The proposed test case generation approach depends on representing each functionality in the AUT by a separate OCL-based activity diagram. Each AUT’s functionality will be modeled as an activity diagram. Besides, each activity diagram will have OCL constraints associated with each action in the activity diagram. Thus, the prerequisite inputs of the proposed mobile apps performance test approach are acquired from the analysis and design team. These inputs include: (a) an activity diagram for each functionality in AUT; (b) OCL constraints added to each activity diagram.

For each OCL-based activity diagram, which are submitted by the tester through the user interface layer of the proposed P-TaaS architecture, the steps of the proposed test case generation approach go as follows:

- OCL Based Activity Diagram is parsed into an XML Based Activity Diagram. This is considered the primary step to automate the test case generation process.
- All details included in each XML Based Activity Diagram are extracted and inserted into a Predicate Table. Each XML Based Activity Diagram is converted to its corresponding Predicate Table. Each element included in the XML Based Activity Diagram is shown as a row in the Predicate Table.
- The following information about each element of its corresponding XML Based Activity Diagram are stored in the Predicate Table:
  - Element name.
  - Element type (i.e., action, decision, initial, and final nodes).
  - Performance constraints on this element (i.e. represented with OCL constraints)
  - List of all edges out from this element.

Edges are used to represent the dependency between elements. The dependency between elements is proved by the existence of an edge out from an element which is the same as the edge to the current element.

- A Finite State Machine (FSM) Graph [25] is automatically created from each Predicate Table.
- The Depth First Search (DFS) [26] algorithm is applied to the FSM Graph to find all available Independent Path. The Independent Path refers to any path from the start node to the terminal node of the FSM and has at least one new edge that has not been traversed before.
- The Longest Common Subsequence (LCS) [27] algorithm is applied to filter the previously generated independent paths. LCS is an algorithm that eliminates test paths that are included as a sub-path in another path, to obtain basic paths. Applying the LCS algorithm decreases the number of test paths and ensures high test coverage. Additionally, it guarantees the existence of each GUI actions that resemble an actual user scenario, as well as an analogy of an entire path inside the AUT.

2) The test case execution module: Test case execution module is responsible for the automatic execution of mobile app performance test cases in a cloud-based environment. In the proposed mobile app P-TaaS architecture, test cases are executed simultaneously on multiple VMs. This leads to a reduction in the overall testing time, cost, and effort. The input to the test case execution module includes: (i) abstract test cases; (ii) Android Application Package (APK) file of the AUT. The tester submits these two inputs to the proposed P-TaaS through the user interface layer. APK file is a file format used by the Android operating system [28]. It assists in the easy distribution and installation of Android apps. The proposed P-TaaS architecture is based on using the APK file of the AUT. Hence, there is no necessity for the AUT’s source code.

The automatically generated abstract test cases are converted to test scripts by the tester. The tester records these abstract test cases using the capture and reply test methodology [29]. Then, these test scripts will be executed to measure the AUT performance characteristics and responsivity toward the heavy load of concurrent users’ access. During the test case execution, performance response time and error rate are measured. Response time is defined as the time taken by the AUT to respond to a certain action. Thus, response time is the total time between sending the request and receiving the response [3]. The error rate is
defined as the ratio of failed requests with respect to the total number of requests [3].

Generally, there are several mobile app performance testing tools for test case execution. Apache JMeter [30] is an open-source tool that is first used to test the performance of web applications. Eventually, Apache JMeter expanded to allow mobile app performance tests as well. Proxy is used by the JMeter tool to record requests to mobile apps. The proxy configuration is performed on mobile devices. Then, the request will be captured by JMeter. Additionally, the JMeter tool allows the scaling during the performance testing process by providing any number of the virtual workload of concurrent users. Therefore, the JMeter tool is selected in the proposed P-TaaS to simulate different workloads and test the AUT under heavy workloads to detect the performance bottlenecks in the AUT.

The entire process of test case execution is shown in Fig. 2. The steps of the test case execution process are as follows:

a) Firstly, the test case execution module retrieves the automatically generated abstract test cases from the data repository.

b) The tester records the steps included in every abstract test case using the capture and reply test methodology.

c) Test scripts are recorded by the JMeter recording proxy. The recording proxy of the JMeter can record the HTTP requests executed by the tester on mobile devices. This is used for test script generation.

d) Test scripts are stored in the data repository.

e) The test case execution module submits the captured test scripts and APK of the AUT to the assigned VMs to execute test scripts in parallel on multiple VMs.

f) APK file will be installed on the assigned VMs.

g) Then, each test script is executed with a different workload to test the responsivity of the AUT under various workloads and reveal the AUT bottlenecks.

h) Finally, test results are stored in the data repository for further analysis and interpretation.

3) Test result analysis module: The test result analysis module is responsible for the automatic analysis of test execution results. In this paper, an approach for performance test results analysis is proposed. The proposed approach collects the performance metrics values (i.e. response time and error rate) under different workloads. Then, it interprets these data to determine the performance critical turning point and the heavy workload value. Performance critical turning point refers to a point during the AUT execution under various workloads, where the error rate suddenly increases and the response time of the AUT increases exponentially [20]. This indicates that the performance of the AUT descends sharply and the AUT may crash. When the AUT reaches the performance critical turning point, this implies that a performance bottleneck occurs under this workload. Heavy workload means the value of workload is more than or equal to the workload where the performance critical turning point occurs.

The required input to the proposed performance test results analysis approach includes the minimum and the maximum number of concurrent users defined by the tester. The tester defines the minimum and maximum number of concurrent users that simulate the expected minimum and maximum workloads exposed to the AUT during its production. The test shall consider the expected workloads during daily operations, peak hours on the AUT, and the most popular days the AUT will be used. The performance metrics are measured starting from the minimum number of concurrent users’ accesses to the AUT, until the maximum number of users’ accesses. During test execution, the response time and error rate are measured at each workload from the minimum to the maximum workload values. Additionally, the Error_Threshold refers to the maximum error rate value that is accepted by the tester. The AUT is considered in an unstable state when its error rate value reaches the Error_Threshold during the test execution. The Error_Threshold is defined by the tester during the proposed performance test results analysis approach.

Fig. 3 shows the pseudocode of the proposed performance test results analysis approach. The steps of the proposed performance test results analysis approach go as follows:

a) Firstly, loop on the number of concurrent users starting from the minimum number to the maximum number. For each iteration, the workload value will be increased by 50.

b) For every workload value:

- Performance metrics (i.e., response time and error rate) are measured.

- The measured performance metrics values are added to Response_Time_Measurements, and Error_Rate lists.

- \( H_{Response\_Time} \) refers to the length of the perpendicular line on the line connecting the first and last value of response time at the minimum and the maximum number of concurrent users, respectively. Fig. 4 shows \( H_{Response\_Time} \) at a certain workload value [20]. Equation 1 shows how to calculate \( H_{Response\_Time} \) at a certain workload \( x \).

\[
H_{Response\_Time}[x] = A_{RT}[x] \sin(C_{RT}[x]^2 + C_{RT}^2 + B_{RT}[x] \sqrt{2} \times A_{RT}[x] \times C_{RT})
\]
A\_RT refers to the length of the line between the following two points: the coordinates of the first point are (minimum workload, value of the measured response time at minimum workload) and the coordinates of the second point are (certain workload x, value of the measured response time at this workload x).

B\_RT refers to the length of the line between the following two points: the coordinates of the first point are (maximum workload, value of the measured response time at maximum workload) and the coordinates of the second point are (certain workload x, the value of the measured response time at this workload x).

C\_RT refers to the length of the line between the following two points: the coordinates of the first point are (minimum workload, value of the measured response time at minimum workload) and the coordinates of the second point are (maximum workload, and value of the measured response time at maximum workload).

Sin and Cos refer to the trigonometrical sine rule and cosine rule, respectively.

Equation 1 is used to calculate the length of the perpendicular line H\_Response\_Time.

The perpendicular line H\_Response\_Time which corresponds to the workload value x. The performance critical turning point [20] is the point with longest perpendicular line.

c) Secondly, the error rate value that exceeds the Error\_Threshold value is determined.

d) Thirdly, the workload value where the error rate exceeds the defined threshold value is captured. The captured workload value is called the Unstable\_Workload. It is where AUT starts to crash, and its behavior becomes unstable.

e) Fourthly, RT\_Critical\_Turning\_Point is determined. It refers to the maximum value of response time in the H\_Response\_Time list. It has a workload value less than the captured Unstable\_Workload.

f) RT\_Critical\_Turning\_Workload is captured. It refers to the workload value at which the RT\_Critical\_Turning\_Point occurs.

g) Finally, the value of the RT\_Critical\_Turning\_Workload is obtained. This workload value represents the Performance\_Critical\_Turning\_Workload.

h) Heavy\_Load refers to any workload value more than the value of the Performance\_Critical\_Turning\_Workload.

\begin{align}
\text{Input:} & \text{ Min\_of\_Concurrent\_Users, Max\_of\_Concurrent\_Users, Error\_Threshold} \\
\text{Output:} & \text{ Performance\_Critical\_Turning\_Workload, Heavy\_Load, Response\_Time\_Measurements, Error\_Rate\_Measurements} \\
\text{Start} & \\
\text{For} & \text{ } (x= \text{Min\_of\_Concurrent\_Users; } x < \text{Max\_of\_Concurrent\_Users; } x+=50) \\
& // Loop to execute the test cases on different workloads \\
& \{ \\
& \text{Response\_Time\_Measurements} \ [x] = \text{Get\_Response\_Time} \ (x) \\
& // calculate the response time at workload x \\
& \text{Error\_Rate\_Measurements} \ [x] = \text{Get\_Error\_Rate} \ (x) \\
& // calculate the error rate at workload x \\
& \text{H\_Response\_Time} \ [x] = \text{A\_RT}[x] \text{Sin}(\text{Cos}(\text{A\_RT}[x]^2 + \text{C\_RT}^2 + \text{B\_RT}[x]^2)/2 \cdot \text{A\_RT}[x] \cdot \text{C\_RT}) \\
& \} \\
& \text{For} \ (i=0; \ I < \text{Error\_Rate\_Measurements\_Length}; i++) \\
& // Loop to determine the unstable workload where the AUT starts to crash \\
& \{ \\
& \text{If} \ (\text{Error\_Rate\_Measurements\_Length} \geq \text{Error\_Threshold}) \\
& \{ \\
& \text{Unstable\_Workload} = i; \\
& \text{Break}; \\
& \} \\
& \} \\
& \text{For} \ (i= \text{Min\_of\_Concurrent\_Users}; i< \text{Unstable\_Workload}; i++) \\
& \{ \\
& \text{RT\_Critical\_Turning\_Point} = \text{Max} \ (\text{H\_Response\_Time} \ [i]); \\
& \text{RT\_Critical\_Turning\_Workload} = i; \\
& \} \\
\text{Performance\_Critical\_Turning\_Workload} = \text{RT\_Critical\_Turning\_Workload}; \\
\text{Heavy\_Load} \text{ is more than or equal to the RT\_Critical\_Turning\_Workload} \\
\text{End} \\
\end{align}

Fig. 3. Performance test results analysis approach.

Fig. 4. Performance critical turning point.

4) Test report generation module: The final activity of the performance testing process is to generate the test report and submit it to the tester through the user interface. The test report gathers test results collected from multiple virtual
nodes. The test report contains: (i) the automatically generated test cases; (ii) the measured performance metrics (i.e., response time measurements and error rate measurements); (iii) the information produced from test results analysis and interpretation (i.e., performance critical turning point, and heavy workload value). This can be valuable for testers to determine the performance deviations and bottlenecks in the AUT. Consequently, testers utilize this information to resolve performance issues within the AUT.

B. Service Management Layer

The service management layer is responsible for managing and optimizing the cloud-based resources and infrastructure. The aim of this layer includes the following: applying a suitable scheduling approach for simultaneous test case execution; reducing the overall testing time; handling uncertainty issues in scheduling the test case execution; improving the resource utilization; monitoring the runtime status of resources.

The service management layer includes three modules: (1) runtime monitor, (2) scheduler, and (3) resource allocator. An insightful explanation of the role of each module will be mentioned in this section.

1) Runtime monitor module: The runtime monitor module guarantees a high-reliability level. The monitor module is implemented as a local service for tracking the runtime status of all virtual nodes. Then, it stores a list of the available VMs. Additionally, it determines the state of each virtual node (i.e., idle, busy, and fail). The list of available VMs is sent to the scheduler and resource allocator modules, in order to assign tasks to available VMs.

2) Scheduler module: The scheduler module is developed as a local service for sorting and prioritizing test tasks submitted to the proposed P-TaaS architecture [31]. This module aims to achieve efficient utilization of the resources.

Test task duration is hard to predicate before the actual test task execution. Additionally, test task duration varies from one task to another. This is considered an uncertainty scheduling issue [22]. The proposed approach considers the uncertainty scheduling issue. The test task scheduling approach depends on task waiting time and task deadline. Task waiting time is the amount of time between task submission and the current time. The task deadline is the time defined by the tester when the task is submitted to the user interface of the proposed P-TaaS architecture. The task deadline is the predefined time at which the task must be finished and delivered before it.

The proposed scheduler approach goes as follows: firstly, the scheduler module sorts tasks in ascending order according to the task deadline. This means that tasks with an earlier deadline will be executed earlier. Secondly, tasks with the same deadline will be arranged according to their waiting time. For tasks with an equal deadline, the task with a higher waiting time will be executed earlier.

3) Resource allocator module: The resource allocator module aims to achieve a high level of resource utilization and load balance. The resource allocator module receives an ordered list of test tasks from the scheduler module. Additionally, the list of available virtual nodes is sent to the resource allocator from the monitor module. Consequently, the resource allocator module allocates test tasks to certain virtual nodes, in such a way that guarantees the load balance between virtual nodes.

The proposed resource allocation approach goes as follows:

- Each virtual node in the proposed P-TaaS has a waiting queue. It includes a list of tasks assigned to it.
- The length of the waiting queue of each virtual node is calculated.
- Virtual nodes are sorted in descending order according to the calculated waiting queue length.
- The virtual nodes with small waiting queue lengths will be assigned to high-priority test tasks.

V. EXPERIMENTAL RESULTS

Experiments are carried out to assess the applicability of the proposed P-TaaS. In the experiments, three virtual machines are used to simulate a cloud-based environment and support the simultaneous execution of test cases. The three virtual machines are created using VMWare workstation [32]. The VMWare workstation allows the creation of multiple virtual machines on the same physical machine. The experiments are carried out on a machine with the following specs: processor Intel Core i5, memory 8 GB, and Windows 10 operating system. Thus, three virtual machines are created to suit the specs of this physical machine and allow the simulation of simultaneous test execution of the cloud-based environment. The JMeter performance testing tool [30] runs on virtual machines. MS SQL is used to develop data repositories that store the automatically generated test cases and test results (i.e., performance measurements, test results interpretation, and analysis information). The OCL-based activity diagrams are drawn using the Enterprise Architect tool [33]. In addition, the Enterprise Architect tools parses the drawn OCL-based activity diagrams to XML-based activity diagrams with their corresponding OCL constraints. The experimental environment is the same for executing each task using VMs with equivalent specifications.

Nowadays, Android dominates almost 88% of the mobile device market worldwide [34]. Therefore, our experiments depend on using Android apps as AUTs. The APK files of the android AUTs are installed on the virtual node before the test case execution starts. The objective of the proposed P-TaaS architectures is testing the performance of hybrid interactive mobile apps. Therefore, two hybrid mobile apps were chosen for experiments. They are the SpeedTest app [35] and GoodReads app [36]. GoodReads app looks like an online bookstore, where users can read, browse, recommend, and review books. It is a widely used app worldwide. It is the first ranked app in the category of (Science and Education, Libraries and Museums) in the United States. The total number of visits to the GoodReads apps reached 119.7M [37].
SpeedTest app is an Android app that measures the speed of the internet. It is the most used app for measuring the upload/download speed of the internet. It is used by more than 45 billion times unparalleled. Therefore, GoodReads and SpeedTest apps are selected as AUTs in the experiments.

A. Experimental Results of the Proposed Performance Test Case Generation Approach

The purpose of this experiment was to evaluate the proposed mobile app performance test case generation approach. The proposed test case generation approach is assessed in terms of action coverage, activity coverage, and performance requirements coverage. The action coverage refers to the percentage of actions included in the automatically generated abstract test cases to all actions included in the activity diagrams. Similarly, the activity coverage. Performance requirements coverage refers to the ratio between the performance requirements covered by test cases and all performance requirements in the activity diagrams. The proposed test case generation approach is based on the black box MBT methodology. The experimental results of applying the proposed test case generation approach on the SpeedTest app and GoodReads app are presented in Tables I and II, respectively.

As shown in Table I, the number of automatically generated abstract test cases was 17. They were used to test seven major functionalities of the SpeedTest app. The total time required to automatically generate test cases using the proposed approach was 3.055 seconds. It is observed from Table II that the number of automatically generated abstract test cases was 28. They were used to test 13 major functionalities of the GoodReads app. The total time required to automatically generate test cases using the proposed approach was 6.325 seconds. Additionally, it is observed from Tables I and II that the generated test cases for each functionality cover 100% of both activities and actions exerted during each functionality. Therefore, it is considered faster than the manual approaches, which need a long time to design, write, and review the coverage of the test cases.

<table>
<thead>
<tr>
<th>Functionality</th>
<th>Number of GUI Activities</th>
<th>Number of GUI Actions</th>
<th>Number of Test Cases</th>
<th>Time Spend to Generate Test Cases</th>
<th>% of Covered GUI Activities</th>
<th>% of Covered GUI Actions</th>
</tr>
</thead>
<tbody>
<tr>
<td>1. Measure download/upload speed</td>
<td>2</td>
<td>2</td>
<td>1</td>
<td>0.211 sec</td>
<td>100%</td>
<td>100%</td>
</tr>
<tr>
<td>2. Show or delete previous results</td>
<td>3</td>
<td>4</td>
<td>3</td>
<td>0.617 sec</td>
<td>100%</td>
<td>100%</td>
</tr>
<tr>
<td>3. Adjust app settings</td>
<td>3</td>
<td>3</td>
<td>3</td>
<td>0.614 sec</td>
<td>100%</td>
<td>100%</td>
</tr>
<tr>
<td>4. SpeedTest support</td>
<td>5</td>
<td>6</td>
<td>3</td>
<td>0.540 sec</td>
<td>100%</td>
<td>100%</td>
</tr>
<tr>
<td>5. Privacy and terms</td>
<td>8</td>
<td>8</td>
<td>5</td>
<td>0.651 sec</td>
<td>100%</td>
<td>100%</td>
</tr>
<tr>
<td>6. Test the speed of the video</td>
<td>4</td>
<td>4</td>
<td>1</td>
<td>0.211 sec</td>
<td>100%</td>
<td>100%</td>
</tr>
<tr>
<td>7. Generate map data</td>
<td>2</td>
<td>2</td>
<td>1</td>
<td>0.211 sec</td>
<td>100%</td>
<td>100%</td>
</tr>
<tr>
<td><strong>Total</strong></td>
<td><strong>27</strong></td>
<td><strong>29</strong></td>
<td><strong>17</strong></td>
<td><strong>3.055 sec</strong></td>
<td><strong>100%</strong></td>
<td><strong>100%</strong></td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Functionality</th>
<th>Number of GUI Activities</th>
<th>Number of GUI Actions</th>
<th>Number of Test Cases</th>
<th>Time Spend to Generate Test Cases</th>
<th>% of Covered GUI Activities</th>
<th>% of Covered GUI Actions</th>
</tr>
</thead>
<tbody>
<tr>
<td>1. Search by book name or author name</td>
<td>2</td>
<td>2</td>
<td>1</td>
<td>0.214 sec</td>
<td>100%</td>
<td>100%</td>
</tr>
<tr>
<td>2. Search by book genre</td>
<td>4</td>
<td>9</td>
<td>2</td>
<td>0.376 sec</td>
<td>100%</td>
<td>100%</td>
</tr>
<tr>
<td>3. Browse recommended books</td>
<td>4</td>
<td>4</td>
<td>2</td>
<td>0.361 sec</td>
<td>100%</td>
<td>100%</td>
</tr>
<tr>
<td>4. Browse best-selling books</td>
<td>2</td>
<td>5</td>
<td>1</td>
<td>0.261 sec</td>
<td>100%</td>
<td>100%</td>
</tr>
<tr>
<td>5. Browse “The Books That Everyone Should Read At Least Once” list</td>
<td>1</td>
<td>2</td>
<td>1</td>
<td>0.261 sec</td>
<td>100%</td>
<td>100%</td>
</tr>
<tr>
<td>6. Browse the featured list of books</td>
<td>4</td>
<td>4</td>
<td>10</td>
<td>2.59 sec</td>
<td>100%</td>
<td>100%</td>
</tr>
<tr>
<td>7. Adjust app settings</td>
<td>3</td>
<td>4</td>
<td>5</td>
<td>0.698 sec</td>
<td>100%</td>
<td>100%</td>
</tr>
<tr>
<td>8. Edit favorite genres</td>
<td>2</td>
<td>3</td>
<td>1</td>
<td>0.259 sec</td>
<td>100%</td>
<td>100%</td>
</tr>
<tr>
<td>9. Enter a reading challenge</td>
<td>2</td>
<td>4</td>
<td>1</td>
<td>0.266 sec</td>
<td>100%</td>
<td>100%</td>
</tr>
<tr>
<td>10. View past challenges</td>
<td>2</td>
<td>4</td>
<td>1</td>
<td>0.277 sec</td>
<td>100%</td>
<td>100%</td>
</tr>
<tr>
<td>11. Update reading progress</td>
<td>2</td>
<td>3</td>
<td>1</td>
<td>0.269 sec</td>
<td>100%</td>
<td>100%</td>
</tr>
<tr>
<td>12. Show the best books this year</td>
<td>2</td>
<td>3</td>
<td>1</td>
<td>0.255 sec</td>
<td>100%</td>
<td>100%</td>
</tr>
<tr>
<td>13. Add kindle notes and highlights</td>
<td>3</td>
<td>3</td>
<td>1</td>
<td>0.270 sec</td>
<td>100%</td>
<td>100%</td>
</tr>
<tr>
<td><strong>Total</strong></td>
<td><strong>33</strong></td>
<td><strong>50</strong></td>
<td><strong>28</strong></td>
<td><strong>6.325 sec</strong></td>
<td><strong>100%</strong></td>
<td><strong>100%</strong></td>
</tr>
</tbody>
</table>
B. Experimental Results of the Proposed Performance Test Results Analysis Approach

This experiment focuses on executing test cases, then analyzing the test results to detect the performance critical point and heavy load of the AUT. The previously generated test cases were executed. Then, the test results were analyzed and interpreted. JMeter tool is used for test execution. The value of the Thread Group variable (i.e., number of simulated concurrent users) starts from 100 users. Then, the Thread Group value increments iteratively by 50, until it reaches 1000. The minimum and the maximum number of concurrent users are defined as a tester input to the proposed P-TaaS architecture. The minimum and the maximum number of concurrent users shall be defined according to the estimation of the owner of the AUT. In the experiments, multiple trials were made to choose the appropriate minimum number of concurrent users. It was noticed that the error rate was 0% and the response time is very small at workloads less than 100. However, the response time and error rate values began to upsurge starting from 100 concurrent users. Thus, the test results analysis experiments use 100 concurrent users as the minimum number. The maximum workload is defined in the experiments as the number of workloads where the AUT’s behavior (i.e., fluctuation and instability) could be observed. Thus, the maximum workload in the experiments was 1000 concurrent users, as shown in Tables III and IV.

Generally, there are dependent and independent variables in any experiment [38]. The goal of the experiment is to monitor the effect of changing the value of the independent variable on the dependent variable. In our experiments, the independent variable is the workload value (i.e., number of simulated concurrent users). The dependent variables are response time and error rate. The value of the H_Respone_Time is a dependent variable on the response time and error rate.

Table III presents the detailed measurements of test execution and result analysis for the first functionality of the GoodReads App (i.e., Search by book name or author name functionality). The first column in Table III includes the number of concurrent users’ access to the AUT. The second column includes the measured response time. The third column includes the calculated H_Respone_Time value for the corresponding workload and response time. The fourth column includes the measured error rate value of the AUT under a certain workload. Error rate value indicates the percent of requests submitted to the AUT with error. The last column includes the time spent on executing the test case within the corresponding workload. It is observed from Table III, that at the workload of 1000 concurrent users, response time increases sharply and H_Respone_Time has the highest value. During the test execution, it was noticed that the error rate became 52.7% at the workload of 1000 concurrent users, which exceeds the threshold error value. This implies that the AUT becomes unstable and may crash at any workload of more than 1000 concurrent users. Thus, it is interpreted that the performance critical turning point occurs at a workload of 1000 concurrent users, a workload of more than 1000 will be considered a heavy load. This leads to an exponential increase in the error rate. Besides, the AUT becomes unstable and may crash. Fig. 5 shows the response time measurement for a different number of concurrent users’ accesses. It is observed that when the number of concurrent users reaches 1000, the response time increases sharply and reaches its peak.

![Fig. 5. Response time measurement for several numbers of concurrent users’ accesses to the goodreads app.](image)

Similarly, Table IV presents the results for the first functionality of the SpeedTest App (i.e., Show upload/download speed). It is observed from Table IV, that at the workload of 850 concurrent users, the response time increases sharply and the H_Respone_Time has the highest value. During the test execution, it was noticed that the error rate became 56.12% at the workload of 850 concurrent users, which is the highest error rate that occurred during the experiment. Thus, it is interpreted that the performance critical

**TABLE III. EXPERIMENTAL RESULTS OF TEST EXECUTION AND ANALYSIS FOR GOODREADS APP**

<table>
<thead>
<tr>
<th>Workload</th>
<th>Response Time (sec)</th>
<th>H_Respone_Time</th>
<th>Error Rate (%)</th>
<th>Test Execution Time (sec)</th>
</tr>
</thead>
<tbody>
<tr>
<td>0</td>
<td>0</td>
<td>0</td>
<td>0</td>
<td>0</td>
</tr>
<tr>
<td>100</td>
<td>1.657</td>
<td>1599.1158</td>
<td>0</td>
<td>102</td>
</tr>
<tr>
<td>150</td>
<td>1.931</td>
<td>1866.810935</td>
<td>0.333</td>
<td>102</td>
</tr>
<tr>
<td>200</td>
<td>2.142</td>
<td>2075.136315</td>
<td>0.25</td>
<td>103</td>
</tr>
<tr>
<td>250</td>
<td>1.991</td>
<td>1940.209504</td>
<td>0.2</td>
<td>103</td>
</tr>
<tr>
<td>300</td>
<td>9.541</td>
<td>9177.330651</td>
<td>0.17</td>
<td>129</td>
</tr>
<tr>
<td>350</td>
<td>19.226</td>
<td>18482.24371</td>
<td>3.86</td>
<td>182</td>
</tr>
<tr>
<td>400</td>
<td>9.246</td>
<td>8899.692469</td>
<td>0.25</td>
<td>175</td>
</tr>
<tr>
<td>450</td>
<td>25.340</td>
<td>24358.72278</td>
<td>36.33</td>
<td>188</td>
</tr>
<tr>
<td>500</td>
<td>4.510</td>
<td>4379.667444</td>
<td>0.4</td>
<td>160</td>
</tr>
<tr>
<td>550</td>
<td>9.592</td>
<td>9243.143812</td>
<td>0.64</td>
<td>129</td>
</tr>
<tr>
<td>600</td>
<td>26.134</td>
<td>25125.38432</td>
<td>12</td>
<td>205</td>
</tr>
<tr>
<td>650</td>
<td>10.818</td>
<td>10426.99425</td>
<td>9.3</td>
<td>143</td>
</tr>
<tr>
<td>700</td>
<td>9.632</td>
<td>9296.884107</td>
<td>8.86</td>
<td>172</td>
</tr>
<tr>
<td>750</td>
<td>13.546</td>
<td>13050.01291</td>
<td>23.4</td>
<td>152</td>
</tr>
<tr>
<td>800</td>
<td>9.256</td>
<td>8950.281099</td>
<td>4.62</td>
<td>141</td>
</tr>
<tr>
<td>850</td>
<td>11.409</td>
<td>11014.08098</td>
<td>14.88</td>
<td>189</td>
</tr>
<tr>
<td>900</td>
<td>13.766</td>
<td>13275.02628</td>
<td>14.67</td>
<td>228</td>
</tr>
<tr>
<td>950</td>
<td>15.958</td>
<td>15379.26698</td>
<td>18.79</td>
<td>113</td>
</tr>
<tr>
<td>1000</td>
<td>78.602</td>
<td>71702.83435</td>
<td>52.7</td>
<td>610</td>
</tr>
</tbody>
</table>
turning point occurs at the workload of 850 concurrent users. A workload of more than 850 will be considered a heavy load. Fig. 6 shows the response time measurement for a different number of concurrent users’ accesses. It is observed that when the number of concurrent users increases, the response time increases sharply and reaches its peak.

### TABLE IV. EXPERIMENTAL RESULTS OF TEST EXECUTION AND ANALYSIS FOR SPEEDTEST APP

<table>
<thead>
<tr>
<th>Workload</th>
<th>Response Time (sec)</th>
<th>H_Respone_Time</th>
<th>Error Rate (%)</th>
<th>Test Execution Time (sec)</th>
</tr>
</thead>
<tbody>
<tr>
<td>0</td>
<td>0</td>
<td>0</td>
<td>0</td>
<td>0</td>
</tr>
<tr>
<td>100</td>
<td>663</td>
<td>647.3549</td>
<td>0</td>
<td>99</td>
</tr>
<tr>
<td>150</td>
<td>685</td>
<td>681.4788</td>
<td>0</td>
<td>100</td>
</tr>
<tr>
<td>200</td>
<td>717</td>
<td>728.3722</td>
<td>0</td>
<td>100</td>
</tr>
<tr>
<td>250</td>
<td>609</td>
<td>653.1596</td>
<td>0</td>
<td>100</td>
</tr>
<tr>
<td>300</td>
<td>723</td>
<td>776.9281</td>
<td>0</td>
<td>101</td>
</tr>
<tr>
<td>350</td>
<td>675</td>
<td>759.3621</td>
<td>0</td>
<td>100</td>
</tr>
<tr>
<td>400</td>
<td>706</td>
<td>811.3094</td>
<td>0</td>
<td>101</td>
</tr>
<tr>
<td>450</td>
<td>767</td>
<td>889.2544</td>
<td>0</td>
<td>101</td>
</tr>
<tr>
<td>500</td>
<td>1,015</td>
<td>1128.796</td>
<td>0</td>
<td>105</td>
</tr>
<tr>
<td>550</td>
<td>1,184</td>
<td>1300.434</td>
<td>0</td>
<td>108</td>
</tr>
<tr>
<td>600</td>
<td>2,646</td>
<td>2639.006</td>
<td>0</td>
<td>116</td>
</tr>
<tr>
<td>650</td>
<td>3,749</td>
<td>3680.732</td>
<td>0</td>
<td>116</td>
</tr>
<tr>
<td>700</td>
<td>1,106</td>
<td>589.3899</td>
<td>0</td>
<td>106</td>
</tr>
<tr>
<td>750</td>
<td>13,978</td>
<td>13453.63</td>
<td>23.9</td>
<td>236</td>
</tr>
<tr>
<td>800</td>
<td>11,976</td>
<td>11538.09</td>
<td>27.12</td>
<td>190</td>
</tr>
<tr>
<td>850</td>
<td>20,647</td>
<td>19859.62</td>
<td>56.12</td>
<td>748</td>
</tr>
<tr>
<td>900</td>
<td>46,544</td>
<td>44749.01</td>
<td>78.37</td>
<td>258</td>
</tr>
<tr>
<td>950</td>
<td>17,977</td>
<td>17301.74</td>
<td>67.3</td>
<td>469</td>
</tr>
<tr>
<td>1000</td>
<td>23,779</td>
<td>22873.01</td>
<td>52.4</td>
<td>623</td>
</tr>
</tbody>
</table>

![Response time measurement for several numbers of concurrent users’ accesses to the speedtest app.](image)

Fig. 6. Response time measurement for several numbers of concurrent users’ accesses to the speedtest app.

The experimental results presented in Tables III and IV, and Fig. 5 and Fig. 6 were closely examined. The observations revealed are as follows:

- There is a fluctuation in the response time, especially with workloads lower than the detected heavy load. The fluctuation means that the response time of the AUT is faster than the preceding run. In the experiments, the fluctuation appears more obviously in Fig. 5 than in Fig. 6. For instance, in Fig. 5 the response time was 9,541 seconds at 300 workloads, the response time was 19,226 seconds at 350 workloads, then the response time decreased to 9,246 seconds at 400 workloads. This fluctuation in the response time is expected and is not considered as a problem during the performance test execution. This fluctuation indicates that the AUT does not clean up its resources. The memory usage metric is used to confirm this. If the memory usage remains high after the test is completed, then this implies that the resources are not cleaned-up at the AUT’s web server. Thus, there are many other performance metrics (e.g., CPU usage, Memory usage, DB response time) that affect the responsivity of the AUT.

- The response time sometimes decreases under high workloads. However, the error rate values increase exponentially. This occurs despite the expectations of high response time value due to the heavy workload. For instance, in Table IV the response time was 46,544 seconds at 900 workloads and the error rate was 78.37%. However, the response time decreased to 17,977 seconds at 950 workloads and the measured error rate was 67.3%. The reason is that 900 workloads were defined as the heavy workload during the experiment where the AUT became unstable. Thus, the high error rate implies that a considerable percentage of the requests returned failed immediately and the average response time calculated by the JMeter decreased.

### C. Experimental Results of the Effect of Utilizing the Simulated Cloud-based Environment in the Proposed P-TaaS

An experiment was conducted to evaluate the effect of utilizing the simulated cloud-based environment in the proposed mobile app P-TaaS. This experiment measures the time spent executing the automatically generated test cases sequentially on one virtual machine. Then, it measures the time spent executing the automatically generated test cases simultaneously on multiple virtual nodes. The experimental results for sequential and simultaneous test execution for the GoodReads app and SpeedTest app test cases are shown in Table V. The sequential execution for GoodReads app test cases takes 25 hours and 52 minutes, but the simultaneous execution only takes 8 hours and 37 minutes. The sequential execution for SpeedTest app test cases takes 18 hours and 18 minutes, but the simultaneous execution only takes 6 hours and 6 minutes. From the results, applying the proposed mobile app P-TaaS shows a vast reduction of time in case of simultaneous test execution.

### TABLE V. EXPERIMENTAL RESULTS FOR SEQUENTIAL AND SIMULTANEOUS TEST EXECUTION FOR SPEEDTEST AND GOODREADS APPS TEST CASES

<table>
<thead>
<tr>
<th>Test Case Execution Approach</th>
<th>SpeedTest</th>
<th>GoodReads</th>
</tr>
</thead>
<tbody>
<tr>
<td>Total Time</td>
<td>18 hours and 18 minutes</td>
<td>25 hours and 52 minutes</td>
</tr>
<tr>
<td>Simultaneous Execution</td>
<td>6 hours and 6 minutes</td>
<td>8 hours and 37 minutes</td>
</tr>
</tbody>
</table>

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VI. DISCUSSION AND LIMITATIONS OF THE PROPOSED P-TaaS ARCHITECTURE

This section discusses the difference between the proposed P-TaaS architecture and other frameworks presented in literature and are mentioned in Section III. These comparisons show that the proposed P-TaaS architecture fulfill the contributions and handle challenges mentioned in Section I. Additionally, this section discusses the limitations of the proposed mobile app P-TaaS architecture.

The proposed P-TaaS architecture is compared with similar cloud-based performance testing frameworks discussed in literature as well as widely used tools in the market. The comparison includes the following criteria: automatic test case generation with high test coverage, simultaneous test execution on multiple virtual nodes in the cloud-based environment, automatic test result analysis, automatic test report generation that includes the performance metrics (i.e., response time and error rate) collected during the test execution at different workloads, and efficient scheduling and resource utilization. The comparison is shown in Table VI. The first five criteria are related to automating the whole performance testing process. The last criterion in Table VI is related to the efficiency of utilizing cloud-based resources. From the comparison, the proposed P-TaaS is considered a comprehensive framework that conducts the whole performance testing process automatically. Moreover, the proposed P-TaaS leverages the cloud-based environment efficiently.

<table>
<thead>
<tr>
<th></th>
<th></th>
<th></th>
<th></th>
<th></th>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td>The Proposed Mobile Apps P-TaaS</td>
<td>Yes</td>
<td>Yes</td>
<td>Yes</td>
<td>Yes</td>
<td>Yes</td>
</tr>
<tr>
<td>ATaaS Framework by Prathibhan et al. [15]</td>
<td>No</td>
<td>Yes</td>
<td>No</td>
<td>No</td>
<td>No</td>
</tr>
<tr>
<td>SOASTA CloudTest [16]</td>
<td>No</td>
<td>Yes</td>
<td>No</td>
<td>No</td>
<td>No</td>
</tr>
<tr>
<td>LoadStorm [17]</td>
<td>No</td>
<td>Yes</td>
<td>Yes</td>
<td>Yes</td>
<td>Yes</td>
</tr>
<tr>
<td>Xamarin Test Cloud [18]</td>
<td>No</td>
<td>Yes</td>
<td>No</td>
<td>Yes</td>
<td>No</td>
</tr>
</tbody>
</table>

TABLE VII. COMPARISON BETWEEN THE PROPOSED SCHEDULING APPROACH AND OTHER APPROACHES IN THE LITERATURE

<table>
<thead>
<tr>
<th>Approach</th>
<th>Uncertainty of Test Execution Time</th>
<th>Load Balance Between Resources</th>
<th>Computation Time</th>
</tr>
</thead>
<tbody>
<tr>
<td>The Proposed P-TaaS Schedule Approach</td>
<td>Exists</td>
<td>Exists</td>
<td>Simple and Low Computation Time</td>
</tr>
<tr>
<td>Fuzzy Sets Theory-Based Approach Proposed by Lampe et al. [21]</td>
<td>Not Exists</td>
<td>Exists</td>
<td>Long Computation Time</td>
</tr>
<tr>
<td>Metaheuristic Methodologies Proposed by Rudy et al. [22]</td>
<td>Exists</td>
<td>Exists</td>
<td>Complex and Long Computation Time</td>
</tr>
</tbody>
</table>

VII. CONCLUSION

The performance of mobile apps is significantly important. Performance testing assesses and guarantees the reliability and stability of mobile apps when exposed to different workloads of concurrent users’ accesses. The lack of performance testing may lead to degradation in mobile apps. Therefore, more attention from the industrial and academic communities is directed to performance testing as a service (P-TaaS). This paper introduced mobile app performance testing based on the TaaS architecture. It accomplishes the entire performance testing process automatically. The proposed P-TaaS adopts efficient approaches for test case generation, simultaneous test execution, test results analysis, and scheduling of test tasks. The experimental results on two different mobile apps prove the effectiveness of the proposed P-TaaS.

VIII. FUTURE WORK

In future work, the proposed mobile apps P-TaaS can be extended to include many directions such as: (i) allowing the performance testing of different types of mobile applications running on different platforms, (ii) including more types of testing as: security testing and regression testing, (iii) using a...
real cloud-based environment for simultaneous test execution on many VMs (e.g., Amazon EC2).

REFERENCES


[37] https://www.similarweb.com/website/goodreads.com/#overview

A Segmentation-based Token Identification for Recognition of Audio Mathematical Expression

SBTI Method for Audio Mathematical Expression Recognition

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Symbiosis Institute of Computer Studies and Research (SICSR), Symbiosis International (Deemed University), Pune, Maharashtra State, India

Abstract—In human-computer interaction, humans can interact with the computer with the help of text, audio, images, speech, etc. Interacting with the computer using speech, speech recognitions in particularly audio segmentation is a challenging task due to accent or way of pronouncing style. To input mathematical symbols, words, functions, and expressions with the help of a keyboard are tedious and time-consuming. Input this with the help of audio, speeds up the input process. In this paper, an SBTI (audio Segmentation Based Token Identification) algorithm is proposed for the recognition of words in an audio mathematical expression. 6 types of audio mathematical expressions are considered for recognition. The proposed algorithm segments the audio file into chunks and from each chunk temporal and spectral characteristics of audio signals are selected to extract the features. The model is trained using a neural network. The proposed algorithm shows a classification accuracy of 100% for the algebraic, quadratic, area, and differentiation expression, 99% for trigonometric expression, and 92% for summation expression.

Keywords—Audio segmentation; classification; feature extraction; neural network; speech recognition

I. INTRODUCTION

In human-computer interaction, various types of data are used such as text, image, audio, and video. Interacting with the computer with the help of audio data increases the speed of interaction. Now a day’s most devices accept audio data as input but input mathematical data with the help of audio is not available. Existing work reveals that few researchers worked on the recognition of audio mathematical expression. To input mathematical symbols, words, functions, and expressions with the help of a keyboard are tedious and time-consuming. In this paper, a novel methodology is designed for the recognition of audio mathematical expressions. Recognition of mathematical expression is a challenging task because it consists of digits, letters, symbols, functions, etc.

Segmenting audio data is an important step in speech processing as well as audio processing applications such as speech recognition [1]. Segmenting audio data is challenging and a need of the day for applications such as speech recognition. Audio segmentation quality [2] affects the performance of recognition of speech. Segmentation of audio data is the process of splitting the audio data into small segments. Continuous audio signals are fragmented into small segments. To segment a sentence into phonemes, words, and syllables, [3] continuous speech segmentation is used. Based on classification techniques, segmentation is categorized into two types [4], [6], [7] classification-dependent segmentation and classification-independent segmentation. In [5] audio is segmented into the voiced and unvoiced parts with the help of ZCR and energy. The purpose of audio segmentation is to find acoustics variations [7] in the audio signal. Audio is segmented into various components of speech such as voiced part, nonvoiced part, noise, silence [8], etc. Segmentation methods are also categorized into three types [6] model-based segmentation, metric-based segmentation, and energy-based segmentation. In energy-based segmentation, silence is detected by thresholding the energy signals. Statistical models are used to design the model-based segmentation [9]. In metric-based segmentation, a distance function is used to determine the boundaries. Speech segmentation is an important step in speech processing [10]. A phoneme or a syllable or a sub-word, is the basic phonetic unit [11] which is based on the language.

Segmentation can be categorized into [18] acoustic audio segmentation and phonetic audio segmentation. Speech segmentation is also categorized into [19] manual segmentation and automatic segmentation. In Manual segmentation, speech signal waveforms are examined and it is segmented but it is very time-consuming and not produced correct results whereas, in automatic segmentation, speech waveforms are automatically segmented into words. Preprocessing of a speech signal is most important in speech recognition. As speech signals are continuous, the first thing is to convert them into digital form with the help of an analog-to-digital converter. In the audio file when silence is present it has no importance because of the lack of information. Silence removal is significant in speech recognition. Silence in the audio file is removed with the help of energy and the threshold value. In an audio file, speech signals are present in the form of frames through which features are extracted by considering the window size and frame rate. Feature extraction is the process of converting an audio signal into a sequence of features called a feature vector. Feature vector consists of the information of audio signals having temporal and spectral characteristics. The selection of feature sets plays a key role to improve the performance of audio segmentation.

In the proposed work, a novel audio segmentation based token identification algorithm is proposed, features are extracted based on various parameters and accuracy for different types of expressions is reported.
II. LITERATURE REVIEW

There are various challenges in audio segmentation and speech recognition such as separating the audio data into regions, multiple classes will be used for different types of segmentation boundaries [1]. To segment audio data various features are considered such as BIC, Boundary confidence, Warping factor variance, and word length [1] where the MAP decoder framework is presented which is applicable for segment features. It has been observed that many researchers have used various audio segmentation techniques such as MFCC, ZCR, energy value are used for the segmentation of recognition of speech in noisy and noise-free environments. In [2], it has been done into syllable units. This can be developed using SVM classifier. In [3] author proposed a mathematical expression is more challenging, hence there is scope to work on it. Researchers reported that the syllable counting method can be used in the segmentation process to improve the result. In this paper, we have proposed a segmentation algorithm based on tokens for the recognition of audio mathematical expressions.

Authors have experimented to segment the speech into phonemes with the help of phonemes specialists, results in known as tedious, expensive and subjective [27].

An optimised parameters of automatic speech segmentation has been done into syllable units. This can be developed using time domain energy-based features and static threshold is used to detect the syllable boundary [28].

Speech Recognition technology is challenging because of sensitivity to the environment (background noise) or the weak representation of grammatical and semantic knowledge [29].

The proposed research study is unique because the dataset is generated as it is not available and samples are collected by varied age group. To segment long utterances is challenging. Here, Complex mathematical expressions are executed using the proposed approach, and recognition accuracy is reported. In this experiment the eight different categories of mathematical expressions are considered because of their different spatial and structural representation.
III. METHODOLOGY SBTI (SEGMENTATION-BASED TOKEN IDENTIFICATION)

Fig. 1 shows the methodology used for implementation of the proposed algorithm of segmentation.

![Methodology Diagram]

**A. Audio Mathematical Expression (AME) Dataset**

To perform the experiment we have selected 6 categories of mathematical expression which include algebraic expression, quadratic expression, area formulae, trigonometric expression, differentiation, and summation. To cover the samples from different age groups, people from the age group of 10 to 55 are used to create the dataset. All these expressions are recorded using the Audacity tool and taken in .wav. Table I shows the number of samples used for this study.

**B. Pre-processing**

In this step, the voiced part from the audio file is selected with the help of ZCR.

**C. Segmentation**

The proposed segmentation algorithm segments audio mathematical expression by considering the parameters such as the number of tokens, frame rate, and duration of audio mathematical expression in milliseconds. Fig. 3 shows the signal of $x^2 + y^2 = z^2$ audio mathematical expression in .wav format which is given as input to the proposed algorithm.

---

**Proposed Segmentation Algorithm 1: SBTI**

<table>
<thead>
<tr>
<th>Input: Audio Mathematical Expression (AME) in .wav format</th>
</tr>
</thead>
<tbody>
<tr>
<td>Output: Segments of AME</td>
</tr>
</tbody>
</table>

**Start**

- $D ← S/ Fr$ \hspace{1cm} // Duration of AME
- $Fr ←$ // Number of samples per second
- $T ←$ // Number of tokens in AME
- $Ch ← D/T$ \hspace{1cm} // Number of chunks
- $St ← 0$ //Starting pint
- $Lt ← 0$ //Ending point

**For** $I ← 1$ to $T$

- $Ch (I) ←$ Seg($St , Lt$)
- $St ← Ch + I$
- $Lt ← Lt + Ch$

**End loop**

**End**

D. **Feature Extraction**

From the audio mathematical expression, we have selected the features such as chroma_stft, rmse, spectral_centroid, spectral_bandwidth, roll-off, zero_crossing_rate, and 21 MFCC coefficients. Chromogram features and root mean square energy are features calculated in chroma_stft and RMSE. Root Mean Square Energy for each frame is calculated in RMSE.

Spectral centroid is a frequency-based feature that computes the “Location of mass”. Equation (1) is used to calculate spectral centroid.

$$f_c = \frac{\sum_{k=0}^{N-1} s(k)f(k)}{\sum_{k=0}^{N-1} s(k)}$$  \hspace{1cm} (1)

where $s(k)$ is the spectral magnitude at frequency bin $k$, $f(k)$ is the frequency at bin $k$.

A variation between higher and lower frequencies is calculated using spectral_bandwidth. Spectral_rolloff computes the shape of the signal. It shows the frequency at which high frequencies deteriorate to 0. ZCR is used to check the voiced and unvoiced parts of speech. It is the rate at which the signal changes from positive to negative or vice-versa. It is computed as

$$ZCR = \sum_{n=1}^{N-1} |S[x(n+1)] - S[x(n)]| / (2(N-1))$$  \hspace{1cm} (2)

In MFCC, the first audio signals are divided into short frames. Calculate the Fourier transform for each frame. Find the log of all filter bank energies. Compute the discrete cosine transform of each Mel log power. MFCC coefficients are calculated as

$$\text{Mel}(f) = 2595 \times \log_{10}(1 + \frac{f}{700})$$  \hspace{1cm} (3)

There are various applications of neural network present such as Machine translation, speech recognition, sentiment analysis, chatbots, named entity recognition, etc. To study such areas machine learning’s sequence model is very useful.
Sequence modeling consists of data in the form of sequences, a data structure.

It is used in supervised learning algorithms. Sequence models are categorized based on the input and output which may be in the form of scalar, audio, text, image, and video form.

The performance of the model is measured using various parameters such as accuracy, loss, F1-score, precision, recall, etc. The accuracy of the model depends on predicted and observed values. It is computed as

$$\text{Acc}(M) = \frac{\text{TrP} + \text{TrN}}{\text{TotP} + \text{TotN}}$$  \hspace{1cm} (4)

Where TrP and TrN are true positive and true negative respectively. TotP is the combination of true positive and false positive samples whereas TotN is the inclusion of false positive and false negative. The model’s performance is visualized using accuracy-loss graphs.

E. Training and Testing

To recognize the speech for audio mathematical expression, the proposed algorithm is used to segment an audio mathematical expression. Audio mathematical expression in .wav format is given as input as shown in Fig. 2. The frame rate and the number of samples present in each expression are computed. Based on the number of samples and frame rate duration of audio mathematical expression is find out. The number of tokens and duration of audio mathematical expressions are used to segment the audio mathematical expression.

Testing accuracy is also less for summation expression as compared to another category because of more number of labels. Table II shows testing accuracy for each category of audio mathematical expression. Fig. 3 shows the graph of training accuracy of summation and differentiation expression. It indicates that initially training accuracy is low but gradually it goes on increasing.

Table III shows the comparative result of the proposed model with the existing result for the recognition of words in continuous speech.

![Image](image_url)

**Fig. 2.** An Audio signal of the mathematical expression “$x^2 + y^2 = z^2$”.

Features such as chroma_stft, rmse, spectral centroid, spectral bandwidth, roll-off, zero_crossing_rate, and 21 MFCC coefficients are extracted from each segment. The parameters used to train and test the NN (Neural Network) model are RELu and Softmax activation function, sparse_categorical_crossentropy loss function, adam optimizer, 50 epochs, and 128 batch size. The total number of samples for each expression is split into Training 80% and testing set 20%.

IV. RESULTS AND DISCUSSION

Table I, represents the samples used for segmentations along with its total number of segments and number of labels used. Table II indicates that training accuracy for algebraic expression, quadratic expression, trigonometric expression, and area formulae whereas it is less for summation expression.

<table>
<thead>
<tr>
<th>Expression Type</th>
<th>No. of Samples</th>
<th>No. of Segments</th>
<th>No. of Labels</th>
</tr>
</thead>
<tbody>
<tr>
<td>Algebraic</td>
<td>160</td>
<td>1280</td>
<td>8</td>
</tr>
<tr>
<td>Quadratic</td>
<td>237</td>
<td>1896</td>
<td>6</td>
</tr>
<tr>
<td>Area</td>
<td>225</td>
<td>675</td>
<td>3</td>
</tr>
<tr>
<td>Trigonometric</td>
<td>102</td>
<td>918</td>
<td>7</td>
</tr>
<tr>
<td>Differentiation</td>
<td>100</td>
<td>600</td>
<td>6</td>
</tr>
<tr>
<td>Summation</td>
<td>100</td>
<td>1000</td>
<td>10</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Expression Type</th>
<th>No. of Samples</th>
<th>No. of Labels</th>
</tr>
</thead>
<tbody>
<tr>
<td>Algebraic</td>
<td>8.66</td>
<td></td>
</tr>
<tr>
<td>Quadratic</td>
<td>0.91</td>
<td></td>
</tr>
<tr>
<td>Area</td>
<td>0.91</td>
<td></td>
</tr>
<tr>
<td>Trigonometric</td>
<td>0.85</td>
<td></td>
</tr>
<tr>
<td>Differentiation</td>
<td>0.89</td>
<td></td>
</tr>
<tr>
<td>Summation</td>
<td>0.60</td>
<td></td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Ref. No.</th>
<th>Dataset</th>
<th>Language</th>
<th>Recognition Rate</th>
</tr>
</thead>
<tbody>
<tr>
<td>[11]</td>
<td>20 unique sentences</td>
<td>Kannada</td>
<td>87.76%</td>
</tr>
<tr>
<td>[12]</td>
<td>150 distinct Words</td>
<td>Punjabi</td>
<td>86.05%</td>
</tr>
<tr>
<td>[17]</td>
<td>25 samples of 0-9 digits, some operators</td>
<td>English</td>
<td>80%</td>
</tr>
<tr>
<td>[20]</td>
<td>74 Mathematical Symbols</td>
<td>French</td>
<td>50%</td>
</tr>
<tr>
<td>[21]</td>
<td>HAMEX</td>
<td>French</td>
<td>50%</td>
</tr>
<tr>
<td>Proposed Algo.</td>
<td>AME</td>
<td>English</td>
<td>85% to 91%</td>
</tr>
</tbody>
</table>
V. CONCLUSION

In Human-computer interaction, speech recognition plays a vital role, but developing systems for speech recognition is challenging. In this paper, a segmentation-based token identification algorithm is proposed to segment an audio mathematical expression and classify it into symbols, letters, digits, Greek letters, and operators. Chroma_stft, Rmse, Spectral_centroid, Spectral_bandwidth, Rolloff, Zero_crossing_rate, and 21 MFCC coefficients were used to extract the features. Results indicated that the proposed algorithm works accurately for the segmentation of audio mathematical expressions. This study will be useful to speed up the entry of mathematics in documents for normal people as well as for blind persons. It will help the researchers to work on the recognition of speech for expressions used in mathematics.

![Graph](image_url)

Fig. 3. Training accuracy of summation and differentiation.

In the future, we would like to expand the proposed segmentation algorithm for audio mathematical expressions having a large number of tokens.

ACKNOWLEDGMENT

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REFERENCES


Evaluating Machine Learning Models for Predicting Graduation Timelines in Moroccan Universities

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Informatics and Applications Laboratory (IA), FSJES, Moulay Ismail University, Meknes, Morocco⁴

Abstract—The escalating student numbers in Moroccan universities have intensified the complexities of managing on-time graduation. In this context, Machine learning methodologies were utilized to analyze the patterns and predict on-time graduation rates in a comprehensive manner. Our dataset comprised information from 5236 bachelor students who graduated in the years 2020 and 2021 from the Faculty of Law, Economic, and Social Sciences at Moulay Ismail University. The dataset incorporated a diverse range of student attributes including age, marital status, gender, nationality, socio-economic category of parents, profession, disability status, province of residence, high school diploma attainment, and academic honors, all contributing to a comprehensive understanding of the factors influencing graduation outcomes. Implementation and evaluation of the performance of five different machine learning models: Support Vector Machines, Decision Tree, Naïve Bayes, Logistic Regression, and Random Forest, were carried out. These models were assessed based on their classification reports, confusion matrices, and Receiver Operating Characteristic (ROC) curves. From the findings, the Random Forest model emerged as the most accurate in predicting on-time graduation, showcasing the highest accuracy and ROC AUC score. Despite these promising results, it is believed that performance enhancements can be achieved through further tuning and preprocessing of the dataset. Insights from this study could enable Moroccan universities, among others, to better comprehend the factors influencing on-time graduation and implement appropriate measures to improve academic outcomes.

Keywords—Machine learning; logistic regression; classification reports; on time graduation; Moroccan universities

I. INTRODUCTION

The integration of technological advancements within higher education has stimulated a shift towards data-driven strategies to manage burgeoning student enrollments and optimize institutional systems. Particularly, on-time graduation, a significant performance metric, is becoming increasingly challenging to predict and manage [1]. This predicament isn’t confined to a single region, as institutions worldwide are contending with it. The case of Morocco, with its unique socio-economic contexts, is even more compelling [2].

This research intends to tackle this critical issue by applying machine learning methodologies to forecast on-time graduation rates at the Faculty of Law, Economic, and Social Sciences at Moulay Ismail University. A comprehensive dataset of 5236 bachelor students who graduated in 2020 and 2021 was utilized. The dataset includes numerous student characteristics such as age, marital status, gender, nationality, socio-economic category of parents, profession, disability status, province of residence, high school diploma attainment, and academic honors. It is posited that the detailed analysis of these variables could reveal valuable insights into the determinants of on-time graduation [3].

To identify the most effective method for predicting on-time graduation rates, a comparative analysis of five machine learning models - Support Vector Machines (SVM), Decision Tree (DT), Naïve Bayes (NB), Logistic Regression (LR), and Random Forest (RF) - was conducted. Each model’s performance was evaluated based on several statistical measures, including classification reports, confusion matrices, and Receiver Operating Characteristic (ROC) curves.

The implications of this research go beyond academic discourse. The findings could provide actionable insights for higher education institutions, particularly in Morocco, facilitating the formulation of effective data-driven strategies for improving on-time graduation management.

The paper is organized as follows: Post the introduction, the methodology, including data collection and analysis procedures, is detailed. Subsequent sections present a comprehensive discussion of the results, interpreting and comparing the performance of the different machine learning models. Potential implications of the findings are then outlined, offering a practical perspective. Finally, the paper concludes with a summary of the research findings and suggests directions for future research.

II. RELATED WORK

Several studies have addressed the question of predicting student success and on-time graduation using machine learning techniques. These studies, while varying in scope and methodology, provide valuable insights into the potential of machine learning in education.

For instance, Marbouti [4] conducted a study using machine learning to predict the success of first-year engineering students based on high school academic performance. They used logistic regression and decision tree models, highlighting the significant role of high school mathematics grades in predicting success.

Similarly, Delen [5] used decision tree, neural network, and logistic regression models to predict students’ graduation status based on demographic and academic data. They found that the
neural network model performed best in predicting student graduation status.

In the context of Moroccan higher education, however, the application of machine learning for predicting on-time graduation remains relatively unexplored. This study contributes to filling this research gap by applying machine learning techniques to a dataset from the Faculty of Law, Economic, and Social Sciences at Moulay Ismail University.

Notably, this work extends beyond the previous studies by comparing the performance of five different machine learning models: Support Vector Machines, Decision Tree, Naive Bayes, Logistic Regression, and Random Forest, in predicting on-time graduation. Moreover, a diverse range of student attributes is incorporated, aiming to create a more comprehensive prediction model.

Through this approach, the aim is to further the understanding of the factors influencing on-time graduation and contribute to the development of more effective strategies for academic success in Moroccan universities.

III. MATERIALS AND METHODS

This section outlines the data collection process, the variables incorporated in the study, the preprocessing steps undertaken, and the machine learning models employed for analysis.

A. Data Source

The dataset was derived from the student records of the Faculty of Law, Economic, and Social Sciences at Moulay Ismail University, encompassing 5236 bachelor students who graduated in the years 2020 and 2021. The dataset was collected and anonymized in strict compliance with data privacy regulations.

B. Dataset Features

The dataset incorporated a variety of student characteristics (Fig. 1) such as age, marital status, gender, nationality, socio-economic category of parents, profession, disability status, province of residence, high school diploma attainment, and academic honors. The target variable was 'Graduate on Time,' a binary variable indicating whether the student graduated within the standard duration of the program. (Table I) summarizes the main variables of this study.

<table>
<thead>
<tr>
<th>Variable</th>
<th>Definition</th>
</tr>
</thead>
<tbody>
<tr>
<td>Age</td>
<td>Age of the student in years</td>
</tr>
<tr>
<td>Marital_status</td>
<td>Marital status of the student</td>
</tr>
<tr>
<td>Gender</td>
<td>Gender of the student</td>
</tr>
<tr>
<td>Nationality</td>
<td>Nationality of the student</td>
</tr>
<tr>
<td>Parents_Categ_sosci</td>
<td>Socio-economic category of the student's parents</td>
</tr>
<tr>
<td>Profession</td>
<td>Profession of the student</td>
</tr>
<tr>
<td>Disability</td>
<td>Indicates whether the student has a disability</td>
</tr>
<tr>
<td>Province</td>
<td>Province of the student's residence</td>
</tr>
<tr>
<td>High_School_Diploma</td>
<td>Indicates the type of high school diploma the student has.</td>
</tr>
<tr>
<td>Academic_honor</td>
<td>Academic honors achieved by the student</td>
</tr>
<tr>
<td>Graduate_on_time</td>
<td>Indicates whether the student graduated on time</td>
</tr>
</tbody>
</table>

C. Label Encoding

The dataset under consideration encompasses a range of features that capture the student's demographic and academic characteristics. The 'Age' feature represents the age of the student in years. 'Marital_status' indicates the student's marital status, represented by numeric values where 1 signifies being Single, 2 stands for Married, 3 denotes Divorced, and 4 implies a Widower. 'Gender' signifies the student's gender, with 0 indicating Female and 1 representing Male. 'Nationality', 'Parents_Categ_sosci', 'Profession', and 'Province' are represented by IDs corresponding to different nationalities, socio-economic categories of parents, professions, and provinces respectively. 'Disability' is a binary indicator that highlights whether a student has a disability, where 0 denotes No and 1 stands for Yes. 'High_School_Diploma' points to the type of high school diploma the student possesses, represented by different IDs for each type of diploma. Academic Honor delineates the academic honors a student has achieved, ranging from 1 (Passing), 2 (Good), 3 (Very Good), to 4 (Outstanding). Lastly, 'Graduate_on_time' is a binary variable that indicates whether the student graduated within the standard duration of the program, represented by 0 (No) and 1 (Yes). These features collectively provide a comprehensive profile of the students' demographic and academic landscape (Table II, Fig. 2).

D. Correlation Features

The features used in this study have differing levels of correlation with the target variable, 'Graduate on Time'. The correlation values signify the strength and direction of the relationship between each feature and the target [6]. These values were computed and visualized through a correlation matrix. A positive correlation indicates that as the feature value increases, the likelihood of on-time graduation also increases, and vice versa. Conversely, a negative correlation means that as the feature value increases, the likelihood of on-time graduation decreases.
TABLE II. METHOD FOR ENCODING VARIABLES

<table>
<thead>
<tr>
<th>Variable</th>
<th>Definition</th>
<th>Possible Values</th>
</tr>
</thead>
<tbody>
<tr>
<td>Age</td>
<td>Age of the student in years</td>
<td>Numeric values</td>
</tr>
<tr>
<td>Marital_status</td>
<td>Marital status of the student</td>
<td>1(Single), 2(Married), 3(Divorced), 4(Widower)</td>
</tr>
<tr>
<td>Gender</td>
<td>Gender of the student</td>
<td>0(Female), 1(Male)</td>
</tr>
<tr>
<td>Nationality</td>
<td>Nationality of the student</td>
<td>IDs corresponding to different nationalities</td>
</tr>
<tr>
<td>Parents_Categ_socio</td>
<td>Socio-economic category of the student’s parents</td>
<td>IDs corresponding to different socio-economic categories</td>
</tr>
<tr>
<td>Profession</td>
<td>Profession of the student</td>
<td>0(No), 1(Yes)</td>
</tr>
<tr>
<td>Disability</td>
<td>Indicates whether the student has a disability</td>
<td>0(No), 1(Yes)</td>
</tr>
<tr>
<td>Province</td>
<td>Province of the student’s residence</td>
<td>IDs corresponding to different provinces</td>
</tr>
<tr>
<td>High_School_Diploma</td>
<td>Indicates the type of high school diploma the student has.</td>
<td>IDs corresponding to different high school diploma</td>
</tr>
<tr>
<td>Academic_honor</td>
<td>Academic honors achieved by the student</td>
<td>1(Passing), 2(Good), 3(Very Good), 4(Outstanding)</td>
</tr>
<tr>
<td>Graduate_on_time</td>
<td>Indicates whether the student graduated on time</td>
<td>0(No), 1(Yes)</td>
</tr>
</tbody>
</table>

Fig. 2. Dataframe after encoding.

In the dataset, the 'Academic Honor', 'High School Diploma', and 'Province' are the features that show the highest correlation with on-time graduation as depicted in the correlation matrix (Fig. 3). The 'Academic Honor' feature, representing the academic performance of students, shows a positive correlation, suggesting that students with higher academic honors are more likely to graduate on time. Similarly, the 'High School Diploma' and 'Province' features also have a positive correlation with on-time graduation, indicating that the type of high school diploma and the province of residence can have an influence on graduation times (Fig. 4).

These highly correlated features are particularly beneficial in predictive modeling, as they provide significant insight into the factors that influence on-time graduation. By focusing on these variables in the machine learning models, it becomes possible to make more accurate predictions and gain a deeper understanding of the factors contributing to graduation times.

![Heat map for checking correlated columns for graduate on time.](image)

**E. Modeling**

In this study, the modeling procedure takes place after the data preprocessing stage. This procedure entails training machine learning algorithms to predict whether a student will graduate on time based on their academic and demographic characteristics. Several well-regarded machine learning techniques were employed, including Support Vector Machine (SVM), Naive Bayes (NB), Decision Tree (DT), Random Forest (RF), and Logistic Regression (LR). These classification models are popular and efficient for dealing with such a binary classification task. The models were trained based on attributes such as Academic Honor, High School Diploma, and Province, aiming to classify students into two categories: those who are likely to graduate on time and those who are not. The Python scikit-learn library was used for data analysis and model implementation. The models were evaluated using a split-test method, partitioning the original dataset into a training set (80%) to train the model, and a test set (20%) to evaluate it. This technique is commonly used in machine learning to assess the effectiveness and efficiency of predictive models. By comparing the performance of the different models, the aim is to identify the one that provides the most accurate predictions for on-time graduation among students at the Faculty of Law, Economic, and Social Sciences at Moulay Ismail University. The entire procedure of the experiment is depicted in (Fig. 5).
1) Machine learning algorithms: In this study, we utilized five different machine learning algorithms, each with its strengths and applicable use cases. These include:

a) Support Vector Machine (SVM): SVM is a widely used classification algorithm that finds the hyperplane in an N-dimensional space that distinctly classifies the data points. It is especially useful in high dimensional spaces and situations where the number of dimensions exceeds the number of samples [7].

b) Naive Bayes (NB): The Naive Bayes classifier is a simple and efficient machine learning algorithm often used in text classification, spam filtering, recommendation systems, etc. It is based on applying Bayes' theorem with the "naive" assumption of conditional independence between every pair of a feature [8].

c) Decision Tree (DT): Decision Trees are a type of flowchart-like structure where each internal node represents a feature (or attribute), each branch represents a decision rule, and each leaf node represents an outcome. They are widely used due to their interpretability and simplicity [9].

d) Logistic Regression (LR): Logistic regression is a type of regression analysis used for predicting the probability of a binary outcome. It’s a statistical model that uses a logistic function to model a binary dependent variable [10].

e) Random Forest (RF): Random Forest is an ensemble learning method that operates by constructing multiple decision trees during training and outputting the majority vote of individual trees for classification problems or average prediction for regression problems. It’s a powerful algorithm known for its robustness and simplicity [11].

F. Performance Indicators

In this study, the performance of the various machine learning models was evaluated using several widely recognized performance indicators:

1) Confusion matrix: The Confusion Matrix is another key performance indicator used in this study. It is a specific table layout that visualizes the performance of an algorithm, typically a supervised learning one. The matrix contrasts the actual and predicted classifications of the instances in a dataset to measure the quality of the output of the classifier. Each row of the matrix represents the instances in an actual class while each column represents the instances in a predicted class (Fig. 6).

In general, the confusion matrix provides four types of classification results with respect to a classification target k [12].

- True positive (TP): correct prediction of the positive class \( c_{k,k} \)
- True negative (TN): correct prediction of the negative class \( \sum_{i \in N \setminus k} c_{ij} \)
- False positive (FP): incorrect prediction of the positive class \( \sum_{i \in N \setminus k} c_{iR} \)
- False negative (FN): incorrect prediction of the negative class \( \sum_{i \in N \setminus k} c_{iL} \)

The Confusion Matrix allows us to compute various other classification metrics, including precision, recall, F1-score, and support. By providing a more detailed view of how the classification model is performing, the Confusion Matrix plays a crucial role in understanding the behavior of the model beyond simple accuracy [13].

2) Classification report: A classification report offers a comprehensive synopsis of how well a classification model has performed. It consolidates key performance metrics such as accuracy, precision, recall, F1-score, and support.

a) Accuracy: Defined as the ratio of correctly predicted observations to total observations, accuracy is the most straightforward performance measure [14]. The accuracy of the model is defined as:

\[
\text{Overall Accuracy} = \frac{\sum \text{TP}_{\text{class}}}{\sum \text{TP}_{\text{class}} + \sum \text{FP}_{\text{class}}} \ #
\]  

(1)

b) Precision: Precision, calculated as the ratio of correctly predicted positive observations to total predicted positive observations, indicates the model’s ability to correctly identify only the relevant instances [15]. It’s defined as:

\[
\text{Precision}_{\text{class}} = \frac{\text{TP}_{\text{class}}}{\text{TP}_{\text{class}} + \text{FP}_{\text{class}}} \ #
\]  

(2)

c) Recall (Sensitivity): Also known as sensitivity, recall measures the model’s ability to identify all relevant instances, defined as the ratio of correctly predicted positive observations to all actual positives [16], it’s defined by equation (3).

\[
\text{Recall}_{\text{class}} = \frac{\text{TP}_{\text{class}}}{\text{TP}_{\text{class}} + \text{FN}_{\text{class}}} \ #
\]  

(3)
d) **F1-Score**: The F1 score combines precision and recall into a single metric by taking their harmonic mean, effectively balancing the trade-off between the two measures [17]. It’s defined by equation (4).

\[
F1 - Score = \frac{2 \times TP_{class}}{2 \times TP_{class} + FN_{class} + FP_{class}}
\]  

(4)

e) **Area Under Curve**: The Receiver Operating Characteristic (ROC) curve, plotting the true positive rate against the false positive rate, indicates the model’s discriminative power. The Area Under the Curve (AUC) offers a single measure summarizing the overall quality of the classifier [18].

**G. Training and Validation**

In this study, the modeling procedure commenced with partitioning the dataset into a training and validation set. As per the widely accepted practices in machine learning research, 80% of the total data was allocated for training the algorithms, while the remaining 20% was set aside for validation [19]. The objective was to ensure a realistic estimate of the models’ performance on unseen data, providing a valuable measure of their generalizability.

The training phase of this research implemented five distinct machine learning models: Support Vector Machine (SVM), Naive Bayes (NB), Decision Tree (DT), Random Forest (RF), and Logistic Regression (LR). These models were selected due to their widespread adoption in predictive modeling tasks of a similar nature and their ability to effectively manage binary classification problems [20].

Python’s scikit-learn library was utilized for the execution of these models [21]. The training of each model was conducted using the ‘fit’ function, while model parameters were optimized through the GridSearchCV function, an exhaustive search over a specified range of parameter values [22].

After the training, the models underwent validation using the validation set. Various performance metrics were employed to evaluate model performance, including Accuracy, Precision, Recall, and the F1-score. All of these metrics were featured in the Classification Report [15]. Additionally, the Receiver Operating Characteristic (ROC) curve was graphed for a visual assessment of the model’s performance [23].

For this study, a 10-fold cross-validation technique was also incorporated during the training phase [19]. This technique subdivides the training set into 10 subsets, and the model is trained 10 times. In each training iteration, nine subsets are used for training, and one is used for validation. The ultimate model performance is computed as the average performance of the ten models. This method helps generate a more dependable performance estimate and mitigates the risk of overfitting [24].

**IV. RESULTS AND DISCUSSION**

**A. Analysis of Results**

The performance of each model is summarized by highlighting key metrics and visual representations from the classification report and confusion matrices.

The SVM model achieved an overall accuracy of 69%. However, it demonstrated high recall (97%) for class 0 but had a low recall (11%) for class 1. This indicates that while the SVM model was able to identify the majority of class 0 instances correctly, it struggled to correctly classify instances from class 1 (Table III, Fig. 7).

**TABLE III. METRIC REPORT FOR SVM**

<table>
<thead>
<tr>
<th>Metric</th>
<th>Class 0</th>
<th>Class 1</th>
</tr>
</thead>
<tbody>
<tr>
<td>Precision</td>
<td>0.69</td>
<td>0.64</td>
</tr>
<tr>
<td>Recall</td>
<td>0.97</td>
<td>0.11</td>
</tr>
<tr>
<td>F1-Score</td>
<td>0.81</td>
<td>0.18</td>
</tr>
<tr>
<td>Support</td>
<td>705</td>
<td>342</td>
</tr>
</tbody>
</table>

![Confusion Matrix for SVM](image)

**Fig. 7.** Confusion matrix for SVM

The Decision Tree model performed better with an accuracy of 74%. It demonstrated a better balance in classifying both classes with a recall of 82% and 57% for class 0 and class 1 respectively (Table IV, Fig. 8).

**TABLE IV. METRIC REPORT FOR DECISION TREE**

<table>
<thead>
<tr>
<th>Metric</th>
<th>Class 0</th>
<th>Class 1</th>
</tr>
</thead>
<tbody>
<tr>
<td>Precision</td>
<td>0.80</td>
<td>0.60</td>
</tr>
<tr>
<td>Recall</td>
<td>0.82</td>
<td>0.57</td>
</tr>
<tr>
<td>F1-Score</td>
<td>0.81</td>
<td>0.59</td>
</tr>
<tr>
<td>Support</td>
<td>705</td>
<td>342</td>
</tr>
</tbody>
</table>

![Confusion Matrix for Decision Tree](image)

**Fig. 8.** Confusion matrix for decision tree.

The Naive Bayes model achieved an accuracy of 70%. While it showed good recall for class 0 (85%), it faced difficulties in classifying class 1 instances correctly, similar to the SVM model (Table V, Fig. 9).
TABLE V. METRIC REPORT FOR NAIVE BAYES

<table>
<thead>
<tr>
<th>Metric</th>
<th>Class 0</th>
<th>Class 1</th>
</tr>
</thead>
<tbody>
<tr>
<td>Precision</td>
<td>0.74</td>
<td>0.55</td>
</tr>
<tr>
<td>Recall</td>
<td>0.85</td>
<td>0.38</td>
</tr>
<tr>
<td>F1-Score</td>
<td>0.79</td>
<td>0.45</td>
</tr>
<tr>
<td>Support</td>
<td>705</td>
<td>342</td>
</tr>
</tbody>
</table>

Fig. 9. Confusion matrix for naive bayes.

The Logistic Regression model performed with an accuracy of 75%. It showed a fairly balanced performance for both classes, with a recall of 89% for class 0 and 47% for class 1 (Table VI, Fig. 10).

TABLE VI. METRIC REPORT FOR LOGISTIC REGRESSION

<table>
<thead>
<tr>
<th>Metric</th>
<th>Class 0</th>
<th>Class 1</th>
</tr>
</thead>
<tbody>
<tr>
<td>Precision</td>
<td>0.78</td>
<td>0.67</td>
</tr>
<tr>
<td>Recall</td>
<td>0.89</td>
<td>0.47</td>
</tr>
<tr>
<td>F1-Score</td>
<td>0.83</td>
<td>0.55</td>
</tr>
<tr>
<td>Support</td>
<td>705</td>
<td>342</td>
</tr>
</tbody>
</table>

Fig. 10. Confusion matrix for logistic regression.

Random Forest was the top-performing model with an accuracy of 77%. It also demonstrated the most balanced performance with a recall of 85% and 62% for class 0 and class 1, respectively (Table VII, Fig. 11).

TABLE VII. METRIC REPORT FOR RANDOM FOREST

<table>
<thead>
<tr>
<th>Metric</th>
<th>Class 0</th>
<th>Class 1</th>
</tr>
</thead>
<tbody>
<tr>
<td>Precision</td>
<td>0.82</td>
<td>0.66</td>
</tr>
<tr>
<td>Recall</td>
<td>0.84</td>
<td>0.62</td>
</tr>
<tr>
<td>F1-Score</td>
<td>0.83</td>
<td>0.64</td>
</tr>
<tr>
<td>Support</td>
<td>705</td>
<td>342</td>
</tr>
</tbody>
</table>

Fig. 11. Confusion matrix for random forest.

The Receiver Operating Characteristic (ROC) curve, in conjunction with the Area Under the Curve (AUC), offer insightful metrics for assessing the classification performance of a predictive model. In the context of this study, the ROC AUC scores reveal that the Random Forest model surpasses its counterparts, recording the highest score of 0.82. This indicates a superior capacity of the Random Forest model in distinguishing between students likely to graduate on time and those who aren’t, across various thresholds.

Following closely, the Logistic Regression model achieved the second-highest ROC AUC score of 0.75. This suggests a commendable proficiency of this model in accurately classifying the students. The Decision Tree model was also noteworthy, with a score of 0.74. The SVM and Naive Bayes models demonstrated similar performance levels, with ROC AUC scores of 0.73 each. Although these scores are lower compared to the Random Forest and Logistic Regression models, they still denote reasonable classification capabilities (Fig. 12).

Fig. 12. ROC curve for SVM, DT, NB, LR, RF.

B. Discussion

In this study, the relationship between a variety of features and the ability to graduate on time was investigated. After a careful data transformation and correlation analysis, three features, 'Academic Honor', 'High School Diploma', and 'Province' were selected based on their strong correlation with the target variable.
Several popular machine learning models were used to construct and validate predictive models. These models included Support Vector Machine (SVM), Naive Bayes (NB), Decision Tree (DT), Random Forest (RF), and Logistic Regression (LR). Each of these models was trained using the train-test split method, and their performance was evaluated using a suite of metrics including accuracy, precision, recall, and the F1 score. Also, the ROC-AUC score was calculated to measure the performance of the models under different classification thresholds.

The results clearly demonstrate that the Random Forest model was superior to all other models, achieving top marks in accuracy, precision, recall, and F1 score for predicting timely graduation of students. In addition, it accomplished an ROC-AUC score of 0.82, a clear indicator of its excellent performance. Although the Decision Tree and Logistic Regression models exhibited commendable performance, they were unable to match the outstanding performance of the Random Forest model.

Future work in this area could include the incorporation of more features, utilization of diverse feature selection techniques, and experimentation with alternative machine learning models. Furthermore, additional data should be considered. Ultimately, the goal is to construct a reliable predictive model capable of accurately identifying students at risk of not graduating on time, thereby enabling early interventions to support these students in achieving success.

V. CONCLUSION

The application of various machine learning models was showcased in this study to predict students’ ability to graduate on time. ‘Academic Honor’, ‘High School Diploma’, and ‘Province’ were used as predictors. The central goal was to identify the model that most accurately could assist educational institutions in pinpointing at-risk students and implementing interventions in a timely manner.

Among all the models tested, the Random Forest model stood out as the most effective, achieving the highest precision, recall, F1-score, and ROC-AUC score. The model’s ability to manage high-dimensional spaces and generate an internal unbiased estimate of the generalization error distinguished it from the other tested models. Therefore, this study underscores the use of machine learning, and particularly ensemble methods like Random Forest, as potent tools in the educational sector.

As a part of future work, further fine-tuning of the Random Forest model is suggested, along with the exploration of other machine learning models and additional student features. A potential avenue for future research may also include the implementation of this model in other educational contexts, allowing for a more comprehensive understanding of its applicability and robustness.

In conclusion, this study accentuates the potential of machine learning in education to predict student outcomes and enable proactive measures. With the ongoing integration of technology into education, the importance of such predictive tools is projected to rise. Such tools equip educational institutions with better understanding of student needs, enabling them to tailor their support services more effectively and, ultimately, assist more students in achieving their educational goals.

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Efficient and Accurate Beach Litter Detection Method Based on QSB-YOLO

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Abstract—Because of the potential threats it presents to marine ecosystems and human health, beach litter is becoming a major global environmental issue. The traditional manual sampling survey of beach litter is poor in real-time, poor in effect, and limited in the detection area, so it is extremely difficult to quickly clean up and recycle beach litter. Deep learning technology is quickly advancing, opening up a new method for monitoring beach litter. A QSB-YOLO beach litter detection approach based on the improved YOLOv7 is proposed for the problem of missed and false detection in beach litter detection. First, YOLOv7 is combined with the quantization-friendly Quantization-Aware RepVGG (QARepVGG) to reduce the model's parameters while maintaining its performance advantage. Secondly, A Simple, Parameter-Free Attention Module (SimAM) is used in YOLOv7 to enhance the feature extraction capacity of the network for the image region of interest. Finally, improving the original neck by combining the concept of the Bidirectional Feature Pyramid Network (BiFPN) allows the network to better learn features of various sizes. The test results on the self-built dataset demonstrate that: (1) QSB-YOLO has a good detection effect for six types of beach litter; (2) QSB-YOLO has a 5.8% higher mAP compared to YOLOv7, with a 43% faster detection speed, and QSB-YOLO has the highest detection accuracy for styrofoam, plastic products, and paper products; (3) QSB-YOLO has the greatest detection accuracy and detection efficiency when comparing the detection effects in various models. The results of the experiments demonstrate that the suggested model satisfies the need for beach litter identification in real-time.

Keywords—Beach litter detection; QSB-YOLO; YOLOv7; Quantization-Aware RepVGG; a simple, parameter-free attention module; bidirectional feature pyramid network

I. INTRODUCTION

Environmental pollution issues have gotten progressively worse in recent years, and beach litter pollution is unquestionably one of the planet's biggest environmental issues. Beach litter is characterized by wide distribution, persistence, and cumulative pollution, with two main sources, land-based and sea-based, including metal products, plastic products, wood, and paper products, etc [1]. The majority of beach litter is made up of materials that break down slowly, and because of poor waste management, it spreads out uncontrollably into the environment and interferes with marine traffic, damages ships, pollutes the nearshore area, degrades the environment, and results in many accidental injuries and deaths of marine life, etc [2]. Additionally, pollutants that persist and build up in beach litter can have an impact on people through the biological chain [2]. Today, China places a high value on reducing beach litter pollution, has created pertinent laws and regulations, and is actively doing research to improve the quality and cleanliness of beaches [3].

Many researchers have conducted a series of studies on the monitoring, sorting, and recycling of coastal litter. For example, Merlino et al. [4] evaluated the reliability of non-expert citizen scientist operators (CSO) to manually tag and classify marine litter from aerial photographs taken by drones. According to the study, CSO can support drone-based marine litter surveys with the right training programs and the provision of user-friendly guidance software, but the study is always labor-intensive [4]. Due to the labor-intensive nature of artificial beach litter sorting and cleanup, automated beach litter detection is a good solution to the pollution problem. The traditional method of beach litter monitoring is to monitor the amount of litter on the beach, which does not specifically localize or identify the beach litter [58]. The beach litter detection algorithm based on deep learning can simultaneously complete the classification and location of beach litter and feed it back to relevant staff, which can not only improve the detection accuracy of recyclable or non-degradable beach litter but also save a significant amount of manpower and material resources needed for beach litter cleaning and reduce environmental pollution to the greatest extent [910]. Deep learning-based beach litter detection is therefore very important.

In this research, we examined a few deep learning approaches that can be used to detect beach litter. However, the majority of deep learning-based object detection algorithms are designed for object detection in natural situations and do not completely apply to such unique scenarios as beach litter detection. This is due to the fact that the universal target detection model has a large number of parameters and a sluggish detection speed, making it unable to fulfill the speed requirements of actual applications and ineffective in dealing with issues like the mutual occlusion of targets and complicated beach litter backgrounds. Beach litter detection also be quite challenging due to the significant size differences between the targets of the litter.

In order to address the aforementioned issues, we suggest the QSB-YOLO beach litter detection method, which reduces model parameters to meet real-time requirements in practical applications, improves feature extraction capabilities of the
model to address the problem of missed and false detection in complex background, and enhances feature fusion network capabilities to enhance network's ability to acquire multi-scale features. The approach also offers improved detection speed and accuracy. The following are the primary contributions of this paper:

- Combining YOLOv7 with the quantization-friendly reparameterization architecture Quantization-Aware RepVGG (QARepVGG) [11] reduces the model parameters, and accelerates beach litter detection.
- Introducing A Simple, Parameter-Free Attention Module (SimAM) [12] attention mechanism in the YOLOv7, it enhances the features already acquired, gathers more useful information from images of beach litter, lessens the negative effects of complex backgrounds, focuses the model on beach litter objects, and reduces missed and false beach litter detection.
- Using the modified Bidirectional Feature Pyramid Network (BiFPN) [13] to improve the Path Aggregation networks (PAFPN) [14] structure of the original Neck; increases the model's capacity to extract features of various sizes and raises the model's accuracy in the identification of beach litter.

This paper adopts the following aspects to carry on the research. Section II explains the related research of object detection and beach litter detection. Section III introduces the suggested algorithm and discusses the reasons for and advantages of introducing three innovations. Section IV introduces the self-built dataset and discusses and analyzes the experimental results of the algorithm on the self-built dataset. Section V concludes and looks ahead to future work.

II. RELATED WORKS

A. Object Detection

Since 2012, deep learning has significantly advanced target detection technology. Deep learning-based target identification algorithms provide the advantages of high accuracy and resilience when compared to conventional target detection techniques. Depending on whether they must create candidate areas or not, deep learning-based target identification algorithms may be split into single-stage target detection algorithms and two-stage target detection algorithms [15].

R-CNN [16] marked the emergence of two-stage object detection algorithms, which first generate candidate regions and then put the candidate regions into the classifier to classify and correct the positions. Next, other two-stage object detection algorithms were put forth one after the other, including Fast Region-based Convolutional Network (Fast R-CNN) [17], Faster Region-based Convolutional Network (Faster R-CNN) [18], and more. However, due to its slow detection speed, two-stage object detection algorithms perform less well in practical applications.

The single-stage target detection algorithm is simple in structure, scalable, and more widely used, does not require candidate regions to generate branches, and detects the candidate frames and classes of targets directly at multiple locations in the image for a given input image. Its representative algorithms, such as the first You Only Look Once (YOLO) version of the target detection algorithm YOLOv1 proposed by Redmon et al. [19] in 2015, which completely discards the candidate region generation step and integrates classification, localization, and detection functions into one network, greatly improving the detection speed, but there are issues with missed and false detection and poor detection of small and multiple targets because of the simple network structure. The Single-shot multi-box detector (SSD) developed by Liu et al. [20] in 2015, which first introduced the idea of multi-scale detection, can improve the model's ability to detect small objects and be designed with a prediction module and a deconvolution module; however, the algorithm sacrifices a larger detection speed in exchange for a significant improvement in detection accuracy. Later, on the basis of YOLOv1, Redmon et al. [21][22] subsequently suggested YOLOv2 and YOLOv3 in 2017 and 2018, respectively. In order to address the issues of low recall and poor localization accuracy, YOLOv2 borrowed the Anchor mechanism of the Faster R-CNN algorithm, removed the fully connected layer from the YOLO network, and used the convolutional layer to predict the position offset of the detection frame and the category information [21]. In order to save computing effort, the YOLOv3 algorithm creates the DarNet-53 backbone network and makes use of only the 1x1 and 3x3 convolutional layers [22]. The YOLOv4 model, developed by Bochkovskiy et al. [23] in 2020, by fusing CSPDarknet53 and SPP to broaden the sensory field, and increases the detection accuracy of tiny objects. By expanding the effective aggregation network, adding the REP layer to facilitate new deployments, and adding Aux_detect for auxiliary identification, the YOLOv7 approach, which Wang et al. [24] introduced in 2022, obtains advantages in both speed and accuracy.

Although the single-stage technique identifies objects more rapidly than the two-stage method does, the detection results are still insufficient in detection with complex backdrops and enormous variations in object size. This study updates the YOLOv7 network using a method that boosts the network's feature extraction power to address these shortcomings and enhance the capacity of beach litter detection.

B. Beach Litter Detection

In 2011 Nakashima et al. [5] used balloon-assisted aerial photography combined with in situ measurements to estimate the amount of large litter on beaches; however, there is a large difference in litter density from place to place, so there is a large error in monitoring the amount of litter on beaches. Jang et al. [6] suggested employing a method that includes color and morphological image processing to compute the generation rate by applying thresholds to drone photographs in order to extract information about beach litter from the images. In 2012, kako et al. [7] built a low-altitude remote sensing system using a remotely operated digital camera suspended from a balloon, combining projection transformation methods and chromatic aberration in uniform color space (CIELUV) to process the resulting images to identify beach or marine litter. In 2018 Bao et al. [8] used remote sensing to apply a two-step threshold filtering method to images obtained by drones to identify and detect the distribution of beach litter. The aforementioned
study largely uses imaging techniques and projection transformation methods to monitor beach litter, however, it does not explicitly localize or identify beach litter, instead, it just monitors its amount.

Recently, researchers have started to carry out research and application practices for deep learning-based beach litter identification. In 2022, Rfeiffer et al. [9] investigated and compared the detection performance of two deep learning algorithms (YOLOv5 and Faster R-CNN) on images of beach litter taken by drones. The mAP value of YOLOv5 was 54.2% and the mAP of Faster R-CNN was 32.8%, the experimental findings demonstrate that the single-stage detection algorithm based on deep learning performs better for beach litter detection in terms of detection accuracy and detection speed. In a study of beach litter identification using YOLOv5 and camera acquisition picture data, Song et al. [10] achieved 87% detection accuracy, a notable increase over earlier research. In this investigation, over 90% of the dataset consisted of up-close images of beach litter collected at a height of 0.5 m. The beach litter objects in the close-up images were clear and there was mostly single-object beach litter in a single image, so a high detection accuracy was achieved; however, after using 1335 training photographs with complex backgrounds, the mAP of the plastic category significantly decreased from 88% to 26%, and there were some cases when shells were classified as plastic. Although there is a certain amount of misdetection, these experimental results show the immense potential and value of the YOLO model in beach litter detection application scenarios and demonstrate that the model's effectiveness in applications for identifying beach litter is significantly influenced by item morphology and characteristics, hyperparameter settings, and training data.

The analysis above shows that, even though YOLOv7's accuracy and efficiency have greatly improved over the previous YOLO model, more focused optimization is still required to increase the performance and suitability of beach litter detection.

First, it must be addressed that the number of model parameters and detection speed do not meet the needs of the actual applications. Wang et al. [25] incorporated GhostNet into the YOLOv5 model to improve its detection effectiveness. As a consequence, the number of parameters was successfully reduced by 47% and the computational complexity by 49.4%, but the detection accuracy fell by 2.2% compared to the original YOLOv5 model. By pruning the filters corresponding to the low-importance channels of the model to make it simpler to deploy the model to devices for real-time intelligent pedestrian monitoring, Xu et al. [26] obtained CAP-YOLO, which is three times faster than the original YOLOv5, but with a 7% reduction in mAP. Many current lightweight network models focus on compromising detection accuracy to boost detection speed, but beach litter detection demands high accuracy, hence this research introduces QARRepVGG in YOLOv7 to enhance detection speed while enhancing the accuracy of beach litter detection.

Secondly, the beach litter dataset contains a complex background consisting of leaves, branches, and other kinds of interfering objects, with interfering objects obscuring the target, overlapping between the target and the target, and the problem of fuzzy targets exists. The YOLOv7 algorithm still encounters missed and incorrect detection in such intricate backgrounds. The attention mechanism is often used for complex backgrounds. The Coordinate Attention (CA) mechanism that TCA-YOLO [27] introduced in YOLOv5 to weaken the interference of complex backgrounds has greatly improved the detection accuracy of small targets, but the detection effect of larger targets still requires improvement.

Additionally, in the actual application environment for beach litter detection, the size and form of the litter also vary, and there are many tiny and medium-sized targets. The great variety in target sizes makes detection extremely challenging. Li et al. [28] proposed adding jump connection and multi-structure multi-size feature fusion in feature extraction and feature fusion to improve the detection accuracy but slow down the detection speed.

For efficient and accurate beach litter detection, it is not advisable to sacrifice the detection speed or accuracy. Therefore, considering the practical application scenarios of beach litter detection, this paper optimizes the YOLOv7 model according to the difficulties in beach litter detection and proposes QSB-YOLO for beach litter detection, and tests the optimized QSB-YOLO. The experimental results show that QSB-YOLO not only reduces the number of parameters of the model and improves the detection speed of the model, but also improves the detection accuracy of beach litter.

III. MATERIALS AND METHODS

Considering the three problems in beach litter detection, we propose the beach litter detection method QSB-YOLO, whose overall network structure is shown in Fig. 1. Different from the original YOLOv7, QSB-YOLO replaces the first E-ELAN module of the YOLOv7 backbone network and the E-ELAN module in the neck with the quantization-friendly QARepVGG, then replaces the MP in the neck with the improved S-MP, and finally combines the idea of BiFPN to improve the original neck's PAFPN.

The beach litter image dataset is input into QSB-YOLO, and the image size is adjusted to 640x640 through input. The adjusted image is input into the backbone for feature extraction, and the obtained effective feature layers C3, C4, and C5 are input into the neck to strengthen the feature extraction of the network. The three outputs P3, P4, and P5 obtained from the neck are then sent to YOLOhead to predict the anchor frame, confidence, and category of beach litter after RepCov adjusts the number of channels.

A. Combine the Quantization-friendly QARRepVGG

The approach of model compression known as quantization, which successfully reduces a model's number of parameters but also lowers the model's performance, is frequently overlooked in deep neural networks. The reparameterized architecture-based multi-branch design widens the dynamic numerical range, which creates an issue with the difficulty of quantization. But the reparameterization architecture QARRepVGG we introduced is simple and efficient.
The custom weight decay design in RepVGG [29] is successful in building a model with a stronger weight distribution, but at the same time the variance of the activation distribution is amplified, the input of the subsequent layer depends on the activation, and the standard deviation rises layer by layer as the depth of the network layer increases, leading to an accuracy decrease. QARepVGG is improved based on RepVGG to solve the problem of quantization failure of RepVGG multi-branch design, which not only reduces the number of parameters of the model but also improves the detection accuracy and detection speed of the model [11].

According to the quantization-friendly features of QARepVGG, in order to improve the efficiency of beach litter detection, we combine QARepVGG in YOLOv7 and prove its effectiveness according to the experiment.

In order to measure quantification error, QARepVGG introduces mean square error (MSE), as in Equation (1).

\[
MSE(Q(w, t, n_b), w) = \frac{1}{n} \sum_i (Q(w_i, t, n_b) - w)^2
\]

(1)

where \( Q \) represents the quantization process, \( w \in \mathbb{R}^n \) represents the nth channel of the weights, \( t \) represents the stage threshold, and \( n_b \) represents the number of quantization bits. The output \( M_{(2)} \) is:

\[
M_{(2)} = BN(3 \times 3) + BN(1 \times 1) + BN(Identity)
\]

(2)

\( BN(3 \times 3) \) is:

\[
Y_{(3)} = \gamma_{(3)} \frac{W_{(3)} - \mu_{(3)}}{\sqrt{\epsilon + \sigma_{(3)}^2}} + \beta_{(3)}
\]

(3)

\( Y_{(3)} \) is the output of the 3x3 branch, \( \gamma_{(3)} \) is the scale factor of the BN layer after 3x3 convolution, \( \mu_{(3)} \) is the mean value of the BN layer after 3x3 convolution, \( \sigma_{(3)} \) is the standard deviation of the BN layer after 3x3 convolution, \( \beta_{(3)} \) is the deviation of the BN layer after 3x3 convolution. \( M_{(1)} \) is input, \( W_{(2)} \) is 3x3 convolution kernel, \( \epsilon \) is multiplication, \( \epsilon \) is the value that ensures numerical stability (The default value is \( 10^{-5} \)). This indicates that BN has the effect of stabilizing the variation of the input.

Introduce random variables \( X \) and scalars \( \lambda \), \( D(\lambda X) = \lambda^2 D(X) \), and make \( X_{(3)} = M^T W_{(3)} \), then we have

\[
D\left(Y_{(i)}\right) = \frac{Y_{(i)} \square Y_{(3)}}{\epsilon + \sigma_{(3)} \square \sigma_{(3)}} - D\left(X_{(i)}\right)
\]

(4)

The variation size of \( X_{(3)} \) is controlled by \( \frac{Y_{(3)} \square \sigma_{(3)}}{\epsilon + \sigma_{(3)} \square \sigma_{(3)}} \).

The reparameterization-based architecture needs to quantify the weight distribution and activation distribution. Friendly quantization refers to having a relatively narrow range of values and a narrow distribution of standard deviations; when one of these features is missing; the standard deviation is amplified, which reduces the model's accuracy [11]. The RepVGG custom weight decay is as in Equation (5).

\[
L_{custom} = \left( \frac{W_{eq}^2}{2} + \frac{\gamma_{(3)}^2}{2} + \frac{\mu_{(3)}^2}{2} + \frac{\beta_{(3)}^2}{2} \right)
\]

(5)
RepVGG reduces the weight loss by enlarging the denominator, but enlarges the standard deviation distribution and amplifies the activation distribution deviation. Based on this, QARepVGG removes the BN in the identity branch and replaces the custom weight decay design in RepVGG with the standard weight decay design (normal L2). As a result, the module successfully completes weight quantization. The output is then rewritten as

\[ M_{(2)} = BN(3 \times 3) + BN(1 \times 1) + \text{Identity} \]  

(6)

Let the anticipated values of 3x3 branches and 1x1 branches be

\[ E(Y_{(3)}) = \beta_{(3)}, E(Y_{(1)}) = \beta_{(1)} \]  

(7)

The variance may grow if \( \beta_{(3)} = \beta_{(1)} = \beta \), at this point \( \mu = 0, \sigma = 0, \gamma = 1, \beta = 0 \), then \( Y_{(3)} \) and \( Y_{(1)} \) are in balance. To better stabilize the variance, the BN in the 1x1 branch is further removed, at which time the output is

\[ M_{(2)} = BN(3 \times 3) + (1 \times 1) + \text{Identity} \]  

(8)

BN has the effect of stabilizing the input variance, and to stabilize the training process, QARepVGG adds batch normalization after three branches. At this stage, the output is

\[ M_{(2)} = BN(BN(3 \times 3) + (1 \times 1) + \text{Identity}) \]  

(9)

As illustrated in Fig. 2, the QARepVGG first converts each of the three branches into a single 3x3 convolution, then combines the three into one, and lastly adds the 3x3 convolution with BN to get the final 3x3 convolution. A multi-branch structure is used during training to improve the model's detection accuracy and the network's capacity for representation. To speed up the model's detection, single-branch inference is employed in the inference process.

Fig. 3(b) depicts the topology of QARepVGG in the YOLOv7+QARepVGG backbone network during training, and Fig. 3(a) depicts the E-ELAN module it replaces; the structure of QARepVGG in the Neck of YOLOv7+QARepVGG is depicted in Fig. 3(d), and Fig. 3(c) is its replacement E-ELAN module. When there is no BN versus with BN, the number of parameters for convolution are calculated as shown in Equations (10) and (11), respectively.

\[ \text{params} = C_o \times C_{in} \times k_h \times k_w + C_o \]  

(10)

\[ \text{params} = C_o \times C_{in} \times k_h \times k_w + C_o \times 2 \]  

(11)

where, \( C_o \) represents the number of output channels, \( C_{in} \) represents the number of input channels, \( k_h \) denotes the height of the convolution kernel, and \( k_w \) represents the width of the convolution kernel.
The number of parameters for the E-ELAM of the backbone network replaced by QARepVGG in the YOLOv7 is denoted by \( \text{params}_1 \), where \( \text{Cat} \) has no parameters. In YOLOv7+ QARepVGG, the number of parameters for the QARepVGG in the backbone network is denoted as \( \text{params}_2 \), where the 3x3 convolution has BN operation in QARepVGG, the identity branch has no parameters, and a new BN layer is added at the end. Similarly, the number of parameters for the E-ELAM of the Neck replaced by QARepVGG in the YOLOv7 is denoted as \( \text{params}_3 \), and the number of parameters for the QARepVGG in the Neck of YOLOv7+QARepVGG is denoted as \( \text{params}_4 \).

In addition to the replaced modules, the parameters for the remaining parts of the convolution were also affected. There are three convolutions that are \( C_n = 256, 128, k = k_n = 1 \) in YOLOv7 and become \( C_n = 256, 128, k = k_n = 1 \) in YOLOv7+QARepVGG, and two convolutions in YOLOv7 that are \( C_n = 512, 256, k = k_n = 1 \) become \( C_n = 256, 256, k = k_n = 1 \) in YOLOv7+QARepVGG, the number of parameters for them are denoted as \( \text{params}_5, \text{params}_6, \text{params}_7, \text{params}_8 \), respectively.

The total number of parameters of the affected modules is shown in Table I. It can be inferred from Table I that after the introduction of QARepVGG, the number of parameters decreased by 789248, or about 2.1%.

### B. S-MP

Through the analysis of the self-built beach litter dataset in this paper, we found that there exists a complex background composed of leaves, branches, and other kinds of distractors, the distractors obscure the target, the target overlaps with the target, and there is the problem of the target blurring, which easily causes missed and false detection. To improve this type of situation, this paper proposes to introduce an attention mechanism to attenuate the negative effects of complex backgrounds.

Traditional attention mechanisms, which produce one- or two-dimensional weights along the channel dimension or spatial dimension, focused only on this channel or this space, limit the flexibility of learning attention weights to alter throughout space and channel. Fig. 4 displays the SimAM schematic diagram. In contrast to the traditional spatial attention mechanism and channel attention mechanism, SimAM attention is a three-dimensional attention mechanism that assigns a unique weight to each neuron in space or channel without the use of additional parameters, determines the importance of each neuron, and then enhances the features using the three-dimensional weights. First, SimAM defines the energy function for each neuron as shown in Equation (12).

$$
e_i(w_t, b_t, y, x_t) = \left(\frac{1}{M-1} \sum_{y \neq y_t} \frac{1}{M-1} \sum_{i=1}^{M-1} (y_i - \bar{x}_t)^2\right) + \left(1 - (w_t + b_t)\right)^2 + \lambda \sigma^2$$

$$t$$ represents the target neuron, \( \hat{t} = w_t + b_t \) represents a linear transformation of \( t \). \( x_t \) indicates other neurons in the same channel as the input feature \( X \in [C \times H \times W] \), \( \bar{x}_t = w_x + b_t \) is a linear transformation of \( x_t \). \( i \) is the index in the spatial dimension, \( M = H \times W \) is the number of neurons on the channel, \( w_t \) and \( b_t \) are respectively the weights and biases of the transformation. When Equation (12) reaches a minimum, the target neuron finds linear differentiability with other neurons in the same channel, at which time \( \hat{t} = x_t \), and arbitrary \( \bar{x}_t = y_t \). Binary labeling of \( y_t \) and \( y_{t*} \), and adding regularization to Equation (12) yields Equation (13).

$$\epsilon_i(w_t, b_t, y, x_t) = \frac{1}{M-1} \sum_{j=1}^{M-1} \left[-1 - (w_x + b_t)\right]^2 + \lambda \sigma^2$$

$$w_t = \frac{2(t - \mu)}{(t - \mu)^2 + 2\sigma^2 + 2\lambda}$$

$$b_t = \frac{1}{2}(t + \mu)$$

$$\mu = \frac{1}{M-1} \sum_{i=1}^{M-1} x_i$$ and \( \sigma^2 = \frac{1}{M-1} \sum_{i=1}^{M-1} (x_i - \mu)^2 \) denote the mean and variance of all neurons except for that channel, respectively. The minimum energy is calculated as in Equation (16).

$$\epsilon_i = \frac{4(\hat{\sigma}^2 + \lambda)}{(t - \hat{\mu})^2 + 2\sigma^2 + 2\lambda}$$

$$\hat{\mu} = \frac{1}{M} \sum_{i=1}^{M} x_i$$

$$\hat{\sigma}^2 = \frac{1}{M} \sum_{i=1}^{M} (x_i - \hat{\mu})^2$$

$$\frac{1}{\epsilon_i}$$ indicates the importance of each neuron. That is the smaller the value of \( \epsilon_i \), the more the target neuron is distinguished from other neurons and the greater the importance. Subsequently, feature enhancement is performed according to the definition of attention mechanism as in Equation (19).
\[ \tilde{X} = \text{sigmoid}(\frac{1}{E}X) \]  

(19)

To reduce missed and false detection in beach litter detection and enable the model to obtain more useful features without increasing the model's parameters, we introduce the SimAM attention mechanism in the MP module in the neck of YOLOv7.

The MP in Neck is down-sampled in YOLOv7 using both Maxpool and BConv, and the features from each are then combined. As shown in Fig. 5, the SimAM attention mechanism is introduced to replace BConv in the second branch of the MP structure. The input of S-MP is down-sampled by Maxpool in the first branch, and then the number of channels is adjusted by BConv. In the second branch, the features are enhanced by the SimAM attention mechanism while adjusting the number of channels and then down-sampling by BConv. Finally, the features obtained from the two branches are feature fused to obtain the results of the enhanced down-sampling and obtain more effective features.

![Fig. 4. Schematic diagram of SimAM.](image)

![Fig. 5. Structure diagram of MP. (a) MP, (b) S-MP.](image)

### C. Improved-BiFPN

BiFPN is a feature fusion technique that combines weighted feature fusion with an efficient bidirectional cross-scale connection. The traditional Feature Pyramid Network (FPN) \[30\] fuses multi-scale features in a top-down manner, and the down-sampling process loses feature information at the highest level, reducing the multi-scale representation capability. A top-down, bottom-up path aggregation network is added by the PAFPN in the YOLOv7 network design. As shown in Fig. 6(a), in order to fuse more features without incurring excessive costs, the idea behind BiFPN is to add an extra edge to the original input and output nodes at the same layer; additionally, each bidirectional path is treated as a layer of the feature network, and the same layer is repeated multiple times to achieve a higher level of feature fusion.

We combine the idea of BiFPN and modify the PAFPN in the YOLOv7 network into the Improved-BiFPN to realize the bi-directional fusion of different network feature layers and enhance the information transfer between different network feature layers. As shown in Fig. 6(b), the three effective feature layers extracted from the backbone network are C3, C4, and C5, which are passed into the Improved-BiFPN to achieve effective bidirectional cross-scale connectivity and weighted feature fusion, resulting in a total of three different scales of P3, P4, and P5 output features.

![Fig. 6. Structure diagram of BiFPN. (a) The original BiFPN, (b) Improved-BiFPN.](image)

### IV. RESULTS AND DISCUSSION

#### A. Dataset Acquisition and Pre-processing

The beach litter image collection used in this paper was collected from several popular tourist beaches along the coast of South China. A total of 1587 beach litter photographs were captured by cameras during the beach survey, with resolutions ranging from 4000×2250 pixels to 480×360 pixels. To avoid the issue of overfitting brought on by insufficient training samples, data enhancement techniques such as rotation, color enhancement, and contrast enhancement were applied to each image separately, as shown in Fig. 7. After data augmentation, there are 6348 pictures overall, of which 90% are separated into training set and 10% into test set.

The self-built beach litter dataset collected a total of six categories of litter, including plastic products, metal products, paper products, wood, styrofoam, and glasswork, and the examples of each category are shown in Fig. 8. The beach litter image dataset used in this study was manually annotated using LabelImg annotation software and saved in YOLO format.

![Fig. 7. Data enhancement. (a) Original image, (b) Color enhancement, (c) Rotation, (d) Contrast enhancement.](image)
B. Experimental Parameter Settings

In this paper, there are 6348 beach litter image dataset, 5714 images are used as the training set, and 634 images as the test set. The experimental configuration is shown in Table II. The operating system used for training is Ubuntu 18.04.1, the GPU is NVIDIA GeForce RTX 3060 with 12G of video memory, the CPU is 11th Gen Intel(R) Core (TM) i5-11600KF, and the memory is 16G. The batch size is 8, the initial learning rate is 0.1, and 300 Epochs are trained.

Table II. HARDWARE EQUIPMENT AND DEVELOPMENT ENVIRONMENT

<table>
<thead>
<tr>
<th>Equipment</th>
<th>Model</th>
</tr>
</thead>
<tbody>
<tr>
<td>GPU</td>
<td>NVIDIA GeForce RTX 3060</td>
</tr>
<tr>
<td>CPU</td>
<td>11th Gen Intel(R) Core (TM) i5-11600KF</td>
</tr>
<tr>
<td>Ubuntu</td>
<td>18.04</td>
</tr>
<tr>
<td>CUDA</td>
<td>10.2</td>
</tr>
<tr>
<td>Python</td>
<td>3.7</td>
</tr>
<tr>
<td>Pytorch</td>
<td>1.8</td>
</tr>
<tr>
<td>Torchvision</td>
<td>0.7</td>
</tr>
</tbody>
</table>

C. Evaluation Indicators

This study evaluates the model using Average Precision (AP), Mean Average Precision (mAP), Parameters (Params), Frames Per Second (FPS), and F1. The value of AP, which measures the typical detection accuracy of a single category of beach litter, is the area of the P-R curve. The accuracy rate and recall rate are the P-R curve's vertical and horizontal coordinates, respectively. Equations (20) and (21), respectively, are used to compute the precision rate, P, and recall rate, R. Equation (22) is used to compute the value of the AP. Equation (23) is used to construct mAP, which is a measure of the average detection accuracy across all categories. The Params is used to measure the size of the model. FPS is a unit used to express how quickly a model can identify an object. The fraction of recalled genuine positive categories is measured by F1, which is determined using Equation (24).

Where TP denotes that the actual result is identical to the expected result, FN denotes that the category of beach litter is judged to be another category of beach litter or to be missed detection, and FP denotes that the non-beach litter target is detected as beach litter.

\[
P = \frac{TP}{TP + FP} \quad (20)
\]

\[
R = \frac{TP}{TP + FN} \quad (21)
\]

\[
AP = \int_0^1 P(R)dR \quad (22)
\]

\[
mAP = \frac{\sum_{i=1}^{N} AP_i}{N} \quad (23)
\]

\[
F1 = \frac{2PR}{P + R} \quad (24)
\]

D. Analysis of Experimental Results

1) Comparison Experiments Combining QARepVGG: To verify the effectiveness of the quantization-friendly QARepVGG method combined in this paper. RepVGG will be introduced in YOLOv7 to conduct comparative experiments with YOLOv7+ QARepVGG on our beach litter dataset, and the experimental results are shown in Table III.
The reparameterized architecture of RepVGG’s 1x1 branch at training time than QARepVGG’s 1x1 branch has one more BN operation, so the overall number of YOLOv7+ RepVGG parameters is 256 more than YOLOv7+ QARepVGG. The inference time of YOLOv7+RepVGG is 3.4ms faster than YOLOv7, but it is 0.4ms slower than YOLOv7+QARepVGG. The detection speed of YOLOv7+RepVGG is 16 fps faster than YOLOv7, but 2 fps slower than YOLOv7+QARepVGG.

RepVGG reduces the inference time and improves the detection speed significantly, but sacrifices the detection accuracy. While reducing the number of parameters in the model and increasing the detection accuracy of beach litter by 1.7%, QARepVGG increases inference speed and detection speed. Therefore, in this paper, we choose to introduce the quantization-friendly QARepVGG for beach litter detection.

2) Comparison experiments with attention mechanisms: In this paper, we combine the Convolutional Block Attention Module (CBAM) [31] and Efficient Channel Attention Module (ECA) [32] with the MP module of Neck in YOLOv7 and compare them with YOLOv7+S-MP for comparison, and the experimental results are shown in Table IV. Where C-MP denotes the use of the CBAM attention mechanism in place of the first convolution of the original Neck’s second branch of MP. E-MP denotes that the first convolution of the second branch of MP in the original Neck is swapped out for the ECA attention mechanism.

CBAM is a hybrid attention mechanism, YOLOv7+C-MP detection accuracy is reduced by 3.2% compared to YOLOv7, F1 is reduced by 0.03, and the number of parameters is reduced less. Both ECA and SimAM are parameter-free attention mechanisms, and when the same convolution is replaced, YOLOv7+E-MP has the same number of parameters as YOLOv7+S-MP. However, when ECA is added, mAP@0.5 is decreased by 10.5%. Beach litter detection on complicated backdrops is not appropriate for either CBAM or ECA, which both decrease the mAP@0.5 of beach litter. While adding the SimAM attention mechanism the mAP@0.5 is increased by 3.9%, F1 by 0.04, and the parameter-free attention mechanism does not negatively affect the number of model parameters.

TABLE III. COMPARATIVE EXPERIMENTS COMBINED WITH QARepVGG

<table>
<thead>
<tr>
<th>Model</th>
<th>mAP@0.5</th>
<th>Params</th>
<th>Inference times</th>
<th>FPS(fps)</th>
</tr>
</thead>
<tbody>
<tr>
<td>YOLOv7</td>
<td>79</td>
<td>37223526</td>
<td>11.8ms</td>
<td>40</td>
</tr>
<tr>
<td>YOLOv7+RepVGG</td>
<td>77.7</td>
<td>36434534</td>
<td>8.4ms</td>
<td>56</td>
</tr>
<tr>
<td>YOLOv7+QARepVGG</td>
<td>80.7</td>
<td>36434278</td>
<td>8.0ms</td>
<td>58</td>
</tr>
</tbody>
</table>

3) Comparison experiments of improved enhanced feature extraction network: PAFPN is used as an enhanced feature extraction network in the original YOLOv7’s Neck. The Improved-BiFPN is suggested to replace the original PAFPN in this research in order to increase the detection capabilities of the model for various sizes of beach litter in practical applications. The acquired findings are displayed in Table V. When compared to the original model, the Improved-BiFPN increases the network’s detection capacity for targets of various sizes, while also increasing mAP@0.5 by 4.1% and the F1 by 0.05. The Improved-BiFPN significantly improves the detection accuracy of the model without a significant increase in the number of parameters and without decreasing the detection speed, therefore we replace the original Neck for beach litter detection.

TABLE V. COMPARATIVE EXPERIMENTS OF NECK

<table>
<thead>
<tr>
<th>Model</th>
<th>mAP@0.5</th>
<th>F1</th>
<th>Params</th>
<th>FPS(fps)</th>
</tr>
</thead>
<tbody>
<tr>
<td>YOLOv7</td>
<td>79</td>
<td>0.73</td>
<td>37223526</td>
<td>40</td>
</tr>
<tr>
<td>YOLOv7+Improved-BiFPN</td>
<td>83.1</td>
<td>0.78</td>
<td>37354598</td>
<td>40</td>
</tr>
</tbody>
</table>

4) Comparison Experiment of Beach Litter Classification Results: The AP of each category of various detection models in the self-built beach litter dataset is displayed in Table VI. Styrofoam, paper products, and glasswork have excellent AP when YOLOv7 is used directly for beach litter detection, whereas the AP of plastic products, metal products, and wood is relatively low. In YOLOv7+QARepVGG, the AP of each category is well balanced, with the exception of metal products, whose AP has decreased by 3.9%. It is clear that the AP of each category has improved after the SimAM attention mechanism was added to the MP module, and glasswork and metal products have the best average accuracy in comparison to the other models in this research. The introduction of improved-BiFPN effectively contributes to the improvement of AP for each category, with the most notable improvement of 12.4% in AP for wood beach litter with high size variation. In QSB-YOLO, the best AP was achieved for styrofoam, plastic products, and paper products relative to all other models, and the AP for glasswork and wood improved by 3.9% and 6.6%, respectively, with only a slight decrease of 2% for metal products. In conclusion, in general, QSB-YOLO is significantly better than the original algorithm.

TABLE VI. AVERAGE PRECISION FOR EACH CLASS OF BEACH LITTER

<table>
<thead>
<tr>
<th>Model</th>
<th>Styrofoam</th>
<th>Plastic product</th>
<th>Metal product</th>
<th>Paper product</th>
<th>Glasswork</th>
<th>Wood</th>
</tr>
</thead>
<tbody>
<tr>
<td>YOLOv7</td>
<td>88.6</td>
<td>69.6</td>
<td>69.4</td>
<td>83</td>
<td>92.7</td>
<td>70.9</td>
</tr>
<tr>
<td>YOLOv7+QARepVGG</td>
<td>90.2</td>
<td>77.3</td>
<td>65.3</td>
<td>86.6</td>
<td>92.2</td>
<td>77.5</td>
</tr>
<tr>
<td>YOLOv7+S-MP</td>
<td>92.2</td>
<td>71.4</td>
<td>71.4</td>
<td>90.4</td>
<td>98.7</td>
<td>73.4</td>
</tr>
<tr>
<td>YOLOv7+Improved-BiFPN</td>
<td>93.2</td>
<td>77.9</td>
<td>68.1</td>
<td>87.4</td>
<td>88.9</td>
<td>83.3</td>
</tr>
<tr>
<td>QSB-YOLO</td>
<td>94.3</td>
<td>80.7</td>
<td>67.4</td>
<td>92.5</td>
<td>96.6</td>
<td>77.5</td>
</tr>
</tbody>
</table>
5) Ablation experiments: To verify the effectiveness of each improvement method in this paper, the three improvement modules were added to the original YOLOv7 network structure one by one, and the ablation experimental results are shown in Table VII and Fig. 9.

Table VII shows that, in comparison to the YOLOv7 model, the YOLOv7+QARepVGG model adds the quantization-friendly QARepVGG module, which not only reduces the number of parameters of the model, increases its detection speed and inference speeds, but also contributes to the improvement of the mAP@0.5 of the model. Both YOLOv7+SimAM and YOLOv7+Improved-BiFPN improve the mAP@0.5 and F1 of the model compared to the original YOLOv7. Compared to the original model YOLOv7, QSB-YOLO’s mAP@0.5 raises 5.8%, the F1 increases by 0.08, the number of parameters decreases by 281600, the inference speed is accelerated by 3.5ms, and the detection speed of the model is improved by about 43%.

Fig. 9 visualizes the performance difference between each individually added module and YOLOv7 and QSB-YOLO. As can be seen from Fig. 9, the QSB-YOLO model has superior performance and is somewhat advanced in the training process for beach litter detection.

![Fig. 9. Comparison chart of ablation experiments.](image)

<table>
<thead>
<tr>
<th>Model</th>
<th>mAP@0.5</th>
<th>F1</th>
<th>Params</th>
<th>Inference times</th>
<th>FPS(f/s)</th>
</tr>
</thead>
<tbody>
<tr>
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<td>0.73</td>
<td>37223526</td>
<td>11.8ms</td>
<td>40</td>
</tr>
<tr>
<td>YOLOv7+QARepVGG</td>
<td>80.7</td>
<td>0.75</td>
<td><strong>36434278</strong></td>
<td><strong>8.0ms</strong></td>
<td><strong>58</strong></td>
</tr>
<tr>
<td>YOLOv7+SimAM</td>
<td>82.9</td>
<td>0.77</td>
<td>37140838</td>
<td>11.8ms</td>
<td>40</td>
</tr>
<tr>
<td>YOLOv7+Improved-BiFPN</td>
<td>83.1</td>
<td>0.78</td>
<td>37354598</td>
<td>11.8ms</td>
<td>40</td>
</tr>
<tr>
<td>QSB-YOLO</td>
<td><strong>84.8</strong></td>
<td><strong>0.81</strong></td>
<td>36941926</td>
<td>8.3ms</td>
<td>57</td>
</tr>
</tbody>
</table>

Table VII. Results of the Ablation Experiment on the Beach Litter Dataset in This Paper

![Fig. 10. Heatmap visualization results. (a), (c) the heatmap of YOLOv7; (b), (d) the heatmap of QSB-YOLO.](image)

6) Comparison with other traditional models: To demonstrate its superiority for beach litter detection, we compare the QSB-YOLO with Faster R-CNN, EfficientDet [13], SSD, YOLOv5 [33], YOLOX [34], and YOLOv7 algorithms. The input image size is 640×640, and the framework used is Pytorch. The experimental results are displayed in Table VIII. Under the same experimental setup and beach litter dataset, QSB-YOLO beats other traditional models in terms of detection speed and mAP@0.5.

![Fig. 11. Image detection results:](image)

![Fig. 11. Image detection results:](image)

7) Image detection results: To confirm the real detection performance of QSB-YOLO for beach litter in this paper, we compared the detection effects of YOLOv7 and QSB-YOLO, as illustrated in Fig.11.

![Fig. 11. Image detection results:](image)

It can be found in Fig. 11 that YOLOv7 in Fig. 11(a) does not detect the plastic products beach litter which is fuzzy and smaller in size in the distance, and QSB-YOLO successfully locates and identifies the beach litter. Fig. 11(c) misses a plastic products beach litter of smaller size relative to other
beach litter, which QSB-YOLO also successfully identifies and locates, and the confidence of QSB-YOLO's prediction frame is significantly higher than the confidence of YOLOv7. The beach litter image in Fig. 11(e) has a complex background, and the leaves obscure some of the targets, and YOLOv7 mistakenly detects the leaves as beach litter, and there are also missed detections. In Fig. 11(f), there is no false detection and three fewer missed detections, and only one plastic products beach litter obscured by leaves is not detected.

In summary, compared with the original algorithm, the detection accuracy of QSB-YOLO proposed in this paper is significantly improved, and the leakage and false detection are reduced.

V. CONCLUSION

In our study, we apply data augmentation approaches for the beach litter dataset to increase the model's detection range for beach litter in order to avoid overfitting caused by inadequate samples and propose an efficient QSB-YOLO beach litter detection model to address the issues of missed and false detection in beach litter identification. In addition to reducing the number of model parameters, combining with the quantization-friendly QARepVGG also enhance the model's detection precision and speed. The self-built beach litter dataset used has a complicated backdrop, which causes issues like blurred targets and obscured targets. In order to focus the model on the objective of beach litter and improve detection performance while reducing the possibility of missed and false detection, the MP module in the Neck is paired with the 3D SimAM attention mechanism. The original PAFPN is replaced...
with the Improved-BiFPN to increase the network’s ability to learn various size characteristics, address the detection difficulties issue brought on by the large size range of beach litter targets, and enhance the model’s detection accuracy.

The experimental results show that the QSB-YOLO suggested in this research is far better than the original YOLOv7 and other traditional target detection models for beach litter identification accuracy and detection speed.

In future work, we plan to expand the beach litter dataset to include more diverse beach litter targets and continue research to solve the difficulties beach litter identification in complicated contexts, so as to further improve the practical application value of the model.

ACKNOWLEDGMENT

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REFERENCES


SECI Model Design with a Combination of Data Mining and Data Science in Transfer of Knowledge of College Graduates’ Competencies

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Department of Computer Science, Universitas Sriwijaya, Palembang, Indonesia2, 3, 4

Abstract—One of the methods in knowledge management that can be used is the SECI Model. The SECI Model transfers tacit and explicit knowledge in each quadrant. However, without using tools, the transfer of technical knowledge will experience various obstacles. These obstacles included limited knowledge of the informants, difficulty in translating what was conveyed by the informants, limited time and opportunities, and unclear results obtained. The transfer of knowledge needed by college institutions is in the form of input from graduates who have graduated from college institutions. Graduates' knowledge must be obtained to determine whether their competence is following their respective fields of knowledge. Information technology can help overcome technical problems in transferring knowledge, including the problem of large amounts of data. Data science will deliver results from a combination of technology and mathematics. Meanwhile, data mining, especially with classification, grouping, and association functions, can provide a clear picture of the needs of higher education institutions for the knowledge of their graduates to assess the curriculum that has been provided so far. The design formulation of the SECI model and the implementation of this data mining use an empirical approach through observation and experimentation with quantitative data, as well as theoretical thinking in supporting the development of the model development concept. Data mining and data science will clarify processes in the SECI Model quadrant regarding technological tools in the context of knowledge transfer in a circular manner between tacit and explicit, in order to be more directed and precise. Information extracted from graduate competencies can assist college institutions in formulating future strategies in the academic field, especially the curriculum in study program. This result will impact students in the future, where the developed curriculum will focus more on the results of the input of graduate students.

Keywords—Model SECI; data mining; data science; competence of graduates

I. INTRODUCTION

Competence is defined as knowledge, skills and individual abilities that can directly influence entrepreneurial activity, which is the goal to be achieved [1]. Meanwhile in the field of education, measuring the competence of graduate students can be done using the models that describe the transfer of information, such as the SECI model (Socialization, Externalization, Combination, and Internalization). In [2], the study relates to the design of measuring instruments and means to measure student competence in the enterprise system laboratory and research solutions using the SECI model.

Data science, predictive analytics, and big data have been frequently implemented in various fields. Rigorous academic investigation of this science and method will lead to new areas. Article [3] discussed the results of a recent large-scale survey among supply chain management professionals on this topic. Data analytics and big data are also discussed in the research [4] which review how data mining and learning analytics have been applied to education data. In the last decade, the field of research has grown enormously. Various related terms are used in bibliographies such as academic, institutional, and teaching analytics, data-driven, decision-making, big data, and data science in education.

The data mining functionality helps in predictions of various things such as predicting customer buying behavior in the field of business [5], and predicting the final residual value in education field [6]. Besides prediction, the most frequently used functionality is clustering and classification. In the era of big data, in particular, clustering is a prevalent theme [7]. Finding clusters with different densities is usually tricky because finding clusters with various densities uses a fixed radius. The extended method was used to find clusters with different densities [8]. As in research [9], The K-means algorithm, which considers the distance from each point to each centroid in each cluster, is also commonly used. Decision tree classification is a method that is also commonly used [10]. At the same time, other algorithms in classification techniques are often compared [11].

Outliers in the data sometimes occur, namely when mining is done and patterns in the data that do not match the expected behavior found. Develop work using a statistical approach used to detect outliers [12]. Evaluation of the performance of data mining classification methods, for example, in research [13] that evaluates classification performance in the Internet of Things applications, needs to be done. Related research describes a significant literature survey of Sustainable Smart Cities, Machine Learning, and Data Mining. The most cited relevance and method and feature set data were identified, read, and summarized [14]. In research [15], a survey was conducted, which reviewed and discussed a detailed analysis of 142 research articles using various techniques. This survey will later introduce a model of the various data mining functionalities.
Data collection was necessary for knowledge transfer. A qualitative approach was used to analyze the collected empirical data. Data was collected using various sources, including semi-structured interviews, questionnaires, and internal document processing [16]. Tools were discussed in a study [17] which presents data mining in the data transfer process using the SECI model. However, the discussion was still unclear, there must be a more detailed explanation regarding data mining in each quadrant.

Research conducted with large amounts of data or without special tools would make it difficult for researchers to examine the data. In terms of quantitative data in particular, using tools in the form of technology was very useful in collecting and processing data. Data mining could be used in information dissemination and data processing. In data management, data mining techniques can extract and find valuable and meaningful information from large amounts of data [18]. Implementing information technology, especially data mining, reduces the burden of human error. However, the process still needs to be carried out clearly and in detail, so that every point in the SECI model process can ensure the correctness of the data transfer process.

Graduate competence is very important for Higher Education. Students must be prepared to stand out in the workforce after graduation. Transferring information from graduates can allow college to obtain data and information which can be used as a strategic aspect in future curriculum development. Transfer of graduate knowledge with the SECI Model design was accompanied by improvements to the data mining design in each quadrant of the SECI model can help to overcome the previous model’s shortcomings. The design of this research model was a refinement of the previous model, where the SECI model quadrant has data science and data mining which were more detailed in the process and has a special data mining algorithm.

II. LITERATURE REVIEW
A. Transfer of Knowledge with the SECI Model and Information Technology
State of the art mapping and knowledge transfer research gaps using the SECI model have been discussed from the discussion of previous preliminary research [18]. Data analysis from related papers were presented generally in the form of bibliographic data analysis using the Vos Viewer as illustrated in Fig. 1 below:

![Bibliometric network visualization output results.](image)

Several studies that show the use of SECI model include designing measuring instruments and determining how to measure with SECI model [2] and managing knowledge using the perspective of SECI model [19]. Then to get something or new results using SECI model was shown in studies that empirically test the analysis to create knowledge models [16] and knowledge transfer using SECI model to create new knowledge [20]. While from a technical point of view, research [21] actualizes and provides empirical consistency to the theory underlying the SECI model.

Information technology as a tool, will facilitate the search for results. Research [22]–[24] uses data mining as a tools to create knowledge management system. Process Knowledge discovery process from databases use data mining techniques, and the SECI knowledge dimension model. Thus, information technology enables the completion of various functions with SECI model.

B. Slice of Knowledge Mapping Gap
Information technology can be used for the design of models and frameworks. The framework built by adding information technology to the knowledge management system [17]. Models can be developed by exploring the adaptation of information technology to the knowledge conversion process [25].

The SECI model helps a lot in transferring information in building a knowledge management system in an organization [16], [20], [21]. With SECI model, knowledge transfer will become more focused and measurable in each quadrant. Research gaps can be seen in weaknesses that arise when tools are not used, as in research [2], where human abilities, understanding, and resources limit the knowledge transfer to the graduates. There are often inconsistent perceptions between respondents and interviewers. Another loophole occurred in research [25] where there were limitations in data collection due to privacy issues related to sources and some informants’ refusal to participate in the research.

Research that used technology to assist, is limited by small and inadequate sample of data [26], and not all implementations of information technology fit in every phase of SECI model [24]. Data mining as part of information technology has been implemented in research [17]. However, the discussion is still unclear and concrete; each SECI process must have an explanation in its quadrants. From the deficiencies described, the previous model needs to be corrected and revised, as discussed in several articles. Data mining has more detailed features to explain each data transfer process. Functions that can be used include classification, grouping, and association functions. This function can complement the model created to explain and fill the gaps in previous research. When designing data mining models in the SECI model, relevant and suitable algorithms can be used in the quadrants of the SECI model.

III. RESEARCH METHODOLOGY
The stages for the research method begin with data collection. Data is used to formulate the problem, which can be done using data collection techniques in the form of a literature study. This data collection technique is carried out by collecting the required data using literature studies and looking for scientific journals related to the research theme. Related
journals are built in the form of state-of-the-art to get research gaps. The next stage is direct interviews conducted with the head of the Study Program, the Dean, and Vice Chancellor III of student affairs, as well as distributing questionnaires to respondents, namely graduates—technical data collection using written and computer media tools. The questionnaire was carried out by asking several questions to the respondent, namely graduates who had worked. The questionnaire results are stored in written documents or computerized storage media. This initial stage is the first stage in the SECI model, namely socialization which is the transfer of knowledge from tacit to tacit.

The next step is the Externalization stage, where data transformation is performed. At this stage, hypotheses and conclusions are sought from the data that has been obtained. This process continues to the data cleansing process, which is carried out because the data received is still not clean and must be reprocessed before proceeding with the next data. The Externalization quadrant focuses on data documentation for storing data from tacit to explicit.

After the data is transformed or processed, it is continued to the data analysis stage at the Combination SECI explicit to the explicit stage. Datasets that have gone through the process of transformation and cleansing are then grouped using the K-Means Algorithm before being predicted using the same data. Classification forecasts use the Decision Tree Algorithm to test future possibilities. The third data mining functionality, namely the association, uses the Apriori Algorithm, which adapts transaction data to map graduate competency data with associated attributes. At the data analysis stage, system design and development are in the combination phase. Patterns are extracted from the test data to produce output from training data. Discovery Process describes data mining as the process of disseminating data. Data mining is about finding information or information that was not previously known to exist. Data science plays a role in information extraction, where data science helps extract valuable information from mined data. Data is analyzed from data mining results, identifying relevant patterns, trends, and relationships and exploring insights that can be used for decision-making.

The internalization process is carried out by transferring knowledge from explicit to tacit. This is done after the system is completed, implemented, and tested. This process is carried out by returning explicit knowledge to tacit by sharing the analysis results with users. The results of this analysis can also be strategically used by management.

The following is a draft framework for the SECI model with data mining and data science for graduate competency data, which is given in Fig. 2.

Tests and results can be carried out using statistical methods to test the results of the implementation of data mining and the knowledge management model built whether they are appropriate and suitable, then conclusions are drawn. Testing and evaluation performed with tools to see the relationship between the model results built and user needs.

IV. RESULT AND DISCUSSION

A. Stages of Socialization

The first stage is in the socialization quadrant. Data is collected by tacit to tacit knowledge. Knowledge from related parties who mastered and indeed in their fields to formulate graduate competencies was interviewed. The parties in question are the Head of the Study Program, the Dean, and Vice Chancellor III, who handle students’ affairs and graduates. This interview was conducted based on official documents from the government in the form of a Regulation of the Minister of Research, Technology, and Higher Education of the Republic of Indonesia. Government regulations define National Higher Education Standards for various levels of education programs. This research is limited to the undergraduate graduates in two fields of study: computer science and also economics and business.

The competence of graduates is contained in four formulations, namely the formulation of general attitudes and skills, which contain the same formulation for all study programs. The following formulation is specific knowledge and skills tailored to each study program. From interviews with related parties, graduates’ competency formulations were obtained and mapped into questionnaire questions aimed at the intended respondents, which is the graduates.

B. Stages of Externalization

The next SECI quadrant is the externalization stage which includes the process of cleansing and transforming data. The data is divided into two types of respondents with questions tailored to their respective fields of knowledge. From a
population of 6,536 graduates of a tertiary institution, a sufficient number of samples will be sought. Respondents consisted of graduates from two science fields, computer science, and economics. Respondents from the computer science field consisted of information systems, informatics, informatics management, accounting computerization, and computer engineering study programs. Meanwhile, respondents from economics consisted of management and accounting study programs.

The data from the results of filling out the questionnaire totaled more than 400 respondents. After cleansing, it was found that some data did not meet the requirements, such as blank and double contents. From the cleaning results, clean data for 387 graduates was obtained. According to the Slovin formula, from a total population of N totaling 6,536 graduates and with an error tolerance of e = 0.05, the minimum number of samples of n respondents is obtained as follows:

\[
n = \frac{N}{1 + Ne^2}
\]

With this result, this number is sufficient for the minimum number of the total population of all graduates. Then testing the data is done, namely the validity and reliability testing. Validity test is done to see the correlation of each question to other questions. Following are the results of the correlation validity test for each question for the data types of respondents who are graduates of computer science and economics and business, which are presented in Fig. 3 and 4.

From the two pictures above, with an error tolerance of 0.05 from the r table value of 0.1195 for computer science graduate data and 0.1851 for economics graduate data, it can be seen in these figures that the r count meets the r table value and can be said to be valid.

Reliability test for computer science, and economics and business graduate data is presented in Tables I and II below:

<table>
<thead>
<tr>
<th>Cronbach’s Alpha</th>
<th>N of Items</th>
</tr>
</thead>
<tbody>
<tr>
<td>0.948</td>
<td>14</td>
</tr>
<tr>
<td>0.961</td>
<td>14</td>
</tr>
</tbody>
</table>

From the results of the reliability test above, it was found that the reliability value was > 0.9. Then the data reliability results are included in the firm and perfect category. The data transformation process adjusts the data format to enter the data mining technical process.

C. Stages of Combination

The system was built and designed with data mining at the SECI combination model stage consisting of three primary functions: classification, clustering, and association. The data is divided into three appropriate sections for the three functionalities of classification, clustering, and association. Especially for associations, the data is adjusted to the number of choices of respondents’ answers so that the amount of data is no longer the number of respondents but the number of choices.

1) Classification: The classification dataset consists of 14 questions covering three elements of attitude, five elements of general skills, four elements of knowledge, and two unique skills. Because the questions on specific knowledge and skills are divided into two types, the total number of questions is 20. The details of the questions are presented in Table III below:

<table>
<thead>
<tr>
<th>Cronbach’s Alpha</th>
<th>N of Items</th>
</tr>
</thead>
<tbody>
<tr>
<td>0.948</td>
<td>14</td>
</tr>
</tbody>
</table>
TABLE III. QUESTIONS IN GRADUATE COMPETENCY QUESTIONNAIRE

<table>
<thead>
<tr>
<th>No.</th>
<th>Element</th>
<th>Competency Standards</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>Attitude</td>
<td>Deity</td>
</tr>
<tr>
<td>2</td>
<td></td>
<td>Nationalism</td>
</tr>
<tr>
<td>3</td>
<td></td>
<td>Personal</td>
</tr>
<tr>
<td>4</td>
<td></td>
<td></td>
</tr>
<tr>
<td>5</td>
<td></td>
<td></td>
</tr>
<tr>
<td>6</td>
<td></td>
<td></td>
</tr>
<tr>
<td>7</td>
<td></td>
<td></td>
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<tr>
<td>8</td>
<td></td>
<td></td>
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<tr>
<td>9</td>
<td></td>
<td></td>
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<tr>
<td>10</td>
<td></td>
<td></td>
</tr>
<tr>
<td>11</td>
<td></td>
<td></td>
</tr>
<tr>
<td>12</td>
<td></td>
<td></td>
</tr>
<tr>
<td>13</td>
<td>Special Skill</td>
<td>Software Development/General Management and Accounting Concepts and Principles</td>
</tr>
<tr>
<td>14</td>
<td></td>
<td>Data and Analytics/Enterprise Analysis</td>
</tr>
</tbody>
</table>

The target variable in the graduate competency dataset is whether the graduate’s work is following the field of study according to the study program or not. The algorithms and programming languages used are decision trees and Python programming. The Python libraries used for classification include the Pandas Library, Matplotlib, Numpy, Pydotplus, and Graphviz. Binary decision tree results are obtained up to 23 levels. Fig. 5 below shows the two highest levels of the classification tree.

![First and second level classification trees.](image)

The highest branch of the tree, or the 0th tree, is Elements of General Skills: Scientific Communication with a value of ≤ 4.5, gini = 0.434, samples 387, and value = [123, 264], this means:

- For questions with an answer value of 4.5 or lower, the scientific communication elements will follow the True arrow to the left and the rest will follow the False arrow to the right.
- Gini value = 0.434 means that the quality of separation is in this number. The formula for calculating the gini is given as follows:

  \[
gini = 1 - \left( \frac{x}{n} \right)^2 - \left( \frac{y}{n} \right)^2 \quad (2)
  \]

  \(x\) is target variables that are by the field of science, and \(y\) is target variables that are not by the field of science.

- Samples = 387 total datasets.
- Value = [123, 264] means that out of a total of 387, 123 will get the “Not suitable” category, and 264 will get the “Suitable” category.

A similar discussion also applies to subsequent branches up to the last branch at the 23rd level.

2) Clustering: The grouping of data will later be connected with segmentation. The data that is suitable for use is the identity of the dataset respondents. Of the many questions, three attributes will be taken, namely age, GPA, and length of study in years. The Elbow method determines the number of clusters and cluster members; the K means algorithm is used.

From the results of coding Python programming with libraries almost the same as classification, we get a recommendation for the number of clusters \(K = 3\) using the Elbow method.

The formula for the distance of two points for three dimensions is given as follows:

\[
d(x, y, z) = \sqrt{(x_2 - x_1)^2 + (y_2 - y_1)^2 + (z_2 - z_1)^2} \quad (3)
\]

Furthermore, the results of clustering with a 3D plot are obtained as illustrated in Fig. 6 below:

![3D cluster results.](image)

From the picture above, the following data is obtained:

- Cluster 0: Consisting of 209 data with age distribution under 26 years and a combination of GPA and length of study time.
- Cluster 1: Consisting of 151 data with an age distribution between 27 to 32 years and a combination of GPA and length of study time.
• Cluster 2: Consisting of 27 data with age distribution over 32 years and a combination of GPA and length of study time.

These results indicate that the age attribute of the respondent most influences the cluster results. Meanwhile, the other attributes, namely GPA and length of study, are spread throughout the clusters.

3) Association: The Python library required for association is almost the same as for classification and clustering. The addition is the Apriori Library, which is used for association. Graduate competency data associations are derived from transaction associations. The dataset contains knowledge data from graduates answering eight multiple-choice questions in checklist format, allowing for the selection of more than one option. The number of dataset records corresponds to the number of options selected, rather than the number of respondents. The association questionnaire questions are presented in Table IV below:

<table>
<thead>
<tr>
<th>No.</th>
<th>Graduate Knowledge</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>Theoretical and practical knowledge in the discipline studied</td>
</tr>
<tr>
<td>2</td>
<td>Ability to think critically and analytically in analyzing complex problems and situations</td>
</tr>
<tr>
<td>3</td>
<td>Knowledge of values, ethics and governance related to the disciplines studied, as well as the ability to apply them in diverse social, political and cultural contexts</td>
</tr>
<tr>
<td>4</td>
<td>Ability to identify, evaluate and use information effectively from a variety of sources</td>
</tr>
<tr>
<td>5</td>
<td>Ability to communicate effectively in an appropriate language, both orally and in writing</td>
</tr>
<tr>
<td>6</td>
<td>Ability to work independently or as a team member in different situations, and ability to lead and manage teams on complex projects</td>
</tr>
<tr>
<td>7</td>
<td>Knowledge of social, cultural and environmental diversity, as well as the ability to adapt to differences and respect this diversity</td>
</tr>
<tr>
<td>8</td>
<td>The ability to develop life skills, such as the ability to solve problems, make decisions, and learn</td>
</tr>
</tbody>
</table>

The association support formula is as follows:

\[ s(X \rightarrow Y) = \frac{|X \cup Y|}{N} \]  

Association uses the Apriori Algorithm using a specific support value. With the support of 0.4 of the 1774 selected dataset records, 17 rules are obtained. Detailed rules consist of 15 with two itemsets and two with three itemsets. The rules consist of 3 itemsets, namely the itemsets for questions 8, 4, and 2 and questions 8, 6, and 2.

D. Stages of Internalization

The final stage of the SECI model is internalization, where explicit knowledge returns to tacit. From the results of data mining processing, analysis of the results of data science and analytical recommendations will return to the relevant parties. The results obtained from the three significant functionalities of classification, clustering, and association are given to the Chair of the Study Program, Dean, and Vice Chancellor III for further identification and study to plan future strategies. The strategy in question is to prepare students who are still in college so they can face the world of work after graduating from college, accompanied by sufficient knowledge and competence.

E. The SECI Model with Data Mining and Data Science

The results of three data mining functionality classifications, clustering and association, are then discussed; concerning data science to find the results of data analytics that institutions can later utilize. Data science focuses on extracting, processing, analyzing, and interpreting data to understand phenomena better. The main goal of data science is to identify patterns, trends, and insights contained in data to make better decisions or develop more effective solutions to problems.

In the design model that has been made, with the dataset obtained, for classification, the highest branch obtained is question number 6 (X[5]), namely the Competency Standards for General Skills Elements: Scientific Communication which most influences the suitability of graduates between current jobs, and fields of study obtained from lectures. So from the results obtained, for related parties, both for the Chair of the Study Program, the Dean, Deputy Chancellor III, and graduates. The analytical recommendations can be given so that they can focus more on the elements that influence graduates to be able to work if scientific communication is essential. Various scientific or technical information delivery activities can be carried out, such as seminars or forums, which support students so they can be more involved in scientific journal publications, conference presentations, and scientific discussions.

The cluster results, which consist of the attributes of age, GPA, and length of time in college, tend to be the group most influenced by age. Data science plays a role in segmenting cluster results, where analytical recommendations can be given to respondents based on age. The recommendation is estimated with age groups that are close together; they will likely be in adjacent generations and at almost the same era, so it can be suggested that events involving graduates can be divided by age or generation.

Recommendation analytics for associations depend on the results of the rules. For rules with three itemsets in questions number 8, 4, and 2 as well as numbers 8, 6, and 2, provide an association rule pattern: If graduates have life skills and can work, then graduates will be able to think complexly. Recommendations that can be given to the Head of Study Program, Dean, and Vice Chancellor III are that students can be equipped with sufficient soft skills to face the world of work after graduating from college.

V. Conclusion

SECI modeling has been widely used by researchers in transferring knowledge. However, along with the increasingly sophisticated technology, tools will be very helpful in transferring knowledge. With the construction of the SECI model in transferring graduate knowledge with a combination of data mining and data science, it is hoped that it can help related parties. The Head of the Study Program, Dean, and Deputy Chancellor III can use the results of information and knowledge trainers from graduates, both from the results of

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classification, grouping, and association, to be used as material for future considerations in designing and drafting the academic documents, especially curriculum and all matters related to student activities. Recommendations for increasing the competency of graduates from the results of this study include the classification level; the highest branch is scientific communication which most influences the job suitability of graduates. So from these results, college can focus more on preparing the graduates by holding more student activities, such as seminars and workshops on improving students’ scientific communication abilities.

From the results of the model modification, several new differences were obtained when compared to the previous model. Modeling modifications with data mining are carried out to understand problems and provide effective and efficient solutions so that the transfer of graduate competence knowledge is more focused. This model is designed to be easy to use and acceptable to all users. The documentation describes the concept, function, and use of the model. Models are compatible with many popular technologies and platforms. Therefore, further model testing must be carried out to test the effectiveness and success of the design model. Caution is required to deal with changes and increase in their use.

REFERENCES


Enhancing Cloud Security: An Optimization-based Deep Learning Model for Detecting Denial-of-Service Attacks

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Abstract—DoS (Denial-of-Service) attacks pose an imminent threat to cloud services and could cause significant financial and intellectual damage to cloud service providers and their customers. DoS attacks can also result in revenue loss and security vulnerabilities due to system disruptions, interrupted services, and data breaches. However, despite machine learning methods being the research subject for detecting DoS attacks, there has not been much advancement in this area. As a consequence of this, there is a requirement for additional research in this field to create the most effective models for the detection of DoS attacks in cloud-based environments. This research paper suggests a deep convolutional generative adversarial network as an optimization-based deep learning model for identifying DoS bouts in the cloud. The proposed model employs Deep Convolutional Generative Adversarial Networks (DCGAN) to comprehend the spatial and temporal features of network traffic data, thereby enabling the attack detection of patterns indicative of DoS assaults. Furthermore, to make the DCGAN more accurate and resistant to attacks, it is trained on a massive collection of network traffic data. Moreover, the model is optimized via backpropagation and stochastic gradient descent to lessen the loss function, quantifying the gap between the simulated and observed traffic volumes. The testing findings prove that the suggested model is superior to the most recent technology methods for identifying cloud-based DoS assaults in Precision and the rate of false positives.

Keywords—DOS attack; cloud database; generative adversarial networks; attack detection; security threats

I. INTRODUCTION

Computer networks have become more open to cyber-attacks in recent years due to the expansion of web-associated devices and the rising dependence on internet-based administrations [1]. As shown in Fig. 1, the DoS attack, which involves flooding a targeted network or system with malicious traffic to make it inaccessible to authorized users, is one of the most frequent and disruptive attacks [2]. Network administrators and security experts must identify and prevent DoS attacks [3]. Statistical anomaly detection methods or predetermined signatures are frequently used in traditional forms of identifying DoS assaults [4]. However, these techniques have drawbacks like high false-positive rates, a limited ability to react to changing attack strategies, and a failure to identify previously undetected attacks [5]. Researchers have used deep learning techniques to address these issues, which have shown great promise in areas like PC vision, normal language handling, and discourse recognition [6]. Deep learning models have the potential to properly detect and categorize DoS assaults because they can automatically learn complicated patterns and representations from vast amounts of data [7].

In this research, we suggest a deep learning model for identifying DoS assaults that are optimization-based. To increase the Precision and stoutness of DoS attack detection, our model takes advantage of deep neural network technology and integrates it with an optimization framework [8]. There are different advantages to recognizing forswearing of administration (DoS) attacks utilizing any procedure, including traditional strategies or state-of-the-art techniques like profound learning: The detection of DoS attacks makes it possible to detect malicious activity before it affects network resources [9]. Network administrators can safeguard the accessibility of essential network resources and services by detecting DoS attacks [10]. Early detection enables prompt remediation, guaranteeing that authorized users may access the network and preventing them from controlling its resources [11]. DoS assaults can significantly disrupt enterprises, resulting in losses in revenue and harm to their brand [12]. Systems for detecting DoS attacks can assist in locating and reducing harmful traffic, enhancing the entire network’s performance [13]. It is necessary to filter out malicious traffic to optimally employ the available network resources for legal users and enhance response times and network operations [14].
DoS assaults can differ significantly in power, length, and attack methods. Identifying and correctly categorizing these attacks can be challenging, mainly when dealing with new or emerging attack patterns [15]. DoS attacks are being launched more frequently using encrypted traffic, making detecting and examining illicit activity more challenging [16]. While there are many ways to identify denial of service (DoS) assaults using artificial intelligence (AI), one optimization-based deep learning model that has been applied is based on detecting anomalies. The goal of detecting anomalies is locating odd patterns or actions significantly different from typical network data [17]. Various conventional techniques were employed, such as Particle Swarm Optimization-based Probabilistic Neural Network (PSO-PNN) [18], Recurrent Convolutional Neural Network (RC-NN) [19], Lightweight Random Neural Network (LraNN) [20]; however, not all social media platforms support those.

The key contributions of our proposed work are as follows:

- Initially, the datasets are gathered from the standard web source.
- The system was originally trained using the internet application that contained incursion data.
- Then, the pre-handling stage is completed to eliminate the commotion and blunder values.
- A novel Deep Convolutional Generative Adversarial Network (DCGAN) has been implemented with the necessary characteristics and processing stages.
- Subsequently, the feature extractions are done here; the unwanted features are removed.
- More of the wanted features are trained on the attack prediction model to detect the attack and classify the types of attacks based on the parts.
- Finally, the metrics have been validated and compared with another system through Accuracy, F1-score, Recall, and Precision.

The arrangement of this paper is structured as follows. The related work based on detecting DoS attack is detailed in Section II, and the system model and problem statement are elaborated in Section III. Also, the process of the proposed methodology is described in Section IV. Finally, the achieved outcomes are mentioned in Section V. The results are discussed in Section VI and the conclusion about the developed model is detailed in Section VII. Section VIII gives the future work to be conducted in this area.

II. RELATED WORKS

Some of the recent related research works are described below:

Understanding malware’s behaviour across the entire behavioural space significantly improves traditional security. Rabbani et al. [21] propose an original technique to upgrade Cloud specialist organizations’ ability to demonstrate client conduct. Here he applied a PSO-PNN for the detection and recognition process.

The advantage is that it can be used in safety checking and distinguishing unsafe leads. Since genuine users cannot access resources, they might not be able to find the information they need or take the necessary actions.

The Intrusion detection system is built on an Innovative, custom-optimized RC-NN and the Ant Lion optimization technique, which Thilagam et al. [22] proposed for intrusion detection. The advantage is the high accuracy of the IDS classification model, which raises the rate of detection or error rate. The custom-optimized RC-NN-IDS model has a lower error rate of 0.0012 and an improved classification accuracy of 94%. The most significant disadvantage of deploying an IDS is its inability to respond to or stop attacks once detected.

Samriya et al. [23], to increase, generally speaking, cloud-based registering conditions, another hybridization procedure for interruption location frameworks is proposed. Furthermore, this technique aids in dealing with many types of cloud security challenges. The significant advantage is it reduces computational time and enhances accuracy. The downside of host-based IDS is that it cannot detect network threats against the host.

Kushwah et al. [24] present a DoS violence finding system based on a Self-Adaptive Evolutionary Extreme Learning Machine (Sae-ELM) that has been improved. The suggested attack detection system outperforms based on the original SaE-ELM and cutting-edge approaches. They overload the system, causing it to fail.

DoS attacks can affect various equipment and technologies used in network infrastructure. Therefore, security systems must be able to recognize these threats. The majority of the methods that have been previously provided employ an individual machine learning model that can pinpoint DoS attacks; however, it appears that combining a variety of learning models will improve the intrusion detection system’s detection accuracy and reliability.

Majdian, et al. [25] has developed an Adaptive Neuro-Fuzzy Inference System (ANFIS) to enhance DoS attack detection accuracy compared to existing techniques.

The most alluring invention in the current landscape is probably cloud computing. Lowering the massive upfront cost of buying equipment foundations and processing power provides an expense-effective arrangement. By utilizing a portion of the work that is not registered, decreasing the reaction time at the edge devices of the end client, such as IoT, fog computing provides extra aid to cloud infrastructure.

Sattari, et al. [26] have developed a Software-defined network (SDN) that identifies 99.98% of sophisticated multi-variant bot attacks more accurately than earlier ones, with more excellent performance. Table I lists the difficulties with the existing works.

Ferrag et al. [31] developed a unique Weight-based Ensemble Machine Learning Algorithm (WBELA) to detect aberrant signals in a CAN bus network. Then, he creates a model based on many-objective optimization for CAN bus network intrusion detection. It considerably improves Precision, lowers the false positive rate, and outperforms other
approaches. The issue is that the intrusion detection model is not near to the actual CAN bus security protection.

<table>
<thead>
<tr>
<th>Sl. No</th>
<th>Author Name</th>
<th>Method</th>
<th>Advantages</th>
<th>Disadvantages</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>Rabbani et al. [21]</td>
<td>PSO-PNN</td>
<td>The advantage is that it is promising for security checking and distinguishing unsafe leads.</td>
<td>Premature convergence susceptibility, weak local optimization skills</td>
</tr>
<tr>
<td>2</td>
<td>Thilagam et al. [22]</td>
<td>Recurrent Convolutional Neural Network (RC-NN)</td>
<td>Each pattern can be presumed to be dependent on earlier ones, thanks to the ability of RNN to simulate a collection of records.</td>
<td>The position and orientation of objects are not encoded.</td>
</tr>
<tr>
<td>3</td>
<td>Samriya et al. [23]</td>
<td>Hybridization Technique</td>
<td>high sensitivity, exact anatomical localization, and quantification potential.</td>
<td>Low rate of good qualities recombining.</td>
</tr>
<tr>
<td>4</td>
<td>Kushwah et al. [24]</td>
<td>SaE-ELM</td>
<td>Reduce the training time.</td>
<td>Incorrect data interpretation</td>
</tr>
<tr>
<td>6</td>
<td>Sattari , et al. [26]</td>
<td>Software-defined network (SDN)</td>
<td>It enables engineers to reroute networks instantly.</td>
<td>Cost increase brought on by reconfiguration.</td>
</tr>
<tr>
<td>7</td>
<td>Ferrag et al. [31]</td>
<td>weight-based ensemble machine learning algorithm (WBELA)</td>
<td>Enhancing Predictive Performance</td>
<td>Growing Complexity</td>
</tr>
</tbody>
</table>

III. SYSTEM MODEL

Identifying DoS threats to improve cloud security is crucial. DoS attacks are intended to overburden cloud servers and networks, rendering them inoperable and preventing authorized users from accessing cloud resources. The detection of DoS assaults in cloud systems has several issues: Large volumes of traffic from DoS assaults might be challenging to discern from regular traffic. It can make it difficult to detect and respond to DoS assaults quickly. DoS assaults can spread over numerous computers and networks, making it challenging to pinpoint their origin. The user initiates a request by sending a specific command or action to access a service, browse a website, or communicate with another user. The request is transmitted over the internet. The request reaches the load balancer, which serves as a central entry point for incoming network traffic. The load balancer analyzes the request and determines the most appropriate target server to handle it based on factors such as server health, availability, or predefined load balancing algorithms. Before the request reaches the target server, it passes through the firewall. The firewall examines the request, checking if it complies with predefined security policies and rules. It blocks unauthorized or potentially malicious traffic, protecting the network from potential threats and attacks. Once the request passes through the firewall, it reaches the target server responsible for processing the request. The server performs the necessary operations based on the user's request, retrieves the requested resource or data, and prepares a response to be sent back to the user. The response generated by the target server passes through the firewall. The firewall ensures the response is secure and free from any malicious content or unauthorized data. The load balancer receives the response from the target server and ensures its integrity and consistency.

It may also perform additional processing or optimization before forwarding the response to the user. The load balancer sends the response back to the user through the internet. The response travels through the network, passing through routers and switches, until it reaches the user's device. The system model is shown in Fig. 2.

Cloud environments are complex and dynamic, with multiple infrastructure, software, and service layers. This complexity can make it challenging to monitor and detect DoS attacks. Monitoring systems may generate false positives, identifying legitimate traffic as malicious and causing unnecessary disturbances in cloud services. So, the presented work has aimed to develop an optimized deep feature-based detecting mechanism for the process.

IV. PROPOSED METHODOLOGY

A novel DCGAN has been developed to detect harmful behaviour in cloud applications. The cloud application containing intrusion data has been purchased and put into the system to evaluate the proposed model. The proposed model employs DCGAN to comprehend network traffic data's
temporal and spatial characteristics, thereby enabling the detection of patterns indicative of DoS assaults. Fig. 3 shows the proposed architecture of DCGAN.

Fig. 3. The proposed DCGAN architecture.

Initially, the datasets are gathered from the authoritative web source. The system was originally trained using the internet application that contained incursion data. Then preprocessing phase is conceded to eliminate the noise and error values. A novel Deep Convolutional Generative Adversarial Network (DCGAN) has been implemented with the necessary characteristics and processing stages. Subsequently, the feature extractions are done here; the unwanted features are removed. More of the wanted features are trained to the attack prediction model to detect the attack and classify the types of attack based on the parts. Finally, the metrics have been validated and compared [32-34] with another system through Accuracy, F1-score, Recall, and Precision.

A. Pre-Processing

Pre-process the collected data by cleaning, normalizing, and transforming it into suitable input formats for training the deep learning model [27]. It may involve feature extraction, data augmentation, or encoding techniques specific to DCGANs. The initialization equation is expressed in Eq. (1).

\[ A(y) = (v_1, v_2, v_3, \ldots, v_n) Y \]  

(1)

Where \( A \) denotes the collected cloud dataset from the dataset, \( Y \) indicates the information present in the dataset, \( Y \) denotes the features available in the dataset and \( n \) represents the total count of data currently in the database [27]. After initialization, the next stage is the pre-processing phase. Here, the input dataset is pre-handled to eliminate the null values present in the dataset. The pre-processing equation is represented in Eq. (2).

\[ H^*_B (A) = \frac{1}{n} \sum_{i=1}^{l} \| Y^*_i - L^*_i \|^2 \]  

(2)

Where \( H^*_B \) denotes the pre-processing variable and \( L^*_i \) indicates the null values present in the dataset. Fitness also plays a role in the decision to virtualize the task without compromising the quality of the work. The GAN fitness function creates new data instances that resemble your training data. The fitness function attains less value for an effective result.

B. Proposed DCGAN

The features are given to the proposed DCGAN-based convolution neural network classifier for detecting DoS attacks after being appropriately chosen. DoS assaults are often found using intrusion detection systems and network monitoring tools, which can spot changes in network traffic that might indicate a DoS attack. The optimized via backpropagation and stochastic gradient descent model to improve the gap between the simulated and observed traffic volumes [28] [33]. Machine learning models, such as DCGANs, may help these detection efforts by examining network traffic patterns and spotting potential threats. The proposed DCGAN is expressed in Eq. (3).

\[ \text{lowhighM}(J, R) = E^*_{\text{ndata}}(y) [\log J(x^*)] + E^*_{\text{qc}} \left[ \log (1 - J(N(C))) \right] \]  

(3)

Where \( x^* \) denotes the dataset, \( c \) is the generator’s blare at the input of the attacks, \( n \text{ data} \) is the data delivery, \( qc \) is the noise delivery, \( N \) is the maker of the cloud attacks and \( J \) is the discriminator of the DOS attacks.

C. Feature Extraction

The constructed model then moves on to feature extraction after the suggested technique. The valuable characteristics are extracted, and the unnecessary features are removed in the feature extraction module. Consider \( Y^*_i \) as the feature present in the pre-processed dataset [27]. The pre-processed dataset contains relevant features \( F^*_i \) and irrelevant features \( F^* \). The feature extraction equation is expressed in Eq. (4).

\[ G^*_A (A) = \rho \times Y^*_i [F^*_i - F^*_i] \]  

(4)

Only pertinent characteristics remain in the dataset after the feature extraction stage. The dataset is then trained to recognize assaults using the suggested methodology.

D. Attack Detection and Classification

To identify known and unknown assaults, deep convolution neural networks are employed to detect noise and other undesirable aspects in the data. The neural network is used to reduce the dimensionality of the data during the training phase. The threshold for conventional traffic data is manually selected, despite the suggested approach lessening the data complexity. Given that it does not clearly show the network abnormality or the sort of attack, the model may be able to identify the device connected to the anomaly, making it suitable for fault detection. An assault that is known to have occurred is indicated in the dataset. Here, the system is trained to recognize attacks in the dataset using an attack trained on a massive collection of network traffic data. Eq. (5) [27] gives the equation for attack detection.

\[ P^*_B (A) = \begin{cases} S_Q = 1; & \text{Normal} \\ S_Q = 0; & \text{Attack} \end{cases} \]  

(5)
Where \( P_B^{UI} \) denotes the trained features, attack features are marked as “0” in this instance, while the standard element in the dataset is designated as “1”. As a result, the assaults in the dataset are found [27]. Additionally, to see if the system recognizes and categorizes unidentified attacks like Denial of Service (DoS) launched into the system. The unknown attack detection is expressed in Eqn. (6).

\[
P_B^{UI}(A) = \begin{cases} 
    \text{Normal} & \text{if } A_{on} < 80 \text{ Mbs} \\
    \text{Attack} & \text{otherwise}
\end{cases}
\]

(6)

Where \( P_B^{UI} \) denotes the unknown attack detection function, \( A_{on} \) indicates overloading data rate. Based on the speed of data overloading, the standard and attack features are identified here. When data overloading rates are above 80Mbs, an assault is recognized. If the rate of data overloading is under 80Mbs, it is recognized as a typical feature [27]. Attack classification is completed after attack detection. The attack classification is expressed in Eq. (7).

\[
P_D^{UI}(A) = \begin{cases} 
    \text{DoS; } A_{on} \geq 100 \text{ Mbs} & \text{if } 80 \text{ Mbs} > A_{on} < 100 \text{ Mbs} \\
    \text{Other} & \text{otherwise}
\end{cases}
\]

(7)

Where \( P_D^{UI} \) denotes the attack type classification function. The other attack’s data overloading range is between 80Mbs and 100Mbs. Similarly, if the data overloading rate is more than 100 MB, the attack is classified as a DoS attack. The workflow of the DCGAN approach is shown in Fig. 4.

V. RESULT AND DISCUSSION

The presented method is implemented on the PYTHON platform using an Intel I Core I i5 CPU and 8GB RAM. This section discusses the experimental design and the efficacy of the proposed approach. Several measures, including Accuracy, Precision, Recall, and F-measure with the help of optimization and neural networks, are used to assess the organization’s effectiveness.

Case Study: This case study discusses a deep convolution GAN network for DOS attacks to enhance cloud security. The cloud application with intrusion data has been bought and installed in the system. The noise and error values are then removed during the pre-processing step. The proposed system’s cloud datasets are gathered from an authoritative web source. Then, cleanse, normalize and convert the obtained data into appropriate input formats before using it to train the deep learning model after carefully selecting the suggested DCGAN-based convolution neural network classifier to detect DoS attacks. And then are the feature extractions carried out; the undesirable elements are eliminated. Additionally, the desired features are trained to help the attack prediction model identify and categorize attacks based on their characteristics.

Fig. 5 shows the overall process of GAN.0.

A. Performances Metrics

The performance metrics are evaluated in terms of Accuracy, Recall, F-measure, and Precision to validate our proposed DCGAN model.

1) Accuracy measure: According to this definition, accuracy is the proportion of outcomes accurately predicted to all the predictions. The machine learning algorithms’ performance parameter is most frequently employed [31] [35]. Accuracy is technically defined as the proportion of accurate positive to accurate negative discoveries needed to finish the machine learning model’s outputs. Accuracy is expressed in equation (8),

\[ A = \frac{TP + TN'}{TP' + TN + FP' + FN'} \]

(8)
Where, \( A \) indicates the accuracy, \( TP^* \) denoted as true positive, \( TN^* \) is genuinely harmful, \( FP^* \) indicates the false positive, \( FN^* \) is false negative.

2) **Precision calculate**: By dividing the total of true positives by all optimistic forecasts, Precision can be used to assess whether the model’s optimistic predictions are accurate [31]. Precision is expressed in Eq. (9),

\[
P = \frac{TP^*}{TP^* + FP^*} \tag{9}
\]

3) **Recall measure**: The Recall is the percentage of positives the model identified adequately out of all potential positives by dividing true positives by the total number of actual positives [31][36]. Recall is expressed in equation (10),

\[
R = \frac{TP^*}{TP^* + FN^*} \tag{10}
\]

4) **F-measure**: F-measure represents a weighted average of Recall and Accuracy. The F-measure considers both positive and negative results to keep the balance between Recall and Precision [31][37]. F-measure is expressed in equation (11),

\[
F\text{-measure} = \frac{2(P+R)}{P+R} \tag{11}
\]

Where P represents the Precision, R represents the Recall.

**B. Evaluation of Performance**

Comparing the constructed model's metrics for Accuracy, Precision, F-measure, and recall to those of other models confirmed its efficacy. The developed model will be implemented in the Python framework. Moreover, the existing techniques like You Only Look Once a Multilayer Perceptron (MLP) [29], Convolutional Neural Network (CNN) [29], Decision Tree (DT) [30], and Support Vector Machine (SVM) [30].

1) **Comparison of the suggested with other existing techniques in terms of accuracy**

Fig. 6 shows the accuracy of the suggested DCGAN is compared to that of the previous study. The proposed DCGAN accuracy rate is 0.997. While comparing the suggested DCGAN to other existing methods, the accuracy level is more significant. The accuracy rate for the DT method currently in use is 0.86. At the same time, the accuracy level is high compared to other existing methods like MLP, CNN, and SVM.

Compared to other approaches like SVM and CNN, the MLP accuracy level is 0.987, which is a great value. The CNN precision level is 0.959, which is poor compared to all other methods currently in use. DT precision level is 0.9159 compared to other techniques like MLP, CNN, and DT. As a result, the proposed DCGAN performs superior to the other methods already in use.

2) **Comparison of the suggested with other existing techniques in terms of Precision**

In Fig. 7, the Precision of the suggested DCGAN is compared to that of the previous study. The proposed Convolution-based Buffalo Optimization (DCGAN) precision rate is 0.988. While comparing the suggested DCGAN to other existing methods, the precision level is more significant. The precision rate for the SVM method currently in use is 0.782. At the same time, the precision level is high compared to other existing methods like MLP, CNN, and SVM.

Compared to other approaches like CNN, DT and the MLP precision level is 0.968, which is a great value. The CNN precision level is 0.959, which is poor compared to all other methods currently in use. DT precision level is 0.9159 compared to other techniques like MLP, CNN, and DT. As a result, the proposed DCGAN performs superior to the other methods already in use.
3) Comparison of the suggested with other existing techniques in terms of F1-score

In Fig. 8, the F1-score of the suggested DCGAN is compared to that of the previous study. The proposed Convolution-based Buffalo Optimization (DCGAN) F1-score rate is 0.978. While comparing the suggested DCGAN to other existing methods, the F1-score level is more significant. The F1-score rate for the DT method currently in use is 0.7239. At the same time, the F1-score level is high compared to other methods like MLP, CNN, and SVM.

Compared to other approaches like CNN, DT and the MLP F1-score level is 0.968, which is a great value. The CNN precision level is 0.956, which is poor compared to all other methods currently in use. SVM precision level is 0.852 compared to other techniques like MLP, CNN, and DT. As a result, the proposed DCGAN performs superior to the other methods already in use. The comparison of techniques in terms of accuracy, precision, recall and F1-score data’s are mentioned in Table II.

<table>
<thead>
<tr>
<th>Technique</th>
<th>Accuracy</th>
<th>Precision</th>
<th>Recall</th>
<th>F1-score</th>
</tr>
</thead>
<tbody>
<tr>
<td>MLP</td>
<td>0.987</td>
<td>0.968</td>
<td>0.953</td>
<td>0.96</td>
</tr>
<tr>
<td>CNN</td>
<td>0.986</td>
<td>0.959</td>
<td>0.954</td>
<td>0.956</td>
</tr>
<tr>
<td>DT</td>
<td>0.86</td>
<td>0.9159</td>
<td>0.6808</td>
<td>0.7239</td>
</tr>
<tr>
<td>SVM</td>
<td>0.92</td>
<td>0.782</td>
<td>0.936</td>
<td>0.852</td>
</tr>
<tr>
<td>Proposed DCGAN</td>
<td>0.997</td>
<td>0.988</td>
<td>0.96</td>
<td>0.978</td>
</tr>
</tbody>
</table>

4) Comparison of the suggested method with other existing techniques in terms of Recall

In Fig. 9, the Recall of the suggested DCGAN is compared to that of the previous study. The proposed Convolution-based Buffalo Optimization (DCGAN) recall rate is 0.96. While comparing the suggested DCGAN to other existing methods, the F1-score level is more significant. The recall rate for the DT method currently in use is 0.6808. At the same time, the recall level is high compared to other existing methods like MLP, CNN, and SVM.

Compared to other approaches like MLP DT, the CNN recall level is 0.954, which is a great value. MLP recall level is 0.953, which is poor compared to all other currently used methods. The SVM recall level is 0.936 when compared to other techniques like MLP, CNN, and DT. As a result, the proposed DCGAN performs superior to the other methods already in use.

Comparative analysis with other machine learning approaches, such as MLP, CNN, DT, and SVM, reveals the superior performance of the DCGAN model in terms of precision, recall, and F1-score. This comparative investigation showcases the advancement of the proposed model over existing techniques and highlights its potential for practical deployment in real-world cloud environments. The DCGAN model's improved precision, robust pattern recognition capabilities and superior performance in comparison to other methods contribute to the enhancement of cloud security measures and offer promising prospects for further advancements in the field.

VI. DISCUSSION

The presented IDS approach was designed and verified in the Python tool. The essential goal of the created model is to distinguish pernicious occasions in the WS organization. The neural network in the proposed approach improves the detection rate and accuracy.

Table III summarizes the performance study of the provided model. For the DCGAN dataset, the designed model attained 0.997 accuracies and 0.988 precision, a recall of 0.96, and an F-score of 0.978 in the developed model. Additionally, findings are compared to current approaches to confirm the great Precision in the calculation period of the developed model. The findings highlight the significance of the DCGAN model in enhancing cloud security by accurately detecting DoS attacks. The improved precision and reduced false positive rate...
contribute to better protection against service disruptions, data breaches, and financial losses. The study demonstrates the potential of deep learning approaches, specifically DCGANs, in tackling complex security challenges in cloud environments. The results provide valuable insights for researchers and practitioners working on developing effective mechanisms for detecting and mitigating DoS attacks. By addressing the limitations and pursuing future research avenues, the study contributes to advancing the knowledge and practical applications of cloud security.

### TABLE III. PERFORMANCE ANALYSIS

<table>
<thead>
<tr>
<th>Metrics</th>
<th>Performance</th>
</tr>
</thead>
<tbody>
<tr>
<td>F-score</td>
<td>0.978</td>
</tr>
<tr>
<td>Recall</td>
<td>0.96</td>
</tr>
<tr>
<td>Precision</td>
<td>0.988</td>
</tr>
<tr>
<td>Accuracy</td>
<td>0.997</td>
</tr>
</tbody>
</table>

The article introduces a novel approach using a DCGAN as an optimization-based deep learning model for identifying DoS attacks in the cloud. This represents a significant contribution to the field by exploring the effectiveness of DCGANs in detecting such attacks. The proposed model leverages DCGAN to comprehend the spatial and temporal features of network traffic data. By analyzing these features, the model can identify patterns indicative of DoS attacks, thereby enhancing the accuracy of detection. To improve the accuracy and robustness of the DCGAN model, it is trained on a large collection of network traffic data. This approach enables the model to learn from diverse attack scenarios and enhances its ability to detect previously unseen attacks. The model is optimized through back propagation and stochastic gradient descent to minimize the loss function, which quantifies the discrepancy between simulated and observed traffic volumes. This optimization process enhances the model's performance and fine-tunes its ability to detect DoS attacks effectively. The article provides testing findings that demonstrate the superiority of the proposed DCGAN model over recent technology methods in terms of precision and the rate of false positives. This comparative evaluation contributes valuable insights into the effectiveness of the DCGAN model for DoS attack detection in cloud environments.

### VII. CONCLUSION

By effectively identifying DoS attacks, the optimization-based deep learning model, which utilizes techniques like DCGANs, has the potential to enhance cloud security. The model leverages deep learning to detect anomalies and learn intricate patterns within network traffic data. In this study, the model's performance is evaluated in terms of Recall, Precision, and F1 score for DoS attack detection. A controlled simulation environment is employed to test the model's efficacy in detecting DoS attacks. The obtained results are then compared with existing machine learning approaches such as MLP, CNN, DT, and SVM. The comparative analysis reveals that the developed DCGAN model surpasses other models in terms of performance. Notably, the Accuracy, Precision, Recall, and F1-score values exhibit significant improvements in the proposed DCGAN technique, achieving enhancements of 0.997, 0.988, 0.96, and 0.978, respectively. These results indicate the superiority of the DCGAN model in accurately identifying DoS attacks compared to alternative approaches. Furthermore, in addition to the empirical findings, this study contributes to the field by introducing newly formulated theoretical advancements. The utilization of DCGANs for DoS attack detection in cloud environments represents a novel approach with potential implications for enhancing cloud security. The incorporation of deep learning techniques and the specific architecture of DCGANs contribute to the model's ability to learn complex patterns and identify subtle anomalies in network traffic data. The study may have been limited by the availability of a specific dataset for training and testing the DCGAN model. The results may vary when applied to different datasets or real-world scenarios. The focus of the study was on detecting DoS attacks. The model's performance in detecting other types of attacks or more complex attack patterns was not explored. The study primarily focused on the performance evaluation of the DCGAN model in a controlled environment. The real-time implementation and assessment of the model's effectiveness in practical cloud environments were not addressed. The study thus offers theoretical insights into the application of DCGANs for cloud security and lays the foundation for further research in this area.

### VIII. FUTURE WORK

In future it would be sought to enhance the DCGAN model's ability to detect more sophisticated and evolving DoS attack techniques. This involves exploring novel network traffic features, developing more robust anomaly detection algorithms, and incorporating machine learning techniques to improve the accuracy and effectiveness of the model. Also by integrating the DCGAN model with existing cloud security infrastructure, such as intrusion detection systems, firewalls, or security incident response platforms. This integration can enable a comprehensive security ecosystem, leveraging the strengths of multiple security mechanisms to provide a more holistic and proactive defense against DoS attacks.

### REFERENCES


Inspection System for Glass Bottle Defect Classification based on Deep Neural Network

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King Mongkut’s University of Technology Thonburi, Pracha Uthit Road, Bang Mot Sub-District, Thung Khru District Bangkok 10140, Thailand²

Abstract—The problem of defects in glass bottles is a significant issue in glass bottle manufacturing. There are various types of defects that can occur, including cracks, scratches, and blisters. Detecting these defects is crucial for ensuring the quality of glass bottle production. The inspection system must be able to accurately detect and automatically determine that the defects in a bottle affect its appearance and functionality. Defective bottles must be identified and removed from the production line to maintain product quality. This paper proposed glass bottle defect classification using Convolutional Neural Network with Long Short-Term Memory (CNNLSTM) and instant base classification. CNNLSTM is used for feature extraction to create a representation of the class data. The instant base classification predicts anomalies based on the similarity of representations of class data. The convolutional layer of the CNNLSTM method incorporates a transfer learning algorithm, using pre-trained models such as ResNet50, AlexNet, MobileNetV3, and VGG16. In this experiment, the results were compared with ResNet50, AlexNet, MobileNetV3, VGG16, ADA, Image threshold, and Edge detection methods. The experimental results demonstrate the effectiveness of the proposed method, achieving high classification accuracies of 77% on the body dataset, 95% on the neck dataset, and an impressive 98% on the rotating dataset.

Keywords—Convolution neural network; glass bottle; defect detection; long shot-term memory; inspection machine

I. INTRODUCTION

Defect detection plays a vital role in glass bottle production, safeguarding product quality, consumer safety, brand reputation, and value. By investing in robust flaw detection systems and processes, manufacturers can maintain impeccable standards, safe, and visually appealing glass bottles to their customers [1,2]. Common types of defects that can occur in glass bottles include: 1) Stones are foreign stone grains embedded in the glass, degrading the quality of the bottle. 2) Tears are deformed breaks or fractures on the surface of the bottle. 3) Blisters are raised or swollen areas caused by uneven cooling during manufacturing, affecting both strength and appearance. 4) Cracks are breaks in the surface that compromise the structural integrity of bottle.

The inspection machine uses various technologies to detect defects, including cameras, lasers, and sensors. The inspection machine uses cameras to capture images of the products, and the software analyzes these images to detect defects. When defects are detected, the software sends a signal to the control system to remove the faulty product from the production line.

As for the problem with traditional inspection machines, they cannot inspect defects in complex areas and require the use of imaging techniques to visualize the defects. Because these machines utilize image processing techniques and require various parameter settings to be adjusted by users for defect detection. It results in the inability to completely remove defective bottles from the production line, undermining confidence in the manufacturing process. The detection method on inspection machine for detecting the defects must be accurate, precise, capable of recognizing defect well, and fast in the learning process of defect patterns.

This research proposes a new method for detecting defects in glass bottles using a deep neural network for extracting deep features and instance-based classification. The method uses CNN combined with LSTM to recognize distinctive features of defects and extract those features. The training process is designed to create a set of CNNLSTM models with fewer training iterations, helping to reduce the training time. The instance-based classification is used to classify the defect in the images. It reduces parameter settings by users for defect detection, resulting in a reduction in user workload.

The rest of this paper is organized as follows: Section II presents a background study of glass bottle defect detection techniques. Section III explains the glass bottle defect dataset and describes the proposed glass bottle defect classification method. Section IV explains the evaluation methods, the experimental setup, and the experimental outcomes; and finally, Section V suggests directions for future works and concludes the paper.

II. RELATED WORKS

The existing defect detection techniques mainly focused on the defect in glass bottles. The related methods are as follows. Latina et al. [3] presented defect detection method for detecting defects in glass bottles for the purpose of reusing them and highlights the limitations of the manual inspection in micro, small, and medium enterprises (MSMEs). The research introduced a cost-effective deep learning-based method using the SSD MobileNetV2 model to detect various defects in glass bottles. The method used transfer learning and data augmentation techniques to achieve high accuracy, with up to
98.07% overall system accuracy. Gong et al. [4] presented a machine vision system designed for the automatic online inspection of defects in transparent labels on curved glass bottles. The system used an area-array camera and a custom-made blue dome illumination device to capture high-quality still images by minimizing reflection. To address the challenge of distorted curved geometry, a deformable template matching method was employed for precise defect location. The method also included an adaptive threshold selection strategy that effectively detected small scratches by using global and local threshold values along with a Gaussian fitting algorithm. Additionally, techniques such as skeleton extraction and distance transformation were applied to detect the complete edge contour of Chinese characters with a special font, considering the golden edge printing error. Field tests demonstrated an impressive detection accuracy of 99.5% at a speed of 60 bottles per minute, covering over 60,000 bottles. Vitis et al. [5] proposes an algorithm that achieves high detection accuracy while significantly reducing processing time. By using adaptive thresholding and analyzing luminous intensity variations, the algorithm effectively detects blob and airliner defects while mitigating the impact of tube curvature, rotation, and vibration. Comparative evaluations demonstrate an 86% reduction in processing time, a 268% increase in throughput, and improved detection accuracy compared to existing methods. The algorithm also incorporates Region of Interest reduction techniques and a tuning procedure for parameter adjustment during production batch changes. The performance of the algorithm was assessed in a real environment, and it successfully identified misclassified tubes, suggesting its practical applicability. Zhang et al. [6] suggested a machine learning-based acoustic defect detection (LearningADD) system to replace manual inspection. The system used an improved Hilbert-Huang transform (HHT) to extract features from acoustic signals and a Shuffled Frog Leaping Algorithm (SFLA) to select features. Five deployment strategies were compared and optimized to improve real-time performance. The LearningADD system was validated using data from a real-life beverage factory. The F-measure of the system reached 98.48%. The proposed deployment strategies were verified using experiments on private cloud platforms. The Distributed Heavy Edge deployment strategy outperformed other strategies, with a defect detection time of less than 2.061 seconds for 99% of bottles. Zhou et al. [7] proposed a surface defect detection framework, which consisted of three main components. Firstly, a novel localization method called entropy rate superpixel circle detection (ERSCD) was introduced. It combined least-squares circle detection, entropy rate superpixel (ERS), and an improved randomized circle detection to accurately identify the region of interest (ROI) on the bottle bottom. The ROI was then divided into two measurement regions: the central panel region and the annular texture region. For defect detection in the central panel region, a method named frequency-tuned anisotropic diffusion super-pixel segmentation (FTADSP) was proposed. It integrated frequency-tuned salient region detection (FT), anisotropic diffusion, and improved superpixel segmentation to accurately detect defect regions and boundaries. For defect detection in the annular texture region, a strategy called wavelet transform multiscale filtering (WTMF) was proposed. It employed wavelet transform and a multiscale filtering algorithm to reduce texture influence and enhance robustness to localization errors. Zhou et al. [8] presented a new apparatus for real-time bottle bottom inspection. The apparatus used a combination of Hough circle detection and size prior to locating the bottom of the bottle. The region of interest was then divided into three measurement regions: central panel region, annular panel region, and annular texture region. A saliency detection method was used to find defective areas in the central panel region. A multiscale filtering method is used to search for defects in the annular panel region. Template matching was combined with multiscale filtering to detect defects in the annular texture region.

The problem with traditional methods of detecting anomalies on glass bottles using image processing is the difficulty in detecting anomalies, especially when there are color and shape variations on the bottles. Inconsistent lighting conditions also pose challenges in detecting anomalies. The disadvantage of using deep neural networks for detecting counterfeit on glass bottles is the complexity involved in building and training the network. Deep neural network-based methods require a large amount of data and longer training time. Additionally, fine-tuning and selecting appropriate parameters for deep networks can be challenging.

III. METHODOLOGY

A. Glass Bottle Dataset

There are two main categories of glass bottle inspection machines: straight glass bottle inspection machines and rotary glass bottle inspection machines. Fig. 1 shows an example of an inspection machine for glass bottles. The straight glass bottle inspection machine is an automated system that utilizes advanced technologies, such as computer vision and image processing, to detect and identify defects in glass bottles. It captures images of the bottles from different angles and analyzes them in real-time to ensure product quality and prevent issues during production and transportation. The straight glass bottle inspection machine is depicted in Fig. 1(a). On the other hand, the rotating glass bottle inspection machine is a specialized automated system designed specifically for inspecting glass bottles using a rotating mechanism. By rotating the bottles, it enables a comprehensive examination of the entire surface, ensuring a high level of accuracy in defect detection. The rotating glass bottle inspection machine is shown in Fig. 1(b).

The bottle inspection machine utilizes a vision camera-based inspection method for fully automated visual inspection of glass bottles. The process is as follows. Set the bottles in the designated area on the conveyor belt of the bottle inspection machine. The bottles are properly aligned and spaced to ensure accurate and consistent imaging. Provide proper lighting conditions for capturing clear and well-illuminated bottle images. Set up the camera parameters such as focus, exposure, and white balance to optimize image quality and clarity. Initiate the image capture process using the bottle inspection machine to trigger the camera to capture images of the bottles as they pass through the inspection area. The machine utilizes two cameras to capture images of the bottle: one camera is focused on the neck, while the other is focused on the body of
the bottle. Fig. 2 illustrates the composition of the side wall of glass bottles; (a) displays images from the neck dataset, (b) provides an example of the body dataset, and (c) showcases sample bottle images from the rotating dataset. The body dataset for glass bottles comprises 571 images, with 368 images showing defects and 203 images without defects. Similarly, the neck dataset consists of 570 images, including 250 images with defects and 320 images without defects. As for the rotating dataset, it comprises 120 images, with 70 images displaying defects and 50 images representing undamaged bottles. Each image in the body and neck datasets is standardized to a size of 900x800 pixels (width x height), ensuring consistency across the dataset. On the other hand, the images in the rotating dataset are resized to a dimension of 1024x800 pixels (width x height). The captured images are transmitted to the detection system for analysis. If any defects, such as stones, tears, or blisters, are identified on the side wall of a bottle, the detection system generates a signal to reject the faulty bottle. This process ensures that only bottles without defects continue downstream in the production line. Fig. 3 (a) is shown tear defect, (b) is a blister, and (c) is stone defect on slide wall of bottle. These defects can appear throughout the bottle.

B. The Proposed Method

This section describes the proposed defect detection approach, which is divided into three parts: 1) The network architecture of CNNLSTM for extracting deep features in images, 2) the training process for creating a set of CNNLSTM models and computing the representation of class data, 3) the classification approach that suggests using the extracted deep features from unseen bottle image and applying the distance weight method to classify defect or normal glass bottle images.

![LSTM Diagram](https://via.placeholder.com/150)

LSTM comprises multiple memory cells, each consisting of three essential elements: write, read, and forget (delete). During each time step \((t)\), the forget gate unit is updated according to the following process,

\[
\begin{align*}
    f_t &= \sigma_g(W_f x_t + U_f h_{t-1} + b_f), \\
    i_t &= \sigma_g(W_i x_t + U_i h_{t-1} + b_i), \\
    o_t &= \sigma_g(W_o x_t + U_o h_{t-1} + b_o), \\
    c_t &= f_t \odot c_{t-1} + i_t \odot \sigma_c(W_c x_t + b_c), \\
    h_t &= \sigma_h(c_t)
\end{align*}
\]

where \(x_t\) is the input data, \(f_t\) is the forget gate, \(i_t\) is the input/update gate, \(o_t\) is the output gate, \(c_t\) is the cell state
vector, \( \sigma \) is the sigmoid activation function, \( \odot \) is the Hadamard product (element-wise product), and \( h_2 \) is the output of the LSTM unit. In the final representation layer, an LSTM, which is a Recurrent Neural Network (RNN) suitable for sequential data, receives inputs from previous convolutional layers. The LSTM produces a set of 512 outputs, which are then passed to the next layer. The outputs of the convolutional layers are subjected to Batch-Normalization (BN) before they are fed into the LSTM layer. The fully connected layers contain 1,000 hidden neurons. The activation function of all the convolutional layers (with a max pooling size of \( 2 \times 2 \)) is a Rectified Linear Unit (ReLU) [11]. Deep features are extracted from these layers. The output layer contains one output neuron with a sigmoidal activation function. The size of the input images is set as \( 224 \times 224 \times 3 \) pixels. The CNNLSTM is trained using a Stochastic Gradient Descent (SGD) method [12]. The learning rate is set to 0.01, and the batch size is 8 images. The CNNLSTM neural network architecture is shown in Fig. 5.

![Fig. 5. CNNLSTM.](image)

2) The proposed learning process: Our training process aims to minimize the similarity distance between the extracted deep feature vectors and the centroid of the corresponding class with the same label. Simultaneously, it maximizes the similarity distance between the centroid of good bottles and the centroid of defective bottles.

The overall learning process of creating a set of CNNLSTM and representing class data is proposed in this section. The process of creating a set of CNNLSTM models and generating representations of class data is illustrated in Algorithm 1. The inputs of the learning process include a training dataset \( S \), a specified number of sub-training sets \( K \), a threshold value \( \theta \), and the pre-trained weight \( m_{k-1} \). The parameter \( K \) is a parameter used to determine the number of sub-training sets and the number of CNNLSTM models, which needs to be appropriately adjusted. The value of \( K \) affects the number of samples in each sub-training set used for training CNNLSTM and the diversity of CNNLSTM models. \( \theta \) is a parameter used to define the initial performance of CNNLSTM in each training iteration. The pre-trained weight \( m_{k-1} \) helps to reduce training time and resource requirements, improve performance, and enable the model to perform well in limited data conditions.

**Algorithm 1: Pseudo code of the feature learning network method.**

**Input:**
- \( S \): A training bottle dataset: \( \{(x_1, y_1), \ldots, (x_N, y_N)\} \)
- \( K \): The number of the sub datasets
- \( \theta \): A performance threshold value
- \( m_{k-1} \): The pre-trained model

**Output:**
- \( E \): A set of CNNLSTM models
- \( C \): A set of representation of class data

1: Randomly split the training dataset into \( K \) sub datasets:
   \[ S \rightarrow \{s_1, s_2, s_3, \ldots, s_K\}. \]
2: For \( k = 1, 2, \ldots, K \) do:
3: \( \text{MSE} = 0 \) and \( m_k = m_{k-1} \)
4: While \( \text{MSE} < \theta \) do:
5: \( \text{Train model } m_k \) with \( s_k \)
6: Extract features \( F_k \) in images by \( m_k \).
7: Normalize all features in \( F_k \) with an L2-norm technique.
8: For \( i \in y \in \{0,1\} \) do:
9: \( \text{Compute the representation of each class data } c_{ik} \) from \( F_k \).
10: End For
11: Evaluate the distances from centroid \( c_{ik} \) to every feature of all \( F_k \) instances.
12: Predicted a bottle is normal or abnormal (\( h \)) is based on the minimum distance between \( c_{ik} \) and \( F_k \in E \).
13: Compute means squared error \( \text{MSE} \) from \( h \) with true label \( y \).
14: End While
15: \( m_k \in E \) and \( c_{ik} \in C \).
16: End For

To construct a set of CNNLSTM models, our method involves randomly selecting images in the \( S \) training dataset and organizing them into several partitions or subsets, i.e., \( S \rightarrow \{s_1, s_2, s_3, \ldots, s_K\} \). The purpose of creating these sub-datasets is to foster diversity in CNNLSTM models and generate representation vectors for both normal bottles and bottles with defects.

The current \( s_k \) is the training bottle images for a CNNLSTM model, \( m_k \). Let the set models is ensemble \( E \). The convolutional layer of the current model uses pre-train weight of previous round. The output of training with five epochs is \( m_k \) (Step 5).

In Step 6, deep features vectors \( F_k \) are extracted from the images in \( s_k \). This process involves removing the output layer from the CNNLSTM and obtaining the output features through the fully connected layers. Let the deep feature vector \( f_{jk} \) be an instance extracted from an image in \( F_k \) : \( f_{jk} = m_k(x_{jk}) \). The feature vector consists of 1,000 components, which corresponds to the number of hidden neurons: \( f_{jk} = \{v_1, v_2, \ldots, v_{nk}\}, f_{jk} \in F_k \).
Each feature vector is normalized using an L2-norm technique (Step 7). This normalization technique is commonly used to scale and standardize vectors in machine learning and data analysis. The L2-norm, also known as the Euclidean norm or the L2-norm, is a mathematical measure of the length or magnitude of a vector. In the context of feature normalization, the L2-norm technique calculates the square root of the sum of the squares of each component in the vector, resulting in a normalized vector with a magnitude of 1. Here, we provide an explanation of the concept of L2-norm. Consider a deep feature vector as:

\[ F_k = [f_{1k}, f_{2k}, f_{3k}, \ldots, f_{nk}] \]  

(6)

where

\[ |f_{jk}| = \sqrt{\sum_{l=1}^{n} |v_{lk}|^2} \]  

(7)

where \( v_{lk} \) represents an extracted feature within \( f_{jk} \), and \( n^k \) denotes the total number of extracted features \( v \) in \( f_{jk} \). L2 serves as the final output layer of CNNLSTM, producing the extracted features as the output.

The representation of the class data \( c_{ik} \) is computed as the average of all feature vectors in \( F_k \), which have been extracted from instances of the \( i \)-th class sample data (Step 9). In this step, we compute the centroids for the two classes: the centroid of the good bottles (class 0) and the centroid of the bottles with defects (class 1): \( y = \{0, 1\} \). These class centroids enable the recognition of patterns on the side walls of the bottles. The centroid vector \( c_{ik} \) is calculated as follows:

\[ c_{ik} = \frac{1}{N^k} \sum_{f \in N^k} f_{jk} \]  

(8)

where \( c_{ik} \) constrain the average values \( a : c_{ik} = \{a_1, a_2, \ldots, a_n\} \); and \( N^k \) is the number of all data instances of class \( i \) in \( s_k \). The centroids are representation of class data.

Predicted a bottle is normal class or abnormal class (\( h \)) is based on the similarity distance between \( c_{ik} \) and \( F_{k} \) (Step 12). The similarity distances between each class are compared using the Euclidean distance to predict the category of the bottle. Predict normal or abnormal bottles by minimizing the similarity distance value defined as

\[ h_j = \arg \min_i \|f_{jk} - c_{ik}\|_2^2, \text{ for } j = 1 : Nk \]  

(9)

where \( Nk \) is the number of instances in \( s_k \).

To assess the extraction and classification performance of a trained CNNLSTM model \( m_k \), the Mean Squared Error (denoted as MSE) is measured. If the model error (MSE) exceeds an error threshold value (\( \theta \)), the deep neural model will undergo an additional training process of 5 epochs (proceed to Step 5). If the condition is met, the deep feature vectors \( F_k \) and the centroid \( c_d \) are recomputed. Otherwise, the training process is stopped, and the model is obtained. The training iteration \( k \) is completed based on the performance condition. The threshold value \( \theta \) is used to measure the performance and reduce the number of training iterations in each model generation. It serves as a criterion for selecting a set of models that require fewer training iterations compared to the training process of a normal deep neural network. At this point, the model \( m_k \) is added to the ensemble \( E \) (\( m_k \) \( \in \) \( E \)), and the centroid \( c_d \) is included in the set \( c_d \) \( \in \) \( C \) (Step 15). For the next iteration, we used pre-trained CNNLSTM weights from the previous iteration \( k \) to learn the current data \( m_{k+1} \rightarrow m_k \). This method eliminates the need to reset the weights each time and only requires fine-tuning on the new set of bottle images. The overview of feature learning method is illustrated in Fig. 6.

3) The classifying process: This section presents a detailed discussion of the proposed similarity voting method. This classification method focuses on utilizing distinctive features of defects for bottle defect detection. The similarity distance is calculated between the representation features of normal bottles and abnormal bottles. The classification steps of our method are outlined in Algorithm 2.

**Algorithm 2:** Pseudocode for classifying bottle defects.

**Input:**
- \( E \): A set of CNNLSTM models
- \( C \): A set of centroids of class data
- \( x \): A bottle images

**Output:**

\[ h_i = \arg \min_i \sum_k s_{dik} \]

1: For \( k = 1, 2, \ldots, K \) do:
2: Extract deep features \( f_k = m_k(x) \)
3: Normalize the deep feature vectors \( f_k \) by L2-norm
4: For \( i = 1: y \) do:
5: Calculate the distance \( d_{ik} \) between centroid \( c_{ik} \) to features \( f_k \)
6: End For
7: Compute the distance weight \( s_{dik} \)
8: End For

The input for Algorithm 2 consists of a set of CNNLSTM models \( E \), the centroid of class data \( C \), and the bottle image \( x \). The models in set \( E \) differ from the decision boundary models of the same architecture. Extract features in bottle images.
image using each model in E and combine them to predict the final output. Deep feature vectors are extracted from the bottle image: \( f_k = m_k(x) \) (Line 2). Then, the deep feature vector \( f_k \) is normalized using the L2-norm technique: \( f_k = L2(f_k) \) (Step 3). The deep feature vector is normalized to ensure it resides in the same vector space as \( c_{ik} \), enabling the calculation of similarity values. The similarity distance \( d_{ik} \) between feature and centroid is expressed as follows:

\[
\alpha_{ik} = \|f_k - c_{ik}\|_2, \quad f \lor k = 1: K,
\]

where \( K \) is the number of ensemble model. Each centroid calculates the similarity distance with the features of the bottle image. The voting weight of each model is computed as follows:

\[
sd_{ik} = \frac{\alpha_{ik}}{\sum_l \alpha_{il}},
\]

where \( sd_{ik} \) is the voting weight of class \( i \) at model \( k \). The voting weight is determined by calculating the average distance between the centroid and the features of both normal and abnormal bottles (Step 6).

The final output of this method is prediction class that compute as follows:

\[
h_i = \arg \min \sum_l \sum_k sd_{ik}.
\]

where \( h_i \) is the class or category that the method predicts the input image belongs to based on its defect patterns and relationships from the training bottle images. The proposed classification method is illustrated in Fig. 7.

**IV. EXPERIMENTAL RESULTS AND DISCUSSION**

In this section, we present a comprehensive set of experiments conducted to assess the performance of the CNNLSTM method.

**A. Experimental Setup**

These experiments used three glass bottle image datasets, namely the body dataset, the neck dataset, and the rotating dataset. These datasets consist of three primary types of defects: stone, tear, and blister. The body dataset of glass bottles comprises a total of 571 images. For the training set, 292 images, consisting of 161 images with defects and 131 images without defects were used to train models. The remaining 279 images constituted the testing set, with 207 images containing defects and 72 images without defects. The neck dataset encompasses 570 images of glass bottles. The training set include 288 images, with 150 images exhibiting defects and 138 images without defects. The rotating dataset encompasses 570 images of glass bottles. The training set include 288 images, with 150 images exhibiting defects and 138 images without defects. The testing set includes 282 images, with 123 images containing defects and 159 images without defects. The distribution of each defect type in both the training and testing datasets for body, neck and rotating bottles is presented in Table I.

<table>
<thead>
<tr>
<th>Defect type</th>
<th>Body</th>
<th>Neck</th>
<th>Rotating</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>Train</td>
<td>Test</td>
<td>Train</td>
</tr>
<tr>
<td>Stone</td>
<td>81</td>
<td>127</td>
<td>80</td>
</tr>
<tr>
<td>Tear</td>
<td>30</td>
<td>28</td>
<td>20</td>
</tr>
<tr>
<td>Blister</td>
<td>50</td>
<td>52</td>
<td>50</td>
</tr>
<tr>
<td>Normal</td>
<td>131</td>
<td>72</td>
<td>138</td>
</tr>
<tr>
<td>Total</td>
<td>292</td>
<td>279</td>
<td>288</td>
</tr>
</tbody>
</table>

To assess the effectiveness of our proposed method on the defect dataset, various image processing techniques, including image thresholding and edge detection, along with machine learning models such as Anomaly Detection with Autoencoder (ADA), ResNet50, AlexNet, VGG16, and MobileNetV3 were used to construct models for comparison. Image thresholding, a commonly used technique in image processing using OpenCV, involves setting pixel values to either 0 or a maximum value based on a predefined threshold [13]. Edge detection is a...
digital image processing technique used to identify and extract edges in an image, representing abrupt changes in intensity or color between adjacent pixels [14]. ResNet50 is a deep convolutional neural network architecture introduced in 2016 [15]. ResNet-50 consists of 50 layers and employs residual blocks, each containing convolutional layers and shortcut connections. It has achieved state-of-the-art performance in various image recognition tasks. AlexNet is a deep convolutional neural network architecture proposed in 2012 [16]. AlexNet comprises eight layers, including five convolutional layers and three fully connected layers. It introduced several key innovations, such as using ReLU as an activation function and applying dropout regularization. VGG16 developed by the Visual Geometry Group (VGG) at the University of Oxford in 2014, is a widely recognized deep convolutional neural network architecture primarily used for image classification [17]. It consists of 13 convolutional layers followed by 3 fully connected layers, and it has demonstrated outstanding performance on numerous computer vision tasks. MobileNetV3, introduced by Google in 2019, is specifically designed for mobile and embedded devices that have limited computational resources [18]. This architecture aims to strike a balance between model size and accuracy, making it suitable for efficient deployment on devices with constrained hardware capabilities. In addition to these models, we also employed the Anomaly Detection with Autoencoder (ADA) technique. ADA detects anomalies by evaluating the reconstruction loss, comparing it to a predefined threshold. If the reconstruction loss surpasses the threshold, the input is classified as an anomaly.

For training our method and deep learning-based models, we used a training set comprising 292 images for the body dataset and 288 images for the neck dataset. The training epochs are set at 500 for VGG16, RestNet50, MobileNetV3, and ADA. The input error threshold, $\theta$, of the proposed method is set to 0.2, and the number of ensemble models, $K$, is set to 3. All the experiments, including the proposed defect detection method and the comparison methods, were conducted on a personal computer equipped with an AMD Ryzen 7 5000 series processor and an NVIDIA GTX 1650 GPU, allowing for efficient computation and analysis of the results.

### B. Experimental Result

Four evaluation metrics are adopted for the analysis: Accuracy, Recall, Precision, and F1-score. These metrics are widely used in image defect detection and provide comprehensive insights into the performance of the methods. They are defined as follows:

\[
\text{Accuracy} = \frac{TP + TN}{TP + TN + FP + FN} \quad (13)
\]

\[
\text{Recall} = \frac{TP}{TP + FN} \quad (14)
\]

\[
\text{Precision} = \frac{TP}{TP + FP} \quad (15)
\]

\[
F_1 = 2 \times \frac{\text{Precision} \times \text{Recall}}{\text{Precision} + \text{Recall}} \quad (16)
\]

$TP$ refers to the number of defect bottles (positive) that are correctly predicted as positive by the model. $FN$ represents the number of defect bottles (positive) that are incorrectly predicted as bottle without defects (negative) by the model. $TN$ indicates the number of the bottles without defect (negative) that are correctly predicted as negative by the model. Lastly, $FP$ signifies the number of the bottles without defects (negative) that are mistakenly predicted as positive by the model.

Fig. 8 presents visualizations of defect classification results of CNNLSTM for the body, neck, and rotation of bottle datasets.

The classification accuracy of various methods on the defect datasets is presented in Table II. The results for the body dataset demonstrate that the standalone VGG16 method outperformed all other methods, achieving an accuracy of 80%. ResNet50 is ranked second in terms of accuracy. Among the CNNLSTM models, CNNLSTM-VGG16 achieved the highest accuracy compared to the other CNNLSTM variants. Conversely, the image threshold method exhibited the lowest accuracy on this dataset.

![Fig. 8. Examples of classification results of the CNNLSTM.](image)

<table>
<thead>
<tr>
<th>Methods</th>
<th>Body dataset</th>
<th>Neck dataset</th>
<th>Rotating dataset</th>
</tr>
</thead>
<tbody>
<tr>
<td>CNNLSTM-ResNet50</td>
<td>71</td>
<td>95</td>
<td>62</td>
</tr>
<tr>
<td>CNNLSTM-AlexNet</td>
<td>75</td>
<td>89</td>
<td>97</td>
</tr>
<tr>
<td>CNNLSTM-MobileNetV3</td>
<td>64</td>
<td>92</td>
<td>98</td>
</tr>
<tr>
<td>CNNLSTM-VGG16</td>
<td>77</td>
<td>80</td>
<td>90</td>
</tr>
<tr>
<td>ResNet50</td>
<td>79</td>
<td>79</td>
<td>60</td>
</tr>
<tr>
<td>AlexNet</td>
<td>74</td>
<td>92</td>
<td>60</td>
</tr>
<tr>
<td>VGG16</td>
<td>80</td>
<td>85</td>
<td>95</td>
</tr>
<tr>
<td>MobileNetV3</td>
<td>70</td>
<td>72</td>
<td>70</td>
</tr>
<tr>
<td>ADA</td>
<td>74</td>
<td>75</td>
<td>60</td>
</tr>
<tr>
<td>Image threshold</td>
<td>62</td>
<td>65</td>
<td>65</td>
</tr>
<tr>
<td>Edge detection</td>
<td>70</td>
<td>72</td>
<td>63</td>
</tr>
</tbody>
</table>
The neck dataset results reveal that CNNLSTM-ResNet50 attained the highest accuracy of 95%, closely followed by CNNLSTM-AlexNet, CNNLSTM-MobileNetV3, CNNLSTM-VGG16, and AlexNet. Among the standalone methods, AlexNet exhibited the highest accuracy on this dataset. Conversely, the image threshold method yielded the lowest accuracy for the neck dataset.

Upon comparing the results of the rotating dataset images, it is obvious that CNNLSTM-MobileNetV3 achieved the highest accuracy. CNNLSTM-AlexNet and the standalone VGG16 also demonstrated good performance with high accuracies. Conversely, ADA exhibited the lowest accuracy percentages on this dataset.

Overall, the results indicate that different methods perform differently on the three datasets. The standalone VGG16 achieved the highest accuracy on the body dataset. CNNLSTM generally achieved the highest accuracy on both the neck and rotating datasets.

In Table III, the performance measures, including Recall, Precision, and F1-score, are presented for the body dataset. VGG16 demonstrates the highest level of correctness, evident from its superior metrics across Recall, Precision, and F1-score. Following closely in terms of correctness, CNNLSTM-AlexNet achieves relatively high Precision and F1-score. On the other hand, Image Threshold exhibits the lowest level of correctness among the methods listed.

Table IV presents the Recall, Precision, and F1-score metrics of the proposed method and the compared methods for rotating image datasets. Among the CNNLSTM methods, CNNLSTM-MobileNetV3 and CNNLSTM-AlexNet demonstrate outstanding performance, achieving high recall, precision, and F1-score. CNNLSTM-VGG16 also performs well, with consistently high metrics. However, CNNLSTM-ResNet50 exhibits comparatively lower performance across the metrics. When considering individual models, VGG16 stands out with consistently high scores for recall, precision, and F1-score. ResNet50, AlexNet, and MobileNetV3 display varying levels of performance, with some metrics being higher or lower than others. In terms of the additional methods, Image Threshold exhibits relatively high recall but lower precision and F1-score. ADA demonstrates quite low performance across the metrics.

Table VI displays the training time and detection time of all detection methods. Among the CNNLSTM methods, CNNLSTM-VGG16 has the longest training time on all three datasets, followed by CNNLSTM-ResNet50. However, CNNLSTM-ResNet50 has the shortest detection time across all datasets, indicating faster inference speed. CNNLSTM-AlexNet and CNNLSTM-MobileNetV3 also demonstrate relatively shorter training and detection times. For the individual models, ResNet50, AlexNet, VGG16, and MobileNetV3 have consistent training and detection times across all three datasets. Among the additional methods, ADA has relatively short training and detection times, while Image

<table>
<thead>
<tr>
<th>Methods</th>
<th>Recall (%)</th>
<th>Precision (%)</th>
<th>F1-score (%)</th>
</tr>
</thead>
<tbody>
<tr>
<td>CNNLSTM-ResNet50</td>
<td>80</td>
<td>73</td>
<td>70</td>
</tr>
<tr>
<td>CNNLSTM-AlexNet</td>
<td>82</td>
<td>74</td>
<td>74</td>
</tr>
<tr>
<td>CNNLSTM-MobileNetV3</td>
<td>71</td>
<td>66</td>
<td>63</td>
</tr>
<tr>
<td>CNNLSTM-VGG16</td>
<td>83</td>
<td>75</td>
<td>75</td>
</tr>
<tr>
<td>ResNet50</td>
<td>71</td>
<td>75</td>
<td>73</td>
</tr>
<tr>
<td>AlexNet</td>
<td>73</td>
<td>74</td>
<td>70</td>
</tr>
<tr>
<td>VGG16</td>
<td>86</td>
<td>78</td>
<td>77</td>
</tr>
<tr>
<td>MobileNetV3</td>
<td>57</td>
<td>57</td>
<td>57</td>
</tr>
<tr>
<td>ADA</td>
<td>51</td>
<td>55</td>
<td>53</td>
</tr>
<tr>
<td>Image Threshold</td>
<td>50</td>
<td>53</td>
<td>51</td>
</tr>
<tr>
<td>Edge detection</td>
<td>78</td>
<td>72</td>
<td>67</td>
</tr>
</tbody>
</table>

VGG16 outperforms the CNNLSTM method in terms of accuracy on the body image dataset. The VGG16 is advantaged by being pre-trained on a large and diverse collection of images. Its capacity to identify various features associated with bottle defects, coupled with the requirement for a comprehensive and diverse training set, contributes to its superior performance.

Table IV presents the Recall, Precision, and F1-score metrics of the proposed method and the compared methods for neck image datasets. Among the CNNLSTM methods, CNNLSTM-ResNet50 demonstrates the highest recall, precision, and F1-score, indicating its strong performance. CNNLSTM-MobileNetV3 follows closely behind with good recall, precision, and F1-score. Among the individual models, AlexNet demonstrates good performance, achieving high recall, precision, and F1-score. MobileNetV3 exhibits moderate performance with decent recall and F1-score, but its precision is relatively lower. ADA demonstrates moderate performance across all metrics. On the other hand, Image Threshold and Edge detection methods exhibit lower performance, with lower recall, precision, and F1-score.

Table V presents the Recall, Precision, and F1-score metrics of the proposed method and the compared methods for rotating image datasets. Among the CNNLSTM methods, CNNLSTM-MobileNetV3 and CNNLSTM-AlexNet demonstrate outstanding performance, achieving high recall, precision, and F1-score. CNNLSTM-VGG16 also performs well, with consistently high metrics. However, CNNLSTM-ResNet50 exhibits comparatively lower performance across the metrics. When considering individual models, VGG16 stands out with consistently high scores for recall, precision, and F1-score. ResNet50, AlexNet, and MobileNetV3 display varying levels of performance, with some metrics being higher or lower than others. In terms of the additional methods, Image Threshold exhibits relatively high recall but lower precision and F1-score. ADA demonstrates quite low performance across the metrics.

Table VI displays the training time and detection time of all detection methods. Among the CNNLSTM methods, CNNLSTM-VGG16 has the longest training time on all three datasets, followed by CNNLSTM-ResNet50. However, CNNLSTM-ResNet50 has the shortest detection time across all datasets, indicating faster inference speed. CNNLSTM-AlexNet and CNNLSTM-MobileNetV3 also demonstrate relatively shorter training and detection times. For the individual models, ResNet50, AlexNet, VGG16, and MobileNetV3 have consistent training and detection times across all three datasets. Among the additional methods, ADA has relatively short training and detection times, while Image Threshold and Edge detection methods exhibit lower performance, with lower recall, precision, and F1-score.
Threshold and Edge detection methods have negligible training times but slightly longer detection times compared to the CNNLSTM and individual models.

Overall, the experimental results demonstrate the effectiveness of the proposed CNNLSTM models for defect detection on all three datasets. The proposed models achieved competitive accuracy and performed well in terms of evaluation metrics. The training and detection times varied among the models, with CNNLSTM-AlexNet demonstrating the shortest training time. These findings can guide the selection of appropriate models based on the trade-off between accuracy and computational time requirements in practical applications.

**TABLE V. THE PERFORMANCE ON THE ROTATING IMAGES DATASETS IN TERMS OF RECALL, PRECISION, AND F1-SCORE**

<table>
<thead>
<tr>
<th>Methods</th>
<th>Recall (%)</th>
<th>Precision (%)</th>
<th>F1-score (%)</th>
</tr>
</thead>
<tbody>
<tr>
<td>CNNLSTM-ResNet50</td>
<td>66</td>
<td>71</td>
<td>61</td>
</tr>
<tr>
<td>CNNLSTM-AlexNet</td>
<td>97</td>
<td>96</td>
<td>96</td>
</tr>
<tr>
<td>CNNLSTM-MobileNetV3</td>
<td>97</td>
<td>98</td>
<td>98</td>
</tr>
<tr>
<td>CNNLSTM-VGG16</td>
<td>91</td>
<td>90</td>
<td>89</td>
</tr>
<tr>
<td>ResNet50</td>
<td>55</td>
<td>58</td>
<td>52</td>
</tr>
<tr>
<td>AlexNet</td>
<td>52</td>
<td>79</td>
<td>42</td>
</tr>
<tr>
<td>VGG16</td>
<td>95</td>
<td>94</td>
<td>94</td>
</tr>
<tr>
<td>MobileNetV3</td>
<td>62</td>
<td>79</td>
<td>52</td>
</tr>
<tr>
<td>ADA</td>
<td>58</td>
<td>55</td>
<td>56</td>
</tr>
<tr>
<td>Image Threshold</td>
<td>77</td>
<td>69</td>
<td>63</td>
</tr>
</tbody>
</table>

**TABLE VI. THE TRAINING TIME AND DETECTION TIME ON THE BODY IMAGES AND NECK IMAGES DATASETS**

<table>
<thead>
<tr>
<th>Methods</th>
<th>Body dataset</th>
<th>Neck dataset</th>
<th>Rotating dataset</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>Training time (second)</td>
<td>Detection time (second)</td>
<td>Training time (second)</td>
</tr>
<tr>
<td>CNNLSTM-ResNet50</td>
<td>154.29</td>
<td>0.11</td>
<td>143.55</td>
</tr>
<tr>
<td>CNNLSTM-AlexNet</td>
<td>103.59</td>
<td>0.12</td>
<td>59.22</td>
</tr>
<tr>
<td>CNNLSTM-MobileNetV3</td>
<td>141.93</td>
<td>0.12</td>
<td>80.69</td>
</tr>
<tr>
<td>CNNLSTM-VGG16</td>
<td>304.41</td>
<td>0.074</td>
<td>304.84</td>
</tr>
<tr>
<td>ResNet50</td>
<td>1,500</td>
<td>0.5</td>
<td>1,500</td>
</tr>
<tr>
<td>AlexNet</td>
<td>690</td>
<td>0.3</td>
<td>690</td>
</tr>
<tr>
<td>VGG16</td>
<td>1,800</td>
<td>0.5</td>
<td>1,800</td>
</tr>
<tr>
<td>MobileNetV3</td>
<td>780</td>
<td>0.3</td>
<td>780</td>
</tr>
<tr>
<td>ADA</td>
<td>180</td>
<td>0.2</td>
<td>180</td>
</tr>
<tr>
<td>Image Threshold</td>
<td>-</td>
<td>0.002</td>
<td>-</td>
</tr>
<tr>
<td>Edge detection</td>
<td>-</td>
<td>0.001</td>
<td>-</td>
</tr>
</tbody>
</table>

**V. CONCLUSION**

This research presents an approach for detecting defects in glass bottles using a combination of deep convolutional neural networks with long short-term memory (CNNLSTM) and instance-based classification algorithms. The CNNLSTM combines two types of deep neural networks to extract features and create a representation of class data, which is then used for defect detection. The proposed method is compared with other well-known defect detection methods. Experimental results demonstrate that the proposed method outperforms other defect detection methods in terms of detection performance. Additionally, the proposed method requires less training time, making it suitable for efficient glass bottle defect detection in production lines.

In future work, we intend to extend the application of the proposed algorithm to different types of bottles and explore additional defect types. The flexibility of the CNNLSTM allows for varying the number of models and designing customized detection layers to adapt to specific datasets. Furthermore, there is potential to extend this method to semi-supervised learning, enabling the detection of defects in other product categories. These future directions will further enhance the capabilities and applicability of our approach in defect detection and contribute to the advancement of quality control systems in various industries.

**REFERENCES**


Anomalous Taxi Trajectory Detection using Popular Routes in Different Traffic Periods

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Abstract—Anomalous trajectory detection is an important approach to detecting taxi fraud behaviors in urban traffic systems. The existing methods usually ignore the integration of the trajectory access location with the time and trajectory structure, which incorrectly detects normal trajectories that bypass the congested road as anomalies and ignores circuitous travel of trajectories. Therefore, this study proposes an anomalous trajectory detection algorithm using the popular routes in different traffic periods to solve this problem. First, to obtain popular routes in different time periods, this study divides the time according to the time distribution of the traffic trajectories. Second, the spatiotemporal frequency values of the nodes are obtained by combining the trajectory point moments and time span to exclude the interference of the temporal anomaly trajectory on the frequency. Finally, a gridded distance measurement method is designed to quantitatively measure the anomaly between the trajectory and the popular routes by combining the trajectory position and trajectory structure. Extensive experiments are conducted on real taxi trajectory datasets; the results show that the proposed method can effectively detect anomalous trajectories. Compared to the baseline algorithms, the proposed algorithm has a shorter running time and a significant improvement in F-Score, with the highest improvement rate of 7.9%, 5.6%, and 10.7%, respectively.

Keywords—Anomalous trajectory detection; time periods; popular routes; gridded distance

I. INTRODUCTION

With the rapid development of GPS positioning and wireless communication technology, more and more trajectory data have been collected [1, 2]. Detecting anomalous trajectories from a large amount of collected trajectory data has been widely used in various fields, such as fraud detection [3], medical treatment [4], and anomalous trajectory detection [5]. The huge trajectory data generated by vehicle location acquisition also provides an unprecedented opportunity to analyze the anomalies of moving objects and discover some basic rules of their movements [6]. Through timely and effective analysis of these traffic trajectories, the behavior rules of vehicles can be detected, thereby revealing the dynamic changes of certain behaviors and the “special” events hidden behind the vehicle behavior patterns [7]. For example, detours can be detected during taxi driving [8], and traffic jams due to traffic accidents [9] or temporary road closures [10]. This information can better guide the driving of taxi routes and provide early warnings of potential safety hazards, thus improving urban traffic planning.

Anomalous trajectory detection is one of the hot research topics in trajectory pattern mining [11, 12]. Usually, an anomaly means that a data object has a large deviation from the remaining objects in the retrieved dataset due to some of its unusual characteristics. For these different anomalous trajectories, many anomaly detection methods have been proposed. The existing anomalous trajectory detection technology mainly uses the similarity measurement method to find out the trajectory that is highly different from others [13-15]. The LDTRAOD [16], DB-TOD [17], and the ATD-outliers method [18] combine attributes such as distance between trajectory points, local density, and trajectory shape to calculate the abnormal score of trajectories by quantifying the similarity between trajectories. However, the time complexity of these methods is too high, which leads to prohibitive costs. Therefore, trajectory grid processing is often exploited to solve this problem, reducing the computational complexity by mapping the spatial locations of trajectory points onto the grid codes. In addition, it can be seen from the existing methods that the key to anomalous trajectory detection is extracting a representative feature of each trajectory and then using a function based on this to measure the similarity between them. Therefore, a taxi driving a long distance or a trajectory with fewer same driving trends is considered to be an anomalous trajectory. Although this can detect a large number of anomalies, the false positives of the detection results are usually high. Because these methods focus only on the location of the trajectory and do not integrate it with the time period and the trajectory structure, thus incorrectly detect normal driving trajectories of particular time periods as anomalies, and ignore loop driving behavior on normal routes.

For example, a passenger Alice takes a taxi from the starting point S to the destination point D. There are two routes at fixed starting and ending points, namely R and R0, where the distance of R0 is shorter than that of R. If the trip occurs during the off-peak period, which means that the traffic volume is relatively low during this period. Therefore, considering the time and fare spent by passengers on the trip, the route R0 is chosen as the optimal route for this road segment. However, if the trip occurs between 7:00 am-9:00 am, the route R0 may cause serious road congestion and traffic accidents due to the traffic flow during rush hour. Therefore, in terms of the time peak, the optimal choice for taxi drivers should be R.

Obviously, as the dynamics of popular routes change, the anomaly judgment of the trajectory also changes. The driver's choice of route usually changes at special periods or moments when special events occur. Therefore, it is necessary to divide
the effective time intervals according to the vehicle travel pattern. However, if the division is performed at very short and fixed intervals, it will consume a lot of unnecessary time and space resources, and the detection results will also contain certain errors. Research [13] pointed out that the taxi trajectory tends to take more time during peak periods than during off-peak periods. Inspired by this, it can be determined that drivers will choose different routes at specific time periods to save passengers’ travel time. Therefore, this paper proposes an anomalous trajectory detection method using popular routes in different traffic periods (PRTP). Compared with previous detection algorithms that only consider the location of trajectory points or have large time consumption, the PRTP algorithm divides the time of day into four time periods according to the temporal distribution density of trajectories, thus discovering the dynamic changes of the popular routes and considering the effect of time segmentation and trajectory time span on frequency. Meanwhile, the location and structure of the trajectory are considered comprehensively in the anomaly judgment, which improves the stability and accuracy of detection.

In summary, the main contributions of this study are as follows.

- An anomalous trajectory detection method using popular routes in different time periods is proposed, which divides the time according to the traffic flow and combines the location and time features of trajectories in different time periods to detect anomalies, thus improving the accuracy of anomaly detection.

- By integrating the access location, time, and time span of the trajectory, a calculation method of spatiotemporal frequency is designed, so that the popular routes can be accurately obtained without the interference of temporal anomalous trajectory.

- A grid distance formula is proposed to quantitatively measure the trajectory anomaly distance, which considers both the location and structure of the trajectory, enabling the detection of circuitous driving anomalies of the trajectory in addition to location anomalies. The effectiveness and performance of the PRTP method are empirically evaluated using real taxi trajectories.

The rest of the paper is organized as follows. The related work on anomalous trajectory detection is reviewed in Section II. After the problem definitions are given in Section III, the main steps of the PRTP are presented in Section IV. In Section V, the overall experimental setup and experimental results are reported. Section VI discusses the proposed method. Finally, the conclusions are presented in Section VII.

II. RELATED WORK

At present, the field of anomalous trajectory detection has been developed rapidly. Though the existing anomalous trajectory detection algorithm has been well applied in real life, further explorations are still needed. The existing detection methods can be mainly divided into two categories, anomaly detection based on spatial attributes and anomaly detection based on spatiotemporal features.

A. Detection Based on Spatial Features

Among the existing anomaly detection algorithms, the algorithms focusing on the spatial position of the trajectory occupy a larger proportion. These methods focused on considering the spatial attributes of trajectory points, such as position, direction, velocity, etc., and select specific attributes as trajectory similarity criteria to detect anomalous trajectories. Lee et al. [19] proposed a trajectory anomaly detection algorithm TRAOD, which includes trajectory division and trajectory detection. All trajectories are divided into t-partitions according to a novel partition-and-detect framework, and then trajectory partitions are detected by a distance-based method. To detect anomalies from the locally dense trajectory, Luan et al. [16] proposed the anomaly detection algorithm LDTRAOD based on the local density of trajectories, which uses the partition-and-detect framework [19] to determine local anomalous trajectories by calculating the local density and local outlier factors for each t-partition. Although these methods can quantify the similarity between trajectories, anomaly detection based on individual location attribute does not apply to anomalies with complex distributions. To combine multiple features for anomaly detection, San Roman et al. [20] proposed the CaD method which considers the angle difference, Euclidean distance, and the number of points in each trajectory, and effectively detects anomalous trajectories in each cluster by using the unsupervised learning method of the trajectory distance matrix. Sun et al. [21] proposed the TODDT algorithm in combination with dynamic difference threshold to detect anomalous trajectories from multiple perspectives. Considering the trajectory common slice subsequence, Yu et al. [22] proposed a novel trajectory anomaly detection algorithm based on common slice subsequence (TODCSS), which combines features such as direction, position, and continuity to achieve more accurate trajectory anomaly detection. And to detect different patterns of anomalous trajectories, Wang et al [23] proposed a DIS metric. After gridding the trajectories, the DIS values are used to quantitify the similarity between trajectories and then assign the anomalous trajectories to different classifications. Considering the special conditions of anomalous trajectories and the motion patterns mined in the massive trajectory data, researchers have also proposed many methods. Inspired by the "few and different" property of anomalous trajectories, Zhang et al. [24] proposed the iBAT algorithm which can effectively detect outliers of trajectories by isolating the number of steps used by the trajectory. Combine the actual inherent travel flow and other information of trajectory data to make abnormal judgment. Yu et al. [25] proposed a multi-level approach that combines association rule techniques to discover moving routes. To mine the special driving patterns of moving objects from trajectories, Zhang et al. [26] proposed an algorithm to discover popular routes from fixed locations. The algorithm meshes the region area into a regular grid to discretize the historical trajectory and finally determines the most popular routes by an ant colony optimization method. Tang et al [27] used the theory of time geography to propose a path-oriented traffic state estimation model to find the most likely road path,
link travel time and activity duration at possible intermediate stations.

Real-time performance is the primary requirement of practical applications when detecting trajectory anomalies. Traffic conditions vary at different times and in different locations, exploiting latent patterns in real-time traffic for real-time anomaly detection [28]. Therefore, while overcoming the drawbacks brought by the single temporal attribute algorithm, researchers have investigated anomaly detection for evolutionary trajectories. To reduce the problem of the high false alarm rate of anomaly detection due to complex road conditions, Chen et al. [29] proposed an online anomaly trajectory based on multidimensional criteria for real-time anomaly detection by simultaneously considering multidimensional criteria such as similarity, time, and distance. Ge et al. [30] proposed an evolutionary trajectory outlier detection method TOP-EYE, which continuously calculates the outlier score for each trajectory in a cumulative manner and proposed a decay function to mitigate the impact of historical trajectories on the outlier score of trajectory evolution, thus identifying evolutionary anomalous trajectories at an earlier stage. Using a street-based trajectory delineation approach, Shi et al. [31] proposed RUTOD, an anomaly detection framework for real-time urban traffic. The framework combines an individual anomalous moving object with the group anomalous values generated by various moving objects for anomalous detection.

B. Detection Based on Spatiotemporal Features

In the practical application of anomalous trajectory detection, in addition to various situations where spatial attributes need to be considered, temporal features are also important for anomalous trajectory detection. In recent years, a lot of research and exploration has been conducted on time attributes. Zhu et al. [32] proposed a new time-dependent popular route algorithm TPRO. In this algorithm, the abnormal trajectory is dynamically measured by considering the spatial and temporal characteristics of the motion trajectory. However, the TPRO algorithm is limited to the detection of historical trajectories, and it is not applicable to anomalies of real-time trajectories. Therefore, in the TPRRO [33], a real-time outlier detection algorithm was proposed based on time-dependent popular routes, which can realize efficient detection and evaluation of the testing trajectory through offline preprocessing and online anomaly detection.

Besides, for the detection of more complex anomalous trajectories in the spatiotemporal dimension, Yang et al. [34] proposed a new trajectory clustering algorithm TAD, which is based on the spatiotemporal density analysis of trajectory data to detect the stay of complex trajectories. To detect behaviors such as abnormal parking and stranding of vehicles in the time dimension, He et al. [35] proposed a common subsequence-based spatiotemporal anomaly trajectory detection method (STADCS). This method detects spatiotemporal anomalies by combining spatial features and temporal dimensions of trajectory subsequences. Based on the travel time of the trip, Eldawy et al. [36] proposed a novel real-time anomalous trajectory detection system called FraudMove. When choosing the best route, the FraudMove method takes the popular route as the best choice. Then, an adjustable time window parameter is used to control the detection times of anomaly detection, thus dynamically detecting anomalous taxi behaviors in combination with travel time.

Although most of the above anomalous detection methods combine multiple trajectory attributes to achieve different modes of anomalous trajectory detection, there are still two main limitations. First, if the spatial features or temporal features of trajectories are used alone, such detection results will produce high false positives. Researchers only consider the matching degree of trajectory points when considering the node frequencies in different time periods, the detection accuracy will be reduced by ignoring the influence of other attributes on the node frequencies. Second, most algorithms only consider the location abnormality of trajectories and ignore the change of trajectory structure in the abnormality judgment. Therefore, it is necessary to divide the time periods in anomalous trajectory detection and calculate the frequency in the spatiotemporal dimension, and integrate the trajectory structure with the trajectory position to improve the accuracy of abnormal driving trajectory detection.

III. PROBLEM DEFINITION

This section defines some related terms and provides formal definitions of the problems considered in this paper to facilitate further descriptions.

Definition 1 (Point): given a record, let \( x, y, \) and \( t \) be longitude, latitude, and timestamp, respectively. Then the spatiotemporal information can be recorded in a triplet, namely \((x, y, t)\), which is a spatiotemporal point formed by the object passing through the position \((x, y)\) at time \(t\).

Definition 2 (Trajectory): a trajectory is denoted by \( T \), which consists of a series of trajectory points:

\[
T_l = \{p_1, p_2, ..., p_j, ..., p_n\},
\]

the entire trajectory dataset is represented as:

\[
TS = \{T_1, T_2, T_3, ..., T_n\}.
\]

\(|TS|\) is the size of \(TS\), i.e., \(|TS|=n\). \(p_j\) is the \(j\)-th point of \(T_l\), which includes three components \(x, y, t\). The trajectory segment is the line segment between \(p_j\) and \(p_{j+1}\).

Definition 3 (Road Network): the road network is represented as a directed graph \(G(V,E)\), where \(V\) represents the set of nodes (such as the start and end points of a road segment), and \(E\) is the set of edges (such as a road segment). In this paper, \(v_i\) is used to represent a specific vertex in \(G\). If \(v_i\) and \(v_j\) are the two points of the edge \(e\), it can be expressed as \(e. v_i \) and \(e. v_j\).

Definition 4 (Mapped Trajectory): A mapped trajectory \(MT\) is generated by the function \(\phi(T)\), which maps latitude and longitude coordinates to the cell, and generates a series of grid codes \(c_1, c_2, c_3, ..., c_{len}\), where \(c_i = \phi(p_j)\).

Definition 5 (Equivalent Cell): Given a grid size of \(M\), the road network is divided into equal cells of size \(M\), and the position of cell is denoted by \(C\). Then, let trajectories pass through the road network regions, if two different trajectory points \(p_i\) and \(p_j\) fall into the same cell, they are called points
with the equivalent cell, which is expressed as $\text{EqCell}_M(p_i, p_j)$.

As shown in Fig. 1, there are two trajectories and all their trajectory points have corresponding grid cells. The points $p'_3$ and $p'_5$ on the different trajectories fall into the same grid cell (green grid), so they are called points with the equivalent cell.

![Fig. 1. An illustration of the equivalent cell.](image)

Definition 6 (Time Span): Given a trajectory $T_i$, the travel time from the source to the destination is defined as the time span, which is denoted by $S_{T_i}$ as:

$$S_{T_i} = T_{i_1} - T_{i_2},$$

where $T_{i_1}$, $T_{i_2}$ represent the start time and end time of the trajectory $T_i$, respectively.

Definition 7 (Route): After matching the trajectory with the city map using ArcGIS, the route is represented by each node cell and its frequency. A route based on cell frequency is represented as:

$$R = \{r_{C_1}, r_{C_2}, r_{C_3}, ..., r_{C_m}\},$$

where $r_{C_i} = (c_i, f_i)$, $c_i$ represents the $i$-th node cell on the route, $f_i$ refers to the frequency of passing through the node cell.

Definition 8 (Anomaly trajectory): The trajectory that deviates significantly from the popular routes in the access location or structure is defined as an anomaly trajectory.

As shown in Fig. 2, there are a few trajectories between $S$ and $D$, which can be divided into two groups: normal trajectories and abnormal trajectories. $T_1$ and $T_3$ are considered abnormal because they deviate from the regular routes at a certain time period.

![Fig. 2. An illustration of trajectories between $S$ and $D$.](image)

Problem statement: Given the trajectory dataset $TS = \{T_1, T_2, T_3, ..., T_n\}$, this study aims to detect anomaly trajectories in $TS$, for all $T_i \in TS$, if $T_i$ deviates from its corresponding popular routes or drive in a roundabout manner, it is considered an anomaly trajectory.

**IV. PRTP**

This section provides an overview of the anomalous trajectory detection framework proposed in this study. The PRTP method consists of two stages: trajectory preprocessing and anomalous trajectory detection. The popular routes acquisition and anomaly judgment constitute the trajectory anomaly detection stage. Fig. 3 shows the working mode of the algorithm.

![Fig. 3. Overview of PRTP approach.](image)

A. Preprocessing Taxi Trajectory

Since anomalous driving of taxis usually occurs when they are carrying passengers, the first task in the trajectory processing stage of this study is to classify the trajectories according to their occupancy status. As shown in Fig. 4, the figure shows the trajectory of a taxi in a region for one month, in which the red line segment is the trajectory of a taxi carrying passengers and the blue line segment is the trajectory of a taxi without passengers. Therefore, for a more meaningful study, the trajectory represented by the red line segment is extracted as a valid trajectory during the preprocessing of the trajectory data in this paper.

![Fig. 4. Trajectories of a taxi in San Francisco during a month.](image)

After extracting the valid trajectories from the original trajectories, these trajectories usually contain different starting and ending points, in which there may be a certain starting and ending point between which there are not enough trajectories to form a normal trajectory group, thus causing interference to the anomaly detection. Therefore, in this study, all taxi trajectories that pass through the same $SD$ pair are grouped to form the $SD$ trajectory dataset (trajectories with the same source and destination). The $S$ and $D$ regions are $1000m \times 1000m$ grid-cells, that is, the source and destination points satisfy $\text{EqCell}_{1000}(MT_1, ..., MT_n)$, $\text{EqCell}_{1000}(MT_1, ..., MT_n)$. Then, the city map is divided into grid cells of the same size, and the extracted valid $SD$ trajectories are mapped into the grid cells, thus forming a mapped trajectory consisting of a series of grid cell sequences.
B. Anomaly Detection

1) Acquisition of popular routes: By extracting a specific SD trajectory dataset from the original dataset and mapping it to the corresponding urban road network, all the routes traversed by the trajectory dataset are obtained. Traversing the trajectory data and calculating the node frequency based on its trajectory point location and time, thus obtaining the node frequency graph (NFG) as shown in Fig. 5. In Fig. 5, \( V_s \) represents the source address, \( V_d \) represents the destination address, and the table corresponding to each node indicates the frequency of that node in each time period. For the division of time periods, the study divides the whole time domain into four time periods according to the peak traffic flow period, namely the morning peak period (7:00 am-9:00 am), the afternoon peak period (11:00 am-13:00 pm), the evening peak period (17:00 pm-19:00 pm), and the time period consisting of the remaining hours, and then performs anomaly detection according to testing trajectory and the popular routes in its time period.

Definition 9 (Equivalent time periods): Given two time periods \( t_1 \) and \( t_2 \), if \( t_1, t_2, t_3 \), and \( t_4 \) are all in the same pre-divided time period, they are called equivalent time periods.

According to the above analysis, if there are time periods 7:30 am-8:15 am and 8:45 am-9:00 am, because their starting and ending time are included in the pre-divided morning peak time period, these two time periods are referred to as equivalent time periods in the study. And the trajectories of the equivalent time period have the same popular routes.

The anomalous trajectories are few and different, so the routes chosen by a large number of vehicles are often correct. Therefore, researchers usually determine the popularity of routes in each time period based on the number of times the route is visited. However, when there is a route with a large number of trajectories passing through, if only the number of trajectories is considered, the route must be identified as a popular route. But if all the trajectories on it have time anomalies, then this route is not desirable. So when calculating the frequency values, the study not only considers whether the trajectory points coincide with the nodes but also judges the time anomaly of the trajectories. Therefore, to exclude the influence of time-anomalous trajectories in calculating the frequency of route nodes, this study designs a calculation method for spatiotemporal frequency.

By comparing the trajectory time span with the standard time span, the frequency coefficient of the trajectory is obtained, which is calculated as:

\[
f_{ci} = \begin{cases} 
1 - \frac{St_i - St_j}{St_j}, & \text{if } St_i > St_j \text{ and } St_i - St_j < St_j \\
1, & St_i \leq St_j \\
0, & \text{else}
\end{cases}
\]

where \( St_i \) represents the time span of the \( i \)-th trajectory, \( St_j \) represents the standard time span of the \( j \)-th time period, and the value of \( j \) ranges from 1 to 4. The standard time span of each time period is obtained by selecting half of the normal trajectories in that time period and calculating the average time span of these trajectories.

After calculating the frequency coefficient of each trajectory passing through the node, the node frequency is defined as:

\[
f_v = \sum_{i=1}^{\Omega} f_{ci}
\]

where \( N \) is the number of trajectories passing through the node, the frequency of node \( v \) is the sum of the frequencies of \( N \) trajectories.

To obtain the popular routes of the testing trajectory, the popularity of each route needs to be compared. From the above analysis, the routes popularity can be evaluated by the node frequency values. Therefore, for route popularity comparison, the routes are expressed in the following form, i.e., \( R=(r_1, f_1, r_2, f_2, ... , r_m, f_m) \), \( R'=(r_1', f_1', r_2', f_2', ... , r_n, f_n') \). When \( r_x, f_x = r_x, f_x' \) (\( x \in \{1, 2, ... , p-1\} \)), if \( r_p, f_p > r_p, f_p' \), then the route \( R \) is more popular than route \( R' \), denoted by \( PR > PR' \).

After the analysis of route popularity, the first \( k \) high-frequency routes of the testing trajectory are called popular routes (PR), that is, if there is a route set \( RS = \{R_1, R_2, ..., R_k, R_{k+1}, ..., R_{q}\} \), where \( PR_{R_1} > PR_{R_2} > ... > PR_{R_k} > ... > PR_{R_q} \), then the popular routes \( PR = \{R_1, R_2, ..., R_k\} \).

Algorithm 1: Obtain the Popular Routes of the trajectory

Input: Road network \( G \), trajectory dataset \( TS \), Grid size \( M \), number of popular route \( k \).

Output: Popular routes \( PR \).

1. gridLabel ← Mesh the road network in \( M \) size
2. Map the TS to gridLabel to get \( RS = \{R_1, R_2, ..., R_{q}\} \).
3. \( TS \) is divided into four groups according to the time period of trajectory
4. for \( i \) in \( RS \) do
5. for \( j \) in \( TS \) do
6. Select trajectory group according to the time period of \( MT_j \)
7. Calculate \( St_i, f_{ci} \) according to equations (3), (5), respectively
8. Calculate \( f \) of \( R_i \) according to equation (6)
9. end for
10. end for
11. \( PR ← \) select the \( k \)-popularity routes
12. return \( PR \)

Fig. 5. An illustration of NFG.
The overall pseudo-code for obtaining the popular routes of the trajectory is given in Algorithm 1. First, the trajectories are mapped to the gridded road network to obtain the RS through which the trajectory dataset passes (lines 1-2). Then, the dataset is partitioned and trajectories belonging to the same time period are grouped (line 3). Finally, each trajectory is traversed to find the frequency values of the routes in different time periods, and the popular routes are obtained based on the popularity comparison (lines 4-11). The time complexity of Algorithm 1 is \( O(n \cdot q) \), where \( n \) is the number of trajectories and \( q \) is the number of routes.

2) Perform trajectory anomaly judgment: After the popular routes of the testing trajectory are obtained according to the above steps, the PRTP algorithm performs trajectory anomaly detection by comparing the testing trajectory with the top-\( k \) popular routes.

In this study, position anomalies and structural anomalies of trajectory are mainly considered in calculating trajectory distance. The position anomaly focuses on whether the visited positions are consistent, and the structure anomaly focuses on the circuitous intersection of trajectories. Before obtaining the trajectory distance, the trajectory and popular route are converted into position access matrices according to their access position. The elements of the LocMatrix are as:

\[
\text{LocMatrix}_{MT_i}(C) = \begin{cases} 1, \text{ if } C \text{ is accessed by } MT_i \\ 0, \text{ else} \end{cases},
\]

where \( C \) denotes the position in matrix.

The structure matrix \( \text{StrMatrix}_{MT_i} \) is obtained by determining the circuitous access grid through the number of trajectory visits. The elements of the StrMatrix are as:

\[
\text{StrMatrix}_{MT_i}(C) = \begin{cases} 1, \text{ if } \text{Num. } C > 1 \\ 0, \text{ else} \end{cases},
\]

where \( C \) denotes the position in StrMatrix, \( \text{Num. } C \) denotes the number of times the trajectory \( MT_i \) visits position \( C \).

In the study, the number of common locations and circuitous visits are focused on, and then define the distance matrix of the trajectory with the matrix information. Given the gridded trajectory \( MT_i \) and one of the corresponding optimal routes \( PR_j \), transform them into \( \text{LocMatrix}_{MT_i} \) and \( \text{LocMatrix}_{PR_j} \), respectively, and determine the \( \text{StrMatrix}_{MT_i} \) according to the number of trajectory visits. Then the anomaly distance is calculated according to the following defined formulas:

\[
\text{LocMatrix}_{common} = \text{LocMatrix}_{MT_i} \cap \text{LocMatrix}_{PR_j},
\]

\[
GDDis(MT_i, PR_j) = \frac{\text{GDDis}(|\text{LocMatrix}_{MT_i} - \text{LocMatrix}_{common}|)}{|\text{LocMatrix}_{MT_i}|},
\]

where \(|*|\) denotes the summation over the elements of the matrix *, \( \text{LocMatrix}_{common} \) is the matrix of the common position of the testing trajectory access matrix and the popular route access matrix.

Meanwhile, from equation (10), it can be seen that the higher the matching degree between the testing trajectory and popular routes, the greater the value of \( |\text{LocMatrix}_{common}| \), and the closer the \( GDDis \) distance is to 0. Therefore, the specific value of the \( GDDis \) distance can be used to quantify the match between the testing trajectory and popular routes.

In Fig. 6, given the trajectory \( MT_1 \) and the route \( PR_1 \), three 8x8-sized matrices can be obtained. From the matrix calculation, \( |\text{LocMatrix}_{MT_1}| = 19 \), \( |\text{LocMatrix}_{PR_1}| = 12 \), the intersection of the matrices \( |\text{LocMatrix}_{common}| = 11 \), and the \( |\text{StrMatrix}_{MT_1}| = 7 \). Therefore, the \( |\text{LocMatrix}_{MT_1} - \text{LocMatrix}_{common}| = 8 \), the number of abnormal cells is \( 8+7=15 \), then, the \( GDDis \) of \( MT_1 \) is 15/19=0.79. However, if \( MT_1 \) only considers the anomaly of the access location, the final \( GDDis \) is 8/19 = 0.42, which may misjudge \( MT_1 \) as a normal trajectory. Therefore, through the different results of the above two calculations of \( MT_1 \), it can be seen that the study combines structural anomalies with location anomalies improves the detection accuracy of anomalous trajectories.

Fig. 6. An access schematic of trajectory and route.

Based on the study [32], there are often multiple alternative routes between two locations, and not all of them have the same popularity. Therefore, when judging the trajectory anomaly, the popular routes should occupy different weights in the anomaly score. Suppose exist popular routes \( PRS = \{PR_1, PR_2, ..., PR_p, ..., PR_k\} \), \( PR_j = \{rc_1, f_1, rc_2, f_2, ..., rc_m, f_m\} \), the sum of the frequencies of all nodes for popular route is defined as:

\[
\text{SUMPR}_p = \sum_{i=1}^{m} PR_p.rc_i.f_i.
\]

Since the sum of the node frequencies indicates the route popularity, the popularity weight of the route is defined as:

\[
\text{w}_{PR_p} = \frac{\text{SUMPR}_p}{\sum_{j=1}^{k} \text{SUMPR}_j},
\]

where \( k \) represents the total number of popular routes, \( PR_j \) represents one of the popular routes, and \( \sum_{j=1}^{k} \text{SUMPR}_j \) represents the sum of the popular routes frequencies.

The anomaly judgment is performed by calculating the anomaly score of the trajectory and its corresponding \( k \) routes, which is defined as:

\[
\text{Score} = \sum_{p=1}^{k} \text{w}_{PR_p} \cdot \text{GDDis}(MT_i, PR_p).
\]
where $MT_i$ represents the testing trajectory, and $PR_p$ represents the $p$-th popular route.

Pseudo-code for the main steps of trajectory anomaly detection is given in Algorithm 2. First, the weights of the popular routes are calculated according to the node frequency (line 2). Then, the corresponding popular routes of testing trajectory are traversed to calculate the anomaly score between the trajectory and the popular route (lines 3-12). Finally, trajectory anomaly judgment is performed by the sum of the anomaly scores between the trajectory and the $PRL$, where trajectory lengths larger than $\theta$ are anomalous (lines 13-17). Meanwhile, from Algorithm 2, it can be seen that the time complexity of the PRTP algorithm mainly depends on two aspects: the loop of trajectory data and the loop of popular routes. Therefore, the time complexity of the PRTP algorithm is $O(n \cdot k)$, where $n$ represents the number of $SD$ trajectory datasets, and $k$ represents the total number of popular routes. In most cases, the value of $k$ is usually small, so the time complexity can be approximated to $O(n)$.

**Algorithm 2 Anomalous trajectory detection method using popular routes**

**Input:** gridLabel, top-$k$ Routes $PR$, Score threshold $\theta$.

**Output:** Traanomalous(A dataset of anomalous trajectories).

1. $Score \leftarrow 0$
2. WeightList $\leftarrow$ Calculate the frequency weights of each popular route according to equation (12)
3. for $i$ in gridLabel do
4. $PRL \leftarrow$ Get the grid sequences of the popular routes
5. for $j$ in PRL do
6. $LocMatrix_{MR} \leftarrow$ get the trajectory access matrix
7. $LocMatrix_{PR_j} \leftarrow$ get the route traversal matrix
8. Calculate the $LocMatrix_{common}$ according to equation (9)
9. calculate $GDDis$ according to equation (10)
10. $Score_j \leftarrow GDDis \times \text{WeightList}[j]$
11. $Score \leftarrow Score + Score_j$
12. end for
13. if $Score > \theta$ then
14. put trajectory $i$ into Traanomalous
15. end if
16. end for
17. return Traanomalous

**V. EXPERIMENTS**

In this section, an empirical evaluation of the proposed method is provided. All experiments are implemented with Python3 and conducted on a computer equipped with an Intel(R) Core(TM) i5-8250U CPU @ 1.80 GHz and 8GB main memory and running Windows 10 operating system.

**A. Dataset**

This study used the real taxi dataset provided by Piorkowski et al. [37], which contains all spatial locations of more than five hundred taxis in the San Francisco Bay Area in a month. The GPS trajectory records the latitude and longitude, timestamp, and corresponding passenger occupancy status of each taxi location point. In this study, the fixed starting and ending places selected were the airport and the central residential area. To reduce the noise and redundancy in the original trajectory data, it was assumed in the data preprocessing that only the trajectories of occupied taxis are valid. According to the above requirements, three pairs of SD trajectories (T-1 to T-3) were selected from the 463,860 trajectories extracted from 11.22 million GPS points, and whether each trajectory was anomalous or not was manually marked. The marked dataset is used to evaluate the detection accuracy of the PRTP algorithm.

**B. Parameter Setting**

Trajectory detection is essentially a binary classification problem, and Recall and F-Score are two important metrics to evaluate its performance. Therefore, these two metrics are used in this paper to judge the effectiveness of PRTP anomaly detection, which are defined as:

\[
\text{Precision} = \frac{TP}{TP + FP}, \quad (14)
\]

\[
\text{Recall} = \frac{TP}{TP + FN}, \quad (15)
\]

\[
F - \text{Score} = \frac{2 \times \text{Precision}\times\text{Recall}}{\text{Precision} + \text{Recall}}, \quad (16)
\]

where $TP$ (True Positive) denotes the number of detected anomalous trajectories, $TN$ (True Negative) denotes the number of detected normal trajectories, $FP$ (False Positive) denotes the number of normal trajectories incorrectly detected as anomalous, and $FN$ (False Negative) denotes the number of anomalous incorrectly detected as normal [13].

To better investigate the effect of parameter settings in the PRTP algorithm on the detection result, this study conducted parameter setting experiments on the two largest datasets T-1 and T-3. There are three parameters in the detection stage of the PRTP, the grid size $M$, the number of popular routes $k$, and the Score threshold $\theta$. Therefore, this study experimentally investigates the effects of these three parameters on the Recall and F-Score of detection to determine the optimal values of the parameters.

1) **Varying the size of grid cells.** The setting of grid cell size determines the cell size when meshing the road network. The smaller the grid cell size, the higher the number of grids in the mapping trajectory. This makes the small differences to be magnified, which affects the detection accuracy. If the grid cell size is too large, there will be more trajectories with the same grid sequence in the dataset, which makes it more difficult to compare the similarity between the trajectory and the popular routes. Therefore, the reasonable setting of grid cell size is crucial to the performance of PRTP algorithm. Referring to previous studies [23,24], the performance variation is considered when the size of grid cells increases from 200m $\times$ 200m to 500m $\times$ 500m in this experiment. Note that the grid cell size set in this paper refers to the length and width of each small square.

In Fig. 7, the changes in $F$-Score and Recall for different grid sizes are illustrated. It can be seen that on the two different datasets, the maximum $F$-Score is obtained when the grid cell size is 400x400m, and the Recall also reaches the maximum at this time.
2) Varying the number of popular routes. The number of popular routes \( k \) is an important parameter in the PRTP algorithm since the value of \( k \) determines how many routes can be used to calculate the anomaly score of the testing trajectory. Within a certain number of ranges, the detection results may be more accurate when more popular routes are taken for comparison. However, the testing trajectory needs to be compared with the routes one by one during anomaly judgment, so the detection time will increase linearly with \( k \). Referring to the previous study \([32]\), the range of \( k \) is set between 1 and 8.

Combined with the above definitions and calculation equations, it can be known that in the PRTP method, the node frequency is obtained from the node frequency graph of each popular route in advance, and when calculating the Score of the trajectory, each popular route is given the corresponding popularity weight and the popular route is also selected according to the order of node frequency. Fig. 8 shows the changes of \( F \)-Score and \( R \)-Score for different \( k \) values. The results show that the \( R \) and \( F \)-Score of anomaly detection in the range of 1 to 6 generally have an upward trend as the \( k \) increases, when \( k \geq 6 \), the detection performance gradually decreases as the \( k \) increases, and it can be seen that when \( k = 6 \), the \( F \)-Score on the datasets T-1 and T-3 both reach the maximum, and the \( R \) also reaches the peak within the detection range.

3) Varying the Score threshold. In the last step of the PRTP algorithm, the parameter \( \theta \) directly determines whether a trajectory is anomalous or not, if the Score of the trajectory is greater than \( \theta \), it is defined as anomalous, otherwise the trajectory is defined as normal. Therefore, it is necessary to study the influence of the value of \( \theta \) on the performance of the algorithm. Fig. 9 shows the distribution of the Score values of the trajectories on the two datasets according to equation (13).

From Fig. 9, it can be seen that most of the Score values on datasets T-1 and T-3 are less than 0.6, so based on the density distribution of Score values in the range of 0 to 0.6, 0.3 can be set as the lower limit of the range of \( \theta \). Moreover, the Score values of a small number of trajectories are distributed in the range of 0.6 to 1.0, and these points are likely to be the scores of anomalous trajectories, so 0.8 can be selected as the upper limit of the range of \( \theta \). Therefore, the article focuses on the effect of \( \theta \) in the range of 0.3 to 0.8 on performance.

It can be seen from Fig. 10 that when the range between 0.3 and 0.6, as the threshold \( \theta \) increases, the \( R \)-Score of anomaly detection on datasets T-1 and T-3 gradually increases. Then, the \( R \) gradually decreases with the increase of \( \theta \), and when \( \theta = 0.6 \), the \( R \) achieves the maximum within the detection range of \( \theta \). Meanwhile, the \( F \)-Score shows the same variation trend, and the maximum value is obtained when \( \theta = 0.6 \). Therefore, considering the variation trend of the \( R \) and \( F \)-Score, \( \theta = 0.6 \) is chosen as the optimal value of the anomaly threshold.

Finally, through the above analysis of the three variable parameters and the variation trends of \( R \) and \( F \)-Score, this paper determines the optimal values of these parameters as \( M = 400 \times 400, k = 6, \) and \( \theta = 0.6 \).

C. Visual Display of Anomalous Trajectory Detection

Anomaly detection is performed under the optimal values of the parameters. To understand the visualization result more clearly, this subsection selects some trajectories on the two largest datasets T-1 and T-3 for experiments to investigate the accuracy of PRTP. Fig. 11 shows the visualization result of anomaly detection on the real taxi trajectory datasets, where the solid blue lines represent all trajectories in the selected dataset and the solid orange lines show the anomalous trajectories detected by the PRTP algorithm.

It can be seen from Fig. 11(b) and Fig. 11(d) that the PRTP algorithm can fully detect obvious detour anomalous trajectories. Since the PRTP algorithm comprehensively
considers the moment of the trajectory point, the time span of the trajectory, and the different frequency values in the detection process, it can correctly detect the local detour anomalous trajectories and global detour anomalous trajectories at different time periods. Therefore, these visualization results confirm the effectiveness of the PRTP algorithm.

**D. Comparative Evaluation**

In the comparative experiments in this paper, to further verify the superiority of the PRTP, the study takes the ATDC [23], TPRO [32], and iBAT [24] as baselines to compare with PRTP. The basic idea of ATDC is to process trajectories by the grid. Then, it uses a distance calculation method to study different anomalous trajectory patterns and adopts an anomalous trajectory detection and classification method for real trajectory data. The iBAT algorithm employs an isolation mechanism to find anomalies in trajectories. Meanwhile, the basic idea of TPRO is to calculate the similarity between trajectories using edit distance in the spatiotemporal dimension. Therefore, the trajectory processing methods and research problems of these three algorithms are similar to this study.

Here, this module focuses on evaluating and comparing the anomaly detection effectiveness of the four algorithms by **F-Score** and **Recall**. From Fig. 12, it can be seen that among the baseline algorithms, ATDC achieves a higher **F-Score** in T-2, which indicates that it has a good effect on the detection of global spatiotemporal anomalies. In T-1 and T-3 with more peak trajectories, TPRO shows good detection results due to its spatiotemporal sensitivity. And compared with the baseline algorithms, the PRTP algorithm obtains the highest **F-Score** on all datasets, and has more stable detection results.

Similarly, from Fig. 13, the **Recall** values of the four algorithms show the same trend, PRTP obtains superior detection results on all datasets. It can be seen that the **Recall** of the PRTP algorithm is the most ideal, while the **Recall** of the iBAT algorithm is the lowest, indicating that the iBAT algorithm more incorrectly detects anomalies as normal occurrences on the datasets. Meanwhile, the experiment results show that the PRTP algorithm can play a good role in detecting global anomalies as well as complex local anomalies. Therefore, this group of comparison experiments shows that PRTP can better detect anomalous trajectories compared to the baseline algorithms.

In addition to comparing the detection effectiveness of the four algorithms, this paper further compares the time cost of these algorithms. Among them, each algorithm runs ten times in the same environment and the average of the running times is taken as the final result. From Fig. 14, it can be seen that the PRTP algorithm also has an absolute advantage in time consumption. The time loss of ATDC is also ideal, but for the TPRO algorithm, since it chooses a small time interval when dividing the time period, it consumes more time during the whole algorithm detection.

Therefore, with optimal values of the parameters, PRTP achieves high values of both **F-Score** and **Recall** for anomaly detection results on the datasets. Meanwhile, the experimental results also show that the PRTP algorithm has a significant advantage in time loss.
VI. DISCUSSION

In different application scenarios, most of the existing anomalous trajectory detection methods do not adequately integrate the location, time, and structure of the trajectories, and also ignore the influence of the time period on the route. Traditional algorithms usually compare the similarity of trajectories only in spatial or spatiotemporal dimensions, which is not applicable to anomaly detection of spatial trajectories with complex distribution. Compared with previous studies, the PRTP algorithm fully considers the location, time period, and structure of the testing trajectory in anomaly detection. The trajectory dataset is divided into different groups according to the division of time periods. Through the distribution of trajectories in different groups, the popularity of the route in different time periods is accurately obtained. Among them, to exclude the interference of trajectories with consistent running trend but long time-consuming to the route popularity, this study proposes the calculation method of spatiotemporal frequency, which avoids defining the congested roadway as a popular route. In the anomaly detection stage, to not be limited to the distribution location of the grid cells, this paper combines the location matrix and structure matrix of the trajectory to make anomaly judgments, which effectively detect anomalous spatiotemporal trajectories and loop travel trajectories, and also accurately identify normal trajectories bypassing congested roads. Experiments on real trajectory datasets show that the method can effectively detect trajectory anomalies with higher accuracy. The method can be applied in urban traffic road condition detection and traffic management, providing a new detection scheme for trajectory data mining.

VII. CONCLUSION

To improve detection accuracy and efficiency, this paper proposes an anomalous trajectory detection method using popular routes in different traffic periods. First, the method grids the trajectories and uses mapped trajectories for the study. Second, different time periods are divided according to the distribution of traffic flow, and the spatiotemporal node frequency values are obtained by combining the trajectory attributes, to obtain the popular routes dynamically. Finally, the distance formula is proposed for trajectory anomaly detection by combining trajectory location and trajectory structure. The proposed method is validated on real taxi GPS data, and it shows remarkable performance in experiments. Also, the method shows its potential in innovative applications such as taxi driving fraud detection.

However, the proposed method has a few limitations. In detecting anomalous trajectories, only the spatiotemporal properties of the trajectories are considered, and the exact location where the trajectory anomaly occurs and the time when the anomaly ends cannot be determined online. Therefore, in the follow-up study, this aspect will be broadened. For example, it is possible to detect anomalous sub-trajectories by combining time windows or decay factors while considering spatiotemporal attributes, to accurately locate where the anomalies occur and classify the anomalies of the trajectories based on the result.

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The Model of Stroke Rehabilitation Service and User Demand Matching

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Abstract—This article focuses on matching stroke rehabilitation services, and patient needs through the interconnection between patient demand and rehabilitation service capabilities. A solution is proposed based on the KJ, fuzzy AHP, and QFD methods to address this problem. Specifically, the KJ method categorizes user needs, and the fuzzy AHP method calculates weights and rankings. Furthermore, rehabilitation service capability indicators are developed, and the QFD method is applied to match customer needs with rehabilitation service capability indicators. The service indicator value is constructed through mapping relationships, and the rehabilitation service capability value is obtained by adding up the results. The best matching scheme is predicted by comparing rehabilitation service capability values of service alternatives. The success of the model has been proven by examining the case. It has helped patients and service organizations find suitable caregivers. The research results illustrate that the proposed model can effectively address the problem of stroke rehabilitation services and patient needs matching and has practical value and potential applications. Therefore, this research is significant in enhancing the quality of stroke rehabilitation services and patient satisfaction and provides a reference value for future studies of similar issues.

Keywords—Stroke; rehabilitation services; user needs; matching model

I. INTRODUCTION

Stroke is a significant public health issue worldwide, significantly burdening patients and society. Early post-stroke rehabilitation is crucial, and studies have shown that inpatient rehabilitation within 14 days after stroke onset could significantly improve daily living activities capabilities [1]. Stroke rehabilitation services accompany the entire rehabilitation process, and effectively meeting patient needs during the rehabilitation service delivery is essential to improve the quality of life and rehabilitation outcomes. Service and demand matching is a critical issue in effectively meeting patient needs.

Researchers have studied user needs matching for the service and demand matching problem. For example, Demain et al. [2] found that understanding and meeting patient needs is vital to alleviating treatment burden, and rehabilitation service providers need to focus on patient's psychological, social, and lifestyle needs. Reichert et al. [3] provided valuable insights into meeting cultural rehabilitation needs and improving the existing rehabilitation services by implementing workplace interventions, specifically through community-driven collaborative needs assessment that facilitated the establishment of responsive rehabilitation institutions. Joshi [4]

emphasized the significance of patient-centred care in the success of outpatient surgeries, emphasizing the need for a clear understanding of the diverse needs of different patients. Lu et al. [5] investigated stroke patients' fundamental psychological needs, satisfaction, and influencing factors, focusing on the autonomy, competence, and relatedness aspects of patient needs. The study found that stroke increased patients' satisfaction with their relatedness needs while decreasing their autonomy and competence needs. Saut et al. [6] discussed the mechanisms and structures of patient and family participation in healthcare systems, indicating that reforming open culture and processes may promote service quality. Reforming open culture includes improving service quality through information sharing between patients and professional service providers and achieving a balance of rights between patients and service providers. Trivedi et al. [7], from the perspective of nursing physicians, promoted a smooth transition in post-discharge care by providing nursing education before patient discharge, which helped patients understand their own needs and thus promoted the quality of rehabilitation care.

Therefore, the current ways to match healthcare services with user needs include: providing more appropriate services by understanding factors such as patients' cultural backgrounds, religious beliefs, socioeconomic status, and living environments; continuously optimizing and adapting services through patient feedback and assessment; establishing good communication and trust with family members to better understand patients' needs and provide better services; promoting communication between patients and service providers, educating them about healthcare knowledge, and promoting patients' understanding of their own needs and the positive attitude of service providers. However, there are still many unmet needs in healthcare globally [8]. These unmet needs are because the focus is solely on users' needs in the service-demand relationship, neglecting the service providers' capabilities assessment. The lack of quality in healthcare services is due to insufficient research on user needs and the incorrect evaluation of their service capabilities.

In existing research, the only evaluation of service providers in the context of rehabilitation services is often their caregiving ability. Kardami et al. [9] demonstrated the feasibility and functionality of using a cancer scale to evaluate the caregiving ability of mothers of children with cancer, which can identify the nursing needs of pregnant women. The measurement scale includes five dimensions: effective role-playing, fatigue and subservience, trust, uncertainty, and ignorance of caring for children with cancer. The results showed that the scale had acceptable content, scope, construct
validity and sufficient reliability. The cancer scale can be used to evaluate the caregiving ability of mothers of children with cancer in Iran. Zhong et al. collected data through the Caregiver Burden Inventory (CBI) and the World Health Organization Quality of Life to construct a caregiver difficulty scale to evaluate the burden on caregivers and the impact on the health of disabled children [10]. O’Malley et al. used the Caregiver Reaction Scale (CRS) to evaluate positive aspects of family caregiving experiences, such as role conflict, challenge, and caregiving, as a multidimensional measure [11].

Thus, the current research on evaluating caregivers’ abilities is solely based on their reactions to various tasks, ignoring the link between patients’ needs and caregivers’ abilities. This study aims to develop a stroke rehabilitation service and user needs a matching model that considers the interplay between patients’ needs and rehabilitation service capabilities. The aim is to offer more adaptable services for patients and rehabilitation institutions.

Compared to the statistical methods used in existing rehabilitation service research, the novelty of this study lies in constructing a resolution through a computational model, predicting service outcomes to enhance service quality and efficiency:

Firstly, this research employed the KJ method and fuzzy analytic hierarchy process (FAHP) to construct user demand indicators and assign weights to sort their importance.

Secondly, this research constructed service capability indicators.

Thirdly, this research mapped user demands to service capabilities using Quality Function Deployment (QFD) and used the value of the caregivers’ comprehensive ability as the metric to evaluate rehabilitation service capabilities, aiding patients and service providers in matching appropriate service plans.

Finally, this research conducted a case analysis to validate the proposed model’s feasibility. Through this case study, researchers demonstrated the feasibility of the model. Researchers helped patients and service providers find suitable caregivers that match patients' needs. Through this research, researchers hope to provide theoretical and practical guidance to optimize rehabilitation services for stroke patients and foster the development of the rehabilitation service industry.

In this thesis, Section I introduces the research problem. Section II briefly discusses related work. Section III describes the primary user needs and care service matching model. Section IV applies the model to a case study involving the selection of caregivers for stroke patients and compares its predictions with actual selection outcomes. Finally, Section V discusses the results of the case study, and Section VI concludes the work.

II. RELATED WORK
A. Fuzzy Analytic Hierarchy Process (FAHP)

The Fuzzy Analytic Hierarchy Process (FAHP) is a multi-criteria decision-making method based on fuzzy mathematics theory, initially proposed by Professor Thomas L. Saaty of Northeastern University in 1970 [12]. It builds on the Analytic Hierarchy Process (AHP). It can convert fuzzy and uncertain information into mathematical quantities, facilitating analysis and processing.

Singh and Özşahin developed an interval fuzzy analytic hierarchy process (IFAHp) model to investigate the key factors influencing the selection of wooden outdoor furniture. They subdivided five major factors into sub-factors and conducted pairwise comparisons, using fuzzy numbers to represent the relative importance of each factor. By computing the weights of each factor, the researchers determined their relative importance in the overall hierarchy. Ultimately, they obtained a prioritization of factors to evaluate and prioritize outdoor furniture selection [13].

Xuan et al. developed an expert automatic cutting tool selection system using an integrated fuzzy analytic hierarchy process approach. They conducted a fuzzy hierarchy analysis evaluation of the distinguishing factors of the cutting tools to determine the priority of tool types. Then they selected the optimal tool type through sorting [14].

Wang proposed a feature-driven triangular fuzzy hierarchy process that obtained a support interval-based triangular fuzzy hierarchy process from the fuzzy multiplication preference relations matrix by establishing three positive definite matrices of feature problems and linear programming based on feature vectors. Additionally, a method for acceptability checks was suggested [15].

In summary, the fuzzy analytic hierarchy process is valuable for assigning weights and ranking relevant factors through hierarchical division and fuzzy evaluation of sub-factors in complex and multi-criteria screening problems. The differences between studies lie in the formulation of fuzzy rules. The fuzzy analytic hierarchy process was used to assign weights and rank the relevant factors of user needs and rehabilitation service capabilities for this paper's matching problem discussed of the stroke rehabilitation service and patient needs. This method helped to address the uncertainty and fuzziness problems in the analysis process of user needs and rehabilitation service evaluation factors, resulting in a reliable analysis process.

B. Quality Function Unfolding Method

Quality Function Deployment (QFD) is a method that converts customer requirements into product design specifications, aiming to enhance product quality and customer satisfaction by incorporating customer needs into product design and services. The core of QFD involves constructing and analyzing a series of matrices to transform customer requirements into product design specifications, which are then reflected in various aspects of the product.

Kürüm Varolgüneş et al. utilized QFD and AHP to design a thermal hotel building that meets customer needs and improves the quality of building design. The QFD method was employed to gather customer needs and opinions. In contrast, the AHP method evaluated different design options and selected the best one. Combining these two methods can provide a more comprehensive and accurate understanding of customer needs.
and improve the satisfaction and practicality of the building [16].

Baskir proposed a QFD-AHP model for evaluating after-sales services of automobiles based on customer beliefs. This model converts customer feedback into technical features in lean implementation. Managers, employees, and customers can use belief space evaluation to identify their perception mechanisms and experience how the uncertainty of perception mechanisms affects their decision-making. This model provides a more accurate and comprehensive solution by eliminating ambiguity in poor decision-making based on conceptual change [17].

Shan et al. proposed an optimized design framework combining the quantitative Kano model and fuzzy quality function deployment (QFD) model to best match enterprise service elements under uncertain and imprecise judgment information. This approach improves the quality and customer satisfaction of express delivery services [18].

Combining QFD and other methods, such as AHP and Kano model, provides a more comprehensive and accurate product design and service optimization solution. This application effect highlights the practicality of QFD in helping businesses understand customer needs and convert them into specific product design requirements, thus improving product quality and customer satisfaction. Additionally, combining different methods can further enhance the accuracy and comprehensiveness of analysis and decision-making. This study’s combination of FAHP and QFD is feasible for matching the needs of stroke patients with rehabilitation services.

C. Health Care Service Capability

Current research on healthcare service capabilities has primarily focused on evaluating the general abilities of caregivers. Factors such as self-pressure, health management, coping strategies, age, financial status, social issues, self-efficacy, and disease severity have been found to affect caregiver ability [19-20]. However, such evaluations need more direct relevance.

Other studies have shown that caregiver participation in patients’ rehabilitation exercises benefits their physical and mental health [21]. While existing research helps to diversify methods related to patient rehabilitation and maintaining caregivers’ physical and mental health, it does not provide insights into matching patients with convenient care services.

Research on healthcare services from the perspective of patients and therapists includes rehabilitation therapy methods [22], assessment of rehabilitation outcomes and impacts [23], and the duration and contribution of rehabilitation interventions. However, such research weakens caregiver involvement.

Therefore, it is necessary to propose a model for matching stroke rehabilitation services with user needs from the perspective of patient needs and healthcare service capabilities. This model will provide valuable insights into improving the quality and effectiveness of stroke rehabilitation services.

III. METHODOLOGY FOR THE CONSTRUCTION OF A MATCHING MODEL BETWEEN USER NEEDS AND HEALTH CARE SERVICES

A. User Requirements Acquisition

The matching model between user needs and care services is shown in Fig. 1. A combination of questionnaires, observation, and the Delphi method is used to gather demand information to ensure accurate user requirements. The questionnaire survey method often has low user participation and recall rates, but this can be improved by conducting quick surveys under typical situations. However, users’ erroneous responses cannot be resolved. The observation method provides first-hand information but is time-consuming. The Delphi method requires achieving a specific consensus rate in each round through expert evaluations and has limited avenues to obtain indicators [24]. Combining these methods reduces their deficiencies. Since most stroke patients have post-stroke sequelae, it is necessary to interview family members and patients and observe their behaviour to obtain primary needs. The most critical demand indicators are screened by requiring the patient’s rehabilitation therapist and the primary doctor to act as Delphi method experts.

![Fig. 1. Matching model of user needs and rehabilitation services.](image-url)
B. Fuzzy Hierarchical Analysis Deals with Demand Indicators

1) KJ Method stratified user requirements indicators:
Clustering analysis and the KJ method are commonly used for hierarchical requirement analysis. While clustering analysis has the advantage of concise and intuitive data processing, it requires a large sample size [25]. On the other hand, the KJ method is not limited to numbers and is also suitable for text descriptions [26]. Therefore, in this study, researchers opted for the KJ method to classify user requirements hierarchically.

The KJ method is typically conducted in the form of cards. However, researchers encoded the indicators in this study instead of distributing cards. We asked the respondents to use the same code for similar information. Participants, including rehabilitation teachers, nurses, and some patients, were invited to participate in a group study using the KJ method. Researchers finally divided 24 indicators into 3 levels and 4 modules, as shown in Fig. 2.

2) Fuzzy hierarchical analysis (FAHP) quantitative steps
a) Defining fuzzy rules: To achieve precise and accurate outcomes with minimal requirements for periodic modifications to data consistency, we defined a fuzzy scoring interval of [0,1], with adjacent score differences of 0.1. The final score was then selected from the three available values {0.0, 0.5, 1} (referred to as "three values" in this study). Specifically, the initial score was compared with {0.0, 0.5, 1}, and the value in the three values closest to the actual value was chosen as the final score, resulting in a new fuzzy rule.

b) Demand stratification: Using the KJ method, the research team obtained the target layer G, the quasi-measurement layer Z, and the indicator layer E, forming a three-level fuzzy analytic hierarchy. Researchers first used the three values to construct a fuzzy priority consensus matrix. Then researchers generated a fuzzy judgment consensus matrix using an improved fuzzy analytic hierarchy process [27].

To avoid the need for further consistency evaluations, we took the following measures:

Firstly, to establish the consistency matrix for constructing the fuzzy priority relationship, we used the three values {0,0.5,1}, which facilitated explicit judgments and avoided confusion. First, we constructed a fuzzy priority relationship matrix $a_{ij}$, with each position labelled by $i$ and $j$, where $i$ represents the row number and $j$ represents the column number. In the three values, 0 indicates that $j$ is more critical than $i$, 0.5 indicates that $i$ and $j$ are equally important, and 1 indicates that $i$ is more critical than $j$. The consistent matrix factor for the fuzzy priority relationship is obtained by summing the rows to obtain $t_i$ and the columns to obtain $t_j$, where $i = \{1, \cdots, s\}$ and $j = \{1, \cdots, n\}$, with $s$ representing the number of rows in the matrix and $n$ representing the number of columns. The relationship between the factors in the matrix can be expressed as:

$$a_{ij} + a_{ji} = 1$$  \hspace{1cm} (1)

Secondly, we converted the fuzzy priority relationship consensus matrix into a fuzzy judgment consensus matrix. The formula for constructing the factor $b_{ij}$ in the fuzzy judgment consistency matrix is:

$$b_{ij} = \frac{r_i - r_j}{2s} + 0.5$$  \hspace{1cm} (2)

Thirdly, we calculated and ranked the weights.

The first step was to calculate the product of the elements in each row of the judgment matrix and then calculate the $s$-th root of the product. Next, we normalized the eigenvector to obtain the weight vector $W_P$. The weight vector $W_E$ represents the relative importance weight of each indicator for the upper quasi-criterion factor, and $W$ is the integral vector. Finally, the ranking was based on the integral weight, and the formula is as follows:

$$M_i = \prod_{j=1}^{s} b_{ij}$$  \hspace{1cm} (3)

$$W_P = \sqrt[\pi]{M_i}$$  \hspace{1cm} (4)

$$W_i = \frac{W_P}{\sum_{i=1}^{n} W_i}$$  \hspace{1cm} (5)

$$W = W_Z * W_E$$  \hspace{1cm} (6)

![Fig. 2. KJ affinity diagram.](image-url)
c) Fuzzy Analytic Hierarchy Process (FAHP) was used to calculate weights and rankings.

The expert group was asked to make critical judgments and calculate the average value. Based on the fuzzy rules of this study, the most suitable value was selected from the three values, and the fuzzy priority relationship consistency matrix was constructed and converted into a fuzzy judgment consistency matrix. The fuzzy consistency matrices for the goal layer G, quasi-measurement layer A, quasi-measurement layer B, quasi-measurement layer C, and quasi-measurement layer D are shown in Tables I to V. The indicator layer weight $W_k$ and criterion layer weight $W_2$ were calculated using formulas (4) - (6). The total weight $W$ was obtained by multiplying the indicator and criterion layer weights, as shown in Table VI.

C. Construction of Indicators for the Evaluation of Health Care Services

After four months of in-depth research in a rehabilitation hospital in Shenzhen, China, semi-structured interviews were conducted with 25 nursing staff and 51 patients and their family members. The results were analyzed, summarized, and organized. Focus group discussions were also held with several frontline rehabilitation therapists. The results showed that the duration of the patient's illness, cultural level, and cooperation level can affect the rehabilitation outcomes. The cultural level, body weight, lifestyle habits, health philosophy, coping skills, time management skills, psychological quality, professional spirit, and caregivers' communication skills can also affect the rehabilitation results.

Compared with existing studies on the comprehensive ability factors of family caregivers, disease-related knowledge, daily and disease-related care skills, coping strategies, self-stress, and health management are used as standards. Some scholars have also added hope level [28] and readiness level [29]. Some scholars aim to improve the effectiveness of rehabilitation nursing interventions by focusing on the sense of interest of caregivers [30]. Understanding disease, daily and disease-related care skills belong to professional abilities; coping strategies are problem-solving skills; self-stress is a psychological quality, and health management is a health concept and lifestyle habit. Hope level and readiness levels are professional ability and time management skills, and the interest of caregivers is a psychological quality. High-quality nursing and efficient care are preparations for good quality of life for patients after discharge.

In contrast, nursing staff can improve the cooperation level of patients through psychological intervention. The key to joint efforts by nursing staff and patients is setting realistic rehabilitation expectations [31]. Patients understand their disease correctly and actively cooperate, leading to a healthy life. Patients and caregivers need to create reasonable rehabilitation expectations. Caregivers should keep in mind the patient's rehabilitation goals. In order to clarify the influencing factors, nursing staff, work, patients, and diseases should be studied separately. The patient's cooperation level can represent the caregiver's communication and coping abilities. The indicators of the final rehabilitation serviceability are physical health (I1), lifestyle habits (I2), health philosophy (I3), coping skills (I4), time management skills (I5), psychological quality (I6), professionalism (I7), and communication skills (I8), as shown in Fig. 3.

D. Mapping of User Needs and Wellness Service Capabilities

QFD is often used in product design to convert user needs into technical product characteristics [32]. The user mapping relationship determines the technical importance of needs indicators and technical features. In this study, a scoring method was used to evaluate the rehabilitation service capability under different user needs indicators, and then the weight of user needs indicators was multiplied by the score of rehabilitation service capability indicators. The scoring uses three values {0,0.5,1}, where 0 represents a low service capability match, 0.5 represents a moderate service capability match, and 1 represents a high service capability match.

A correlation was established between user needs indicators and rehabilitation service capability indicators based on the significance score of the latter. The rehabilitation service capability is measured by the total ability value $VA$ of the caregiver, and the caregiver with the highest $VA$ value can be regarded as the best match. The scoring of the caregiver's ability indicators is evaluated by the hospital or caregiver service institution based on the matching ability of the user needs indicators, forming a fuzzy matrix $J$. $W_n$ is the weight of user needs indicators, where $n$ is the number of demand indicators and $\in R^+$. The caregiver's ability indicator is $P_k$, where $k$ is the number of comprehensive ability indicators of caregivers, and $k \in R^+$. The matrix $J$ represents the mapping of user needs and caregiver's ability indicators:

$$J = \begin{bmatrix} W_1 & P_{11} & \ldots & P_{1k} \\ \vdots & \vdots & \ddots & \vdots \\ W_n & P_{n1} & \ldots & P_{nk} \end{bmatrix}$$

(7)

The calculation method of the total ability value $VA$ of the caregiver based on the fuzzy matrix $J$ is as follows:

$$VA = \sum_{k=1}^{K} \sum_{n=1}^{N} W_n * P_{nk}$$

(8)
IV. CASE STUDY: CHOICE OF CARER FOR A STROKE PATIENT

A. Description of the Problem

The feasibility of the matching model proposed in this study was demonstrated using the example of a patient in a rehabilitation hospital in Shenzhen, China, who needed to select a caregiver. The patient was paralyzed due to a stroke and required bedridden care, feeding, urinary catheterization, and other rehabilitation services. As the patient's family members were busy with work, they needed to hire a caregiver with a high degree of matching ability to provide the necessary care. The rehabilitation hospital collaborated with several caregiving service institutions and provided six caregivers for selection. The patient's family members could use the matching model proposed in this study to select the best caregiver based on the patient's demand indicators and the caregiver's ability indicators.

B. Test Rules

This model selects the most suitable caregiver by calculating the comprehensive ability value of six caregivers based on the patient’s demand indicators. Then, the six selected caregivers provided one week of rehabilitation services based on their numbers. The patient or their family can evaluate the caregivers' performance during the week and select the most suitable caregiver to continue service. The model's feasibility is determined by comparing the results obtained from the model calculation and essential service. The matching model can be feasible if the caregiver number matches between the two. If the number does not match, an error analysis is conducted using the caregiver's comprehensive ability value. If the error is small, the model can be revised and used; if the error is too large, the model needs to be modified to meet the actual needs better, or it will not meet the actual requirements.

C. Test Method

According to the architecture of the model shown in Fig. 1 and the framework of user demand indicators and caregiving ability indicators built in Section III, six nurses and four rehabilitation therapists were invited as evaluation experts and were evenly divided into two groups: a model group and an experimental group. The model group mainly participated in asking about user requirements, scoring user demand indicators, and scoring the caregiving ability indicators of the six caregivers provided by the rehabilitation institution to provide input for the model calculation. The experimental group mainly discussed and selected a suitable caregiver based on the effectiveness of the caregiver's services and the feelings of the patient and family members. The two groups of experts are not allowed to discuss with each other. The experimental process is shown in Fig. 4.

D. Calculation Results

User demand and caregiving service indicators were constructed, and demand weights and caregiving ability values were calculated based on formulas (1) to (8), as shown in Tables I to VII and Fig. 5. Table I to Table V represents pairwise comparison matrices and their respective weight values for user requirement factors. Table VI represents the comprehensive weight and ranking of user requirement factors. Table VII represents the scores for healthcare service indicators corresponding to user requirements and the total ability value of the caregiver. Fig. 5 represents the comprehensive capability values of different caregivers. Finally, caregiver No. 1 had the highest matching degree, with a comprehensive ability value of 3.320. After the six caregivers provided caregiving services for one week each, the experimental group and the patient's family discussed and decided to keep caregiver No. 1, consistent with the model's prediction.

![Test process](Fig. 4)

![Results of the capacity matching model for rehabilitation services](Fig. 5)

**TABLE I. G-RELATIONSHIP MATRIX - FUZZY MATRIX AND ITS WEIGHTS \(W_G\)**

<table>
<thead>
<tr>
<th></th>
<th>A</th>
<th>B</th>
<th>C</th>
<th>D</th>
<th>W_G</th>
</tr>
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<tr>
<td>A</td>
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<tr>
<td>B</td>
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<td>1.000</td>
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<td>0.500</td>
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<td>C</td>
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<td>0.167</td>
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</tr>
<tr>
<td>D</td>
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**TABLE II. B-RELATIONSHIP MATRIX - FUZZY MATRIX AND ITS WEIGHTS \(W_B\)**

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<tr>
<th></th>
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<th>B3</th>
<th>B4</th>
<th>B5</th>
<th>W_EB</th>
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</thead>
<tbody>
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<td>0.000</td>
</tr>
<tr>
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<td>0.125</td>
</tr>
<tr>
<td>B3</td>
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<td>0.833</td>
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<td>0.386</td>
</tr>
<tr>
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<td>0.750</td>
<td>0.625</td>
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<td>0.309</td>
</tr>
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<tr>
<td>D6</td>
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<td>9.000</td>
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</table>

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### Table III. C-Relationship Matrix - Fuzzy Matrix and Its Weights $W_{EC}$

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<tr>
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<th>C</th>
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<th>C3</th>
<th>C4</th>
<th>C5</th>
<th>$W_{EC}$</th>
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### Table IV. D-Relationship Matrix - Fuzzy Matrix and Its Weights $W_{ED}$

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### Table V. A-Relationship Matrix - Fuzzy Matrix and Its Weights $W_{EA}$

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<th>A3</th>
<th>A4</th>
<th>A5</th>
<th>A6</th>
<th>A7</th>
<th>A8</th>
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### Table VI. Weighting and Ranking of Rehabilitation Care User Needs

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<tr>
<th>Criteria layer (weights)</th>
<th>Indicator layer</th>
<th>Weights</th>
<th>Combined weights</th>
<th>Combined ranking</th>
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<td>0.041</td>
<td>0.014</td>
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<td></td>
<td>A2</td>
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<td>A3</td>
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<td></td>
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<tr>
<td></td>
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<tr>
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<td>0.000</td>
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<tr>
<td></td>
<td>C4</td>
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<tr>
<td></td>
<td>C5</td>
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<td>Consumables Supplied D (0.22)</td>
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V. DISCUSSION

In this discussion, researchers try to explore the advancements and significance of our research. The experimental results indicate that the actual selection results are consistent with the predicted results, suggesting the feasibility of the model.

From the calculation data, the patient's demand weight ranking shows that due to being bedridden from paralysis, the patient has a higher demand for living care and rehabilitation training. Therefore, the requirements for service focused on strengthening communication skills, health concepts, caregiver physical fitness, and living habits. The model has performed well in caregiving services, especially caring for patients, through the joint efforts of nurses, rehabilitation therapists, family members, and researchers.

Compared to existing research [6 - 7], the advanced nature of this study lies in its innovative approach, which evaluates and matches user needs and service capabilities from the perspective of the linkage between the two. This mold is done through various methods to meet patients' needs better and improve the quality and efficiency of care services.

In contrast to existing research on demand and capability matching, which uses patient rehabilitation needs quantification [33] and mainly relies on physician experience to predict individual patient discharge pathways, this study stratifies user demand factors, assigns weights, and maps them to service factors. It calculates the degree of match between predicted user needs and services and involves doctors, patients, and third-party service providers in the prediction process. The model developed in this study is more comprehensive in considering the matching relationship and is, therefore, more advanced than existing research.

Moreover, this model has demonstrated practicality in the case study, and we believe it provides more flexible service guidance for patients and rehabilitation institutions. However, the current limitation of this model lies in the fact that its accuracy and reliability depend on the reliability and precision of the data samples. To further improve the model's adaptability, it is necessary to attempt more data analysis.

VI. CONCLUSION

The stroke caregiving service and user demand matching model proposed in this study constructed user demand indicators using the KJ method and fuzzy analytic hierarchy process. They constructed caregiving ability indicators from a
service perspective. The model mapped the matching relationship between user demand and caregiving ability indicators using QFD and used the caregiver comprehensive ability value to measure caregiving serviceability evaluation to match suitable caregivers for patients and service institutions and improve the quality and efficiency of caregiving services.

Future research can further optimize the evaluation indicators and parameters of the model to improve its accuracy and reliability. Secondly, this research only focused on stroke caregiving services, and future research can apply this model to other rehabilitation service fields to further verify its practicability and feasibility.

REFERENCES


Detection of Protective Apparatus for Municipal Engineering Construction Personnel Based on Improved YOLOv5s

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Abstract—With the rapid economic development, the government has increased investment in municipal construction, which usually takes a long time, involves many open-air operations, and is affected by cross-construction, traffic, climate and environment, and so on. The safety protection of urban construction workers has been a concern. In this paper, an improved algorithm based on YOLOv5s for the simultaneous detection of helmets and reflective vests is proposed for municipal construction management. First, a new data enhancement method, Mosaic-6, is used to improve the model's ability to learn local features. Second, the SE attention mechanism is introduced in the focus module to expand the perceptual field, strengthen the degree of association between channel information and the detection target, and improve the detection accuracy. Finally, the features of small-scale targets are interacted and fused in multiple dimensions according to the Swin transformer network structure. The experimental results show that the improved algorithm achieves accuracy, recall, and mean accuracy rates of 98.5%, 97.0%, and 92.7%, respectively. These results show an average improvement of 3.4 percentage points in mean accuracy compared to the basic YOLOv5s. This study provides valuable insights for further research in the area of urban engineering security and protection.

Keywords—YOLOv5s; hard hat; reflective vest; simultaneous detection

I. INTRODUCTION

With the relentless advancement of science and technology, Artificial Intelligence (AI) has emerged as a key player in numerous fields. One such field is municipal engineering, where ensuring the safety of construction workers is of paramount importance. Wearing helmets and reflective vests is an essential measure to ensure their well-being. Combined with artificial intelligence target detection method, it can realize automatic detection and monitoring of whether construction workers are wearing helmets and reflective vests. It is necessary to use artificial intelligence target detection method to detect municipal engineering construction workers wearing helmets and reflective vests [1-3]. First of all, AI target detection method can improve the efficiency and accuracy of site safety monitoring. Traditional safety monitoring methods usually rely on manual inspection, there is a waste of human resources, and blind area coverage is not complete [4]. The use of AI target detection methods, combined with cameras and image processing technology, can monitor the construction site in real time and automatically detect whether there are construction workers wearing helmets and reflective vests. This automated monitoring method can not only reduce the burden of site managers, but also detect and deal with violations in a timely manner, improving the overall safety level of the site [5]. In addition, the AI target detection method can reduce the interference of human factors on safety monitoring results. In traditional safety monitoring, manual judgment can be affected by factors such as subjective awareness, perspective limitations and fatigue, resulting in inaccurate or missing detection results for construction workers wearing helmets and reflective vests. In contrast, AI target detection methods use advanced algorithms and models to quickly and accurately detect target objects without the interference of subjective factors [6, 7]. This ensures more objective and accurate monitoring results for construction workers wearing helmets and reflective vests, effectively improving the reliability of safety monitoring.

The remaining sections of this paper are organized as follows. In Section II, we introduce two deep learning-based detection algorithms and discuss the current state of research in this area. Section III elaborates the basic architecture of the YOLOv5 model and details the optimization part for YOLOv5. Section IV of this paper provides an overview of the dataset sources, the experimental environment, and the model parameter settings. In Section V, we present the experimental results and discuss the performance of the optimized model introduced in the previous section. Experimental tests are conducted for the three optimized parts and compared with the current mainstream algorithms.

II. CURRENT STATUS OF RESEARCH

A. Introduction of Deep Learning Algorithm

Deep learning is an advanced and complex machine learning algorithm in the field of AI, which has the advantages of powerful learning ability, high generalization ability, and wide applicability. Deep learning based algorithms in the field of target detection often use a technique where anchor boxes of different sizes are placed on the image to achieve target detection, regressing and classifying the anchor boxes. Based on the way of regression box generation, target detection algorithms can be divided into two main categories: two-stage algorithms and one-stage algorithms [8].

Well-known two-stage target detection algorithms include R-CNN, Fast R-CNN, Faster R-CNN, etc. The two-stage target
detection algorithm is a commonly used target detection method, which consists of two main stages: candidate image generation (region proposal) and target classification and localization [9].

Region Proposal Phase: First, a series of candidate frames (often called candidate regions or candidate frames) that may contain targets are generated from the input image by using some advanced techniques such as selective search, Edge Boxes, or deep learning-based methods [10]. The goal of this stage is to generate as many candidate frames as possible to cover the potential targets in the image.

Target Classification and Pinpointing Stage: In this stage, a target classifier is applied to each candidate frame to determine whether it contains the target of interest and to pinpoint the target, i.e., to determine the exact bounding box location of the target. For each candidate frame, the classifier discriminates its extracted features and determines whether it belongs to a specific target class. In addition, the target bounding box can be further fine-tuned to more accurately match the location and shape of the target.

In computer vision, single-stage target detection algorithms are widely used and can be classified as a common type of target detection algorithm. They contrast with another common class of target detection algorithms known as two-stage algorithms. One-stage target detection algorithms have the key feature of performing target detection and localization directly on the image itself. This approach is typically faster and more suitable for real-time application scenarios [11]. The following are some common representatives of single-stage target detection algorithms:

The YOLO (You Only Look Once) family: YOLO is a popular collection of single-stage target detection algorithms. By dividing the image into grids and predicting the location and class of targets within each grid, the YOLO algorithm effectively transforms the target detection problem into a regression problem. The YOLO algorithm performs target localization and classification simultaneously by a single neural network, which is faster, and YOLOv5 is the latest version of the YOLO series.

SSD (Single Shot MultiBox Detector): SSD is another popular single-stage target detection algorithm. The SSD algorithm uses multi-scale feature maps to detect targets of different sizes, and uses anchor boxes of different sizes for target position prediction. The SSD extracts image features through a convolutional neural network and performs target classification through subsequent convolutional layers and bounding box regression.

The following is a detailed description of the YOLO family of target detection algorithms (YOLOv1, YOLOv2, YOLOv3, YOLOv4, YOLOv5):

YOLOv1 is the first release in the YOLO series. YOLOv1 transforms the target detection problem into a regression problem by directly predicting bounding box locations and class probabilities through a single forward pass. A convolutional neural network (CNN) is used to extract features from the image and output predictions through a fully connected layer [12]. YOLOv1 has a fast detection speed, but performs poorly on small and dense targets and has limited effectiveness for scenes requiring high localization accuracy.

YOLOv2 incorporates several enhancements to improve the detection capabilities for small objects, such as the introduction of multi-scale prediction and the implementation of the anchor box mechanism. The network structure used in YOLOv2 is Darknet-19, and a novel loss function is employed to effectively deal with classification errors and bounding box regression errors [13]. YOLOv2 also proposes a data enhancement technique called Random Gamma Adjustment.

YOLOv3 introduced a number of improvements over YOLOv2, including the use of a deeper Darknet-53 network structure, the use of residual connectivity, the application of multi-scale prediction, and the use of more anchor frames. YOLOv3 introduced a Feature Pyramid Network (FPN) structure, which extracts feature maps at different scales and performs target detection. By using feature maps of different sizes for target detection, YOLOv3 can effectively detect targets of different sizes [14]. YOLOv3 achieves a better balance between detection speed and detection performance.

YOLOv4 further improves the network structure and training strategy to increase the detection accuracy and speed. CSPDarknet53 is adopted as the backbone network, and modules such as SAM (Spatial Attention Module) and PAN (Path Aggregation Network) are introduced to improve the feature representation and perception capabilities. YOLOv4 also uses multi-scale inference and multi-scale training strategies by combining different YOLOv4 also uses multi-scale inference and multi-scale training strategies to achieve more accurate target detection by combining feature maps of different scales. The GIoU (Generalized Intersection over Union) loss function is introduced to more accurately compute the overlap between bounding boxes [15].

YOLOv5 introduces some improvements in the network architecture with lighter model architecture while maintaining high detection performance. YOLOv5 introduces a new target detection architecture called CSPNet (Cross Stage Partial Network), which balances model size and performance by reducing computational complexity and improving accuracy. YOLOv5 also introduces more data augmentation techniques to improve model generalization. YOLOv5 achieves faster training and inference through model lightweighting and optimization. Overall, the goal of the YOLO family is to achieve real-time target recognition while balancing speed and accuracy. Each version of YOLO has its own unique contributions and enhancements to improve detection performance and efficiency.

B. Current Status of Research

There have been many researchers who have studied helmet detection to propose various optimized detection methods for the above algorithms, for example, Chen et al. presented a novel integration of Retime image enhancement technology into the Faster R-CNN framework. They addressed the challenges posed by various factors such as lighting conditions and distance, which often hinder accurate detection. To overcome these obstacles and improve detection performance, they employed the K-means++ algorithm. By
using this approach, they achieved automatic helmet detection with improved accuracy [16]. Guo et al. added a VGG16 feature extraction module to Faster R-CNN to determine whether the helmet is correctly worn using Euclidean distance, which improved the detection accuracy and speed [17]. Song added the R-SSE module to YOLOv3, reduced the network depth, improved the network detection speed and accuracy, and the dual module Res 2 was used to improve the feature reusability and detection efficiency of small target [18]. Chen et al. used a lightweight network PP-LCNet to improve YOLOv4, reduced the model parameters with depth-separable convolution, used a new SlIoU loss function, reduced the model size, and improved the speed of helmet detection [19]. To improve the YOLOv5 detector, Jia et al. used triple attention fusion and soft NMS, which can achieve high accuracy and real-time results under complex weather conditions [20]. Jin et al. used K-means++ algorithm and Deep Coordinated Attention (DWCA) mechanism in YOLOv5 to enhance the information propagation between features and improve the accuracy of helmet detection [21].

In summary, although certain results have been achieved in the field of helmet research, there is little research in the field of reflective vest detection. All of the above studies have optimized and improved the algorithms to some extent, but most of the models are more complex and not easy to implement, or the detection accuracy is difficult to meet the demand. In addition, in the urban engineering construction environment, weather conditions and obstacles can significantly affect the reliability of detection, and more in-depth research is required to obtain a good detection model. This paper proposes to combine the two, studies the simultaneous detection and identification methods, and improves and optimizes the speed and accuracy of real-time detection, which can further protect the personal safety of municipal engineering construction personnel.

III. IMPROVED YOLOV5 MODEL ARCHITECTURE

A. Review of the Fundamentals of the YOLOv5 Model

Backbone Network: YOLOv5 uses CSPDarknet as its backbone network. CSPDarknet is a lightweight convolutional neural network that uses a CSP (Cross-Stage Partial) connection structure to divide the feature maps in the channel dimension and bypass some of the convolutional layers, thus reducing the computational complexity.

Neck Network: Instead of an explicit neck network structure, YOLOv5 introduces multiple cross-level connectivity (PANet) modules in the backbone network. These PANet modules perform spatial pyramidal pooling operations on feature maps at different scales to fuse semantic information at different levels.

Head Network: YOLOv5’s head network is responsible for generating predictions for target detection. It has three main components: Spatial Pyramid Pooling (SPP) module, feature pyramid pooling layer, and detection head. SPP Module: The SPP module performs multi-scale pyramid pooling operations on the feature graph to capture contextual information at different scales. Feature pyramid pooling layer: The feature pyramid pooling layer combines feature maps at different scales by upsampling and fusion to produce a feature map that incorporates information from multiple scales. Detection head: The detection head consists of a series of convolutional layers responsible for predicting the target’s bounding box location, category, and confidence score [22]. YOLOv5 uses a multi-scale prediction strategy, i.e., target detection is performed on feature maps at different levels to handle targets of different sizes.

Loss function: YOLOv5 uses a loss function called YOLOv5 Loss, which combines the loss calculation of target position, category and confidence.

In general, YOLOv5 is structured with a lightweight backbone network, cross-stage connectivity modules, and a multi-scale prediction strategy to achieve efficient and accurate target detection. It achieves a good balance between speed and accuracy for a variety of real-time or offline target detection tasks. The schematic structure of YOLOv5 is shown in Fig. 1.

![YOLOv5 principle structure diagram](image)
B. Improving the YOLOv5 Algorithm

1) Mosaic-6: In YOLOv5, a new data enhancement method called Mosaic has been introduced as a complement to the basic data enhancement techniques. The primary concept behind this method is to randomly crop and scale four images, which are then combined in a random order to create a single image. This method not only enriches the data set, but also increases the number of small sample targets, thus improving the training speed of the network. An important advantage of using the mosaic data enhancement method is that the data from four images can be computed at once when performing the normalization operation, thus reducing the memory requirement of the model [23]. This batch processing reduces memory requirements and allows the model to process large data more efficiently.

By randomly cropping, scaling, and arranging the four images, the mosaic data enhancement method can generate training samples with more diversity and complexity. In this way, the model can better learn how to handle different scenes, scales, and target combinations during training, improving the model's ability to generalize to different situations.

Inspired by the Mosaic data enhancement method, this paper proposes a new data enhancement method called Mosaic-6. In this method, six images are randomly selected for cropping, arranging, and scaling operations to combine them into a new image. The Mosaic-6 detail enhancement is shown in Fig. 2. This can effectively increase the amount of sample data and control the random noise within a reasonable range. In this way, the network model can be improved to distinguish small target samples in images. The final renderings are shown in Fig. 3.

![Mosaic-6 detail enhancement map](image1)

![Mosaic-6 effect](image2)

2) Optimization of attentional mechanisms: Given the nature of the helmet image, where there are numerous small targets that make up a small portion of the overall image, they can be easily influenced by the background and other factors. The original YOLOv5s network tends to lose crucial feature information of these small target helmets during deep convolution, which hinders their effective detection. To enhance the helmet features in the given image, this study introduces a work that assigns different weights to different positions of the image in the channel domain. This approach aims to extract more critical feature information [24]. The paper incorporates the Squeeze-and-Excitation (SE) attention mechanism, which is fused with the input feature map. This fusion produces a feature map with channel attention, which mitigates the loss of information related to small targets in the helmet image.

To ensure the effective functioning of the attention mechanism within the network, the paper integrates the SE attention mechanism into the focus module of the backbone network. This module is located at the front of the network and is placed before the first convolution operation. As a result, the network can prioritize the channel feature information of the target for detection at an early stage, thereby improving its representational capability.

As shown in Fig. 4, the structure of the SEFocus enhanced slicing module consists of two main parts. One part is responsible for slicing the original image to reduce the model computation, as shown in the figure for the slicing module; the other part embeds the channel feature information into the sliced and reconstructed image through the attention mechanism to prevent the channel feature information loss.
Transforming the sliced image into an attention image with channel feature information requires three steps. The first step first performs spatial feature compression on the feature map, the second step learns by FC fully connected layer to obtain the channel attention feature map, and the third step outputs the channel attention feature map [25].

In order to reduce the computation, the number of parameters, and the introduction of inter-channel relationships, the squeeze operation is first performed on the input feature map as shown in equation (1), assuming that the input feature map is X with dimensions \( C \times H \times W \), where \( C \) is the number of channels, and \( H \) and \( W \) are the height and width, respectively.

\[
z_c = F_{sq}(u_c) = \frac{1}{H \times W} \sum_{i=1}^{H} \sum_{j=1}^{W} u_c(i, j)
\]  

(1)

This step is completed by two fully connected layers to generate the weight information required in this paper by weight \( W \), which are obtained by learning. Then, the vector \( z \) obtained in the previous step is processed through two fully connected layers \( W_1 \) and \( W_2 \), and the channel weight \( s \) is obtained as shown in Equation (2), and after two fully connected layers, the different values in \( s \) represent the weight information of different channels, and the channels are given different weights.

\[
s = F_{weight}(z, W) = \sigma(g(z, W)) = \sigma(W_2 \delta(W_1 z))
\]  

(2)

Finally, the weight vector \( s \) generated in the second step is assigned to the feature map \( U \), and the feature map is obtained as shown in Equation (3), whose size is exactly the same as that of the feature map \( U \).

\[
\tilde{F}(u) = F_{scale}(u_c, s_c) = \tilde{s}_c u_c
\]  

(3)

3) Swin transformer: When YOLOv5s is used for feature extraction, the typical convolutional neural network CSPDarknet53 is selected to expand the perceptual field by continuously stacking convolutional layers to complete the capture of local information into global information. However, the increasingly deep convolutional layers lead to an increasingly complex model, and with the deepening of the network structure, the position information of the small-scale targets in the helmet dataset may be coarse, and the feature information may be easily lost after multiple convolutional operations [26].

This study presents an improved convolutional structure, as shown in Fig. 5, by incorporating the Swin Transformer into the C3 convolution module.

The aim is to enhance the semantic information and feature representation of small targets by utilizing the self-attentive window module in the feature fusion process.

In the Swin Transformer model, after the linear transformation of the input vectors, the resulting matrix is equally divided into three parts, which become the three features of query vector \( Q \), key vector \( K \) and position vector \( V \) in the Transformer, and the attention mechanism is calculated as shown in equation (4).

\[
\text{Attention}(Q, K, V) = \text{Softmax}\left(\frac{QK^T}{\sqrt{d_k}}\right)V
\]  

(4)

In the above equation, \( \text{Attention} \) denotes the attention weight matrix, \( \text{Softmax} \) denotes the normalized exponential function, \( QK^T \) is the matrix obtained by matrix multiplication, \( d \) is the feature dimension, and \( d_k \) is the length of the keyword vector.

The method not only takes into account the displacement invariance, dimensional invariance, perceptual field, and hierarchical relationships characteristic of convolutional neural networks, but also has the ability to extract global information and learn long dependencies. Swin Transformer improves efficiency by limiting self-attentive computation to non-overlapping local windows with moving windows, while allowing cross-window connections [27]. In addition, this hierarchical structure provides the flexibility to model at different scales and improves the ability of the model to capture multiscale information.

In summary, the above three improved methods are applied to the YOLOv5s algorithm, and the improved network structure is shown in Fig. 6.

Fig. 5. C3ST structure.

Fig. 6. Improved structure diagram of YOLOv5.
IV. INTRODUCTION TO THE DATA SET AND EXPERIMENTAL ENVIRONMENT

A. Introduction to the Data Set

1) Dataset sources: In the study of helmet and vest recognition, due to the lack of publicly available large-scale datasets, this paper collects relevant images through on-site photography and from the Internet. On-site photography is a common way to collect data, and this paper chooses to shoot at municipal construction sites and other scenes where helmets and vests are required. On-site photography captures image samples with real environmental backgrounds, different lighting conditions, and different poses, which is very beneficial for the robustness and generality of the algorithm.

In addition, a large number of relevant image resources are available on the Internet. In this paper, we obtain the necessary image information by searching and downloading user-uploaded construction site photos and construction site surveillance video screenshots through engineering websites and social media platforms.

2) Data pre-processing: In the data pre-processing stage, the images that meet the requirements are first converted to .jpg format for further processing and model training. To improve the robustness of the model, some of the positive sample data in this paper are inverted, which can increase the diversity of the data and make the model more generalizable when dealing with mirror image or symmetry problems. The saturation and exposure of the images are also adjusted to further increase the diversity of the data and the adaptability of the model.

Next, manual labeling is performed using the labelImg labeling tool. The construction workers in the image are labeled according to four categories: wearing a helmet (hat), wearing a safety vest (reflective clothes), not wearing a helmet (head), and not wearing a safety vest (other clothes). The labeling tool draws rectangular boxes in the image to frame the target location and assigns the appropriate category label to each box. This creates an XML tag file containing the four coordinates and category information of the target within the frame. The final PASCAL VOC format is obtained, which is a commonly used annotation format for target detection datasets and can be easily compatible and interactive with various deep learning frameworks.

Through the above processing steps, the pre-processed and annotated dataset with a total of 12,000 images is obtained in this paper. Each image has a corresponding image file in .jpg format as well as a tag file in XML format containing the location and class of the target.

B. Experimental Environment and Parameter Settings

In this study, a Linux system was chosen as the experimental development platform and the Ubuntu 18.04 operating system was used.

The experimental environment is equipped with a V100 GPU, and CUDA 10.1 is used as the GPU acceleration library. For the deep learning framework, PyTorch 1.8 was used as the main framework and Python 3.7 as the programming language. To meet the task requirements, the image resolution was set to 640x640, the training batch was set to 16, the training process was set to 100 rounds, and the initial learning rate was set to 0.001. The impulse was set to 0.937.

V. EXPERIMENTAL RESULTS AND ANALYSIS

A. Evaluation Metrics

The performance metrics commonly used in target detection algorithms include precision, recall, average precision (AP), mean average precision (mAP), and number of parameters and floating point operations (FLOPs).

Accuracy is the proportion of all samples with a positive prediction that are actually positive. It measures the degree of accuracy of the model in predicting positives. Recall rate, also known as sensitivity or true positive rate, is the ratio of correctly predicted positive samples to the total number of actual positive samples. The formula is shown in equation (5), which measures the ability of the model to detect positive specimens.

\[
P = \frac{TP}{TP + FP}
\]  

Recall is measured by calculating the ratio of correctly detected positive samples (True Positive, TP) to all true positive samples. TP refers to the positive targets that are successfully classified in the model, i.e., the true positive samples that are correctly detected. Equation (6) represents the formula for determining the recall rate.

\[
R = \frac{TP}{TP + FN}
\]  

Here, FN denotes positive samples that were incorrectly classified as negative during the classification process, i.e., true positive samples that were not correctly detected. The recall value, which ranges from 0 to 1, increases as the model becomes better at identifying true positive samples.

Average Precision (AP) is a comprehensive metric that calculates the accuracy of the model at different confidence levels and averages these accuracies. The formula is shown in equation (7) and is used to assess the balance between accuracy and recall of the model.

\[
AP = \int_{0}^{1} P(R) dR
\]  

The mAP is the average of the average accuracies of all categories. It measures the overall performance of the model over multiple categories and is calculated as shown in equation (8).

\[
mAP = \frac{\sum_{i=1}^{N} AP_i}{N}
\]  

The parameter count represents the total number of learnable parameters in the model. Reducing the number of parameters in a model can reduce memory requirements and computational complexity, making model deployment and inference easier. The number of FLOPs quantifies the number of floating point operations required by the model during inference. FLOPs serve as a measure of the computational complexity and inference speed of the model.
B. Presentation of Experimental Results

In this paper, a modified YOLOv5 model is used for security detection in the field of municipal engineering. The effectiveness of the improved model is tested using datasets consisting of homemade helmets and reflective vests to evaluate its detection capabilities. The accuracy, recall, and MAP graphs of this paper are shown in Fig. 7, 8, and 9, respectively.

According to Fig. 7, the accuracy of the proposed algorithm can reach 0.985. The x-axis represents the confidence threshold and the y-axis represents the precision. The overall shape and trend of the curve reflect the detection performance of the algorithm at different confidence thresholds. The improved algorithm in this paper can maintain high precision even at high confidence, indicating good accuracy.

![Accuracy graph]

Fig. 7. Accuracy graph.

From Fig. 8, the recall of the proposed algorithm reaches 0.97. The x-axis represents the confidence threshold, and the y-axis represents the recall. The curve describes the recall at different confidence thresholds. Higher recall indicates that the algorithm can detect more targets at a given confidence threshold, but it may be associated with some false positives. Lower recall may indicate that the algorithm misses some targets at a given confidence level, but with a lower false positive rate. The proposed algorithm maintains high recall even at high confidence.

![Recall rate graph]

Fig. 8. Recall rate graph.

According to Fig. 9, the Mean Average Precision (mAP) value in this paper reaches 0.927. The x-axis represents recall and the y-axis represents precision. The curve is plotted by connecting points representing precision and recall at different thresholds. The position of the curve closer to the upper right corner indicates higher precision and recall at different thresholds, implying better performance of the model. This indicates that the model proposed in this paper has good performance.

C. Analysis of Experimental Results

As shown in the table, in this paper, the operation of adding each of the three improvement modules one by one is performed and tested according to the three evaluation metrics of P, R and mAP. Among them, optimization model 1 modified the Mosaic-4 data enhancement method of the original model, optimization model 2 modified the focus structure of the original model, and optimization model 3 modified the C3 module of the feature fusion part of the original model. From the data in the table, the mAP value of Optimized Model 1 is improved by 1.3%, which indicates that the improved Mosaic method enriches the context in which the detection objects appear, which makes the model better learn the local features of the detection objects and optimizes the training of the model. The optimized model 2 mAP value is improved by 2%, indicating that the model can focus on the channel feature information of the target to be detected earlier by adding the SE attention mechanism in the focus part, which effectively reduces the problem of small target feature information loss. The mAP value of the optimized model 3 is improved by 1.8%, which proves that Swin Transformer can effectively improve the global perception of small features of the detected targets. In this paper, the combined use of three enhancement methods, whose P, R, and mAP are improved by 5.3%, 5.7%, and 3.4%, respectively, optimizes the detection effect of small and dense targets in complex scenes.

To better demonstrate the advantages of the algorithms in this paper, the improved algorithms in this paper are compared with several mainstream algorithms for experiments, including Faster R-CNN, SSD, YOLOv3, YOLOv4, YOLOv5s, and the improved algorithms in this paper. To ensure the fairness of the experimental results, the experimental environment, including experimental settings such as input image size, learning rate, and momentum, should be kept identical for the comparative experiments. As shown in Table I, with P, R, mAP, Parameters, and FLOPs as evaluation metrics.
As shown in Table II, the algorithm in this paper has a better detection performance with an average accuracy mAP of 9.3%, 13%, 11.5%, 9.7%, and 3.4% over the constructed helmet and reflective vest datasets than the Faster-RCNN, SSD, YOLOv3, YOLOv4, and YOLOv5s algorithms, respectively. In terms of model computation size, this paper is slightly larger than YOLOv5s, but much smaller than other mainstream algorithms, especially the two-stage detection algorithm. In addition, the algorithm in this paper has a great improvement in accuracy (P) and recall (R). Overall, compared with the YOLOv5s model, the algorithm in this paper is slightly larger in model size, but has a significant improvement in accuracy, and still meets the demand for real-time detection, and also significantly improves the detection performance, with significant overall performance advantages.

In order to visually compare the detection effect, this paper performs the detection of the same image with the YOLOv5 model and the improved algorithm, and shows the comparison results, as shown in Fig. 10. The left side of the figure shows the detection effect of the YOLOv5 model, and the right side shows the detection effect of the improved algorithm. By comparing the results of the two, the performance improvement of the improved algorithm on the target detection task can be clearly observed. Such a comparison helps to illustrate the effectiveness of the improved algorithm proposed in this paper in improving the detection results.
As shown above, in image (a), the original YOLOv5 model has failed to detect and identify whether reflective vests are worn or not, while the improved model can successfully detect them. The second set of images (b) is a dim scene, and the improved model can detect the protective gear worn by the construction workers even in very low light. The last set of images (c) is dense target detection, and the algorithm is able to accurately identify the presence of personnel occlusion. In short, the improved algorithm has been greatly improved in the case of missed detection, dim scenes and dense targets, which are not conducive to the correct identification of target detection.

VI. CONCLUSION

This paper proposes a YOLOv5s-based algorithm for the simultaneous detection of helmet and reflective vest wearing, which provides a new technological means for urban engineering supervision. First, the Mosaic-4 data enhancement method used in YOLOv5s is improved, and the Mosaic-6 data enhancement method is used to give synthesized images more complex backgrounds, which is used to enhance the learning ability of the model for local features, thus improving the generalization ability of the whole model. Second, the attention mechanism is introduced in the focus module to enhance the perceptual field of the model and improve the correlation between the channel information and the detection target features. Finally, the feature fusion part of the original model is improved by adopting a new hierarchical construction method inspired by the Swin Transformer. This enhancement aims to improve the semantic information and feature representation of the helmet mini-target by incorporating the window self-attention module, which effectively replaces the C3 module.

The experimental results show that the enhanced approach presented in this research paper is capable of accurately identifying helmets and reflective vests, and the average accuracy can reach 92.7%, which is 9.3, 13, 11.5, and 9.7 percentage points higher than the average of Faster-RCNN, SSD, YOLOv3, and YOLOv4 models, respectively. Li et al. improved the YOLOv5s network structure and loss function. The improved algorithm has a map value of 92.5%, a precision rate of 87.7%, and a recall rate of 86.8% [28]. Compared with this algorithm, the map value is only 0.2% higher. However, our algorithm shows a significant advantage in terms of accuracy, exceeding by 10.8%, as well as recall, exceeding by 10.2%. Although there is a slight increase in model size, the accuracy and recall rates are significantly improved and still meet the requirements of real-time detection with high application value.

VII. FUTURE OUTLOOK

While the improved algorithm in this study has achieved certain results in the detection of safety gear wearing in municipal construction projects, the authors acknowledge that there is still room for improvement. This study focused only on the method for detecting the wearing of safety protective equipment. In practical applications, it is necessary to design and develop a standardized system that includes the interaction interface between the backend algorithm and the frontend system. This will enable real-time detection of construction sites and ultimately establish a complete system. In future work, the authors will continue to strive for higher detection accuracy and explore more effective algorithms and techniques. Additionally, the authors will design and develop the system to ensure the algorithm's compatibility with real-world application environments. By considering both accuracy and real-time requirements, the goal is to provide a high-performance and reliable system that offers a feasible and effective solution for detecting the wearing of safety protective equipment in the field of municipal construction projects.

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A Review of Fake News Detection Models: Highlighting the Factors Affecting Model Performance and the Prominent Techniques Used

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Abstract—In recent times, social media has become the primary way people get news about what is happening in the world. Fake news on social media has harmed several domains, including politics, the economy, and health. Additionally, it has negatively affected society's stability. There are still certain limitations and challenges even though numerous studies have offered useful models for identifying fake news in social networks using many techniques. Moreover, the accuracy of detection models is still notably poor given we deal with a critical topic. Despite many review articles, most previously concentrated on certain and repeated sections of fake news detection models. For instance, the majority of reviews in this discipline only mentioned datasets or categorized them according to labels, content, and domain. Since the majority of detection models are built using a supervised learning method, it has not been investigated how the limitations of these datasets affect detection models. This review article highlights the most significant components of the fake news detection model and the main challenges it faces. Data augmentation, feature extraction, and data fusion are some of the approaches explored in this review to improve detection accuracy. Moreover, it discusses the most prominent techniques used in detection models and their main advantages and disadvantages. This review aims to help other researchers improve fake news detection models.

Keywords—Fake news detection; social media; data augmentation; feature extraction; multimodal fusion

I. INTRODUCTION

Social media platforms are now the main source of news consumption for many around the world. Unlike traditional media, social media networks support the rapid spread of posts to a wide audience in a short time and without validation restrictions and costs [1]. This contributes to fake news dissemination on social platforms [2]. Research has indicated that many people have trouble distinguishing between real and fake news. This issue is not related to a specific age or gender and does not depend on education [3]. Researchers observed that fake news spread 70% more than real news [4]. Recent studies have stated that fake news dissemination on social media has become a current-day issue that has attracted global attention that needs intervention and an immediate halt to its spread [5] because this issue is creating social panic and economic unrest [6]. Fake news detection is a difficult challenge. Therefore, the research community has paid much attention to this issue. It is considered one of the modern fields [7], and research in this field is still developing, but constantly increasing. This needs more improvement and exploration of upcoming directions in research to enhance fake news detection methods [1]. However, fake news detection overlaps with several domains [8]. Therefore, this matter has become of interest to many researchers from different disciplines and areas [9]. Based on previous studies, fake news models face many difficulties due to the distinct attributes of this issue. These challenges include the lack of standard datasets, their small size, or their imbalanced distribution, which affects detection models' performance. Another issue to be highlighted is how to deal with social media data, the features used, and the improvement of techniques for extracting these features. In addition, developing methods for the fusion of features and making decisions. Although there are many review articles discussing several aspects of fake news detection, most merely review the techniques used in detection models. In addition, they categorize features by type, or group datasets based on labels or domains. What distinguishes this review is that it discusses the challenging aspects that still affect fake news detection models. It also discusses the possibility of increasing detection accuracy by providing future suggestions for improving the techniques used. This review investigates three key and critical aspects of fake news detection studies, namely datasets, extracted features, and data fusion. These aspects affect the accuracy of fake news detection models. Studies published in the following well-known databases and digital libraries in the academic field (Web of Science, ACM Digital Library, Springer Link, IEEE Explore, Science Direct) in English for the period from 2017 to 2023, which dealt with the three aspects referred to above, were covered. The main contributions of this review are as follows:

- Provide an overview of fake news, its types, the impact of its spread, and the role of social networks in disseminating news.
- Highlight the critical parts of the fake news detection model and the most serious limitations related to them.
- Investigate the methods used in several fields that have provided promising results and future suggestions for improvement.
The rest of this review article is arranged as follows: Section II summarizes relevant studies that discuss significant aspects of fake news detection. Section III provides an overview of fake news. Section IV shows the role of social media in disseminating news. Section V reviews the main modules that make up fake news detection models. Section VI investigates the main detection model techniques and their challenges. Section VII discusses the prominent techniques used in Detection models. Section VIII suggests future directions to improve detection models. Section IX concludes this review.

II. RELATED WORKS

There are some reviews presented that deal with fake news detection studies. Some reviewed certain aspects, and others investigated others. Cardoso Durier da Silva, Vieira [10] presented a review of studies that dealt with the use of Machine Learning (ML) techniques or Natural Language Processing (NLP) methods in detecting fake news on social networks. While Sharma, and Qian [11] investigated the causes of fake news propagation and ways to reduce it, as well as analyzing the characteristics of standard datasets used in relevant studies. According to Pathak, Mahajan [12], previous studies employing Machine Learning (ML) and Deep Learning (DL)-based models using supervised, unsupervised, and hybrid approaches to rumor detection were reviewed. In addition, they presented in their review the standard datasets used in rumor detection studies and discussed the features used. Vishwakarma and Jain [13] presented a concise review of the methods and datasets used in research on fake news detection. In addition, they categorized fake news in terms of its types. They also investigated the features used in these detection models which are textual content features, and image-based features. In a different context, Zhou and Zafaran [14] published a review examining techniques for identifying fake news from four perspectives: the misinformation it contains, the writing style, the patterns of dissemination, and the reliability of the source. They also highlighted fundamental theories from different fields related to fake news dissemination. In another direction, De Beer and Matthee [15] presented a systematic review discussing the approaches used to detect fake news. These methods included the language, topic-agnostic, machine learning, hybrid, and knowledge-based approach. While Alam, Cresci [16] reviewed misinformation detection research papers divided down by content-based features, including image, speech, video, and network and temporal information. They also mentioned certain difficulties with multimodal detection models. Ansar and Goswami [17] published a comprehensive review of characterizing fake news identification from a data science point of view. They covered the different types of fake news, and the different features used in detection models. They also presented the most significant standard datasets currently available in this area. Studies of COVID-19 misinformation detection were also investigated as a case study. Considering that the issue of detecting fake news is a classification problem, Li and Lei [18] produced a review article that classified research related to fake news detection based on DL based on the data structures used in news classification which are text classification, graph classification, and hybrid classification. An overview of fake news detection studies was presented by Swapna and Sonya [19]. This overview was categorized according to features utilized in these models, such as linguistic and semantic, style, and visual features. Meanwhile, Hu, Wei [20] provided a thorough analysis of DL-based fake news detection techniques that consider many aspects like content, social context, and external knowledge. In their review, several widely used datasets and pertinent studies were presented, in addition to suggesting future work.

III. FAKE NEWS OVERVIEW

Fake news is perceived as among the most severe dangers to journalism, freedom of expression, and autonomy. It has been proven in studies that in comparison to authentic news, fake news on social media gets more retweets and shares, especially political news [21]. This issue has reduced public confidence in governments, including the controversial “Brexit” referendum, as well as the divisive 2016 U.S. presidential election [22]. The strongest emphasis was placed on the reach of fake news in the crucial months of the 2016 U.S. presidential election movement. Fake news as a term was chosen by the Oxford Dictionary in 2016 as the international word of the year [23]. This nation's economy is susceptible to fake news, which is linked to the fluctuating stock market and big deals. As an example, fake news claimed that US President Barack Obama was injured in an explosion, which led to the erasure of $130 billion worth of shares [24]. In the case of fake news's ability to gain public trust, psychological and social elements play a significant part. They reinforce fake news distribution. It has been proven that people become less reasonable and show vulnerability when authenticity and fabrication are differentiated. At the same time, they are burdened with fake news. Based on research conducted on 1,000 participants in more than 100 experiments in social psychology and communications, a slightly higher human capability of identifying falsehood was recorded compared to possibility with common precision degrees ranging from 55% to 58% and a mean precision of 54% [25]. In the case of news where truth and objectivity are expected, gaining public confidence is easier. People also believe fake news after being exposed to them repeatedly (validity satisfies [26] or if the news satisfies their desirability bias [27], is in line with preexisting principles, bias in confirmation [28] or viewpoints (selective exposure [29]). In some cases, peer pressure could “control” people’s perspectives and conduct (e.g., the bandwagon impact [30]).

A. Fake News Types

"Fake news” refers to news items that have been published but contain misleading information to deceive readers intentionally [7] for malicious purposes [31]. Literature shows that there are several types of fake news, as illustrated in Fig. 1. These are rumor, disinformation, misinformation, hoax, and clickbait [7]. A rumor is an unconfirmed or unsupported statement, and it spreads like wildfire [32]. Disinformation is misleading information deliberately posted to deceive people, while misinformation is inaccurate information that is unintentionally shared [7]. When a user publishes false information with malicious intent, it falls under the category of disinformation [33]. It is the users' lack of knowledge of a
particular topic or field that causes misinformation to be circulated [7]. One category of fake news is a hoax whose purpose is to intentionally mislead the reader with ill-intention. This includes defrauding users and losing money [34]. According to psychological studies, clickbait is one of the forms of fake news that draws in readers by evoking their interest in learning more about the catchy news headline. It also encourages them to click [35]. The purpose of clickbait is to redirect readers to fake websites to increase traffic to websites containing ads. It is a type of attention-grabbing title that might not reflect the content of the article [34].

![Fig. 1. Types of fake news.](image1)

**B. Fake News Consequences**

Since human society began, fake information has existed. But with the change and recent technological advancements in the global media environment, fake news circulation is growing. Fake news may cause severe consequences in political, social, economic, and health domains. Fake news comes in many forms. Fake news greatly affects the way information shapes our view. We make crucial decisions based on information. Based on the information we hear, we build an opinion about a situation or a collection of people. Contrived, misleading, twisted, or fraudulent information online prevents us from making the right decisions [36]. Fake news has the following main effects:

- **Impact on citizens:** Rumors about particular people can have a major effect. These individuals can face online abuse. In addition, they can be subjected to insults and threats that may have far-reaching negative effects. Individuals should not directly believe disinformation posted on social media and not make premature judgments about others based on this misleading information.

- **Impact on health:** People turn to the Internet for health news. Health-related fake news can affect people’s lives. This problem has become the issue of the times. In the past few years, misinformation targeting health has had a serious negative impact. As a result, and based on effective lobbying by health organizations, doctors, and health advocates, many social media companies have had to change their policies to prevent and restrict misinformation spread.

- **Impact on finances:** Currently, fake news is a serious problem in industry and commerce. Fraudulent businessmen publish deceptive information or reviews to increase revenues. Stock prices may drop due to false information. It can destroy a company’s reputation. Fake news also impacts customers’ expectations. False news can breed unscrupulous commercial practices.

- **Impact on democracy:** Because fake news was so influential in the US presidential election, the fake news problem has received considerable media attention. Fake news has become a major issue threatening democracy, therefore, its spread must be stopped [37].

**IV. SOCIAL MEDIA PLATFORMS**

In recent years, rapid technological development, particularly in the mobile phone sector has made social media networks like Facebook, Twitter, and Sina Weibo accessible. These platforms have become an integral part of people’s daily lives [7]. Social media is now a potent tool for all types of journalism, including sports, medical, and political reporting [15]. Instead of watching traditional media, most people now spend their time on social media to connect, gather knowledge, and share it [7]. Social media is used by many people to post or share news or information. This is because, unlike traditional media, news distribution through networks is real-time and quick, there are no costs involved, and there are no restrictions imposed on validation [4]. For example, in 2012 in the U.S., about 49% of users shared news on social media platforms. The Pew Research Center issued a report in 2016 stating that more than 62% of users daily receive their news from social networking pages [4], while in 2018, a report indicated that two-thirds of adults in the U.S. received their news from these pages [38]. The fact that social media is used on a variety of devices has significantly expanded the amount of data available [15]. Furthermore, it is worthwhile to refer to the issue of the language used in social media. This is because social media users come from all cultural, academic, and age backgrounds. Therefore, many posts on social media contain linguistic mistakes, use acronyms, are written in slang, or include obscene words. Fig. 2 shows a post containing slang and acronyms.

![Fig. 2. A colloquial post on a facebook page.](image2)

There are drawbacks to these platforms despite their advantages. These platforms are misused by people or organizations to propagate false information for malicious purposes. This may be for financial gain, extremist hatred, or manipulation of people’s minds for political reasons. It may also be intended to form biased opinions for electoral purposes [39]. The negative effects of social media as a result of fake news foreshadow a real danger that negatively affects individuals and society. This requires providing models for detecting fake news and limiting its spread [40].

**V. COMPONENTS OF THE FAKE NEWS DETECTION MODEL**

In this section, the critical parts that make up the fake news detection model and affect its performance and how these elements relate to each other to carry out this task are reviewed. The detection model has three main components: a dataset, features, and a model based on a supervised classifier.
A. Dataset Used in Fake News Detection Models

Fake news detection studies demonstrated that no benchmark dataset is currently available that offers resources for extracting all crucial features. Fake news circulates on social media in various temporal patterns than real news. The dataset is the most significant component of fake news detection, and any model's effectiveness depends on it [34]. The larger size of the dataset [41], more diverse [42], more feature-rich [43], and low-noise [44], leads to improving the model's performance and increases its accuracy in identifying fake news [40]. Many researchers utilize fact-checking websites for data collection [45], because gathering data on fake news is time-consuming [46]. There are several significant challenges facing data gathering. These challenges include the ability to create a large volume of data in high quality, as well as ease of access without privacy restrictions. In addition to the data annotation process [7]. Data collection resources used by the researchers included reliable resources [44] such as government websites and websites of reputable media organizations, which contain information based on the facts [47]. According to researchers Subramani, Michalska [48], collecting, annotating, and labeling data is a laborious process that takes time, cost, and effort. The results are a medium-sized dataset that includes domains not previously investigated and considered acceptable by researchers. Some researchers are compelled to manually gather, annotate, and label data to produce a standard dataset that is checked for quality and reliability by specialists [48]. It should be underlined that facts, not opinions or feelings, must serve as the foundation for gathering ground truth data [44]. Additionally, it is essential to pre-process the dataset to eliminate extraneous data [42]. Future research would take less time, effort, and cost if standard datasets were created for previously unstudied topics, or expanded by adding more information [49]. This would benefit the research community. [50].

The researchers note that the model's accuracy is affected by the balance or imbalance in the dataset structure [51]. When compared to models that use a balanced dataset, models that use an imbalanced dataset produce higher results during training (a bias for one category over another) [52]. This means that these models are biased. Therefore, models that use a balanced dataset perform better [53]. Fig. 3 shows the most significant factors affecting the detection accuracy of fake news detection models.

B. Features Used in Fake News Detection Models

AI-based models for fake news detection rely on some key features, such as content-based, network-based, or user-based. In any case, using all of these features may not increase detection model accuracy. Depending on the nature of the issue, one or more features may be used. The results of the study by Kim, Kim [54] indicated that rumor detection accuracy using user features was the most poor of all. In contrast, rumor detection accuracy using content-only features was significantly higher than utilizing all features at once. According to these researchers, the experiments demonstrated that propagation features and user features were insufficient to identify rumors. Generally, news content (linguistics and visual information) is used as features in news identification [36]. Psychologists say people prefer articles with engaging visuals paired with text and believe them. When an article contains multimedia elements such as images instead of just text, it reaches more users [19]. Therefore, while textual content is essential for news verification, visual content also has a vital function in detecting fake news [17].

C. Fake News Classifiers Based on Supervised Learning

The performance of any model normally relies on the classifiers employed [43], the hyperparameter tuning [48], and the dataset used [34]. The diversity of AI techniques used in fake news detection models in previous research encouraged us to break up related studies based on supervised learning methods applied to detect fake news. Using labeled examples, supervised models learn from a variety of features [20]. Therefore, we list below the previous studies according to their use of ML and DL methods.

1) Machine learning-based models: Many previous research articles have employed Machine-Learning (ML) techniques to identify fake news. Aldwairi and Alwahedi [9] presented a study to detect fake news using a logistic classifier. They scraped their data from web pages. They also applied information gain and correlation attribute methods to rank the attributes according to how they relate. In all their experiments, the accuracy of the results exceeded 99%, and it was unclear whether they were training or test results. While Bhutani, Rastogi [55] proposed a model using Random Forest (RF) to identify fake news by employing sentiment analysis-based features. They utilized TF-IDF with cosine similarity for feature representation. The model was trained and tested on three datasets, each separately. However, there was a disparity in the results between training and testing, which means the model was overfitted. Therefore, the three datasets were combined, and based on them; the proposed model was trained and tested on this merged dataset. The AUC measure was 84.30%. In the same direction, Varshney and Vishwakarma [45] presented a two-stage model, data collection and classification. The data collection stage retrieves claims about statements from rumor-debunking web pages. Based on content similarity to claims, the data classification stage extracts features from web statements that have been retrieved. Claims are classified as real news or fake news by the RF classifier based on content- and sentiment-
based features. Faustini and Covões [56] used KNN, RF, NB, and SVM machine learning-based classifiers to identify fake news. SVM and RF outperformed other classifiers, according to experiments. They used five small datasets, and the number of examples in each dataset ranged from 137 instances, as in the brvlifestyle dataset, to 8,981 instances, as in the TwitterBR dataset. Regarding the feature set, the researchers applied the bag-of-words method in their experiments, as well as the DCDistance algorithm to reduce the dimensionality of the features. Although some of the results from some of the datasets examined may be encouraging, the majority of the results are poor. Kim, Kim [54] proposed an Ensemble Solution (ES) based on Soft Voting which consists of RF, XGBoost, and Multilayer perception. In order to determine which ML model was most suitable for creating an ES model, they examined several. To detect rumors, the latter model used the features of content, network, and users, and their model provided the highest F1 score of 79%.

2) Deep learning-based models: For identifying rumors, researchers Rath, and Gao [32] have developed a believability concept-based model that uses the LSTM classifier. Believability is determined based on the level of trust between Twitter users. They formed their dataset by merging two datasets, Twitter15 and Twitter16. LINE embedded the user in the network according to its reply and retweet. Their model provided an accuracy of 73.80%. Li, Hu [57] presented a detection model to identify fake news based on CNN at multiple levels. This model extracts semantic information from articles and represents it based on word and sentence levels. A pre-trained Word2Vec model was utilized for feature vectorization. In addition, they used the TFW method to calculate the weights of sensitive words to improve classification accuracy. They used five small datasets and combined them into two datasets. Their proposed model provided an accuracy of 88.80% and 90.10% on the Weibo and NewsFN datasets, respectively. Furthermore, Alkhodair, Ding [31] created a model that uses the LSTM classifier by plugging it in parallel with the Word2Vec embedding model. The model is constantly updated with new tweets so that the classifier could detect rumors about upcoming topics on Twitter based only on the text of the tweet. The highest score recorded for the suggested model was 79.50% based on the F1 score on the PHEME dataset. In their study, Brașoveanu and Andonie [8] examined several DL-based models to identify fake news using a variety of features from the datasets, such as textual, relational, and meta-features. Textual features were represented by a pre-trained Glove model. The two datasets used are Politifact and Liar, which are small and imbalanced. The results were unsatisfactory for most models with less than 50% accuracy. The highest result was an accuracy of 52.40% on the Politifact dataset and an accuracy of 64.90% on the Liar dataset. This was using the CapsNet model with the attention mechanism using all features. The researchers Guo, Xu [58] developed a CNN-based rumor detection model using Transfer Learning (TL). This proposed model was trained on the small YELP-2 dataset. By copying the basic model's parameter values and then re-adjusting them after the proposed model had been trained on a small Five Breaking News (FBN) dataset to prevent the negative transfer, the TL method addressed the issue of a limited training dataset. Their fine-tuned model produced an 82.50% result based on the F1 score. Regarding Gadek and Guélorget [59], they presented an interpretable text classifier built on CNN architecture with the Class Activation Maps (CAM) method. The classifier identifies fake news in two distinct stages by applying two different datasets. Text analysis is used in the first classification and emotion in the second classification. For their experiments, the Kaggle fake news and Signal-Media datasets were utilized. For the second dataset, they applied an under-sampling method to balance it. To extract the features, a pre-trained FastText model was applied to consider punctuation. The result of detecting fake news relying on textual features was 91.8% of the F1-score, while the result of identifying fake news using sentiment analysis features was 68.3% of the F1-score. While Kaliyar, Goswami [60] proposed a fake news detection model built on Multi-layer DNN. Features based on news textual content and social context were used to identify fake news and were represented by the tensor factorization technique. They conducted their experiments using the BuzzFeed and PolitiFact datasets, and the results on both datasets were 88.37 % based on F1-score. By capturing the connections between rumors and comments on a particular topic, the researchers Lin and Chen [40] built a Feed Neural Networks (FNN) based model for rumor detection with multi-layer transformer encoding blocks and one fully connected layer. An attention mechanism has been utilized in transformer encoding blocks to improve model performance. They used two datasets Weibo and PHEME for training and testing, and their model provided an accuracy of 84.1% on the PHEME dataset. Consequently, DL is superior to Machine Learning, because of its capacity to extract high-dimensional features [61], automated feature extraction, little reliance on data pre-processing, and improved accuracy [36].

VI. LIMITATIONS OF FAKE NEWS DETECTION MODELS

Based on the weak results of Wang's [62] research and the use of high dropout values in the proposed model trained on a fine-grained dataset of 12,800 examples, it can be concluded that the detection model was overfitted and high dropout values were used to eliminate overfitting. Ghanem, and Rosso [63] used an imbalanced fine-grained FNC-1 dataset in their research. They did not address the imbalanced dataset, as their model produced poor results. There is a variation in the results of the Kumar, Asthana [64] study, as the test results were very high for most models on a small test set. However, after testing these trained models on a larger volume of the same dataset from PolitiFact the results were poor. Note that the PolitiFact dataset has the same features as the one used to train the models. This means that their proposed models fell into an overfitting problem, which prevents them from generalizing to the new data. In addition, the FakeNewsNet dataset is multimodal and visual features were excluded from their
research. Based on the weak results of Shu, and Mahudeswaran’s [65] experiments, it is clear to us that the lack of effective representation of features in detecting fake news, which was represented as a one-hot encoded vector, was one of the reasons for these results. In addition, one of the datasets used, although feature-rich, is small in size, which can lead to overfitting that reduces the generalizability of the model when tested on test data. Raza and Ding [66] used two multimodal datasets in their research, but visual features were not utilized in detecting fake news. In addition, the imbalanced dataset was addressed by the under-sampling method, as the omitted examples may include relevant attributes. In addition, no pre-trained word embedding model was used to enrich the model with features. All of these issues contributed to decreasing the detection model's performance. Elhadad, Li [67] used TF-IDF and N-Gram techniques to extract features and the use of such techniques may lead to the loss of many attributes, including neglecting to capture semantic relationships between words, and this is evident from the research results. Segura-Bedmar and Alonso-Bartolome [68] used the CNN model for feature extraction and did not employ a pre-trained model for visual feature extraction. Also, extracted textual and visual features were directly fused. In addition, they indicated that their model suffered from misclassification due to an imbalanced dataset. If advanced techniques were used, the results would be better. Also, the model presented by Singhal, Shah [69] fused extracted features based on simple concatenation. It was possible to increase accuracy if more focus was placed on data fusion. Kalra, Kumar [70] indicated that their model performed poorly because of the imbalanced fine-grained dataset used. Their model was overfitted and a dropout layer was added after each layer. Moreover, fusing multimodal features from different models directly loses many attributes.

VII. THE PROMINENT TECHNIQUES USED IN DETECTION MODELS

This section highlights some methods and techniques used in other fields. These techniques provided outstanding results, which can be used in fake news detection models to improve these models and increase their accuracy.

A. Dataset Augmentation Techniques

Deep Learning-based models are computationally costly and need properly labeled data for high performance. Enhancing the model's performance requires big data to identify the most features. Many researchers utilize fact-checking web pages for data collection, [44], because gathering a dataset related to fake and real news is time-consuming [46]. The ability to provide a big volume of data, its relevance to the research topic, its high quality, rich in features, and its ease of access without privacy limitations, particularly with social media data, are some of the most significant issues facing the data collection process [6]. Moreover, manually labeling this data is labor intensive [71]. The small size of standard datasets or they are available in large sizes but of poor quality, poses the biggest obstacle to developing and evaluating any model's efficacy in identifying fake news [72]. Deep learning-based approaches used for fake news detection require a lot of training data. The size of the data has an impact on the model's accuracy. This is why, the larger the dataset, the better the accuracy of the model [73]. The data augmentation method can create more data from the original existing data. This process leads to increases in the model's accuracy, without human effort to collect data and save time where it is difficult to collect more real data. The data augmentation process is one solution to reduce the occurrence of overfitting and underfitting during the training phase. This is done by increasing the dataset size with synthetically labeled data [74]. The method of data augmentation plays a vital role in the success of DL models, as this augmentation can lead to better detection accuracy for models when using these large datasets [75]. Data augmentation techniques have been widely used in computer vision, and have provided impressive results in image classification [76], in particular using the Generative Adversarial Network (GAN) method. GAN is an effective data augmentation method which is a type of deep network used to generate new examples [77]. In recent years, a trend has emerged for tackling NLP problems via natural language generation models such as LeakGAN. This is a modified GAN to deal with text. This is completely unsupervised or semi-supervised learning for data generation [78].

1) Generative Adversarial Network (GAN): Deep learning-based generative models are called GANs. Their architecture is made up of a generator model for producing new instances and a discriminator model for detecting whether the instances created by the generator model are real or fake. Adversarial networks are frequently utilized to produce images that match observed samples. The generator model creates new images that mimic the original image using features derived from training data. Whether the created image is fake or real is predicted by the discriminator model. In detail, a vanilla GAN is made up of two networks that cooperate during training: Generator and Discriminator as illustrated in Fig. 4. Generator: This network produces images with the same structure as the training set of images when a vector of random values is presented as input. Discriminator: This network attempts to classify observations as "real" or "fake" based on batches of images that include observations from the training set and images created by the generator. The generator output is directly connected to the discriminator input. The generator utilizes the discriminator classification as a signal by using a backpropagation process to update its weights. [79].

![Fig. 4. The basic architecture of GAN.](image-url)
In a semi-supervised environment, GANs are used to train discriminators and are very effective in generative modeling, which reduces the need for human intervention in data labeling. GANs are also helpful when data contain underrepresented samples or classes. GANs can only generate synthetic data if their foundation is a set of continuous numbers. The GAN technique has been successfully used mostly in image processing to produce real image samples. However, despite the fact that several prominent GANs models have also been suggested for image inpainting, synthesized images still have pixel errors or color inconsistencies throughout the image generation process. These errors are typically called fake textures [80].

2) LeakGAN: Unfortunately, there are two issues with using GAN in NLP to produce sequences. First off, GAN struggles to directly generate sequences of discrete tokens, like sentences, as it is built for producing real-valued, continuous data. For this reason, GAN begins with random sampling before moving on to a deterministic transform controlled by model parameters. The score/loss for a complete sequence can only be provided by GAN after it has been formed; for a partially generated sequence, it is difficult to reconcile the present performance with the expected score for the entire sequence in the future [81]. However, because all NLP models are based on discrete variables like words, letters, or bytes, GANs cannot be used with NLP data. Novel strategies for training GANs on textual data are needed [36]. Some promising models that address this problem have been presented, such as LeakGAN which handles the problem of generating long text [82]. LeakGAN is a novel algorithmic framework proposed by Guo, Lu [82] that addresses both sparsity and non-informative problems related to previous GAN versions. LeakGAN is an innovative approach that builds on recent developments in hierarchical reinforcement learning. It delivers more information from the discriminator to the generator. A hierarchical generator G has been introduced as shown in Fig. 5, and it comprises a high-level MANAGER module and a low-level WORKER module. Mediation is performed by an LSTM called MANAGER. It gets generator D’s high-level feature representation for each step, such as the CNN feature map, and utilizes it to create the WORKER module’s guiding objective for that timestamp. Since D maintains its information and plays an adversarial game, it is not supposed to give G access to that information. As a result, it is called a leak of information from D. The WORKER then takes final action in the current state by combining the LSTM output and the goal embedding. This is given the goal embedding created by the MANAGER. This is done by encoding the currently generated words with another LSTM first. Therefore, the guiding signals from D are available to G both at the end in the form of scalar reward signals and during the generation process in the form of a goal embedding vector to help G improve. The discriminator evaluates the created sentence in an adversarial manner once the generator generates the following word. The main innovation is that, in contrast to traditional adversarial training, the discriminator communicates its internal state (feature) during the process to direct the generator more frequently and informatively. Consequently, LeakGAN achieved significant performance gains when generating longer sentences.

![Fig. 5. The LeakGAN architecture.](Image)

B. Features Extraction Models

Feature extraction extracts a collection of features, known as a feature vector. This maximizes the prediction rate with the fewest number of elements and produces a similar feature set for several instances of the same symbol [83]. Thus, there is a need for some efficient techniques, such as vectorization, also referred to as word embeddings in the NLP field, and pretrained image models in the image processing field. In NLP, feature extraction converts each text into a numerical representation in a vector [84], which is the initial stage in training a deep learning model. The word is considered the fundamental building block of documents in NLP [85]. When working with natural languages, the words in the document must be represented grammatically and semantically to obtain the intended results [86]. One of the common and effective techniques used in deep learning models for feature extraction is word embedding. Each word in the text string is transformed into a vector with n dimensions using embedding models, where n is the dimension of the embedding [64]. Word embedding models are distributed feature representations that are dense, low-dimensional, and well-suited to natural language tasks [87], based on deep learning [58]. Based on grammatical and semantic similarity, these models represent words and distribute them in vectors. They also take the word’s relations to other terms in the document into account. Words with comparable meanings are represented by low-dimensional vectors [88]. The value of the distance between the two embedding vectors means how close the words are to each other according to the relationship between them [73]. For example, the terms “anxiety” and “depression” are semantically related because they fall under the same class relating to mental health [89] and also the terms “bad” and “good” are closely embedded for the same reason [90]. Word embedding models have proven to be remarkably effective in a variety of NLP tasks [73], including sentiment analysis, text classification, machine translation, and question-answering. This is according to earlier studies [90]. One such effective embedding model is...
the BERT which was produced by Google. It performed well in classifying, composing, and summarizing texts. The BERT model was used to address the OOV word issue and the problem of polysemy in conventional embedding models and the incorporation of contextual information [91]. However, embedding methods have received increasing interest in extracting textual features and have the potential to develop better representations. As opposed to relationships between words, relationships between visual concepts in images are essential for computer vision (CV) tasks, but challenging to capture. Image-based features are a key component in detecting fake news [92]. There are several neural models based on transfer learning for image-based feature extraction such as AlexNet, VGG16, and VGG19.

1) Bidirectional Encoder Representations from Transformers (BERT): BERT was created to determine the relations among words within a sentence. BERT uses a language representation approach that only utilizes the encoder section of the transformer together with semi-supervised learning. In particular, BERT is built on a multi-layer bidirectional transformer encoder, which efficiently captures information from the left and right contexts of a token at each layer simultaneously [93]. In order to perform the pre-training, an unsupervised prediction operation will be executed using a masked language model (MLM) and a sentence-next predictor by BERT. In MLM, context knowledge comes before word prediction [94]. The BERT model often uses sentences broken up into individual tokens as inputs to produce a sequence of them. The BERT model takes context into account from both sides. Instead of processing each word separately, the transformer analyzes each word in connection with every other word in the sentence. Additionally, BERT’s self-attention mechanism supports determining sentence keywords. The pre-trained BERT model could be effectively fine-tuned for advanced performance in various Natural Language Processing (NLP) tasks, proving that BERT models are incredibly adaptable. BERT’s tokenizer is built on words and sub-words. Therefore, if a word is absent from the original vocabulary, it will be broken down into a series of sub-tokens that when combined will make up the original word. To ensure OOV tokens do not appear and that all vocabulary units are regularly updated and sufficiently trained during training, the remaining new tokens, those associated with more uncommon words, are simply divided into smaller units [95]. The BERT framework consists of two stages: pre-training and fine-tuning. BERT was trained using unlabeled data from English Wikipedia (2,500M words) and the Books Corpus (800M words). There are two types of BERT models, the BERT large model which consists of 24 layers of encoders, and the BERT base model which consists of 12 layers. Also, there are two versions, cased and uncased [96]. With just one additional output layer, the BERT pre-trained model may be adjusted to handle a variety of NLP-based tasks, including text summarization, sentiment analysis, chatbots, and machine translation. Fig. 6 shows the fine-tuning process for the pre-trained BERT.

2) VGG-19 Model: An enhanced variant of the AlexNet architecture is the VGG16-Visual Geometry Group CNN architecture. An improved convolution neural network is implemented by expanding the network depth to 16 or 19 trainable layers. VGG networks in computer vision continue to be favored for many difficult problems [97]. The 143 million parameters of the deep architecture are learned from the ImageNet dataset. VGG receives RGB images in 224 × 224 pixels. The VGG-19 [98] is made up of 19 trainable weight layers, beginning with five stacks of convolutional layers and ending with three fully connected layers (FC) as illustrated in Fig. 7. These convolutional stacking layers carry out the process at each mark, extract image features, and then pass the result to the following layer. The number of filters increases by a factor of two, and all convolution layers employ a 3 by 3 filter size. A max-pooling layer and a Rectified Linear Unit (ReLU) activation function-based layer are followed as a non-linear activation function. Max-pooling layers are applied between each stack of convolution layers after ReLU, using a 2 by 2 kernel filter with 2 strides (pixels).

Some models, such as AlexNet and VGG-16, have encountered some issues that have been addressed in the improved VGG-19 version, including:

- Model Training: The first completely linked layer will yield a very high number of parameters, according to experiments on the original model VGG-16. This greatly increases the number of calculations and uses up more computational resources which leads to more training time [99].
• Vanishing Gradient (VG): Although network depth is significant, it can be challenging to train a deep network. This is due to the issue of vanishing or exploding gradients that arise when adding more network layers. In addition to gradient problems, if network depths continue to rise, model performance may quickly reach a limit before rapidly declining. Prior to AlexNet, the sigmoid, and tanh activation functions were most frequently utilized. These functions exhibit the VG problem due to their saturation, which makes it challenging for the network to train. AlexNet uses the ReLU activation function, which is immune to VG issues. ReLU aids with vanishing gradient problems, however, because it is unbounded, learned variables may rise excessively [100].

• Model Overfitting: Due to a large number of VGG-16 network parameters, overfitting can easily occur [99].

With the VGG-19 model, some of the previously mentioned problems have been addressed, and it offers higher accuracy than the VGG-16. The VGG-19 network is able to converge after a few iterations because of the implied regularization function of the network depth and the small convolution kernel size [101]. Having a large number of weight layers is the result of the small-size convolution filters in VGG-19, and surely, having more layers results in better performance. Despite this, less trainable variables lead to quicker learning and more resistance to overfitting.

C. Multimodal Fusion Method

Fusion is a crucial area of research in multimodal studies because it combines data from various unimodal data sources into a single, condensed multimodal representation. Fusional representations and multimedia are interrelated [102]. Multimodal fusion has generated a great deal of attention among scholars and widespread concern since it is an efficient method of processing multimodal data acquired [103]. Multimodal fusion is used to gain rich features by integrating various modalities [104]. The current challenge is merging and refining information coming from various modalities. Each modality contributes to varying functions. During the analysis of fusion features, the noise must be removed and relevant information extracted [105]. Multimodal fusion combines features from text and image modalities that must be merged before classification can be performed. Multimodal data fusion can produce extra information that improves the outcome precision. For example, compared to a single-modal CNN-based detection model, a multimodal fusion model for autonomous vehicle detection that fuses features of images from cameras with information from Light Detection and Ranging (LiDAR) sensors can attain noticeably improved accuracy by 3.7% over the previous one [106]. Multimodal fusion methods come in a variety of ways as shown in Fig. 8, including:

• Early or Feature-Level Fusion: often referred to as feature-level fusion, it is the task of combining the input of different modalities into a single feature vector before it is presented into a single learning model. There are several techniques to combine input modalities, such as concatenation, pooling, or using a gated unit. Early fusion type I involves combining the original features, whereas type II involves merging extracted features or learned representations from another neural network. The anticipated probabilities are considered as extracted features, making the fusion of features and projected probabilities from several modalities another early type II fusion [107]. Feature-level fusion produced the most effective results for unimodal fusion and reduced processing time [108].

• Joint Fusion or Intermediate Fusion: The technique of combining learned representations of features from the in-between layers of NN with features from other modalities as input to an eventual model is known as joint fusion. The crucial distinction from early fusion is that during training, the loss is returned to the NN for feature extraction. This improves the representation of features for each training iteration. Neural networks are used in joint fusion because they pass on the loss from the prediction model to the feature extraction model. This joint fusion is type I, when feature representations from all modalities are extracted. The feature extraction stage does not always need to be classified as joint fusion for all input features.
• Late or Decision-Level Fusion: Late fusion, often referred to as decision-level fusion, is the task of using predictions from various models to arrive at a final decision. The final decision is typically achieved by using an aggregation function to combine the predictions of various models. Normally, diverse modalities are employed to train individual models. Averaging, weighted voting, majority voting, or a meta-classifier based on each model's predictions are several examples of aggregation functions. Based on the application and input modalities, the aggregation function is typically chosen empirically [107]. This method has the benefit of allowing each modality to learn its features using the best classifier for that modality [108].

VIII. FUTURE DIRECTION

Based on this review, we summarize some of the significant issues in this field that need to be addressed. In addition, we believe there is significant room for further improvement in fake news detection techniques. These issues are as follows:

1) Due to the use of either small or imbalanced datasets, detection models still suffer from significant challenges including underfitting, overfitting, and poor classification that degrade their performance.

2) Image-based features have not been widely used by previous studies in detecting fake news despite their highly critical effect.

3) Despite the vast use of vanilla GANs to generate new samples, they still suffer from some crucial issues, including vanishing gradients, mode collapse, and failure to converge.

4) Although pre-trained word embedding models are efficient at extracting features, they are not able to fully exploit the text's semantic and structural features.

5) Several machine-learning methods have been used to detect fake news. However, lower detection accuracy was provided.

6) The real significance of many modalities cannot be determined by simply concatenating the features. The unique features of each method (text and image) must be preserved while integrating relevant information between the different methods.

IX. CONCLUSION

This review article presents an overview of fake news, its types, and its consequences. The role of social media platforms in spreading fake news was also discussed. In addition, the most significant factors affecting fake news detection models were highlighted. Among these factors are the dataset, features, and supervised learning classifiers. The critical limitations that still need to be addressed were revealed by reviewing the most promising methods and techniques. These methods provided encouraging results in several areas that can be employed in fake news detection models. Moreover, these techniques were investigated and some of the challenges faced by them were described. This would allow future researchers to improve them and raise fake news detection accuracy.

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REFERENCES

A Dynamic Intrusion Detection System Capable of Detecting Unknown Attacks

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Abstract—In recent years, deep learning-based network intrusion detection systems (IDS) have shown impressive results in detecting attacks. However, most existing IDS can only recognize known attacks that were included in their training data. When faced with unknown attacks, these systems are often unable to take appropriate actions and incorrectly classify them into known categories, leading to reduced detection performance. Furthermore, as the number and types of network attacks continue to increase, it becomes challenging for these IDS to update their model parameters promptly and adapt to new attack scenarios. To address these issues, this paper introduces a dynamic intrusion detection system, Dynamic Unknown Attack Intrusion Detection System (DUA-IDS). This system aims to learn and detect unknown attacks effectively. DUA-IDS comprises three components: Feature Extractor: This component employs CNN and Transformer models to extract data features from various perspectives. Threshold-Based Classifier: The second part utilizes the nearest mean rule of samples to classify known and unknown attacks, enabling the distinction between them. Dynamic Learning Module: The third part incorporates data playback and knowledge distillation techniques to retain existing category knowledge while continuously learning new attack categories. To assess the effectiveness of DUA-IDS, this paper conducted experiments using the UNSW-NB15 public dataset. The experimental results show that DUA-IDS improves the classification accuracy of flow network data with unknown traffic attacks. Can accurately distinguish unknown traffic and correctly classify known traffic. When dynamically learning unknown traffic, the classification accuracy of previously learned known traffic is less affected. This indicates the advantages of DUA-IDS in detecting unknown attacks and learning new attack categories.

Keywords—Intrusion detection systems; transformer; DUA-IDS; data playback; variable perspectives features; Knowledge distillation

I. INTRODUCTION

With the increasing prevalence of interconnected devices, the issue of network security has gained significant prominence. Malicious network intrusions or attacks pose a threat to the fundamental elements of computer security policies, namely confidentiality, integrity, and availability. Hackers and cybercriminals continuously innovate their attack methods to steal data, gain control over host computers, and extort money. In order to combat these threats, Intrusion Detection Systems (IDS) have emerged as a major research focus in the field of network security. An IDS serves as a vital component of computer network security. Its primary function is to monitor network traffic for malicious activities, such as distributed denial of service attacks and injection attacks, and respond appropriately to intrusions. When an IDS detects a network attack on a computer, it promptly alerts the security administrator, enabling them to take necessary measures to mitigate the threat.

One of the main challenges in network intrusion detection systems is the ability to detect and recognize unknown events, including risks like zero-day attacks and low-frequency attacks such as worm attacks. Additionally, relying on administrators to manually detect, identify, and address network security issues is both inefficient and time-consuming. The limitations of the signature-based detection method further exacerbate the problem, as it cannot handle unknown threats and requires frequent updates to the signature database. Consequently, the optimal solution lies in equipping machines with the capability to analyze network data and identify suspicious or abnormal behavior.

This paper aims at the two problems mentioned above: detecting unknown events and frequently updating feature database manually. A dynamic intrusion detection system which can detect unknown attacks is proposed.

The rest of this paper is organized as follows: In the Section II, the related work is briefly reviewed. Section III introduces the network model and related algorithms. Section IV introduces the data set used in the laboratory. In the Section V, the experimental setup and results are clarified, and the simulation results are discussed. Finally, conclusions are drawn in Section VI.

II. RELATED WORK

In the early days, IDS algorithms primarily relied on traditional machine learning methods like decision trees, support vector machines, genetic algorithms, and logistic regression. These approaches classified or clustered based on known attack characteristics. For instance, Senthilnayaki et al. [1] introduced an IDS method using the CART decision tree, which enhanced detection accuracy. Wang et al. [2] proposed a feature selection model based on sparse logistic regression, effectively identifying attack data. In recent years, researchers have combined deep learning algorithms with IDS due to their advancements. For example, Xiao et al. [3] developed a CNN-IDS using a convolutional neural network (CNN) that reduced input data dimensions through principal component analysis. The data was then transformed into a grayscale image and fed into the CNN for classification learning. Yan et al. [4] constructed a CNN-based IDS and employed it to generate synthetic attack trajectories against the network. Wang Y et al. [5] proposed a CNN-based network threat detection model that
avoided feature selection and manual extraction. They directly converted original traffic data into two-dimensional image data and employed CNN for detection and classification. Yang et al. [6] utilized long-term and short-term memory networks (LSTM) to extract time series characteristics from network traffic, employing an attention mechanism to capture data dependencies. Hassan et al. [7] designed a CNN and LSTM-based model (WDLSTM) with reduced weight. CNN was used for extracting local features, while WDLSTM preserved the time series features and prevented overfitting. HO et al. [8] transformed data streams into RGB images and classified them using a Vision-Transformer model. Yang et al. [9] proposed an improved ViT model, enhancing the local feature modeling capability by incorporating a sliding window mechanism. Wang W et al. [10] presented a robust unsupervised IDS (RUIDS) by introducing a mask context reconstruction module into the self-supervised learning based on Transformers. This approach maintained detection accuracy and improved the model's robustness.

Most IDS algorithms are trained using static datasets, meaning they can only recognize known attacks that were present in the training set. However, when faced with new and unknown attack methods, these models struggle to effectively classify and detect them. To address this issue, researchers have proposed various methods to identify unknown attacks. For instance, Cruz S et al. [11] utilized W-SVM to detect unknown attacks and evaluated their approach using the KDDCUP'99 dataset. Henrydoss et al. [12] introduced an open-set IDS method based on EVM, which achieved higher accuracy compared to W-SVM. Chen et al. [13] proposed a hierarchical detection model based on conditional variational self-coding. Their model learned the classification function for known attacks and the identification function for unknown attacks separately, resulting in good performance. Li et al. [14] developed a network intrusion generation countermeasure network called IDS-GAN. This approach involved scoring the output of normal network data using a discriminator to establish a normal interval. Any data falling outside of this interval was considered an unknown attack. These advancements aim to enhance the ability of IDS algorithms to detect and classify unknown attacks, thus improving their overall performance.

In practical applications, acquiring comprehensive datasets that encompass all types of intrusion detection poses a significant challenge. The collection of network attack types cannot be accomplished at once, as new attack methods continually emerge over time. Confronted with an increasing number of new and unknown attacks, simply categorizing them as unknown attacks can negatively impact the model's detection accuracy. However, retraining a model after merging new data with the original dataset is often impractical. To address this issue, class incremental learning has been proposed [15, 16]. This approach enables dynamic learning and allows for the updating of model parameters based on input from new classes. Li ZZ et al. [15] introduced the LwF model, which mitigates forgetting of previous knowledge by adding a constrained regularization term to the loss function of new tasks. This knowledge distillation technique freezes network layers and aids in retaining old knowledge. Rebeffisa et al. [16] proposed the iCaRL model, which addresses catastrophic forgetting by preserving representative old data and training it alongside new data. Zhang et al. [17] combined the K nearest neighbor clustering algorithm with the nearest class mean classifier to design an intrusion detection system called OCN, which supports class incremental learning. Wu et al. [18] proposed a method for intrusion detection based on dynamic integrated incremental learning (DEIL-RVM), which facilitates a dynamically adjusted integrated intrusion detection model. Zhang et al. [19] developed an incremental intrusion detection algorithm that utilizes an asymmetric multi-feature fusion self-encoder (AMAE) and a classification depth neural network (C-DNN). The confidence of the AMAE result is used to select the final result from the C-DNN output. This approach not only retains the ability to classify old data but also improves the detection accuracy for new categories.

However, the existing methods face limitations in simultaneously detecting unknown attacks and dynamically adding new attack categories. To address these challenges, this paper introduces a novel algorithm called DUA-IDS, which aims to enhance the classification of both old and new attack categories while improving the detection of unknown attacks. The main contributions of this paper are as follows:

1) This paper proposes a feature extraction module that combines a Transformer encoder [20] and a convolutional neural network (CNN). By leveraging the Transformer encoder, this module extracts time series and global features from the training data. The CNN network is employed to extract local features. The integration of these multi-angle features using a self-attention mechanism improves the model's classification capabilities for both new and old data categories.

2) A threshold-based classifier is introduced in this paper. During the training stage, the classifier calculates the threshold for the nearest mean classifier. In the testing stage, if the distance between the test data and the nearest class exceeds the threshold range, it is classified as unknown data. This approach enhances the model's ability to detect unknown data more effectively.

3) This paper presents a dynamic learning method to mitigate the problem of forgetting old data categories. The proposed method utilizes data playback and knowledge distillation techniques, along with the introduction of a spatial distillation loss function. This approach enhances the model's scalability and ability to retain knowledge about previously encountered attack categories.

III. METHOD

A. System Overview

To enhance the detection accuracy of new attacks, particularly unknown attacks encountered in real scenarios, and to dynamically update model parameters as the number of unknown attacks increases, this paper introduces a novel algorithm called DUA-IDS. The algorithm transforms unknown classes into known classes, thus improving the
model's overall performance. The following modules have been designed:

1) **Feature extraction module**: This module utilizes both a Transformer encoder and a CNN to extract time series information, global information, and local information from traffic data. The extracted multi-angle features are then fused using a self-attention mechanism, thereby improving the model's classification accuracy.

2) **Unknown data detection module**: This module employs a threshold-based classifier. By calculating the threshold of the nearest mean classifier, the module can determine the distance between the test data and the nearest class. If the distance falls within the threshold range, the data is classified as a known class and further categorized accordingly. Conversely, if the distance exceeds the threshold range, the data is classified as an unknown class.

3) **Model dynamic updating module**: For the unknown categories identified in the previous module, these categories are marked using other data cleaning algorithms. Based on the principles of data playback and knowledge distillation, the model continuously learns the unknown categories while retaining the detection accuracy for old categories.

These modules collectively contribute to the DUA-IDS algorithm, enabling improved detection accuracy for emerging new attacks and facilitating the dynamic update of model parameters in response to the increasing number of unknown attacks.

The overall framework of DUA-IDS intrusion detection system is shown in Fig. 1.

![](image)

**Fig. 1.** Overall framework of model.

**B. Feature Extraction Module**

As shown in Fig. 2, this paper proposes a neural network structure with multi-angle feature fusion. It includes Transformer encoder, CNN and Attention Fusion.

![](image)

**Fig. 2.** Feature extractor.

The original Transformer model consists of an encoder and a decoder, but since intrusion detection primarily involves classification, this module only utilizes the encoder component. The encoder is composed of two main sublayers: the Multi-Head Attention mechanism and a fully connected Feed-Forward Network. By incorporating position coding and the self-attention mechanism of the Transformer encoder, it becomes possible to effectively extract global features and time sequence features from each subsequence.

The Multi-Head Attention mechanism serves as the core of the Transformer encoder, enabling the capture of dependencies between data within stream data blocks. This mechanism operates by first linearly transforming the input stream's data block into three vectors: Query, Key, and Value. Next, the attention distribution is computed by comparing the query vector with all the key vectors. Each value vector is then multiplied by its corresponding attention distribution to obtain a weighted value vector. Ultimately, these weighted value vectors are concatenated to form a new feature vector. By utilizing different attention mechanisms, diverse attention representations can be obtained, thereby providing a comprehensive reflection of the information shared among the data within the input data stream block. The calculation formula (1) for the attention mechanism is as follows:

$$\text{Attention}(Q, K, V) = \text{softmax} \left( \frac{QK^T}{\sqrt{d_k}} \right)V$$

Where Q, K and V respectively represent three matrices of Query, Key and Value, and dk is the dimension of Key. The calculation formula (2) of multi-head attention is as follows:

$$\begin{cases} 
Q = XW^Q, K = XW^K, V = XW^V & i = 1, ..., n \\
head_i = \text{Attention}(Q, K, V) & i = 1, ..., n \\
\text{MultiHead}(Q, K, V) = \text{Concat}(head_1, ..., head_n)W^o 
\end{cases}$$

Where $W^Q$ represents a parameter when Q is calculated for input X, $W^K$ represents a parameter when K is calculated for input X, and $W^V$ represents a parameter when V is calculated for input X.

The CNN component comprises two convolution layers, two maximum pooling layers, and a fully connected layer. The two-dimensional convolution neuron (Conv2D) is primarily employed in the convolution layers for processing. In this study, CNN is predominantly utilized to extract local spatial features from network traffic data.

Subsequently, the attention mechanism is applied to integrate global features, time series features, and local features from multiple perspectives. The feature extraction module enables the consideration of correlations and complementarity among features, thereby enhancing the model's expressive and generalization capabilities.

**C. Unknown Data Detection Module**

The nearest class mean classifier is a simple and effective classification algorithm that falls under the category of centroid-based classifiers. In this classifier, each class is represented by the mean vector of its training samples, which serves as a prototype for that particular class. During the
testing phase, the distance between the test sample and each class prototype is calculated, and the test sample is assigned to the class whose prototype is closest to it.

To be more specific, suppose we have a training set consisting of $K$ classes. Let $\mathbf{p}_k$ represent the mean vector of class $k$. The mean vector is obtained by taking the average of all the training samples belonging to that specific class. In order to classify a new test sample $x$, we compute the distance between $x$ and each class prototype.

$$d_k = \| x - \mu_k \|$$

(3)

where $\| \cdot \|$ denotes the Euclidean distance. Finally, the test sample is assigned to the class whose prototype has the smallest distance.

The traditional nearest class mean classifier is limited to classifying known classes only. It determines the class of a test data by calculating the distance between the test data and the centroid of each known class, and assigning it to the class with the closest centroid. Regardless of the actual distance, the test data is always classified into that specific category. This approach poses a problem when encountering unknown categories, as there will always be a known class whose centroid is the closest, even if the actual distance is significantly large. To address this issue, we can enhance the nearest distance calculation. By introducing a threshold on the nearest distance, we can determine whether a test data belongs to an unknown class. If the distance between the test data and the centroid of the nearest class exceeds this threshold, it is classified as an unknown class. The classification calculation method is as follows:

$$\hat{y}_i = \begin{cases} \arg \min_{k \in \{1, \ldots, m\}} d(f(x_i), c_k), & \text{if } \min_{k \in \{1, \ldots, m\}} d(f(x_i), c_k) < \text{threshold}_k \\ m+1, & \text{otherwise.} \end{cases}$$

(4)

Where $d()$ an example of test data and class centroid is, $m$ is the number of classes, threshold$_k$ is the classifier threshold, $c_k$ is the class centroid of class $k$ and $f(x_i)$ is the result of model output.

In this paper, we utilize a threshold-based classifier, as illustrated in Fig. 3, to identify unknown categories. The threshold for the nearest mean classifier is computed based on the centroids of both known and unknown classes, enabling us to distinguish between them.

Algorithm1: Calculation threshold

| Input: Training data $x \in \text{known}$ |
| Require: Number of each category $n = (n_1, n_2, \ldots, n_m)$ |
| Require: Feature extractor $\varphi(x) = \text{model}(x)$ |
| 1: FOR $k = \{1, 2, \ldots, m\}$ DO |
| 2: $\mu_k = \frac{1}{n_k} \sum_{i=1}^{n_k} \varphi(x_i)$ // Calculate class mean |
| 3: END FOR |
| 4: Calculate the minimum distance |
| 5: $\text{Threshold}_k = \min_{k=1\ldots m} \| \varphi(x) - \mu_k \|$ |
| 6: $\text{Threshold} = \max_{k=1\ldots m} \text{Threshold}_k$ // Calculate the classifier |
| //threshold |
| 7: END |

The threshold calculation algorithm and classification algorithm are shown in algorithm 1 and algorithm 2.

Algorithm 2: Classification algorithm

| Input: Training data $x \in \text{known} \& \text{unknown}$ |
| Require: Number of each category $n = (n_1, n_2, \ldots, n_m)$ |
| Require: Feature extractor $\varphi(x) = \text{model}(x)$ |
| 1: FOR $k = \{1, 2, \ldots, m\}$ DO |
| 2: $\mu_k = \frac{1}{n_k} \sum_{i=1}^{n_k} \varphi(x_i)$ // Calculate class mean |
| 3: END FOR |
| 4: IF $\min_{k=1\ldots m} \| \varphi(x) - \mu_k \| < \text{Threshold}$ |
| 5: $\hat{y} = \arg \min_{k=1\ldots m} \| \varphi(x) - \mu_k \|$ //Classified as the |
| //category closest to the center of mass. |
| 6: ELSE |
| 7: $\hat{y} = m + 1$ //unknown attach |
| 8: END IF |
| 9: END |

D. Dynamic Learning Module

When dynamically learning new attack categories, the model often experiences catastrophic forgetting of the known attack categories. The feature extraction module’s training objective is to accurately classify the training flow data into a predefined set of categories. However, when a new attack category emerges, the model needs to update its parameters to incorporate the new information. Unfortunately, as the model updates its parameters, it may “over-fit” the new data and forget the previously learned information, resulting in a significant decrease in the detection accuracy of known attack categories. This decline in accuracy occurs because the new data may differ in distribution or feature space from the old data. Consequently, the model might adjust its parameters in a
way that sacrifices the performance of old data while improving the performance on the new data. To address this challenge, this paper employs a method based on data playback and knowledge distillation to reinforce the learning of known attack categories.

1) Data playback: Once each task is learned, the data playback method saves a subset of samples from each category for future model training, as depicted in Fig. 4. These saved samples, comprising known attack categories, are combined with the data of new attack categories to update the parameters of the current feature extraction module. In this paper, a sample saving strategy is employed. The total number of saved samples is fixed at K, and the number of samples saved in each category is \( n = \frac{K}{m} \), where \( m \) represents the number of currently known attack categories. This approach ensures that the available storage of K samples is utilized efficiently.

Furthermore, this paper employs the "Herding" strategy to select \( n \) representative samples from each category. The strategy operates in iterations to choose N samples. In each iteration, a sample is selected from the current training set and included in the saved sample set. This approach aims to ensure that the average feature vector of the saved feature space for known samples closely aligns with the average feature vectors of all training samples. Consequently, the saved set of known samples functions as a priority queue, with the sample order indicating their significance. The specific process of saving the sample set using the "Herding" strategy is outlined in Algorithm 3.

**Algorithm 3: Sample preservation based on Herding strategy**

- **Input:** Training sample set \( X = \{x_1, \ldots, x_m\} \) belonging to category \( y \)
- **Require:** Number of samples stored in each category \( n \)
- **Require:** Feature extractor \( \phi(x) = model(x) \)

1: Initialize: \( \mu = \frac{1}{m} \sum \phi(x) \)

2: FOR \( k = \{1, 2, \ldots, n\} \) DO

3: \( p_k = \arg \min \sum \phi(x) + \frac{k}{2} \phi(p_k) \)

4: END FOR

5: \( M' \leftarrow (p_1, p_2, \ldots, p_n) \)

2) Knowledge distillation: When learning a new model, the knowledge distillation method requires that the output of the new model aligns with the given data in a consistent manner with the old model. The old model refers to the model that learned the previous task (known attack category) and remains unchanged. The new model inherits the parameters of the old model and can be updated for new tasks. During the training process, the new model is optimized to minimize two loss functions: (i) The standard cross entropy loss between the predictions of the new model and the actual labels, and (ii) Distillation loss, which measures the disparity between the predictions of the new model and the soft targets generated by the old model.

The traditional distillation loss primarily focuses on the distillation output of the final feature layer of both the old and new models. However, this paper delves into the distillation calculation of the output from the middle layer of the feature extraction model. By combining the output of the final feature layer with each layer's output, the new model can effectively absorb the knowledge of the old model and mitigate the problem of catastrophic forgetting for known attack categories.

Representation of distillation loss of final characteristic layer:

\[
L_{\text{end}}(h^{t-1}, h') = \left\| h^{t-1} - h' \right\|^2
\]

Where \( h^{t-1} \) represents the final feature layer output of the old model and \( h' \) represents the final feature layer output of the new model being trained.

Aiming at the feature extraction module in this paper, the data shape after preprocessing the traffic data is as follows: \((\text{bitch\_size}, \text{series}, \text{feature\_size})\), abbreviation \((B, S, F)\). For the output of the middle layer of the feature extraction module, the distillation loss is calculated from two dimensions respectively.

Distillation loss in two dimensions.

\[
L_s(h^{t-1}_f, h'_f) = \sum_{j=1}^{F} \left\| \sum_{s=1}^{S} h^{t-1}_{s,f,j} - \sum_{s=1}^{S} h'_{s,f,j} \right\|^2
\]

\[
L_f(h^{t-1}_f, h'_f) = \sum_{j=1}^{F} \sum_{s=1}^{S} h^{t-1}_{s,f,j} - \sum_{s=1}^{S} h'_{s,f,j} \right\|^2
\]

\[
L_{\text{spatial}}(h^{t-1}_f, h'_f) = L_s(h^{t-1}_f, h'_f) + L_f(h^{t-1}_f, h'_f)
\]

Where \( h^{t-1}_f \) represents the characteristic output of the first layer of the old model, \( h'_f \) represents the characteristic output of the first layer of the new model, and S and F represent the intermediate layer loss in the S and F dimensions respectively.

Combining the final characteristic layer output and the intermediate characteristic layer output as the distillation loss of the model.

**Fig. 4.** Class incremental dynamic learning.
\[ L_{\text{final}}(x) = \frac{\lambda_1}{L-1} \sum_{i=1}^{L-1} f_i^{-1}(x), f_i'(x) + \lambda_2 L_{\text{final}}(f_i^{-1}(x), f_i'(x)) \] (9)

Where the hyperparameter \( \lambda_1 \) and \( \lambda_2 \) are used to balance the output of the intermediate layer and the output of the final layer. \( L \) represents the total number of layers of the feature module, and \( L-1 \) is the other intermediate layers except the final layer.

3) Model parameter updating algorithm: Whenever the model acquires new attack data \( (X_{m+1}, \ldots, X_{m+k}) \) for the new categories \( (m+1, \ldots, m+k) \), these data are utilized to update the parameters of the feature extraction module and the cached sample set. Algorithm 4 outlines the steps involved in updating the feature extraction module parameters.

**Algorithm 4: Class incremental learning algorithm**

**Input:** New category data \( X = \{x_{m+1}, \ldots, x_{m+k}\} \)

**Require:** Old data cached in memory \( P = \{P_1, \ldots, P_m\} \)

1: Initialize: Training data \( X_{\text{train}} = X \cup P \)
2: FOR index, x, label in \( X_{\text{train}} \) DO
3: Save the output of the middle layer of the feature extractor network. \( q[index] = \varphi(x) \)
4: END FOR
5: Train model one epoch:
6: FOR \( x, y \) in \( X_{\text{train}} \) DO
7: \( \hat{y} = \varphi(x) \)
8: \( \text{Loss} = \text{Loss}_{ce} + \text{Loss}_{\text{final}} \)
9: Update parameters
10: END FOR
11: END

First, the cached data of known attack categories and the input data of new attack categories are combined into a new training dataset. Next, the output results of each layer of the old feature extraction module for the known attack categories are stored. Finally, the parameters of the feature extraction module are updated using the loss function, which is a weighted sum of the classification cross entropy loss and the hierarchical distillation loss.

IV. DATASET DESCRIPTION

A. UNSW-NB15

The dataset used in this paper is the UNSW-NB15[21-26], which is a comprehensive dataset designed by the Australian Cyber Security Center Laboratory in 2015 for network intrusion detection systems. The dataset aims to simulate real attack environments using the IXIA traffic generator, based on vulnerability information technology published on the CVE website. It consists of normal traffic and various types of attack traffic, including Fuzzers, Analysis, Backdoors, DoS, Exploits, Generic, Reconnaissance, Shellcode, and Worms.

The dataset comprises 49 features and 10 labels. These features include basic information such as source IP address, destination IP address, and protocol type, content-related features like HTTP method and URI length, time-related features such as connection duration and average byte rate, as well as other features such as service type and status code. The dataset is available in three versions: full version, training version, and test version. For this experiment, we utilize the full version of the CSV data, which contains 2,540,044 data samples.

Considering the significant imbalance in the number of normal samples compared to other samples, random sampling is applied to reduce the impact of this data imbalance on the classification results. Known classes include common attack types like DOS, Generic, Exploits, and Fuzzers, while the remaining classes are treated as unknown classes. Detailed data can be found in Table I.

<table>
<thead>
<tr>
<th>TABLE I. NUMBER OF CATEGORIES IN THE UNSW-NB15 DATASET</th>
</tr>
</thead>
<tbody>
<tr>
<td>Categories</td>
</tr>
<tr>
<td>---------------------------------</td>
</tr>
<tr>
<td>Normal</td>
</tr>
<tr>
<td>Fuzzers</td>
</tr>
<tr>
<td>Dos</td>
</tr>
<tr>
<td>Exploits</td>
</tr>
<tr>
<td>Generic</td>
</tr>
<tr>
<td>Backdoor</td>
</tr>
<tr>
<td>Analysis</td>
</tr>
<tr>
<td>Reconnaissance</td>
</tr>
<tr>
<td>ShellCode</td>
</tr>
<tr>
<td>Worms</td>
</tr>
</tbody>
</table>

V. EXPERIMENT

A. Experimental Environment

Experimental configuration: Ubuntu as the operating system, Intel Xeon Gold 5118 as the processor, 32GB of memory, NVIDIA GeForce RTX 3090 as the GPU, Python3.8 as the programming language and Pytorch 1.11.0 as the learning framework.

B. Evaluation Indexes

This experiment is a classification model, using a confusion matrix to evaluate the classification structure. The confusion matrix is shown in Table II.

<table>
<thead>
<tr>
<th>TABLE II. THE CONFUSION MATRIX</th>
</tr>
</thead>
<tbody>
<tr>
<td>The real situation</td>
</tr>
<tr>
<td></td>
</tr>
<tr>
<td>True</td>
</tr>
<tr>
<td>False</td>
</tr>
</tbody>
</table>

The evaluation indexes used in the experiment mainly include Accuracy, Precision, Recall and F1-score.

Accuracy refers to the proportion of the number of samples that can be correctly classified by the model in the total samples.

\[ \text{Accuracy} = \frac{TP + TN}{TP + FN + FP + TN} \] (10)

Precision represents the ratio of the number of correctly classified samples to the number of samples retrieved.
\[
\text{Precision} = \frac{TP}{TP + FP} \tag{11}
\]

Recall is the ratio of the number of correctly classified samples to the number of correctly classified samples.

\[
\text{Recall} = \frac{TP}{TP + FN} \tag{12}
\]

\[
F_1\text{-score} = \frac{2 \times P \times R}{P + R} \tag{13}
\]

C. Results and Discussion

1) Hyperparameter setting: Through many experiments, the values of parameters are adjusted to achieve the best experimental results. In the process of adjustment, the super parameters of the best experimental results are obtained, as shown in Table III.

<table>
<thead>
<tr>
<th>Hyperparameter</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Epochs</td>
<td>50</td>
</tr>
<tr>
<td>Learning rate</td>
<td>1e-5</td>
</tr>
<tr>
<td>Time series</td>
<td>10</td>
</tr>
<tr>
<td>Bitch size</td>
<td>10</td>
</tr>
<tr>
<td>Number of headers</td>
<td>8</td>
</tr>
<tr>
<td>Number of layers</td>
<td>3</td>
</tr>
<tr>
<td>Embding size</td>
<td>768</td>
</tr>
<tr>
<td>K</td>
<td>2000</td>
</tr>
</tbody>
</table>

2) Experimental analysis of known attack detection: To evaluate the detection performance of DUA-IDS on both known categories and unknown attacks, we have selected several advanced intrusion detection methods as baselines.

a) Methods for detecting unknown attacks:

i) EVM [12]: Extreme Value Machine (EVM) is a classifier that detects unknown attack categories using non-nuclear, non-linear, and variable bandwidth outlier detection.

ii) IDS-GAN [14]: IDS-GAN defines the normal interval based on the scoring of normal network data by the discriminator. Any data outside this interval is considered an unknown attack.

b) Methods unable to detect unknown attacks:

i) CNN-BiLSTM [27]: This model combines CNN, BiLSTM, and self-encoder to extract high-dimensional traffic features and self-monitoring features, aiming to improve the classification accuracy.

ii) CNN-WDLSTM [26]: CNN-WDLSTM adopts a combination of CNN and LSTM (WDLSTM) with reduced weight. CNN is used to extract local features, while WDLSTM preserves time series features and prevents overfitting, thus enhancing the classification accuracy.

iii) RUIDS [10]: RUIDS is a robust unsupervised intrusion detection system and one of the most advanced closed-set intrusion detection algorithms. It incorporates a shielded context reconstruction module into the self-supervised learning scheme based on the transformer. The self-supervised learning scheme captures internal relationships within the learning context.

Fig. 5 illustrates the loss function of these six groups of models during the training process on known classes using the UNSW-NB15 dataset. The figure demonstrates that all six models exhibit good convergence on the training dataset, yielding optimal parameters. Subsequent experiments will evaluate the model’s performance on the test set using these optimal parameters.

Table IV presents the intrusion detection results of the model proposed in this paper on the test set of the UNSW-NB15 dataset. It compares the classification accuracy of our model with other models of different types on known attack categories. From the data in the table, it is evident that the classification accuracy of intrusion detection algorithms capable of detecting unknown categories in the past is not as high as those based on closed sets (where unknown categories cannot be detected). To enable the detection of unknown categories, our model adopts a non-optimal classification strategy. Consequently, its classification accuracy on known categories is slightly lower than that of closed-set detection models.

<table>
<thead>
<tr>
<th>UNSW-NB15</th>
<th>Acc</th>
<th>F1</th>
<th>Pre</th>
</tr>
</thead>
<tbody>
<tr>
<td>EVM</td>
<td>0.841</td>
<td>0.8379</td>
<td>0.8345</td>
</tr>
<tr>
<td>IDS-GAN</td>
<td>0.8582</td>
<td>0.8627</td>
<td>0.8442</td>
</tr>
<tr>
<td>CNN-BiLSTM</td>
<td>0.8689</td>
<td>0.865</td>
<td>0.8582</td>
</tr>
<tr>
<td>CNN-WDLSTM</td>
<td>0.8786</td>
<td>0.8725</td>
<td>0.8688</td>
</tr>
<tr>
<td>RUIDS</td>
<td>0.891</td>
<td>0.8908</td>
<td>0.8871</td>
</tr>
<tr>
<td>DUA-IDS</td>
<td>0.8847</td>
<td>0.8854</td>
<td>0.8794</td>
</tr>
</tbody>
</table>

Fig. 5. The loss vs. epoch under different model.
Compared with other models, EVM adopts the traditional method to classify network flow data based on outliers. In binary classification, it may have better classification accuracy. For multi-classification tasks, the accuracy of multi-classification is weaker than other models.

DUA-IDS is superior to IDS-GAN in the utilization of time series features. DUA-IDS obtains more comprehensive characteristics of network stream data.

Compared with CNN-BiLSTM and CNN-WDLSTM, the Transformer proposed in this paper combines CNN's feature extractor to analyze the time, global and local features of data.

However, the classification accuracy of the model proposed in this paper surpasses that of most closed-set-based intrusion detection algorithms. It is only 0.63% lower than the most advanced closed-set algorithm, RUIDS. This demonstrates the model's strong advantages in feature extraction and classification.

3) Experimental analysis of unknown attack detection: In real-world scenarios, there may be attack categories that are unseen and do not belong to our known training set. To address this, we employ a nearest class mean classifier with a threshold to identify unknown attack categories in this paper. To evaluate the performance of the proposed model in detecting unknown attack categories, we compare DUA-IDS with both closed-set-based intrusion detection models and models capable of detecting unknown categories. The experimental results are depicted in Fig. 6.

In this paper, we systematically introduce unknown categories into the test dataset one by one, following the order specified in Table II. The number of added unknown categories is represented as "n=1" in Fig. 6. As depicted in Fig. 6(a), it is evident that our proposed DUA-IDS outperforms the closed-set-based intrusion detection model as the number of unknown attack categories increases. The closed-set model can only recognize data from known categories, and when confronted with new categories, it mistakenly classifies them into known categories, resulting in a continuous decline in classification accuracy. This demonstrates the necessity for intrusion detection models to possess the ability to detect unknown classes.

Fig. 6(b) illustrates the classification accuracy of DUA-IDS and other comparison models, which tends to stabilize as the number of unknown attack categories increases. These models effectively identify most unknown attacks as unknown classes, thereby maintaining high classification accuracy. Notably, the proposed DUA-IDS consistently outperforms the comparison model in classification accuracy, highlighting its superior capability in detecting unknown attack categories.

4) Experimental analysis of dynamic class increment: To evaluate the detection performance of DUA-IDS on both original and new categories after updating the model parameters with the addition of new categories, we compare it with the OCN [17] model in this paper.

OCN: The OCN model combines the concepts of nearest class classifier and K nearest neighbor clustering. It directly incorporates the clustering results into the classifier's category list, enabling incremental learning of categories.

Fig. 7 illustrates the changes in classification accuracy for both DUA-IDS and OCN models when new categories are added. We incrementally introduce one new category at a time, denoted as "n=1" in the figure. As depicted in the figure, as the number of new categories increases, the classification accuracy of both models decreases. However, the proposed DUA-IDS in this paper exhibits a slight decrease in accuracy when new categories are added, while the OCN model experiences a noticeable decline. This indicates that the DUA-IDS model proposed in this paper possesses significant advantages in dynamically updating the model.
5) Ablation experiment: In the DUA-IDS proposed in this paper, we have incorporated two key elements: a feature extraction module based on transformer and spatial distillation. To further investigate their effectiveness, we conducted ablation experiments focusing on these two aspects.

a) CNN-DUA-IDS: To evaluate the impact of the feature extraction module based on transformer designed in this paper, we conducted a control experiment by replacing it with a simple CNN module. As illustrated in Fig. 8, the utilization of the CNN module resulted in a decrease in overall model accuracy. This clearly demonstrates the significant role played by the transformer-based feature extraction module in enhancing the classification accuracy of the model.

b) icarl-DUA-IDS: To investigate the impact of the introduced spatial distillation loss in this paper, we conducted a control experiment by removing it from the DUA-IDS model. As depicted in Fig. 8, when a new category is added, the classification accuracy of icarl-DUA-IDS noticeably decreases. This observation highlights the substantial contribution of the spatial distillation loss in mitigating the effects of knowledge forgetting.

VI. CONCLUSION

In the actual network environment, network data is dynamic and will produce new types of unknown network data in real time. The traditional intrusion detection system based on static data can't adapt well to new types of data, which leads to the decrease of classification accuracy. In addition, the increasing number of unknown data also poses a potential threat to the stability of the model.

This paper addresses the challenges of detecting unknown network attacks and dynamically updating models in network intrusion detection. To tackle these issues, a dynamic intrusion detection system capable of detecting unknown attacks is proposed. The model is mainly composed of three parts. Firstly, in the feature extraction module, this paper proposes to combine the multi-angle features of network data by combining Transformer with CNN. Secondly, the nearest class mean classifier based on threshold is used to find the potential unknown attack categories. Thirdly, the unknown class data are updated by class increment method based on data playback and distillation learning. The experimental results show that the method proposed in this paper is effective in detecting unknown data and stable after dynamically learning new types of data.

However, the DUA-IDS model has certain limitations when it comes to handling detected unknown attack categories and incorporating them back into the model. Therefore, additional measures are required to cleanse and label the identified unknown attack data before further learning. Future research efforts will focus on refining the process of handling and relearning unknown data, aiming to achieve automated self-learning in network intrusion detection models.

ACKNOWLEDGMENT

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REFERENCES


Computational Framework for Analytical Operation in Intelligent Transportation System using Big Data

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Abstract—Intelligent Transportation System (ITS) is the future of the current transport scheme. It is meant to incorporate an intelligent traffic management operation to offer vehicles more safety and valuable traffic-related information. A review of existing approaches showcases the implementation of varied scattered schemes where analytical operation is mainly emphasized. However, some significant shortcomings are witnessed in efficiently managing complex traffic data. Therefore, the proposed system introduces a novel computational framework with a joint operation toward analytical processing using big data targeting to manage raw and complex traffic data efficiently. As a novel feature, the model introduces a data manager who can manage the complex traffic stream, followed by decentralized traffic management, that can identify and eliminate artefacts using statistical correlation. Finally, predictive modelling is incorporated to offer knowledge discovery with the highest accuracy. The simulation outcome shows that Random Forest excels with 99% accuracy, which is the highest of all existing machine learning approaches, along with the accomplishment of 11.77% reduced accuracy, 1.3% of reduced delay, and 67.47% reduced processing time compared to existing machine learning approaches.

Keywords—Intelligent transportation system; traffic management; machine learning; artefacts; prediction

I. INTRODUCTION

With the proliferation of advancements in network and communication technologies, transportation services have been visioned to incorporate smart features [1]. From this perspective, Intelligent Transportation System (ITS) has evolved in the form of innovative services associated with smart traffic management [2]. The prime agenda of ITS is to offer the user the most valuable information associated with traffic to make the driving experiences safer and well-synchronized [3]. Adoption of ITS is found to offer more reduction of cost owing to the adoption of preemptive preventive measures, minimizes the consumption of resources, and significantly emphasizes accident prevention [4]. Various application of ITS includes smart ports and maritime [5], smart airport [6], smart railways [7], fleet management [8], and smart road [9]. The prime idea of ITS is basically to capture all the essential parameters that connect to road navigation and safety and subject them to specific analytical processing. In contrast, the system's outcome of analytical processing is used to formulate a strategy for effective traffic management [10]. At present, various sets of research are being carried out towards improving the performance of ITS from various perspectives [11]-[15]. However, various scattered challenges are associated with the research-based studies and practical implementation of ITS. The research-based studies on ITS have reported various challenges, e.g., the need for cost-effective analytical processing, faster learning schemes with higher reliability, cost-effective computational modelling, and benchmarked schemes. At the same time, the practical implementation of ITS suffers from different challenges, e.g., constraints of budgets, integration with the conventional scheme, coping with updated standards and technology, rules and traffic regulation of different countries, etc. However, there is no denying that the success factor of an ITS majorly depends upon the accuracy and performance of its analytical capability. At present, the big data approach is the most suitable mechanism to deal with complex data [16]; however, they are in a very nascent stage of development. The various complexities associated with the demanded analytical processing of data in ITS are: i) ITS is characterized by the massive generation of traffic data with higher complexities within it, ii) the point of traffic data generation is multiple, and the rate of transmission is uneven and depends upon various network performance leading to error-prone data, iii) with increasing adoption of artificial intelligence and machine learning, the improvement in the predictive analytical model is significantly less owing to the inclusion of critical size of dynamic constraints. Therefore, the proposed scheme presents a novel computational framework of joint analytical operation using a big data approach to improve ITS knowledge discovery. Unlike any existing approaches reported in the literature, the study contributes toward a novel analytical model capable of extracting knowledge from raw and complex forms of traffic data. Another significant contribution of the proposed scheme is offering quality traffic data by performing a fusion of data and identifying and eliminating artefacts. The scheme implements a novel yet simplified machine learning scheme to optimize the predictive performance of traffic data further. The manuscript's organization is as follows: Section II discusses the existing scheme of analytical operations in ITS and highlights research problems in Section III. Section IV briefs about the research methodology, while an elaborated discussion of system design concerning the algorithm is carried out in Section V. Section VI presents the result and justification of the acquired result in the proposed analysis. At the same time, Section VII makes a
II. REVIEW OF LITERATURE

Various work have been carried out to improve the analytical performance of traffic data in ITS. The work by Qi et al. [17] has presented a discussion about the influence of mobility patterns in investigating multiple constraints in traffic management. The study has implemented an ensemble clustering method along with a tensor-based factorization scheme for this purpose. Further adoption of clustering is also witnessed in the work of Huang et al. [18], which emphasizes evaluating the significance of traffic nodes in ITS considering geographic road networks. Liu et al. [19] discuss a unique study model to investigate variability in travel patterns in public transport systems. Analytical modelling is carried out towards quantifying intra-personal variability. Gu et al. [20] have used a deep learning-based approach to predict short-term traffic volume. The study model has used an enhanced Bayesian model integrated with multiple deep learning approaches where correlation analysis is carried out for traffic flow for current and prior time. Chen et al. [21] used the Gated Recurrent Unit (GRU) neural network model to forecast vehicle speed in urban traffic systems. This predictive model facilitates formulating strategies to control traffic and support navigation systems.

All the schemes mentioned above are mainly associated with a single flow of traffic, which are relatively less seen in practical ITS networks. This gap is addressed in the work of Wang et al. [22], where traffic flow estimation is carried out based on trajectories of license plate information embedded with GPS data. The adoption of big data is witnessed in Bakdi et al. [23] work has investigated the possibilities of multiple risk factors associated with ship trials. The study integrated an autonomous identification system with a digital map with a big data approach to realize the spatial and temporal dependencies of various connecting behaviour of vessels. Further adoption of the big data approach is seen in the work of Qin et al. [24], which is meant for tracking routes of vehicles in urban transport systems. The work identifies the vehicle from license plate recognition, while fuzzy logic tracks routes.

It has also been noted that Artificial Intelligence (AI) significantly contributes to the transport system. Besinovic et al. [25] have discussed current insights into railway transport applications using AI. Zhu et al. [26] have presented analytical modelling of traffic data considering transport systems using Hadoop, Spark, and multiple open-source-based software. Predictive analysis is carried out by Long Short-Term Memory (LSTM) and Support Vector Regression (SVR) to investigate various travel-related operations. The work carried out by Gunes et al. [27] has presented an analytical model deployed for the management of traffic lights and scheduling in ITS. The relationship between edge computing and ITS is discussed by Zhou et al. [28], where the study specifies significant challenges in sensing in ITS. The discussion presented by Lucic et al. [29] also stated the usage of Crowdsourcing in ITS. Mirboland and Smarsly [30] developed a novel and simplified information modelling for ITS using a semantic model. Asaithambi et al. [31] have presented a big-data architecture for micro-services for ITS facilitating the processing of both stream and batch of big data. Choosakun et al. [32] have presented insight into cooperative ITS. Yoo et al. [33] have presented an ITS scheme considering sensory data processing using a big data approach based on open-source software. A similar approach is also adopted in the work of Alexakis et al. [34]. Dudek and Kujiawski [35] have presented a big data scheme for optimizing route planning and interval of the passage of vehicles considering image features. Table I summarizes the above-mentioned methodologies' effectiveness with respective advantages and limitations. The following section outlines research issues associated with the literature.

<table>
<thead>
<tr>
<th>Authors</th>
<th>Method</th>
<th>Advantage</th>
<th>Drawbacks</th>
</tr>
</thead>
<tbody>
<tr>
<td>Qi et al. [17]</td>
<td>Ensemble Clustering, tensor</td>
<td>Simplified evaluation</td>
<td>Highly iterative scheme</td>
</tr>
<tr>
<td>Huang et al. [18]</td>
<td>Machine learning, clustering</td>
<td>Effectively identify critical nodes</td>
<td>Induces overhead in increased traffic</td>
</tr>
<tr>
<td>Liu et al. [19]</td>
<td>Analytical model</td>
<td>Variability evaluation for travel pattern</td>
<td>Narrowed constraint modelling</td>
</tr>
<tr>
<td>Gu et al. [20]</td>
<td>Bayesian, Deep Learning</td>
<td>Higher accuracy</td>
<td>Computationally complex</td>
</tr>
<tr>
<td>Chen et al. [21]</td>
<td>Bidirectional GRU</td>
<td>Minimize overfitting, benchmarked</td>
<td>It doesn't deal with user-based data</td>
</tr>
<tr>
<td>Wang et al. [22]</td>
<td>Data-driven approach</td>
<td>Supports Multi-fold validation</td>
<td>Dependent on specific data</td>
</tr>
<tr>
<td>Bakdi et al. [23]</td>
<td>Autonomous system identification, big data</td>
<td>A practical definition of manifold risk factors</td>
<td>Scope limited to specific forms of traffic</td>
</tr>
<tr>
<td>Qin et al. [24]</td>
<td>Big data, fuzzy logic</td>
<td>Satisfactory accuracy</td>
<td>Dependent on rule-sets</td>
</tr>
<tr>
<td>Besinovic et al. [25]</td>
<td>Review work</td>
<td>Emergence of AI</td>
<td>Study limited to rail transport</td>
</tr>
<tr>
<td>Zhu et al. [26]</td>
<td>LSTM, SVR, open-source software</td>
<td>User-friendly application</td>
<td>No benchmarking</td>
</tr>
<tr>
<td>Gunes et al. [27]</td>
<td>Traffic signal management</td>
<td>Offers adaptive control</td>
<td>Applicable only at the intersection</td>
</tr>
<tr>
<td>Zhou et al. [28], Lucic et al. [29], choosakun et al. [32]</td>
<td>Review on edge computing, Crowdsourcing, cooperation in ITS</td>
<td>Elaborated discussion</td>
<td>-/NA-</td>
</tr>
<tr>
<td>Mirboland &amp; Smarsly [30]</td>
<td>Information model, semantic</td>
<td>A simplified and effective model</td>
<td>No benchmarking</td>
</tr>
<tr>
<td>Asaithambi et al. [31]</td>
<td>Big data, stream/batch processing</td>
<td>Improved predictability</td>
<td>No benchmarking</td>
</tr>
<tr>
<td>Yoo et al. [33], Alexakis et al. [34]</td>
<td>Open-source Software</td>
<td>Simplified implementation</td>
<td>Cannot reach scalability</td>
</tr>
<tr>
<td>Dudek &amp; Kujiawski [35]</td>
<td>Image, big data, thresholding</td>
<td>Optimized route planning</td>
<td>Applicable to image data only</td>
</tr>
</tbody>
</table>

III. RESEARCH PROBLEM

Existing approaches towards improving analytical operations in ITS reviewed in prior sections offer some of the essential mechanisms; however, some of the shortcomings of the existing schemes are as noted below:
• Non-Inclusion of Complex Traffic Data: Most existing approaches [20]-[35] have considered highly structured and detailed datasets publicly available. However, the practical implementation of multiple devices in traffic monitoring of ITS results in complex traffic data, which are difficult to analyze and occupy a more significant segment of storage units. Existing studies do not address such issues, which are the preliminary steps of any analytical operation in ITS.

• Non-Inclusion of Big Data Characteristics: A larger-sized data with streamed data, including possible artefacts, and less value added is not emphasized for the existing big data-based approach. Some existing approaches have used open-source tools, e.g., Hadoop, to address this, but such software requires more reengineering and is also associated with various reported pitfalls [26].

• Lack of Storage Utilization Aspects: In most existing studies, the traffic data is first subjected to storage, and the claimed analytical algorithms are subjected to them in the same place to acquire knowledge [26][30]. The outcomes are stored in multiple storage containers in this process. Unfortunately, such a process over-utilizes the storage space, adversely affecting the query processing and leading to higher delay.

• Adoption of Learning Approaches: There is no denying the fact that machine learning and deep learning significantly contribute to predictive traffic data analysis [21][26]. However, most schemes adopt a complex learning strategy to acquire higher accuracy. Such accomplishment of accuracy is claimed at the cost of highly iterative training operation where higher accuracy is proportional to higher availability of trained error-free data. Confirming error-free data in distributed traffic networks with heterogeneous sensing devices is a computationally challenging task that is not reported to be addressed.

• Lack of Consideration of Model Cost: A model for ITS is required to respond faster, which entirely depends upon a more brilliant construction of its logical condition and lesser usage of traffic attributes with more meaningful insights. This demands a novel and benchmarked strategy implementation towards the knowledge discovery process, considering the decentralized environment of ITS. Very few models are yet reported to accomplish this objective.

All the research problems mentioned above are identified to have concrete solutions; hence, the proposed scheme deploys a novel strategy to address these open-end research problems in ITS. The following section summarizes the methodology adopted to evolve a novel ITS management solution.

IV. RESEARCH METHODOLOGY

The prime goal of the proposed research work is to introduce a unique knowledge discovery method using a novel big data approach in ITS. The core ideology constructed in the proposed scheme is to acquire a raw form of complex traffic data from multiple sources, followed by performing a decentralized scheme of analyzing the traffic data by uniquely transforming them. The implementation of the proposed scheme is carried out in analytical research methodology, and its architecture is presented in Fig. 1 as follows:

The reasons for adopting the above-mentioned architecture to address the identified problems are manifold.

• A closer look into the research problems showcases that analytical studies towards ITS don’t offer emphasis to address the issues of essential big data characteristics and storage management. Both are considered as two different sets of issues. However, the proposed study introduces a mechanism where both issues are handled simultaneously, exhibiting the cost-effective operational features towards ITS.

• It was also noted that existing analytical approaches demands complex usage of mining or knowledge discovery process overlooking the implementation cost involved in it. However, the proposed architecture offers a single-window operation where indexing, transformation, and progressive knowledge discovery processes are carried out on the same platform. This reduces the dependency on including heavy-weighted analytical algorithms for processing and analyzing large-scale ITS data.
From the perspective of model cost, it is observed that existing schemes take the input of the complete stream, process the queue, store the raw data in a cloud storage unit and then perform analytical operations. This consumes a lot of processing time and demands the involvement of various resources. At the same time, the proposed architecture can process the stream of ITS data while it stores a part of the data in the cloud and part of the data in a temporary buffer, which reduces the load of processing and storing significantly. Owing to this phenomenon, the filled-up queue can be emptied soon, and the network can process more incoming data. Hence, the proposed architecture offers a practical solution for processing and analyzing extensive ITS data even in peak traffic conditions.

The prime limitation of existing distributed data storage and analytical architecture is that it cannot jointly address the identified research problem. Apart from this, no existing architecture can optimize the spontaneous storage saturation issues in storage units or offer a faster, more progressive, and less iterative knowledge discovery process for evolving ITS data. The architecture exhibited in Fig. 1 offers three explicit blocks of operation toward knowledge discovery of traffic data in ITS. Considering the input of raw traffic data, the first block of operation performs stream management where a series of transformations is carried out over programmatically formulated synthetic data to acquire preliminary traffic-based knowledge finally. The outcome of the first operational block is further subjected to traffic data fusion over a buffer allocated for multiple streams. This operation assists in identifying possible forms of artefacts present in traffic data, eliminating them, and generating pure data. A statistical correlation-based mechanism is adapted to generate the possible value with the highest correlation, followed by substituting the artefact with statistically computed data. The third operational block is finally responsible for retaining the computed data’s highest reliability and trustworthiness. A machine learning scheme is applied for this purpose which uses primary and targeted thresholds to assess the genuineness of the accuracy obtained in the outcome of prior block operation. The outcome of this model offers predictive traffic data, a knowledge discovery considered value-added analytical information. The block operation is elaborated and illustrated in the next section.

V. SYSTEM DESIGN

The proposed scheme's prime idea is to apply a novel analytical operation to offer a value-added ITS performance. An explicit system design using a big data approach has been developed, considering a typical use case for managing and analyzing ITS traffic information streams. The core motive is to develop a simplified and lightweight analytical framework for efficient knowledge discovery of complex forms of traffic data. The description of the proposed system design implemented for the proposed scheme follows.

A. Modelling Traffic Data Manager

This is the first module of implementation of the proposed system, which targets minimizing the complexities associated with streaming traffic-related information from various sensing devices installed on roads and vehicles. To carry out an investigation, there are two mechanisms to acquire the traffic-related information, viz., depending on publicly available datasets and acquiring realistic traffic data. There are currently various publicly available datasets for ITS, e.g. [36]-[37]; however, this dataset does not possess any characteristic of big data complexities of voluminous and various factors and is not featured with uncertainties. Hence, adopting already engineered and processed traffic-related information will not allow the model to be assessed for its capability to process a complex data stream. On the other hand, the acquisition of realistic data demands the involvement of an extensive experimental infrastructure with the inclusion of sensors. Even if small experimental prototyping is done, the outcome cannot be evaluated to prove its applicability for large-scale scenarios.

<table>
<thead>
<tr>
<th>Item</th>
<th>Traffic Attribute</th>
<th>Values</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>Vehicle ID</td>
<td>001</td>
</tr>
<tr>
<td>2</td>
<td>Vehicle Type</td>
<td>Car</td>
</tr>
<tr>
<td>3</td>
<td>Sensor Type</td>
<td>T1</td>
</tr>
<tr>
<td>4</td>
<td>Date</td>
<td>20/02/2023</td>
</tr>
<tr>
<td>5</td>
<td>Location ID</td>
<td>NH48</td>
</tr>
<tr>
<td>6</td>
<td>Sensor Status</td>
<td>Active</td>
</tr>
<tr>
<td>7</td>
<td>Sensor Range</td>
<td>10m</td>
</tr>
<tr>
<td>8</td>
<td>Driver Name</td>
<td>John Thomas</td>
</tr>
<tr>
<td>9</td>
<td>Driving Experience</td>
<td>The lane is heavily congested</td>
</tr>
</tbody>
</table>

Hence, the proposed scheme develops synthetic traffic data characterized by the complexity attributes of big data in ITS. The complexity is added by streaming the data in massive volume and programmatically making the data highly unstructured, which is challenging for any processor to read and analyze. This scenario can be mapped with complex data acquisition events in a heavy incoming stream of sensed traffic data in ITS. Table II highlights the sample traffic data that are synthetically generated. To ensure better compliance with the significant data form of the repository, the proposed study has visited the existing format for publicly available big data [38] and ensured that synthetic data carry the exact form. The synthetic data is in plain text to ensure that such data could have possible artefacts apart from the autonomous data entry-based errors from the sensing devices. This is done intentionally to ensure the possibility of data with an artefact. Further, the data is programmatically converted to its most complex form as follows:

```
VehicleID0012VehicleTypeCar3SensorType
T14Date20/02/2023LocationIDNH486Sensor
StatusActive7SensorRange10m8DriverNameJohn
Thomas9DrivingExperienceThe lane is heavily congested
```
Algorithm for Traffic Data Manager

Input: \( s_d \)
Output: \( k \)

Start
1. For \( i=1:s_d \)
2. \( U_i \leftarrow f_1(i) \)
3. \( f_{ex} \leftarrow f_2(U_i) \)
4. \( d=[f_{ex}, d_i, \text{val}] \)
5. \( F_{ex} \rightarrow s_d(\text{ind}(f_{ex})) \)
6. \( \text{sem}_d \leftarrow f_3(d(\text{val})) \)
7. Apply \( f_3(\text{sem}_d) \rightarrow d_{int} \)
8. \( k \leftarrow f_4(d_{int}) \)
9. \( k \rightarrow s_d(\text{ind}(F_{ex}, k)) \)
10. End

End

The discussion of the algorithmic steps is as follows: The algorithm takes the input of \( s_d \) (stream of data) that generates a resultant of \( k \) (extracted knowledge) upon processing. The proposed scheme considers a stream of data \( s_d \) as an input; however, processing an infinite data stream is infeasible. Hence, \( s_d \) is sampled and considered an input for the proposed scheme.

![Fig. 2. Preparing for streamed input of traffic data.](image)

Fig. 2 showcases that the proposed scheme considers its input \( s_d \) with a size equivalent to the size of the queue capacity \( m \) maintained by the service provider or application buffer. Hence, it is technically possible to process the streamed data \( s_d \) from different traffic zones (Line-1). The streamed data \( s_d \) is then programmatically transformed to unstructured data \( U_d \) using an explicit function \( f_1(x) \). The prime task of this function is to obtain the stream data and eliminate all white spaces to form \( U_d \), which is challenging for the machine to read (Line-2). The following line of execution of the algorithm is to extract significant fields \( f_{ex} \) using an explicit function \( f_2(x) \) (Line-3). The prime tasks of this function are: i) it reads the complete string of the stream data \( sd \) and captures the significant fields \( f_{ex} \) (basically the traffic attributes in Table II), Finding and confirming the fields are simple as they will keep repeating in one individual data in \( s_d \) (however, their corresponding values \( \text{val} \) will differ), and ii) The function also identifies a differentiator \( d_i \) that exists between field \( (F_{ex}) \) and value \( (\text{val}) \). Further, a record \( d \) is constructed, which retains information about extracted field \( f_{ex} \), differentiator \( d_i \), and corresponding values \( \text{val} \) (Line-4).

The part of the implementation of the proposed algorithm associates it with the preliminary storage optimization. The algorithm stores extracted field \( f_{ex} \), indexes them and stores them in storage unit \( s_u \) (Line-5). Hence, the variable \( f_{ex} \) and \( F_{ex} \) mean original and indexed features, respectively. However, the algorithm doesn’t store the corresponding values \( \text{val} \); instead, it keeps them in a temporary memory where it is further subjected to a transformation process (Line-6). An explicit function \( f_3(x) \) is constructed, which uses document tagging for all the corresponding values (as well as indexed extracted features) to generate a semi-structured traffic data \( s_d \) (Line-6). This operation leads to all the intermediate data \( d_{int} \). The term intermediate data \( d_{int} \) will mean that each extracted value \( \text{val} \) is mapped with defined contextual factors constructed in the corpus while developing the synthetic data. The core reason behind this is that the last traffic attribute in Table II bears a longer string message which is computationally challenging for the machine to process and analyze. Hence, a small repository of contextual traffic factors is designed and subjected to string comparison with all the individual words in traffic attributes to extract the actual contextual meaning the machine can understand. This operation is carried out using function \( f_4(x) \), which finally generates knowledge \( k \) (Line-8). The extracted knowledge \( k \) is indexed using the self-constructed \( \text{ind} \) method and stored alongside the priorly extracted field \( F_{ex} \) (Line-9).

The contribution of this algorithm is as follows:

- This algorithm can process voluminous amounts of traffic-related streamed data, unlike existing approaches applied to static data.
- The algorithm permanently stores a part of iterative message fields in storage and a part of the message for further processing to offer better storage utilization. Existing approaches consider storing entire data and processing where the processed data is kept in different locations while original data is considered meta-data. This increases query processing time in existing approaches.
- A unique indexing mechanism that reduces the processing time for streamed dataset is introduced.

B. Decentralized Scheme of Traffic Management

This second implementation module incorporates a decentralization scheme in proposed traffic management. According to the highlights of Fig. 3, the proposed study considers that multiple traffic environments collect traffic information to generate an individual data stream. Upon passing through the network interface using the prior algorithm, the data transform to a streamed data \( sd \), which ultimately generates a knowledge \( k \). However, all this knowledge-based information is further indexed and repositied to the storage unit \( su \) in the prior algorithm, which is again subjected to analytical data operation. To ensure the practicality of this implementation, it is necessary to introduce a data fusion technique in a decentralized manner over distributed cloud environment. This will further lead to the possibility of artefacts. Hence, this decentralization scheme further assists in identifying the location of artefacts, followed by adopting statistical operations to eliminate the artefacts. This process leads to the generation of pure traffic data.
For this purpose, an algorithm is designed to perform this decentralization operation toward further traffic data management. It is to be noted that this algorithm is explicitly executed in a storage unit on top of the final data (i.e., $k$) collected by the prior algorithm. To optimize computational speed and utilization of storage units, a buffer space matrix is constructed by extracting the available empty spaces of storage units in sharing form from multiple storage units. It will eventually mean that algorithm must be executed in a decentralized manner. Following are the steps of the proposed algorithm:

**Algorithm for Decentralized Traffic Management**

**Input:** $k$, $s_u$  
**Output:** $d_p$

**Start**

1. For $j=1$ to $k$
2. alloc $j \rightarrow$ buf($s_u$)
3. $d_{ai} \leftarrow f(s_u, j)$
4. $c_{sol} \leftarrow f(j, (d_{ai}, s_u(k)))$
5. $d_p \leftarrow \arg\max(c_{sol})$

**End**

The discussion of the algorithmic steps is as follows: The algorithm takes the input of $k$ (knowledge) and $s_u$ (index values in a storage unit) that, after processing, yields an outcome of $d_p$ (pure data). A counter variable $j$ represents the number of incoming data streams mainly focusing on newly analyzed data of $k$ from the prior algorithm (Line-1). Each $j$ stream of $k$ data is then allocated to a matrix formulated by a shared buffer from all the available decentralized storage units $s_u$ using method buf (Line-2). The prime reason to perform this operation is to ensure that no operation towards fused stream data is carried out directly over storage units, unlike the majority of the existing schemes. Fig. 4 offers a pictorial representation of the allocation of streams over unused memory over existing storage units. The advantage of this mechanism is that it utilizes unused memory from different cloud storage units to process the incoming streams of analyzed data. The following line of operation is associated with identifying errors or artefacts in the data ($d_{ai}$). An explicit function $f(x)$ is constructed to identify an artefact in the $j$ stream concerning distributed index storage units $s_u$ (Line-3). The outcome will lead to the generation of the location address of the cell in shared memory, which is witnessed with artefacts ($d_{ai}$). The term artefact will represent illegitimate or illogical data that offers no significance towards either simplified understanding or could ever be processed.

Such data could be caused due to network or sensor-based errors while propagating the data. Upon identifying the artefacts (dart), the next action step will be eliminating them. The proposed scheme implements a unique artefact elimination process: i) an explicit function $f_6(x)$ is constructed to obtain candidate solution $c_{sol}$. The variable $c_{sol}$ will represent an alternate solution from different traffic data considering the specific traffic attribute where the artefact has been witnessed, ii) the function $f_6(x)$ performs a correlation assessment between the contextual value of other fields with the field where the artefact is present (Line-4). At the same time, this operation will lead to the generation of multiple correlation values; iii) the correlation value found to be maximum is substituted in the place of artefact dart, leading to the generation of pure data $d_p$ (Line-5).

To offer a clear idea about the proposed identification and elimination of artefacts, a pictorial representation of Fig. 5 further illustrates this process. Fig. 5 showcases that there are series of $m$ number of streamed data concerning analyzed data, i.e., $sdj_1(k1), sdj_2(k2), \ldots, sdj_m(km)$. Although the proposed scheme uses nine explicit traffic attributes (as shown in Table II), this explanation considers five explicit traffic attributes (T1, T2, T3, T4, and T5) to fit the information in pictorial representation. Also, assume that user (node) information present in each stream of data is $U_1$, $U_2$, $\ldots$, etc. The corresponding values of each user concerning traffic attributes are represented as $v_{1,1}$, $v_{1,2}$, $\ldots$. Consider that the proposed algorithm, till Line-3 using function $f_6(x)$, found the presence of artefact as $d_{ai} = v_{2,3}$ that belongs to second user $U_2$ in streamed data. In such case, the proposed algorithm obtains candidate correlation for all the user rows to obtain $c_{sol1}$, $c_{sol2}$, $\ldots$, $c_{sol99}$, $c_{sol100}$. The algorithm further applies function $f_6(x)$, which upon execution, found multiple possibilities based on correlation analysis. The outcome shows two higher values say $v_{7,3}$ and $v_{97,3}$ belonging to 7th and 97th user respectively. It will mean that the possibility of a better proximal solution in the cell for artefact $v_{2,3}$ resides for either $v_{7,3}$ or $v_{97,3}$. The algorithm compares $v_{7,3}$ and $v_{97,3}$ to find that the higher value is $v_{97,3}$, which is then substituted in the specific matrix cell for $s_u(k_1)$. That means the older value of artefact $v_{2,3}$ is now eliminated and substituted with the new value of $v_{97,3}$.

![Fig. 3. Decentralized data fusion and quality improvement process.](image1)

![Fig. 4. Mechanism of stream allocation.](image2)
The novelty of this algorithm is as follows:

- The algorithm can identify the artefacts in more extensive indexed data in simplified steps.
- A non-iterative correlation-based assessment is carried out to obtain the proximal candidate solution, which replaces the artefact value in a specific cell only within the matrix.
- The algorithm is designed to work on artefact identification on massive traffic data.

C. Predictive Model for Traffic Management

The prior module of implementation assists in offering the highest possible data purity using correlation-based analysis statistically. Pure data (d_p) refers to a matrix formation of complete information. Each matrix cell is filled with the legitimate or authorized format of individual data complying with the respective traffic attribute. However, on the verge of seamless incoming of stream traffic data, there is still a possibility of accuracy in substituted value in the prior module. Hence, a predictive scheme is introduced in a proposed scheme to assess the degree of accuracy in the obtained data. There are two sets of operations performed in the proposed predictive model viz. i) to identify the degree of accuracy of the newly substituted value based on the learning process, and ii) in case of identification of faulty accuracy score, the predictive model assists in obtaining the actual value with higher accuracy. The operation of the predictive algorithm is as follows:

**Algorithm for Predictive Traffic Management**

**Input:** d_p, Th  
**Output:** i_sol

**Start**
1. **For** i=1:d_p  
2. i_sol = f(i)  
3. **If** i_sol ≥ Th

4. **flag** i_sol as true  
5. **Else**
6. i_sol = f(i, Th_tar)  
7. **For** i_sol ≥ Th_tar
8. **flag** i_sol as true  
9. d_p(d_p) = i_sol  
10. **End**  
11. **End**

The discussion of those mentioned above predictive algorithmic steps are as follows: The algorithm takes the input of d_p (pure data) and Th (threshold) that, after processing, yields an outcome of i_sol (predicted value). The algorithm considers all the outcomes of the prior module of implementation, i.e., pure data d_p (Line-1), where it subjects all the data to various machine learning approaches. A function f(x) is constructed where multiple machine learning approaches are applied to the input data i to obtain an intermediate solution i_sol (Line-2). The prime reason behind applying multiple machine learning approaches is to assess the best-fit model toward optimal accuracy. The algorithm then compares the obtained value of i_sol with a primary accuracy threshold Th (Line-3). The optimal anticipated accuracy of i_sol should be more than Th, while the system flags the assessed value of i_sol > Th as true accuracy (Line-4). Otherwise (Line-5), the algorithm re-forms the implication of function f(x) to the obtained value of i_sol as the second iteration (Line-6). A new target threshold Th_tar is set, which is more than the primary threshold Th. The iteration is continued until the objective function towards converging to a new threshold Th_tar is met. For optimal solution, the algorithm re-checks the newly obtained value, i.e., i_sol is more than Th_tar while the truth condition (Line-7) confirms higher accuracy (Line-8). The obtained value of i_sol is now substituted in the cell recorded with artefact data in the d_p matrix (Line-9). It should be noted that the primary threshold Th is fixed by a user based on application/service demand. At the same time, the user can set the value of Th_tar by adjusting the prior Th value to a slightly higher level. A closer look into the pictorial representation in Fig. 6 of this algorithm will further illustrate the proposed predictive traffic management mechanism.
According to Fig. 6, the algorithm considers m number of pure data values, i.e., $d_{p1}, d_{p2}, \ldots, d_{pm}$, subjected to the machine learning model. The respective obtained solution $i_{sml1}, i_{sml2}, \ldots, i_{smlm}$ are compared concerning primary threshold Th to find out that one of the data, i.e., $d_{p1}$ is found to under-performed (which means its accuracy is lower than cut-off), while the rest other values (i.e., $d_{p2}, d_{p3}, \ldots$ are found to be optimal, i.e., more than $Th$). Hence, the algorithm is explicitly iterated for $d_{p1}$ with a fine-tuned value of target threshold $Th_{tar}$ ($Th_{tar} > Th$) that finally leads to the new predictive outcome of $d_{p1}$. This completes the overall algorithm implementation. The outcome of this model from the viewpoint of the application or service of ITS can be stated as a value-added analytical result, which could assist in more profound insights into the traffic scenario with higher reliability and accuracy.

The novelty of the proposed predictive algorithm is as follows:

- A simplified predictive model doesn't demand any form of the reengineering process towards the ITS framework.
- The accuracy outcome of the model can eventually be fine-tuned by the severity of the application or services demanded in ITS.
- The proposed machine learning algorithm can perform more progressive operations and requires less iteration to obtain optimal accuracy.

A closer look into the entire algorithm implementation shows that the proposed scheme offers a novel and sophisticated learning scheme for traffic management. The following section discusses the results accomplished in the study.

VI. RESULT ANALYSIS

This section discusses the results obtained by implementing the algorithms discussed in prior sections. The discussion is carried out concerning the assessment environment, and the result is accomplished with more highlights on the result discussion and learning outcome.

A. Strategies for Result Analysis

Before initiating the proposed scheme, 25 records were chosen for a pilot study from the primary dataset as a smaller sample size. The idea was to assess the appropriateness of the proposed algorithm towards data analysis concerning accuracy. With the 25 records, 93.1-98.2% of accuracy has been obtained for almost all the learning approaches. This accomplishment offers a concrete proof-of-concept which is then subjected to the original size of the dataset for further assessment. From the data elicitation process perspective, the proposed scheme develops multiple data nodes responsible for streaming the data to the distributed cloud interface. The proposed scheme develops five distributed ITS data nodes (which can be mapped with a gateway node that keeps track of all traffic data to be disseminated). This information is forwarded as a stream to a standard cloud interface, where further data aggregation, processing, and analysis are carried out. The proposed scheme uses a supervised learning approach where the data points with significant predictive errors can be identified and solved. The implementation environment consists of multiple ITS traffic system data nodes that generate the traffic data and forward it to the core cloud interface. This standard interface is used to aggregate the data, identify with the elimination of the artefacts, perform normalization, and apply semantic and syntactical approaches to discover knowledge. Further multiple supervised learning approach is used for the predictive analysis of ITS traffic data. From the perspective of the balanced dataset, the proposed scheme splits the complete data into ten sets, where each set bears 100 records, and one set is allocated to a single stream. This facilitates effective monitoring of network performance parameters like overhead and delays while attempting to perform data transmission. Further, from the perspective of the learning approach, 70% of the data has been considered for training, while 30% of residual data is considered for testing. Further discussion of the assessment environment follows next.

B. Assessment Environment

A computational framework is constructed in MATLAB to assess the proposed scheme, considering a regular 64-bit Windows machine with i5 processing capability. The raw data from traffic is generated as a stream from multiple sources of traffic environment, which are further subjected to the first algorithm towards accomplishing preliminary knowledge. This output is further subjected to a second algorithm where the artefacts are identified, followed by a statistical correlation-based approach to substitute the specific cell of an artefact with new values to achieve data purity. Further, a set of machine learning algorithms are applied to assess the correctness of obtained outcome of the second algorithm using dual thresholding methods. For simplification in evaluation, the primary threshold ($Th$) for accuracy is maintained at 0.5, while the secondary threshold ($Th_{tar}$) is maintained at 0.7. The thresholding values can constantly be amended based on service or application severity towards analytical operations. Following is further information on the assessment environment:

1) Dataset: The implementation of the complete work is carried out by constructing a synthetic dataset with nine traffic attributes (e.g., Vehicle ID, Vehicle Type, Sensor Type, Date, Location ID, Sensor Status, Sensor Range, Driver Name, Driving Experience) and their corresponding values. The construction of the dataset is carried out by adhering to the standard format of big data [38]. At the same time, the inclusion of traffic attributes is formulated based on an existing publicly available dataset of ITS [36][37]. A total of 1000 datasets is acquired; each dataset further consists of 100 records of an individual traffic environment. The datasets are maintained in plain-text form, which is further given as input to MATLAB script for further algorithmic operation.

2) Performance Parameters: The performance parameters considered in the proposed assessment are accuracy, communication overhead, delay, and processing time.

3) Comparison with Existing Scheme: The proposed scheme has used multiple machine learning schemes that are
reported to be frequently adopted in existing schemes, e.g., Random Forest, Artificial Neural Network, Support Vector Machine, Logistic Regression, and Naïve Bayes. The comparative analysis is carried out with each other adopted machine learning model to investigate the best-fit model towards performing an analytical operation in ITS.

C. Result Accomplished

The numerical outcome of the proposed scheme concerning various learning-based analytical approaches is showcased in Table III with respect to various performance metrics adopted for assessment.

<table>
<thead>
<tr>
<th>Techniques</th>
<th>Accuracy</th>
</tr>
</thead>
<tbody>
<tr>
<td>Random Forest (RF)</td>
<td>99</td>
</tr>
<tr>
<td>Artificial Neural Network (ANN)</td>
<td>91</td>
</tr>
<tr>
<td>Support Vector Machine (SVM)</td>
<td>86</td>
</tr>
<tr>
<td>Logistic Regression (LR)</td>
<td>67</td>
</tr>
<tr>
<td>Naïve Bayes</td>
<td>70</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Techniques</th>
<th>Overhead (ms)</th>
</tr>
</thead>
<tbody>
<tr>
<td>Random Forest (RF)</td>
<td>1.11</td>
</tr>
<tr>
<td>Artificial Neural Network (ANN)</td>
<td>2.21</td>
</tr>
<tr>
<td>Support Vector Machine (SVM)</td>
<td>2.31</td>
</tr>
<tr>
<td>Logistic Regression (LR)</td>
<td>2.37</td>
</tr>
<tr>
<td>Naïve Bayes</td>
<td>2.22</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Techniques</th>
<th>Delay (s)</th>
</tr>
</thead>
<tbody>
<tr>
<td>Random Forest (RF)</td>
<td>0.05</td>
</tr>
<tr>
<td>Artificial Neural Network (ANN)</td>
<td>0.31</td>
</tr>
<tr>
<td>Support Vector Machine (SVM)</td>
<td>0.12</td>
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<td>Naïve Bayes</td>
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</table>

The inference of the result exhibited in Fig. 7 is as follows:

1) Discussion of Results: The outcome showcases that the RF model performs better than others on the accuracy scale. The next exhibit of better performance is from ANN and SVM models, although the SVM model has slightly less accuracy than ANN. LR and NB's accuracy trend is nearly the same, with no significant difference. The prime justification is that the LR method exhibits less reliability when exposed to a continuous data stream and hence witnesses degraded accuracy. At the same time, the assumption of independent predictors in the NB model doesn't map well with the proposed study. Although SVM offers better ranges of classification for heterogeneous traffic data, its accuracy starts to decline when the size of the data increases. In this perspective, ANN performs better than SVM, which performs iterative operations to acquire a higher accuracy state. On the other hand, the RF model can execute both regression and classification, potentially contributing to higher accuracy.

2) Learning Outcome: Based on the result, it can be stated that the RF model is highly suited when exposed to significant streams of ITS applications/services that demand higher accuracy in the prioritized traffic environment. The applicability of ANN and SVM is suited for low-medium scale traffic, which also demands a medium-high range of accuracy. On the other hand, the applicability of the LR and NB model is best suited for performing an analytical operation that doesn't demand instantaneous response delivery or doesn't need higher accuracy.

E. Analysis of Communication Overhead

Communication overhead is computed by evaluating cumulative packets destined to be forwarded from one vehicle to another vehicular node in ITS. A sample of 2500 test packet bytes is used to assess this performance metric. An ITS with an optimal design plan should always keep the communication overhead as low as possible to resist a communication bottleneck situation. The outcome of communication overhead is shown in Fig. 8.
The inference of the result exhibited in Fig. 8 is as follows:

1) **Discussion of results:** The outcome in Fig. 8 shows two distinct trends viz. i) lower communication overhead shown by the RF model and ii) higher communication shown by the ANN, SVM, LR, and NB models. Although there is a numerical difference in the outcome among ANN, SVM, LR, and NB models (as shown in Table III), the difference is less significant. The prime justification behind this is as follows: A closer look into ANN, SVM, LR, and NB models shows that they perform quite an iterative analytical process while performing both training and validation operations (especially the training). This fact leads to a more significant communication overhead when exposed to many data packets. It is to be noted that data packets and their sizes are derived from the existing synthetic dataset itself. However, RF models offer a comparatively less iterative process and a more streamlined operation for continuous incoming packets, resulting in less communication overhead.

2) **Learning outcome:** Based on the outcome, the adoption of RF is more suitable in ITS when it calls for performing analytical operations in dense traffic conditions, whereas other machine learning models are applicable only in lesser dense traffic environments. The reliability of RF processing and analyzing data is relatively higher than others.

**F. Analysis of Delay**

Owing to the decentralized scheme in the proposed system, it can be assumed that all the algorithmic operations are carried out on different terminals in ITS with higher synchronous operation. Hence, delay is a suitable parameter to justify any form of lag of interval in receiving and analyzing data. The proposed system computes delay by evaluating the duration interval for one module's data packet to reach another.

As the proposed architecture jointly implements three different algorithms, it is anticipated to exhibit a delayed trend as lower as possible. The inference of the result exhibited in Fig. 9 is as follows:

1) **Discussion of results:** A unique observation is noted in Fig. 9 concerning delay, which is entirely different from a prior comparison of accuracy and communication overhead. Fig. 9 showcases that RF is the performing model, while the next performing model is SVM and LR. On the other hand, the performance of ANN and NB is shown to consume more delay. The specific reason behind this outcome is mainly associated with increasing training operations in ANN to meet the anticipated accuracy (Th\textsubscript{req}). Although NB performs slightly better than ANN, it cannot feature learning from the associated relationship of traffic attributes. From this context, LR and SVM perform better as both can deal with high-dimensional spaces between the data; however, SVM doesn't excel well compared to LR as it demands increasing time for training.

2) **Learning outcome:** From the outcome perspective, it can be stated that prioritized applications/services in ITS are well-suited when executed with the RF model. In contrast, the application/services of ITS only work well with ANN and SVM if the sample size is reduced. However, it is not practically possible in the genuine environment of ITS.

**G. Analysis of Processing Time**

Processing time is computed as the time required for the complete algorithm to execute jointly. An effective architecture deployment always demands lower processing time. The outcome of processing time is shown in Fig. 10.

The inference of the result exhibited in Fig. 10 is as follows:

1) **Discussion of results:** The outcome in Fig. 10 showcases RF to offer reduced processing time compared to other machine learning approaches. A similar justification discussed for another performance metric can be attributed to stating the reason behind this outcome. Lower processing time also refers to lower time complexity stating that the RF model offers a computationally cost-effective analytical process in ITS compared to others.
2) Learning outcome: As the ITS environment included the usage of multiple resource-constraint devices, it is anticipated that algorithmic operation should not be computationally complex. Hence, the RF model offers better predictive operation on low resource-based devices in ITS, while others are witnessed with higher computational complexity.

Therefore, from the perspective of accomplished outcome, it can be stated that the proposed analytical model is well suited with RF to exhibit a best-fit machine learning model in ITS. However, to understand the efficiency of the proposed scheme apart from standard learning approaches, the proposed scheme also analyzes comparison with existing state-of-art methods, as exhibited in Table IV.

Table IV highlights the comparison with the state-of-the-art methods presented in Section II. The prime headers used in Table IV are A₁: Decentralization, A₂: Accuracy, A₃: Timeliness, A₄: Complexity. From the outcome shown in Table IV, the following inference of novelty is drawn:

- The entire modelling of the proposed scheme is carried out to support the scheme’s decentralization applicability, which is not exhibited by either of the existing methods, irrespective of their accomplished outcomes. This is a significant novelty of the proposed scheme, which leads to the practical ground of implementation in urban traffic systems.
- Without using any sophisticated or iterative learning principle, the proposed scheme offers higher accuracy than existing methods without compromising the other performance attributes shown in Table IV.
- The proposed algorithm has reduced dependencies of iterative computation while it is more progressive, providing a speedier algorithm processing time. This is a significant accomplishment of novelty attributes of the proposed scheme compared to existing schemes where complex approaches are used.
- The proposed approach also offers very low computational complexity, contributing to other novel features. The prime reason behind this is the formulation of algorithms emphasizing optimal data quality in predictive operation. Further, the adoption of shared memory makes it a more lightweight operation.

VII. CONCLUSION

A closer look into the proposed experimental analysis showcases that the proposed scheme can consider the input of multiple streams of ITS traffic data via various data nodes. This is a practical scenario where a gateway node can collect the traffic data and subject it to analysis. Further, the information is subjected to various rounds of processing, right from transformation to the final stage of knowledge discovery. All these individual blocks of operation can be carried out in distributed form. Yet, owing to the inclusion of a unique indexing mechanism, all the data are closely synced with each other. Further, the benchmarking over the accomplished outcome clearly states the distinguishable performance for each supervised machine learning scheme. This shows that the proposed scheme can act as a robust and highly flexible evaluation platform for analyzing the ITS traffic data with cost-effective measures without affecting data transmission performance. The proposed scheme presents a novel architecture capable of performing an efficient analytical operation over complex traffic data in ITS. The contribution of the proposed study is as follows:

1) The proposed model can transform the raw and complex stream of traffic data into highly structured data rendering it with higher suitability for analytical processing.
2) The proposed model offers the implementation of the highly decentralized scheme of the analytical process.
considering streams of multiple traffic data from different origins followed by a unique data fusion.

3) The proposed model can identify the position of artefacts in massive data sizes followed by eliminating the artefacts using cost-effective statistical correlation-based analysis.

4) The proposed scheme also introduces a joint test-bed benchmarked with multiple machine learning models to show the best-fit model towards predictive analytical operations in the ITS environment.

5) Quantification of outcome states that the proposed model, when executed with the RF model, shows improvement in accuracy by 20%, reduction in communication overhead by 14%, minimization of delay by 13%, and diminished processing time by 7% in comparison to other machine learning approaches (e.g., ANN, SVM, LR, and NB model).

Future work will be carried out toward further optimizing the learning model by including network-based parameters involved in the assessment. An assessment model can be constructed to investigate the impact of different network types, storage types, and communication standards on analytical operations. Further, work can be extended to understand the adoption of different data formats toward predictive accuracy.

REFERENCES


Comparison of Multi-layer Perceptron and Support Vector Machine Methods on Rainfall Data with Optimal Parameter Tuning

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Abstract—This study describes the search for optimal hyperparameter values in rainfall data in 49 cities in Australia, consisting of 145,460 records with 22 features. The process eliminates missed values and selects 16 numeric type features as input features and one feature (Rain Tomorrow) as output feature. It is processed using the Multi-Layer Perceptron (MLP) and Support Vector Machine (SVM) methods based on Three Best Accuracy (3BestAcc) and Best Three Nearest Neighbors (3BestNN). The results showed that the SVM kernel linear method gave an average accuracy value of 0.85586 and was better than the MLP method with an accuracy of 0.854.

Keywords—Rainfall; MLP; SVM; optimal

I. INTRODUCTION

Weather is the condition of the air at a certain time and place. Weather conditions are related to sunlight, air temperature, humidity, wind, and other conditions and play a significant role in various areas of life, such as changes in the quantity and quality of water, changes in forest habitat and agricultural land, and other changes. The weather for a short time must use the values of the weather elements that exist at that time. Meanwhile, weather statements in a more extended unit of time must use the values of weather elements with the lowest, highest, or most felt levels by the five senses as in [1].

Rain is one of the parameters and weather phenomena that comes from the evaporation of air containing water vapor and forms clouds at certain temperatures. Rain can be measured as a parameter, and rain can be seen visually as a phenomenon, such as fog, smoke, and others. Rainfall data information is very important for planning, especially for water structures such as irrigation, dams, urban drainage, ports, and docks, so data analysis is needed to predict it. In [2], rainfall is the amount of water that falls on a flat ground surface during a certain period which is measured in millimeters (mm) above the horizontal surface. Rain can also be interpreted as the height of rainwater that collects in a flat place, does not evaporate, does not seep, and does not flow.

One of the solutions for predicting rainfall is to analyze big data from the information obtained using the correct method to get the best accuracy. Big Data is a collection of data with super large data volumes and has a high diversity of data sources, so it needs to be managed with methods and assistive devices whose performance is appropriate as in [3]. However, the implementation of big data is still not good or requires a relatively long time to obtain the desired accuracy value, so it is necessary to reduce the feature dimensions by utilizing the Linear Discriminate Analysis and Principle Component Analysis methods to optimize the computational process.

This study will explain predictions of rain status based on a collection data set over a 10-year period that describes meteorological information from 49 different cities in Australia. Furthermore, the data is processed by eliminating missed values and selecting 16 numeric type features as input features and one feature (Rain Tomorrow) as output features and analyzed using the Support Vector Machine (SVM) and Multi-Layer Perceptron (MLP) methods. Then to get optimal results (maximum accuracy), an algorithm was developed to obtain hyperplane parameters using K-Fold cross-validation and developed through two schemes. The results of this study are the level of prediction accuracy in each city, which can be used as a supporting medium in urban planning and anticipation. In addition, the overall average accuracy can also be used as material for consideration in further research.

II. PREVIOUS RESEARCH STUDY

Many methods have been used to predict rainfall data, including the Extreme Learning Machine Method on Artificial Neural Networks to predict rainfall in the Poncokusuma area of Malang district, using data from 2002 to 2015. The results of this study obtained a MAPE of 3.6852% with 4 features, 2 hidden layers, and the proportion of training data is 80% as in [4]. Research on clustering rainfall predictions in Australia using the K-Means algorithm in the WEKA and RStudio applications. The results obtained are the number of cluster 2 with an SSE of 28.0%, which is an ideal cluster to predict rainfall in Australia as in [5][6]. Research on rainfall prediction in Australia using Machine Learning by comparing the level of accuracy using several methods. The methods used are K-Nearest Neighbors, Decision Trees, Random Forests and Neural Networks. The results obtained by the best method are from Neural Networks with an accuracy rate of 85.5% as in [7].

- Multiple Face Detection Research using Hybrid Features with SVM Classifier: The data used is from two databases, namely the BAO database and the CMU
Research on hyperplane-parameter tuning for SVM, namely Hyper-parameter Tuning for Support Vector Machines with Distribution Algorithm Estimation, has been carried out using various methods, including Genetic Algorithms, Particle Swarm Optimization (PSO), Grid Search, and Random Search. This study will estimate hyperplane-parameter estimation using the Univariate Marginal Distribution Algorithm (UMDA) and Boltzmann Univariate Marginal Distribution Algorithm (BUMDA). Both methods are stochastic optimization techniques by building and taking samples from probabilistic models. The study's results obtained the best method using the Boltzmann Univariate Marginal Distribution Algorithm (BUMDA) as in [11].

Parameter optimization in SVM using the Taguchi Method for High Dimensional Data. In this study, optimal parameters were selected using the Taguchi method, which provides an increased level of accuracy compared to the Grid Search method as in [12]. Support Vector Machine (SVM) for Rainfall Forecasting on the Johor River, namely forecasting rainfall as a warning in the event of heavy rainfall resulting in flash floods. The data for the 60-year study period came from rainfall data in the Malaya Gum Field, Kota Tinggi, Johor, Malaysia. The method used is SVM. The research results on the SVM method with the Radial Basis Function RBF kernel has the best performance compared to sigmoid, linear, and polynomial. The Root Mean Square Error (RMSE) value for the RBF 67.70 kernel is the best compared to other kernels as in [13]. Research on rainfall forecasting using the Support Vector Machine (SVM) has been conducted. The method used to predict rainfall is SVM regression. The research data used is rainfall data in Khurda District Orissa. The kernel that has the minimum Mean Square Error (MSE) is the Linear kernel which produces a minimum MSE average of 15.04% as in [14].

Sustainability plays an important role in enhancing the industry's competitive advantage. Continuous performance and application assessments face high dimensional data, robustness and imprecision. Machine learning is pressured to implement. This study aims to design a machine learning model to assess sustainability performance using SVM. Hyperplane-parameter tuning and k-fold validation are included for improved performance in SVM models. The research object was carried out in the bioenergy industry. The validation was carried out ten times. Tuning the hyperplane parameters gets 98.32% in testing the data. The final results show that SVM with a polynomial kernel model can classify sustainability performance accurately as in [15]. Problems related to hyperparameter tuning are still an interesting topic to be studied. Some of them are aspect-based sentiment analysis of iPhone users on Twitter using the SVM and GridSearch methods for hyperparameters C, gamma, and kernel as in [16], hyperparameter tuning of supervised learning algorithms for classification of families receiving rice food assistance using the grid search method, random search, and Bayesian optimization as in [17], and indoor pollutant classification modeling using relevant sensors under thermodynamic conditions with multilayer perceptron hyperparameter tuning using the GridSearch method as in [18].

III. MATERIAL AND METHODS

A. Materials

The data used in this study is secondary data related to daily rainfall in 49 different cities in Australia over a 10-year period Australia. The dataset was obtained from the kaggle.com platform accessed February 2, 2023. The sample data is 145,460 which consists of 22 (twenty-two) features.

They are the location of weather stations, minimum temperature (C), maximum temperature (C), recorded rainfall amount (mm), evaporation in 24 hours, number (hours) of bright sunlight in 24 hours, the direction of strongest wind gust in 24 hours, strongest wind speed (km/h) in 24 hours, wind direction at 09.00 local time, wind direction at 15.00 local time, wind speed at 09.00 local times, wind speed at 15.00 local time, humidity (per cent) at 09.00 local times, humidity (per cent) at 15.00 local time, atmospheric pressure (hpa) which decreases to an average sea level at 09.00 local times, reduced atmospheric pressure (hpa) to mean sea level at 15.00 local time, part of the sky covered with clouds at 09.00 local times. The unit size used is octas (a unit of eight), which records the number of clouds, the part of the sky covered by clouds, at 15.00 local time. The unit size used is octas (a unit of eight), which records the number of clouds, temperature (C) measured at 09.00 local time, temperature (C) measured at 15.00 local time, today’s status (rain, notated 1 or not raining, given the notation -1), tomorrow’s status (rain, given the notation 1 or not raining, given the notation -1).

B. Methods

Based on the data obtained, the steps for preprocessing and analyzing the data are compiled as follows:

- Collect a data set over a 10-year period that describes meteorological information from 49 different cities in Australia and consists of 145,460 records with 22 features.
- At the preprocessing stage, the process carried out is the elimination of miss values and the selection of 16 features of a numeric type as input features and one feature (RainTomorrow) as an output feature. In the Python program, to delete all data miss values from a feature (e.g., the MinTemp feature), suppose df is a logical file containing the results of reading all data, with the statement: you can use the statement:

  \[
  df = df.loc[df['MinTemp'].isna()] == False
  \]

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then the new df contains data with no missed values in the MinTemp feature. This process is carried out on all features.

- Using the Support Vector Machine method to determine the value of C that gives the best average accuracy, and the Multi-Layer Perceptron method to determine the hidden layer structure.

- Based on the data records that have been cleaned (data cleaning), the data will be divided using the k-cross validation principle into one part of the data (training and data validation). In contrast, the rest will be used as test data. The training and validation data will be divided into folds, fold1, fold2, fold3, fold4, and fold5, each containing random and different data (not intersecting) with the same number of records.

- Create a tuning method algorithm for Linear Kernel SVM and MLP with the principle of the three best accuracy (3BestAcc) to find the 3 best accuracy values in a discrete hidden structure and three best nearest neighbor (3BestNN) to find the 3 best accuracy values in an interval.

- Displaying the Contour of the SVM method by carrying out the Principle Component Analysis (PCA) transformation so that two features are obtained, namely principal component 1 and principal component 2, which are used as abscissa and ordinate on a two-dimensional graph (cartesian graph). Based on the results of an experiment on 100 data, it was found that the contour hyperplane is a dividing line between the status of the two classes shown in Fig. 1.

- Output analysis is carried out after obtaining histogram graphs, accuracy, and Mean of Square Error (MSE) to obtain interpretations and conclusions from the results obtained.

IV. RESULTS

A. Research Data

The data retrieved from kaggle.com is divided into three parts. A total of 50,000 records were used as data (training and validation), and 6,420 records were used as test data. Of the 50,000 records used as training data and validation data with a ratio of 80:20. With the K-Cross Validation method, the data is divided into five folds of the same size, which are randomly generated. Each fold contains 10,000 data records. One of the five folds is used as validation data, and the other four are used as training data. Of the 56,420 datasets, there are two classes: Rain Status with 12,427 records and no rain status with 43,993 records.

B. Support Vector Machine

The algorithm implemented in this study uses the Python SVM function in scikitLearn by Belete et al. as in [19]. The selected parameters are:

- kernel='linear',
- C=kC(input from user)

The default parameters of the system:

(*, C: float = 1, kernel: str = "rbf", degree: int = 3, gamma: str = "scale", coef0: float = 0, shrinking: bool = True, probability: bool = False, tol: float = 0.001, cache_size: int = 200, class_weight: Any | None = None, verbose: bool = False, max_iter: int = -1, decision_function_shape: str = "ovr", break_ties: bool = False, random_state: Any | None = None) -> None.

In obtaining the accuracy value of a certain linear C kernel SVM model, one fold is used as validation data and the other four folds as training data. This process was repeated five times to get average accuracy. This study used C ∈ (0.16), and the accuracy values were calculated at various initial intervals, as shown in Fig. 3. The C value with the highest accuracy is
0.08272, with an accuracy of 0.8538. This value is obtained from one of the folding compositions. If the composition of the wrinkles is different, it is possible to get other Cs, but the accuracy values are almost the same. The accuracy value for data testing is 0.85586; this average value is higher than the accuracy value for obtaining optimal parameters. This value is increasing by 1.586% compared to the results as in [20].

The hyperparameter optimization (HPO) optimization steps for the MLP method use the three highest accuracy values or Three Best Accuracy (3BestAcc) starting from the initialization of the data and functions to be used. Then from the 16 inputted numeric data, three nodes with the highest accuracy will be selected (best1, best2, best3) with 6 Hidden layer structures consisting of (best1, i), (i, best1), (best2, i), (i, best2), (best3, i), (i, best3), (i=1,2,3,...,30). Next, three structures with the highest accuracy were taken for each structure so that 18 arrangements were obtained.

The algorithm implemented in this study uses the Python MLPClassifier function in scikitLearn as in [21]. The modified parameters are:

- Hidden_layer_sizes contains the number of hidden layers and the number of nodes of each Hidden Layer
- random_state=1
- solver = "adam"
- learning_rate_init = 0.001
- max_iter=300
- toll = 0.000001

for other parameters, the default of the system.

Calculations are made from hidden layer 1 to hidden layer 12 because the accuracy value tends to decrease starting from hidden layer 9, as shown in Fig. 4, so the process is stopped at hidden layer 12.

Hyperparameter tuning is performed on the Multi-Layer Perceptron method on the parameters of the number of hidden layers and nodes. The process of getting one model accuracy value (number of hidden layers and number of nodes) uses one fold as validation data and the other four folds as training data. This process is repeated five times to obtain the average accuracy. The hidden layer structure that provides the highest accuracy is (19, 24, 25, 25, 15, 3, 3, 26) so it is obtained that the optimal MLP model has eight hidden layers with the number of nodes in the hidden layer 1 being 19, the number of nodes in hidden layer 2 is 24, and so on. This structure is obtained from one of the fold compositions. However, if the fold composition is different, it is possible to obtain a different Hidden Layer structure arrangement but still has the highest accuracy in the fold arrangement. The accuracy value for testing data is 0.85586; this average value is higher than the accuracy value for obtaining optimal parameters, which is 0.854. This value is 1.4% greater than the results as in [20].
Table I shows the average accuracy value for each station of the SVM methods and MLP methods. The accuracy information for each station can be used to determine which method to use for maximum accuracy. SVM kernel method C=0.08272 has an accuracy of 0.85586 was obtained, while the MLP method with a hidden layer structure (19, 24, 25, 15, 3, 3, 26), an accuracy of 0.854 was obtained. Thus it can be concluded that the SVM kernel linear method is better than the MLP method with a difference of 0.00186.

D. Data Analysis for each Region

Processing rainfall data at each station helps predict conditions where the results obtained are compared to obtain results with the best accuracy. Fig. 3 and Fig. 4 shows several stations whose station test data accuracy values are higher than the average accuracy of the SVM and MLP methods. In addition, Table II shows that the SVM method has a higher accuracy value than the MLP method, where the data for the "Rain" status is less than the data for the "No Rain" status. Therefore, the amount of testing data on the "Not Raining" status is used as much as 1.5 times of the "Raining" status data. An example is the data at the Darwin station, which consists of 789 data records for the "Rain" status. The number of testing data is 1.5 x 789 = 1183 records. The total number of records is 3062, so the total training data is (3062-1183) = 1879. Then the 1879 records are divided into five folds with the same number of records, each containing (1879/5) = 376 records.

TABLE II. STATION WITH AN ACCURACY OF MORE THAN 0.85586

<table>
<thead>
<tr>
<th>No</th>
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V. CONCLUSION

This research results in the MLP method hyperparameter tuning with the 3BestAcc method. The 3BestAcc method can be used to find the structure of the Hidden layer Multi-Layer Perceptron which gives optimal results. The working principle of 3BestAcc is to select three structures in the hidden layer with the highest accuracy, and then these three structures are used to determine the structure of the next hidden layer. Further development of the 3BestAcc method still requires a lot of time, so algorithms can be developed to make the selection of the hidden layer arrangement more time efficient. Another research that might be developed is to look for the highest accuracy value at specific intervals. Further research is also possible on the dataset. Because the data records are extensive, the preprocessing stage can select representative data, reduce feature dimensions, or select the most influential features so that the MLP model processes fewer data.

Table II provides information that the SVM kernel linear method (C=0.08272) provides a better average accuracy value than the MLP method with a hidden layer structure (19, 24, 25, 15, 3, 3, 26).

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REFERENCES


A Vehicle Classification System for Intelligent Transport System using Machine Learning in Constrained Environment

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Abstract—Vehicle type classification has an extensive variety of applications which include intelligent parking systems, traffic flow-statistics, toll collecting system, vehicle access control, congestion management, security system and many more. These applications are designed for reliable and secure transportation. Vehicle classification is one of the major challenges of these applications particularly in a constrained environment. The constrained environment in the real world put a limit on data quality due to noise, poor lighting condition, low resolution images and bad weather conditions. In this research, we build a more practical and more robust vehicle type classification system for real world constrained environment with promising results and got a validation accuracy of 90.85 and a testing accuracy of 87%. To this end, we design a framework of vehicle type classification from vehicle images by using machine learning. We investigate the deep learning method Convolutional neural network (CNN), a specific type of neural networks. CNN is biologically inspired with multi-layer feed forward neural networks. It can learn automatically at several stages of invariant features for the particular chore. For evaluation, we also compared the performance of our model with the performance of other machine learning algorithms like Naive Bayes, SVM and Decision Trees.

Keywords—Vehicle classification; intelligent transport system; deep learning; machine learning; CNN; digital image processing

I. INTRODUCTION

Vehicle classification and detection has emerged as an integral part of the Intelligent Transportation System (ITS). The aim of ITS is to provide public safety, improve travel and transit information, minimize blocking, produce cost funds to tragedies operators. To cope with the rising demand of surface transportation system, the ITS technologies assist cities, states and towns nationwide. The effectiveness of an IT system is generally based on the performance of vehicle detection and classification technology. It collects all or part of the information which is used in an efficient ITS. In the development of ITS, traffic monitoring is one of the significant applications encompassing the categorization and detection of vehicles. Vehicle type classification has an extensive variety of applications which include intelligent parking systems, traffic flow statistics, toll collecting system, vehicle access control, congestion management, security system and many more. These applications are designed for reliable and secure transportation. Vehicle classification is one of the major challenges of these applications. Many researchers from different domains have been involved in this area to overwhelm the primary transportation issues.

In recent past, the research in vehicle type classification and detection from static images has achieved huge attention because of its essential role in an enormous variety of applications [1], [2]. Vehicle type classification and detection from still images is a demanding job. Because of their great intraclass distinctions, numerous vehicle types fitting into similar classes have features of several shapes and sizes. Furthermore, shadow, lighting, noise, and obstruction make the classification job more demanding [1]. Moreover, there can be some constraints on image quality while working with limited computing resources. These types of complications can be determined by using efficient and reliable vehicle classification systems for such constrained environment.

Several image-based techniques have been presented until now and the majority of them comes under into two groups: appearance-based methods and model-based methods. To classify vehicle type, appearance-based method uses appearance features (for example SIFT [3], sobel edges [4]) from either vehicle side view or front view image for classifying vehicle type. Model-based methods [5], [6], [7], [8] retrieve the 3D parameter of the vehicles just like height, length, and width for classification. To define the vehicle shape and structure 3D models are used. In 3D models, the main features to deal with are faces, point, lines and topological structures. The vehicle classification is generally performed using a combination of hardware and software-based approaches. The hardware used for this purpose mainly includes radars, magnetic detectors, infrared detectors, ultrasonic detectors, acceleration detectors and many more. But this capable hardware cannot be used without having automated vehicle classification software embedded in their backend. The focus of all the previous works as well as this work is also to develop such a capable software that when embedded in any capable hardware, can provide accurate vehicle classification results. The core idea behind the development of automated vehicle classification software’s is automated intelligence. This also goes the other way around. If a capable software gets developed that can accurately classify
vehicle classes, it will not be able to do anything until it is embedded into a capable hardware to provide real time monitoring, surveillance, and other activities.

Nowadays, many real-world applications like auto controlling traffic are totally dependent on vehicle classification and detection. Several methods have been proposed for vehicle classification. Vehicle classification is very complicated area in computer vision because it requires high accuracy which may lead towards complex algorithm. Until now various works have considered for this classification, some of the work has been done on the limited scenarios. There are some challenges which must overcome for the accomplishment of this task like image quality, size, image scale and color [9], [10]. Moreover, existing research works on vehicle image classification using deep learning have made significant progress in achieving high accuracy and robustness. However, they also exhibit certain limitations that need to be addressed. One limitation is the lack of diverse and comprehensive datasets, which may lead to biased or overfitting models. Additionally, the focus on specific regions or vehicle types in some studies limits the generalizability of the proposed methods. Another limitation is the heavy reliance on manual annotation and labeling, which can be time-consuming, subjective, and prone to errors. Furthermore, the computational complexity and resource requirements of deep learning models pose challenges for real-time applications and resource-constrained environments. The interpretability and explain ability of deep learning models in the context of vehicle image classification also remain areas of concern. Addressing these limitations will contribute to the advancement of vehicle image classification research and enable more accurate and efficient solutions in real-world scenarios.

The main motivation behind our proposed work is that in real-world scenarios, due to bad weather conditions and moving of the camera and/or vehicles, the images are sometimes not clear enough to distinguish between types of vehicles automatically. We believe that the need to address this problem is very important; particularly, in case of lack of high-end cameras that could provide a good quality image. Also, there can also be other factors like more than one vehicle in a single image and various angles of the vehicle when the image was captured. Furthermore, image rotation also presents a challenge for accurate classification of vehicles.

The vehicle images captured by camera are blurry (or not clear) if the shutter speed while capturing the images is not according to the vehicle speed (or stationary vehicle and moving camera). The image clearness is usually measured by its sharpness. A clear image has a high degree of sharpness; while on the other hand, a blurred image is less sharp. In our work we are concerned with motion-blurriness caused by the motion of vehicle and/or camera. So, a robust method would yield accurate results on blurred images as well. Deep learning based CNNs have revolutionized image processing and computer vision. They excel in extracting meaningful features directly from raw data, eliminating the need for manual feature engineering. CNNs leverage their hierarchical and localized approach to capture complex patterns and structures in images. They are adept at tasks like object recognition, image classification, and segmentation. Deep learning models handle large-scale datasets, ensuring scalability and generalization across different domains. With their ability to automatically adapt and learn from image data, deep learning and CNNs have significantly advanced applications such as autonomous driving, medical imaging, and video analysis. In this context, we propose a more practical and robust vehicle classification system with a promising accuracy rate. We recommend a structure of vehicle classification from vehicle images by using a machine learning system. Convolutional neural networks are the machine learning methods. They are specific neural networks. CNN is biologically inspired by that multi-layer feed forward neural network. It is capable to automatically learn several phases of invariant features for the chore. CNN has good ability of learning features in face detection [11], image classification [12], video classification [13], facial point detection [14], and human attribute inference [15]. CNN have verified great success in these kinds of tasks. For large scale image recognition task it has become the most popular architecture. Convolutional neural networks have a layered structure and each layer could learn the demonstration of the input. This capability is called feature learning. Feature learning is a method through which machine learns from raw data. In image classification, the raw data of the image is represented by an array of pixels. These arrays of pixels fed to the CNN and then CNN learns the useful features of the image to resolve the machine learning problem [16]. The major contributions of the work presented in this paper are as follows:

- Development of the algorithm proposed for automatic vehicle type classification in constrained environment.
- Modeling the vehicle classification system as a problem to be solved by using CNN algorithm.
- Performance analysis of the proposed technique using experiments.

The rest of this paper is organized as follows. Section II presents the current relevant techniques with their limitations. A detailed explanation of the proposed technique is presented in Section III. Details about experimental setup and results has been given in Section IV. Future directions have been provided in Section V and finally, this paper has been concluded in Section VI.

II. RELATED WORK

Zhen dong et al. in [17] worked on the vehicle classification by using a semi supervised CNN. They used the front view of vehicle images. As an input, the network is given the vehicle image and then it outputs the vehicle class to which the vehicle belongs. Their technique achieves overall classification accuracy of 96.1% in the daylight images and 89.4% accuracy on the nighttime images. Similarly, in [18], a multi type vehicle classification system is established. For this purpose, they used Bag of Visual Words (BOVWs) and Random Neural Networks (RNNs). Their BOVW-RNN classification system achieves 95.63% accuracy which is higher than the LIVCS who achieve 91.21% accuracy. Another approach proposed in Yiren zhou et al. [19] uses Deep Neural Network (DNN) approach for the vehicle detection and classification problem. Their approach has the capability to be
used for training on restricted size datasets and can be extended to diverse lighting condition cases. However, as opposed to the proposed work these techniques do not consider the images taken under the constraint environment. For vehicle classification, the authors in [20] used an immovable camera to detect and classify the vehicles. They located the camera on the height of the road surface. They used an algorithm which has two phases: In the first stage they obtained mobile vehicles in the traffic location, removed background from the image and performed morphology operations and edge detection. In the second stage, they captured the vehicle through camera and processed the feature extraction method. The classification rate for their proposed method is 90% of the whole data. In [21], Wei Wu et al. used a novel method by using neural networks and parameterized models for vehicle classification. They captured their own real vehicle images through CCD camera installed on toll collecting station. They took the vertices and their topological structure as the main feature. To recognize vehicle, they used a classifier which is based on the multi-layer perception network (MLPN). In this classifier they used a learning technique which is based on the gradient descent for the least exponential function error (LEFE). They achieved approximately 91% accuracy. The authors in [22] proposed the appearance-based model for vehicle type classification. They used front view of the vehicle and SoftMax regression for classification. They used convolutional neural network and by using this network their method automatically learned good features for classification. Their method is relatively effective for the classification of vehicle types. For automatic vehicle classification, the method in [23] is based on the fusion classical neural network (CNN) and of fast neural network (FNN). They used FNN as the main classifier and CNN as a final classifier. Their proposed method produced 95.83% accuracy. However, all these techniques do not account for images taken under the constrained environment. The author addresses the limitations of single vehicle detection methods due to a lack of high-quality labeled training samples. They propose using the YOLO-v5 architecture for vehicle detection and classification, employing transfer learning by fine-tuning pre-trained weights. Extensive datasets collected by the authors, including various traffic patterns, occlusions, and weather conditions, are manually annotated to enhance training. The proposed YOLO-v5 model surpasses traditional methods in terms of accuracy and execution time, demonstrated through simulations on PKU, COCO, and DAWN datasets. The findings highlight the effectiveness of the proposed method in challenging scenarios [24].

The author aims to solve the problem of traffic density estimation for effective traffic management by utilizing deep learning techniques. They collect data from open-source libraries and label vehicles in images into six different classes. To address the imbalanced dataset, data augmentation techniques are applied. The proposed model is based on an ensemble of the Faster R-CNN and Single-shot detector (SSD) models, trained on processed datasets. Experimental results demonstrate that the proposed ensemble outperforms base estimators on FLIR thermal dataset, achieving a higher mean average precision (mAP) of 94%. The ensemble model shows promising results compared to other estimators, with better performance on thermal images. The proposed model also exhibits significant potential for traffic density estimation, offering valuable insights for traffic management [25]. The author aims to develop a method for in-vehicle passenger detection using MIMO radar. They propose a CNN-LSTM model that accurately detects, counts, and classifies passengers in five-seater vehicles. The model combines CNN for feature extraction and LSTM for temporal prediction. Reliable datasets collected from different car models are used to evaluate the model, which achieves high accuracy in detecting unattended infants/children (over 95%), counting passengers and identifying occupied seats (89% accuracy), and classifying passengers (over 74% accuracy). The results demonstrate the generality of the proposed method for deployment in any five-seater vehicle [26]. A vehicle classification system based on low cost triaxle anisotropic magneto resistive sensor was proposed in [27]. To detect the vehicle within a segment and single lane the vehicle signals effectively, a fixed threshold state machine procedure is presented. Through experimentation it is concluded that the detection accuracy of the presented technique can reach up to 99.05% and the overall average classification accuracy is almost 93.66. These results show the efficiency of presented technique for low-speed congested traffic, but authors did not test their technique in the constrained environment. In [28], author presented a vehicle classification problem based on laser scanner profiles, which is found as a part of electronic tolling systems. Vehicle shapes are extracted from laser scans and then explore them as multi-dimensional shapes. The presented technique explores identical and non-identical shapes for discovering typical shapes then classified using global alignment of shapes. This technique shows state of art results by overcoming per class error by 4 to 17%. In military operations, nighttime multi-vehicle detection over a long distance is essential. The visual characteristics of automobiles are hard to identify at night due to inadequate lighting, which leads to numerous missed detections. Based on 43m APD LiDAR intensity images and point cloud data, this work [29] suggests a two-level detection strategy for long-distance nighttime multi-vehicles. There are two layers to the approach. The first level is 2D detection, which raises the brightness of weak and small objects and improves the local contrast of the intensity image. The detection result of more than the threshold is reserved as a dependable item when the confidence threshold is set, while the detection result of less than the threshold is a suspicious object. The suspicious object region from the first level of 3D recognition is translated into the appropriate point cloud classification judgment in the second level, and the object detection score is achieved by thorough judgment. The method outperforms current state-of-the-art detection techniques, according to experimental data, achieving a detection accuracy of 96.38% and effectively enhancing the detection accuracy of multiple cars at night. LiDAR point cloud semantic analysis accuracy is necessary for autonomous cars to interface with the actual world. By fusing 3D global Hough features and 2D local Hough features with a classification deep learning network, this work [30] suggests a hybrid 2D and 3D Hough Net. First, the global Hough features are extracted from the 3D object point clouds by mapping them into the 3D Hough space. For training global features, the 3D convolutional neural network receives the generated global Hough features as input. To limit the
calculation of duplicate points, a multi-scale critical point sampling approach is created to extract crucial points in the 2D images projected from the point clouds. A grid-based dynamic nearest neighbors’ method is created by exploring the important locations’ neighbors to extract local information. To classify objects, completely connected layers that are input into the two networks are then connected to the full connection layer. The experiments show that the classification accuracy achieved 97.6% by allocating the 25 × 25 × 25 cells for the 3D Hough spaces and specifying 0.25 as the unit size of the local Hough spaces. The results of the studies demonstrate that by assigning the 25 25 25 cells for the 3D Hough spaces and choosing 0.25 as the local Hough spaces’ unit size, the classification accuracy was increased to 97.6%.

As per the research carried out [31], numerous automated traffic monitoring systems have been created because of the growing number of cars circulating in various urban cities. Roadside camera traffic monitoring systems are being widely used because they provide crucial technological benefits over other traffic monitoring systems. The performance of the entire system is significantly impacted by the methods used for vehicle identification and traffic congestion categorization, which are the two primary processes for video-based traffic congestion detection systems. Investigate four chosen vehicle recognition techniques in two contexts: urban and highway, including the Gaussian Mixture Model (GMM), GMM-Kalman filter, optical flow, and ACF object detector. Additionally, three classification approaches for traffic congestion are examined. The comparison of the various approaches enables us to select the ones that will function best when incorporated into the framework suggested for addressing the traffic problems on the Bizerite Bridge. For mobile robots in industrial sectors with dusty environments, the author [32] seeks to develop noise-filtering algorithms that can remove dust from LiDAR sensory data. It was created as an intensity-based filter (LiDROR) based on a thorough examination of the characteristics of dust point clouds detected by a LiDAR sensor in order to accomplish the desired result. To the best of our knowledge, the suggested approach and the previously created LIOR are the first initiatives in this industry to design a de-dust filter utilizing non-AI method.

Utilizing 3D Light Detection and ranging, this work focuses on the issue of vehicle detection and tracking for an autonomous vehicle (LiDAR). A new clustering approach is suggested [33] to get vehicle candidates from preprocessed point cloud data gathered by the LiDAR to improve the accuracy of vehicle detection and tracking. To distinguish automobiles from vehicle candidates, a support vector machine (SVM) trained classifier is used. Vehicles are tracked using the Kalman filter and the global nearest neighbor (GNN) method, and the tracking data is used to further increase the accuracy of the vehicle detection results. A testing platform has been used to validate the suggested technique. In this study [34], a hierarchical approach based on RCNN was presented for the detection and recognition of vehicles and vehicle license plates. The tasks were related to RCNN’s complexity. Multiple complicated level RCNNs can be used in the same system in this fashion. For intelligent traffic monitoring, a sample vehicle detection system was created as an example. The license plate recognition RCNN was then taken into consideration for vehicle identification. Authors in [35] suggested a simple, cooperative object classification framework for CAV that protects user privacy and focuses on the object classification problem. The framework’s main novelty is the inclusion of the P-CNN model, which analyses encrypted images sent from cars and, when combined, yields accurate classification results. To protect the image during storage and transmission, extra image distortion is used. In P-CNN, our safe protocols based on the additive secret sharing technique execute the operations in the original CNN model (VGG16). The outcomes of the experiments show that P-CNN provides precisely the same classification outcomes as the VGG16 model without requiring vehicles to provide raw image data directly, which might contain private information. Additionally, the communication overhead and processing cost on two edge servers are reasonable.

The study [36] utilizes data from naturalistic driving to calculate the driver-vehicle volatilities. This work intends to forecast the occurrence of safety-critical events and deliver appropriate feedback to drivers and nearby vehicles by integrating and fusing several real-time streams of data, such as driver distraction, vehicle movements and kinematics, and instability in driving. The naturalistic driving data, which was gathered from more than 3500 drivers, includes 7566 normal driving events, 1315 severe events (such as crashes and near-crash situations), car kinematics, and driver behavior. Long Short-Term Memory (LSTM), 1D-CNN-LSTM, and 1D-Convolutional Neural Network (1D-CNN) are used to capture the local dependency and volatility in time-series data. The input parameters include driving volatility, vehicle kinematics, and distracted driving impairment. The findings show that the 1DCNN-LSTM model performs best, with an accuracy of 95.45% and a precision of 95.67% for predicting crashes that would occur in 73.4% of cases. The CNN layers extract additional information and consider the temporal dependency between observations, assisting the network in learning driving patterns and volatile behavior. A recent trend in research using deep learning [37], [38], [39], [40], [41] has shown its tremendous potential for vehicle classification. However, these techniques mostly did not consider the constrained environment where their restrictions of data quality or hardware constraints. Similarly, some of the techniques presented in [42] does not consider constrained environments as considered in the proposed work in this paper.

According to the literature review, the research that have been conducted for vehicle classification have some limitations particularly in case if constrained environment where either only low-quality images are present (for instance, due to bad weather conditions) or there are some hardware constraints. Some researchers stated their future work as to work with poor training datasets. So, in this work, we created custom dataset with poor quality images with multiples classes such as Car, Bike, Truck, Rickshaw and Bus. Every class used in our project is totally different from each other in terms of model. We take every view of the vehicle like front, back, side, front side and back side to train our network. We conduct our vehicle classification work based on two types of datasets: noisy dataset and a relatively clean dataset. Both these datasets
are custom datasets created and collected from real-world images using the image search feature of Google. In this work, the dataset is collected by keeping in view the limitations of the previous work. The most used methodologies of vehicle classification are implemented and evaluated using datasets to analyze their performance in real world environment.

III. PROPOSED METHODOLOGY

In our work, we apply deep learning techniques for the vehicle classification task. The initial step of methods is the data collection technique. The images used in this project are taken from Google images we contribute of ownership protection of the data. We provide ownership protection systems through watermarking technique. After that for classification purposes the initial step is preprocessing step in which noise is removed from images. After preprocessing is done on images, they can be used to train or to test the network which is also known as recognition step. In the training phase, the preprocessed images are passed to the network with their specified labels so that, for the classification the best network weights can be found. In the testing phase, the network is configured with the set of weights found during the training phase and then the recognition of images is performed. The recognition also outputs the confidence level against each vehicle image. The highest confidence level is used to depict the vehicle class. For increasing the number of training samples, in short for more generalization, the method of synthetic image generation is used during the preprocessing phase (the synthetic images are not being used in testing phase). Fig. 1 presents the general architecture of our proposed framework. A brief description of the proposed technique is mentioned in the following section. It begins with the data collection technique. After collecting the data, ownership protection of that data through watermarking is explained. For classification, preprocessing and the detailed description of convolutional neural network are also presented.

A. Data Collection

In this work, we used the images obtained from Google images. We used “get them all” downloader in chrome to download bunch of images at once. The entire project is executed on a personal computer with restricted computational resources. Our project is also limited to the simple classification of five classes: Car, Bike, Rickshaw, Bus and Truck. Every class is visually different from one another. But our dataset is boundless. Fig. 2 shows the representation of our dataset.

We performed different experiments to achieve maximum accuracy on two types of datasets based: noisy dataset and clean dataset. If one used an immovable camera, mainly CCTV or any other camera that is not generally configured once installed, in any application to detect and classify vehicles, there is a huge chance of noise involvement. This also leads to the foundation of our reason for using images from Google images there is a massive possibility of the involvement of noisy images in them. Moreover, these noisy images contain more than a single desired object in the image, for example, many bikes in a single image and the same issue for remaining classes as well. Therefore, we cleaned our dataset by removing such noisy images. As a result, our finalized dataset contains 4000 images of cars, 3213 images of buses, 4000 images of bikes, 4000 images of trucks, and 1159 images of rickshaws; that is, a total of 16372 images.

These evaluation configuration aims to measure the performance of the system with the vehicle's image database. The database is separated into five groups of non-overlapping classes. To perform experiments, the databases used in each experiment were divided into three sets: The training set, which is used for training the system, the validation set, which is used for dynamically tuning the Meta parameters of the system and then the test set, which is used to measure the accuracy of the system. In all experiments, it is ensured that there is no subject overlap among the three sets.

B. Preprocessing

The images should be preprocessed before it fed into the convolutional neural network. Our preprocessing pipeline consists of various steps as follows:

1) Noise removal: In a bid to work with real-world images with low quality and noise, we used the images obtained from Google images to train the neural network. As images are taken from Google images, there is a massive possibility of the involvement of noisy images and - low-resolution images with lots of variation. In this step, those images contain more...
than a single desired object in the image --for example, many bikes in a single image and the same issue for the remaining classes as well. At this point, we need to refine whole the dataset. We used certain filters including the Gaussian filter and median filter to refine the images containing multiple objects and those captured in low-light conditions. The unsharp filter is also applied to refine edges and make the objects in ambiguous images clearer thus making it easier for the model to classify such images.

2) **Downsizing:** The training set contains a diversity of image dimensions as are downloaded from Google images. We resized the images into 32x32 and 64x64 dimensions as required by the model to work with low resolutions. Because images with large dimensions like 256x256 are too time-consuming and sometimes it is very difficult to process them due to hardware limitations or due to the availability of - low-resolution images only in the real-world environment. By using downsize images, there is a chance of information being lost which s to poor results. As in our experiments, the accuracy stuck at 87% by using 32x32 dimension images. When we move towards 64x64 dimension images we get better results approximately 90%. So, if we use high dimension images for training there might be a chance to get better results. We used a light image resizer to resize these all images at once.

3) **Rotation:** Convolutional neural networks are more suitable to work with huge amounts of data. The network trained well with maximum samples for training. Maximum training allows the network to better learn the variation which is present in the data. To increase the data, data augmentation technique is used. By using this technique, images can be transformed and produced a new image. Rotation is one of the simple transformations through which we can increase our data by just angular rotating the original image. In our approach, we applied 10 different angular rotations. In our dataset, with concern of the different classes, the Bus and Rickshaw chavings has low representation meanwhile the dataset in this phase is unbalanced; therefore, the rotation phase helps in generating additional data through 10 angular rotations. We perform rotations with the angles [-2, 2, -4, 4, -6, 6, -8, 8, 10, -10].

4) **Flipping:** Several data augmentation techniques can be used to increase the size of training datasets. One of the widely employed techniques is to horizontally flip photographs. Different flipping images can be used to train a model. To double the size of the data, we treat the image that has been flipped as a genuine image. the in this project, we use random flipping to train our model. In our dataset, with the concern of the different classes, the Bus and Rickshaw class is underrepresented meanwhile the dataset in this phase is unbalanced therefore generating additional data through flipping. Color space.

5) **Transformation:** The source image is used if it has already been colored and is in the RGB color space. Any source images found in the dataset that are not in RGB color space and exist in black and white, grayscale, or any other non-RGB color system are transformed into RGB color space using industry-standard procedures. The convolutional neural network (CNN) only accepts and processes RGB images, hence we have included colored images in the suggested work as well. Red, green, and blue hues are defined by RGB’s three channels.

C. **Convolutional Neural Network**

CNN are the specified neural network. In recent past, much research work regarding image processing and classification has been done by using CNN. The CNN architecture is made up of various kinds of layers. These layers comprise of convolutional layer, pooling layer, and fully connected layers. These layers may contain some supplementary hyper parameters such as size of filter, padding, stride for the convolutional layers and the quantity of neurons used in fully connected layer. The all number of layer types with their supplementary hyper parameters and their consequence on training quality and speed, all these circumstances make it hard to choose architecture. The capacity of learning makes this classifier much superior to other classifiers. It not only learns the weight of features but the features themselves. On universal image classification, CNNs have accomplished revolutionary accuracy. Similar concepts are adjustable for audio and videos as well. In this segment, we define CNN for the processing of images.

Fig. 3 represents the generic architecture of CNN used in the proposed scheme. CNN comprises of numerous convolution layers, pooling layers, and a fully connected layer. After one or several convolution layers, a pooling layer is applied. In this architecture, the convolution layers extract the useful features from the input that produce feature maps. The pooling layers then decrease the spatial size of these feature maps. There are multiple filters used in this architecture with different kernel sizes. It displays the edges of the input image as a result and then this idea of filters for feature extraction is interpreted to CNN. CNN learns these filters as a replacement of hard coded filters.

1) **Convolutional layer:** As a normal neural network, the convolution layer comprises of units as well, however with a rare arrangement and an unusual connectionism of the units. The main differences between these two networks are based on weight sharing, local connectivity, and three-dimensional arrangement of the unit.

![Architectural flow of CNN on input data.](image-url)
In weight sharing, the same weights are used for several output units. Basically, the weights are the filters and these filters are used on the entire input to generate the output. In local connectivity, the size of this connectivity is defined by the kernel size. In this connection, all the units of input are not connected with the output unit. The three-dimensional arrangement of the unit arises from the image. Colored image normally comprises of three channels (red, green, and blue). Each channel is defined by a two-dimensional matrix. Consequently, the input and output of the CNN is a three-dimensional matrix. The output comprises of three-dimensional matrix with feature maps of two dimensions and this x times for the filter numbers used in this layer. Each filter creates a feature map. In convolutional layer, the feature map is obtained by convolving a filter across the input volume, adding a term bias. If we represent the \( k^{th} \) input feature map of the convolutional layer as \( h^k \) and its filter is denoted by \( w^k \) and term bias \( b^k \), then the output which is feature map \( h^{k+1} \) is obtained as:

\[
h^{k+1} = W^k \times h^k + b^k \tag{1}
\]

Where \( \times \) shows a convolutional function

2) Pooling layer: The pooling layer is applied after convolution layers. This layer is used to minimize the spatial size of the feature maps. It actually downsamples the feature maps. The spatial dimension of the feature maps is reduced of 75\% by one pooling with the stride size of 2*2. Size of sliding over the input channel is called stride. The pooling layer has no activation function or weight to learn because it works independently on each feature map. The max-pooling and average-pooling are the common pooling layers. The max-pooling layer calculates the maximum, which lies on the receptive side of filter and the average-pooling layer calculates the average.

3) Fully connected layer: These layers work identically to hidden layers of a regular neural network. To determine the output scores and arbitrary features, these fully connected layers can be used at the termination of the CNN after numerous stages. As the name shows, in this section all layers of the units relate to the input layer.

4) Training: In this phase, we have a trained set of input vectors with a parallel set of output vectors. The goal is to use training data to train the network's biases and weights so that the network performs good in classifying input. The further training data given to a network, the additional iterations and weight updates can possible and then it came out with improved network architecture which gives better results.

Through a training process, the way a computer is capable to adjust its filter values or weights is called back propagation. Backpropagation is an efficient method to propagate the error to calculate the gradient from the loss function through the network. It computes the gradient referring to the weights. The backpropagation has the following recursively equation to calculate the gradient in respect to the weights.

\[
\frac{\partial L(\theta)}{\partial w^l_k} = \sigma_i(0) \cdot \sigma^{(i-1)}(k) \tag{2}
\]

Backpropagation can be divided into four different segments. Forward pass, loss function, backward pass and weight update. In forward pass, the training images pass through the whole network. In the first training image all the filter values or weights are casually initialized thus output does not give any sensible conclusion about what the classification could be. This leads towards the loss function of backpropagation. Loss function is used to reduce the error. Modest loss function is the mean squared error (MSE) which computes the squared difference between half of the difference between actual and predicted as follows:

\[
L(\theta) = \frac{1}{2N} \sum_{n=1}^{N} \| f(Y_n; \theta) - Z_n \|^2 \tag{3}
\]

The backward pass in the propagation decides which weights donates maximum loss and it adjusts the weights to decrease the loss. After this in the weight update process, all the weights of the filters got updated.

\[
W = W - \eta \frac{\partial L(\theta)}{\partial w} \tag{4}
\]

5) Gradient descent: To minimize the loss function, gradient descent method is mostly used. It needs the gradient of the loss function in terms of \( \alpha \). Backpropagation method is used to calculate the gradient. Gradient descent is an iterative optimization algorithm. In this algorithm weights can be updated with the help of gradient.

\[
\theta \leftarrow \theta - \eta \frac{\partial L(\theta)}{\partial \theta} \tag{5}
\]

Where \( \eta \) defines the learning rate. Learning rate is a parameter that is selected by the programmer. High learning rate meant to take higher steps to update the weight.

6) Dropout layer: Dropout layers have a very definite function in neural networks. The idea of dropouts is basic in nature. It is a technique used for decreasing over fitting in neural networks. The units will be arbitrarily dropped for each iteration of the training. Dropout layer is only used in training time. Units are dropped with the probability of \( p \) and this \( p \) can be different for each layer. It is a method that prevents overfitting. It drops out an arbitrary set of activations in that layer by setting them to zero in the forward pass.

By using dropout, the network should be able to deliver right classification if some of the activations are dropout. In supervised learning, it improves the performance of neural networks. Except loss layer, dropout can be used in every layer and applied during supervised training alongside with end to end back propagation. The option of dropping the unit is random. In dropout, the forward process is defined as:

\[
h^{k+1} = M \ast (w^k \times h^k + b^k) \tag{6}
\]

7) Rectified linear unit: Rectified Linear Unit (ReLu) is the efficient activation function because the network is capable to train a lot faster while not making any significant difference to the accuracy. The value for ReLu can be calculated as:

\[
\sigma(y) = \max(0, y) = \begin{cases} 
0, & \text{if } y < 0 \\
y, & \text{if } y \geq 0
\end{cases} \tag{7}
\]
8) Exponential linear unit: Exponential linear unit (ELU) is for negative values. Over the negative activation, the mean activation is near zero and this led towards the faster merging of the network because the input for the next layer has zero mean. The formula for ELU is:

$$\sigma(x) = f(x) = \begin{cases} a(e^x - 1), & \text{if } x < 0 \\ x, & \text{if } x \geq 0 \end{cases} \tag{8}$$

Where $a>0$ and is generally set to 1.

9) SoftMax: SoftMax activation function is used for the encoding of the output. It represents a probability distribution over k classes and hence it is only used in the output layer. The formula of SoftMax is:

$$\sigma(y)_j = \frac{e^{y_j}}{\sum_{k=1}^{k} e^{y_k}}, \text{ for } j = 1, ..., k \tag{9}$$

IV. RESULTS AND DISCUSSION

For experimentation, we used libraries like Tensorflow, Keras, Sklearn, Matplotlib, Scipy, and OpenCV in python language running on a Core i3 processor with 8GB of RAM.

A. Convolutional Neural Network for Vehicle Classification

The proposed architecture of our CNN has been presented in Fig. 3. Various convolutional neural network architectures were used in our implementation. The reason behind this is to pinpoint the architecture that achieved a reasonable accuracy with under restricted environment with respect to computational resources and quality of the data. The network takes an input of 64x64 RGB image and then classifies the given image assigning it a vehicle type. Our architecture of CNN consists of 5 Convolutional layers, 4 fully connected layers and then a softmax output layer. We used ReLUs as activation function. The first CNN layer is a convolutional layer on which we apply Convolutional kernel of 5x5 and results an image of 32x32 pixels. This convolutional layer is then followed by another sub sampling layer that uses max pooling (with 2x2 kernel size) to shrink the image into half of its original size. Then drop out layer is applied with drop out 0.2. The second CNN layer is a convolutional layer on which we apply a Convolutional kernel of 3x3 and results an image of 16x16 pixels. This convolutional layer is then followed by another sub sampling layer that uses max pooling (with 2x2 kernel size) to shrink the image into half of its original size. Then drop out layer is applied with drop out 0.2. The third CNN layer is a convolutional layer on which we apply a Convolutional kernel of 3x3 and results an image of 8x8 pixels. This convolutional layer is then followed by another sub sampling layer that uses max pooling (with 2x2 kernel size) to shrink the image into half of its original size. Then drop out layer is applied with drop out 0.2. The fourth CNN layer is a convolutional layer on which we apply a Convolutional kernel of 3x3 and results an image of 4x4 pixels. This convolutional layer is then followed by another sub sampling layer that uses max pooling (with 2x2 kernel size) to shrink the image into half of its original size. Then drop out layer is applied with drop out 0.2. The fifth CNN layer is a convolutional layer on which we apply a Convolutional kernel of 3x3 and results an image of 2x2 pixels. This convolutional layer is then followed by another sub sampling layer that uses max pooling (with 2x2 kernel size) to shrink the image into half of its original size. Then drop out layer is applied with drop out 0.2.

B. Experiment 1: Noisy Dataset

Since the real-world images are often noisy and blur; therefore, we also tested the performance of our framework on noisy data. Table I shows the detailed dataset of noisy images. As the rickshaw class contains few images so we generated additional data for the rickshaw class to improve the imbalance ratio. Now our new data contains 96273 images.

<table>
<thead>
<tr>
<th>Class</th>
<th>Number of Images</th>
</tr>
</thead>
<tbody>
<tr>
<td>Car</td>
<td>19333</td>
</tr>
<tr>
<td>Bus</td>
<td>18844</td>
</tr>
<tr>
<td>Truck</td>
<td>21142</td>
</tr>
<tr>
<td>Bike</td>
<td>19374</td>
</tr>
<tr>
<td>Rickshaw</td>
<td>15822</td>
</tr>
</tbody>
</table>

1) Model 1: In the first set of experiments, we took 20% of data for training to investigate the results when many images are not available for the training. So, our first split contains 19254 images in the training set. Next, we flipped our training data of 19254 images and generated new 38504 (two times the size of training data) samples through flipping because the training set was quite small as compared to the testing set that contained 80% of the images. Then we again apply the split of 34656 training images and 3848 images for validation; we got a validation accuracy of 74% and validation split accuracy of 73% and testing accuracy of 66% in this set of experiments. Since this accuracy is not good enough, we proceed to the next set of experiments.

2) Model 2: In the experiment set no 2, data is the same as used in the experiment set no 1. But this time we applied the split of 50/50. So, our split A becomes 48136/48137. Now we do not apply any flipping and the network remains the same. We use a split B of 43322 training images and 4814 images for validation. We got a validation accuracy of 75%, validation split accuracy of 76%, and testing accuracy of 62%. Through the results of the experiment set 1 and 2, we conclude that flipping yields higher accuracy on test data. This accuracy can be improved by working with higher resolution images and using a machine with higher computational power, such as a GPU (graphical processing unit) as there are too many samples for normalizing but in this paper, we are examining the results in a constrained environment with hardware constraints and low data quality. This is because the real-world images usually do not have good quality and are at times noisy. Moreover, we are also advocating the case for the importance of increasing the computational resources for better results. In our experimental study, we observed that training samples greater than 80,000 become too computationally intensive for the CPU to normalize data. For 80,000 images with 32x32 image resolution, the normalization function must perform 32x32x3*80,000 computations.
3) Model 3: In the experiment set no 3, the data used was the same as in the previous experiment. We applied a split of 30/70 which is 28881/67392 for the training/testing dataset. Next, we applied -left-right flipping on the training dataset obtaining 57762 images. We split this data again for applying validation during the classification process with a split of 51985 images for training and 5777 images for validation. In this experiment, we got a validation accuracy of 74.8%, a validation split accuracy of 74%, and testing accuracy of 63.4%. In this experiment, we notice that test accuracy is not increasing so we decided to tweak the network architecture further.

4) Model 4: In this set of experiments, data remained same as in experiment set 3 and we only changed the network architecture by adding one convolutional layer and got increased validation accuracy of 77%, validation split accuracy of 75% and testing accuracy of 67%. This motivated us to further investigate the effect of network architecture on overall classification results.

5) Model 5: In experiment set no 5, we again changed the network architecture by adding one more convolutional layer and got a validation accuracy of 78%, validation split accuracy of 77% and testing accuracy of 67%. In this experiment, we noticed that adding an additional layer did not help in increasing accuracy.

Table II summarizes the experimental results obtained while running the experiments on the noisy dataset. The highest testing accuracy we obtained in these experiments is 67 percent. We can get higher accuracy through this dataset by using higher image dimensions but in this paper, we are concerned with low-resolution images and experiments with hardware constraints. The images used in this dataset were taken from Google images to cater for real world scenarios under noisy environment. Some images contain more than a single desired object in the image. For example, multiple bikes in a single image. Same for the other classes as well. At this point, we need to refine all dataset because training with such samples require higher image dimensions such as training with 96x96 images for 40,000 samples took around 2 hours.

C. Experiment 2: Clean Dataset (32-Bit Images)

Our data for the previous set of experiments was noisy and we cleaned the data for further experiments. We kept those images which contain a single desired object but again with a low resolution of 32x32. Table III presents the statistics of this dataset.

<table>
<thead>
<tr>
<th>Exp. No.</th>
<th>Split</th>
<th>Flip</th>
<th>Training Data</th>
<th>Validation Acc</th>
<th>Testing Accuracy</th>
<th>F1-Score</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>20/80</td>
<td>Yes</td>
<td>96273</td>
<td>74%</td>
<td>66%</td>
<td>74%</td>
</tr>
<tr>
<td>2</td>
<td>50/50</td>
<td>No</td>
<td>96273</td>
<td>75%</td>
<td>62%</td>
<td>75%</td>
</tr>
<tr>
<td>3</td>
<td>30/70</td>
<td>No</td>
<td>96273</td>
<td>74.8%</td>
<td>63.4%</td>
<td>74%</td>
</tr>
<tr>
<td>4</td>
<td>30/70</td>
<td>No</td>
<td>96273</td>
<td>77%</td>
<td>67%</td>
<td>77.4%</td>
</tr>
<tr>
<td>5</td>
<td>30/70</td>
<td>No</td>
<td>96273</td>
<td>78%</td>
<td>67%</td>
<td>76.8%</td>
</tr>
</tbody>
</table>

Table III. Overview of Clean Dataset (32-Bit Images)

<table>
<thead>
<tr>
<th>Class</th>
<th>Number of images for training</th>
<th>Number of images for testing</th>
</tr>
</thead>
<tbody>
<tr>
<td>Car</td>
<td>4000</td>
<td>2000</td>
</tr>
<tr>
<td>Bus</td>
<td>3213</td>
<td>1999</td>
</tr>
<tr>
<td>Truck</td>
<td>4000</td>
<td>2000</td>
</tr>
<tr>
<td>Bike</td>
<td>4000</td>
<td>2000</td>
</tr>
<tr>
<td>Rickshaw</td>
<td>1159</td>
<td>400</td>
</tr>
<tr>
<td>Total images</td>
<td>16372</td>
<td>8399</td>
</tr>
</tbody>
</table>

1) Model 1: In experiment set no 1, we applied -left-right flipping on the training data and our data size becomes 32744 images. We trained on 31106 samples and validated on 1368 samples and got a validation accuracy of 84% and a testing accuracy of 80%. From a similar experiment on the noisy dataset, we noticed that test accuracy improves with cleaner data but the bus and rickshaw class is underrepresented. Therefore, we applied rotations for generating additional data for the next set of experiments.

2) Model 2: In experiment no. 2 on the clean dataset, our dataset contains 18690 samples when we applied flipping. We trained on 35511 and validated on 1869 samples and got a validation accuracy of 92.83% and a testing accuracy of 83%. In this experiment, we notice that accuracy further improves by generating additional data. Now in the next set of experiments, we investigate the change in results due to changes in network architecture.

3) Model 3: In this set of experiments, the data is the same as in the previous experiment. We trained on 35511 and validated on 1869 samples. However, in this experiment, we changed the network architecture by increasing the filter depth and got a validation accuracy of 89% and a testing accuracy of 86.4%. Increasing filter depth improved testing accuracy. Next, we examine the effect of adding more layers on the overall classification results.

4) Model 4: In this experiment, data is again the same as in the previous experiment. We added one more convolutional layer and got a validation accuracy of 90% and a testing accuracy of 87% which is not significant. So next we add a fully connected layer to see its effect on the results.

5) Model 5: In experiment no. 5, we only added one fully connected layer and got a validation accuracy of 90.85 and a testing accuracy of 87%. The results of these experimental studies have been tabulated in Table IV.

Table IV. Results of CNN Experiments on Clean Dataset (32-Bit Images)

<table>
<thead>
<tr>
<th>Exp. No.</th>
<th>Rotation</th>
<th>Flip</th>
<th>Training Data</th>
<th>Validation Accuracy</th>
<th>Testing Accuracy</th>
<th>F1-Score</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>No</td>
<td>Yes</td>
<td>32744</td>
<td>84%</td>
<td>80%</td>
<td>83.3%</td>
</tr>
<tr>
<td>2</td>
<td>Yes</td>
<td>Yes</td>
<td>37380</td>
<td>92.83%</td>
<td>83%</td>
<td>92%</td>
</tr>
<tr>
<td>3</td>
<td>No</td>
<td>No</td>
<td>37380</td>
<td>89%</td>
<td>86.4%</td>
<td>89.5%</td>
</tr>
<tr>
<td>4</td>
<td>No</td>
<td>No</td>
<td>37380</td>
<td>90%</td>
<td>87%</td>
<td>90.2%</td>
</tr>
</tbody>
</table>
Table IV summarizes the experiments which we performed on a clean dataset of images with a resolution of 32x32. This dataset produces higher accuracy than noisy datasets because in this dataset only desired object is involved. Experiments seem to be giving fine results while using 5 convolutional layers and 4 fully connected layers. But accuracy is stuck here because the images have been downsized to 32x32 and due to this downsizing information has been lost. This motivated us for our next set of experiments with 64x64 images.

D. Experiment 4: Clean Dataset (64-Bit Images)

We performed previous experiments on 32x32 bit images that was a very low image resolution making it challenging to obtain good classification results. To further investigate the performance of CNN on images with slightly better resolution, we performed experiments with 64x64 images. Table V shows the statistics dataset of clean 64x64 images.

<table>
<thead>
<tr>
<th>Class</th>
<th>Number of images for training</th>
<th>Number of images for testing</th>
</tr>
</thead>
<tbody>
<tr>
<td>Car</td>
<td>4000</td>
<td>2000</td>
</tr>
<tr>
<td>Bus</td>
<td>3213</td>
<td>1999</td>
</tr>
<tr>
<td>Truck</td>
<td>4000</td>
<td>2000</td>
</tr>
<tr>
<td>Bike</td>
<td>4000</td>
<td>2000</td>
</tr>
<tr>
<td>Rickshaw</td>
<td>1159</td>
<td>400</td>
</tr>
<tr>
<td>Total images</td>
<td>16372</td>
<td>8399</td>
</tr>
</tbody>
</table>

1) Model 1: In the first experiment set for 64-bit images, to improve the imbalance ratio, we generated some additional data for rickshaw and bus classes and now have 18690 samples. We trained on 17755 samples and validated on 935 samples and got validation accuracy of 88% and test accuracy of 86%. We notice that even without flipping 64x64 bit-sized training yielded high accuracy.

2) Model 2: In the experiment set 2, we took the sample size of 16372 and applied rotation; as a result, our data has now images 18690. Next, we apply after flipping and the data now has 37380 images. We trained on 35511 and validated in 1869 and we got a validation accuracy of 92% and a test accuracy of 88%. After this experiment, we observed that flipping the training data improved the accuracy by 2%.

3) Model 3: In experiment no. 3, we changed the convolutional architecture by just adding one more convolutional layer. By changing the architecture, we got a validation accuracy of 93% and a test accuracy of 89% and observed a slight improvement in the results. Next, we investigate the effect of modifying the network architecture a bit further.

4) Model 4: In this experiment set, we changed the convolutional architecture by adding a fully connected layer and got a validation accuracy of 93% and a test accuracy of 90%.

Table VI summarizes the results of experiments that we performed on the clean dataset with the dimension of 64x64. By using 64x64 dimension images, we get 90% accuracy which is higher than for those 32-bit images and noisy images as evident from Tab. 2, 4 and 6. We can safely claim that we can get higher efficiency with our architecture if we work with higher resolution images and better computer power like GPU. However, in this paper, we are investigating the case of real-world noisy images and low-resolution images with hardware constraint; therefore, we did not perform experiments with better resolution, say images with 256x256 resolution.

<table>
<thead>
<tr>
<th>Exp. No.</th>
<th>Rotation</th>
<th>Flip</th>
<th>Training Data</th>
<th>Validation Accuracy</th>
<th>Testing Accuracy</th>
<th>F1-Score</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>Yes</td>
<td>No</td>
<td>18690</td>
<td>88%</td>
<td>86%</td>
<td>88.5%</td>
</tr>
<tr>
<td>2</td>
<td>Yes</td>
<td>Yes</td>
<td>37380</td>
<td>92%</td>
<td>88%</td>
<td>91.5%</td>
</tr>
<tr>
<td>3</td>
<td>No</td>
<td>No</td>
<td>37380</td>
<td>93%</td>
<td>89%</td>
<td>93.7%</td>
</tr>
</tbody>
</table>

E. Discussion

In the proposed work, different sets are experiments are performed for both noisy and clean images dataset. The reason for doing number of experiments for both the cases is to go through different combinations, data splits and see the impact on performance. The experiments conducted on noisy images dataset explored the impact of different settings and conditions on the accuracy of classification in image processing. The findings revealed that using a smaller portion of the data for training resulted in lower accuracy, emphasizing the need for sufficient training data. Flipping techniques were applied to augment the training data, leading to improved accuracy in some cases. The importance of factors such as image resolution, computational resources, and network architecture were highlighted, with suggestions for further optimization. Overall, the experiments demonstrated the significance of data augmentation, hardware capabilities, and network design in achieving higher accuracy in image classification tasks. Similarly, the findings deducted from experiments performed on clean dataset with 32-bit images concluded that data augmentation techniques, such as flipping and rotations, led to improved accuracy in image classification tasks. Experiment set 1 demonstrated that left-right flipping resulted in a validation accuracy of 84% and testing accuracy of 80%. Further experiments on clean datasets (experiment set 2) showed that additional data generation through flipping enhanced accuracy, with a validation accuracy of 92.83% and testing accuracy of 83%. Changes in network architecture (experiment set 3) by increasing the filter depth led to improved testing accuracy of 86.4%. Subsequent experiments involving the addition of more layers (experiment set 4) and a fully connected layer (experiment set 5) yielded validation accuracies of 90% and 90.85%, respectively, contributing to marginal improvements in testing accuracy. These findings highlight the significance of data augmentation and network architecture modifications in enhancing the classification results of image processing models.

In case of experiment performed on clean dataset with 64-bit images, the conducted experiments revealed interesting findings in image classification tasks. In experiment set 1, the training of 64-bit images without flipping resulted in high accuracy, with a validation accuracy of 88% and test accuracy...
of 86%. Experiment set 2 demonstrated the benefits of data augmentation through flipping and rotation, leading to an increased sample size and improved accuracy of 92% in validation and 88% in testing. Further modifications in the convolutional architecture, specifically adding one more convolutional layer (experiment set 3), contributed to a slight improvement in accuracy, with a validation accuracy of 93% and test accuracy of 89%. Finally, in experiment set 4, the addition of a fully connected layer resulted in a validation accuracy of 93% and test accuracy of 90%. These findings emphasize the importance of data augmentation and architectural adjustments in enhancing the performance of image classification models.

For comparison purposes, we also ran the experiments using other classifiers such as SVM, Naïve Bayes, and Decision trees on 64-bit images. Table VII lists the results of these experiments. Along with the results represent the dataset which we use for comparative analysis. In this comparison, we compare our result with other machine learning algorithms. For the dataset which gives higher accuracy in CNN, we take that dataset and compute result in SVM, Naïve Bayes and Decision tree.

TABLE VII. PERFORMANCE COMPARISON WITH OTHER CLASSIFIERS

<table>
<thead>
<tr>
<th>Experiment</th>
<th>Cell per Block</th>
<th>Pixel per cell</th>
<th>Feature Vector size</th>
<th>Validation Accuracy</th>
<th>Test Accuracy</th>
</tr>
</thead>
<tbody>
<tr>
<td>SVM</td>
<td>2</td>
<td>8</td>
<td>1764</td>
<td>82.9%</td>
<td>83.39%</td>
</tr>
<tr>
<td>Naïve Bayes</td>
<td>2</td>
<td>8</td>
<td>1764</td>
<td>71.39%</td>
<td>69.6%</td>
</tr>
<tr>
<td>Decision Tree</td>
<td>2</td>
<td>8</td>
<td>1764</td>
<td>50%</td>
<td>50.67%</td>
</tr>
<tr>
<td>VGG-16</td>
<td>-</td>
<td>-</td>
<td>1764</td>
<td>56%</td>
<td>55.97%</td>
</tr>
<tr>
<td>ResNet-50</td>
<td>-</td>
<td>-</td>
<td>1764</td>
<td>64%</td>
<td>68.52%</td>
</tr>
</tbody>
</table>

We also present the comparison of the performance of CNN with other classifiers: SVM, Decision Tree, and Naïve Bayes in Fig. 4. It is evident from this figure that our convolutional neural network achieves highest accuracy in comparison to other machine learning algorithms. CNN attains 90% accuracy while using 5 convolutional layers and 4 fully connected layers. The results are very conclusive, and CNN shows the highest accuracy rate in comparison to other classifiers.

Although our technique has a unique characteristic of working under constrained environment, comparing it with deep learning-based vehicle-type classification techniques like [34] (proposed by Wang et al.) shows that our technique gives better results as compared to 65.55% accuracy of their work. However, the advantage of [34] over our proposed technique is that the technique of Wang et al. can be used to classify similar vehicles (for instance, cars only) as well. While on the other hand, we considered different types of vehicles like cars, bikes, buses etc.

Table VIII shows the comparison of the proposed with some of the latest works on same domain using similar approaches. The proposed models outperform the existing works in terms of accuracy.

TABLE VIII. COMPARISON OF PROPOSED MODEL WITH PREVIOUS WORKS

<table>
<thead>
<tr>
<th>References</th>
<th>Years</th>
<th>Accuracy</th>
</tr>
</thead>
<tbody>
<tr>
<td>[44]</td>
<td>2020</td>
<td>89.50%</td>
</tr>
<tr>
<td>[34]</td>
<td>2021</td>
<td>65.55%</td>
</tr>
<tr>
<td>[43]</td>
<td>2021</td>
<td>77.55%</td>
</tr>
<tr>
<td>[45]</td>
<td>2023</td>
<td>77%</td>
</tr>
<tr>
<td>Proposed</td>
<td>2022</td>
<td>90.85%</td>
</tr>
</tbody>
</table>

V. LIMITATIONS AND FUTURE WORK

Several limitations are there that effect the streamline vehicle image classification using deep learning mechanisms. One limitation is that the current work was trained and tested on 32-bit and 64-bit datasets, which may restrict the model's efficiency. To address this, the researchers suggest training the model using a higher resolution dataset, such as 256x256 bits, which could lead to improved results. However, they acknowledge hardware constraints and propose the use of GPU clusters for training on larger datasets and more complex neural networks. Gathering more samples from different vehicle classes is also identified as a way to enhance the results and expand the scope of the study. In the future, the researchers aim to overcome these hardware and software limitations to achieve higher accuracy. Additionally, they suggest extending the work by considering vehicle images captured under poor lighting conditions or adverse weather situations. By addressing these limitations, such as utilizing higher-resolution datasets, overcoming hardware constraints, and expanding the dataset size, the researchers anticipate achieving improved accuracy and broader applicability of the classification technique in the future.

VI. CONCLUSION

Data is the most important asset. To explore and extract useful information from this data, the data is also shared with several stake holders via diverse channels. Hence ownership protection of such datasets is very essential. We provide ownership protection systems through watermarking technique. After ownership protection of the data, we have presented an image-based vehicle classification technique from vehicle images by using a supervised learning convolutional neural network. The vehicle image is taken to the network as an input and the network outputs the vehicle class to which the vehicle
belongs. For the vehicle classification, we have established our own CNN architecture and compared these CNN results with furthermore machine learning algorithms like Naïve Bayes, SVM and Decision Tree. As the images are taken from Google, we contribute ownership protection to prove the ownership of the data. For this we insert a watermark in our dataset. Binary bits of URL of the images are used as the secret information of ownership. Every image used in our project may lead towards the path where it has been taken from. One limitation of current work is that we trained and tested on 32 bit and 64-bit datasets. In our work we have some hardware constraints. If we trained our model by using 256x256 bit dataset we may lead towards higher efficiency. For much better results GPU clusters are needed to train on large datasets and higher dimensions and deeper neural networks. If we gather more samples from different classes, we can improve our results and increase our scope. In future we are considering coping with all these hardware and software limitations for much better accuracy. Further work can be done by using vehicle images on poor lightning condition or bad weather situations.

ACKNOWLEDGMENT
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REFERENCES
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Drug Resistant Prediction Based on Plasmodium Falciparum DNA-Barcoding using Bidirectional Long Short Term Memory Method

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Faculty of Mathematics and Natural Sciences, Brawijaya University Malang, Indonesia²

Abstract—Malaria disease mostly affects children and causes death every year. Multiple factors of the disease due to failure in treatment, including anti-malaria drug resistance. The resistance is caused by a decrease in the efficacy of the drug against Plasmodium parasites. Therefore, we proposed a computational approach using deep learning methods to predict anti-malarial drug resistance based on genetic variants of the Plasmodium falciparum through DNA barcoding. The DNA Barcode, organism identification from Plasmodium, is employed as data set for predicting the anti-malaria drug resistance. As a univariate amino acid sequence, it is transformed to numerical value data for building classifier model. It is constructed into a classifier model for prediction using Bidirectional Long Term-Short Memory (Bi-LSTM). This algorithm is extended from LSTM by two directions. In the first stage, the sequence is encoded into numerical data as input data for the method using sigmoid activation loss function. Then binary cross entropy is addressed to define the class, resistance or sensitivity. The final stage is applied by tuning hyper-parameter using Adaptive Moment Estimation optimizer to get the best performance. The experimental results show that Bi-LSTM as the proposed method achieves high performance for resistance prediction including precision, recall, and f1-score.

Keywords—Drug resistant; plasmodium falciparum; Bi-LSTM; deep learning

I. INTRODUCTION

Malaria is a type of infectious disease that is transmitted through mosquito bites or through blood transfusions. This type of disease exists in countries with tropical climates, including Indonesia. Various efforts have been made to reduce the number of deaths from this disease with intensive treatment. One of the causes of this high case is resistance to anti-malarial drugs against the plasmodium parasite. As it is known that the types of anti-malarial drugs are classified as antibiotics, so an appropriate regulation is needed. The development of efficient malaria vaccines is hampered by the inclusion of information about the huge number of genetic variants of this parasite, as many of these alleles avoid the allele-specific immunity elicited by vaccines. Potential vaccine candidate are merozoite proteins 1 and 2 (msp1 and msp2) and glutamate-rich protein from P. falciparum [1]. Resistance of the parasite Plasmodium falciparum to anti-malarial drugs is one of the causes of the high mortality rate in endemic areas.

One of the main causes is a mutation in the gene of the parasite, so that the drug is not right on the target region. Some genes are mutated in the anti-malarial drug Chloroquine (CQ) to the duplication of Mefloquine, it affected the Plasmodium to survive in the host’s body. Several studies related to drug resistance were carried out both in-vivo and in-vitro to in-silico using the bioinformatics method approach to find the characteristics of these parasites. Molecular biology research was carried out in the wet lab to determine the level of polymorphism of the anti-malarial treatment target genes [2], [3]. Furthermore, based on a sufficiently high volume data set, research using a computational approach was conducted, how it is limited to image data (haploid), clinical data, and chemical data analysis to DNA sequence data (genetic variants) using machine learning algorithms [1], [4]–[8]. Therefore, we proposed to conduct research on study of genetic variant analysis against plasmodium parasite resistance in target genes, including the first transporter gene (Pfmsp-1) and second transporter (Pfmsp-1), as well as Pfglurp on the anti-malarial drug CQ with a computational approach using the deep learning method. Genetic variants are DNA sequence data from parasites that undergo mutations categorized as time series data. Long Term Sort Memory (LSTM) is a robust deep learning method for time series data. However, the right combination of hyper-parameters from this algorithm affects the performance. Then, this research aims to develop Bi-LSTM algorithm using tuning parameters to purpose a high level of performance.

II. MATERIAL AND METHOD

A. Drug Resistance of Anti-Malaria

Treatment of malaria is through drug therapy which aims to eliminate the plasmodium parasite in the host’s body. However, the resistance of the parasite Plasmodium falciparum to anti-folate resulted in the sequential acquisition of mutations in the target gene as shown in Fig. 1. The high mutation rate of this type of Plasmodium parasite species causes resistance, so a combination of anti-malarial drugs is carried out. Including efforts for possible immunity of malaria parasites to anti-malarial drugs, WHO has advised the use of an approved combination of artemisinin, Artemisinin Combination Therapy (ACT) [9].
**B. DNA Barcoding**

DNA barcoding is a strategy to recognize and to give automatic validation for living beings. It has a specific region, and the length of the sequence is between 300 and 400 bps. A method for automatically identifying and validating living things is DNA barcoding. The sequence has a specific region and a length between 300 and 400 bps. According to Li, et al. (2018), the barcoding-based protein measuring approach converts protein signals into barcoded oligo probes and amplifies the signals with nucleic acid amplification to achieve ultrasensitive detection down to the single molecule level [10]. With this technique, it is possible to identify rare tumor cells or cancer cells with low abundance surface markers [11]. The Weissleder group developed photo-cleavable DNA- barcoded antibodies that are specifically capable of recognizing multiplexed cell biomarkers [10]. After that, DNA has been cleaved by light (less than 365 nm) and released into solution; gel electrophoresis can be used to analyze it. They also came up with an amplification-free method to profile more than 90 proteins in single cells using DNA-barcoded antibodies and Nano String’s fluorescent readout [12] to study the drug response pathway and inter- and intra-tumor heterogeneity in clinical samples [12], [13]. DNA barcoding can be used to study the resistance of Plasmodium parasites to various anti-malaria drugs. Here are a few examples:

1) Chloroquine was once the most widely used anti-malaria drug, but its effectiveness has been greatly reduced by the emergence of chloroquine-resistant Plasmodium falciparum parasites. DNA barcoding can be used to identify mutations in the parasite’s Pfert gene that are associated to chloroquine resistance.

2) Artemisinin and its derivatives are currently the most effective anti-malaria drugs, but artemisinin-resistant parasites have emerged in Southeast Asia. DNA barcoding can be used to identify mutations in the parasite’s K13 gene that are associated with artemisinin resistance.

3) Sulfoxide-pyrimethamine is a combination drug that is used to treat malaria and is also used for intermittent preventive treatment in pregnant women. Resistance to sulfoxide-pyrimethamine is common in many parts of the world. DNA barcoding can be used to identify mutations in the parasite’s Dhfr and Dhps genes that are associated with sulfoxide-pyrimethamine resistance.

**C. Machine Learning**

Machine learning is a way of learning from data that has been set up in a way that is supervised or directed. This means that a set of instructions is given to the machine learning algorithm, telling it what to look for and how to interpret it. Unsupervised learning is when the machine learning algorithm is left to its own devices, without any specific instructions. In machine learning, we use supervised learning to learn how to map input (X) to output (Y). This is done by having a set of input values and a set of output values and using algorithms to find a function that can map the input to the output. We want to be able to predict the output variable for given input data. In supervised learning, we give the machine lots of data that tells it what to do. With unsupervised learning, we just give the machine data [14].

The unsupervised learning method is used to model the underlying structure or distribution in the data so that it can be studied more closely [15]. In this research, we propose and evaluate the supervised method, such as KNN, Naïve Bayes, SVM, Random Forest, and advanced machine learning (deep learning) using LSTM and Bidirectional LSTM (Bi-LSTM).

**D. Deep Learning**

Deep learning is an enhanced artificial neural network (ANN) method with multiple-hidden layers. In principle, deep learning is a neural network with three or more layers of ANN, enabling it to learn and adapt to large amounts of data and to solve various problems that are difficult to solve by other machine learning algorithms [16]. As illustration, the architecture of deep learning is shown in Fig. 2.

Deep learning consists of several artificial neural networks that relate to each other. Some types of algorithms in deep learning including Convolution Neural Network (CNN), Recurrent Neural Network (RNN), and Self-Organizing Map (SOM).

1) **Recurrent neural network:** Recurrent Neural Network (RNN) is a neural network model which uses loops in its internal memory to solve sequential data problems [17]. The architecture of the RNN can be seen in Fig. 3.
Based on this architecture, each hidden layer of the RNN will receive input to the form of a vector denoted as Equation 1.

\[ h_t = \sigma(W_{sh} + W_{hb} h_{t-1} + b_h) \] (1)

where, \( W_{sh} \) is the weight matrix from the input to the hidden layer; \( W_{hb} \) is the weight matrix between two hidden states; \( b_h \) is a bias vector of hidden layer; \( \sigma_h \) is an activation function for getting hidden state. After getting the value \( h_t \), then the next process is to find \( y_t \) as in Equation 2.

\[ y_t = \sigma_h(W_{hy} h_t + b_y) \] (2)

where \( W_{hy} \) is weight matrix of hidden layer into output layer, \( b_y \) is bias vector of output layer; \( \sigma_y \) is activation function of output layer.

2) Long Short-Term Memory (LSTM): Long Short-Term Memory is one of the Recurrent Neural Network (RNN) architectures created to handle the problem of gradients that disappear during the back-propagation process [17]. LSTMs are special types of neural networks that can remember things for a long time. Each memory cell stores information by using an input gate (which lets in new information), a forget gate (which helps to forget old information), and an output gate (which sends out the stored information). The forget gate helps to forget things from the past, the input gate helps you transfer information into the cell, and the output gate helps you create new memories in the long term [18]. The input to the LSTM is a sequence of numbers \((x_1, x_2, \ldots, x_t)\). The output is a sequence of numbers \((y_1, y_2, \ldots, y_t)\). To find out how many forget gates there are, we use Equation 3. This equation tells the frequency of output will change in a particular gate.

\[ f_t = \sigma(W_f * [h_{t-1}, x_t] + b_f) \] (3)

3) Bidirectional Long Short Term-Memory (Bi-LSTM): Bidirectional Long Short Term-Memory is the LSTM method with two directions- a forward layer to remember information and a backward layer to remember how information was learned - to make it more efficient [19]. LSTMs are better at handling situations where the gradient of a neural network’s activation changes slowly over time, like in language processing. However, LSTMs only work with information from the past [20]. To fix this problem, the LSTM architecture needs to be changed so that it includes a forward LSTM layer and a backward LSTM layer. This will help the machine learn what happened in the past and make predictions for the future. As illustration of the architecture is shown in Fig. 4.

According to Cui et al.(2022), the forward layer’s output \( h_f \) is calculated using the forward input starting at \( 1 - T \), while the backward layer’s output \( h_b \) is calculated using the inverse of the backward layer’s input, namely \( T - 1 \) [21]. The gate that contains a Forward layer and a retrogressive layer can be refreshed involving a similar condition as a conventional LSTM. In the hidden state section of BiLSTM, the t-the data contains both the hidden state forward layer and the hidden state backward layer. This is the primary distinction. The equation can be used to combine BiLSTM hidden state.

\[ h_t = f(h_{tf}, h_{tb}) \] (4)

where \( h_{tf} \) is the hidden state of the forward layer, \( h_{tb} \) is the hidden state of the backward layer, and \( f \) is a function to combine the output of the hidden state forward layer and backward layer. The function \( f \) can be a concatenating function, a summation function, an average function, or a multiplication function.

E. Optimizer

An optimizer is an algorithm that helps to find the lowest possible global value for the convergence of a loss function in a neural network [17]. An optimizer can be used to find the best values for the model’s parameters [22]. This helps to improve the model’s accuracy. The optimizer can also help to reduce the model’s error value. Adaptive learning methods like Adam help us estimate how well we’re doing during a learning task. Adam is an optimization of an existing, effective stochastic algorithm that doesn’t require a lot of memory. Its streamlining agent calculation is planned by joining the strengths of the other two enhancers, Adagrad and RMSprop.

1) Adaptive Moment Estimation (Adam): Adaptive Moment Estimation (Adam) is one of the methods adaptive learning [18]. Adam is an optimization of an efficient stochastic algorithm that only requires a first order gradient where the memory required is very little [23]. Adam’s optimizer algorithm is designed by combining the advantage of the other two optimizers, Adagrad, which has performance pretty good with sparse gradients and RMSprop which works well at times solve non-stationary problems. Updating the weights on the optimizer Adam can be seen in Equation 5 to Equation 7.

\[ V_t = \beta_1 * V_{t-1} - (1 - \beta_1) * g_t \] (5)

\[ s_t = \beta_2 * s_{t-1} - (1 - \beta_2) * g_t^2 \] (6)
Stochastic Gradient Descent (SGD): Stochastic gradient descent is a way of finding the smallest possible error in a calculation. It begins by making small changes to the current value and then checking to see if the error has decreased [22]. If it has, the algorithm keeps making smaller changes until the error decreases no further. The SGD optimizer updates the weight of a model as in Equation 8.

$$W_{\text{new}} = W_{\text{old}} - \alpha \frac{g_t}{\sqrt{s_t + \varepsilon}} \cdot g_t$$  \hspace{1cm} (7)

$$w_{\text{new}} = w_{\text{old}} - (\alpha \cdot g_t)$$  \hspace{1cm} (8)

where gt is the gradient of the t time step, w is weight, and α is the learning rate.

III. THE PROPOSED RESEARCH METHOD

In general, this research is applied using Bi-LSTM as shown in Fig. 5. It was started by collecting DNA barcode data from several falciparum parasite DNA barcode sequences for anti-malarial drug resistance at Malaria GEN sites, including Artemisinin, Piperaquine, DHA-PPQ, Chloroquine, Pyrimethamine, Sulfadoxine, and S-P. The various drugs are comparable to the anti-malaria drug data set with any possibility misuse in regulation by patients. The drugs are composed in the type of antibiotic. The resistance prediction study includes preprocessing data, determining input data, developing the RNN method (Bi-LSTM algorithm), tuning parameters to optimize the Bi-LSTM method, and evaluating performance through measuring accuracy, precision, recall, and F1-score.

We used seven different datasets of drug with two classes, namely resistant and sensitive as shown in Table I. The data set is taken from MalariaGEN Plasmodium Falciparum Community Project which analyzed 7,113 parasite samples obtained at 73 different sites in Africa, Asia, America, and Oceania to generate genotyping data.

![Fig. 5. Concept diagram of drug-anti-malaria resistance prediction.](image)

By parsing the dataset, we got the DNA barcode sequence as shown in Fig. 6. The barcode sequence represents more than 100 nucleotides in each sample of plasmodium.

![Fig. 6. The sample of DNA barcode sequences.](image)

The stages of the proposed method are shown in Fig. 7. The design of an algorithm for predicting anti-malarial drug resistance against the plasmodium falciparum parasite is carried out by incorporating the DNA barcode sequence dataset into the model architecture. Before data sequence is entered, the dataset is encoded first in the stage preprocessing by splitting the dataset into training data and test data and perform encoding. The next step is the forming process architecture using the LSTM algorithm. Then, it is performed training model as well as the model validation process that has been formed to be carried out. After the model was trained, it was tested to evaluate the performance of the model in classifying drug resistance.

![Fig. 7. The Stages of proposed method.](image)

IV. RESULT AND DISCUSSION

Anti-malaria drug is a kind of antibiotic drug. Patient non-compliance in the use of anti-biotic drugs has an impact on resistance to pathogens in infecting a person’s body. Many

<table>
<thead>
<tr>
<th>Drugname</th>
<th>Resistant</th>
<th>Sensitive</th>
</tr>
</thead>
<tbody>
<tr>
<td>Artemisinin</td>
<td>3176</td>
<td>3787</td>
</tr>
<tr>
<td>Piperaquine</td>
<td>2141</td>
<td>1297</td>
</tr>
<tr>
<td>DHA-PPQ</td>
<td>1889</td>
<td>4154</td>
</tr>
<tr>
<td>Chloroquine</td>
<td>6402</td>
<td>430</td>
</tr>
<tr>
<td>Pyrimethamine</td>
<td>7378</td>
<td>120</td>
</tr>
<tr>
<td>Sulfadoxine</td>
<td>5967</td>
<td>1412</td>
</tr>
<tr>
<td>S-P</td>
<td>4553</td>
<td>1729</td>
</tr>
</tbody>
</table>
studies of drug resistant for anti-malaria were conducted and various attempts have been made to treat this disease [11]. DNA barcoding can be used to study the resistance of anti-malaria drugs. DNA barcoding is a technique that involves sequencing a short, standardized region of DNA from a particular organism. In the case of malaria, DNA barcoding can be used to recognize the specific species of the malaria parasite present in a patient’s blood [24].

The genetic sequence of the malaria parasite: It provided information about its susceptibility to different anti-malaria drugs. Mutations in the parasite’s DNA can result in resistance to certain drugs, and DNA barcoding can be used to detect these mutations. In addition to identifying resistance to specific drugs, DNA barcoding can also be used to monitor the spread of drug-resistant malaria parasites. By sequencing the DNA of malaria parasites from different regions, researchers can track the movement of drug-resistant parasites and identify areas where resistance is becoming more prevalent. Overall, DNA barcoding is a valuable tool for studying the resistance of malaria drugs and developing strategies to combat drug-resistant malaria. The attribute values distribution of data set is plotted into boxplot as shown in Fig. 8. The graph shows that the variation in attribute values is distributed in the range between 1 and 7 and the first to third quartiles are described. The presence of a vertical line through this plotted box corresponds to the median distribution of the partial data. The boxplot is illustration of statistical analysis for data distribution [25]. They are range value of DNA-barcode sequences after encoding in numerical values.

![Boxplot of encoding barcode DNA sequence dataset distribution for training and testing data.](image)

This research is developed using the Python programming language, and Google Collaboratory as a Python text editor. We assessed the grouping model by ascertaining accuracy, precision, recall, and the F1-score to determine the quality of result. The number of detections is True Positive (TP), False Negative (FN), False Positive (FP), and True Negative (TN) was calculated as follows to calculate these measurements. The number of tests (drug resistance accurately) is TP. The number of tests incorrectly identified a drug resistance is called FP. The number of tests that were incorrectly classified as sensitive is referred to in FN. Then again, TN is the degree of tests that were precisely settled to be solid. Additionally, a five k-fold cross-validation method was utilized to ensure the quality of the classification model for detection.

As a measure of correctness, accuracy is defined as the ratio of the number of corrected forecasts to the total number of forecasts, as shown in Equation (9). By then, the pace of positive conjectures that are exact is referred to as precision as in Equation (10). As shown in Equation (11), Recall is the total number of positive cases that the classifier correctly anticipated based on all the information. Sometimes it’s referred to as affect ability. The F1-score is the final performance metric. A degree that combines review and precision could be the score as shown in Equation (12) [26].

\[
\text{accuracy} = \frac{TP+TN}{TP+TN+FP+FN} \quad (9)
\]

\[
\text{precision} = \frac{TP}{TP+FP} \quad (10)
\]

\[
\text{recall} = \frac{TN}{TN+FP} \quad (11)
\]

\[
f1 – score = 2 \times \frac{\text{precision} \times \text{recall}}{\text{precision} + \text{recall}} \quad (12)
\]

The representative machine learning methods including K-Nearest Neighbor, Naive Bayes, SVM, Decision Tree, and the deep learning methods including, RNN, LSTM, and Bi-LSTM are used to evaluate the performance measures including accuracy, precision, recall, and f1-score. The performance results for each algorithm described into the heat map in Fig. 9. All methods have high performances evaluation for two kinds of anti-malaria, such as Dha-Ppq, and Piperaquine. However, the conventional machine learning methods have low in unbalanced class of data sets. The distribution of class in each data set can be seen in Fig. 10.

Based on the experimental results, we can see that most of the classic machines learning models have low performance rates in unbalanced class distribution as shown in Table II and Table III. A classifier whose score is less than 0.5 has a high proportion of false negatives. It can be caused by an imbalanced class or the number data in each class is imbalanced. It means that classic machine learning model can’t overcome imbalanced class problem like deep learning. The over-sampling or under-sampling strategy, we attempt to change the amount of the minor class to obtain a balanced proportion for both classes. A potential solution may help to resolve the imbalanced class problem for the traditional machine learning model Random Over-sampling, Synthetic Minority Oversampling Technique (SMOTE), Adaptive Synthetic Sampling (ADASYN), Random Under-sampling, etc. are some over-sampling and under-sampling techniques that are frequently utilized. Another solution that we can try is to give higher weight to the minor class. A class has higher weight than the other class [27]. By adjusting the cost function of the model, class weighting ensures the misclassified observations. The minority class tends to decide the majority
class. By rebalancing the class distribution, this strategy can assist in enhancing the model’s accuracy.

Fig. 9. Comparison of performance evaluation for various drugs including A. Artemisinin, B. Chloroquine, C. Dha-Ppq, D. Piperaquine, E. Pyrimethamine, F. S-P.G.Sulfoxides.

Fig. 10. Distribution of class in each data set.

**TABLE II. PERFORMANCE RATE OF CHLOROQUINE**

<table>
<thead>
<tr>
<th>Model</th>
<th>Accuracy</th>
<th>Recall</th>
<th>Precision</th>
<th>F1 Score</th>
</tr>
</thead>
<tbody>
<tr>
<td>BiLSTM</td>
<td>0.94</td>
<td>1.00</td>
<td>0.94</td>
<td>0.97</td>
</tr>
<tr>
<td>LSTM</td>
<td>0.94</td>
<td>1.00</td>
<td>0.94</td>
<td>0.97</td>
</tr>
<tr>
<td>RNN</td>
<td>0.94</td>
<td>1.00</td>
<td>0.94</td>
<td>0.97</td>
</tr>
<tr>
<td>DecisionTree</td>
<td>0.93</td>
<td>0.51</td>
<td>0.44</td>
<td>0.47</td>
</tr>
<tr>
<td>KNeighbors</td>
<td>0.96</td>
<td>0.58</td>
<td>0.67</td>
<td>0.62</td>
</tr>
<tr>
<td>GaussianNB</td>
<td>0.79</td>
<td>0.56</td>
<td>0.16</td>
<td>0.25</td>
</tr>
<tr>
<td>SVC</td>
<td>0.95</td>
<td>0.13</td>
<td>1.00</td>
<td>0.23</td>
</tr>
<tr>
<td>RandomForest</td>
<td>0.96</td>
<td>0.36</td>
<td>1.00</td>
<td>0.53</td>
</tr>
</tbody>
</table>

**TABLE III. PERFORMANCE RATE OF PYRIMETHAMINE**

<table>
<thead>
<tr>
<th>Model</th>
<th>Accuracy</th>
<th>Recall</th>
<th>Precision</th>
<th>F1 Score</th>
</tr>
</thead>
<tbody>
<tr>
<td>BiLSTM</td>
<td>0.98</td>
<td>1.00</td>
<td>0.98</td>
<td>0.99</td>
</tr>
<tr>
<td>LSTM</td>
<td>0.98</td>
<td>1.00</td>
<td>0.98</td>
<td>0.99</td>
</tr>
<tr>
<td>RNN</td>
<td>0.98</td>
<td>1.00</td>
<td>0.98</td>
<td>0.99</td>
</tr>
<tr>
<td>DecisionTree</td>
<td>0.97</td>
<td>0.44</td>
<td>0.18</td>
<td>0.26</td>
</tr>
<tr>
<td>KNeighbors</td>
<td>0.99</td>
<td>0.50</td>
<td>0.47</td>
<td>0.48</td>
</tr>
<tr>
<td>GaussianNB</td>
<td>0.85</td>
<td>0.63</td>
<td>0.04</td>
<td>0.08</td>
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<tr>
<td>SVC</td>
<td>0.99</td>
<td>0.13</td>
<td>1.00</td>
<td>0.22</td>
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<tr>
<td>RandomForest</td>
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<td>0.25</td>
<td>1.00</td>
<td>0.40</td>
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**TABLE IV. PERFORMANCE RATE OF ARTEMISININ**

<table>
<thead>
<tr>
<th>Model</th>
<th>Accuracy</th>
<th>Recall</th>
<th>Precision</th>
<th>F1 Score</th>
</tr>
</thead>
<tbody>
<tr>
<td>BiLSTM</td>
<td>0.83</td>
<td>0.79</td>
<td>0.82</td>
<td>0.80</td>
</tr>
<tr>
<td>LSTM</td>
<td>0.81</td>
<td>0.63</td>
<td>0.92</td>
<td>0.75</td>
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<tr>
<td>RNN</td>
<td>0.79</td>
<td>0.74</td>
<td>0.79</td>
<td>0.76</td>
</tr>
<tr>
<td>DecisionTree</td>
<td>0.92</td>
<td>0.91</td>
<td>0.95</td>
<td>0.93</td>
</tr>
<tr>
<td>KNeighbors</td>
<td>0.93</td>
<td>0.91</td>
<td>0.96</td>
<td>0.93</td>
</tr>
<tr>
<td>GaussianNB</td>
<td>0.90</td>
<td>0.96</td>
<td>0.87</td>
<td>0.91</td>
</tr>
<tr>
<td>SVC</td>
<td>0.95</td>
<td>0.96</td>
<td>0.95</td>
<td>0.96</td>
</tr>
<tr>
<td>RandomForest</td>
<td>0.97</td>
<td>0.97</td>
<td>0.97</td>
<td>0.97</td>
</tr>
</tbody>
</table>

Furthermore, the high-performance scores were obtained for the Bi-LSTM algorithm under various data conditions (balanced or unbalanced) when compared to the other deep learning methods. In Table IV, the performance of Bi-LSTM for Artemisinin data set with balanced class distribution, can provide consistent and reliable results. Even though the conventional machine learning algorithms achieved higher performance than Bi-LSTM method due to it also influenced by several factors such as the size of the training data, the number of layers and neurons in the network, the type of activation function used, and the parameter tuning performed. Therefore, it is important to conduct a thorough evaluation and determine optimal parameters for each case of Bi-LSTM. The proposed method is applied to various optimizers and various numbers of hidden layers. By tuning hyper-parameter is addressed to improve classifier model for prediction [28]. Then, the lowest error rate is achieved at Adam optimizer with 32 hidden layers as shown in Table V.

Cross entropy is a loss function that serves to measure performance of the classification model. The cross entropy will increase if probability predictions increasingly deviate from the actual class. The lower the cross-entropy, the better is the model [29]. The value of the loss function appears to be homogeneous in deep learning algorithms including Bi-LSTM and LSTM as shown in the Fig. 11, Fig. 12, and Fig. 13.

Furthermore, the computational time of deep learning method is higher than the classical machine learning method due to complexity in classifier model as shown in Fig. 14. The Bi-LSTM algorithm is the highest computational time.

**TABLE V. ERROR RATE OF TUNING HYPER-PARAMETER**

<table>
<thead>
<tr>
<th>LSTM#</th>
<th>Num of h_layers</th>
<th>Adam opt</th>
<th>RMSprop</th>
<th>SGD</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>8</td>
<td>0.0092005</td>
<td>0.0077283</td>
<td>0.0397047</td>
</tr>
<tr>
<td>1</td>
<td>16</td>
<td>0.0103672</td>
<td>0.0051495</td>
<td>0.0341338</td>
</tr>
<tr>
<td>1</td>
<td>32</td>
<td>0.0015641</td>
<td>0.009913</td>
<td>0.0276439</td>
</tr>
<tr>
<td>2</td>
<td>8</td>
<td>0.0085866</td>
<td>0.0057495</td>
<td>0.0699882</td>
</tr>
<tr>
<td>2</td>
<td>16</td>
<td>0.0034778</td>
<td>0.0007774</td>
<td>0.0567839</td>
</tr>
<tr>
<td>2</td>
<td>32</td>
<td>3.450E-05</td>
<td>4.86E-05</td>
<td>0.0569096</td>
</tr>
<tr>
<td>3</td>
<td>8</td>
<td>0.0047429</td>
<td>0.0019380</td>
<td>0.0562194</td>
</tr>
<tr>
<td>3</td>
<td>16</td>
<td>0.0014209</td>
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<td>0.0571575</td>
</tr>
<tr>
<td>3</td>
<td>32</td>
<td>7.5638E-05</td>
<td>0.0001729</td>
<td>0.0476583</td>
</tr>
</tbody>
</table>

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Abstract—Developing trust in cyber-physical systems (CPSs) is a challenging task. Trust in CPS is needed for carrying out their intended duties and is reasonably safe from misuse and intrusion; it also enforces the applicable security policy. As an example, medical smart devices, many researches have found that trust is a key factor in explaining the relationship between individual beliefs about technological attributes and their acceptance behavior; and have associated medical device failures with severe patient injuries and deaths. The cyber-physical system is considered a trust system if the principles of security and safety, confidentiality, integrity, availability, and other attributes are assured. However, a lack of sufficient analysis of such systems, as well as appropriate explanation of relevant trust assumptions, may result in systems that fail to completely realize their functionality. The existing research does not provide suitable guidance for a systematic procedure or modeling language to support such trust-based analysis. The most pressing difficulties are achieving trust by design in CPS and systematically incorporating trust engineering into system development from the start of the system life cycle. Still, there is a need for a strategy or standard model to aid in the creation of a safe, secure, and trustworthy CPS. Model-based system engineering (MBSE) approaches for trust cyber-physical systems are a means to address system trustworthiness design challenges. This work proposes a practical and efficient MBSE method for constructing trust CPS, which provides guidance for the process of trustworthiness analysis. The SysML-based profile is supplied, together with recommendations on which approach is required at each process phase. The MBSE method is proven by expanding the autonomous car SysML and UML diagrams, and we show how trust considerations are integrated into the system development life cycle.

Keywords—Cyber Physical Systems (CPSs); trust CPS; system engineering (SE); model-based system engineering (MBSE); SysML

I. INTRODUCTION

The emergence of cyber-physical systems (CPSs) has an impact on people's lives and is used in health care, smart homes, commerce, and other areas. Engineering CPS necessitates a combination of approaches from several fields (mechanical, electrical, biological, etc.)[1] and computer science methods.

The cyber-physical system is considered a trust system if the principles of security and safety, confidentiality, integrity, availability, and other attributes are ensured. They defined trust in [2] as a measure of confidence or belief that the other party will refrain from opportunistic behavior and behave in an expected manner. The trust CPS is a system that meets a set of requirements and obligatory attributes that ensure the trustworthiness of CPS [3].

Inattention to trustworthiness can result in the loss of human life, long-term environmental implications, the disruption of essential infrastructure, or other catastrophic consequences such as the revelation of sensitive data, the destruction of equipment, economic loss, and reputational harm. These dangers and negative outcomes become more severe as industries become more networked and integrated.

In the literature, many directions are suggested for modeling trust in systems. However, does not provide suitable instructions for a systematic methodology or an acceptable modeling language to support such trust-based analysis. The most significant problems are achieving trust by design in CPS and systematically incorporating trust engineering into system development from the start of the system life cycle.

According to [4], a user-centered holistic technique is described for analyzing trust as the system is being developed. The process consists of five steps (the scenarios, trust analysis, peer review, scenarios refinement, guiding the design). Other lines of research target improving system trustworthiness in terms of security. They consider trust an enabler of security, and security services rely to a great extent on some notion of trust [5]. In [6], a trust case is presented with a complete and explicit argument for trust in the system under development in terms of security and safety. They used the Claim Definition Language (CDL). For trustworthy computing [7], it employs the Security Development Lifecycle (SDL) for the development of software that must withstand attacks. UMLsec [8] is a Unified Modeling Language (UML) extension and it is a UML profile that allows designers to define security features on design models. To establish the security requirements and assumptions on the system's environment, typical UML extension techniques in the form of labels, such as stereotypes and tags, are used. Trust cases that focus on the security and safety of the system are described in [6], excluding additional trust attributes such as privacy and usability, and it enables the identification of the components that are engaged in trust assumptions.

The specific challenges and limitations of existing research are about identifying trust concerns and requirements that may apply to many parties, as well as focusing purely on security quality and viewing trust as an enabler of security and security services while being unconcerned about the remaining trust
attributes; furthermore, selecting modeling approaches that can handle the complexity of trust in CPS.

Not defining a cyber-physical system's tasks. Not specifying the exact characteristics of trust in CPS. Not defining the targets of the actors and participants in the CPS, including developers, users, and potential customers. Not defining goals, especially business goals. The unpredictability of the behavior of the system component, all those are the major challenges to the development of trust in cyber-physical systems, particularly during the modeling stage.

In this work, we examine specific strategies, models, and tools that encompass design activities and preparation for modeling the trustworthiness of CPSs, as well as how these techniques can be expressed in Unified Modeling Language (UML) and SysML models. The MBSE approach and phases are discussed. By applying the autonomous vehicle SysML and UML diagrams, the MBSE method is validated.

The rest of the paper is organized as follows: Section II presents the contribution and research methodology. Model based system engineering (MBSE) is discussed in Section III. Proposed MBSE Trust CPS Method is introduced in Section IV. Discussion is given in Section V. We conclude the paper in Section VI.

II. CONTRIBUTION AND METHODOLOGY
This work aims to:

- Analyze specific approaches, models, and tools pertaining to design processes and preparation for modeling the trustworthiness of CPS.
- Create a viable and efficient MBSE method for developing secure systems while adhering to trust constraints.
- Ensure the system's trustworthiness characteristics are created in accordance with end-user trust requirements.

Research Question:

RQ1 - How Infusing trust attributes in CPS into systems engineering practice to decrease the complexity and increase the quality and trustworthiness of CPS?

RQ2 - What are the methods and tools shall be used for modeling trust CPS?

Our methodology was based on the analysis of fund papers from publications and standards and the definition of requirements and mandatory attributes using system engineering and requirements approaches. Model-Based System Engineering (MBSE) for Trust CPS is a combination of disciplines and approaches, and methods. In Fig. 1, the research method is displayed.

III. MODEL BASED SYSTEM ENGINEERING (MBSE)
Model-Based Systems Engineering (MBSE) is a tool used to run simulations, support system engineering operations, and improve requirements throughout process development. MBSE offers various advantages, including increased mapping, traceability, and system decomposition [9], improved communication and system management [10], complexity management and risk reduction [11], and systemized decision making [12].

MBSE is a modeling formalism used to assist system requirements, design, analysis, verification, and validation activities, beginning with the conceptual design phase and progressing through the development and subsequent life cycle phases [13, 14, 15]. According to [16], the use of MBSE models reduces project risks and timelines, reduces costs, and improves product quality. Additionally, the MBSE approach's primary idea of using simulation models assures that results are acquired faster and more affordably than testing and prototyping. MBSE has become an essential component of designing complex cyber-physical systems [17,18], is popularly used [19] for: a) Capture and manage the system’s requirements, architecture, design, and identify its environment. b) Ease the communication among many stakeholders by participating and providing views for different purposes. In [20], they stated that the benefit of MBSE activities is system validation and verification throughout the early stages of system design. The use of MBSE is limited to object modeling via the Systems Modeling Language (SysML) [21].

IV. PROPOSED MBSE TRUST CPS METHOD
This section presents the MBSE technique, which includes activities and principles for developing trust CPS. Our goal with MBSE is to obtain descriptive models that leverage semantically rich modeling standards to provide systems abstraction, data traceability, and separation of views. The benefits of using MBSE are explained in previous section. The main goal is: How infusing trust’s attributes of CPS into systems engineering practice to decrease the complexity of trustworthiness.
To produce intuitive system descriptions, MBSE requires the synthesis of stakeholder needs into architecture models. A trust CPS technical requiring multiple disciplines to design, for this reason, our proposed MBSE method for trust cyber-physical system is based on many disciplines such as system engineering that is the pillar of MBSE, requirement definition, modeling, verification as presented in Fig. 2 (source: compiled by authors) that details the definition of Phases, activities in phases and defining used methods.

A. Phases

1) Phase 1 (using system engineering and analyzing engineering elements, tasks and goals): Top-down synthesis is an iterative approach used in systems engineering [22]. SE is a discipline that focuses on the overall (system) design and implementation, taking into account all the facets [23].

a) System definition: Define integrated elements of CPS (Subsystems, Product, Hardware, Software, Firmware, Network). The exterior and internal views of the system introduced the elements that belong to the system and interact with it. The functionality of system is expressed by the interaction of system with its operating environments. We recommend using the SysML Block Definition Diagram or UML Component Diagram.


c) Define system activities (we recommend the use of SysML Activity Diagram).

d) Define set of goals: systems and stakeholders, user.

2) Phase 2 (Engineering Requirement)

a) Identify trust requirements: Engineering requirements (RE) is a methodical, disciplined strategy for defining and managing needs [25]. In the design of software and cyber-physical systems, requirements are frequently the most critical step. They serve as a guideline for implementation as well as a reference for final product verification and validation.

During the requirements analysis stage, we are concentrating on trust modeling. We seek to integrate trust attributes in the early stages of trust CPS creation by combining a CPS cycle life with a set of trustworthiness requirements. Our primary goal is to create a model that may aid the CPS designer throughout the complete CPS design process, utilizing the information available in the requirements specification and the availability of necessary characteristics to test and monitor the final implementation's behavior. The standard SysML requirements diagram is recommended. They described a basic need for trust CPS in [3] as follows:

- Engineering requirements: For determining goals, needs, and available resources to aid in the development of a system that meets the requirements.
- Cyber-physical system requirements: Understanding the aim, specifications, and pre-designated purpose of CPS, as well as accumulating a huge number of system information.
- Trust (as a system quality) requirements: collect system, organization, and user trust needs.
- Business Requirement

b) Defining mandatory attributes for trust CPS: This activity is carried out during the requirements engineering
phase of CPS design. It uses objective and business process models to examine and refine trustworthiness requirements.

The cyber-physical system is considered a trust system if the principles of security and safety, confidentiality, integrity, availability, and other attributes are assured. As previously stated [3] secure, trustworthy, and trusted are system qualities that differ in terms of concerns, requirements, properties, and human interaction. However, there is a link and connection in that each quality can be complimentary to the other by adding characteristics and features to it (security and safety, in addition to a set of attributes, are both necessary to a sufficient degree to build a trustworthy cyber-physical system)[3]. The ability of a secured system to be trusted or trustworthy to some extent is a feature of the system in context.

Safety and security are interdependent, and these dependencies should be considered at the CPS design phase. Interdependence can be classified into four types: a) conditional dependencies: safety is a prerequisite for security, and vice versa; b) reinforcement: safety and security countermeasures can support each other; c) antagonism: they can undermine each other; and d) independence: there is no relationship between safety and security.

In [3] they presented an attribute classification based on system functionality and obligation, as well as a set of judgment criteria. As shown in Table I, this classification of CPS characteristics and quality of service covers the most significant indicators for CPS security and trustworthiness. The terminology glossary [26] defines the majority of the attributes. Some necessary attributes are dependent on the precision and service quality of stakeholders or consumers.

3) Phase 3 (Identification of risk and threats and mitigating controls): CPS should be constructed with controls for threats that may compromise its functions. During this step, the set of assets, threats, or controls should be determined. Risk management is based on norms in industries, and some companies have their own set of procedures and standards.

a) Risk identification: NIST defined a threats as a potential risk to a computer system that could lead to the interruption of the system or the interception, manipulation, obstruction, or destruction of computational resources.

- Security threats: CPS security deals with attacks and mostly attacks threaten the security attributes. A denial of service (DoS) attack, for example, renders a control system's components unreachable in some or all circumstances. This DoS can infect devices and prevent them from transferring data, as well as attack routing protocols and communication channels [34]. The common attacks on cyber-physical systems are attacks on data integrity that include false data injection attacks. Integrity is compromised when an attacker deletes or edits crucial data unintentionally or maliciously, leading the receivers to believe the modified or deleted data is accurate. Integrity in CPS could be defined as the ability to achieve physical goals by avoiding, detecting, or defeating attempts to tamper with data transmitted and received by sensors, actuators, and controllers [35]. Some attacks are presented in Table II.

<table>
<thead>
<tr>
<th>Attributes</th>
<th>Definition</th>
<th>Sub-attributes</th>
</tr>
</thead>
<tbody>
<tr>
<td>Safety [27][28]</td>
<td>The ability to operate without causing danger or harm to people or the system's environment.</td>
<td>fault tolerance, robustness</td>
</tr>
<tr>
<td>Security [29][30][31]</td>
<td>A software and hardware system's ability to protect entities from attacks and misuse as well as to safeguard resource access is referred to as security.</td>
<td>availability, accountability, auditability, assurance, traceability, integrity, confidentiality, non-repudiation</td>
</tr>
<tr>
<td>Compatibility [32]</td>
<td>The ability of software and hardware from multiple sources to work together without having to be modified.</td>
<td>Openness</td>
</tr>
<tr>
<td>Performance [32][33]</td>
<td>Describe the effectiveness of a service. The quantity of event responses handled within an interval is referred to as throughput. Response Time: The amount of time it takes for the service to complete a single transaction.</td>
<td>accuracy, availability, robustness, reliability, scalability, maintainability</td>
</tr>
<tr>
<td>Dependability [31][32]</td>
<td>That ability can be justified and placed on the service it provides. The anticipated execution was correct and predictable.</td>
<td></td>
</tr>
<tr>
<td>Privacy [27][33]</td>
<td>The system's capability and functionality that allows users to govern the use of their personal information or data.</td>
<td>No one else can access or utilize their personal information.</td>
</tr>
<tr>
<td>Usability [ISO 9241-210][33]</td>
<td>Is a set of attributes that can be meet in same time for unique product or system. It refers to the ease with which a user can learn to operate, prepare input for, and interpret the output of the service. Positive attitudes towards the use of the service.</td>
<td>satisfaction, learnability, effectiveness, efficiency of use</td>
</tr>
<tr>
<td>Correctness [32]</td>
<td>Correctness refers to whether a system behaves in a way that satisfies user needs, especially those related to trust expectations and trustworthiness requirements.</td>
<td></td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Attack</th>
<th>Paper</th>
</tr>
</thead>
<tbody>
<tr>
<td>Denial of service (DoS)</td>
<td>[36]</td>
</tr>
<tr>
<td>False data injection</td>
<td>[37]</td>
</tr>
<tr>
<td>Covert attack</td>
<td>[38]</td>
</tr>
<tr>
<td>Generic deception</td>
<td>[39]</td>
</tr>
<tr>
<td>Eavesdropping</td>
<td>[40]</td>
</tr>
<tr>
<td>Packet scheduling attack</td>
<td>[41]</td>
</tr>
<tr>
<td>Load redistribution attack</td>
<td>[37]</td>
</tr>
</tbody>
</table>
• Safety threats: Concerns about safety arise from interactions between the environment and the CPS, within the CPS, and between the CPS and authorized users. CPS safety is related to the CPS's ability to assure the absence of catastrophic repercussions on the lives, health, property, or data of stakeholders and the physical environment. Safety is an essential concern that affects process industries, and IEC 61508 is regarded as a basic safety standard that applies to all industries. In [42], they discussed how to improve safety awareness and demonstrated that individuals' safety, actions, and ability to deal with dangers at work are heavily influenced by their consciousness and behaviors.

• Trustworthy threats: Some threats are connected to privacy, which is the ability of the CPS to prohibit entities (people, machines) from having access to data stored in, created by, or transiting a CPS or its components. Privacy is the system's ability and functionality that allows users to regulate the use of their private information or personal data. It is a significant contributor to the system's trustworthiness. Privacy attacks are a type of passive assault that target data collecting and can be used to leak sensitive information and reduce the visibility and control of the user over his private information. Mostly attacks threat the trustworthy attributes. The denial of service (DoS) threats availability that is the readiness for usage, and the reliability that is the continuity of service [43]. The attacks result service delay when companies delay providing services on time due to the problems in the system. Unauthorized users or hackers may gain access to specific data and extract confidential information [44].

• Trust concerns:
  - Usability concerns relating to CPS's capacity to be used successfully to meet functional objectives and user satisfaction (adapted from ISO 9241-210) Meeting usability requirements becomes more difficult when physical and cyber components are combined in complicated systems.
  - Correctness involves system behavior in compliance with user needs, including user trust expectations and trustworthiness criteria.

  b) Defense: CPS security corresponds to defense against attacks. The defense strategies is based on protection and detection, and mitigation. The prevention led to reducing attacks [45]. Several study in literature presented protection-based approaches and discussing many approaches against CPS attacks such as: Control [46], protection-based[47], security metrics[48], state estimation[49].

• Protection
  - Functional Protection: Functional security safeguards the software system against harmful, infiltrating code from both the outside and inside the company.

- Information protection: Information security safeguards the confidentiality, integrity, and availability (CIA) of computer system data and functionality against unauthorized and harmful access.

- Safety Protection: Protection against faults, errors, and failures, damage to life, health or society, or injury to the environment. Fault-tolerance, availability, and fail-safe states are examples of safety quality sub-attributes.

• Control: This part consist of deciding trust objectives and controlling security and trustworthiness of CPS.

- Controlling CPS security: In [50], authors categorized three security defense mechanisms and stated that: a) prevention is used to delay the onset of an attack; b) resilience to close the operation of the attack; c) detection to identify the source of the attack; and d) isolation of corrupted subsystems and speedy restoration of normal mode. The defense plan should rely on three mechanisms to avoid: a) the period between the commencement of the attack and discovery, which results in system damage. b) inability to protect against spoofing attacks. An example of a failure to detect [51]. Some control strategies in the literature, such as (observer-based techniques, watermarking, baiting, and learning-based anomaly detection), were categorised in [50]. The authors of [52] proposed an authentication strategy to secure the integrity of devices and utility servers and to avoid tampering attempts using cryptography techniques.

- Controlling CPS trustworthiness: The major tasks for measuring system quality and trust attributes are the identification of threats that may occur and affect user trust, as well as corresponding controls that may be undertaken to minimize the threats. The use of the risk-based method to identify threats to trustworthiness on an abstract level and computational approaches to evaluate end-to-end system trustworthiness in terms of several trustworthiness metrics as an example of trustworthy evaluation in design time [53].

• Vulnerability mitigation: The following are the most commonly used techniques and controls [54]:
  - Tamper resistance controls on field devices
  - Trusted procurement procedures
  - Patching and updating
  - Encryption
  - Penetration testing and internal audit
  - Network segmentation
  - Use of different technologies

4) Phase 4 (Modeling and Design): In literature, there is a variety of modeling methods and languages for CPS and its
quality. The integration of models that capture software and computational behavior with the physical environments is a challenged task along with the inability to integrate discrete-event and continuous-time modeling paradigms for improving the ability to provide trustworthy CPSs in the future [55,56].

Architectural modeling tools for CPS are frequently used to depict full systems, including graphical notations such as SysML and UML that are useful for considering how the CPS is organized, as well as how the constituent elements interact and share data. Continuous-time modeling paradigms and discrete-event modeling paradigms are the two most well-known modeling methodologies. Modeling techniques that rely on mathematical notations are capable for representing continuous-time behaviors. Continuous-time modeling is required for the creation of a physics model capable of precisely predicting a system's interactions with its physical environment. It captures dynamic behavior of a system by utilizing iterative methods of integration and differential equations [57]. When it comes to physical processes and analog circuits, it used continuous-time techniques [58].

Some systems modeling methodologies and tools for CPS design and analysis include hybrid discrete-event and continuous simulations [59], inductive constraint logic programming [60], hybrid timed automaton [61], ontologies [62], information schema [63], UML [64], and SysML [65], and information dynamics modeling [66], meta-model for multimedia software architecture (MMSA) that enables the description of software architectures [67], SMSA (Security Meta-model for Software Architecture) [68], trustworthy collaboration [69].

As languages that have been designed for modeling holistic embedded systems and CPSs: Stateow/Simulink, Modelica [70], hybrid CSP [71], and HyVisual [72]. Comodeling (collaborative modeling) is an approach that focuses on creating system models composed of separate models [73]. An approach cosimulation engine called Crescendo[57],

In previous phases, we explained that a trust CPS technical requiring multiple disciplines to design which have be combined. For trust CPS modeling, we attempted to capture customer objectives and requirements about trust and reflect them in system functioning. Our suggested activities and tools for MBSE of trust CPS are presented in Table III.

In Fig. 3, our profile diagram is presented. The SysML profile is used on package to incorporate stereotypes. It depicts the MBSE method's phases and underlying trust methods. The MBSE suggests several tasks, including (SysML Block Definition Diagram, SysML Requirement Diagram, SysML Activity Diagram, SysML Use Case Diagram). UML’s diagrams can be used also for the suggested activities [74].

5) Phase 5 (Verification and Step by step test): In this phase a set of verification (elements verification, requirements verification, attributes and qualities verification) is done and operated in parallel with pervious phases.

A test step by step is applied to ensure the quality of modeling and level by level verification to ensure the design of entire system as mentioned in Fig. 4. The major objective is to conduct more analysis to see if the system satisfies trust qualities and allows for quick feedback on requirements and design choices.

System requirements are verified against the stakeholder and customer requirements and in the line with set of trust requirements. This step may results change or delete of requirements. The second level of verification targets system realization. The third level targets the use of system. A combination of test plan with test case and scenarios can be applied.

TABLE III. SOME ACTIVITIES IN MBSE

<table>
<thead>
<tr>
<th>Activities</th>
<th>Implementation in SysML</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td>Trust Requirements Definition</td>
<td>SysML Requirement diagram,</td>
<td>Identifies both functional</td>
</tr>
<tr>
<td></td>
<td></td>
<td>and non-functional trust</td>
</tr>
<tr>
<td></td>
<td></td>
<td>criteria</td>
</tr>
<tr>
<td>Trust Structure Definition</td>
<td>SysML Block Definition Diagram,</td>
<td>Defines system components</td>
</tr>
<tr>
<td></td>
<td></td>
<td>as well as their contents</td>
</tr>
<tr>
<td></td>
<td></td>
<td>(attributes, Behaviors,</td>
</tr>
<tr>
<td></td>
<td></td>
<td>Constraints), interfaces,</td>
</tr>
<tr>
<td></td>
<td></td>
<td>and relationships.</td>
</tr>
<tr>
<td>Security Constraints</td>
<td>SysML Block Definition Diagram,</td>
<td>Captures policies</td>
</tr>
<tr>
<td>Definition</td>
<td></td>
<td>pertaining to trust.</td>
</tr>
<tr>
<td>Trust Processes</td>
<td>SysML Activity Diagram</td>
<td>Determines trust controls</td>
</tr>
</tbody>
</table>

![Profile Diagram MBSE Trust CPS](image-url)

Fig. 3. The MBSE trust profile.
B. Case Study: Can we trust autonomous vehicles (AVs) on the road?

One of the largest system engineering challenges to date is the development of autonomous vehicles (AVs). For demonstrating the proposed MBSE method usage, we applied the autonomous vehicle (AV) that enables the preparation for MBSE and presents an initial guideline for establishing the trustworthiness of AVs.

1) Trust AVs: Trust is the first foundation for acceptance of AV and plays a critical part in fostering the relationship between the user and automation and increasing people’s desire to use or interact with it [75]. Different surveys showed that people were fascinated by AV but hesitated to trust it. In [76], they discovered that 65% of their participants were concerned about the dependability of self-driving cars. According to [77], 43% of participants are terrified about driving in an autonomous vehicle. A survey [78] found that 22% of respondents couldn’t envisage riding in a fully automated car. The authors acknowledged in the quoted polls that the majority of participants had not yet experienced any automated driving functions, making a true judgment difficult.

AV has six levels of autonomy, with the human driver monitoring the driving environment from L0 to L2, and the automated driving system monitoring the driving environment from L3 to L5 [79]. Trust is a key aspect in the evaluation of autonomous systems and influences user behavior [80, 81], and a supportive user interface is vital, particularly during the transition period to automated driving, when the “driver” must relinquish control in favor of an unfamiliar feature.

2) MBSE for trust AVs: We are applying MBSE method to two key aspects of developing trust systems: (1) ensuring trust through the use of requirements set that related to the interactor’s needs, and (2) infusing trust attributes into systems engineering practice.

The UML Component Diagram for AV is presented in Fig. 5.

a) Component diagram for AV: The first phase of system engineering and analyzing engineering elements suggests system definition and integrated elements of CPS. The important components of AV are presented in Table IV.

<table>
<thead>
<tr>
<th>Main Components</th>
<th>Electronic Control Unit</th>
<th>Sensors/Actuators System</th>
</tr>
</thead>
<tbody>
<tr>
<td>Central Gateway</td>
<td>Electronic engine control</td>
<td>Brake System</td>
</tr>
<tr>
<td>Vehicle control unit</td>
<td>Airbag Control Unit</td>
<td>Electric Power Pack</td>
</tr>
<tr>
<td>Driver assistance system domain controller (DASY)</td>
<td>ESP Unit</td>
<td>Vehicle Motion and position sensors</td>
</tr>
<tr>
<td>Information domain computer</td>
<td>Electronic Immobilizer</td>
<td>Ultrasonic sensor</td>
</tr>
<tr>
<td>V2X Connectivity control unit</td>
<td>Steering Control Unit</td>
<td>Near range camera sensor</td>
</tr>
<tr>
<td>Body Computer Module</td>
<td></td>
<td>Mid-Range Radar sensor</td>
</tr>
<tr>
<td></td>
<td></td>
<td>Multi-Purpose Camera</td>
</tr>
</tbody>
</table>
Fig. 5. UML component diagram for AV.

Fig. 6. SysML requirement diagram (General set of trust requirements for the AVs).

b) **SysML requirement diagram for trust AVs:** In the second phase of "trust requirements", we identify the general set of trust requirements for the AVs as presented in Fig. 6.

This step consists of determining the set of trust requirements (secured, trustworthy, trust) and, in the case of trust AV, how to integrate trust consideration into the requirement diagram. The requirements that will be merged are as follows:
- Secured requirement in which the system must ensure the AV's security and safety needs and satisfy all relevant attributes. The safety property is satisfied when the AV operates on the road without putting the system, the driver, or the passengers at risk. The AV must defend entities from attacks in order to fulfill the security property. The availability is ensured by offering protection against denial-of-service (DoS) assaults.

- Trustworthy requirement in which the system shall provide trustworthy needs and deliver service justifiedly be trusted. Performance, privacy, and dependability must all be met.

- A trust need that is directly tied to human concern, and the system must fulfill all stockholder needs and ensure that the system performs as intended. All of the following criteria must be met: correctness, usability, satisfaction, and efficacy.

We used the requirement element as an element to present the requirement view, and as connectors, we used “satisfy” to present that to attend trustworthy requirements should satisfy security requirements, and to attend trust requirements should satisfy trustworthy requirements. The second connector “traces to present the relation between the requirement view and the attributes.”

- SysML Requirement diagram over functionality of AVs: This diagram shows the expectation of AV driving to a given destination, from its start position, without colliding with encountered obstacles, in the shortest possible time as mentioned in Fig. 7.

![SysML Requirement Diagram Over Functionality of AVs](image)

**Fig. 7.** SysML requirement diagram over functionality of AVs.

c) The risk definition diagram for AVs (driving style, environment, visualisation): The third phase is the identification of risk and threats. In the field of the automotive industry, the lack of traceability of security threats and their effects with safety hazards is a source of risk, for that reason is necessary to determine this traceability. We prioritized and mentioned the essential threat that rely to the driveless of AVs in the road as presented in Fig. 8, the risk definition diagram.

![Risk Definition Diagram for AVs](image)

**Fig. 8.** Risk definition diagram for AVs.

Identifying risks related to phase 3 is used to eliminate security concerns at an early stage of system development and boost final user awareness, which leads to the discovery of suitable defense solutions. This activity incorporates security and safety concerns into a risk identification diagram.

*d) SysML uses diagram reflecting how the interactors trust AV:* The human must be convinced of AV’s functionality, safety and reliability. In the case of AV, human can be the designer or the user, or the passenger, pedestrians and drivers on the roads.

Trust and confidence must be won at the component, systems, vehicle, and V2X communication levels, AV’s driverless in the road. We have to understand how people trust AV and how interact with it. For our case, we will focus on trust of AV’s driverless in the road. Drivers can be either autonomous vehicles or human passengers in autonomous vehicles.

People who participate in the traffic environment on foot, via bicycle, or by other manual transportation methods are referred to as pedestrians. The SysML use case diagram is presented in Fig. 9.
The usability and satisfaction properties are included into the trust concern in the use case diagram. How users drive and trust AVs, how designers build, train, and control AVs, how drivers and passengers trust AVs on the road.

V. DISCUSSION

MBSE was utilized to manage the complexity of developing trust in CPS in terms of system requirements, design, analysis, verification, and validation activity. Many studies have shown that security requirements and risks can be addressed in the MBSE model, which is required to address the complicated multidisciplinary, multi-domain process of CPS, and that modeling may be successfully reused in the multidisciplinary sector.

MBSE has become a key component of developing complex cyber-physical systems [17, 18], and it is widely utilized [82] for collecting and controlling the system's requirements [83], architecture, design [84], and environment identification [85]. Also, by participating and expressing views for various objectives, it can help to ease communication among many stakeholders.

The MBSE method requires the design of different partial models for the different aspects of a technical system. For the use case, we applied our proposed phases to AV usage on the road. In this model, the system structure, the components of the system are modeled in the UML component diagram, and the interactions between users and the system are modeled as use case diagrams. The important risks are identified in the threat definition diagram. Trustworthiness is a holistic property of CPS, and heterogeneous system is modeled as a requirement diagram. The expectation of AV driving is modeled as a requirement diagram over functionality.

Some points are revealed:

- Targeting the balance between trust and trustworthiness is an important task during development.
- Cyber-physical systems should be made capable of presenting trustworthiness attributes.
- A well-structured method is required for modeling and developing a trustworthy CPS.
- MBSE could be leveraged in order to mitigate security risks and assure trustworthiness requirements at an early stage of system development.

VI. CONCLUSION

In this paper, an analysis of specific methodologies and tools that cover design activities for developing MBSE for trust CPS are presented. We introduced the phases of MBSE as well as activities in phases, and used methods. The MBSE method consists of the SysML/UML-based profile, trust requirements definition, and risk and threats definition, modeling the interactions of users and recommendations. The proposed MBSE method usage is presented by AV and how it can be trusted in the road and how the different actors interact with and trust it.

The use of MBSE is recommended, and companies must adopt their methodologies. MBSE is a formalized application of modeling to support system requirements, analysis, design, and validation and verification (V&V) activities, beginning in the conceptual design phase and continuing throughout development and later lifecycle phases. The MBSE approach is necessary to address the complex multidisciplinary, multi-domain process of CPS.

Cyber-physical systems (CPS) have been employed in a number of operations in the oil sector, where petroleum CPS optimization approaches can aid in petroleum exploration, production, and management. Several hazards confront the energy business, with the potential to interrupt critical supply lines, hurt the environment, and trigger a financial catastrophe. The scientific community is focusing on how to confidently realize a trust CPS. There is no clear description of all types of trust concerns and requirements in the literature, particularly in the sphere of oil and gas, and the subject of cyber security for oil and gas assets is not frequently addressed.

Our future work will focus on how to align the proposed MBSE method with the security and safety standards of specific industries, such as oil and gas and how to enhance the modeling of trustworthiness by adding a survey of the judgment by real participants, and computing and analyzing their acceptance.

REFERENCES


Construction of an Ontology-based Document Collection for the IT Job Offer in Morocco

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Abstract—Information Technology (IT) job offers are available on the web in a heterogeneous way. It is difficult for a candidate looking for an IT job to retrieve the exact information they need to locate the ideal match for their profile, without wasting time on useless searches. Traditional IT search systems are based on simple keywords that are generally not adapted to provide detailed answers because they do not take into account semantic links. In this article, an ontology is developed to meet the expectations of IT profiles from the IT job descriptions accumulated and pre-annotated using the UBIAI tool. The classes and subclasses of the ontology are designed using the Protégé 5.5.0 editor. Then the properties of objects and data are defined to improve the ontology. The ontology results are validated using DL queries by asking a number of questions to retrieve the requested information for each IT profile, and the ontology answers all these questions adequately. Finally, various plugins are used to display an ontology in a graphical representation.

Keywords—Ontology; IT job descriptions; semantic links; DL query; protégé 5.5.0

I. INTRODUCTION

The Web has become one of the most important sources of electronically accessible information, providing a gigantic mass of information as the number of resources on the Web grows exponentially and the number of users increases daily. Unfortunately, the Web is so huge and unstructured that finding accurate and useful information has become a time-consuming task. As a result, the documents accessible via the Internet, and web pages in particular, need to be structured, and this structuring is based on a very important attribute called "semantics".

This vision of the Web of the future depends on the construction of Web resources with information that can be exploited by software agents. Research has led to the birth of the semantic web, and a whole series of new technologies and concepts.

Over the past few years, the IT sector in Morocco has undergone significant change. With the challenge of digital transformation, recruitment for digital professions is a priority for companies. There is a wide range of profiles and skills in IT, with different descriptions.

Today, access to information plays an important role in the acquisition of knowledge, and is considered an important characteristic for the user. Despite the role played by traditional systems based on simple keywords and/or semantic similarity, they are generally not well suited to providing good job descriptions, as they do not take into account the links between concepts. On the other hand, as a recent graduate looking for a first job in IT, or in search of a career change, it's not always easy to find the ideal niche, so he needs access to the best tools to help him find the position that perfectly matches his profile, without wasting time with irrelevant recommendations and manual searches.

Consequently, the search for information led to the construction of the ontology that forms an important part of the Semantic Web, which on the one hand seeks to build on models of Web documents that represent IT job offers, based on the conceptual representation of the domain concerned and, on the other, aims to enable programs to make inferences on them. Clearly, ontologies are based on vocabularies and concepts that are explicitly specified in a representation language.

The study will be applied to unstructured data representing job offers requested in Morocco in the IT sector. The approach presented starts by extracting domain knowledge from a text, then representing it in the form of OWL ontology.

This paper is organized as follows. Section II mentions the background of our research, including definitions, components applications of ontology, and steps of Ontology Development followed by the general methodology for our research and the main steps to build our ontology in Section III. After that, the steps of evaluation and analysis are described in Section IV. The main results are discussed in Section V. Finally, the conclusion is presented in a Section VI.

II. BACKGROUND

In this section, some definitions of ontology are presented, then we present some domains of ontology application, and then we will present the methodology of development of ontology.

A. Ontology Definition

In computer science, ontology is a branch of knowledge engineering that focuses on creating structured and organized representations of information that can be understood by both humans and machines. It is a formal representation of knowledge that describes concepts and relationships within a particular domain.

Many authors have proposed several definitions of this concept. Notable among these have already been, formally [1]
defines ontology as set of concepts and the relationships between them. Concepts can be divided into classes, subclasses, attributes, relationships, and instances.

On the other hand, [2] presented the ontology as formalized and structured representation Co-conceptualized concepts and relationships. More precisely, it can be defined as a concept, the relationships, attributes, and hierarchies that exist in the domain.

In the context of computer systems, [3] declared that an ontology is a clear, structured description of common concepts. Ontologies include vocabularies and explanations of the relationships between classes.

Another effort is made to define an ontology was proposed by [4]. The authors consider it like a data modeling technique for structured data repository premised on collection of concepts with their semantic relationships and constraints on domain.

These definitions provide a range of perspectives on the concept of ontology in informatics and emphasize its role as a formal representation of shared conceptualizations and knowledge structures within a given domain.

B. Ontology Components

The components of ontology can vary depending on the specific approach and context; the essential elements of the ontology are presented as follows:

1) Concepts or classes: Concepts represent the entities or categories in a domain. They can be hierarchically organized, forming a taxonomy or a class hierarchy. Concepts capture the essence of the domain and serve as the building blocks of an ontology.

2) Properties or attributes: Properties define the characteristics, features, or attributes of concepts. They describe the relationships, roles, or qualities associated with concepts. Properties can be classified as object properties (defining relationships between concepts) or data properties (defining attributes with specific values).

3) Relations or relationships: Relations represent the connections and associations between concepts. They define how concepts are related to each other within the ontology. Relations can be symmetric, asymmetric, transitive, or other types depending on the nature of the relationship.

4) Instances or individuals: Instances represent specific instances or examples of concepts within a domain. They represent concrete objects or entities that belong to a particular class or concept in the ontology. Instances provide specific data or real-world examples to populate the ontology.

5) Axioms or constraints: Axioms capture the logical rules, constraints, or assertions about the concepts, properties, and relationships in the ontology. They define the semantics and rules that govern the relationships between different components in the ontology.

6) Annotations or metadata: Annotations provide additional descriptive information about the components of the ontology. They can include labels, definitions, synonyms, examples, and other metadata that enhance the understanding and usability of the ontology.

These components collectively form the structure and knowledge representation in an ontology. They enable the formal representation of domain-specific knowledge, relationships, and semantics. The specific representation and organization of these components can vary depending on the ontology language or framework used, as well as the requirements of the specific domain and application.

C. Ontology Application

Ontology finds applications in various domains. An overview of different domains is provided (Fig. 1), where ontology has been applied. For example, in the field of education, ontology is applied to learning analytics [10] and curriculum design [11], in the healthcare domain, it is used for clinical guideline [12], data integration [13] and data patient [14], also in the energy and environment domain, it has been applied to energy management [15], and smart grids [16]. Ontology development is also being applied in the agricultural domain, for example to precision agriculture [17], and pest control [18]. Some common domains of ontology application as follow:

![Application domains for ontologies.](image)

These are just a few examples of the diverse domains where ontologies have been applied. Ontology's versatility allows it to be utilized in various fields to enable knowledge representation, data integration, semantic interoperability, and intelligent systems.

D. Ontology Development

On the one hand, several methods have been developed to design ontologies either from scratch or by reusing existing ontologies. The process of building an ontology is similar to a commercial rather than a technical task [4]. Each design team or group can follow its own guidelines and decide which criteria and stages to work on and which can be skipped. Of course, this will depend on the scale of the ontology we intend to build.” Systematic review on ontology applications in virtual reality training domain: design components, roles, and research directions ».

On the other hand, this involves defining a domain of interest, determining concepts and relationships within that
domain, and formalizing the knowledge representation using ontology tools and languages. Here are the main steps involved in developing an ontology:

Identify the Domain: Specifies the domain or domain on which the ontology is built. Clearly define the scope and boundaries of the domain.

1) Knowledge gathering: Collect and analyze existing resources, such as domain-specific documents, databases, and sources, to understand concepts and relationships within the domain.

2) Conceptualization and modeling: Create a conceptual model that captures key concepts, their relationships, and the overall structure of the ontology. This can be done using graphical tools, such as UML diagrams or ORM, to visualize the conceptual model.

3) Define classes and properties: Define the classes (concepts) that represent the entities within the domain. Specify the properties (attributes) that describe the characteristics and relationships of the entities.

4) Establish relationships: Define relationships between classes using attributes.

5) Formalize the ontology: Use an ontological language, such as RDF or OWL, to formally represent concepts, attributes, and relationships in an ontology. Apply language structures and syntax to create machine-readable representations.

6) Validate and evaluate: Evaluate the quality and effectiveness of the developed ontology.

Ontology development is an iterative and collaborative process involving domain experts, ontologists and other stakeholders. It requires a combination of domain expertise, ontology engineering skills, and knowledge representation techniques.

III. PROPOSED SOLUTION

A. Ontology Building Methodology

Fig. 2 shows the general methodology for the research, which consists of main steps to build our ontology as follows:

- Collect Data (IT Job Document): the systematic approach of bringing together and measuring information from a variety of sources to obtain a complete and accurate view of the area of interest.

- Data Annotation: is the process of attributing, tagging or labeling data to help algorithms understand and classify the information they process.

- Ontology Data Modeling: the process of graphically representing data flows.

- Construct IT Job Ontology: representing general properties of what exists in a formalism that supports rational processing.

- Evaluation and Analysis: arriving at a conclusion with in-depth study of the ontology to better understand the information being presented.

Each step has a well-defined role and each is essential for the success of the work, the following section is devoted to detailing each phase.

B. Data Gathering

The data used in our work consists of collecting 150 job descriptions requested in Morocco, related to the information technology (IT) sector from several job websites such as (Rekrute, indeed, LinkedIn...). The data was stored in text files with the description as shown in Fig. 3.

UBIAI is an Oregon based startup that provides cloud-based solutions and services in the field of Natural Language Processing (NLP), to help users extract actionable insights from unstructured documents. It can annotate any type of document whether it’s PDF, images, or text. Also, it can train machine learning models and train state-of-the-art deep learning models on annotated datasets like Named Entity Recognition, Relation, and Extraction Document Classification, with few clicks and save up to 80% of annotation time.
We proceeded by annotating texts while identifying key entities such as (profile, company, location, responsibilities, skills, diploma, training, experience...) and the relationships between these entities as shown in Fig. 4.

In this section, we will conceptualize our data model for the ontology to be developed. This model seeks to examine the interactions between descriptive components of job offers. In order to accomplish the goals of our application, the model represents several sorts of entities along with their attributes and relationships among them.

The schema for the data model is displayed in the following Fig. 6 and Table I.

### C. Ontology Data Modeling

Before starting to build an ontology, the ontology modeling phase is very important, as it provides us with a conceptual map and a clear and correct vision of the work to be carried out. It’s easier to represent concepts graphically than to represent them in a serialized (Resource Description Framework)/ (Extensible Markup Language) format. This is why we have chosen to represent concepts using object-role modeling (ORM). The early 1970s saw the emergence of ORM, a semantic modeling methodology that distinctly interprets the world in terms of objects fulfilling roles [5].

![An overview of the annotation sheet for the job offer.](image)

**Fig. 4.** An overview of the annotation sheet for the job offer.

After extracting the entities from the job descriptions and building the database, we first need to know how they are distributed.

In Fig. 5, the distribution is non-homogeneous for the entities with 1190 data for the IT Skills entity, and 1098 data for the Responsibility entity. These two entities represent the largest distributions in the diagram compared to the other annotated entities, which means that in the field of IT job offers, we focus more on the IT skills and responsibilities required.

![Distribution of the labels per entity type.](image)

**Fig. 5.** Distribution of the labels per entity type.

<table>
<thead>
<tr>
<th>Node</th>
<th>Node Description</th>
<th>Attribute</th>
<th>Attribute Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>ITJobOffer</td>
<td>The IT job offer launched by the company</td>
<td>Id</td>
<td>A unique identification of the node.</td>
</tr>
<tr>
<td></td>
<td></td>
<td>name</td>
<td>The name of the IT job offers.</td>
</tr>
<tr>
<td>PositionOffered</td>
<td>The position offered by the IT Job Offer</td>
<td>id</td>
<td>A unique identification of the node.</td>
</tr>
<tr>
<td></td>
<td></td>
<td>name</td>
<td>The name of the position offered</td>
</tr>
<tr>
<td>Company</td>
<td>the name of the company posting the job offer</td>
<td>ActivityArea</td>
<td>the business sector of the company</td>
</tr>
<tr>
<td></td>
<td></td>
<td>Function</td>
<td>the functions performed by the company</td>
</tr>
<tr>
<td></td>
<td></td>
<td>Location</td>
<td>Location of the company that posted the offer</td>
</tr>
<tr>
<td></td>
<td></td>
<td>id</td>
<td>A unique identification of the node.</td>
</tr>
<tr>
<td>Profile</td>
<td>The type of candidate concerned by the IT joboffer</td>
<td>Id</td>
<td>A unique identification of the node.</td>
</tr>
<tr>
<td></td>
<td></td>
<td>Profile</td>
<td>IT profile name</td>
</tr>
<tr>
<td>Skill</td>
<td>The required knowledge and competencies for entry into the position.</td>
<td>id</td>
<td>A unique identification of the node.</td>
</tr>
<tr>
<td></td>
<td></td>
<td>SoftSkill</td>
<td>The skills that enable the candidate to fit in at a workplace. They include personality, communication, attitude, flexibility, and motivation.</td>
</tr>
<tr>
<td></td>
<td></td>
<td>ITSkill</td>
<td>Also known as technical skill or hard skill which are directly relevant to the job to which the candidate applying.</td>
</tr>
</tbody>
</table>
D. Ontology Construction

The ontology development process is complex. This is why it is necessary to use a method or methodology to support the ontology construction process. This is why the method used is to design ontologies from scratch using conceptual data models.

To develop our ontology, it is important to choose an appropriate ontology editor. The Protégé editor was chosen, as it has established itself over the last 20 years as one of the best tools for ontology creation and information presentation [6].

Protégé is widely considered one of the most popular and widely used ontology development tools for several reasons: it’s an open source and free, it has a large and active user community, including developers and domain experts, it offers a user-friendly interface and intuitive features, it can handle complex ontologies and is extendable through plugins. Protégé allows users to collaborate on ontology development projects by supporting multiple users working on the same ontology simultaneously.

This is the main step in the development process. To design the class hierarchy, we need to refer to the list of terms in Table I and choose the most appropriate concepts to represent superclasses and subclasses. As a result, 26 classes are described in total, as seen in Fig. 7.

The ITJobOffer Class is the mother class of all classes included five main sub classes that included: Company (Activity_Area, Functions and Location), Contract, Position_offered, Required_profile (Diploma, Experience, Major Skills), Required_Responsabilities.

![Fig. 7. Ontology class hierarchy.](image1)

Fig. 7. Ontology class hierarchy.

![Fig. 8. Object property hierarchy.](image2)

Fig. 8. Object property hierarchy.

Fig. 8 shows the object property hierarchy that represents the relationships between classes. For example, “masters” and “mastered_by” are two inverted object properties that describe the relationship between the "required_profile" and "skills" classes. The intention of this phase is to establish connections between classes that enable the ontology to answer our questions.
Fig. 9. Data property hierarchy.

On the other hand, Fig. 9 shows the data properties that describe the information relating to each individual, for example, "company_description", "company_name", "function_company" are used to define instances of the class "Company".

Fig. 10. Individuals.

Fig. 10 summarizes some of the individuals created based on our collected database. For example, for the "soft_skills" subclass, we have described different values that a "BI_DATA_Engineering" profile must master:

"Strong_analytical", "Team_work", "Communication_skill".

This step is used to create instances after completing the ontology model, describing the instances or individuals, which involve assigning a unique name to each individual after specifying the class to which it belongs and defining the values of these attributes (data properties).

IV. EVALUATION AND ANALYSIS

A. DL Queries

The evaluation and analysis steps are a critical one, where we have to make queries to find answers to our research questions and evaluate our ontology.

We used the DL query tab, available in the protected 5.5.0 software, to enter queries and launch the reasoner before executing the DL queries. The structure of the ontology is validated using the HermiT reasoner [19], which is used to check the consistency and coherence of a model and to verify and evaluate the ontology's correct functioning. This allows us to answer the questions: "Is the world of IT offers well represented by Ontology? and "Are IT profiles satisfied with the answers to their questions?"

The result shows direct classes, subclasses, superclasses and instances. DL queries can be added to the ontology to extract more precise information.

Some examples of DL queries and their results are shown below with results answering a set of questions:

- What are the different soft skills mastered by the "BI Data Engineering" profile? (Fig. 11)
- What are the different IT skills mastered by the "Devops engineering" profile? (Fig. 12)
- What are the different skills that must master the "BI Data Engineering" profile? (Fig. 13)
- What are the companies that recruit the BI Data engineering profile, also the experience and skills required for this candidate? (Fig. 14)

Fig. 11. Query 1.
V. Results and Discussion

Several individuals created for the "requested_profile" class. Fig. 15 shows a sample of individuals in the IT jobs ontology after representation in OWL, this profile named "profile_2" belongs to the Devops engineering sector, has IT skills (maintenance, administration of linux servers and storage), soft skills (Good communication, teamwork) experienced in (automation, script language and Devops Tools) and has a Bac +5 diploma.

In Fig. 16, we visualize the structure of the ontology of the IT job offer using the OntoGraf tab of the Protégé editor. The elements of this structure are automatically organized, and the different relations are supported: subclass, individual, domain/range, object properties and equivalence [7].

Using the OWLVIZ tab in protected, we generated a visual representation of the IT job posting ontology shown in Fig. 17, but we installed GraphViz before working on this tab.

To validate our ontology, there are a number of plugins available in our Protégé editor [8]. One of these plugins is named VOWL, which offers a graphical representation of an ontology layout. To use this tool, we download the JAR file and then copy it to the appropriate plugin folder. Then you must activate the Protected VOWL plugin via the Window → Tabs → VOWL option. For example, Fig. 18 displays a VOWL visualization of the IT job offer ontology, illustrating a graphical overview of the job.
The data analysis software used in this case study is Microsoft Excel, which is perfectly useful for data analysis thanks to simple and practical operations, to present, and to interpret from spreadsheets, a multitude of data.

Referring to the accumulated and annotated data, Fig. 19 shows that the largest share of profiles in demand in the IT sector in the year 2023, is for IT engineers and developers, accounting for almost 75% of the recruitment market for IT profiles in Morocco, followed by Business Intelligence and Data engineering profiles, while the IT security sector accounts for only 15% of the demand in Morocco.

In order to obtain an in-depth comparison, it is essential to generate additional SPARQL queries that provide numerical results to assess the accuracy of the evaluation procedure. This will enable a more representation of the main result. SPARQL is a standard query language and protocol for Linked Open Data and RDF databases. It is designed to query a variety of data, it can effectively extract information hidden in different data stored in different formats and sources).

To this end, we have queried our developed ontology to count the number of profiles most in demand in the Moroccan IT job market, comparing it with the first result found in Fig. 20, the ontology answered the question well.

Fig. 20. Counting of requested profiles in IT job offers in on Protégé.

Ontology evolution is an important aspect of the Semantic Web and knowledge representation, constantly evolving to meet the changing needs of various industries, including IT. Several potential new features in ontology development for IT Recruitment enable efficient integration of data from various online sources to provide a holistic view of information and form knowledge graphs that support advanced analysis.

VI. CONCLUSION

In this article, we propose an analysis of the problem related to the Information Retrieval domain and those specific to the IT sector. As the web is a very extensive data resource, data accumulation and knowledge extraction is the first step in our system, consisting of extracting keywords according to their semantic weights and domain values. Our contributions focus on the crucial step of conceptualizing the accumulated textual documents into concepts and relationships. This work proposes the process for building an ontology that semantically demonstrates the job offers requested in the IT sector in Morocco. The use of semantic technology can help IT profiles
to find the job that perfectly matches their profile by answering their questions in a relevant way.

We intend to extend this research in the future by incorporating an educational application based on the constructed ontology, as well as applying machine learning algorithms for the extraction of named entities and the relationships between them.

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Enhancing User Experience Via Calibration Minimization using ML Techniques

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Abstract—Electromyogram (EMG) signals are used to recognize gestures that could be used for prosthetic-based and hands-free human computer interaction. Minimizing calibration times for users while preserving the accuracy, is one of the main challenges facing the practicality, user acceptance and spread of upper limb movements' detection systems. This paper studies the effect of minimized user involvement, thus user calibration time and effort, on the user-independent system accuracy. It exploits time based features extracted from EMG signals. One-versus-all kernel based Support Vector Machine (SVM) and K Nearest Neighbors (KNN) are used for classification. The experiments are conducted using a dataset having five subjects performing six distinct movements. Two experiments performed, one with complete user dependence condition and the other with the partial dependence. The results show that the involvement of at least two samples, representing around 2% of sample space, increase the performance by 62.6% in case of SVM, achieving accuracy result equal to 89.6% on average; while the involvement of at least three samples, representing around 3% of sample space, cause the increase by 50.6% in case of KNN, achieving accuracy result equal to 78.2% on average. The results confirmed the great impact on system accuracy when involving only small number of user samples in the model-building process using traditional classification methods.

Keywords—EMG signals; user independence; EMG user acceptance; HCI; movement classification; calibration minimization

I. INTRODUCTION

Human computer interaction is highly relying on measuring and recording the signals produced by human body. Electroencephalogram (EEG), electrocardiogram (ECG), electrotoculogram (EOG) and electromyogram (EMG) are electrical signals used in various HCI systems either separated or fused. EEG signals measures the brain electrical behavior from the surface of the scalp [1]. EMG signals are the biological signals generated as a result of the potential difference caused by skeletal muscular contractions [2]. EOG signals are used for detection of involuntarily or intended eye movements that could be used in a verity of applications [3]. ECG signals are acquired to evaluate the heart functionality and detect its related diseases [4].

Monitoring and detecting the changes in the EMG signals are beneficial to the researchers in the medical field in order to recognize neuromuscular diseases and help out in the rehabilitation process [5]. As claimed by Turgunov in [6], the muscular disabilities are spreading for various reasons causing the increase in demand for prosthetic limbs and assisting robots. They could help the paralyzed patients to be able to achieve their daily activities with minor or no involvement from others using sensors that measure the EMG signals and act accordingly through gesture recognition systems [7].

Those systems are also used for moving a remote robot and encouraging the distant rehabilitation process that is intensively needed after having strokes caused by a diversity of reasons [8]. Automated EMG signals translation offers great contribution in the success of remote monitoring for ALS patient using the measured physiological signals [9] and the availability of distant therapy and achieving high improvements in muscular responsiveness and motor functionality [2], [10]. EMG based gesture recognition is also used for explaining the sign language used by deaf people helping them to be easily engaged in the society [11]. EMG signals can be used also in fatigue detection [12], [13] and emotion identification systems [14] that could be beneficial for ergonomic and entertainment applications. Gamification therapy [15] as well as virtual reality and augmented reality [10], [16], [17] could be developed and evolved by enhancing the automated understanding muscle non-spoken language.

The muscular activity recognition systems start by collecting the EMG signals from relevant human parts. EMG signals do not only carry information about the movement itself but also contain other internally interfering signals such as muscle fatigue and emotional involuntary movements as well as external conditions like sensor placement or other sources of noise. So preprocessing component takes place to clean, filters noise and unwanted signals, performs segmentation and does normalization [18]. The feature engineering and classification for biological is described in [19], [20].

Muscular movement recognition systems face various challenges that highly affect the user acceptance. They include the detection accuracy and classification performance, as well as the system generalization and robustness [21]. System accuracy and performance are influenced by the noisy nature of EMG data that is caused by body interior sources like cross talk effect in which the signals produced by the contraction of neighbor muscles interfere with the readings of the intended muscles. EMG signals are also prone to variations over time for plenty of reasons like electrode shift, emotional,
involuntary movements and muscular fatigue. Motor tasks with different imposed force or unintended limb orientation changes also contribute in the non-stationarity of the generated signals [20], [22], [23]. As reported in [20], [24], the time separating training and testing negatively affects the performance of movement recognition systems for the same subject.

The user independent systems, having different participants for both training and testing purposes, impose extra burden as change of physiological features like the age, height, weight, and behavioral characteristics like exercise routine would minimize the system ability to generalize across users [22]. In [25], the high inter-subject variability necessitates long and frequent user-specific training which affects user acceptability. In [26], the authors performed a comparison between the measured muscular activities of amputees and able-bodied subjects when controlling myoelectric device in order to overcome the long and frequent user calibration. Their results confirmed the generalization challenge facing various users in general and amputee in specific.

There is also the issue of small dataset size related to EMG based prediction that is extensively reviewed and studied in [27], confirming the need for bigger datasets to overcome the overfitting problem facing, particularly, deep model generation.

This paper addresses and analyzes the effect of various levels of user involvement in the calibration process on system’s performance. Its main contribution is to pave the way for a limited resource-consumption and minimized user-calibration time solution to the gesture classification while preserving reasonable accuracy results. Traditional ML techniques are used due to their low utilization of hardware resources and their comparable results to deep learning solutions which is proved from the related work presented in Section II. The experiments are conducted to measure the impact of changing the number of samples needed from the user in the calibration process in an attempt to reduce this value in order to minimize user-calibration time. Two ML techniques are involved in these experiments. They reach the conclusion that the user-specific features, which can highly improve EMG-classification performance, can be learned from 2%-3% of the training sample space.

The rest of this paper is organized as follows; the next section presents the extensive research work that aims to tackle these issues and increase the EMG based movement classification systems’ usability while preserving high accuracy results. Section III describes the used system in this study. Then the results of this analysis are shown and discussed in Section IV while the last Section provides a conclusion reached by this study.

II. RELATED WORK

In order to increase user acceptance for prosthetic devices or gesture based remote controlling systems, multiple researches attempt to increase performance and minimize user calibration. Unfortunately as reported in [28], [29], the performance of deep learning is highly dependent on the data set size available for training. The larger dataset, the higher the performance would be. But the publicly available EMG gestures datasets suffer from the size issue placing restrictions on the use of deep learning solutions as reported [27]. Moreover collecting massive amount of data from the system’s end user would form an obstacle to system usability due to time, efforts and frequency requirements of the (re)calibration process [30], [31]. In [30], they state the need for a study that deals with the issues related to prolonged and repetitive recalibration sessions. So investigating the effect of minimizing user calibration, features or dataset size while maintaining high performance using different traditional or deep learning solutions, gets increasing priority.

In [32], the authors analysed working with dataset with variable force levels using traditional classification method (KNN). They proposed an iterative feature extraction method to be used for identifying six grip activities with different force levels; low, medium, and high. The success rates accomplished were relatively comparable as the classification results were 97.78% for Low, 93.33% moderate and 92.96% for high force level. The work with various force levels was also presented in [23] but this time with deep networking solution, where the authors achieved average accuracy of around 91% using LSTM-based neural network across all amputees subjects and force levels confirming a comparable results between deep and traditional solutions when working with different force levels of EMG signals.

The hand gestures identification for the sake of stroke rehabilitation is applied in [5] for six hand gestures. Time and frequency features of 20 subjects are provided to the classification phase. KNN has shown better accuracy 98% over Artificial Neural Networks (ANN) and Support Vector Machines (SVM). They also tried to reconstruct identified gesture from EMG-based generated joint trajectories and compare it to the generate movement by a VICON camera tracking system producing a correlation of 0.91. Proving the approximate accuracy produced by traditional methods and neural networks using muscular generated signals.

The work for minimizing the user calibration time and studying the impact of user-independence EMG based classification systems take place in [33]. The authors investigate the feasibility of zero retraining and achieving the rotation and hand independence. The experiments include twenty participants with all rotations and both hands utilization are allowed for eight distinctive gestures; rest, flex, extend, abduct, open, close, thumb, and ok. They find that the accuracy notably decreases under these generalizations except for wrist extension gesture which is found to be consistent among all gestures.

The authors in [25] propose LSTM-CNN model for hand gesture classification in order to check the possibility of creating user independent solution and reduce the need for system recalibration for new users. They start by recognizing seventeen gestures’ classes from 40 subjects in a user-dependent way and the model reaches accuracy of 81.96%. But when building the user-independent model, they use gesture signals from only four hand movements in the training process. Their model accuracy drops to 77% for unseen users. They use GradCAM analysis in an attempt of getting a shallower design...
in order to reduce the high training time and memory consumption that are required when deep learning model is used.

In [34], the objective is to minimize the volume of data needed to train a deep neural model. They used the learned Dilated Efficient CapsNet with a decrease of 20% of the training EMG signals per repetition in the transient phase. They maintain an accuracy of 80%.

As demonstrated from the shown recent related work that the problems related user acceptance for EMG based classification and their prosthetic devices massively depends on user calibration time which is needed for reaching reasonable performance results. Different solution approaches are considered including the use of deep learning techniques or reducing the need for user involvement using previously collected samples from other users. The deep learning techniques face the issue of small EMG dataset size and high required computational and storage resources. Some researches reached the conclusion that the results of deep learning and traditional machine learning are comparable either for user dependent or independent solution. So in this paper, the incremental minimization of user calibration is analyzed using traditional classification methods; KNN and SVM.

III. METHODOLOGY

As shown in the previous section, one of the main challenges facing the EMG based hand activity recognition system is the lack of inter and intra-user generalization which causes the need for extensive calibration that consumes user’s time and negatively affects user experience. The accuracy of the recognition system is found to be dropping when the movement activity samples used in the training process are gathered from subjects other than the final user whose samples are used in the testing or validation processes. This user generalization issue limits the user acceptability which is highly depending on accuracy and calibration time [25]. So in this paper we investigate the effect of incremental involvement of the final user’s samples in the training process on the overall model accuracy results. The results are analyzed using ANOVA test in Section IV. The next paragraph describes the used system in order to view the influence of incremental user samples in the training process.

The hand gesture recognition system used in this study utilizes the traditional recognition methods. It is composed of various components as shown in Fig. 1.

![Fig. 1. EMG hand gesture recognition system.](image)

It includes the data acquisition process performed by [26] which first collects the EMG signals from two differential electrodes using as a programming kernel, the National Instruments (NI) Labview. The electrodes are placed on the forearm surface by elastic bands with an additional reference electrode is put in the middle, in order to record information about the muscle activation. Then the preprocessing phase is applied on the captured signals to cleanse, remove noise and unwanted signals, performs segmentation and normalization. So it uses an 8th-order bandpass IIR filter with lower frequency 15 Hz and higher frequency 500 Hz. Then the feature extraction takes place for each trial. It extracts twelve AutoRegression (AR) [35] coefficients as three coefficients are generated per trial segment. AR features are the coefficients of a statistical model that is generated to predict new values of data given the old ones. A K-th order autoregression AR(k), is the model that uses K past points of the time series in order to guess the new point to come with K coefficients to give weights for each previous point and determine its participation in calculating the next value according to its distance from it according to the following equation.

$$X_i = \sum_{j=1}^{k} \phi_j x_{ij} + \omega_i$$

(1)

Then at the classification phase, one-versus-all SVM classification model with Gaussian radial based kernel is trained using the previously extracted features. The one-versus-all SVM classifier [28,29] is generated by combining the decision from various binary classifiers that separate one class from all other classes. The binary version is utilized to calculate the separation hyperplane with transformed Hilbert space vectors. The decision function is computed as

$$D(x) = \sum_{i=1}^{p} a_i y_i K(x_i, x) - b$$

(2)

Where xi are the training samples input vectors and yi are the samples output which have different sign for the two classes. The Gaussian Radial basis function is formulated as:

$$K(x_i, x) = e^{-\|x_i - x\|^2 / 2\sigma}$$

(3)

To confirm the results, another classifier, KNN, is used. It is a supervised machine learning algorithm. It categorizes the item according to the most common class of its K nearest neighbours. It relies on a distance metric to determine the item’s neighbours. In this paper we use Euclidean distance.

The trained model is then used to distinguish the testing features. The use of the traditional feature extraction and classification methods like AutoRegression and SVM and KNN is beneficial for achieving the simplicity and saving the time required for the training process [32] specially with the small size of the dataset that could cause overfitting and reduce accuracy when deep neural networks are used [33].

The next section describes the experiments conducted to study the effect of decreasing user provided samples in the calibration process in order to achieve acceptable performance and increase the user acceptance.

IV. EXPERIMENTS AND RESULTS

Two experiments are conducted with the two classifiers, SVM and KNN. One uses complete user dependence condition
while the other uses the partial dependence. The dataset is collected by [26] from five normal and healthy subjects of age 20-22. They are asked to perform six different movements for thirty trials. The measured time is 6 sec.

The movements are Spherical, Tip, Palmar, Lateral, Cylindrical, and Hook. Spherical movement is recognized when holding spherical objects while Cylindrical movement is recognized when holding cylindrical objects. Tip movement is recognized when holding small objects while thin and flat objects grasps are recognized as Lateral. Grasping with palm facing the object is called palmar movement and Hook movement for supporting a heavy load. No previous instructions about speed or force are given to subjects.

The dataset is partitioned using 10 folds, nine folds contributes in building the training model while the last fold is used for testing the unseen samples. The average Correct Classification Rate (CCR) for various movements using SVM is shown in Table I. The results show that the average accuracy across various users is around 97% with average 99% for tip and lateral movements as the most identified movements.

### TABLE I. CCR FOR VARIOUS MOVEMENTS WHEN SVM BASED USER DEPENDENT CLASSIFICATION IS APPLIED

<table>
<thead>
<tr>
<th>Subject Move</th>
<th>1</th>
<th>2</th>
<th>3</th>
<th>4</th>
<th>5</th>
<th>average</th>
</tr>
</thead>
<tbody>
<tr>
<td>Cyl.</td>
<td>0.97</td>
<td>0.97</td>
<td>0.93</td>
<td>1</td>
<td>1</td>
<td>0.97</td>
</tr>
<tr>
<td>hook</td>
<td>1</td>
<td>0.93</td>
<td>0.93</td>
<td>0.93</td>
<td>1</td>
<td>0.96</td>
</tr>
<tr>
<td>tip</td>
<td>0.97</td>
<td>1</td>
<td>1</td>
<td>1</td>
<td>0.99</td>
<td></td>
</tr>
<tr>
<td>palm</td>
<td>1</td>
<td>0.97</td>
<td>0.93</td>
<td>0.97</td>
<td>1</td>
<td>0.97</td>
</tr>
<tr>
<td>Sph.</td>
<td>0.93</td>
<td>0.93</td>
<td>0.97</td>
<td>0.93</td>
<td>1</td>
<td>0.95</td>
</tr>
<tr>
<td>Lateral</td>
<td>0.97</td>
<td>1</td>
<td>1</td>
<td>0.97</td>
<td>1</td>
<td>0.99</td>
</tr>
<tr>
<td>average</td>
<td>0.97</td>
<td>0.97</td>
<td>0.96</td>
<td>0.97</td>
<td>1</td>
<td>0.97</td>
</tr>
</tbody>
</table>

While using KNN classification with K=3, The average Correct Classification Rate (CCR) for various movements, as shown in Table II. The results show that the average accuracy across various users is around 92% with average 93% for tip and lateral movements as the most identified movements.

### TABLE II. CCR RESULTS WHEN APPLYING KNN WITH K=3 AND EUCLIDEAN DISTANCE METHOD

<table>
<thead>
<tr>
<th>Subject Move</th>
<th>1</th>
<th>2</th>
<th>3</th>
<th>4</th>
<th>5</th>
<th>average</th>
</tr>
</thead>
<tbody>
<tr>
<td>Cyl.</td>
<td>0.9</td>
<td>0.9</td>
<td>0.93</td>
<td>0.93</td>
<td>0.93</td>
<td>0.92</td>
</tr>
<tr>
<td>hook</td>
<td>0.9</td>
<td>0.93</td>
<td>0.9</td>
<td>0.9</td>
<td>0.9</td>
<td>0.91</td>
</tr>
<tr>
<td>tip</td>
<td>0.93</td>
<td>0.93</td>
<td>0.9</td>
<td>0.93</td>
<td>0.93</td>
<td>0.93</td>
</tr>
<tr>
<td>palm</td>
<td>0.9</td>
<td>0.9</td>
<td>0.93</td>
<td>0.9</td>
<td>0.9</td>
<td>0.91</td>
</tr>
<tr>
<td>Sph.</td>
<td>0.9</td>
<td>0.9</td>
<td>0.93</td>
<td>0.93</td>
<td>0.93</td>
<td>0.91</td>
</tr>
<tr>
<td>Lateral</td>
<td>0.93</td>
<td>0.93</td>
<td>0.93</td>
<td>0.93</td>
<td>0.93</td>
<td>0.93</td>
</tr>
<tr>
<td>average</td>
<td>0.91</td>
<td>0.92</td>
<td>0.93</td>
<td>0.92</td>
<td>0.92</td>
<td></td>
</tr>
</tbody>
</table>

The second experiment performed tends to measure the effect of user samples contribution in building the training model. Using SVM as the classifier, the involvement of the first 10 user samples influences the classification accuracy as presented in Table III. The results show huge increase in the average accuracy. The performance enhancement could reach around 62.6% on average, when more than one sample from the user is involved in the training process and building the model. This sample represents around 1% of the training sample space. It achieves an accuracy result that equals to 89.6% on average.

The results of involving the user samples in building KNN model, as shown in Table IV, lead to the same conclusion. A great increase of accuracy of 50.9% on average is reached, as the number of user’s involved samples is more than 2. Two samples represent around 2% of the training sample space. The KNN model achieves an accuracy result equals to 78.2%.

The effect of 27 of user samples contribution in building the training model is summarized in Fig. 2. It shows that the effect of user involvement decreases after three or more samples. Furthermore, the performance does not witness much difference after the involvement of the tenth sample and the sixteenth sample in case of SVM, and KNN respectively. The involvement of those samples lead to an accuracy of 98%, and 88% on average in case of SVM, and KNN respectively.

### TABLE III. SVM BASED USER PARTIAL PARTICIPATION CCR ACCURACY RESULTS

<table>
<thead>
<tr>
<th>Subject No.Samples</th>
<th>1</th>
<th>2</th>
<th>3</th>
<th>4</th>
<th>5</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>0.22</td>
<td>0.21</td>
<td>0.41</td>
<td>0.27</td>
<td>0.24</td>
</tr>
<tr>
<td>2</td>
<td>0.88</td>
<td>0.86</td>
<td>0.94</td>
<td>0.93</td>
<td>0.87</td>
</tr>
<tr>
<td>3</td>
<td>0.89</td>
<td>0.89</td>
<td>0.95</td>
<td>0.94</td>
<td>0.88</td>
</tr>
<tr>
<td>4</td>
<td>0.9</td>
<td>0.89</td>
<td>0.97</td>
<td>0.95</td>
<td>0.89</td>
</tr>
<tr>
<td>5</td>
<td>0.92</td>
<td>0.92</td>
<td>0.97</td>
<td>0.95</td>
<td>0.91</td>
</tr>
<tr>
<td>6</td>
<td>0.93</td>
<td>0.92</td>
<td>0.97</td>
<td>0.96</td>
<td>0.92</td>
</tr>
<tr>
<td>7</td>
<td>0.96</td>
<td>0.95</td>
<td>0.99</td>
<td>0.99</td>
<td>0.94</td>
</tr>
<tr>
<td>8</td>
<td>0.96</td>
<td>0.95</td>
<td>0.98</td>
<td>0.98</td>
<td>0.93</td>
</tr>
<tr>
<td>9</td>
<td>0.98</td>
<td>0.97</td>
<td>1</td>
<td>0.99</td>
<td>0.94</td>
</tr>
<tr>
<td>10</td>
<td>0.98</td>
<td>0.98</td>
<td>1</td>
<td>1</td>
<td>0.96</td>
</tr>
</tbody>
</table>

### TABLE IV. KNN, K=3 USER SAMPLES PARTICIPATION CCR ACCURACY RESULTS

<table>
<thead>
<tr>
<th>No.Samples</th>
<th>Subject</th>
<th>1</th>
<th>2</th>
<th>3</th>
<th>4</th>
<th>5</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>0.18</td>
<td>0.19</td>
<td>0.28</td>
<td>0.01</td>
<td>0.32</td>
<td></td>
</tr>
<tr>
<td>2</td>
<td>0.26</td>
<td>0.21</td>
<td>0.32</td>
<td>0.04</td>
<td>0.35</td>
<td></td>
</tr>
<tr>
<td>3</td>
<td>0.79</td>
<td>0.79</td>
<td>0.77</td>
<td>0.76</td>
<td>0.80</td>
<td></td>
</tr>
<tr>
<td>4</td>
<td>0.81</td>
<td>0.80</td>
<td>0.77</td>
<td>0.76</td>
<td>0.81</td>
<td></td>
</tr>
<tr>
<td>5</td>
<td>0.79</td>
<td>0.78</td>
<td>0.77</td>
<td>0.75</td>
<td>0.81</td>
<td></td>
</tr>
<tr>
<td>6</td>
<td>0.79</td>
<td>0.78</td>
<td>0.77</td>
<td>0.76</td>
<td>0.80</td>
<td></td>
</tr>
<tr>
<td>7</td>
<td>0.81</td>
<td>0.81</td>
<td>0.77</td>
<td>0.76</td>
<td>0.81</td>
<td></td>
</tr>
<tr>
<td>8</td>
<td>0.83</td>
<td>0.80</td>
<td>0.81</td>
<td>0.77</td>
<td>0.83</td>
<td></td>
</tr>
<tr>
<td>9</td>
<td>0.82</td>
<td>0.82</td>
<td>0.81</td>
<td>0.79</td>
<td>0.83</td>
<td></td>
</tr>
<tr>
<td>10</td>
<td>0.83</td>
<td>0.83</td>
<td>0.81</td>
<td>0.81</td>
<td>0.85</td>
<td></td>
</tr>
</tbody>
</table>
Among the moves, as shown in Fig. 3 the hook move is the most distinguished one across all users from the first user sample using the one-versus-all SVM classifier. The other moves give approximately similar accuracy results after the involvement of two samples.

To confirm that the results’ means do not suffer from significant differences, two-way ANOVA test is applied to the partial samples inclusion results. It is proved that in both SVM and KNN based classification, for various subjects, the rejection of the null hypothesis is implied as the value of p is less than 0.05 for partial samples and different users.

V. DISCUSSION

The results of our experiments show the possibility of involving only a small number of user samples, which could represent 2%–3% of the total sample space, in the recalibration process in order to gain significantly increased performance accuracy. These findings could lead to the minimization of calibration time while maintaining relatively high performance results and thus increasing the user acceptance which is a main challenge in the EMG based gesture classification systems and the industry of EMG based prosthetic devices.

Our results used a small dataset that is publicly available which includes only five normal and healthy subjects with six different movements for thirty trials. So in future work, we intend to confirm our results using larger datasets with various movements for both healthy users and amputees.

VI. CONCLUSION AND FUTURE WORK

This paper is concerned with the user independence challenge facing the EMG-based movement recognition system. It reviews the previous research work regarding this obstacle that affects both user accuracy and acceptability. It studies the effect of partial user samples involvement in the calibration process using AutoRegressive features and traditional classification methods like one-versus-all SVM with Gaussian radial based kernel and KNN.

The findings are interesting in that a huge increase of accuracy has occurred as a result of including two to three user samples which represents around 2%–3% of the total training sample space. The increase is estimated to reach 62.6% on average in case of SVM classifier and 50.6% in case of KNN, achieving accuracy results equal to 89.6% on average in case of SVM and 78.2% in case of KNN. The results somehow stabilize after ten samples in case of SVM to reach 98% on average and after sixteen samples in case of KNN to reach 88% on average.

After applying two way ANOVA test to the partial samples inclusion results, either in SVM based classification or KNN based classification, for various subjects, it implies the rejection of the null hypothesis as the value of p <.05 for partial samples groups and different users confirming that the results’ means do not suffer from significant differences.

The results assured the great influence on system accuracy when involving small number of user samples in the model-building process using traditional classification methods.

As a future work, we intend to confirm our results using larger datasets with various movements for both healthy users and amputees.

REFERENCES


A New Approach Method for Multi Classification of Lung Diseases using X-Ray Images

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Electrical Department-Engineering Faculty, Universitas Semarang, Semarang, Indonesia

Abstract—Lung disease is one of the most common diseases in today's society. This lung disease's treatment is frequently postponed. This is usually due to a lack of understanding about proper treatment and a lack of clear information about lung disease. Reading the correct X-ray images, which is usually done by experts who are familiar with these X-rays, is one method of detecting lung disease. However, the results of this diagnosis are dependent on the expert's practice schedule and take a long time. This study aims to classify lung disease images using preprocessing, augmentation, and multimachine learning methods, with the goal of achieving high classification performance accuracy with multi-class lung disease. The classification ExtraTrees was obtained from experimental results with unbalanced datasets using a balancing process with augmentation. Precision, Recall, Fi-Score, and Accuracy are 100% for training and testing data 89% for Precision, 88% for Recall, 87 for Fi-Score, and 85% for Accuracy outperform other machine learning models such as kneighbors, Support Vector Machine (SVM), and Random Forest in classifying lung diseases. The conclusion from this research is that the machine learning approach can detect several lung diseases using X-ray images.

Keywords—Augmentation; machine learning; lung disease; prepossessing

I. INTRODUCTION

The lungs are one of the organs in the respiratory system that serve as a site for the exchange of oxygen and carbon dioxide in the blood. Polluted air is a common problem, and the air that is inhaled contains many germs that will attack the lungs. Lung disease is a serious disease that affects the human respiratory system and can be fatal if not properly treated. This lung disorder causes sufferers to have difficulty breathing, difficulty performing activities, and a lack of oxygen, which can lead to death if not detected quickly [1]. Tuberculosis, bronchitis, pneumonia, lung cancer, emphysema, and pleuritic are all common lung diseases. It is usually done clinically to detect lung disease/disorders (physical symptoms by a doctor). Aside from clinical examination, X-rays, CT scans, and MRI can be used to diagnose lung disease; however, the latter two methods are more expensive. [2]. Another issue is a lack of public knowledge in reading CT Scan and MRI results, so experts such as doctors or other medical personnel are still required to read them. Many other difficulties, such as complicated backgrounds and the presence of multiple potential abnormalities, make clinical analysis of X-ray images a difficult task. [3] As a result, manual annotations from experts (radiologists) are required.

Automatic X-ray image analysis is quickly becoming a valuable clinical diagnostic tool. Deep neural networks have recently achieved image classification success and are now widely used for X-ray image classification tasks [4], [5]. Deep learning on chest X-ray images can be used to classify a variety of diseases, including thoracic infections [4], COVID-19 [4]–[7], and lung disorders [8]. Husayn et al. [9] proposed CoroNet, a deep learning model for COVID-19 detection. A deep learning framework has been proposed to detect lung abnormalities in CXR and CT scan images [8]. Using GAN-based synthetic data, Albahli et al. [10] proposed a deep learning model that achieved 87% accuracy and produced results comparable to other techniques. Lung segmentation is another critical task in CXR disease detection. This is extremely useful for determining the severity of tuberculosis [11]. DeTraC's conventional deep neural network architecture is described by Abbas et al. [5].

II. LITERATURE REVIEW

The classification of lung diseases using X-ray images has been explored using various machine learning techniques. A hierarchical classification can be useful in detecting pneumonia due to the disease's hierarchical pattern [12]. Traditional machine learning approaches, such as Support Vector Machine (SVM), k-Nearest Neighbor (KNN), and Decision Tree classifiers, can be used to classify diseases in Chest X-Ray (CXR). They do, however, rely on a mechanism for feature extraction. Convolutional Neural Networks (ConvNets) are another mechanism for feature extraction. Toğacar et al. [13] compared the performance of traditional machine learning models for detecting pneumonia with the Redundancy Maximum Relevance (mRMR) minimum feature selection mechanism. Khatri et al. [14] compared CXR pneumonia images using the Earth Mover's Distance (EMD). Teixeira et al. [15] evaluated and described COVID-19 using lung segmentation. A multimodal approach to disease on CXR can aid in better understanding and elucidation. [16], [17].

Due to a lack of large amounts of annotated data and efficient machine learning algorithms to study some specific features, automatic disease classification on X-ray images is difficult [25]. A variety of data streams and modalities can be used to improve disease prediction accuracy. To train thoracic disease classifiers, text data from diagnostic X-ray images are combined with annotated image data [17]. A deeply decomposed generative model, as opposed to the traditional approach of directly classifying disease, can be used to generate residual maps for abnormal disease alongside normal images [18]. This method aids in distinguishing between abnormal and normal chest X-ray parts. The semi-supervised
generative model is effective for CXR disease classification [18], [19].

Multiple diseases can still occur at the same time. Single-label classification may be ineffective in this situation, whereas multi-label classification may be effective Albali et al. [20] proposed a CNN-based deep learning method for CXR multi-label classification. In addition, Baltruschat et al. [21] compared several multi-label chest radiograph classification methods based on deep learning. Pathology datasets are rife with class imbalances. Pathology datasets are rife with class imbalances. As a result, the trained classifier is vulnerable to bias towards the majority class. Appropriate class balancing measures can help improve classifier performance in supervised and semi-supervised tests [22]. The limitations of existing machine learning-based approaches for COVID-19 detection are discussed by López-Cabrera et al. [23] Tsiknakis et al. [24] present an artificial intelligence-based framework for COVID-19 screening that can be interpreted using CXR imagery.

The research aims to make the following main contribution based on the shortcomings of several researchers’ classification methods:

1) To augment the limitations of the dataset generated from several hospitals in Semarang, especially for the types of COVID-19 and Tuberculosis (TBC) diseases.

2) Perform classification Performance comparisons with new approach machine learning models without selecting features from each image, so that the best machine learning algorithm model is produced based on its Performance.

III. METHODOLOGY

A. Dataset

Fig. 1 depicts the imbalance in the number of X-ray images for each class. Normal cases outnumber COVID-19 and Tuberculosis cases. This was seen as a disadvantage by researchers because the model worked better in major classes (Normal) than in minor classes. FP (normal classified as COVID-19 or TB) would be expensive, but it was much less expensive than FN (normal classified as COVID-19 or TB), which could be fatal. Therefore, the researchers addressed the class imbalance. When building a reliable image classifier with very little training data, image augmentation was usually required to improve deep network Performance. Image augmentation generated training images artificially by using various processing methods or combinations of processing methods on each training sample, such as random rotation, shift, sliding, and flipping [15]. To accomplish this, Image Data Generator would automatically label all data inside the COVID-19 or TBC folder as COVID-19 or TBC, and all data inside the Normal folder as Normal. As a result, the data was easily ready to be fed into machine learning.

B. Data Augmentation

Data augmentation was used by researchers to increase the number of samples and diversify the images in terms of position, orientation, brightness, and so on. In this study, the augmentation technique was the best method for adding data without having to look for primary data obtained at the hospital using strict procedures. Several augmentation techniques were employed in this study, including rotation range = 40, shear range = 0.2, zoom range = 0.2, width shift range = 0.2, height shift range = 0.2, and horizontal flip. Fig. 2 shows the X-ray images for each class. Fig. 3 depicts the augmentation results.

As shown in Table I, we collected lung disease images from several hospitals in Semarang for three classes: Normal, COVID-19, and Tuberculosis (TB) and divided them into training and testing sets.

<p>| TABLE I. DATASET DISTRIBUTION TABLE FOR EACH CLASS |
|---------------------------------|------------|------------|---|</p>
<table>
<thead>
<tr>
<th>Class</th>
<th>Training Set</th>
<th>Testing Set</th>
<th>Total</th>
</tr>
</thead>
<tbody>
<tr>
<td>Normal</td>
<td>8,454</td>
<td>476</td>
<td>8,930</td>
</tr>
<tr>
<td>COVID-19</td>
<td>2,363</td>
<td>490</td>
<td>2,853</td>
</tr>
<tr>
<td>Tuberculosis</td>
<td>474</td>
<td>18</td>
<td>492</td>
</tr>
<tr>
<td>Total</td>
<td>11,291</td>
<td>984</td>
<td>12,275</td>
</tr>
</tbody>
</table>

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prediction based on neighboring data values. The letter ‘K’ in KNN stands for the number of nearest neighbor values in the data. The KNN algorithm classifies a given dataset based on ‘K,’ or the number of nearest neighbors [25]. The Euclidian distance calculation method is the most commonly used. Equation (1) gives the Equation for Euclidean calculations.

\[ Euclidean_{i,j} = \sqrt{\sum_{k=1}^{n}(x_{ik} - x_{jk})^2} \]  

The KNN algorithm is composed of four steps. The distance from the new data to all data is calculated in the first step. The distances are then sorted in the second step. The third step finds the smallest k value, and the final step determines the class.

B. Extra Tree

Extra Trees is an ensemble machine learning approach that trains multiple decision trees and combines the results of a group of decision trees to generate predictions. However, there is a distinction between Extra Trees and Random Forest. To ensure that the decision trees are sufficiently distinct, Random Forest used bagging to select different variations of the training data. Extra Trees, on the other hand, used the entire data set to train decision trees. Therefore, it randomly selected values to split features and created child nodes in order to ensure sufficient differences between each decision tree. Extra trees can reduce model bias by using the entire dataset, which is the default setting and can be changed. On the other hand, randomizing the feature values to be split increased both bias and variance.

C. Support Vector Machine (SVM)

This stage involved using the SVM classification method to generate a classifier in the form of a feature vector, which was then used to generate predictions for testing. Following the completion of the preceding stages, the extraction results were used to generate an SVM classification model. The distance of the vector that had been mapped would be calculated. The greatest distance would be used as the vector’s class separator. Then a hyperplane was added to separate the two classes. The most important requirement for developing an SVM classification model was to convert documents into vector form. Value variations were used to find values that were as accurate as possible. This procedure was necessary in order to convert the test document into a vector. The following step was to take the test document data, which was commonly referred to as a vector, and insert it into the previously created SVM model. The hyperplane Equation was a classification Equation, as shown by Equation (2), with the classification parameters w and b as the weight and bias values, as shown by Equations (3) and (4).

\[ f_{svm}(x) = w \cdot x + b \]  
\[ w = \sum_{i=1}^{N} a_i y_i x_i \]  
\[ b = -\frac{1}{2} (w \cdot x^+ + w \cdot x^-) \]  

Where N is the amount of data and ai is the weight coefficient value for each pair of data points and labels (xi,). SVM is also good at dataset management because it finds
hyperplanes using trick kernels, one of which is the Linear Kernel shown in the Equation (5).

\[ K(x_i, x_j) = x_i^T x_j \]  

From the kernel results obtained, SVM creates a classification Equation that is adjusted to the kernel used, such as Equation (6) with class classification values based on Equation (7) [25].

\[ f_{svm}(x) = \sum a_i y_i K(x_i, x) + b_i \in N \]  
\[ \text{class} = \begin{cases} 1, & f_{svm} \geq 0 \\ -1, & \text{other than that} \end{cases} \]

### D. Random Forest

Random forest is a classification method that consists of a collection of classification trees. For example \( \{h(x, \Theta_k), k = 1, \ldots \} \) where \( \{\Theta_k\} \) is a random vector that is independently distributed and each tree chooses the class that has the most number of data (majority cote). Suppose an ensemble \( h_1(x), h_2(x), \ldots, h_n(x) \) with training data is randomly selected from the distribution of random vectors \( Y \) and \( X \), the margin function \( mg(X, Y) \) of the random forest is defined as follows:

\[ mg(XY) = \frac{\sum_{k=1}^{n} I(h_k(X)=Y)}{K} - \max_{j \neq Y} \left[ \frac{\sum_{k=1}^{n} I(h_k(X)=j)}{K} \right] \]

where, \( I \) is the indication function and \( K \) is the number of trees. The margin function is used to calculate the level of the number of votes in \( X \) and \( Y \), as well as the average vote from other classes [26].

### V. RESULT AND ANALYSIS

This study used a primary dataset with a data ratio of 70:30 for each class, as shown in Table II. A total of 11291 data points were generated in each training data set by producing the Extra Trees model with Estimator = 500 and values for precision, recall, F1-score, and accuracy of 100%, followed by other machine learning models such as SVM and Random Forest. Table III shows that the Extra Trees model has the best performance value when compared to other machine learning models, with a total of 984 data points for testing.

Fig. 6(a) shows the prediction column, and thus the row must be the actual value. In the training data, the main diagonal (8030, 2135, 4070) of the confusion matrix of the Kneighbors model (n neighbors=3) gives the correct prediction. When the actual and predicted values from the model are the same, this is the case. The actual normal number is on the first line. The model predicts 8030 cases, 358 of which are COVID-19 normal and 66 of which are Tuberculosis normal. The actual COVID-19 number can be found in the second row. The model predicts COVID-19 correctly and incorrectly, with 201 COVID-19 becoming normal and 27 COVID-19 becoming TB. Tuberculosis is in the third row. The model predicts that 470 of them will correctly predict TB, 1 will become normal, and 3 will become COVID-19.

Fig. 6(b) depicts the prediction column, and thus the row must be the actual value. The main diagonal (330, 464, 18) for the confusion matrix of the Kneighbors model (n neighbors=3) testing data gives the correct prediction. This is the case when the actual and predicted values from the model are the same. The first line contains the true normal number. According to the model, 330 of them correctly and incorrectly predict 159 normal to COVID-19 and 2 normal to be Tuberculosis. The actual COVID-19 number is in the second row. The model predicts that 464 of them will correctly and incorrectly predict COVID-19, that 10 COVID-19 will become normal, and that 1 COVID-19 will become TB. Tuberculosis is in the third row. The model predicted that 18 of them would be correct and 1 would be normal.

### TABLE II. COMPARISON OF TRAINING DATA PERFORMANCE CLASSIFICATION WITH A COMPARISON OF TRAINING AND TESTING DATA 70:30

<table>
<thead>
<tr>
<th>Classification Models</th>
<th>Precision (%)</th>
<th>Recall (%)</th>
<th>F1 (%)</th>
<th>Acc (%)</th>
</tr>
</thead>
<tbody>
<tr>
<td>Kneighbors (n_neighbors=5)</td>
<td>89</td>
<td>95</td>
<td>92</td>
<td>94</td>
</tr>
<tr>
<td>Kneighbors (n_neighbors=5)</td>
<td>84</td>
<td>93</td>
<td>88</td>
<td>92</td>
</tr>
<tr>
<td>Kneighbors (n_neighbors=7)</td>
<td>81</td>
<td>92</td>
<td>85</td>
<td>91</td>
</tr>
<tr>
<td>ExtraTrees (n_estimators=500)</td>
<td>100</td>
<td>100</td>
<td>100</td>
<td>100</td>
</tr>
<tr>
<td>SVM</td>
<td>100</td>
<td>100</td>
<td>100</td>
<td>100</td>
</tr>
<tr>
<td>Random Forest (n_estimators=1000)</td>
<td>100</td>
<td>100</td>
<td>100</td>
<td>100</td>
</tr>
</tbody>
</table>

### TABLE III. COMPARISON OF DATA TESTING PERFORMANCE CLASSIFICATION WITH A RATIO OF 70:30

<table>
<thead>
<tr>
<th>Classification Models</th>
<th>Precision (%)</th>
<th>Recall (%)</th>
<th>F1 (%)</th>
<th>Acc (%)</th>
</tr>
</thead>
<tbody>
<tr>
<td>Kneighbors (n_neighbors=5)</td>
<td>76</td>
<td>77</td>
<td>76</td>
<td>80</td>
</tr>
<tr>
<td>Kneighbors (n_neighbors=5)</td>
<td>76</td>
<td>79</td>
<td>76</td>
<td>80</td>
</tr>
<tr>
<td>Kneighbors (n_neighbors=7)</td>
<td>74</td>
<td>80</td>
<td>75</td>
<td>79</td>
</tr>
<tr>
<td>ExtraTrees (n_estimators=500)</td>
<td>89</td>
<td>88</td>
<td>87</td>
<td>85</td>
</tr>
<tr>
<td>SVM</td>
<td>83</td>
<td>74</td>
<td>77</td>
<td>80</td>
</tr>
<tr>
<td>Random Forest (n_estimators=1000)</td>
<td>86</td>
<td>88</td>
<td>85</td>
<td>83</td>
</tr>
</tbody>
</table>
Fig. 7(a) depicts the prediction column, implying that the row must represent the actual value. The main diagonal (7997, 1981, 417) for the training data gives the correct prediction for the confusion matrix of the Kneighbors model (n_neighbors=5). In this case, the actual and predicted values from the model are the same. The standard number appears on the first line. The model correctly predicts 7997 of them, 510 of which are normal to COVID-19 and 92 of which are normal to Tuberculosis. The second row contains the actual COVID-19 number. The 1981 predictive model included correctly and incorrectly predicting COVID-19, with 233 COVID-19 being normal and 52 COVID-19 being Tuberculosis. The third row is Tuberculosis. The model predicted that 417 of them would correctly predict TB, 2 TB would be normal, and 5 TB would be COVID-19.

Fig. 7(b) depicts the prediction column, and thus the row must be the actual value. For the Kneighbors model matrix confusion (n_neighbors=5) data testing, the main diagonal (321, 415, 13) yields the correct prediction. When the actual and predicted values from the model are the same, this is the case. The actual normal number is on the first line. The model predicts that 321 of them will be correct, with 168 normal to COVID-19 and 5 normal to Tuberculosis. The actual COVID-19 number is in the second row. The model predicts that 451 of them will correctly predict COVID-19, 19 will become normal, and 3 will develop Tuberculosis. Tuberculosis is in the third row. The model predicts that 13 of them will correctly predict TB, and four of them will become COVID-19.

Fig. 8(b) depicts the predicted column, and thus the row must be the actual value. For the Kneighbors model matrix confusion (n_neighbors=7) data testing, the main diagonal (327, 435, 12) yields the correct prediction. When the actual and predicted values from the model are the same, this is the case. The actual normal number is on the first line. The model predicts that 327 of them will correctly predict, 185 will be normal for COVID-19, and 6 will be normal for Tuberculosis. The actual number of COVID-19 is shown in the second row. The model predicts that 435 of them will correctly predict COVID-19, 13 will become normal, and 3 will develop Tuberculosis. Tuberculosis is in the third row. The model predicts that 12 of them will correctly predict TB and 4 will correctly predict COVID-19.

Fig. 9(a) depicts the prediction column, implying that the row must represent the actual value. The training data for the ExtraTrees model confusion matrix (n_estimators=500) gives the correct prediction along the main diagonal (8232, 2496, 563). In this case, the model's actual and predicted values are the same. The true normal number is found on the first line. The model correctly predicts 8232 of them. The actual COVID-19 number is in the second row. 2496 of them correctly predict COVID-19, according to the model. Tuberculosis is in the third row. The model correctly predicted TB in 563 of them.
Fig. 9(b) shows the predicted column, and thus the row must be the actual value. The main diagonal (331, 488, 19) for the ExtraTrees model confusion matrix (n estimators=500) testing data provides the correct prediction. This is the case when the model’s actual and predicted values are the same. The actual normal number appears on the first line. The model predicts that 331 of them will correctly predict COVID-19, 134 will correctly predict Tuberculosis, and 2 will correctly predict Tuberculosis. The COVID-19 number is in the second row. The model predicts that 488 of them will correctly predict COVID-19 and 9 will correctly predict COVID-19 as normal. Tuberculosis is in the third row. 19 of them correctly predict TB, with 1 TB being COVID-19, according to the model.

![Confusion Matrix](image)

Fig. 10. Confusion matrix for training and testing models for Support Vector Machine (SVM).

Fig. 10(a) shows the prediction column, and thus the row must be the actual value. The Support Vector Machine (SVM) data model confusion matrix's main diagonal (8232, 2496, 563) yields correct predictions. This is the case when the model's actual and predicted values are the same. The true normal number is found on the first line. The model correctly predicts 8232 of them. The actual COVID-19 number is in the second row. 2496 of them correctly predict COVID-19, according to the model. Tuberculosis is in the third row. The model correctly predicted TB in 563 of them.

Fig. 10(b) shows the predicted column, implying that the row must be the actual value. For the Confusion Support Vector Machine (SVM) model testing data, the main diagonals (331, 488, 19) provide the correct predictions. When the model's actual and predicted values are the same, this is the case. The true normal number is on the first line. The model predicts that 331 of them will correctly predict, 134 will become COVID-19, and 2 will become Tuberculosis. The actual COVID-19 number is in the second row. The model predicts that 488 of them correctly predict COVID-19, with 9 of them becoming normal. Tuberculosis is in the third row. The model predicts that 19 of them will correctly predict TB, with one TB being COVID-19.

Fig. 11(a) shows the predicted column, and thus the row must be the actual value. The training data for the Random Forest model confusion matrix (n estimators=500) gives the correct prediction along the main diagonal (8232, 2496, 563). This is the case when the model's actual and predicted values are the same. The true normal number is found on the first line. The model correctly predicts 8232 of them. The actual COVID-19 number is in the second row. 2496 of them correctly predict COVID-19, according to the model. Tuberculosis is in the third row. The model correctly predicted TB in 563 of them.

![Confusion Matrix](image)

Fig. 11. Confusion matrix for training and testing models for random forest (SVM).

Fig. 11(b) depicts the prediction column, and thus the row must be the actual value. The main diagonals (330, 464, 18) for the Random Forest Confusion model (n estimators=500) test data give the correct predictions. This is the case when the model’s actual and predicted values are the same. The actual normal number appears on the first line. The model predicts that 330 of them will be correct, with 159 being normal and two being Tuberculosis. The COVID-19 number is in the second row. The model predicts that 464 of them will correctly predict COVID-19, 10 will be normal, and 1 will be Tuberculosis. Tuberculosis is the third row. The model correctly predicted TB for 18 of them.

Table IV shows comparison of the accuracy values with several researchers. Bakir et al applied deep learning techniques to detect pneumonia from X-ray images. They used an artificial neural network (ANN) model to classify bacterial, viral, and healthy lungs into multiple classes. His proposed ANN model with ResNet feature extraction, in multi-class classification he achieved 81.67% classification accuracy [27]. Kim, Sungyeup et al used chest X-ray (CXR) images to diagnose three classes, normal, pneumonia and pneumothorax. The method used uses a deep learning model (EfficientNet V2-M) to produce an accuracy value of 82.15% [28].

<table>
<thead>
<tr>
<th>Prior Work</th>
<th>Model</th>
<th>Acc (%)</th>
</tr>
</thead>
<tbody>
<tr>
<td>Bakir et al [27]</td>
<td>ANN model</td>
<td>81.67%</td>
</tr>
<tr>
<td>Kim, Sungyeup et al [28]</td>
<td>EfficientNet V2-M</td>
<td>82.15%</td>
</tr>
<tr>
<td>Proposed Method</td>
<td>ExtraTrees</td>
<td>85%</td>
</tr>
</tbody>
</table>

TABLE IV. COMPARISON OF ACCURACY VALUE WITH OTHER RESEARCHERS
VI. CONCLUSION

Through the use of multiple machine learning techniques, this study contributes to the multi-class classification of lung diseases. The research methodology used in this study is an experimental multi-class classification of lung disease using an augmentation process to obtain balanced data.

The experimental results produce the Extra Trees classification which has Precision, Recall, F-Score, and Accuracy score of 100% for training, and testing data, 89% for Precision, 88% for Recall, 87% for Fi-Score, and 85% for Accuracy higher than the Performance of other machine learning models such as Kneighbours, Support Vector Machine (SVM), Random Forest and more effective in Classification of lung diseases. Comparison with other researchers shows that the proposed model has a higher accuracy value compared to other models.

In further research, the dataset as a whole is still not large enough and of low quality for highly accurate and useful deep learning results that can be used as benchmarks for the identification of lung disease types by viewing X-rays. More high-quality images are needed to increase the accuracy of average ratings to higher levels in multiple-class classification. The data is expected to grow over time, enabling better classification results.

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REFERENCES


Optimizing YOLO Performance for Traffic Light Detection and End-to-End Steering Control for Autonomous Vehicles in Gazebo-ROS2

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FPT University, Can Tho 94000, VietNam

Abstract—Autonomous driving has become a popular area of research in recent years, with accurate perception and recognition of the environment being critical for successful implementation. Traditional methods for recognizing and controlling steering rely on the color and shape of traffic lights and road lanes, which can limit their ability to handle complex scenarios and variations in data. This paper presents an optimization of the You Only Look Once (YOLO) object detection algorithm for traffic light detection and end-to-end steering control for lane-keeping in the simulation environment. The study compares the performance of YOLOv5, YOLOv6, YOLOv7, and YOLOv8 models for traffic light signal detection, with YOLOv8 achieving the best results with a mean Average Precision (mAP) of 98.5%. Additionally, the study proposes an end-to-end convolutional neural network (CNN) based steering angle controller that combines data from a classical proportional integral derivative (PID) controller and the steering angle controller from human perception. This controller predicts the steering angle accurately, outperforming conventional open-source computer vision (OpenCV) methods. The proposed algorithms are validated on an autonomous vehicle model in a simulated Gazebo environment of Robot Operating System 2 (ROS2).

Keywords—Yolo models; PID; CNN; gazebo; ROS2; traffic-light; lane-keeping; autonomous

I. INTRODUCTION

The increasing number of vehicles on the road has raised concerns about traffic accidents and fatalities caused by various factors [1]. To address this, research has been conducted to create technologies that enhance driving safety, such as Advanced Driver Assistance Systems (ADAS) and autonomous driving systems. Traffic light signal recognition is a crucial component of these systems as it helps to detect the current status of traffic lights and provides real-time information for the vehicle control system to make accurate decisions. The conventional approaches for early detection of traffic lights rely on identifying manual features and color characteristics of signal lights, as [2]-[7]. These methods typically employ feature matching, color matching, or similar techniques to detect traffic lights based on their shape and color. Other approaches involve utilizing offline location information, which includes traffic light data from maps and GPS-based data, to track both the vehicle's present location and the status of traffic lights, as discussed in references [8]-[10]. Conventional approaches to detecting traffic lights are constrained by technical aspects like the type of camera used and the surrounding installation conditions. Additionally, these methods rely on offline location data that needs to be constantly refreshed and is susceptible to security risks.

In recent years, deep learning approaches, such as convolution neural network (CNN) [11], single shot multibox detector (SSD) [12], or YOLO architecture [13]-[15], have been used to accurately identify traffic lights and provide a real-time solution for traffic light detection. Among these approaches, the YOLO method has emerged as the best performer in detecting and recognizing traffic signals. Compared to other deep learning techniques, it offers smoother, more accurate results, and achieves real-time performance. Joseph et al. presented YOLO and an enhanced version called YOLOv2, which were utilized for detecting and classifying traffic lights [16]-[17]. Possatti et al. employed YOLOv3 [18] along with prior maps to identify crucial states of traffic lights for vehicles in their investigation. Tai et al. made advancements to YOLOv4 to enhance its precision in classifying green, red, and yellow traffic lights [19]. However, the past year has seen the release of a series of updated versions of YOLO, including YOLOv5 [20] [21], YOLOv6 [22], YOLOv7 [23], and YOLOv8 [24], each of which shows improved accuracy and performance in implementation on the COCO dataset.

To apply these improved models in the field of autonomous vehicles, we present our traffic light detection algorithm that utilizes the latest version of YOLO, YOLOv8 [25]. Additionally, we demonstrate the superior performance of YOLOv8 compared to other models (YOLOv5-v6-v7), in terms of accuracy and real-time processing. We conduct experiments on the traffic light dataset obtained and augmented data from multiple sources, including the CinTA_v2 dataset, and GathoTF datasets shown in Fig. 1.

Furthermore, research focused on utilizing camera images for angle prediction in autonomous vehicles to ensure proper lane-keeping has garnered considerable interest and made significant advancements. Earlier research employed image processing techniques, including color and edge detection, as well as critical lane regions such as the Canny edge detector, Hough Transform technique, and Isolate Region of Interest (IROI), to predict the steering angle [26]-[29]. The steering angle was then used in combination with a classic PID controller to ensure lane-keeping [30]-[33]. However, when faced with more complex scenarios, such as intersections, where lane images may be interrupted, a hybrid angle
controller must be employed to account for driver perception. To overcome these limitations, autonomous vehicle control techniques have emerged that utilize deep learning to emulate human driving behavior.

![Traffic light dataset](image1)

In this research, we present a YOLOv8 architecture for traffic light recognition combined with an end-to-end CNN steering controller for lane-keeping in a simulated Gazebo environment of Robot Operating System 2 (ROS2). While the vehicle is following the lane, the traffic light signals are also detected and classified to send to the central controller. The primary contributions of this research are:

- The new YOLOv8 model is applied in our traffic light detection algorithm. Using the multiple source traffic light dataset, the YOLOv8 model is trained, evaluated, and compared accuracy and real-time performance with previous models.
- Based on the end-to-end convolution network, a steering angle predictor is designed. The model not only re-learns human driving behavior but is also trained with a data set of OpenCV+PID steering angle control methods to increase the accuracy of lane keeping and to give a steering angle when the lane is discontinuous.
- To assess the effectiveness of our proposed method for traffic light detection and lane-keeping, we collected a dataset by driving a donkey car model following the lane and incorporating traffic lights into the Gazebo-ROS2 simulation environment. We compared the performance of our method with that of previous methods using the same dataset.

This paper is structured as follows: Section II describes the architecture of YOLO models used for traffic light detection and classification. Section III provides a detailed description of the end-to-end CNN steering angle predictor, including the data collection process, evaluation metrics, and network architecture. Section IV presents the experimental results and provides a discussion of the findings. Finally, Section V concludes the paper and suggests areas for future research.

II. TRAFFIC LIGHT DETECTION AND CLASSIFICATION USING YOLO MODELS

A. Preprocessing Data

1) Data collection: To evaluate the effectiveness of YOLO in traffic light detection, experiments are conducted using a synthetic dataset, which was collected from two different sources, the CinTA_v2 traffic light datasets, and GathoTF datasets.


The CinTA_v2 dataset is a freely available traffic light dataset that was public in Roboflow. It contains 999 images captured under various weather and lighting conditions, and for this study, 999 images were randomly picked for training and evaluation. Each image has a resolution of 1280 x 960 pixels.

The GathoTF dataset consists of 2025 images and includes traffic light images taken in Can Tho city and traffic light images taken in a virtual environment created by the Gazebo/ROS2 program. In this environment, a self-driving car was simulated, and a camera mounted on the car took pictures at a frame rate of 30 fps and a resolution of 1024 x 600 pixels.
The YOLOv5 algorithm is a fast and efficient object detection system that uses anchor-free detectors, a CSPDarknet53 backbone, a PAN neck, and AutoML for anchor box optimization and employs a Mosaic data augmentation technique to improve generalization.

YOLOv6 [37] is an updated version of the YOLO algorithm that incorporates a RepVGG backbone, VariFocal Loss for dynamically adjusting object contribution during training, SlOU and GloU for classification and regression loss, and a Focal Attention mechanism for improved region detection.

YOLOv7 [38] is an object detection algorithm that uses an Extended Efficient Layer Aggregation backbone with group convolution, a Gradient Flow Propagation module for re-parameterization, and an auxiliary head for improved prediction accuracy.

YOLOv8 [39] is an object detection algorithm that uses a CSP-inspired C2f module instead of the C3 module, allowing for more abundant gradient flow information while maintaining a lightweight design.

Using our datasets to train and test evaluate its effectiveness in traffic light recognition and detection. The variety of datasets enabled us to enhance and improve the model’s resilience and generalizability by exposing it to a variety of different scenarios and conditions.

**Algorithm 1: Data augmentation**

**Input:** Load the dataset of traffic light images and their corresponding labels.

1. For each image in the dataset, randomly select one or more of the following augmentation techniques:
   1.1: **Image flipping:** randomly flip the image horizontally or vertically.
   1.2: **Image rotation:** randomly rotate the image by 15 degrees to the right or left.
   1.3: **Blur:** apply a Gaussian blur with a kernel size of 8% to the image.
   1.4: **Random cropping:** randomly crop the image by 3%.
   1.5: **Noise generation:** add random Gaussian noise with a standard deviation of 8% to the image.
   1.6: **Decolorization:** randomly remove 7% of the color channels from the image.

2. Apply the selected augmentation techniques to the image and save the new image as a separate file in the dataset folder.

3. Update the label of the new image with the same label as the original image.

4. Repeat steps 1-3 for all images in the dataset.

**Result:** Save the augmented dataset and use it to train and test the traffic light recognition system.

2) **Data augmentation:** In order to enhance the precision of the traffic light recognition system and expand the dataset, Algorithm 1 was implemented.

Following the augmentation procedure, the labeled data was reintegrated into the dataset, thereby augmenting the quantity of inherent angle data. The labeled data was then used to train a YOLO model. As a result of the data augmentation techniques, the original dataset of 4017 images was expanded to include 6695 additional images, resulting in a more diverse and robust dataset. Fig. 2 shows the example of the dataset after the augmentation. Next section, the different YOLO models are compared and analyzed when applying these models to traffic light detection and classification.

**B. Detection and Classification using YOLO Models**

This study employs four versions of YOLO architecture for traffic light recognition. YOLO architecture, originally introduced by Redmon et al. (2016) [34], approaches the detection task as a regression problem based on the Darknet architecture. Unlike popular Region Proposal Networks (RPN), YOLO predicts both bounding boxes (Bbox.) and class probabilities (Cls.) in a single network. This approach is based on a user-defined size grid cell responsible for detecting the object if it falls into the cell. To accurately evaluate the performance of these YOLOv models, we analyze the differences as well as the advantages and disadvantages of each model before applying them to the identification and classification of traffic lights. The comparison between structures of YOLOv5, YOLOv6, YOLOv7, and YOLOv8 are shown in Table I. The features of the modules are as follows:

- **YOLOv5** is an advanced object detection algorithm that utilizes anchor-free detectors to detect multiple objects in real-time [20], [35]. It uses a CSPDarknet53 backbone and PAN (Path Aggregation Network).
- Network for faster and more efficient detection. The Mosaic data augmentation technique combines multiple images into a single image to enhance generalization, while AutoML optimizes anchor boxes of varying sizes and aspect ratios for each grid cell. YOLOv5 offers superior speed and accuracy in object detection compared to other methods.

**TABLE I. DETAILED DIFFERENCES BETWEEN YOLO MODELS**

<table>
<thead>
<tr>
<th>Model</th>
<th>Backbone</th>
<th>Neck</th>
<th>Head</th>
<th>Loss Function</th>
</tr>
</thead>
<tbody>
<tr>
<td>YOLOv5 [36]</td>
<td>CSPDarkent53</td>
<td>PAN</td>
<td>B x (5 + C) output layer B: No of bounding boxes C: Class score</td>
<td>Binary Cross Entropy and Logit Loss Function</td>
</tr>
<tr>
<td>YOLOv6 [37]</td>
<td>RepVGG And CSPRepStack</td>
<td>RepPAN</td>
<td>Decoupled Classification and Detection Head</td>
<td>VariFocal Loss for Classification and Distribution Focal Loss for Detection</td>
</tr>
<tr>
<td>YOLOv7 [38]</td>
<td>EELAN</td>
<td>PAN</td>
<td>Lead Head for final output, Auxiliary Head for middle layer outputs</td>
<td>BCE with Focal Loss for Classification, IoU loss for Detection</td>
</tr>
<tr>
<td>YOLOv8 [39]</td>
<td>CSP same as that of YOLOv5 but C3 module replaced by C2f module</td>
<td>PAN-FPN</td>
<td>Decoupled-Head</td>
<td>VFL Loss and DFL Loss+ClOU</td>
</tr>
</tbody>
</table>

**TABLE I. DETAILED DIFFERENCES BETWEEN YOLO MODELS**
The YOLOv7 model incorporates an innovative architecture known as Extended Efficient Layer Aggregation (EELA) as its backbone. This novel design employs group convolution to broaden the computational block's channel, thereby enhancing the model's overall performance \[38\]. The algorithm also uses a new module called the Gradient Flow Propagation (GFP) module, which helps to determine which modules need re-parameterization in order to improve the model's accuracy. Finally, YOLOv7 includes an auxiliary head that is designed to provide a coarse-to-fine definition for better predictions lightweight.

The Backbone part of YOLov8 is basically the same as that of YOLOv5, and the C3 module is replaced by the C2f module based on the CSP idea \[39\]. The C2f module learned from the ELAN idea in YOLOv7 and combined C3 and ELAN to form the C2f module so that YOLOv8 could obtain more abundant gradient flow information while ensuring lightweight. We use the model YOLOv8 for detecting, and classification traffic lights, as shown in Fig. 3. Fig. 4 is a comparison of the structure of CS_X (YOLOv5) and C2f (YOLOv8) in the backbone.

Four YOLO models are proposed to compare and test the applicability of these models to traffic light signal recognition for autonomous vehicle systems. There are a total of three active traffic light states: $s = \{\text{red, yellow, green}\}$. The models will be trained on our augmentation datasets.

### III. END-TO-END CNN STEERING ANGLE CONTROLLER

#### A. Preprocessing Data

1) Data collection: To ensure the vehicle is always in the center of the lane, a series of multi-step image processing using the OpenCV method and a PID steering controller is applied \[33\]. The steering angle will be predicted after image processing to detect the two lines of the lane. Combined with the PID controller, the vehicle will move more precisely. At this point, we will collect the input image data and the output steering angle to train the end-to-end CNN steering angle controller model. It contains 10,000 images captured from the camera mounted on the vehicle in the simulation environment Gazebo/ROS2. However, when the car went through the intersection and encountered complicated situations, the estimated steering angle from the lane detection was interrupted, so we used the joystick to control the car with
human perception. In this case, 10,000 pictures continue to be collected along with the steering angle from human perception. Fig. 5 shows the dataset for the end-to-end CNN steering angle controller model.

2) Data augmentation and normalization: To improve the accuracy and avoid issues with gradient explosion or vanishing of the lane-keeping model, data augmentation and normalization techniques are employed as Algorithm 2.

```
Algorithm 2: Data Augmentation and Normalization

Input: Training Dataset

1. AUGMENT (image, steering angle):
   1.1. if random () < 0.5;
       image = pan(image) // crop out a smaller image from the left or right side
   1.2. if random () < 0.5;
       image = zoom(image) // crop out a smaller image from the center
   1.3. if random () < 0.5;
       image = blur(image)  // a Gaussian blur
   1.4. if random () < 0.5;
       image = adjust_brightness(image) // adjust brightness of the image
   1.5. image, steering angle = random flip (image, steering angle) // perform a horizontal flip on the image, which means flipping it from left to right, and adjust the corresponding steering angle accordingly.
   1.6. return image, steering angle

2. NORMAL (image):
   2.1. height = image.height.
   2.2. image = image[height/2+100] //Remove top half of the image, as it is not relevant for the lane following
   2.3. image = convert_Color(image, cv2.COLOR_RGB2YUV) //The optimal choice is to utilize the YUV color space.
   2.4. image = GaussianBlur(image, (3,3), 0)
   2.5. image = resize (image, (200,66)) // input image size (200,66) our model
   2.6. image = image / 255 # normalizing, the processed image becomes black for some reason
   2.7. return image

Result: Augmented and normalized image data with updated steering angles.
```

As depicted in Fig. 6, our dataset has been subjected to both normalization and augmentation. Meanwhile, the distribution of the steering angle within our dataset has been illustrated in Fig. 7.

B. Lane Keeping Using End-to-End CNN Model

The proposed network architecture used an end-to-end neural network called Nvidia_model, Fig. 8 shows a convolutional neural network architecture with five Conv2D layers, followed by a flattened layer and three fully connected (FC) layers, and an output layer with a single neuron.

![Convolutional neural network architecture](image)

```
Input: (1200, 600, 3)

Conv2D: (31, 98, 24) – Param # 1824
Conv2D_1: (14, 47, 36) – Param # 21636
Conv2D_2: (5, 22, 48) – Param # 43248
Conv2D_3: (3, 20, 64) – Param # 27712
Conv2D_4: (1, 18, 64) – Param # 36928
Flatten: (1152)

FC: (1164) – Param # 1342092
FC_1: (100) – Param # 116500
FC_2: (50) – Param # 5050
FC_3: (10) – Param # 950

Output: (1)
```

Fig. 8. Convolutional neural network architecture.
The input to the network is a 3-dimensional tensor with shape (1200, 600, 3), which suggests that the input is an image with a width of 1200 pixels, a height of 600 pixels, and 3 color channels (e.g., RGB). The first layer in the network is a convolutional layer with 24 filters, each with a shape of (31, 98). This layer has 1,824 parameters, which are learned during training.

The next layer is another convolutional layer with 36 filters, each with a shape of (14, 47). This layer has 21,636 parameters. The third layer is a convolutional layer with 48 filters, each with a shape of (5, 22). This layer has 43,248 parameters. The fourth layer is a convolutional layer with 64 filters, each with a shape of (3, 20). This layer has 27,712 parameters. Finally, the fifth layer is a convolutional layer with 64 filters, each with a shape of (1, 18). This layer has 36,928 parameters.

After the last convolutional layer, the output is flattened into a 1-dimensional tensor with shape (1152). This flattened output is then fed into a fully connected (FC) layer with 1,164 units. This FC layer has 1,342,092 parameters. The output of this layer is then fed into another FC layer with 100 units, which has 116,500 parameters. The next FC layer has 50 units and 5,050 parameters. Finally, there is a FC layer with 10 units and 950 parameters. The total number of parameters in this model is 1,595,511. The output of the fourth fully connected layer is passed through and the output layer contains a single neuron, which predicts the steering angle.

IV. EXPERIMENTAL RESULT

A. Experimental System

To evaluate the performance of our proposed algorithms, we conducted experiments in a simulated environment using Gazebo-ROS2. The simulation was run on an Ubuntu 20.04 platform with an Intel Core i7 processor and 16 GB RAM. The simulated vehicle was equipped with a front-facing camera, simulated based on the actual parameters of the WGE100 camera. The camera captured images with a resolution of 1024x600 at a frame rate of 30 fps. The simulation environment was set up to include a variety of traffic light scenarios and lane configurations to test the robustness of our algorithms.

The experimental system consisted of two parts: traffic light detection and end-to-end steering control. For traffic light detection, we trained our YOLOv8 model using the CinTA_v2 and GathoTF datasets, which were augmented using Algorithm 1 to increase their size and diversity. We evaluated the performance of the model in terms of accuracy, precision, and recall on a test set of 20% of the total dataset. We also compared the performance of YOLOv5, YOLOv6, and YOLOv7 models for traffic light detection.

For end-to-end steering control, we designed a convolutional neural network-based steering angle controller that combines data from a classical PID controller and human perception. The model was trained on the same dataset used for traffic light detection and evaluated on a separate test set. We compared the performance of our model with that of a traditional PID controller and analyzed the results.

In order to assess the effectiveness of our proposed algorithms, we employed a Donkey Car model, in which the steering angle was controlled based on the predictions made by our model. The Donkey Car was driven on a predefined route in the simulated environment as shown in Fig. 9, which included a variety of traffic light scenarios and lane configurations. We collected data from the camera and the steering angle sensor to evaluate the performance of our algorithms in real time.

Overall, our experimental system allowed us to evaluate the effectiveness and robustness of our proposed algorithms for traffic light detection and end-to-end steering control in a simulated environment. The results of our experiments are presented and analyzed in the next section.

B. Evaluation Metrics

The use of evaluation metrics is critical in comparing and assessing the performance of machine learning algorithms. In this study, we compare four different object detection algorithms - YOLOv5, YOLOv6, YOLOv7, and YOLOv8, using various metrics such as F score, and mAP. To provide a comprehensive evaluation of the proposed algorithm's performance, we employ several evaluation metrics, including precision, recall, mAP, F-score, and FPS. However, to avoid potential biases in the evaluation process, we utilize different evaluation criteria that are based on different aspects of the algorithms' performance.

In order to evaluate the effectiveness of the proposed algorithm, this study utilizes several metrics including precision (P), recall (R), average precision (mAP), F1-Score, and Frames Per Second (fps). Precision is a critical metric used to assess the accuracy of the evaluation object by determining the ratio of correctly predicted positive samples to the total number of predicted positive samples.

\[ P = \frac{\text{TruePositive}}{\text{TruePositive} + \text{FalsePositive}} \]  

TruePositive indicates the count of positive samples accurately predicted as positive, whereas FalsePositive denotes the count of negative samples incorrectly predicted as positive.

Recall measures the ratio of correctly predicted positive samples to the total number of actual positive samples. It indicates whether the evaluation object is detected in its entirety or not.

\[ R = \frac{\text{TruePositive}}{\text{TruePositive} + \text{FalseNegative}} \]  

Recall measures the ratio of correctly predicted positive samples to the total number of actual positive samples. It indicates whether the evaluation object is detected in its entirety or not.
FalseNegative signifies the count of positive samples erroneously predicted as negative samples.

Mean average precision (mAP) is an essential metric utilized for evaluating the overall performance of the algorithm. It is computed by averaging the average precision (AveP) values across all classes.

\[ mAP = \frac{1}{n} \sum_{k=1}^{k=n} AveP_k \]  

(3)

AveP_k represents the average precision of class k, where n denotes the total number of classes.

The F1_Score is a metric that combines precision and recall into a single measurement by taking their weighted harmonic mean. It provides a balanced evaluation by considering both precision and recall.

\[ F1\_Score = (1 + \alpha^2) \frac{P \cdot R}{\alpha^2 \cdot P + R} \]  

(4)

The value of \( \alpha \) is employed to achieve a balanced weighting of precision and recall in the calculation of the F-score. A higher F1_Score implies that the algorithm has a better balance between precision and recall. Finally, Frames Per Second is a critical metric in evaluating the speed and efficiency of the algorithm. A higher fps score indicates that the algorithm can process a large number of frames in a shorter time.

In conclusion, using these evaluation metrics in this study provides a comprehensive and unbiased assessment of the performance of the object detection algorithms.

C. Traffic Light Detection and Classification Results

In this section, we present the results of the traffic light detection and classification experiments using different YOLO models. We used a combined dataset of CinTA_v2 and GathoTF traffic light datasets for training and testing. The dataset was split into three parts: Training Set (76%), Validation Set (19%), and Testing Set (5%).

We trained YOLOv5, YOLOv6, YOLOv7, and YOLOv8 models on the combined dataset, and evaluated their performance on the Testing Set. The results are summarized in Table II. The YOLOv8 model achieved the best performance in terms of precision, recall, and F1 score, with an F1 score of 0.8947, and mAP_0.5 of 0.9192 outperforming other models by a significant margin. The results demonstrate the effectiveness of the proposed approach in detecting and classifying traffic lights accurately and efficiently.

Additionally, we assessed the influence of data augmentation on the performance of the YOLOv8 model. We trained the model with and without augmentation and subsequently compared their respective performances on the Testing Set. The findings of this evaluation are reported in Table II. The augmented dataset significantly improved the performance of the model, with an increase of 0.0313 in F1 score, and 0.0148 in mAP_0.5 indicating that data augmentation is an effective technique for enhancing the robustness and generalizability of the model.
Fig. 10. Precision for traffic light detection and classification using YOLO models.

<table>
<thead>
<tr>
<th>YOLO Model</th>
<th>Precision</th>
<th>Recall</th>
<th>F1_Score</th>
<th>mAP_0.5</th>
</tr>
</thead>
<tbody>
<tr>
<td>YOLOv5</td>
<td>0.7821</td>
<td>0.7787</td>
<td>0.7733</td>
<td>0.8216</td>
</tr>
<tr>
<td>YOLOv6</td>
<td>0.9139</td>
<td>0.8106</td>
<td>0.8631</td>
<td>0.8929</td>
</tr>
<tr>
<td>YOLOv7</td>
<td>0.9201</td>
<td>0.8289</td>
<td>0.8675</td>
<td>0.9028</td>
</tr>
<tr>
<td>YOLOv8</td>
<td>0.9317</td>
<td>0.8427</td>
<td>0.8947</td>
<td>0.9192</td>
</tr>
</tbody>
</table>

To further analyze the performance of the proposed approach, we generated precision curves for each YOLO model, as shown in Fig. 10. The curves indicate that the YOLOv8 model achieved the highest precision values for detecting and classifying traffic lights, followed by YOLOv7, YOLOv6, and YOLOv5 models.

Furthermore, we visualized the resulting images generated by the YOLOv8 model to illustrate the effectiveness of our approach in detecting and classifying traffic lights. Fig. 11 shows sample images from the Testing Set with bounding boxes and labels generated by the YOLOv8 model. The model successfully detected and classified the traffic lights, with high precision and recall values.

Overall, the results demonstrate that the proposed approach using YOLOv8 with data augmentation achieves superior performance in traffic light detection and classification, providing a real-time solution for autonomous driving systems in complex scenarios.

D. Lane-Keeping Results

In addition to traffic light detection, our research also focuses on improving lane-keeping performance for autonomous vehicles. We propose a convolutional neural network (CNN)-based steering angle controller that combines data from a classical PID controller and human perception to predict the steering angle. In order to evaluate the performance of our proposed steering controller, we conducted experiments in a simulated environment using the Gazebo-ROS2 platform.

We collected a dataset of driving behaviors from a human driver using the OpenCV+PID steering angle control method, which we used to train and validate our CNN-based steering controller. The dataset consists of a donkey car model driving in a simulated environment with different road conditions and lighting conditions.

To evaluate the performance of our steering controller, we conducted experiments in the same simulated environment. We compared the performance of our proposed method with a baseline PID controller and a CNN-based steering controller trained with a traditional CNN architecture.

In order to evaluate the performance of our proposed steering controller, we conducted experiments in the same simulated environment and compared our method with a
baseline PID controller and a CNN-based steering controller trained with a traditional CNN architecture. We use a mathematical equation to calculate the percentage of accuracy by summing the prediction error, dividing by the overall validation angle, and multiplying by 100 to measure the accuracy of our proposed method.

\[
\text{accuracy} = 100 - \frac{\sum_{i=1}^{N}(y_{\text{actual}}_i - y_{\text{pred}}_i)}{\sum_{i=1}^{N}(y_{\text{true}}_i)} \times 100 \tag{5}
\]

where \(y_{\text{actual}}\) is the actual angle value, and \(y_{\text{pred}}\) is the predicted angle value.

The evaluation of our lane-keeping performance involved a comparison between the actual steering angle of the vehicle and the predicted steering angle generated by the End-to-End CNN model. This analysis is depicted in Fig. 13.

We also evaluated the performance of our model using the MSE loss function, and the learning curve for the End-to-End CNN model using the MSE loss function is depicted in Fig. 12. After 20 epochs, the MSE value was 230.8, and the learning curve shows a decreasing trend. The curve indicates that the model's performance improves as the number of epochs increases, with diminishing returns after a certain point. Although the curve appears to be approaching a stable solution, further training may be required to confirm this.

Our results demonstrate that our proposed CNN-based steering controller outperforms the baseline PID controller and the traditional CNN-based steering controller in terms of accuracy and smoothness of the steering control. Our proposed method achieved an accuracy of 86.46%, as measured by the percentage of accurately predicted steering angles. The results indicate that our proposed method can effectively predict the steering angle and improve the lane-keeping performance of autonomous vehicles.

V. CONCLUSION

In conclusion, the study presents an optimized approach for traffic light detection and End-to-End steering control for autonomous vehicles using YOLOv8 and a CNN-based steering angle controller. The proposed methods are evaluated in a simulated environment and achieved high performance in both traffic light detection and lane-keeping tasks. The results show that YOLOv8 outperforms other YOLO models in traffic light detection, while the CNN-based steering angle controller achieves a high accuracy rate. The study contributes to the development of advanced autonomous driving systems that can improve driving safety and reduce traffic accidents.

REFERENCES


DefBDet: An Intelligent Default Borrowers Detection Model

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Abstract—The growing popularity and availability of online lending platforms have attracted more borrowers and lenders. There have been several studies focusing on analyzing loan risks in the financial industry, however, defaulting loans still remains an issue that needs more attention. Hence, this research aims to develop an intelligent prediction model that is able to predict risky loans and default borrowers, named the Default Borrowers Detection Model (DefBDet). We seek to help loan lending platforms to approve lending loans to those who are expected to comply with re-payments at the agreed time. Previous works developed a binary classification prediction model (either default or repaid loan). Repaid loans include loans being repaid on or after the loan deadline date. DefBDet, on the other hand, is a novel model, it can predict a loan status based on a multi-classification bases rather than a binary class bases. Hence, it can additionally identify expected late repaid loans, so that special conditions are assigned before loan being approved. This study employs seven different Machine Learning models, using a real-world dataset from 2009-2022 consisting of around 255k loan requests. Statistical measures such as Recall, Precision, and F-measure have been used for models’ evaluation. Results show that Random Forest has achieved the highest performance of 85%.

Keywords—Default borrowers; default loans; loan risks; machine learning models; prediction model

I. INTRODUCTION

Nowadays, data more than ever before is being generated at an extremely fast rate, even more; the volume of data being produced every day is extraordinary [1] [2]. Over the next few years up to 2025 [3], global data creation is expected to grow to more than 180 zettabytes [3]. Furthermore, the global financial data analytics market was valued at $7.6 billion in 2020 and is forecasted to reach $19.8 billion (more than the double value) by 2023 [4].

Default loans data-based assessment is being widely used in financial organizations around the world to assist organizations in either approving or rejecting loan requests [5]. In addition, the growing popularity of loans shows that, in the US [6], more than 20 million persons were owed $178 billion in personal loans as of the first quarter of 2022. That’s more than the double of what was in 2015, only $88 billion were owed from personal loans [6].

One of the reasons for the rising demand for lending is the availability and accessibility of online lending platforms, also the simplicity of completing the loan applications process (with no or limited conditions), which led to a jump in loan requests in 2022 [6]. Lending is risky in that repayments are not always guaranteed, thus, increasing the number of defaulters which is expected to reach 6% in 2023 [7][8]. It has been noticed that the number of willful default cases has increased in 2021 [9], compared with the earlier two years. In addition, loans are considered the dominant asset in the banking sector, they represent nearly 75% of the total amount of the banking assets [10].

Therefore, it is indeed critical to manage the lending activities in a way that controls the borrower's compliance and maintains the financial institution's performance, assets, and liquidity. In contrast, failure to manage loan compliance would likely affect the economy at large [10].

These reasons push toward the need to detect defaulters at early stages (i.e. prior to the loan request being approved), which essentially means identifying the loan requester who will be potentially a defaulting borrower. This would definitely help loan lending organizations to approve loan requests of only committed borrowers and conserve financial institutions' resources. The above-mentioned challenges confirm the importance of monitoring loan requests by developing an intelligent default borrower’s identification model with an early warning system that is capable of alerting financial institutions of potential losses and preventing crises, which in turn is the aim of this paper.

Machine Learning (ML) plays an essential role in the financial sector, especially in the loans domain [11]–[21]. In this work, we develop an intelligent prediction model that is able to recognize the compliance of borrowers against the loan's repayment. Also, it is able to identify expected late repaid loans (loan being fully repaid after loan's maturity date). Thus, we develop a novel model named, Default Borrowers Detection Model (DefBDet). This model is able to predict the loan status based on a multi-classification bases rather than binary class bases, which was the case in previous works [11]– [21].

Hence, the main objective, in this work, is to improve the utilization of classification algorithms in the financial loans field. We seek to assess default loan risk prior to issuing the loan lending decision. ML techniques have been employed in order to develop DefBDet using real loan data. We used the family of supervised ML algorithms, such as Decision Tree (DT), Random Forest (RF), ID3, Deep Learning (DL), Gradient Boosted Decision Trees (GBDT), Support Vector Machines (SVMs), and Naïve Bayes (NB).
The rest of the paper is organized as follows. The next section provides a summary of past related studies. Later, the processes involved in building the model, including data collection and processing, model training and testing, and associated findings and results are presented. The final section concludes with a summary of the work and the suggested plan for future work.

II. LITERATURE REVIEW

ML has been utilized intensively in the financial industry along with its subfields, including but not limited to, the stock market [22]–[24], insurance [25]–[27], and fraud detection [28]–[34].

The stock market field is an attractive topic due to the abundance of data being generated at high and irregular rates. Besides, it plays an important role in the economy; investors are also continually looking to predict future transactions to avoid certain associated risks. The study in [24] has constructed a model to predict stock market future trends. In addition, along with historical stock market prices it considered sentiment analysis using text polarity of financial news. Regarding the insurance field, this study [27] has summarized the role of Data Mining (DM) in the insurance industry and how DM enhances the decision-making using insurance data. For fraud detection, it can be committed in different ways and areas, such as in banking, insurance, government, and healthcare sectors [34]. The paper in [28] provides a comprehensive review of existing research works and literatures on the applications of DM to fraud detection in finance.

A. Predicting Default Loans and Risk Detection

Businesses use predictive analytics to identify potential risks and opportunities for their organizations [35]. Recently, it has been observed that the number of willful defaulters in the financial sector is significantly rising [36]. Therefore, systematic identification to predict and detect willful default behavior is indeed essential. The Corporate Finance Institute [37] has defined the loan as: the sum of money that one or more individuals or companies borrow from banks or other financial institutions, to financially manage planned or unplanned events. In doing so, the borrower must pay back with interest and within a specified period of time [37]. Following we show some research performed in this domain, this including but not limited to [11]–[21].

Due to the increase in using the Internet and submitting loan applications online, it is defiant for financial institutions to evaluate all loans manually. Thus, predicting whether a borrower is going to default is becoming an extremely urgent need and draws much attention from researchers. The author in [11] proposed a supervised default loan prediction method based on deep metric learning, the method extracts the features of a loan itself, models the hidden relationships in loan pairs, and calculates the probability of default. The challenges were the imbalanced defaulted samples compared with total data of loans, hard decision boundaries due to loans binary label, and the heterogeneous loan features that have different data types. The proposed method has a higher accuracy compared to Support Vector Machines, Logistic Regression, Naive Bayes, and Multi-layer Perceptron.

Another work in [12], the aim was to identify a comprehensive list of factors along with building a data model for early prediction of whether the loan will become a Non-Performing Assets (NPA) or not. They explored different classification techniques and considered Neural Network, because of its higher accuracy. The model covers the loan end-to-end processes, starting from loan request where the factors of NPA are early detected, followed by loan monitoring, where the model can identify outliers and possible requests to be defaulted. And the last stage is closing the loan, whether it is fully repaid or declared as NPA.

In [13], the authors presented a study for predicting whether a peer-to-peer (P2P) loan application will be repaid or defaulted by employing different classification models. This work was aiming to find interesting relations among the attributes of loan application by applying association rules. The used dataset by LendingClub was classified as whether a loan will default or not (Yes/No). The most effective classification model was achieved using Random Forest and its accuracy was 71.75%.

Using the LendingClub dataset, a two-phase model is proposed in [14], the first phase predicts loan acceptance or rejection by applying Logistic Regression with recall score of 77.4%. The second phase, on the other hand, predicts loans requests either will be defaulted or fully paid by applying Deep Neural Network with recall score of 72%.

Another study [15] wants to predict loan defaulters using the LendingClub dataset. The author encodes loan status (Current, Fully paid, Issued) as Normal, and encodes (Default, Charged off, In Grace Period, Late) as Default. The performance evaluation was applied and compared using Random Forest algorithm with other three ML methods, namely, Decision Tree, Logistic Regression, and Support Vector Machines, while Random Forest still performs the best.

In another prediction work on LendingClub dataset [16], researchers label the dataset records as follows: any loan that (Defaulted, Charged off, or Late on repayments) was classified as Negative examples, while classified any loan that was (Fully paid or Current) as Positive examples. Naïve Bayes has been used which performs the best with default prediction rate compared with other models.

This study [17] presented clustering for loan risk analysis on big data using the k-mean clustering algorithms. Researchers used different datasets related to loans, including the Bondora\textsuperscript{3} dataset. The clustering of Bondora was divided into two classes: Default risk and Non-default risk.

\textsuperscript{3}A non-performing asset (NPA): is a classification used by financial institutions for loans and advances on which the principal is past due and on which no interest payments have been made for a period of time. – Corporate Finance Institute.

\textsuperscript{2}LendingClub is a peer-to-peer lending company headquartered in San Francisco, California.

\textsuperscript{3}Bondora is one of the leading non-bank digital consumer loan providers in Europe.
The author in [18] developed a DM model for predicting loan default among P2P loans. The work was specifically for small business owners and employed using Boosted Decision Tree model. In this study, the class labels are (Pay in full and Charged off).

Another work utilized Bondora dataset [19] is mainly focused on the prediction of loan default using ML algorithms. The author used Status attribute to predict loan default, which is an existing attribute in the dataset. Status reflects the loan payment status and has polynomial values as follows: Current, Repaid, and Late. The work converted the polynomial values into binary, while the loan records that having Repaid status are treated as Not default and Late status are treated as Default, whereas the records having Current status are excluded as they play no role in the default classification.

In view of this and after reviewing the above literature, the majority of studies [11]–[21] have classified the loans into binary classification (i.e. repaid or not), also, they have issues in determining the loan payment activities and the default stages. In more detail, the Repaid status in previous works does not only consider that the loan was paid on the agreed time (i.e. loan repaid before the loan deadline), but also considers late loan payment (i.e. payment after the loan deadline, such as in the case of default) as Repaid too.

Furthermore, it is worth noting that Repaid status does not reflect exactly whether or not the borrower defaulted prior to the loan being fully repaid. In this case, the default behavior is not detected properly. Alternatively, additional investigation is required in the dataset to determine the actual loan status whether default behavior happened or not before the loan is flagged as Repaid, as it is the key challenge to predicting the defaulters. There is a crystal-clear necessity for multi-classifications, such as Late Repaid classification, while it is not applicable that we include the Late Repaid records in Repaid classification. As a result, having a precise multi-classification will improve the quality of the loan prediction model and can help financial institutions to accurately determine the compliance stage of the borrower, and therefore, take the decision to lend or not. Multi-classification of repayment can differentiate between lenders who are compliant to close their loans on the agreed time and lenders who are delayed repaying the loan.

III. DATA DESCRIPTION
A. Data Collection and Data Description

Bondora is known to be one of the leading non-bank digital consumer loan providers in Europe and has been operating since 2009 [38]. It is licensed as a credit provider under the Estonian Financial Supervision Authority [39].

It is worth noting that Bondora dataset has been extensively used in different published works, including but not limited to [17], [19], [40]–[43].

Bondora’s real-world data is publicly available on the internet4 [44]. It contains raw data related to loan requests from 2009 till today (download day is 22/9/2022) and it is daily updated. The dataset consists of 122 attributes and around 255k records.

Each record belongs to a loan request, which includes data about the borrower and the loan application, such as borrower demographical data, loan duration, purpose of the loan, dates of payment, and also information regarding the current loan's status (Repaid, Late, Current), amount of principal and interest, default date, etc.

Due to the enormity of the dataset, we analyzed it in-depth and determined the most relevant attributes to the study’s purpose. Table I provides a full description of the selected attributes in the dataset.

<table>
<thead>
<tr>
<th>Feature Name</th>
<th>Brief Description</th>
<th>Data Type</th>
</tr>
</thead>
<tbody>
<tr>
<td>LoanId</td>
<td>A unique ID given to all loan applications</td>
<td>Categorical</td>
</tr>
<tr>
<td>PartyId</td>
<td>A unique ID given to the borrower</td>
<td>Categorical</td>
</tr>
<tr>
<td>NewCreditCustomer</td>
<td>Did the customer have prior credit history in Bondora False: Customer had at least 3 months of credit history in Bondora True: No prior credit history in Bondora</td>
<td>Categorical</td>
</tr>
<tr>
<td>LoanDate</td>
<td>Date when the loan was issued</td>
<td>Date</td>
</tr>
<tr>
<td>MaturityDate_Original</td>
<td>Loan maturity date according to the original loan schedule</td>
<td>Date</td>
</tr>
<tr>
<td>MaturityDate_Last</td>
<td>Loan maturity date according to the current payment schedule</td>
<td>Date</td>
</tr>
<tr>
<td>Age</td>
<td>The age of the borrower when signing the loan application</td>
<td>Numeric</td>
</tr>
<tr>
<td>Gender</td>
<td>Male, Female, and Undefined</td>
<td>Categorical</td>
</tr>
<tr>
<td>Amount</td>
<td>Amount the borrower received on the Primary Market. This is the principal balance of your purchase from Secondary Market</td>
<td>Numeric</td>
</tr>
<tr>
<td>LoanDuration</td>
<td>Current loan duration in months</td>
<td>Numeric</td>
</tr>
<tr>
<td>UseOfLoan</td>
<td>Loan consolidation, Real estate, Home improvement, Business, Education, Travel, Vehicle, Health, and Other</td>
<td>Categorical</td>
</tr>
<tr>
<td>Education</td>
<td>Primary education, Basic education, Vocational education, Secondary education, and Higher education</td>
<td>Categorical</td>
</tr>
<tr>
<td>NrOfDependants</td>
<td>Number of children or other dependents</td>
<td>Categorical</td>
</tr>
<tr>
<td>EmploymentStatus</td>
<td>Unemployed, Partially employed, Fully employed, Self-employed, Entrepreneur, and Retiree</td>
<td>Categorical</td>
</tr>
<tr>
<td>OccupationArea</td>
<td>Other, Mining, Processing, Energy, Utilities, Construction, Retail and wholesale, Transport and warehousing, Hospitality and catering, Info and telecom, Finance and insurance, Real-estate, Research, Administrative, Civil service &amp; military, Education, Healthcare and social help, Art and entertainment, Agriculture, and Forestry and fishing</td>
<td>Categorical</td>
</tr>
<tr>
<td>HomeOwnershipType</td>
<td>Homeless, Owner, Living with</td>
<td>Categorical</td>
</tr>
</tbody>
</table>

4 To download the loan dataset (https://www.bondora.com/en/public-reports#secondary-market-archive)
While examining the dataset, we discovered that 53 records had missing values in the majority of the attributes.

Due to the limited number of rows, we decided to exclude them from the dataset. The removed records are approximately less than 0.25% of the total dataset.

Also, some attributes in the dataset were numeric data type, and to facilitate the analysis, we converted these attributes from numeric to categorical data type by following discretization process [45]. Table II shows each attribute with original values before applying discretization and the transformed values after the discretization process.

### TABLE II. BONDORA’S ATTRIBUTES AFTER DISCRETIZATION PROCESS

<table>
<thead>
<tr>
<th>Attribute</th>
<th>Previous Values</th>
<th>New Values</th>
<th>Corresponded Values</th>
</tr>
</thead>
<tbody>
<tr>
<td>Age [46]</td>
<td>Values range from 18 to 77</td>
<td>Young adults</td>
<td>(From age 18 to 28)</td>
</tr>
<tr>
<td></td>
<td></td>
<td>Middle-aged adults</td>
<td>(29 to 39)</td>
</tr>
<tr>
<td></td>
<td></td>
<td>Old-aged adults</td>
<td>(40 to 58)</td>
</tr>
<tr>
<td></td>
<td></td>
<td>Seniors</td>
<td>(&gt;=59)</td>
</tr>
<tr>
<td>IncomeTotal</td>
<td>Values range from 0 to 1,012,019</td>
<td>No income</td>
<td>(Total between 0</td>
</tr>
<tr>
<td></td>
<td></td>
<td>Very low income</td>
<td>to 10)</td>
</tr>
<tr>
<td></td>
<td></td>
<td>Low income</td>
<td>(11 to 1,000)</td>
</tr>
<tr>
<td></td>
<td></td>
<td>Middle income</td>
<td>(1,001 to 2,000)</td>
</tr>
<tr>
<td></td>
<td></td>
<td>High income</td>
<td>(2,001 to 3,000)</td>
</tr>
<tr>
<td></td>
<td></td>
<td>income</td>
<td>(&gt;=3,001)</td>
</tr>
<tr>
<td>Amount</td>
<td>Values range from 6 to 15,948</td>
<td>0 – 1,500</td>
<td></td>
</tr>
<tr>
<td></td>
<td></td>
<td>1,501 – 3,001</td>
<td></td>
</tr>
<tr>
<td></td>
<td></td>
<td>3,001 – 4,501</td>
<td></td>
</tr>
<tr>
<td></td>
<td></td>
<td>4,501 – 6,000</td>
<td></td>
</tr>
<tr>
<td></td>
<td></td>
<td>More than 6,000</td>
<td></td>
</tr>
<tr>
<td>LoanDuration</td>
<td>Values range from 1 to 120</td>
<td>1 – 30</td>
<td></td>
</tr>
<tr>
<td></td>
<td></td>
<td>31 – 60</td>
<td></td>
</tr>
<tr>
<td></td>
<td></td>
<td>61 – 90</td>
<td></td>
</tr>
<tr>
<td></td>
<td></td>
<td>91 – 120</td>
<td></td>
</tr>
</tbody>
</table>

BONDORA’s dataset provides Loan Status, which indicates the loan payment status, it has three possible values, Repaid (meaning that loan has been fully paid back to the investor either on or before the loan maturity date, or after the loan maturity date), Late (the loan has not yet been fully paid to the investor and has exceeded the loan maturity date), and Current (the loan is still in progress, i.e. maturity date has not been reached yet). Almost 30% of the dataset consists of Late loans, and 35% of both Current loans and Repaid loans, Fig. 1 illustrates in a pie chart the ratio of each status of the total dataset.

The original dataset lacks the ability to show the borrowers status regarding paying the loan on time or not. Hence, if the Loan Status is Repaid, it does not clearly indicate whether the loan was repaid on time (on or before the loan maturity date) or not. In other words, it does not show the cooperation of the borrower to repay the loan by the maturity date. Knowing that the loan was repaid on time saves the lending company resources and means that the borrower is trustworthy and dependable.

As a result, and since the dataset has no direct indicator regarding the borrower’s compliance in repaying the given loan on the agreed time, we defined a new polynomial feature named “Borrower’s compliance status”, which precisely represents the situation of the loan in terms of repayment according to the data in each record.

The new feature measures the borrower compliance regarding paying the loan’s last payment before the maturity date of the loan or not.

It consists of four values, Repaid on-time, Late repaid, Defaulted, and in progress.

A loan can be labelled as "Repaid on-time", meaning that the loan was repaid before or on the maturity date of the loan (the dataset consist of 76k record or almost 30% of records were labelled as Repaid on-time), "Late repaid" on the other hand, which means that the loan was repaid after the loan maturity date (15k or 6% of records were labelled as Late repaid). Also, a loan can be labelled as "Defaulted", meaning that the loan is not fully repaid yet, and has passed the maturity date of the loan too (71k or 28% of records were labelled as Defaulted), or labelled as "In progress", which means the current loan is still open and not meet any of the above status (93k or 36% of records were labelled as In progress).

This feature was defined using the following existing attributes in the dataset: MaturityDate_Original, LastPaymentOn, DefaultDate, and Status.
In this study, ML algorithm was determined after reviewing existing works and literatures in this scope [11]–[21], in addition to the study’s purpose which is to predict the borrowers' compliance against the loan re-payment.

Developing a prediction model is indeed valuable for financial institutes and banks as it enables business owners to take actionable decisions at the right time. It helps to avoid borrowers who have common characteristics with other defaulters. Thus, preventing tangible and intangible losses that are expected to occur, or in critical conditions, escalating the case to the court. Therefore, the prediction model will support and enhance the approval or the rejection of loan applications that are expected to default.

In this research, we applied various classification algorithms to build DefBDet which belongs to the family of supervised ML algorithms. These algorithms are DT [48]–[50], ID3 [51], [52], RF [53]–[55], DL, GBDT [18], [56], SVMs [57], [58], and NB [59].

B. Model Evaluation and Assessment

In this study, we mainly focus on several performance evaluation metrics for classifier including F-measure, Precision, Recall. Below are brief definitions of each measure:

- According to [60], F-measure was first introduced by Cornelis Joost van Rijsbergen [61], it combines Recall and Precision with an equal weight [62]. F-measure formula as following (1) [60]:

\[
F1 = \frac{2 \times \text{Recall} \times \text{Precision}}{\text{Recall} + \text{Precision}}
\]

(1)

- Precision or Confidence is the proportion of predicted positive cases that are correctly real positives [62]. Precision formula as following (2) [63]:

\[
\text{Precision} = \frac{\sum \text{TP}}{\sum \text{TP} + \text{FP}}
\]

(2)

Recall or Sensitivity is the proportion of real positive cases that are correctly predicted positive. This measures the Coverage of the real positive cases [62]. Recall formula as following (3) [63]:

\[
\text{Recall} = \frac{\sum \text{TP}}{\sum \text{TP} + \text{FN}}
\]

(3)

Accuracy is a common measure for evaluating a classification model’s ability to discriminate between classes. And because the class labels are imbalanced, the F-measure will capture a balance between Recall and Precision and weights them equally better than the Accuracy in terms of describing the overall model performance [64], [65]. The evaluation metrics and performance measures of each algorithm are detailed in Table III.

V. RESULTS AND DISCUSSION

In this study, we have employed ML classification technique (a supervised learning method) in order to predict the compliance status of loan borrowers. We used seven different classification algorithms and evaluated its performance by F-measure, Precision, and Recall.

The results show that the RF and DT have outperformed all other classification algorithms. In addition, they both have very comparable results (the obtained results for seven algorithms are displayed in Table III). The DT has the highest F-measure score of 85.69%, while RF ranked in the second place with very minor differences of 85.29%.

It is worth noting that the outcomes of all performance measures are considered essential and crucial. However, from business prescriptive, specifically the financial field, we are concerned that the loan is provided to the client who will comply and is willing to repay on time. The financial organizations do not want to lose this type of client (borrowers), i.e. financial organizations are always keen to provide loans to clients who comply with payments on time. Thus, in terms of a performance measure, this is described as precision, which predicts positive cases that are correctly real positives [62]. In other words, the model can predict only the positive borrowers (who will repay on time), than identifying all positive borrowers.

Financial organizations are not willing to provide a loan to someone who has a slight probability of defaulting, and this is due to the importance of managing the financial, human, and digital resources effectively to be sustained in the market, besides the organization’s reputation among competitors.

In DefBDet, the precision based on RF is above 92%, on the other hand, the precision of DT is almost 89%, indicating that the RF model has a strong ability for detecting correct positive cases than DT.

Other reasons why we choose RF algorithm over the other ML algorithms are that RF runs efficiently on enormous datasets and works well with unbalanced class labels. We can conclude obviously that the RF algorithm outperforms other ML approaches according to the mentioned reasons above.
TABLE III. EVALUATION METRICS COMPARISON OF THE SEVEN ALGORITHMS

<table>
<thead>
<tr>
<th>Rank based on F-measure</th>
<th>Algorithm</th>
<th>F-measure</th>
<th>Precision</th>
<th>Recall</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>DT</td>
<td>85.69%</td>
<td>88.93%</td>
<td>82.67%</td>
</tr>
<tr>
<td>2</td>
<td>RF</td>
<td>85.29%</td>
<td>92.14%</td>
<td>79.39%</td>
</tr>
<tr>
<td>3</td>
<td>GBDT</td>
<td>84.19%</td>
<td>88.93%</td>
<td>79.92%</td>
</tr>
<tr>
<td>4</td>
<td>NB</td>
<td>79.52%</td>
<td>77.09%</td>
<td>82.11%</td>
</tr>
<tr>
<td>5</td>
<td>DL</td>
<td>77.95%</td>
<td>73.24%</td>
<td>83.30%</td>
</tr>
<tr>
<td>6</td>
<td>ID3</td>
<td>77.22%</td>
<td>83.54%</td>
<td>71.78%</td>
</tr>
<tr>
<td>7</td>
<td>SVMs</td>
<td>59.92%</td>
<td>54.78%</td>
<td>66.13%</td>
</tr>
</tbody>
</table>

VI. CONCLUSION AND FUTURE WORK

This study has developed an intelligent prediction model identifying default loans in lending communities; its main aim was improving the lending decision-making process. In other words, DefBDet model aims to detect the expected default borrower before the approval of the loan request, which can reduce default loans and at the same time maintain the expected target of returns.

We can clearly infer that DefBDet could reduce the defaulting loans with positive consequences for the efficiency of the financial institutions, by eliminating loan requests that have been detected and are expected to default. Furthermore, it can identify loans that are expected to be late repaid (i.e. loans being repaid after the loan maturity date has passed). Late repaid loans can be an issue to financial institutions if not being closely monitored.

Previous works were able to binary classify loan requests to either fully repaid or defaulted. The case of fully repaid, includes both loans repaid on or before the loan deadline and loans being repaid after the loan deadline date has passed. However, we think expected Late repaid loans needs special attention before loan approval being issued. Thus, DefBDet is a multi-class model; it aims to identify expected late repaid borrowers, so that additional conditions and/ or close monitoring are given. DefBDet can classify a loan to Repaid on-time, Late repaid, and Default.

In addition, the results provided by the model can be generalized to any lending activities or financial institutions. In general, the DT and RF algorithms showed a better performance, the overall performance was higher than 85%, compared with other classification models like SVMs, ID3, GBDT, NB, and DL. However, from a business point of view, correctly identifying defaulting borrowers is very crucial; it can lead to saving financial institutions resources. Financial organizations are not willing to provide a loan to someone who has a probability of defaulting. Thus, high precision value is indeed favorable. In DefBDet, the precision using RF was above 92%, on the other hand, the precision of DT was nearly 89%. As a result, The RF algorithm was adopted in DefBDet.

In the future, we seek to employ DefBDet prediction model on a local dataset to explore the diversities between international and local datasets. In addition, we aim to introduce an additional borrower class (label), which leads to improving DefBDet to be able to predict an additional multiclass label in order to precisely identify the compliance stage of the borrowers before repayment.

ACKNOWLEDGMENT

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Effects of Training Data on Prediction Model for Students' Academic Progress

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Abstract—The ability to predict students’ academic performance before the start of the class with credible accuracy could significantly aid the preparation of effective teaching and learning strategies. Several studies have been conducted to enhance the performance of prediction models by emphasizing three key factors: developing effective prediction algorithms, identifying significant predictor variables, and developing preprocessing techniques. Importantly, none of these studies focused on the effect of using different types of training data on the performance of prediction models. Therefore, this study was conducted to evaluate the effects of differences in training data on the performance of a prediction model designed to monitor students’ academic progress. The findings showed that the performance of the prediction model was strongly influenced by the heterogeneity of the values of the predictor variables, which should accommodate all the existing possibilities. It was also discovered that the application of training data with different characteristics and sizes did not improve the performance of the prediction model when its heterogeneity was not representative.

Keywords—Decision tree; effects of training data; heterogeneity; prediction; students’ academic performance

I. INTRODUCTION

The prediction of students’ academic performance enables higher education institutions to improve the quality of their graduates. This is because it helps to prepare strategies and allocate resources to assist students at risk of academic failure. However, the benefits of these strategies can be minimal when the prediction results are not provided early or when they have poor performance. It is also important to note that the earlier detection of students having the potential to fail academically can lead to immediate implementation of necessary actions by relevant stakeholders [1]. Therefore, several studies have been conducted on early detection of students’ academic performance.

Early detection models were divided into two types which include those developed to screen superior students and those designed to deal with students having the potential to experience academic failure. The first type was commonly used to evaluate prospective students' future academic performance to make informed decisions about their admission [2], [3]. Meanwhile, the models to determine students with the potential to experience academic failure were applied to increase the number of students who graduate on time. Some of those were applied at the beginning of the semester using demographic and academic data of the previous semester [4], [5]. However, the accuracy of these models was found to be lower than the early detection models which were not applied at the beginning of the semester. This was because there were several predictor variables such as test scores that have a significant influence on the model can only be obtained after the course has started [6]. The effort to solve this problem led [6] to develop a two-stage prediction model. The first prediction was made at the beginning of the semester to obtain a list of students with the potential to fail in order to take immediate action while the second was after the midterm exam to monitor their academic progress.

Several studies have been conducted to improve the performance of prediction models with a focus on three aspects which include developing prediction algorithms, obtaining different predictor variables with significant influence on the model, and developing pre-processing techniques. The prediction algorithm was generally developed by integrating specific techniques. For example, an algorithm that integrates Collaborative Filtering and Artificial Immune Systems was developed by [7] to predict student grades based on recommended courses. The results showed that the developed algorithm provided very accurate results. OKC algorithm was also developed by [8] to improve the performance of prediction models using imbalanced datasets without resampling. This algorithm was a hybrid of One-class support vector machine, K-nearest neighbor, and Classification and regression tree algorithms (abbreviated ad OKC). Research [9] performed an ensemble of seven prediction methods with a majority voting technique to improve the accuracy of this prediction model.

 Scholars have been striving to identify the factors influencing the performance of existing prediction models in order to enhance their accuracy. This has led to the application of demographic and academic data as variable predictors to predict students’ academic performance. The academic data can be divided into secondary and higher education data. Moreover, efforts are currently being made to determine other factors having the potential to strongly influence student academic performance in order to improve the prediction model performance. For example, [10] applied students’ motivation, social, and managerial aspects to complement demographic and academic data as predictor variables. The managerial and social aspects were also used by [11] to develop a prediction model. Meanwhile, psychological aspects such as common talents were utilized by [3] to predict the future performance of prospective students. The study also implemented predictor variable weighting to improve the performance of the model. Other studies were also observed to have used psychological aspects in the form of personality to predict academic performance [12]–[15].
Preprocessing has also been applied to improve prediction model performance [11]. It was discovered that [11] added feature selection before the model formation process to eliminate variables with little effect on the performance and to increase the accuracy. A similar attempt was made by [16] through the combination of different feature selection techniques and data transformation to obtain the best prediction model performance. Apart from feature selection, [9], [17], [18] also applied resampling to overcome imbalanced classes to improve model performance.

Several researchers studied the effect of training data to improve the performance of predictive models [19]-[21]. Research conducted by [19] shows that if there is harmonization in the dataset, classifier performance can be improved. So, it is important to choose the appropriate training and testing data for harmonization. The training data order also greatly affected the performance of the classifier [21]. On exactly the same training dataset, classifier performance can vary from 10% to 100%.

These findings showed that several attempts had been made to improve the performance of prediction models but there was a need for more evidence on the effect of using training data with different characteristics on this performance. Therefore, this study was conducted to evaluate the effect of differences in training data on the performance of the prediction model developed to monitor students’ academic progress. This article was divided into four parts which include the introduction section followed by the methodology, analysis of the results, and conclusion of the research.

II. RESEARCH METHODOLOGY

This research was conducted using the simulation method through the steps shown in Fig. 1. Data were collected through the questionnaires distributed to students at a private higher education institution in Surabaya and academic data retrieved from the higher education information system. The data collected were prepared according to the trial scenario to be implemented in the subsequent process. This was followed by the development of the prediction model according to the existing scenario. Finally, the model’s performance was evaluated with a focus on the differences in the training data used in developing the model.

A. Participants and Datasets

The dataset was compiled from 246 private higher education students in Surabaya, Indonesia. The focus was on three different academic years from 2019-2020 to 2021-2022 and three programs including Informatics Engineering, Business Information Systems, and Multimedia. The data obtained were integrated into the student's academic data to form 3,169 instances after the preprocessing stage. Student academic data was collected from 87 subjects grouped in 22 knowledge areas.

<table>
<thead>
<tr>
<th>Predictor Variable</th>
<th>Data Type</th>
<th>Details</th>
</tr>
</thead>
<tbody>
<tr>
<td>Program (X₁)</td>
<td>Nominal</td>
<td>Informatics Engineering, Business Information Systems, and Multimedia</td>
</tr>
<tr>
<td>Sex (X₂)</td>
<td>Nominal</td>
<td>Male, female</td>
</tr>
<tr>
<td>Age (X₃)</td>
<td>Numeric</td>
<td>16, 17, …</td>
</tr>
<tr>
<td>Father’s academic background (X₄)</td>
<td>Ordinal</td>
<td>Not in school, Elementary school, Middle school, High school, Undergraduate, Masters, Doctoral degree</td>
</tr>
<tr>
<td>Mother’s academic background (X₅)</td>
<td>Ordinal</td>
<td>Not in school, Elementary school, Middle school, High school, Undergraduate, Masters, Doctoral degree</td>
</tr>
<tr>
<td>College entrance path (X₆)</td>
<td>Nominal</td>
<td>Regular, Collaboration, non-academic achievement scholarships, academic achievement scholarships</td>
</tr>
<tr>
<td>The specialization chosen while in high school (X₇)</td>
<td>Nominal</td>
<td>Natural Sciences, Social Sciences, Languages, others</td>
</tr>
<tr>
<td>High school city (X₈)</td>
<td>Nominal</td>
<td>Surabaya, Java outside Surabaya, outside Java</td>
</tr>
<tr>
<td>Average Math score at 12th grade (X₉)</td>
<td>Numeric</td>
<td>66.5 - 99</td>
</tr>
<tr>
<td>Average English score at 12th grade (X₁₀)</td>
<td>Numeric</td>
<td>70-100</td>
</tr>
<tr>
<td>The Average score at 12th grade (X₁₁)</td>
<td>Numeric</td>
<td>73.6-97.93</td>
</tr>
<tr>
<td>Cumulative credits (X₁₂)</td>
<td>Numeric</td>
<td>Total credits that have been collected since the first semester</td>
</tr>
<tr>
<td>CGPA (X₁₃)</td>
<td>Numeric</td>
<td>0-4</td>
</tr>
<tr>
<td>Number of courses taken in the semester (X₁₄)</td>
<td>Numeric</td>
<td>3-9</td>
</tr>
<tr>
<td>Number of course participants (X₁₅)</td>
<td>Numeric</td>
<td>4-108</td>
</tr>
<tr>
<td>Previous course grade (X₁₆)</td>
<td>Ordinal</td>
<td>A-E</td>
</tr>
<tr>
<td>Length of time repeat (X₁₇)</td>
<td>Numeric</td>
<td>How many semesters before, last time taking the course</td>
</tr>
<tr>
<td>Position difference (X₁₈)</td>
<td>Numeric</td>
<td>The difference between the course positions taken in the curriculum and the student’s current position</td>
</tr>
<tr>
<td>Pass percentage (X₁₉)</td>
<td>Numeric</td>
<td>The percentage of passing of this course in previous class by the same lecturer</td>
</tr>
<tr>
<td>Prerequisite grade (X₂₀)</td>
<td>Ordinal</td>
<td>A-E</td>
</tr>
<tr>
<td>Prerequisite time taken (X₂₁)</td>
<td>Numeric</td>
<td>How many semesters before, was the last time this prerequisite course was taken</td>
</tr>
</tbody>
</table>

This study used two-stage prediction. The first was applied for the early detection of student academic performance in a course. The predictions were made before the start of lectures using 21 predictor variables as shown in Table I. Meanwhile, the target class was divided into two, Pass or Fail. The results obtained can be used by students, lecturers, and higher...
education stakeholders to devise appropriate strategies to ensure the success of students that have been predicted to fail at the end of the semester. The second stage was applied after lectures had been received for half a semester and the midterm exam scores had been released. This led to the addition of two predictor variables including the midterm exam scores (range: 0 – 100) \(X_{22}\) and the number of students’ absence from lectures during the first half of the semester (range: 0 – 7) \(X_{23}\). The results obtained can be used by lecturers to monitor students’ academic progress.

B. Preprocessing

This stage was used to remove irrelevant data, overcome missing values, transform data to the format required by the algorithm to be used, and prepare data according to the research scenarios. The first time, irrelevant data such as students’ ID, course code, and other attributes not needed as well as those considered to be incomplete and impossible to complete were removed from the dataset. Example of those considered incomplete include are the data related to courses opened for the first time in the program without any record of the pass percentage from lecturers in the previous period.

The data were later transformed by discretizing the continuous data to form categories in order to increase the accuracy of the prediction model [16]. This was followed by the conversion of the categorical data into a numeric form. It was done because the prediction will be carried out using the Decision Tree algorithm with Python programming which requires data to be in a numerical form. Moreover, the application of numeric data had the ability to make some Machine Learning algorithms run efficiently [5].

Finally, the structure of data was prepared to be appropriate for the four scenarios: to evaluate the effect of using training data from certain knowledge areas, different admission years, specific courses, and different training data sizes. The details of each scenario are described in the evaluation section.

C. Model Development and Evaluation

The Decision Tree method was used to develop the prediction models. The method is more popular than the others [1], [22], [23]. Its application was due to the fact that the resulting model tree is usually easy to understand and the conversions are directly in the form of IF-THEN rules.

The dataset consists of 2941 instances for the Pass target class known as major data and 228 instances for the Fail target class classified as minor data. The existence of imbalance in the number of each target class can reduce the performance of the prediction model [8], [24], [25].

The evaluation was conducted to reduce the effect of the target class imbalance using ten-fold stratified cross-validation technique. The data were divided into ten sections with each having a balanced proportion for each target class. It is pertinent to note that one section was used as test data while nine sections were applied as training data. The test process was repeated ten times to avoid deviations using the test data from each section.

The evaluation was conducted using four different scenarios run in different ways as indicated in the following explanations:

1) First scenario: The dataset consists of 22 knowledge areas. The knowledge areas used refer to the Computer Science Curricula 2013 from the ACM – IEEE Computer Society [26]. The dataset was divided into 22 sub-datasets with each containing instances from a particular knowledge area. A total of five knowledge areas with the highest number of instances were used as indicated in the statistics presented in the following Table II. Moreover, a sub-dataset will be compiled which is a combination of those five sub-datasets for comparison.

<table>
<thead>
<tr>
<th>Knowledge Areas</th>
<th>Number of Major Class</th>
<th>Number of Minor Class</th>
</tr>
</thead>
<tbody>
<tr>
<td>Programming Languages (PL)</td>
<td>820</td>
<td>85</td>
</tr>
<tr>
<td>Computational Science (CN)</td>
<td>628</td>
<td>95</td>
</tr>
<tr>
<td>Algorithm and Complexity (AL)</td>
<td>664</td>
<td>58</td>
</tr>
<tr>
<td>Software Development Fundamentals (SDF)</td>
<td>471</td>
<td>23</td>
</tr>
<tr>
<td>Discrete Structures (DS)</td>
<td>371</td>
<td>88</td>
</tr>
</tbody>
</table>

2) Second scenario: The dataset was divided into three sub-datasets with the first containing instances of students that entered higher education in 2019, the second for those in 2020, and the third for those in 2021 as indicated in the following Table III. Moreover, a sub-dataset containing combined instances from the three sub-datasets was compiled for comparison.

<table>
<thead>
<tr>
<th>Admission Year</th>
<th>Number of Major Class</th>
<th>Number of Minor Class</th>
</tr>
</thead>
<tbody>
<tr>
<td>2019</td>
<td>226</td>
<td>32</td>
</tr>
<tr>
<td>2020</td>
<td>2073</td>
<td>150</td>
</tr>
<tr>
<td>2021</td>
<td>642</td>
<td>46</td>
</tr>
</tbody>
</table>

3) Third scenario: The dataset was divided into several sub-datasets with each containing instances from specific courses. The five courses with the highest number of instances and the most significant number of minor data were used as indicated in Table IV. Moreover, a sub-dataset containing instances from different courses was compiled for comparison.

<table>
<thead>
<tr>
<th>Courses</th>
<th>Number of Major Class</th>
<th>Number of Minor Class</th>
</tr>
</thead>
<tbody>
<tr>
<td>Algorithm and Programming (AP)</td>
<td>194</td>
<td>39</td>
</tr>
<tr>
<td>Statistics (S)</td>
<td>143</td>
<td>64</td>
</tr>
<tr>
<td>Discrete Mathematics (DM)</td>
<td>180</td>
<td>18</td>
</tr>
<tr>
<td>Web Programming (WP)</td>
<td>120</td>
<td>23</td>
</tr>
<tr>
<td>Computer Network (CN)</td>
<td>126</td>
<td>9</td>
</tr>
</tbody>
</table>
4) Fourth scenario: The dataset was divided into eight sub-datasets with 50, 100, 500, 1000, 1500, 2000, 2500, 3000, and 3169 instances respectively. These instances were selected randomly from the dataset using a simple random sampling technique.

The trial conditions from the first to the third scenarios were balanced by ensuring the number of instances in the comparison sub-dataset was equal to the number of training data in the other sub-datasets. The instances in the comparison sub-dataset were selected randomly from the dataset using a stratified random sampling technique while the test data used were the same as those applied for each sub-dataset.

Accuracy was used as the performance measure and it was determined as the ratio of the correctly predicted number of instances to the total number of instances, as indicated in (1).

\[
\text{Accuracy} = \frac{TP+TN}{TP+TN+FP+FN} \tag{1}
\]

True Positive (TP) is the number of instances predicted to fail from instances with the failed target class. True Negative (TN) is the number of instances predicted to pass from instances with the target class passed. False Positive (FP) is the number of instances predicted to fail from instances with the target class passed. False Negative (FN) is the number of instances predicted to pass from instances with the failed target class.

The classes of each predictor and target variable were not distributed equally and this means applying only the accuracy measure can cause confusion [5], [16]. Therefore, F-measure was introduced and it involved calculating the harmonic average between recall and precision values, as indicated in (2), (3), and (4).

\[
\text{Recall} = \frac{TP}{TP+FN} \tag{2}
\]

\[
\text{Precision} = \frac{TP}{TP+FP} \tag{3}
\]

\[
F\text{-measure} = \frac{2\times\text{Recall} \times \text{Precision}}{\text{Recall} + \text{Precision}} \tag{4}
\]

III. RESULTS AND DISCUSSION

A total of five analyses were conducted in the first scenario as indicated in Tables V and VI. It was discovered that there was an increase in the prediction model's performance from the first to the second stage. This showed that the addition of appropriate predictor variables increased the accuracy as well as the performance of the model for unbalanced data distribution without increasing the amount of minor data. Furthermore, a higher increase in the F-measure value compared to the accuracy value indicated a rapid increment in the predictive ability of minor data compared to the majority data.

The prediction model performance of most comparison sub-datasets was found to be better than for specific knowledge areas but the variation was relatively small. This showed that the academic performance of students in certain knowledge area was more clustered. As a result, heterogeneity in the training data in each sub-dataset needed to be distributed appropriately. The combination of different kinds of knowledge areas in the comparison sub-dataset complemented its heterogeneity, thereby, increasing its ability to accommodate heterogeneity in the test data.

Tables VII and VIII show the predicted results for the four models formed according to the second scenario. The pattern of the prediction model's performance from the first to the second stage was found to be the same as the first scenario. It was also discovered that there was a significant increase in the F-measure but not as much as in the first scenario.

The test results for the sub-dataset with a particular admission year showed better performance than the comparison sub-dataset. This means that the heterogeneity in the training data of each sub-dataset was well distributed to accommodate the one in the test data. Meanwhile, the heterogeneity of the datasets for each admission year was found to be different and their combination to form a prediction model reduced the performance compared to the application of datasets from a particular admission year. Meanwhile, the difference was minimal. Based on these results, it can be concluded that student academic performance varies for each year of admission. As a result, the prediction of student performance with a certain admission year will be better if the training data was taken from the same admission year.

<p>| TABLE V. FIRST SCENARIO TEST RESULTS (ACCURACY) |</p>
<table>
<thead>
<tr>
<th>Knowledge Areas</th>
<th>Accuracy (%)</th>
<th>Comparison Models</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>1st Stage</td>
<td>2nd Stage</td>
</tr>
<tr>
<td>PL</td>
<td>87.85</td>
<td>92.26</td>
</tr>
<tr>
<td>CN</td>
<td>87.68</td>
<td>87.84</td>
</tr>
<tr>
<td>AL</td>
<td>87.95</td>
<td>93.21</td>
</tr>
<tr>
<td>SDF</td>
<td>92.53</td>
<td>93.34</td>
</tr>
<tr>
<td>DS</td>
<td>81.27</td>
<td>84.30</td>
</tr>
</tbody>
</table>

<p>| TABLE VI. FIRST SCENARIO TEST RESULTS (F-MEASURE) |</p>
<table>
<thead>
<tr>
<th>Knowledge Areas</th>
<th>F-measure (%)</th>
<th>Comparison Models</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>1st Stage</td>
<td>2nd Stage</td>
</tr>
<tr>
<td>PL</td>
<td>15.25</td>
<td>56.34</td>
</tr>
<tr>
<td>CN</td>
<td>46.44</td>
<td>47.74</td>
</tr>
<tr>
<td>AL</td>
<td>16.76</td>
<td>54.82</td>
</tr>
<tr>
<td>SDF</td>
<td>10.67</td>
<td>37.57</td>
</tr>
<tr>
<td>DS</td>
<td>42.55</td>
<td>58.60</td>
</tr>
</tbody>
</table>

<p>| TABLE VII. SECOND SCENARIO TEST RESULTS (ACCURACY) |</p>
<table>
<thead>
<tr>
<th>Admission Year</th>
<th>Accuracy (%)</th>
<th>Comparison Models</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>1st Stage</td>
<td>2nd Stage</td>
</tr>
<tr>
<td>2019</td>
<td>84.45</td>
<td>84.85</td>
</tr>
<tr>
<td>2020</td>
<td>92.58</td>
<td>93.66</td>
</tr>
<tr>
<td>2021</td>
<td>92.88</td>
<td>94.19</td>
</tr>
</tbody>
</table>

<p>| TABLE VIII. SECOND SCENARIO TEST RESULTS (F-MEASURE) |</p>
<table>
<thead>
<tr>
<th>Admission Year</th>
<th>F-measure (%)</th>
<th>Comparison Models</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>1st Stage</td>
<td>2nd Stage</td>
</tr>
<tr>
<td>2019</td>
<td>30.71</td>
<td>42.45</td>
</tr>
<tr>
<td>2020</td>
<td>34.03</td>
<td>47.83</td>
</tr>
<tr>
<td>2021</td>
<td>37.45</td>
<td>57.43</td>
</tr>
</tbody>
</table>
TABLE IX. THIRD SCENARIO TEST RESULTS (ACCURACY)

<table>
<thead>
<tr>
<th>Courses</th>
<th>1st Stage</th>
<th>2nd Stage</th>
<th>1st Stage</th>
<th>2nd Stage</th>
</tr>
</thead>
<tbody>
<tr>
<td>AP</td>
<td>74.69</td>
<td>87.10</td>
<td>76.41</td>
<td>85.36</td>
</tr>
<tr>
<td>S</td>
<td>66.26</td>
<td>76.45</td>
<td>70.60</td>
<td>72.55</td>
</tr>
<tr>
<td>DM</td>
<td>85.32</td>
<td>90.97</td>
<td>83.87</td>
<td>86.39</td>
</tr>
<tr>
<td>WP</td>
<td>84.71</td>
<td>86.00</td>
<td>76.95</td>
<td>88.76</td>
</tr>
<tr>
<td>CN</td>
<td>88.90</td>
<td>94.12</td>
<td>91.15</td>
<td>95.60</td>
</tr>
</tbody>
</table>

TABLE X. THIRD SCENARIO TEST RESULTS (F-MEASURE)

<table>
<thead>
<tr>
<th>Courses</th>
<th>1st Stage</th>
<th>2nd Stage</th>
<th>1st Stage</th>
<th>2nd Stage</th>
</tr>
</thead>
<tbody>
<tr>
<td>AP</td>
<td>19.18</td>
<td>63.92</td>
<td>21.21</td>
<td>43.07</td>
</tr>
<tr>
<td>S</td>
<td>42.42</td>
<td>60.70</td>
<td>46.83</td>
<td>51.74</td>
</tr>
<tr>
<td>DM</td>
<td>19.00</td>
<td>49.33</td>
<td>19.00</td>
<td>41.38</td>
</tr>
<tr>
<td>WP</td>
<td>57.57</td>
<td>53.00</td>
<td>43.65</td>
<td>63.00</td>
</tr>
<tr>
<td>CN</td>
<td>6.67</td>
<td>25.00</td>
<td>24.00</td>
<td>40.00</td>
</tr>
</tbody>
</table>

The test data results for the datasets from specific courses presented in Tables IX and X were observed to be similar to those of the previous evaluation such that the second stage was found to perform better than the first stage, except for the F-measure of the Web course Programming. However, the decrease in the F-measure value of the Web Programming course, 8%, was much lower than the average increase, 178%, for the other course sub-datasets and 68% in the comparison sub-dataset. The average increase in the accuracy value of the five sub-datasets was also estimated at 8%-9% and this was much lower than the F-measure value.

The test results for each particular course sub-dataset did not show any particular pattern when compared with the comparison sub-dataset. In some courses, the model's performance was better than the comparison sub-dataset and this means the heterogeneity in the training data of each sub-dataset was not properly distributed, thereby, indicating the inability to accommodate the heterogeneity in the test data. Moreover, the combination of different kinds of courses in the comparison sub-dataset complemented the heterogeneity in the comparison sub-dataset and this allowed the accommodation of heterogeneity in the test data.

In the fourth test scenario, nine types of analysis were performed using different dataset sizes as indicated in Fig. 2 and Fig. 3. It was discovered that the performance of the prediction model in the second stage was better than the first stage, except for the accuracy of the dataset with 2000 instances. This was observed to have a similar pattern as the first scenario trial.

There was no pattern showing that an increment in the dataset's size led to an increase in the accuracy value. Even though, the most extensive dataset size provided the highest accuracy and F-measure in this trial. This was observed to be in line with the research conducted by [5]. It also showed that a small dataset could provide a credible accuracy rate as long as it had the ability to identify key indicators.

The tree model generated from datasets of different sizes is presented in Fig. 4, starting from the smallest in Fig. 4(a) to the largest in Fig. 4(i). It was observed from each of the trees produced that the predictor variable played a crucial role in the prediction model and the size of the trees formed was different. This means the model can perform effectively when the training data represent the heterogeneity in the test data. It was also discovered that the training data with a large size but lacks the ability to accommodate heterogeneity in the test data could produce poor performance. An increment in the size of the training data was expected to increase the heterogeneity in order to accommodate the test data but this could not always be achieved. Therefore, the problem can be anticipated through other aspects such as the selection of a suitable predictor variable.

**IV. CONCLUSION**

This research was conducted to determine the effect of using different training data on the prediction model of students’ academic performance in some courses. The dataset

![Fig. 2. The accuracy of the results for the 4th scenario.](image2)

![Fig. 3. The F-measure of the results for the 4th scenario.](image3)

![Fig. 4. Trees generated from the 4th scenario.](image4)
used in this research was limited to a private higher education institution in East Java, Indonesia which was taken during the Covid 19 pandemic. The prediction model was developed in two stages to be used as a tool to monitor students’ academic progress. The trials were conducted based on four different scenarios, including the effect of using sub-datasets from certain knowledge area, admission years, courses, and different dataset sizes. The results showed that the performance of the prediction model in the second stage was mostly better than the first, even though the average accuracy of the first was higher than 80%. These findings showed that the addition of appropriate predictor variables can improve model performance thereby increasing confidence in the results provided by the model to monitor academic progress.

The results showed that the performance of the model was greatly influenced by the heterogeneity of both predictor and target variables, and this was necessary to accommodate all possible outcomes. Therefore, the use of datasets with specific characteristics or sizes can only improve the prediction model's performance when the heterogeneity of the dataset is representative of the larger population. This means, there is a need to ensure the data's heterogeneity is considered to achieve satisfactory performance measures. Research that will be considered further is to develop a synthetic data oversampling strategy to increase the heterogeneity of the dataset so that the performance of the predictive model can be improved.

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REFERENCES


A Novel Framework for Detecting Network Intrusions Based on Machine Learning Methods

Batyrbek Omarov, Nazgul Abdinurova, Zhamshidbek Abdulkhamidov
Suleyman Demirel University, Kaskelen, Kazakhstan

Abstract—In the rapidly evolving landscape of cyber threats, the efficacy of traditional rule-based network intrusion detection systems has become increasingly questionable. This paper introduces a novel framework for identifying network intrusions, leveraging the power of advanced machine learning techniques. The proposed methodology steps away from the rigidity of conventional systems, bringing a flexible, adaptive, and intuitive approach to the forefront of network security. This study employs a diverse blend of machine learning models including but not limited to, Convolutional Neural Networks (CNNs), Support Vector Machines (SVMs), and Random Forests. This research explores an innovative feature extraction and selection technique that enables the model to focus on high-priority potential threats, minimizing noise and improving detection accuracy. The framework's performance has been rigorously evaluated through a series of experiments on benchmark datasets. The results consistently surpass traditional methods, demonstrating a remarkable increase in detection rates and a significant reduction in false positives. Further, the machine learning-based model demonstrated its ability to adapt to new threat landscapes, indicating its suitability in real-world scenarios. By marrying the agility of machine learning with the concreteness of network intrusion detection, this research opens up new avenues for dynamic and resilient cybersecurity. The framework offers an innovative solution that can identify, learn, and adapt to evolving network intrusions, shaping the future of cyber defense strategies.

Keywords—Attack detection; intrusion detection; machine learning; information security; artificial intelligence

I. INTRODUCTION

The omnipresence of network systems and the growing dependence of various industries on these platforms have amplified the necessity for robust cybersecurity measures. A significant component of such measures is the effective detection of network intrusions [1]. Traditionally, the detection of such intrusions has been performed by rule-based systems, which, while effective in certain circumstances, have been found lacking in the face of more complex and evolving cyber threats [2].

In the field of cybersecurity, intrusion detection systems (IDS) play a pivotal role. IDS work as a security guard, continuously monitoring network traffic and promptly identifying possible threats [3]. The most commonly employed IDS are signature-based and anomaly-based. Signature-based IDS detect known threats by matching them against an existing database of threat signatures. Conversely, anomaly-based IDS identify deviations from the 'normal' network behavior as potential threats. While these traditional methods provide a certain degree of security, their shortcomings are becoming more apparent in the contemporary landscape of cyber threats [4].

Signature-based IDS are inherently limited by their dependence on the existing database of threats. As they can only detect previously encountered threats, their effectiveness dwindles when faced with novel, unknown threats [5]. On the other hand, anomaly-based IDS, while theoretically capable of detecting new threats, often suffer from high false-positive rates due to the challenge of defining what constitutes 'normal' behavior.

In recent years, the rise of machine learning has provided a promising avenue for overcoming these limitations [6-8]. Machine learning, with its ability to learn from data and make decisions, has shown significant potential in numerous fields, including cybersecurity [9]. Particularly, machine learning methods can address the limitations of traditional IDS by learning from past data, improving over time, and adaptively identifying new threats.

This paper introduces a novel framework for detecting network intrusions, harnessing the power of machine learning. This framework moves beyond the rule-based systems, offering a more flexible, adaptive, and intuitive approach to network security.

The main impetus for this research comes from the increasing complexity of cyber threats and the consequent need for more advanced detection techniques. The landscape of network intrusions has seen an evolution from relatively straightforward threats to sophisticated attacks that can bypass conventional security measures.

Therefore, the primary objective of this research is to develop a machine learning-based framework for network intrusion detection that can effectively identify and respond to both known and unknown threats. We aim to leverage the predictive and adaptive capabilities of machine learning to achieve higher detection accuracy and lower false-positive rates than traditional IDS.

This research also seeks to address the issue of scalability in network intrusion detection. As network systems grow in size and complexity, the amount of network data that needs to be monitored also increases, posing a significant challenge to conventional IDS. Our machine learning-based framework is designed to handle this increased scale and complexity effectively.
The world of cybersecurity is at a juncture where traditional techniques of network intrusion detection are proving insufficient against the complex and evolving landscape of cyber threats. To address this, our research explores the integration of machine learning techniques into a novel framework for detecting network intrusions. This research aims not only to enhance the efficacy of intrusion detection but also to pave the way for future advancements in cybersecurity.

II. LITERATURE REVIEW

The literature review presented herein gives a comprehensive overview of various machine learning methods applied in the realm of network intrusion detection systems (IDS). These methodologies encompass both the traditional algorithms and the emerging paradigms.

A. Traditional Machine Learning Methods in Intrusion Detection

Decision Trees (DT) have been widely applied in IDS due to their interpretability and efficiency in handling large-scale datasets. DT-based models, such as the C4.5 algorithm, have shown impressive results in terms of detection accuracy and speed [10]. However, they tend to overfit the training data, which leads to poor generalization in the face of new threats.

K-Nearest Neighbors (KNN) is another popular algorithm in the IDS domain because of its simplicity and effectiveness [11]. Its major advantage lies in its ability to detect local patterns, making it powerful for identifying unusual behavior. However, its performance deteriorates with high-dimensional data, which is common in network intrusion detection.

Naive Bayes (NB) classifiers, based on Bayes' theorem, have been used due to their capability to handle a vast number of features effectively. Nevertheless, the assumption of feature independence in NB often leads to suboptimal performance since features in network traffic data are usually interrelated [12].

Support Vector Machines (SVM) are another choice, often praised for their high accuracy and robustness against overfitting. Despite these benefits, SVMs have two critical limitations: computational complexity with large datasets and sensitivity to parameter selection [13].

Logistic Regression (LR) is a statistical technique often applied to binary classification problems, including IDS [14]. LR models are easily interpretable and handle noisy data well. However, they often perform poorly when there are non-linear relationships in the data.

Random Forest (RF) is a widely applied ensemble learning method in IDS, combining multiple decision trees to reduce overfitting and improve prediction accuracy [15]. RF can handle high-dimensional and large-scale data efficiently, but it may produce biased predictions if the features have different scales.

Adaptive Boosting (AdaBoost) is another ensemble technique that combines weak classifiers to form a strong classifier. AdaBoost has been used in IDS to improve the performance of base classifiers like DT and NB [16]. However, it is sensitive to noisy data and outliers.

Artificial Neural Networks (ANN) can model complex non-linear relationships, which are common in IDS tasks. ANN, such as Multilayer Perceptron (MLP), have been used to build IDS due to their ability to learn and generalize from the input data [17]. However, they can be computationally expensive and are often considered as ‘black-box’ models due to their lack of interpretability.

B. Related Works

The domain of network intrusion detection has witnessed a proliferation of research efforts in recent years. These studies have ventured into different machine learning algorithms, feature selection techniques, and evaluation metrics to enhance the IDS's performance.

Several researchers have explored the potential of traditional machine learning models in the IDS domain. For instance, a study [18] explored the use of the Decision Tree (DT) model for intrusion detection. Their work highlighted the utility of DT in classifying network intrusions, yet underlined its tendency to overfit when handling new, unseen data. Another investigation [19] deployed K-Nearest Neighbors (KNN) to detect abnormal network traffic. Despite the promising results, the study acknowledged that KNN's performance deteriorated with high-dimensional datasets.

Naive Bayes (NB) and Support Vector Machines (SVM) have also been adopted in this field. Next study [20] examined the application of NB and found it capable of handling numerous features effectively, albeit with suboptimal performance due to the assumption of feature independence. On the other hand, the work by [21] presented SVM as a powerful tool in detecting network intrusions, with its drawback being its computational complexity and parameter sensitivity.

In the realm of ensemble learning, Random Forest (RF) and Adaptive Boosting (AdaBoost) have been at the forefront of numerous investigations. The study [22] employed RF for network intrusion detection and demonstrated its effectiveness in handling large-scale data. However, it also noted a possible bias in predictions if the features had different scales. Similarly, the research by [23] used AdaBoost to improve IDS performance. Despite achieving promising results, their work indicated the model's sensitivity to noisy data and outliers.

Artificial Neural Networks (ANN) have been the focus of several studies due to their ability to model complex relationships. Next work [24] deployed ANN in the form of a Multilayer Perceptron (MLP) for intrusion detection. The research indicated that despite ANN's high detection rates, its 'black-box' nature posed a significant challenge for interpretation.

While the cited works have significantly contributed to the IDS field, they mostly concentrate on a single machine learning method. This narrow focus tends to overlook the intricate nature of network intrusion detection. Furthermore, the changing landscape of cyber threats necessitates a system that can continuously adapt and learn, an aspect that is often neglected in these studies.
In contrast to the aforementioned research, our work presents a novel framework that harnesses a diverse blend of machine learning methods. By doing so, it captures the complexity of network intrusion detection and leverages the strengths of each method, thereby overcoming the individual weaknesses. Moreover, our framework is designed to evolve with the changing landscape of cyber threats, making it a more robust and sustainable solution.

C. Discussion

The above literature reveals that each of these machine learning methods has its strengths and weaknesses. Traditional methods such as DT, KNN, NB, SVM, and LR are generally simple and interpretable but may struggle with the complexity and high dimensionality of IDS tasks. On the other hand, ensemble methods like RF and AdaBoost offer improved accuracy and robustness but may suffer from bias or sensitivity to noisy data.

ANN provides a powerful tool for modeling complex relationships, yet their interpretability and computational efficiency are often questioned [25]. These considerations underscore the necessity for a robust machine learning-based framework that can effectively utilize these methodologies' strengths while mitigating their shortcomings.

Although substantial research has been conducted in the field of machine learning-based IDS, many gaps still remain. First, most studies focus on using a single machine learning method, which may not fully capture the complexities of network intrusion detection. Second, many existing models fail to consider the evolving nature of cyber threats, which require IDS to continuously learn and adapt [26].

This research aims to fill these gaps by proposing a novel framework that utilizes a variety of machine learning techniques to improve detection accuracy, reduce false positives, and adapt to the changing landscape of network intrusions.

III. PROBLEM STATEMENT

As a flowchart of the proposed system, we take Fig. 1 [26]. Fig. 1, the suggested framework incorporates four sequential stages. These stages comprise: (1) Data Cleaning, where irrelevant and erroneous data entries are removed or corrected; (2) Data Transformation, which involves normalizing and restructuring the cleansed data; (3) Feature Engineering, where significant attributes are extracted and selected; and finally (4) Classification using Machine Learning, which involves training models on the refined data to detect network intrusions.

A. Data Cleaning

The first stage of the proposed system is Data Cleaning, a critical process aimed at purging the data of inconsistencies, inaccuracies, and redundancies. This involves the identification and rectification of various data issues such as missing or incomplete data, duplicate entries, inconsistent data formats, or erroneous data entries. The process also includes outlier detection and treatment, as these anomalous data points can have a considerable impact on the subsequent analysis if left untreated [27].

The principal motivation behind this stage is to improve the quality of the data and thereby enhance the performance and reliability of the ensuing steps. The integrity and quality of the data are key determinants of the efficacy of any data-driven system, hence this phase is paramount. It sets a strong foundation for the remaining processes, ensuring they are not skewed by noise or inaccuracies in the data. The output from this phase is a cleaner, more reliable dataset, which provides a more accurate and dependable basis for the following stages of the system.

B. Data Transformation

Upon completing the data cleaning process, the system transitions to the Data Transformation phase. In this stage, the cleaned data is converted into a format that facilitates subsequent stages of the system. This typically involves two main processes: normalization and restructuring [28].

Normalization is a procedure that standardizes the scales of numerical features in the dataset, ensuring that all features contribute equally to the analysis irrespective of their original scales. This is crucial for machine learning methods that are sensitive to the scale of the input features.

![Fig. 1. Architecture of the proposed system for network intrusion detection.](www.ijacsa.thesai.org)
Restructuring, meanwhile, refers to reformatting the dataset into a structure that's more conducive to the feature engineering and machine learning processes. This may involve actions like encoding categorical variables into numerical formats, or transforming complex data structures into a simpler form that can be processed more easily by the system.

C. Feature Engineering

The Feature Engineering phase forms the crux of the system. During this stage, the transformed data is analyzed to identify the most significant features that should be input into the machine learning model. This involves a combination of domain knowledge, exploratory data analysis, and statistical techniques.

Feature extraction is employed when the original feature set is transformed into a set of derived features, which are expected to better represent the underlying problem. These new features may be simpler, may capture more complex relationships, or may simply be more relevant for the problem at hand.

Feature selection, on the other hand, involves selecting the most relevant features from the original or extracted feature set. This is typically performed through methods such as correlation analysis, mutual information, or wrapper methods. By removing irrelevant or redundant features, this process reduces the dimensionality of the problem, thereby improving the computational efficiency and potentially enhancing the performance of the machine learning model.

D. Classification using Machine Learning

The final phase in the system, Classification using Machine Learning, leverages the outputs of the preceding stages to learn patterns in the data and classify network activities as normal or intrusive. This stage employs a chosen machine learning algorithm, which is trained using the processed data from the earlier stages.

Training the model involves inputting the feature vectors into the model and allowing it to learn the relationships between the features and the target variable. Once the model has been trained, it can then be tested using unseen data to evaluate its performance.

The ultimate aim of this stage is to generate a predictive model that can accurately and effectively detect network intrusions. This model forms the core of the proposed system, and its efficacy directly determines the success of the system as a whole. The choice of machine learning algorithm, the tuning of its parameters, and the evaluation of its performance all form critical parts of this phase.

IV. DATASET

The NSL-KDD dataset, a benchmark dataset commonly used in the realm of network intrusion detection research, is a refined version of its predecessor, the widely recognized KDD'99 dataset [29]. The KDD'99 dataset was created from the 1998 DARPA Intrusion Detection Evaluation Program conducted by MIT Lincoln Labs, providing a realistic representation of network traffic data with a variety of simulated attacks.

A. Description of the Dataset

The KDD'99 dataset, while serving as an indispensable resource for researchers, was noted for having a number of significant issues. These included the presence of a large number of duplicate records, creating an artificial bias in the system, and an unrealistic distribution of the different classes of network intrusions. To address these limitations and provide a more accurate testing ground for researchers, the NSL-KDD dataset was proposed as an improved version.

The NSL-KDD dataset eliminates the redundancies present in the original dataset by removing duplicate entries, thereby creating a more balanced and realistic representation of network traffic. It comprises of approximately 125,973 records for training (KDDTrain+) and 22,544 records for testing (KDDTest+), including a variety of intrusion types. Fig. 2 provides a graphical representation of the NSL-KDD dataset, delineating its class distribution. The dataset is composed of approximately 53% instances labeled as 'normal', representing legitimate network behavior, while the remaining 47% signify various types of 'attack' classes. Furthermore, Fig. 2 presents a detailed breakdown of the NSL-KDD dataset by protocols. In this distribution, TCP protocol related data constitutes a significant majority at 82%, followed by UDP protocol related data at 12%, with ICMP protocol data accounting for the remaining 7%.

Each category represents different types of network attacks. For instance, DoS attacks aim to make a machine or network resource unavailable, U2R attacks exploit vulnerabilities to gain unauthorized root access, R2L attacks exploit vulnerabilities to gain local access, and Probe attacks scan a network to gather information or find known vulnerabilities.

Fig. 3 provides an illustrative overview of the distribution of 'flags' across the 'normal' and 'attack' classes in the NSL-KDD dataset. In network communication, flags are employed to indicate the status of a certain connection, or to signal various types of events or errors. These flags can serve as powerful indicators of anomalous or malicious behavior in network traffic, hence their distribution across the different classes is of significant interest.

This figure presents a comparative analysis, providing a visual breakdown of how different flag values are distributed between 'normal' and 'attack' instances. By presenting the data in this manner, it allows for a more nuanced understanding of the relationship between flag values and the class of the connection. This, in turn, can provide important insights into how different flag values might be associated with different types of network traffic, and how they can be utilized in the detection of network intrusions.
It's important to consider that the effectiveness of using flags as indicators of malicious activity can depend on several factors, including the nature of the network environment and the specific types of attacks that are prevalent. As such, the distribution presented in this figure serves as a starting point for deeper analysis and discussion on the role of flags in network intrusion detection.

The target variable, meanwhile, is a binary class label indicating whether the connection was normal or an attack, along with a detailed label specifying the type of attack if it was an intrusion. The variety and depth of features, coupled with the extensive range of attack types, make the NSL-KDD dataset an excellent resource for developing and testing network intrusion detection systems.

The NSL-KDD dataset provides a rigorous and realistic testing ground for machine learning algorithms, enabling a detailed evaluation of their performance in intrusion detection tasks. Despite the advancements in this field, the NSL-KDD dataset remains a pertinent choice for researchers, continuing to provide valuable insights into the efficacy of various methods and systems. As such, it forms an ideal resource for our study, offering a robust and diverse dataset to evaluate the effectiveness of our proposed system.

B. Data Preprocessing

As presented in Fig. 4, we have utilized the boxplot method to examine the distribution and presence of outliers across all columns in the NSL-KDD dataset. By graphically representing these parameters, we can easily identify interquartile ranges, detect potential outliers, and understand the overall data distribution. In this instance, Fig. 3 serves as a valuable visual aid to understand the statistical nuances of our chosen dataset, enabling us to discern any aberrant data points effectively.
Fig. 4. Box plots for each feature.
V. EVALUATION PARAMETERS

In the realm of predictive modeling, accuracy is a fundamental metric utilized to quantify the performance of a model. It represents the proportion of correct predictions made by the model in relation to the total number of predictions. Higher accuracy values indicate a superior ability of the model to correctly classify or predict outcomes. Equation (1) explains accuracy considering true positives (TP), true negatives (TN), false negatives (FN), and false positives (FP) [30].

\[
\text{accuracy} = \frac{TP + TN}{TP + FN + TN + FP} \tag{1}
\]

Precision, an important measure in classification tasks, assesses the model's capacity to accurately predict positive instances [31]. It is defined as the ratio of true positives to the sum of true positives and false positives. Higher precision signifies that the model's positive predictions are largely accurate, minimizing false positive occurrences.

\[
\text{precision} = \frac{TP}{TP + FP} \tag{2}
\]

Recall, often referred to as sensitivity or true positive rate, is a critical metric in the domain of classification problems [32]. It gauges the model's effectiveness in identifying all relevant instances, calculated as the ratio of true positives to the sum of true positives and false negatives. A higher recall implies fewer instances of false negatives, ensuring that the model captures most positive observations.

\[
\text{recall} = \frac{TP}{TP + FN} \tag{3}
\]

The F-score, also known as the F1 score, is a composite metric that harmonizes precision and recall in a single measure [33]. It is the harmonic mean of precision and recall, which equally weights both measures. A high F-score implies that both the precision and recall of the model are high, thus representing an optimal balance between false positives and false negatives.

\[
F_1 = \frac{2 \cdot \text{precision} \cdot \text{recall}}{\text{precision} + \text{recall}} \tag{4}
\]

The aforementioned evaluation metrics - accuracy, precision, recall, and F-score - are paramount in gauging the performance of a classification model in a comprehensive manner. Accuracy quantifies the proportion of correct predictions made by the model, while precision evaluates the model's ability to correctly identify positive instances. Recall assesses the capacity of the model to detect all relevant instances, and the F-score serves as a combined measure that balances both precision and recall. These measures together provide a holistic assessment of a model's predictive capabilities, and each has unique importance depending on the specific objective of the classification task at hand.

VI. EXPERIMENTAL RESULTS

The following section, "Experiment Results," details the outcomes of our investigative study, aimed at evaluating the effectiveness of the proposed novel framework for detecting network intrusions. Leveraging the NSL-KDD dataset, various machine learning methods were applied and examined in the context of network intrusion detection. The performance of the models was measured using key metrics such as accuracy, precision, recall, and F-score, providing a comprehensive assessment of the results. This section intends to elucidate the experimental findings, elucidating the comparative efficacy of the employed methods, and highlight the potential advantages and limitations of the proposed framework in a real-world context.

Presented in Fig. 5 are confusion matrices associated with the application of six distinct machine learning models — as they are utilized in the complex problem domain of network intrusion detection. Each confusion matrix serves as a crucial visual tool, succinctly encapsulating the performance of a given model by displaying the interplay of true positive, true negative, false positive, and false negative predictions. Consequently, these matrices offer a nuanced understanding of model performance, not only showcasing the number of correct and incorrect predictions, but also highlighting the nature of errors made. By drawing upon these comprehensive insights, we can systematically compare the respective models' effectiveness in detecting network intrusions, thus paving the way for a data-driven selection of the most adept methodology.

Fig. 6 showcases the Receiver Operating Characteristic (ROC) curves and associated Area Under the Curve (AUC) values for six machine learning models employed in the task of network intrusion detection.

Each ROC curve graphically depicts the true positive rate (sensitivity) versus the false positive rate (1-specificity), at various threshold settings, enabling a clear representation of the trade-off between sensitivity and specificity for the given models. The AUC, on the other hand, provides an aggregate measure of the model's performance across all possible classification thresholds.

Through an evaluation of the ROC-AUC plots, we can comparatively assess the performance of the models, offering insights into their relative effectiveness in distinguishing between normal and attack instances in network traffic data. This detailed analysis aids in identifying the most promising model for intrusion detection. The experimental outcomes reveal that the k-Nearest Neighbors (kNN) algorithm displays superior performance in identifying network intrusions, as evidenced by its superior metrics across the evaluation parameters. Conversely, indications of overfitting are discernible in the performance of both the Random Forest and Support Vector Machines (SVM) classifiers. Overfitting is a modelling error which occurs when a function is too closely aligned to a limited set of data points, thereby impeding the model's ability to generalize to new data. This propensity for overfitting within these classifiers may compromise their effectiveness in a real-world network intrusion detection context. Overall result of ROC-AUC curves of the applied six methods show that, the proposed framework for intrusion
detection based on machine learning methods is applicable for practice. In case of intrusion detection framework, the best ROC-AUC curve model can be used as a main algorithm in intrusion detection system.

Fig. 5. Confusion matrices for machine learning methods in network intrusion detection problem.

Fig. 6. AUC-ROC curves for machine learning methods in network intrusion detection problem.
Fig. 7. Train and test accuracies of machine learning methods in network intrusion detection problem.

Fig. 7 offers a comparative visualization of the training and test accuracies achieved by the applied machine learning methods in the context of network intrusion detection on the NSL-KDD dataset.

Training accuracy provides a measure of the model's performance on the training dataset, reflecting the ability of the model to fit the given data. Test accuracy, on the other hand, measures the model's performance on an unseen dataset, indicative of the model's capacity to generalize beyond the training data.

The graph depicted in Fig. 7 enables us to scrutinize the interplay between these two types of accuracies for each applied method. By comparing training and test accuracies, we gain insights into potential overfitting or underfitting scenarios, which are critical for understanding the effectiveness of the models. High training accuracy accompanied by low test accuracy often suggests overfitting, whereas a low training accuracy may indicate underfitting.

This comparative assessment serves to inform subsequent decisions about model selection and the need for potential adjustments in model complexity or training procedures to optimize performance.

Fig. 7 provides a comparative analysis of the training and test precisions attained by the applied machine learning methods in the context of intrusion detection using the NSL-KDD dataset.

Training precision assesses the model's ability to accurately predict positive instances within the training dataset, while test precision evaluates the model's performance on unseen data. By comparing the training and test precisions depicted in Fig. 8, we can discern insights into potential overfitting or underfitting scenarios.

Examining the interplay between training and test precisions aids in determining the models' ability to generalize and effectively classify network intrusions. Higher training precision with a significant drop in test precision may indicate overfitting, highlighting the need for adjustments to enhance generalization. Conversely, low training and test precisions may suggest underfitting, necessitating model refinement or reconsideration of feature selection.

These findings from Fig. 8 contribute to the understanding of model performance and guide the selection of the most suitable methods for intrusion detection tasks.

Fig. 9 provides a comparative analysis of the training and test recalls achieved by the applied machine learning methods in the domain of intrusion detection using the NSL-KDD dataset.

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Training recall measures the model's ability to correctly identify the positive instances within the training dataset, while test recall assesses the model's performance on unseen data. By examining the interplay between training and test recalls depicted in Fig. 9, we gain insights into the models' capacity to generalize and accurately capture relevant instances of network intrusions.

Discrepancies between training and test recalls can signify potential overfitting or underfitting. High training recall accompanied by a significant drop in test recall may indicate overfitting, necessitating measures to enhance generalization. Conversely, low training and test recalls may suggest underfitting, warranting model refinement or reassessment of feature selection.

The findings presented in Fig. 9 contribute to the understanding of model performance and guide the selection of appropriate methods for effective intrusion detection in real-world scenarios using the NSL-KDD dataset.

VII. DISCUSSION

The results obtained from the experiments conducted on the proposed novel framework for detecting network intrusions based on machine learning methods provide valuable insights into its efficacy and comparative performance. In this discussion, we delve into the key findings, highlight the strengths and limitations of the framework, and address potential avenues for further improvement.

The experimental results demonstrated that the k-Nearest Neighbors (kNN) algorithm exhibited superior performance in detecting network intrusions on the NSL-KDD dataset. This is evident from its high accuracy, precision, recall, and F-score values compared to the other applied methods. The kNN algorithm's ability to identify similar instances based on proximity in feature space, without making strong assumptions about the underlying data distribution, likely contributed to its success in this context. The results highlight the potential of instance-based methods, such as kNN, in handling network intrusion detection tasks.

On the other hand, the Random Forest classifier and Support Vector Machines (SVM) showed indications of overfitting. Overfitting occurs when a model becomes too closely aligned with the training data, leading to poor generalization to unseen data. This is a common challenge in machine learning, particularly when dealing with complex datasets such as network intrusion detection [34]. To mitigate overfitting, techniques such as regularization, feature selection, or hyperparameter tuning can be employed to improve the models' generalization capability.

The experimental results also provided insights into the performance of other applied methods. Decision Tree, Gaussian Naive Bayes, and Logistic Regression exhibited competitive performance, albeit slightly lower than that of kNN. These methods have their own strengths and weaknesses, and their suitability may vary depending on the specific requirements and characteristics of the intrusion detection problem at hand. Further exploration of these methods, including ensemble techniques such as AdaBoost, could potentially yield improved results.

The framework's utilization of the NSL-KDD dataset, a well-established benchmark dataset, adds credibility to the results. The dataset's diverse range of network intrusion types and comprehensive set of features enable a realistic evaluation of the proposed framework. However, it is important to acknowledge that the NSL-KDD dataset itself has some limitations, such as the inclusion of preprocessed data and the potential bias introduced during data collection [35]. Future studies should consider incorporating other datasets and real-world network traffic to further validate the framework's performance in practical settings.

The comparative analysis of training and test accuracies, precisions, recalls, and F-scores provided crucial insights into the generalization capabilities of the models. Discrepancies between training and test performance metrics can signify overfitting or underfitting [36]. In this context, attention should be paid to the models exhibiting high training performance but significantly lower performance on the test set, indicating overfitting. Strategies such as regularization techniques, cross-validation, or early stopping can help alleviate overfitting issues and enhance the models' generalization.

Moreover, the findings suggest that careful consideration should be given to the selection and tuning of hyperparameters for each machine learning algorithm [37]. Fine-tuning the algorithms' parameters, such as the number of neighbors in

![Fig. 9. Train and test recall of machine learning methods in network intrusion detection problem.](image-url)
kNN or the maximum depth of decision trees, can significantly impact their performance [38]. Conducting a comprehensive hyperparameter search using techniques like grid search or Bayesian optimization can potentially yield further performance improvements [39].

While the proposed framework demonstrated promising results, it is important to acknowledge its limitations. Firstly, the evaluation was conducted on a specific dataset, and the performance may vary when applied to other datasets or real-world network environments. The framework's adaptability to different network architectures, traffic patterns, and attack scenarios remains an area for future exploration [40]. Additionally, the framework primarily focused on supervised learning methods [41], neglecting the potential benefits of unsupervised or semi-supervised approaches in network intrusion detection [42]. Future research could incorporate hybrid models or anomaly detection techniques to further enhance the framework's capabilities.

In conclusion, the experimental results presented in this study highlight the effectiveness of the proposed novel framework for detecting network intrusions based on machine learning methods. The k-Nearest Neighbors algorithm emerged as a top-performing method, outperforming the other applied algorithms in terms of accuracy, precision, recall, and F-score. The results underscore the importance of careful model selection, hyperparameter tuning, and addressing overfitting issues to ensure optimal performance. The findings contribute to the existing body of knowledge in network intrusion detection and provide a foundation for further research and development of robust and adaptable frameworks for network security. Last time, machine learning are used in different areas from medicine to smart cities [43-45]. In this research, we applied machine learning in network intrusion detection problem. As the obtained results show, machine learning gives high efficiency in this area, too.

**VIII. CONCLUSION**

In this study, we proposed a novel framework for detecting network intrusions based on machine learning methods and evaluated its performance on the widely used NSL-KDD dataset. The experimental results demonstrated the effectiveness of the framework in identifying network intrusions, with the k-Nearest Neighbors (kNN) algorithm emerging as the top-performing method. The framework's comprehensive evaluation metrics, including accuracy, precision, recall, and F-score, provided a comprehensive assessment of its performance.

The results highlight the importance of selecting appropriate machine learning algorithms and fine-tuning their hyperparameters to achieve optimal performance in network intrusion detection tasks. The kNN algorithm's success can be attributed to its ability to leverage proximity-based learning and handle complex patterns in the dataset. Furthermore, the comparative analysis of training and test accuracies, precisions, recalls, and F-scores shed light on potential overfitting issues and underscored the significance of model generalization.

While the proposed framework demonstrated promising results, it is important to acknowledge its limitations. The evaluation was primarily conducted on the NSL-KDD dataset, and the framework's performance may vary on different datasets or real-world network environments. Additionally, the framework focused on supervised learning methods and neglected the potential benefits of unsupervised or semi-supervised approaches. Exploring hybrid models and incorporating anomaly detection techniques could enhance the framework's capabilities in detecting previously unseen attacks.

In conclusion, the proposed novel framework presents a viable approach for network intrusion detection based on machine learning methods. The experimental results validate its effectiveness and highlight the importance of careful algorithm selection and hyperparameter tuning. The framework can serve as a foundation for further research in developing robust and adaptive intrusion detection systems, safeguarding network security in an evolving threat landscape. Future studies should expand the evaluation to include other datasets and consider the integration of unsupervised and semi-supervised approaches for improved performance and versatility.

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A Dynamic Model for Risk Assessment of Cross-Border Fresh Agricultural Supply Chain

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Abstract—The cross-border trade of Fresh Agricultural Products (FAP) is widespread in the current society, and the demand for it is also increasing. The cross-border fresh agricultural product Supply Chain (SP) itself has strong complexity and high costs, and it also bears many risks. In order to alleviate the adverse impact of risk factors interfering with cross-border fresh agricultural product SPs and improve the overall SP efficiency, this study proposes a system dynamics model based on cross-border fresh agricultural product risk factors. The experiment first studied the possible risk factors in the SP of FAP. After discussing the causal relationship between possible risks, subjective and objective weighting methods were introduced to weight risk factors. After that, a system dynamics model of the cross-border fresh agricultural product SP was constructed for the purpose of enhancing product quality and the overall efficiency of the SP. In the system dynamics model constructed, risk factors are introduced for simulation experiments. It is demonstrated that the suggested model can truly reflect the dynamic changes of the actual SP, and can obtain the operational rules of the system.

Keywords—Cross-border fresh agricultural products; supply chain management; risk identification; system dynamics model; risk weighting

I. INTRODUCTION

As a large country of agricultural production and consumption, the rational development of agriculture is the basis of the national economy and the necessary condition for the survival of other material sectors. Today, with the continuous advancement of economic globalization, the import and export scale of China's agricultural products has been firmly in the forefront of the world [1]. The increasing trade in Fresh Agricultural Products (FAP) has deepened international exchanges, but it also poses challenges to the Supply Chain (SP) management of cross-border FAP [2]. FAP, compared to other products, have the characteristics of perishability and are not easy to preserve, so their requirements for logistics and transportation are high. In addition, customs inspection of cross-border trade is relatively strict, and transportation time and distance are relatively longer, resulting in more risks. The cross-border fresh agricultural product SP has a higher complexity and a greater likelihood of risk occurrence [3]. Therefore, effective identification and timely avoidance of risk factors in the cross-border fresh agricultural product SP is crucial to ensuring the quality of FAP. System dynamics model is a qualitative and quantitative research method that can simulate and analyze the connection and development of complex problems from a holistic perspective, and is applicable to this research topic [4]. Currently, the academic community has conducted relevant research on the SP risk of cross-border FAP, and has also achieved certain results. However, it mainly focuses on SP inventory control, ordering strategies, and other aspects. There are few studies that use risk factors and system dynamics models to find the impact of risk variables on the SP [5]. Therefore, this experiment takes the cross-border fresh agricultural product SP as the research object, and introduces subjective and objective weighting methods to weight its risk factors. Based on this, a system dynamics model is constructed to explore the best solution for enhancing the overall efficiency of the SP.

The innovation of this study lies in: (1) The introduction of subjective and objective weighting methods to weight the risk factors of cross-border fresh agricultural products supply chain. (2) The system dynamics model of cross-border fresh agricultural products supply chain is constructed.

The study is divided into five parts. The first part is the introduction, which introduces the research background and significance; the second part is a literature review, which introduces the current development status of supply chain factor identification and system dynamics model. The third part is the establishment of system dynamics model for operational risk assessment of cross-border fresh agricultural supply chain. The fourth part is the performance verification of the constructed model. The last part is the summary of the full text and the prospect of future research.

II. RELATED WORK

Currently, researchers have discussed the methods for identifying SP risk factors. Zhao and other researchers used topic analysis, fuzzy cross impact matrix multiplication analysis, and other methods to effectively manage the complexity and vulnerability of agricultural food SPs. The experiment classified risks based on their dependencies and driving forces, thereby helping to determine the relationship between risks and the most critical risk factors. The final result promoted the study of risk factors in the agricultural food SP [6]. To explore the sustainable impact of flood risk drivers on agricultural SPs, scholars such as Yazdani proposed a multi standard approach to assess flood risk in crop regions. This method ranked agricultural projects affected by floods to detect the best projects, thereby mitigating the greatest impact of flood risk on crop areas. This method had important practical significance for preventing flood risk [7]. In order to determine the impact of standardized management systems on selected risks in the SP, Zimon et al. applied basic data analysis methods to investigate logistics company staff. The
experiment was conducted using a questionnaire survey, and the results showed that improving SP management can help managers [8]. Wang and other researchers conducted empirical research on the SP risk management of express delivery companies. The experiment mainly started with exploring the relationship between innovation capability and SP risk, and established a partial least squares method for structural equations based on the survey data. The study found that there is a negative correlation between them, so enterprises can try to decrease the negative impact of SP risk by developing their own innovation capability [9]. From the perspective of risk factors in the halal food SP, Khan S and other teams used a hierarchical fuzzy analysis method to rank the identified risk factors. This method can effectively rank the risk factors in the halal food SP, thereby helping managers take effective measures to mitigate risks. This result has practical significance for studying the risk management of halal food SP [10].

System dynamics model is a structural model that simulates the dynamic changes of a system, and has a wide range of applications in various fields. Based on system dynamics models, researchers such as Papachristos developed a simulation model that combines standard competitive dynamics theory. This model could be used as a powerful tool for enterprises to obtain competitive advantage. Simulation experiments on four standard competition cases showed that the simulation results of are consistent with real cases. Therefore, the proposed model could lay the foundation for theoretical and empirical research on standard competitive strategy [11]. Liu et al. established an environmental assessment model for construction waste using a system dynamics model. The impact on the environment, economy, and society was analyzed in the experiment. The final simulation results can provide reference value for construction waste treatment [12]. Rathore and other scholars used system dynamics models to promote the interaction between dynamic feedback effects and risks in grain transportation. The experiment used a system dynamics model that considers the value of the risk index to observe the impact of the risk value on grain inventory levels and vehicle capacity. The model proposed in the experiment could help improve food supply through comprehensive risk control in the SP, and can improve the efficiency of the food SP [13]. Sayyadi and other researchers proposed an integrated approach based on system dynamics and analytical networks to assess the sustainability of transportation policies. This method evaluated and ranks five policies, namely, travel sharing, reducing travel rate, reducing road network length, vehicle ownership, and average driving kilometers, through the third indicator of congestion degree, fuel consumption, and emissions. The research results verified the effectiveness of the constructed system [14]. Researcher Oleghe developed a system dynamics model based on end-to-end agribusiness and aquaculture SP models from the perspective of capacity expansion of aquaculture companies. This model covers unique dynamics related to the aquaculture SP and enables simulation of company working capital management rules. Experiments have verified that the proposed model can be applied to companies’ management of working capital under different financing modes and capacity expansion rates [15].

To sum up, both have relevant applied research. However, there are still many limitations in the above research. For example, the construction of the index system of risk factors is not perfect, the effect of risk simulation is not good, and the risk management method still needs to be further improved. At present, there are no relevant studies combining risk factor variables with system dynamics models, and the description of cross-border FAP risk factors is insufficient. Therefore, this study introduced subjective and objective weighting methods, weighted its risk factors, and established a system dynamics model to explore ways to improve the overall efficiency of SP.

III. CONSTRUCTION OF A SYSTEM DYNAMICS MODEL FOR RISK ASSESSMENT OF CROSS-BORDER FRESH AGRICULTURAL PRODUCT SP OPERATION

A. Research on Main Risk Factors of Cross-Border Fresh Agricultural Product SP

The SP of traditional cross-border FAP is mainly composed of three main bodies: overseas SPs, domestic buyers, and relevant consumers. Its sales channels are mainly agricultural markets, supermarkets, and distribution outlets. The traditional sales channel that occupies the main position is the supermarket. The purchaser of the store first proposes an order demand to the overseas supplier; after receiving the order, the overseas supplier shall ship the goods by sea or air. During this process, FAP need to undergo customs clearance, inspection and quarantine before reaching the domestic purchaser's warehouse [16,17]. To reduce costs, cross-border e-commerce and other direct procurement operation models have gradually emerged in recent years. This mode is mainly for cross-border e-commerce enterprises to directly connect with overseas agricultural product suppliers and deliver products to consumers through front-end warehouses [18,19]. The operation mode of cross-border e-commerce has greatly shortened the length of the SP, effectively reducing production costs. Fig. 1 illustrates the overall operational process of the traditional cross-border fresh agricultural product SP and the cross-border e-commerce fresh agricultural product SP. The cross-border fresh agricultural product SP under both modes has similar operational links, namely, the supply link, the transportation link, and the sales link. In fact, various unexpected situations and constraints in reality have posed challenges to the SP management of cross-border FAP. Therefore, the three links of the SP are faced with varying degrees of risk possibilities.
Currently, FAP are vulnerable to risks due to the international situation, cold chain transportation, bottlenecks, import and export food safety, and other conditions. FAP are vulnerable to environmental changes. During the COVID-19 epidemic, people’s demand for FAP increased sharply. Meanwhile, FAP appeared to be out of stock and not delivered in a timely manner. In this context, the cross-border fresh agricultural product SP is also subject to many restrictions, resulting in a longer overall warehousing time for goods. Secondly, FAP has high requirements for cold chain logistics of enterprises. Cold chain technique is a key technology to ensure the quality of FAP, so it has significant constraints on the SP of FAP. The cold chain logistics technology for FAP in China is still immature, and compared to other developed countries, the overall loss rate of products is relatively high, as shown in Fig. 2. Thirdly, the customs clearance efficiency of China’s cross-border fresh agricultural product SP is low, which leads to low supply efficiency and poor operation of the entire chain [20]. Finally, China attaches more importance to the safety issues of imported and exported food quality, but the technology for safety testing needs to be strengthened. Compared to other cross-border commodities, FAP are perishable and resistant to bumps, with higher transportation requirements and costs [21]. Therefore, it is particularly important to accurately identify the actual operational risk factors in their SP.

The risk factors of the cross-border fresh agricultural product SP mainly include the problems that are prone to occur in the operation of the SP and the statistical analysis results of existing SP accidents that have occurred. The main risk factors can be divided into three aspects, namely, SP risk factors, transportation chain risk factors, and sales chain risk factors. In the SP, supply delay risk, inventory risk, quality safety risk, and supply risk are prone to occur. Supply delay refers to the delay in delivery at the source caused by suppliers’ inability to prepare goods from the origin on time, which has a negative impact on the operational efficiency of the entire SP [22]. Inventory risk refers to the phenomenon of excess inventory or shortage caused by supplier information lag or stock preparation delay. The quality safety risk is that the quality of FAP at the source of the supplier does not meet the standard. Supply risk refers to the insufficient supply quantity and poor product quality of the overall product. The risk factors faced in the transportation process mainly include shipment delay risk, cold chain risk, unexpected risk, and customs clearance risk. The risk of delivery delay refers to the failure of suppliers to deliver goods on time, resulting in poor operation of the total SP. Cold chain risk refers to the possibility of product loss caused by substandard cold chain technology. Sudden risks refer to unexpected events such as weather disasters and wars encountered during transportation. The destruction of products by sudden risks is irreversible, and the losses brought to the SP are irreparable. Customs clearance risk refers to the problems of excessive customs clearance time and low customs clearance efficiency during the quarantine process of commodity export and import. The final sales process also faces major risk factors such as sales delay risk, market risk, sales inventory risk, and cold chain risk. Fig. 3 demonstrates the causal relationship between various types of risks. In the figure, S represents an overseas supplier; P represents the domestic purchaser; “+” indicates positive feedback and “-” indicates negative feedback.

![Fig. 1. Operation process of two SPs.](image1)

![Fig. 2. Comparison of fresh agricultural product consumption rates between China and developed countries.](image2)
Fig. 3. The causal relationship between various types of risks.

B. Construction of System Dynamics Model for Cross-Border Fresh Agricultural Product SP

The previous section describes the causal relationship of risk factors in the SP of FAP. Common risks in cross-border fresh agricultural product SPs include supply delay risk, inventory risk, quality and safety risk, supply risk, shipment delay risk, cold chain risk, sudden risk, customs clearance risk, sales delay risk, market risk, sales inventory risk, and cold chain risk. For the establishment of this system dynamics model, it is first necessary to convert the above risks into state variables in the system. The above variables can be determined as state variables in the model, and their corresponding changes are rate variables. When using system dynamics to establish a risk assessment model for the operation of the fresh agricultural product SP, it is required to adopt some methods to define the functional relationship between various risk factors [23]. Among them, the most commonly used is the linear functional relationship, which is to establish system dynamics equations by calculating the weights of various risk factors. The weight determination method starts from three aspects: subjective weight, objective weight, and comprehensive integration weight [24]. The overall process of comprehensive empowerment through subjective and objective empowerment methods is shown in Fig. 4.

Firstly, the subjective weight is determined using the Order Relationship Analysis Method (G1) [25]. Assuming that there are \( N \) risk factors, ranking them according to the importance of each risk indicator can obtain the ranking results shown in Eq. (1):

\[
a_1^* \geq a_2^* \geq \ldots \geq a_n^*
\]

In Eq. (1), \( a_i \) represents the evaluation criteria and indicators. The ratio between \( a_i \) and \( a_{i-1} \) is the degree of relative importance, recorded as \( t_i \). After reasonably assigning a value to \( t_i \). Calculate the weight coefficient using Eq. (2):

\[
m_i = \left(1 + \sum_{i=2}^{n} \prod_{k=i}^{n} t_k \right)^{-1}
\]

In Eq. (2), both \( m_i \) and \( t_{n-1} \) represent weight coefficients. When using the G1 method to determine the weight, there is a situation where team experts' scores are inconsistent, so it is necessary to discuss the importance level. Assuming that there are \( x( x \geq 1) \) experts in total, among which \( z \) experts have the same results for ranking indicators, there is an Eq. (3):

\[
t_i^* = \frac{1}{a} \sum_{i=1}^{x} t_i
\]

In Eq. (3), \( t_i^* \) represents the ratio of the relative importance between indicators \( a_i \) and \( a_{i-1} \). Then, the weight coefficient of this \( z \) expert can be determined as:

\[
m_i = \left(1 + \sum_{i=2}^{n} \prod_{k=i}^{n} t_k \right)^{-1}
\]

\[
m_{i-1} = t_i m_i \quad (i = n, n-1, \ldots, 2)
\]

Aside from that, assuming that the evaluation results of the remaining \( x - z \) experts are different, averaging them can obtain Eq. (5):

\[
m_i = \frac{\sum_{j=1}^{x} m_j}{x - z}
\]

Finally, combining the two types of results, Eq. (6) can be obtained:

\[
m_i = k_1 m_i^* + k_2 m_i
\]

In Eq. (6), \( k_1 = \frac{z}{x} \quad , k_2 = \frac{x - z}{x} \). After determining the subjective weight, it is necessary to discuss the objective weight. The difference from subjective weight is that the final result of objective weight does not rely on subjective judgment, but rather decides the indicator weight based on the information of the sample itself. Entropy weight method is a commonly used objective weighting method, which mainly determines the weight through the degree of variation of indicators. The size of the entropy value can reflect the degree of variation of the index. The larger the entropy value, the greater the degree of variation and the more information it covers, so the higher the weight value. Suppose there is a judgment matrix \( \left( A^P \right) \) that contains the judgment values of \( n \) experts on \( m \) risk factors, and the specific expression is:

\[
P = \begin{bmatrix}
A_{11} & A_{12} & \ldots & A_{1m} \\
A_{21} & A_{22} & \ldots & A_{2m} \\
\vdots & \vdots & \ddots & \vdots \\
A_{n1} & A_{n2} & \ldots & A_{nm}
\end{bmatrix}
\]
Standardize the data contained in Eq. (7) to obtain Eq. (8):

\[ P_{ij} = \frac{A_{ij} - \min(A_{ij})}{\max(A_{ij}) - \min(A_{ij})} \quad (8) \]

In Eq. (8), \( \min(A_{ij}) \) denotes the minimum value in the original judgment matrix; \( \max(A_{ij}) \) denotes the maximum value in the original judgment matrix; \( P_{ij} \) represents a standardized value. The equation for calculating the proportion of each index is shown in Eq. (9):

\[ r_{ij} = \frac{P_{ij}}{\sum_{j=1}^{n} P_{ij}}, \quad (i = 1, 2, \ldots, m) \quad (9) \]

Eq. (9) represents each indicator value’s weight, so the entropy weight calculation Equation for the \( i \) index is:

\[ w_i = \frac{1 - E_i}{m - \sum_{i=1}^{m} E_i} \quad (10) \]

After determining the subjective and objective weights, comprehensive weighting can be performed. This model can combine the advantages and characteristics of subjective and objective weighting methods to enhance the scientificity and rationality of the final result. The linear weighting method can not only compensate for the shortcomings caused by the uneven numerical values of other methods, but also have a relatively concise calculation process for finding the optimal combination weight. Therefore, this experiment uses a linear weighting method to restructure the weights. The basic calculation Equation is as follows:

\[ Q_i = \beta m_i + (1 - \beta)w_{ij}, \quad 0 \leq \beta \leq 1 \quad (11) \]

In Eq. (11), \( \beta \) denotes the proportional coefficient. The overall calculation equation is shown in Eq. (12):

\[ Q_i = \frac{m}{m-1} \left[ \frac{2}{m} (P_1 + 2P_2 + \ldots + mP_m) - \frac{m+1}{m} \right] \quad (12) \]

In Eq. (12), \( P_{ij} \) represents the weight value of the \( P_{ij} \) values of each indicator factor after being sorted in ascending order. In addition to the above risk variables, there are also some boundary risk variables that cannot directly obtain data. For the assignment of boundary risk variables, this experiment was conducted using expert scoring. The scoring rules are shown in Eq. (13):

\[ a_{ij} = (x + y + n)/7 \quad (13) \]

In Eq. (13), \( x, y, n \) represents the minimum, maximum, and most likely values that boundary risk factors may have an impact on the fresh agricultural product SP. \( i \) represents the \( i \)-th risk factor; \( j \) represents the \( j \)-th expert. Average all the obtained results to obtain the corresponding boundary risk value. Draw a system flow diagram for all risk variables based on the causal relationship diagram in the previous section. Fig. 5 shows the basic form of a system dynamics model for the cross-border fresh agricultural product SP.

![Fig. 5. Basic form of system dynamics model for cross-border fresh agricultural product SP.](www_ijacsa_thesaio rg)
IV. SIMULATION EXPERIMENT ON SYSTEM DYNAMICS MODEL OF CROSS-BORDER FRESH AGRICULTURAL PRODUCT SP

A. System Dynamics Model Test of Cross-Border Fresh Agricultural Product SP

To make the logic of the constructed model and functional equation reasonable, and ensure that the system operation can reflect the actual situation to a greater extent, model verification is conducted before conducting model simulation experiments. First, set the constants for the overall model, and the values set are obtained based on the relevant data in the industry report. To verify whether the model can truly, stably, and continuously reflect the actual change rules, experiments were conducted to test the realistic reproducibility of the model. In order to make the model truly reproducible, experiments were conducted to examine both extreme and actual situations. Table I shows the settings for some of these variables. The experiment was conducted using Vensim software.

Fig. 6 shows the model test results under extreme conditions where the market demand is zero. Set the market demand to a limit condition of 0, with the simulation starting at 0, ending at 100 months, and step length of 1 month. Then observe the operation effect of the system. When the market demand level is 0, the buyer does not sell the product and will not place an order. Therefore, both the sales and order ratios are 0. When the buyer does not need to place an order, the seller will not provide the goods and will not ship them. Therefore, both the supply ratio and the shipment ratio are 0. The company's inventory level remains at its initial level of 0. The model can be tested under extreme conditions.

<table>
<thead>
<tr>
<th>Variable</th>
<th>Numerical Value</th>
<th>Variable</th>
<th>Numerical Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Market price</td>
<td>101.22</td>
<td>Order Price</td>
<td>60.89</td>
</tr>
<tr>
<td>Purchaser smoothing time</td>
<td>2.13</td>
<td>Supplier smoothing time</td>
<td>2.16</td>
</tr>
<tr>
<td>Purchaser's expected inventory time</td>
<td>3.27</td>
<td>Supplier's expected inventory time</td>
<td>3.47</td>
</tr>
<tr>
<td>Purchaser inventory adjustment time</td>
<td>4.32</td>
<td>Supplier inventory adjustment time</td>
<td>5.24</td>
</tr>
<tr>
<td>Supplier Cost Quality Factor</td>
<td>45000</td>
<td>Cold chain cost coefficient</td>
<td>40000</td>
</tr>
<tr>
<td>Supplier attrition rate</td>
<td>0.1</td>
<td>Purchaser's loss rate</td>
<td>0.22</td>
</tr>
<tr>
<td>Supplier non refrigerated rate</td>
<td>0.1</td>
<td>Supplier non refrigerated rate</td>
<td>0.56</td>
</tr>
<tr>
<td>Supplier unit inventory cost</td>
<td>5.67</td>
<td>Supplier's unit transportation cost</td>
<td>20</td>
</tr>
<tr>
<td>Supplier unit supply cost</td>
<td>10.19</td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

Fig. 6. Model test results under extreme conditions with market demand of 0.

After that, use the model to test the actual situation. Similarly, set the start time to the 0th month, the end time to the 100th month, and the step length to 1 month. At the same time, set the risk of sales delay in the 10th to 20th months; Risk of supply delay in the 20th to 40th months; the risk of shipment delay occurs in the 60th to 80th months; sudden risks occur in the 50th month. Fig. 7 shows the changes in purchasing inventory during this period. In the 10th month, there was a risk of sales delays, i.e. a decrease in the purchaser's sales rate, resulting in an increase in inventory and a decrease in the supplier's delivery rate. In the 50th month, there was a sudden risk that the purchaser's inventory decreased, leading to an increase in the supplier's delivery rate until it stabilized. In the 60th month, the risk of delivery delay occurred, and the supplier's delivery rate decreased, resulting in a slight decrease in the purchaser's inventory. In the 80th month, the risk of shipment delay disappeared and the purchaser's inventory gradually returned to normal levels. The operation of the model coincides with the actual situation.

Fig. 7. Purchaser inventory changes.

Fig. 8 shows the inventory changes of suppliers during the above time period. In the 10th month, the supplier's shipment rate suddenly decreased, and the supplier's inventory was still able to meet downstream ordering requirements, so the supply
rate was 0. After that, when the supplier's shipment rate began to pick up, the supply rate also began to gradually rise. However, the growth rate of the supply rate is slower than the delivery rate, until the two are equal, the supply inventory decreases to the minimum. When the supplier's inventory is at the lowest level, the supply rate will peak in a short period of time. Then, when the supplier's shipment volume remains basically constant, the supply rate will lag behind the inventory. The risk of shipment delay occurs between 60 and 80 months, and the supplier's inventory changes exhibit a similar pattern. The operation of this model is consistent with the actual situation.

**B. Simulation Analysis of Key Risk Variables in the Cross-Border Fresh Agricultural Product SP System Dynamics Model**

Fig. 9 shows the level of impact of supplier quality input on the SP. As suppliers invest more in quality, product quality will gradually improve, thereby promoting an increase in market demand. As a result, the overall profit of the market has increased. However, for suppliers, the marginal benefits generated by continuing to increase investment in product quality are diminishing. The growth rate of product quality and market demand has stabilized, and high investment also means high costs. Therefore, the total profits of buyers and suppliers will gradually increase to a certain level and remain constant. This indicates that increasing the level of quality input from suppliers can improve the overall revenue of the SP, but there is an upper limit.

Fig. 10 shows the impact of the purchaser's cold chain investment on the SP. The increased investment of buyers in the cold chain has led to the improvement of product quality, which has promoted the increase in market demand. Therefore, the overall interests of the market will increase accordingly. However, after investing in the cold chain to a certain level, the marginal income will suddenly rise and then decline after accumulating to a certain level. This is because cold chain logistics requires a certain amount of upfront investment. In the early stages of cold chain transportation, a large amount of money needs to be spent, resulting in a small return on investment in the early stage. High investment brings high costs. Thus the overall returns of buyers and suppliers will gradually increase until they remain unchanged. As a result, increasing the investment of suppliers in the cold chain can improve the overall efficiency of the SP, but also has an upper limit.
Fig. 11 shows the impact of various solutions on the SP system. There are four different schemes in the figure: Scheme 1 is the initial setting; Scheme 2: Increase input for suppliers; Scheme 3: Increase the degree of investment for the purchaser; Scheme 4 is the degree of joint cooperation investment. It can be seen from the figure that suppliers and purchasers will increase their investment in the product, which will improve the final quality of the product and the overall profit of the SP. When all members of the SP jointly increase their investment, both the benefits of both parties and the overall benefits of the SP are higher than when one party alone increases its investment level. Therefore, the two parties should actively establish a cooperative relationship and choose appropriate incentives. Suppliers increase their investment in product quality to improve quality products; the purchaser should increase investment in the cold chain and improve the cold chain management ability. This will maximize benefits while ensuring the overall efficiency of the SP.

V. CONCLUSION

Nowadays, the increasing demand for cross-border FAP poses a significant challenge to the management of their SP. To reduce the interference of risk factors on the cross-border fresh agricultural product SP, and thereby enhance the efficiency of the overall SP, the experiment weighted the possible risk factors and constructed a system dynamics model. The test results of the system dynamics model show that the proposed method can pass the test under extreme conditions, and its operation is consistent with the actual situation. Simulation experiments were conducted to introduce risk factors into the constructed system dynamics model. The simulation results indicate that increasing the supplier's quality input can enhance the overall revenue of the SP, but there is an upper limit; increasing the investment of suppliers in the cold chain can enhance the overall efficiency of the SP, but it also has an upper limit. Therefore, to ensure the overall efficiency of the SP and maximize benefits, suppliers increase their investment in product quality and improve the products’ quality; the purchaser should increase investment in the cold chain and optimize the cold chain management ability. The proposed model can truly reflect the dynamic changes of the actual SP, and has good practicality. This experiment still has some limitations, that is, it fails to consider external macro risks and other factors, so the subsequent research can start from here.

REFERENCES


Design of an Educational Platform for Professional Development of Teachers with Elements of Blockchain Technology

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Abstract—This paper presents an in-depth examination of the development and implementation of an innovative platform for teacher professional development, incorporating features of blockchain technology. The platform manifests a revolutionary step in enhancing teacher training, creating a secure, transparent, and decentralized approach for maintaining continuous professional development records. Using blockchain's inherent properties, the platform ensures immutable record-keeping and instills credibility in teachers' career progression, empowering educators through direct ownership of their professional development milestones. Additionally, the platform fosters a culture of lifelong learning, encouraging educators to actively engage in their professional growth, while providing reliable evidence of their achievements. Alongside highlighting the design aspects of the platform, the paper delves into potential challenges and solutions associated with the incorporation of blockchain technology into educational contexts. Through this innovative intersection of technology and education, the platform showcases the potential of blockchain in reshaping and enriching professional development strategies for teachers, thereby elevating educational standards and practices across the board.

Keywords—Blockchain; professional development; artificial intelligence; teaching; learning

I. INTRODUCTION

The advent of the digital age has brought about transformative changes across industries worldwide. In education, technology's role in shaping instructional strategies, fostering collaboration, and enhancing student outcomes is well-recognized [1]. However, an area less explored, but no less critical, is the use of technology to facilitate and optimize the professional development (PD) of teachers [2]. To that end, this paper aims to describe the design and development of a novel platform that leverages elements of blockchain technology to revolutionize the PD landscape for teachers.

Teacher PD refers to a comprehensive set of specialized training, formal education, and advanced professional learning aimed at enhancing teachers' pedagogical skills and knowledge [3]. The effectiveness of a teacher's professional development program is central to the quality of education imparted. Research has indicated that high-quality PD positively impacts teacher quality, which, in turn, improves student achievement [4]. However, the traditional methods of documenting and validating PD efforts often involve arduous, time-consuming paperwork, and a lack of transparency and credibility.

Blockchain technology, renowned for its robust security, decentralization, and immutable record-keeping capabilities, can provide efficient solutions to these challenges [5]. While blockchain has mostly been associated with cryptocurrencies like Bitcoin, its potential extends far beyond, having disruptive implications for sectors like healthcare, finance, and supply chain management [6]. In the realm of education, blockchain holds promise for academic credential verification, student record management, and, as explored in this paper, enhancing teacher PD.

In response to the need for a more efficient, transparent, and credible system for managing teacher PD, we have developed a unique platform incorporating blockchain technology. The platform serves as a decentralized ledger for recording, verifying, and sharing professional development activities [7]. It provides teachers with direct control over their records and the ability to share their verified qualifications and skills seamlessly, fostering greater trust and collaboration within the educational community.

The application of blockchain technology to teacher PD also aligns with the increasing emphasis on lifelong learning [8]. The platform supports a culture of continuous learning and improvement, allowing educators to document their professional growth over time reliably [9]. By reinforcing the value of PD and enhancing its accessibility and transparency, blockchain technology can contribute significantly to enriching teacher quality and, consequently, student learning experiences [10].

This paper provides a comprehensive overview of the platform's design, development, and implementation processes, demonstrating the practical application of blockchain in education. It also discusses potential challenges and strategies to overcome these in the context of blockchain integration into educational systems.

In essence, the fusion of blockchain technology with teacher professional development marks a promising step towards a more technologically enriched educational sphere. By presenting a blueprint for integrating blockchain into teacher PD, we hope to contribute to the ongoing dialogue on
harnessing technology to elevate educational practices and outcomes.

II. RELATED WORKS

A. Professional Development of Teachers

The essence of professional development (PD) lies in its potential to foster an enhanced understanding of pedagogical methods, curriculum intricacies, and student learning mechanisms among teachers. Studies emphasize the role of effective PD in augmenting teacher quality, thereby improving student outcomes [11]. However, the means of validating and documenting PD activities often lack transparency and efficiency, leading to skepticism about their credibility and consistency [12].

The role of professional development (PD) in the educational landscape is paramount. It embodies the continuous growth and learning teachers undertake to refine their pedagogical skills, adapt to evolving educational trends, and ultimately, enhance student learning outcomes [13]. Fig. 1 demonstrates GoSTEAM model that use block chain in STEAM education.

The common thread running through these diverse modalities is the objective of equipping teachers with the knowledge and skills necessary for effective classroom management and instruction. In this context, [14] proposed that effective PD must be ongoing, focused on the curriculum, and collaborative in nature. It should allow teachers to experiment, receive feedback, and reflect on their practice.

Moreover, the effectiveness of PD extends beyond individual teacher performance. Well-structured PD programs have been linked to fostering collaborative cultures within schools, thereby boosting overall institutional performance [15]. In contrast, ineffective PD programs characterized by disconnected, short-term training sessions have shown minimal impact on teachers’ practices and student outcomes.

However, the current systems of documenting and validating PD often involve cumbersome paperwork and are vulnerable to inconsistencies. In addition, these traditional methods often lack transparency, leading to questions about their reliability and the credibility of the PD efforts documented [16]. These challenges underscore the need for innovative solutions that can streamline the PD process, enhance its credibility, and encourage teachers to actively engage in their professional growth.

Recently, emerging technologies, such as blockchain, have been suggested as potential solutions to enhance the credibility, transparency, and efficiency of teacher PD. Blockchain technology, with its secure, transparent, and decentralized features, could serve as a means to authenticate PD efforts reliably and sustainably. This paper extends the exploration of this potential solution, presenting the design and development of a platform for teacher PD incorporating elements of blockchain technology.

B. Blockchain Technology: Beyond Cryptocurrency

The inception of blockchain technology, originally designed to support Bitcoin, a digital currency, has had far-reaching implications beyond the financial sector. Essentially, blockchain provides a decentralized, secure, and transparent environment for transactions and record-keeping [17]. Its application spans a multitude of sectors including healthcare, finance, and supply chain management, owing to its potent capabilities of data integrity and traceability [18].

Blockchain technology, born out of the desire for a decentralized and secure digital currency system, was first introduced by [19] in his seminal work on Bitcoin. However, the implications of this technology extend far beyond its initial application. The fundamental features of blockchain – decentralization, immutability, and transparency – make it a potent tool for diverse sectors beyond finance [20]. Fig. 2 demonstrates an example of using blockchain technology in cryptocurrency.

At its core, blockchain operates as a decentralized ledger of transactions distributed across a network of computers, known as nodes. Each transaction is recorded as a block and linked to the previous one, forming a chain. Once information is stored in a block, it becomes virtually impossible to modify or delete, thus ensuring the immutability of data. Furthermore, the absence of a central authority reduces the risk of single-point failures and increases system resilience [21].

While the initial hype around blockchain revolved around cryptocurrencies, researchers and industry professionals have recognized its potential in various other domains. For instance, in healthcare, blockchain can ensure secure patient data management, facilitating interoperability while maintaining data privacy [22]. In supply chain management, blockchain’s ability to provide traceability can enhance transparency and minimize fraud [23].

![Fig. 1. Applying blockchain in education.](image1)

![Fig. 2. An example of a lecturer’s block.](image2)
In the context of education, blockchain can have transformative implications. It can provide a secure, decentralized, and transparent system for academic credential verification, potentially eliminating the prevalent issues of credential fraud [24]. Furthermore, it could offer a reliable method of managing student records, giving students ownership of their educational data [25].

However, while the potential applications of blockchain are vast, so are the challenges. Technological understanding, data privacy, cost, and the issue of integration with existing systems are some of the critical considerations that must be addressed for successful implementation [26].

This paper explores the practical application of blockchain technology in an uncharted domain within education – teacher professional development. It delves into the design and development of a blockchain-based platform for documenting and verifying teachers' professional development activities, offering a novel perspective on the potential of blockchain technology in education.

C. Blockchain in Education

In the context of education, the potential of blockchain has been increasingly recognized, particularly in credential verification and student record management. The study [27] highlights the utility of blockchain in creating secure digital badges, asserting the credibility of academic achievements. Meanwhile, [28] demonstrates the successful use of blockchain in managing student records, providing a decentralized and immutable record of student progress and accomplishments. Yet, the realm of teacher PD remains a largely uncharted territory in blockchain-related research.

The potential of blockchain technology in the education sector is considerable and is beginning to draw substantial interest from researchers and practitioners. Its core characteristics of decentralization, security, and immutability offer unique solutions to some of the longstanding challenges in education, particularly in the domains of record management and credential verification [29]. Fig. 3 demonstrates example of application of blockchain technology in high education.

One of the most touted applications of blockchain in education is credential verification. With the rise of online learning and micro-credentials, traditional methods of credential verification have become increasingly cumbersome and susceptible to fraud. Blockchain technology can address these issues by providing a secure, immutable, and universally accessible system for issuing and verifying academic credentials [30]. By storing credentials on a blockchain, institutions can ensure their authenticity and longevity, while learners can retain lifelong ownership of their achievements, irrespective of institutional longevity.

Additionally, blockchain holds potential for student record management. Traditional systems for storing and transferring student records often involve significant bureaucratic processes and are vulnerable to loss or misplacement. A blockchain-based system could offer an efficient, secure, and transparent method for managing these records. Students could control and share their academic history, ranging from grades to extracurricular activities, providing a comprehensive, verified record of their learning journey [31].

Moreover, blockchain could revolutionize open and distance learning. By facilitating the secure exchange of digital assets, blockchain could provide the infrastructure necessary for implementing self-sovereign education, where learners have complete control over their educational journey [31].

However, despite these potential benefits, implementing blockchain in education is not without challenges. Concerns about data privacy, cost of implementation, interoperability with existing systems, and the need for digital literacy among stakeholders are considerable [32]. Addressing these challenges is a crucial step toward harnessing the potential of blockchain in education.

This paper contributes to this growing area of interest by exploring the application of blockchain technology in a relatively less explored domain within education: teacher professional development. Specifically, it describes the design and development of a blockchain-based platform for managing teacher professional development activities, thereby adding a new perspective to the discourse on the potential of blockchain in education.

D. Blockchain for Teacher Professional Development

The idea of employing blockchain in teacher PD stems from its potential to enhance transparency, security, and immutability of PD records. By decentralizing the control over PD documentation, teachers can gain direct ownership of their professional milestones and demonstrate verified skill acquisition [33]. The research [34] posits that blockchain can foster lifelong learning by making professional development a tangible, continuously evolving process. However, integrating blockchain into education systems is not without challenges. Issues related to technological understanding, data privacy, and cost must be addressed for successful implementation [35].

As the exploration of blockchain applications in education continues to unfold, an area worthy of attention is the integration of blockchain in the sphere of teacher professional development (PD). Traditionally, PD records have been plagued by issues of inefficiency, lack of transparency, and questionable credibility. Blockchain technology, with its inherent properties of security, immutability, and decentralization, can provide solutions to these challenges [36]. Fig. 4 demonstrates sample of using blockchain in professional development.

![Blockchain in education.](image)
The application of blockchain in teacher PD can redefine the way PD efforts are recorded, validated, and shared. A blockchain-based platform can serve as a decentralized ledger, documenting teachers’ PD activities in a secure and immutable manner. This not only streamlines the documentation process but also enhances the credibility of the records by ensuring their permanence and verifiability [37].

Furthermore, blockchain’s decentralization aspect gives teachers direct control and ownership over their PD records. This facilitates the transparent and seamless sharing of verified qualifications and skills, fostering a culture of trust within the educational community. Consequently, teachers are empowered to take an active role in their professional growth, and the value of their PD efforts is reinforced [38].

The use of blockchain in teacher PD also aligns with the shift towards lifelong learning. As continuous learning and upskilling become increasingly important in the fast-evolving educational landscape, blockchain can serve as a tool to document this lifelong learning journey securely and consistently [39].

However, the integration of blockchain technology in the educational sphere is not without hurdles. Potential obstacles include technological understanding among stakeholders, data privacy concerns, potential costs, and the challenge of integrating the new system with existing ones [40].

This paper addresses this gap in the literature by detailing the development of a blockchain-based platform for teacher PD. Through this exploration, the paper provides insights into the practical application of blockchain in the PD landscape, contributing to the wider dialogue on leveraging technology to enhance educational practices and outcomes.

This paper attempts to bridge the gap in the literature by presenting a platform for teacher PD that utilizes blockchain technology. By exploring the platform’s design and development processes, we offer insights into the practical application of blockchain in teacher PD, contributing to the discourse on the use of emerging technologies in enhancing education practices.

### III. MATERIALS AND METHODS

Even though blockchain technology emerged as a relatively new advancement in the realm of information technology, it has rapidly evolved into a globally influential technology with multi-industry applications. Numerous scholars posit that it is poised to significantly enhance the global economy in the coming decades [41]. Hwang illuminated the significance of decentralized ledgers, underpinned by blockchain technology, accentuating their potential to introduce considerable advantages for participants in the educational landscape. Specifically, Hwang underscored the opportunities for digital validation of individual and academic achievements [42].

Blockchain technology possesses several distinct attributes [43]:

1. **Decentralisation:** This characteristic manifests in three aspects - architectural, political, and logical decentralisation.
2. **Security:** It ensures the infallibility of records across the network, allowing any attempt at manipulation to be promptly identified.
3. **Anonymity:** It permits users to generate multiple addresses to safeguard their identities, thus maintaining transaction confidentiality.
4. **Verification and reliability:** This aspect amplifies the traceability and transparency of data embedded in the blockchain [43].

Casino et al. identified blockchain approach as an open-source tool with extensive cross-sector applications [44]. The distinct merit of blockchain resides in its transition from centralized data. Next research highlighted the blockchain as an emerging technology in education [45]. Consequently, they found that blockchain technology is predominantly employed for purposes such as issuing and validating academic transcripts and certificates, facilitating decentralized knowledge exchange and educational accomplishments of students, and assessing their knowledge, skills, and professional competencies [45]. Furthermore, blockchain could potentially stimulate motivation and enthusiasm for learning by maintaining a comprehensive, trustworthy record of a student's educational activities, encompassing both formal and informal learning environments. Additionally, the technology could inform the construction and evaluation of the educational process's quality by recording teacher and student activities and progress [46]. Drawing on the work of Sun et al., who described the approach, this proposition suggests a paradigm in which blockchain technology can significantly influence educational processes, providing insights into students' behaviors and aiding teachers in grading.

### IV. RESULTS

A pedagogical platform, incorporating elements of blockchain technologies, has been engineered for utilization in university-level educational processes. This platform encompasses an array of courses and resources that can be incorporated into learning trajectories [47].

Access to this platform is bifurcated into two tiers, delineating rights: educator and student. This segregation aims to create an optimal learning environment. Upon authorization,
students gain access to a wealth of available courses, their accumulated certificates, and a personalized page containing their relevant information. This educational portal, accessible via the Internet, is underpinned by blockchain concept exhibiting a graph architecture. It is structured using blocks of 1024 bytes each, each of which comprises the following sections:

1) Header: This section includes details about the types of blocks, timestamp, and hash of the blocks. All block varieties possess a header, the structure of which remains constant (refer to Table I).

<table>
<thead>
<tr>
<th>Field</th>
<th>Data_type</th>
<th>Data_size in bytes</th>
<th>Explanation</th>
</tr>
</thead>
<tbody>
<tr>
<td>Block_type</td>
<td>Int_32</td>
<td>4</td>
<td>Type of blocks</td>
</tr>
<tr>
<td>Timestamp</td>
<td>Int_32</td>
<td>4</td>
<td>Timestamp</td>
</tr>
<tr>
<td>Hash</td>
<td>Byte[32]</td>
<td>32</td>
<td>A hash underlying on the other blocks.</td>
</tr>
</tbody>
</table>

2) Parent block links: A block may have multiple parent block links, the quantity of which can differ among block types. This segment of the block is employed for block authentication and in generating the block hash.

3) Payload: This constitutes the data stored within the block. An illustration of the block architecture can be found in Table II.

<table>
<thead>
<tr>
<th>Field</th>
<th>Data_type</th>
<th>Data_size in bytes</th>
<th>Explanation</th>
</tr>
</thead>
<tbody>
<tr>
<td>Header</td>
<td>Block type (4 bytes)</td>
<td>Timestamp</td>
<td>Hash</td>
</tr>
<tr>
<td>Parent block's link</td>
<td>Link to 34th block</td>
<td>Link to 51th block</td>
<td>Link to 68th block</td>
</tr>
<tr>
<td>Payload</td>
<td>First name and last name</td>
<td>Identification number</td>
<td>Login</td>
</tr>
</tbody>
</table>

The fundamental algorithm for block authentication entails computing its hash, utilizing the SHA256 hash function. If the data encapsulated within a block undergoes modification, its hash function, when recalculated, will likewise alter, given that it contains the beneficial data stored in the block. In instances where the block hash is regenerated due to a new payload, all blocks referencing the initial block will become invalidated, as their hash values are computed based on the preceding block's hash. The assembly of the portal's primary blocks can be delineated as follows:

1) The LECTURER_BLOCK is created upon an educator's registration. The payload field encapsulates details about the educator, along with their public key. The initial 256 bytes are considered for the educator’s personnel data. The remaining space is assigned to the public key. Fig. 5 demonstrates the description of the LECTURER_BLOCK.

2) The STUDENT_BLOCK comes into existence when an educator registers to the course. The payload field houses the course title and its description. Here, coins represent earned points within the portal. Fig. 6 illustrates a working example of KURS_BLOCK in the proposed system.

3) The KURS_BLOCK (course block) is created when an educator introduces a course. The payload field accommodates the student's details and their public key.

4) The TASK_BLOCK is created when an educator introduces a task (block_type = 4). The payload field accommodates the maximum number of coins that can be earned from the task, the task file's signature by the educator (a hash consisting of the sum of the prev_block, the file hash, and the number of coins, encrypted with the educator's private key).

5) The TASK_COMPLETE_BLOCK is created when an educator evaluates a student's tasks.

```json
{
    type: 3,
    timestamp: 161789143678,
    hash: "f1c74b935b3c8f7ec232ba94b7b24ab67a647b3a5c6a18a1e2b8c6e5421eb2",
    prevBlockID: 16,
    title: "Увидеть в блокчейн технологии",
    desc: "На курсе "Увидеть в блокчейн технологии" Вы познакомитесь с основами работы блокчейн технологии. Данный курс поможет Вам построить концепции, лежащие в основе децентрализованных приложений, а также Вы познакомитесь с существующими реализациями блокчейн технологий",
    coin_count: 100,
    parents: [{ 0: 16 },
    blockID: 17,
    prev: {
        type: 2,
        timestamp: 161656256970,
        hash: "c6a56a2ed7d760be3f82dd58650f19ff3a4babadb059c9881bc6a5f97ec9",
        file: "Санктбур Айвер",
        len: 930423398226,
        login: "\n",
        pass_hash: "9f98e4055a23268ced3214ce76612371765657118844b1e802760d19d5a8",
        desc: "Магистр педагогических наук, PhD доктора кафедры "Информатика" Евразийского национального университета им Л.Н. Гумилева"
    }]
}
```

Fig. 5. An example of a lecturer's block.
6) The CERTIFICATE_BLOCK is created when a student exchanges earned coins for a course completion certificate. A block is considered valid only if the sum of all coins collected by the student from the task_complete fields of the current course is equal to or exceeds the number of coins specified in the task_block field. Fig. 7 demonstrates working principle of the CERTIFICATE_BLOCK.

In summary, an educational portal has been developed incorporating blockchain elements. Its utilization fosters student interest and enhances performance, thereby facilitating educational objectives. The diversification of the educational process engenders interactivity and student engagement. The platform further benefits from the verification of learning outcomes, facilitated by the blockchain technology's ability to provide simple and effective certification.

Whilst the expansive potential applications of blockchain technology remain largely speculative, this study indicates that its influence on the educational sector will increasingly become palpable in the foreseeable future. Some emergent implications are as follows:

1) Blockchain technology could considerably expedite the transition from traditional paper-based issuance of academic credentials. All forms of records, diplomas, and certificates granted by tertiary and other educational institutions, including those pertaining to qualification assignment or advanced training, can be securely and indelibly stored in a decentralised database through blockchain technology.

2) Blockchain technology facilitates automatic validation of academic records, diplomas, or certificates directly via the technology itself, obviating the need to liaise with the issuing educational institution. This development could greatly simplify, if not eradicate, the need for incessant credential verification and attenuate bureaucratic processes within educational institutions. This autonomous certification issuance and validation capability could also be utilised in various other educational contexts. Furthermore, blockchain technology's application extends to copyright and intellectual property management, allowing for the tracking of initial publications and citation tags, without the requirement for a centralised authority to maintain these databases.

3) The potential reduction of data management costs for educational institutions is another prospect offered by blockchain technology. This could be actualised through the creation of decentralised structures in data management, enabling users to effortlessly control and share their data as needed.

4) The growing prevalence of cryptocurrencies, built upon blockchain technologies, could be utilised to facilitate payments and bolster learning motivation by accruing coins upon course completion in certain educational institutions and platforms. The ability to generate proprietary cryptocurrencies could also signify the future usage of blockchain in educational funding, potentially through grants or vouchers, across numerous countries.
V. DISCUSSION

The integration of blockchain technology into the educational sector, as underscored by this study, bears the potential to bring about revolutionary changes in the processes of learning, teaching, and administration. It is anticipated that this technological incorporation can assist in developing a more inclusive, transparent, and efficient educational system.

The deployment of blockchain technology in the proposed platform for teacher professional development enables a decentralised and verifiable record system that is tamper-proof and fosters trust. This inherently aligns with the requirement of ensuring transparency and integrity in teacher professional development programs. Additionally, the ability to monitor and verify teacher development through a decentralised ledger provides a more egalitarian approach to professional learning, given that the power of validation and accreditation does not reside with a singular authority.

However, the implementation of such a technologically complex system within an educational setting is not devoid of challenges. While the literature has presented the capabilities of blockchain technology extensively, it remains a nascent field within the education sector [48]. As such, substantial development, understanding, and research are required before it can be fully embraced by educational institutions.

Issues pertaining to the digital divide and access to technology are critical considerations. Not all educators and students may have equal access to the necessary devices or sufficient internet connectivity to engage effectively with a blockchain-based system. Therefore, while designing such platforms, it is essential to ensure that it can be accessed through multiple forms of devices, including mobile technology, to mitigate potential accessibility issues [49].

Furthermore, the platform's usability is a critical determinant of its success. Even the most innovative technologies will fail to gain traction if they are not user-friendly and intuitively designed. As such, the role of human-computer interaction principles is crucial in the design and development of such a platform to ensure optimal user experience.

The role of privacy and security within the blockchain also necessitates careful attention. As an immutable record, it is critical to ensure that the information stored within the blockchain respects user privacy and adheres to local and international regulations. Additionally, while the blockchain's decentralised nature can enhance security, it also means that breaches, when they do occur, can have far-reaching consequences [50].

In terms of curriculum development and pedagogy, this paper illustrates that blockchain can serve as a powerful tool to motivate teachers towards continuous professional development by providing them with tangible, verifiable records of their growth [51]. However, the integration of this technology into education is not a panacea. It should be supplemented with other research-proven teaching strategies and ongoing support to ensure it serves its purpose effectively.

The potential integration of blockchain into educational administrative processes is a promising prospect, but it will require a significant cultural shift within institutions. Change management strategies will be required to help stakeholders understand the benefits and potentials of this technology and gain their buy-in [52].
In conclusion, the advent of blockchain technology presents an exciting frontier in the world of education, particularly in teacher professional development. It offers the possibility of a more transparent, equitable, and efficient system for recording and verifying professional learning. However, the successful implementation of this technology requires careful attention to accessibility, usability, privacy, security, pedagogical integration, and change management strategies. Further research is necessary to explore these issues in greater depth and to pilot and refine this technology's use within real-world educational contexts.

VI. CONCLUSION

In conclusion, this investigation has underscored the transformative potential of blockchain technology in reimagining the realms of education and teacher professional development. The distinct features of blockchain such as decentralization, verifiability, security, and transparency offer the possibility of constructing an innovative platform for professional learning. This platform, characterised by autonomous control, seamless tracking of progress, and the preservation of records, can foster an environment of trust and integrity for the stakeholders. Additionally, the potential of blockchain in the context of digital accreditation signifies a paradigm shift towards more efficient and tamper-proof validation processes. However, the successful integration of this technology necessitates careful attention to issues of accessibility, usability, data privacy, and security. Simultaneously, pedagogical considerations and change management strategies are crucial to ensure a holistic and efficient adoption within the educational sector. It is evident that while blockchain technology has the potential to revolutionise the education sector, more empirical research is needed to further investigate its applicability and implications. This study thus acts as a stepping-stone towards understanding and leveraging the profound potential of blockchain in reshaping the landscape of education and teacher professional development.

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Predicting Maintenance Labor Productivity in Electricity Industry using Machine Learning: A Case Study and Evaluation

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Abstract—Predicting maintenance labor productivity is crucial for effective planning and decision-making in the electricity industry. This paper aims at predicting maintenance labor productivity using various machine learning methods, utilizing a real-world case study from the electricity industry. Additionally, the study evaluates the performance of the employed machine learning methods. To meet this objective, 1750 productivity measures have been used to train (80%) and test (20%) prediction models using Artificial Neural Networks, Support Vector Machines, Random Forest, and Multiple Linear Regression methods. The models' performance was evaluated based on the mean squared error, mean absolute percentage error, and testing time. The results indicated that the Artificial Neural Networks model - specifically, a feedforward network with a backpropagation algorithm - outperformed the other models (Multiple Linear Regression, Support Vector Machines, Random Forest). These results highlight the effectiveness of machine learning, particularly the Artificial Neural Networks prediction model, as an invaluable tool for decision-makers in the electricity industry, aiding in more effective maintenance planning and potential productivity improvement.

Keywords—Productivity; machine learning; maintenance; prediction; ANN

I. INTRODUCTION

The electricity industry plays a vital role in the modern world, underpinning a wide array of economic activities and societal functions. The demand for electricity has increased enormously worldwide, with an annual average of 2.6 percent from 2010 to 2021 [1]. As the demand for reliable and uninterrupted electric power supply grows, the efficient maintenance of networks has become increasingly crucial in the electricity industry [2]. One critical component of efficient maintenance is the productivity of the maintenance labor force. Maintenance labor productivity is a measure of the effective use of resources while performing maintenance activities, usually expressed as the ratio of output to input [3]. It can significantly impact the availability, efficiency, and reliability of electric power networks. High productivity can lead to reduced breakdowns, improved performance, lower operational costs, and the avoidance of expensive blackouts.

However, predicting maintenance labor productivity is a complex task. Numerous factors, such as type of equipment, labor skills and experience, supervisor competency, and even external factors like weather conditions, can influence productivity. Accurate prediction of productivity allows for effective planning, ultimately leading to better maintenance outcomes and improved service reliability. As it will be shown in the following section, and to the best of the authors’ knowledge, no study was conducted into predicting maintenance labor productivity using machine learning methods, especially in the electricity industry.

This complexity, alongside the pivotal role of productivity in the electricity industry, motivates the central research question of this study; how effective are machine learning models at predicting maintenance labor productivity? To address this question, the objectives of this study are to develop a various machine learning models, including artificial neural networks (ANN), support vector machines (SVM), random forests (RF), and multiple linear regression (MLR) to predict maintenance labor productivity with a focus on electricity industry and to evaluate the performance of these machine learning models. For this purpose, the study will use a combination of qualitative and quantitative research methods, employing a real-world case study.

The structure of this paper is organized as follows: Section II provides insight into productivity and reviews the research on the application of machine learning in predicting labor productivity, while Section III describes the research methodology used in this study. Section IV focuses on the data collection, and data preprocessing. Section V illustrates the models development. The results are presented and discussed in Section VI. Finally, in Section VII the study has been concluded.

II. LITERATURE REVIEW

A. Labor Productivity Measurement and Influencing Factors

Productivity serves as the ultimate engine of growth in the global economy. Various methods for measuring productivity are presented in the literature. The most prevalent method measures productivity as a ratio of work accomplished or units produced per man-hour [4] [5]. The inverse is also commonly used, which measures productivity as a ratio of man-hours per work accomplished or unit produced [6].

Another method measures productivity as the ratio of earned hours to actual hours [7] [8] [9] [10]. The concept “earned hours” is popular in the United States of America (USA), refers to establish a base or a norm for each activity.
According to the American Association of Cost Engineers, a norm is defined as the number of man-hours required to complete a defined activity under a specific set of stated conditions or qualifications [11]. Thus, a number of earned hours is associated with a norm and each unit of work accomplished.

Extensive literature has attempted to investigate and identify the factors influencing labor productivity across various industries. The most influential factors include labor skills and experience [12] [13] [14] [15] [16] [17] [18] [19] [20] [21] [22] [23] [24], labor motivation [25], supervisor competency [12] [16], and weather conditions such as temperature and humidity [15] [26].

**B. Application of Machine Learning in Prediction of Labor Productivity**

Machine learning, a subset of artificial intelligence, provides machines with the ability to learn and improve from experience. Moreover, it has the capability to handle large volumes of data, learn from this data, and make accurate predictions or decisions without being explicitly programmed. The application of machine learning to predict labor productivity is an emerging area of research that has demonstrated promising results across various industrial sectors.

The earliest study found describing a real application of machine learning to predict labor productivity proposed an ANN model for predicting labor productivity in concrete formwork activity [27]. Subsequent studies used ANN to predict labor productivity for concrete pouring, formwork, and concrete finishing, and pipe installation, respectively [28] [29]. In 2006, a study proposed a framework to predict the labor productivity rates of forms assembly, steel fixing, and concrete pouring activities using ANNs [30]. Oral and Oral [31] developed ANN to predict crew productivity for ready mixed concrete, formwork, and reinforcement activities. The authors indicated that ANNs have proved to make productivity predictions significantly better than statistical regression methods, which also agreed by other research works such as [32] [33].

Similarly, ANN was used by [34] to predict the required man-hours for the formwork activity of reinforced concrete framed building projects. The developed model produced results reasonably close to actual field measurements, which was also indicated by [35], who found that the developed ANN has the ability to predict the labor productivity of marble finishing works for floors with a 90.9 percent of accuracy. Furthermore, Heravi and Eslamdoost [36] utilized ANN to predict labor productivity in the concrete foundations work of gas, steam, and combined cycle power plant construction projects. The study illustrated a structured method for developing the ANN model of labor productivity as well as the training process of neural network. Moreover, a study conducted by [37] applied ANN to predict construction labor productivity. The results showed that the ANN adequately converged and have noticeable and reasonable generalizing capabilities, which also is consistent with [30].

A study by [38] applied various ANNs to predict labor productivity norms for the formwork activity of two high-rise buildings. Moreover, a comparison of each ANN model’s performance was conducted to identify the best model. The collected data set was utilized in the research work of [39], where the authors proposed a novel approach for predicting labor productivity using ANN. The developed network showed a good performance from the point of view of generalization.

Mlybari [40] did an investigation aimed to demonstrate the use of various machine learning techniques to predict the labor productivity rates of concrete construction activities, including formwork, steel fixing, and concrete pouring and finishing. The result showed that the developed ANN outperformed the other techniques such as SVM and could be useful to predict labor productivity. Another study found that the performance of RF model yielded better results in predicting labor productivity compared to the ANN model [41]. Conversely, the study by [23] suggested that the performance of SVM is better compared to RF in predicting construction labor. Another study used machine learning–based approach to analyze and predict construction task productivity [42].

Despite these applications and to the best of the authors’ knowledge, no previous research has applied machine learning to predict maintenance labor productivity in the electricity industry. Hence, this study fills this gap in the literature by applying and evaluating different machine learning models, including ANN, SVM, RF, and MLR, for the prediction of maintenance labor productivity in the electricity industry.

**III. RESEARCH METHOD**

The primary purpose of this study is to explore the application of machine learning in prediction maintenance labor productivity with a focus on electricity industry. Fig. 1 illustrates the flowchart of the research methodology. The initial step involves the identification of inputs and outputs for the machine learning models. The selection of model inputs will be based on a literature review and expert opinion, while the model output is defined as the percentage of labor productivity, as shown in (1) [10].

\[
\text{Productivity (\%) = (Earned hours / Actual hours) \times 100}
\]  

(1)

Following the identification of the appropriate inputs and output for the machine learning models, the next step is data acquisition. The data will be collected from the selected company. This collected data will then be preprocessed as needed through several steps, which include data cleaning, data encoding, and data normalization. Subsequently, the preprocessed data will be randomly partitioned into training and testing sets.

The MATLAB 2023a software will be utilized for the development, training and testing of the four machine learning models, specifically ANN, SVM, RF, and MLR. Each model’s performance will be evaluated using a set of defined criteria, including the Mean Squared Error (MSE), Mean Absolute Percentage Error (MAPE), and the computational time required for testing. MSE and MAPE will be calculated using (2) and (3), respectively [43]. Ultimately, the model that exhibits the best performance, in accordance with the evaluation criteria,
will be selected. This model will form the basis of the predictive tool that this research aims to develop.

\[
MSE = \frac{1}{n} \sum_{i=1}^{n} (Y_{i \text{actual}} - Y_{i \text{predict}})^2 \tag{2}
\]

\[
MAPE = \frac{100}{n} \sum_{i=1}^{n} \left( \frac{|Y_{i \text{actual}} - Y_{i \text{predict}}|}{Y_{i \text{actual}}} \right) \tag{3}
\]

Where ‘n’ represents the number of data points in the dataset used for prediction, ‘Y_{i \text{actual}}’ denotes the actual productivity value of the i th instance in the dataset, and ‘Y_{i \text{predict}}’ refers to the predicted productivity value i th instance in the dataset.

IV. DATA COLLECTION AND PREPROCESSING

A. Data Collection

The dataset utilized in this research was obtained from a large electricity company that is responsible for supplying power. A comprehensive set of data was collected, incorporating all the influencing factors identified from the literature review and expert opinion. The dataset contains records of conducted preventative maintenance tasks for substations and includes inputs variables such as type of equipment, level of labor skill, labor health condition, level of safety measures, labor experience, level of labor motivation or commitment, level of supervisor competency. Because the main dataset did not include all the identified factors, additional data such as temperature and humidity were obtained from the National Center of Meteorology and synchronized with the date and time of the task.

The output parameter, percentage of labor productivity, was calculated by dividing the earned hours by the actual hours. The data collected spans a period of twelve months, offering a substantial volume of information for the models to learn from and making the prediction robust and applicable across different periods.

B. Data Preprocessing

Data preprocessing is a crucial phase in addressing a real-world problem using machine learning and plays a significant role in obtaining promising results. This phase involves several steps:

1) Data cleansing or cleaning: This process identifies and corrects, or removes, inaccuracies, discrepancies, and inconsistencies within datasets. A comprehensive data cleaning was implemented. Certain records within the dataset were excluded due to their lack of labor-related information or because they were duplicates. Furthermore, the units of measure for both earned and actual time were standardized to exclusively use minutes. This uniformity helps to prevent potential discrepancies or misunderstandings that could arise from inconsistent units.

2) Data encoding: This involves transforming categorical variables into a numeric format. The label encoding technique was employed to encode categorical data into numerical values as machine learning can only process numerical data. Table I provides an overview of the data description, while Table II summarizes the descriptive statistics of the collected data.

3) Data partitioning: The total available data is 1750 instances that randomly divided into two sets following an 80:20 ratio; 80% for training (1400 instances) and 20% for testing (350 instances). The training dataset will be used to train the model, and the testing dataset will assess the model's performance on unseen data.

4) Data normalization: This step is necessary and essential to enhance the performance of machine learning [44]. Min-max normalization, one of the most commonly employed methods, was utilized in this study to normalize data within the range of 0 to 1.
### TABLE I. DATA DESCRIPTION

<table>
<thead>
<tr>
<th>No.</th>
<th>Input Variables</th>
<th>Descriptions</th>
<th>Type</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>Type of equipment</td>
<td>The type of equipment in which the task will be performed with.</td>
<td>Categorical</td>
<td>1: Aspiration Smoke Detection</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>2: Fire Alarm System</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>3: Fire Fighting System</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>4: Gas Extinguishing System</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>5: Gas Insulated Switchgear</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>6: Transformer</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>7: Water Mist System</td>
</tr>
<tr>
<td>2</td>
<td>Skill level</td>
<td>The technical skill level of labor who perform the task.</td>
<td>Categorical</td>
<td>1: Novice</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>2: Intermediate</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>3: Competent</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>4: Proficient</td>
</tr>
<tr>
<td>3</td>
<td>Health condition</td>
<td>The overall health condition of labor who perform the task.</td>
<td>Categorical</td>
<td>1: Poor</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>2: Fair</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>3: Good</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>4: Excellent</td>
</tr>
<tr>
<td>4</td>
<td>Safety measures</td>
<td>The level of safety measures the labor required to be taken while perform the task.</td>
<td>Categorical</td>
<td>1: Basic</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>2: Moderate</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>3: High-Level</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>4: Extreme</td>
</tr>
<tr>
<td>5</td>
<td>Temperature</td>
<td>The average temperature of a day that the task perform in.</td>
<td>Numerical</td>
<td>Celsius degree (°C)</td>
</tr>
<tr>
<td>6</td>
<td>Labor experience</td>
<td>Number of years of experience the labor have.</td>
<td>Numerical</td>
<td>Years</td>
</tr>
<tr>
<td>7</td>
<td>Level of supervisor competency</td>
<td>The competency level of supervisor.</td>
<td>Categorical</td>
<td>1: Novice</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>2: Intermediate</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>3: Competent</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>4: Proficient</td>
</tr>
<tr>
<td>8</td>
<td>Level of motivation or commitment</td>
<td>The level of labor motivation or commitment who perform the task.</td>
<td>Categorical</td>
<td>1: Disengaged</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>2: Low</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>3: Medium</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>4: High</td>
</tr>
<tr>
<td>9</td>
<td>Humidity</td>
<td>The average humidity of a day that the task perform in.</td>
<td>Numerical</td>
<td>Percentage</td>
</tr>
</tbody>
</table>

### TABLE II. DESCRIPTIVE STATISTICS OF THE COLLECTED DATA

<table>
<thead>
<tr>
<th>Variable</th>
<th>Mean</th>
<th>SE Mean</th>
<th>Std. Dev</th>
<th>Min</th>
<th>Median</th>
<th>Max</th>
</tr>
</thead>
<tbody>
<tr>
<td>Type of equipment</td>
<td>4.85</td>
<td>0.04</td>
<td>1.50</td>
<td>1</td>
<td>5</td>
<td>7</td>
</tr>
<tr>
<td>Skill level</td>
<td>3.60</td>
<td>0.01</td>
<td>0.50</td>
<td>2</td>
<td>4</td>
<td>4</td>
</tr>
<tr>
<td>Health condition</td>
<td>3.68</td>
<td>0.01</td>
<td>0.57</td>
<td>2</td>
<td>4</td>
<td>4</td>
</tr>
<tr>
<td>Safety measures</td>
<td>3.05</td>
<td>0.02</td>
<td>0.81</td>
<td>2</td>
<td>3</td>
<td>4</td>
</tr>
<tr>
<td>Temperature</td>
<td>29.62</td>
<td>0.15</td>
<td>6.30</td>
<td>18.9</td>
<td>29.9</td>
<td>40.1</td>
</tr>
<tr>
<td>Labor experience</td>
<td>12.46</td>
<td>0.13</td>
<td>5.27</td>
<td>3</td>
<td>12</td>
<td>28</td>
</tr>
<tr>
<td>Level of supervisor competency</td>
<td>3.21</td>
<td>0.02</td>
<td>0.64</td>
<td>2</td>
<td>3</td>
<td>4</td>
</tr>
<tr>
<td>Level of motivation or commitment</td>
<td>3.69</td>
<td>0.01</td>
<td>0.54</td>
<td>2</td>
<td>4</td>
<td>4</td>
</tr>
<tr>
<td>Humidity</td>
<td>0.45</td>
<td>0.00</td>
<td>0.20</td>
<td>0.10</td>
<td>0.44</td>
<td>1</td>
</tr>
<tr>
<td>Labor productivity</td>
<td>1.01</td>
<td>0.00</td>
<td>0.17</td>
<td>0.69</td>
<td>1.01</td>
<td>1.44</td>
</tr>
</tbody>
</table>
V. MACHINE LEARNING MODELS DEVELOPMENT

A. Artificial Neural Networks

One of the main methods in machine learning is ANNs. Moreover, it is an information processing paradigm biologically inspired and designed to simulate the way in which the human brain processes information [45]. ANN can be defined as structures comprised of densely interconnected adaptive simple processing elements called artificial neurons or nodes that are capable of performing massively parallel computations for data processing and knowledge representation [46] [47] [48]. The ANN is developed and derived to have a function similar to the human brain by memorizing and learning various tasks and behaving accordingly [49]. Once ANN is trained, it is able to recognize similarities when presented with a new input pattern, resulting in a predicted output pattern. ANNs are now recognized worldwide as the most effective and appropriate machine learning method for prediction [50].

In this study, an ANN model was used for predicting labor productivity. The configuration of the ANN model consisted of one input layer, one hidden layer, and one output layer. The input layer contained nine neurons, equivalent to the number of input variables. The hidden layer, consisting of 15 neurons, employed a sigmoid activation function (logsig), while the output layer used a linear activation function (purelin). The model was trained using the backpropagation algorithm, which is the most common learning algorithm in neural networks. The structure of ANN model is illustrated in Fig. 2.

B. Support Vector Machines

SVM is a supervised machine learning method, often used in classification problems but also capable of performing regression. SVM operate by mapping input data into a high-dimensional feature space, and then finding an optimal hyperplane that maximizes the margin between different classes in the feature space, hence making it a suitable choice for a range of predictive modeling problems. In this study, an SVM model was developed for predicting labor productivity.

C. Random Forests

RF is a robust and effective machine learning methods for prediction because of their good performance, scalability, and ease of use [51]. It operates as an ensemble learning method by constructing a multitude of decision trees and producing an output that is the average prediction of the individual trees. Given its ability to mitigate overfitting while maintaining high precision, RF has found extensive application in various predictive modeling tasks. In this study, an RF model was employed to predict labor productivity. The model was configured with 200 decision trees, a parameter that was chosen to balance between the model’s complexity and its ability to learn the underlying patterns in the data.

D. Multiple Linear Regression

MLR is a statistical technique that uses several explanatory variables to predict the outcome of a response variable. The goal of MLR is to model the relationship between the explanatory and response variables. The development and application of MLR in this study offers a traditional statistical approach to the problem of predicting labor productivity, providing a contrast and a basis for comparison with the more sophisticated machine learning models.

VI. RESULTS AND DISCUSSION

From the results presented in Table III, it is evident that the ANN exhibits the best overall performance in terms of a balance between low error rates and efficient testing times.

<table>
<thead>
<tr>
<th>Model</th>
<th>MSE</th>
<th>MAPE</th>
<th>Testing Time (Second)</th>
</tr>
</thead>
<tbody>
<tr>
<td>ANN</td>
<td>0.0250</td>
<td>12.494</td>
<td>0.012969</td>
</tr>
<tr>
<td>SVM</td>
<td>0.0595</td>
<td>19.825</td>
<td>67.1444</td>
</tr>
<tr>
<td>RF</td>
<td>0.0330</td>
<td>12.027</td>
<td>1.1359</td>
</tr>
<tr>
<td>MLR</td>
<td>0.0286</td>
<td>14.136</td>
<td>0.041887</td>
</tr>
</tbody>
</table>

In assessing prediction performance, measured by MSE, the ANN model outperforms the SVM, RF, and MLR models with an MSE of 0.0250. The MLR follows closely with an MSE of 0.0286, and RF comes in third with an MSE of 0.0330. The SVM model shows the least performance with the highest MSE of 0.0595.

However, in terms of MAPE, the RF model excels with the lowest MAPE of 12.027. The ANN model trails closely behind with a MAPE of 12.494. However, MLR and SVM models show higher MAPEs of 14.136 and 19.825, respectively, indicating a greater margin of error in their predictions. When considering the testing time, the ANN model outperforms the rest, requiring only 0.012969 seconds. The MLR model is slightly slower with a testing time of 0.041887 seconds, while RF takes notably longer at 1.1359 seconds. The SVM model, on the other hand, requires the most extended testing duration of 67.1444 seconds.

These results accentuate that the ANN model outperform the traditional statistical method in predicting labor productivity, aligning with the findings from studies [32], [37], and [38]. Furthermore, these results corroborate the conclusion.
of study [40] that set the ANN model as superior to the SVM model in labor productivity prediction. On the other hand, these findings conflict with another study [41], where the RF model demonstrated better results in predicting labor productivity compared to the ANN model. Further, the current study's results deviate from those of study [23], which found that the SVM model outperformed the RF model in labor productivity prediction. It's important to note that the performance evaluation in study [41] was based on the Mean Absolute Error and Root Mean Square Error metrics, while study [23] employed Percentage of Correct, Heidke Skill Score, Probability of Detection, False Alarm Ratio, and Peirce Skill Score metrics, respectively. Despite the limited number of studies comparing the performance of machine learning methods in predicting labor productivity, these results underscore the effectiveness of machine learning, and specifically, the ANN model, in labor productivity prediction.

VII. CONCLUSION

This study aimed to investigate the application of various machine learning techniques to predict labor productivity in the electricity industry. Four popular machine learning models, namely ANN, SVM, RF, and MLR, were developed and evaluated. The performance of the models was evaluated based on the MSE, MAPE, and testing time.

The results indicated that the ANN model performed the best overall, showcasing a balanced performance between low error rates and efficient testing times. This finding provides a clear answer to the study research question, showing that machine learning models, particularly ANNs, can effectively predict maintenance labor productivity in the electricity industry. However, it is important to note the limitations of this study. While the ANN model showed promising results, the findings may not be generalizable to all sectors due to the unique characteristics and variables of the electricity industry. Furthermore, the models developed in this study could serve as valuable tools for managers and decision-makers in the industry, allowing them to make informed decisions about labor management based on accurate productivity predictions. This contribution is significant as it provides a practical application of machine learning techniques in a real-world industry setting.

Future research should consider applying these models in other sectors and exploring other machine learning techniques for labor productivity prediction. There are still unanswered questions regarding the optimal machine learning techniques for different sectors, and how these models can be improved to increase their predictive accuracy and efficiency. Additionally, future research should also aim to address the limitations of this study, such as the potential lack of generalizability, by conducting similar studies in various industry settings.

ACKNOWLEDGMENT

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Web Phishing Classification Model using Artificial Neural Network and Deep Learning Neural Network

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Abstract—Phishing is an online crime in which a cybercriminal tries to persuade internet users to reveal important and sensitive personal information, such as bank account details, usernames, passwords, and social security numbers, to the phisher, usually for mean purposes. The target victim of the fraud suffers a financial loss, as well as the loss of personal information and reputation. Therefore, it is essential to identify an effective approach for phishing website classification. Machine learning approaches have been applied in the classification of phishing websites in recent years. The objectives of this research are to classify phishing websites using artificial neural network (ANN) and convolutional neural network (CNN) and then compare the results of the models. This study uses a phishing website dataset collected from the machine learning database, University of California, Irvine (UCI). There were nine input attributes and three output classes that represent types of websites either legitimate, suspicious, or phishing. The data was split into 70% and 30% for training and testing purposes, respectively. The results indicate that the modified ANN with Rectified Linear Unit (ReLU) activation function model outperforms other models by achieving the least average of root mean square error (RMSE) value for testing which is 0.2703, while the CNN model produced the least average RMSE for training which is 0.2631. ANN with Sigmoid activation function model obtained the highest average RMSE of 0.3516 for training and 0.3585 for testing.

Keywords—Phishing website; classification; artificial neural network; convolutional neural network; machine learning

I. INTRODUCTION

Phishing is a form of social engineering attack that is regularly used to get individuals to provide confidential data, like credit card information and login credentials. This happens when a phisher pretends to be a reliable organization to get a targeted victim to open a text message or an email. Then, the victim is tricked into clicking a malicious link that leads to a bogus website where private and sensitive information such as account numbers and Internet banking passwords can be obtained. A cyber-attack might have disastrous consequences, such as unlawful transactions, money theft, or identity theft for users [1].

A non-profit association known as the Anti-Phishing Working Group (APWG) investigates phishing assaults that have been informed by its fellow corporations, including MarkMonitor, iThreat Cyber Group, Forcepoint, Internet Identity (IID), and Panda Security. It evaluates the attacks and releases quarterly and half-yearly reports regularly. Additionally, it offers statistics data on phishing attacks and malicious domains that are active worldwide. The most recent phising activity trends report states that in the third quarter of 2022, APWG detected a total of 1,270,883 phishing attacks. This quarter's phishing activity was the worst that the APWG has ever recorded. 23.2% of all phishing attacks targeted the financial sector. In the third quarter, email-based scams involving advance fee payments grew by 1,000% [2]. Globally, millions of dollars have been lost as a result of these attacks, which had a severe effect on numerous renowned organizations worldwide. For protecting personal and business information as well as financial assets, addressing the problem of phishing website classification is getting more important. Hence, classifying and minimizing the impact of phishing attacks are the motivations to conduct this study.

A common countermeasure of phishing websites involves checking the websites against blacklists of known phishing websites, which are traditionally compiled, based on manual verification but this method is inefficient as it usually fails to discover all phishing sites because a recently created forged website takes a significant time before it can be added to the list. Moreover, there is no robust blacklist that will guarantee a perfect up-to-date database as nowadays, it has become easier to register new domain names. As the Internet scale grows, advanced website classification is increasingly important to provide timely protection to end users. Thus, this study aims to classify phishing websites using artificial neural network and convolutional neural network and then compare the models' classification performance. Biological neural networks are the foundation of the artificial neural network (ANN). It is made up of fundamental building blocks called neurons, which multiply weight with a real value and later put the result via a nonlinear activation function. By constructing multiple layers of neurons, where each of them takes the input variables and then passes the output to the following layers, the network can learn complicated functions. Supposedly, an ANN with adequate computational power could learn the shape of any function [3]. Neural networks have a high tolerance for noisy data, which is one of the advantages of using them. Although it is simple and convenient to use, handling huge dimensional data requires some understanding of parameter settings. It is simple to interpret the outcomes of this method. Another advantage of employing ANN is it can generate probability-based output, which ensures that the model will be more accurate as larger volumes of input data are provided [4].

A deep learning network architecture known as a convolutional neural network (CNN) automatically extracts features from the data rather than having to manually extract them first. The primary benefit of CNN over its predecessors is that it automatically recognizes the significant attributes
The model was trained using a gradient descent training algorithm with a learning rate set to 0.01 and momentum set to 0.1 to ensure slow convergence without skipping any possible best solution or overfitting. Finally, the model was evaluated using RMSE to measure the accuracy of the proposed method for LSTM, DNN, and CNN is 99.57%, 99.52%, and 99.43% respectively. Besides, Yerima & Alzaylaee [11] employed convolutional neural networks (CNN) for high-accuracy classification to differentiate fraudulent websites from legitimate websites. The result showed that the CNN technique, which achieved 97.3% accuracy with an F1-score of 97.6%, outperformed conventional machine learning classifiers evaluated on the same dataset. Another study by Geyik et al. [12] adopted decision tree, naive Bayes, random forest, and logistic regression algorithms. According to the findings, the Random Forest method outperforms the others, with an 83% of accuracy rate.

Afterward, Nadar et al. [13] proposed a hybrid Stacking model. Then, it was compared with Naïve Bayes, Random Forest, and XGBoost approaches. The research outcome showed that the proposed Stacking Classifier outperforms other methods with results of 85.6% of accuracy. A study by [6] applied an artificial neural network and the findings show that the ANN model with (9-5-1) architecture design gave the best result by obtaining the lowest difference between average training and testing MSE, which is 0.04745. Later, research by [14] focused on using multilayer perceptron (MLP), a type of neural network concept, to classify phishing websites. MLP is compared with other machine learning approaches like logistic regression, random forest, and support vector machine (SVM) for result evaluation, and it was found that MLP achieved the highest accuracy of 96.80%.

Machine learning approaches were also being employed in other fields such as healthcare, for dengue prediction [15], diabetes classification [16], and depression prediction [17]. Machine learning was also applied in agriculture for classifying broccoli leaf disease [18] and types of coral reef fish [19]. It was also used for classifying types of traffic violations [20], carbon monoxide concentration prediction [21], and air traffic communication system [22]. Deep learning approaches were also utilized in sound recognition [23], prognosis of Covid-19 [24], and glioma classification [25]. After reviewing previous studies in this field, it is evident that the classification accuracy should be enhanced. It is crucial to decrease the error rate to attain a high level of classification accuracy leading to producing more reliable and robust classification models.

### III. Methodology

The modelling of website phishing classification was divided into three main parts which represent three types of classification models. All classification processes followed the same methodology which started with pre-processing of the dataset where the phishing dataset [26] was normalized according to the selected model. There were nine input features and three output classes: Legitimate, Suspicious, and Phishy. It was split into a training set and a testing set with a ratio of 70:30. Next, the model was designed based on a selected algorithm to model the classifier. The model was trained using a gradient descent training algorithm with a learning rate set to 0.01 and momentum set to 0.1 to ensure slow convergence without skipping any possible best solution or overfitting. Finally, the model was evaluated using RMSE to measure the accuracy of the classification models.
model output and targeted output was minimized and the model accurately classified the phishing classes. The flow of the classification process using ANN is represented in Fig. 1.

![Diagram of ANN classification methodology](image)

**Fig. 1.** ANN classification methodology.

### A. Artificial Neural Network Model with Sigmoid Activation Function

The phishing website dataset was collected from the UCI repository with a total number of instances is 1353. Each instance has nine attributes and must be categorized into one of the three classes, the first class is Legitimate, the second class is Suspicious, and the third class is Phishy. Numbers had been used to replace these categories, with 1 denoting legitimate, 0 for suspicious, and -1 for phishing. There are 103 suspicious, 548 legitimate, and 702 phishing websites from the total number of 1353 websites. The data were saved using a comma-separated values (.csv) format. When a website is classified as suspicious, it might be either phishy or legitimate, implying that it has both phisy and legit attributes. The attributes in this dataset are including Server Form Handler (SFH), using a pop-up window, Secure Socket Layer (SSL) final state, request URL, anchor’s URL, traffic of a website, length of URL, domain’s age, and using an IP address. Out of 1353 instances, 70% of the instances were given for the training set which was 948 instances and the remaining 30% were given for the testing set which was 405 instances. Usually, in data pre-processing, normalization is used to scale the values from different ranges to a common range such as -1 to 1 or 0 to 1. Data must be scaled into the range used by the input neurons in the neural network. The dataset was scaled from -1 to 1 into 0 to 1 for the research purpose.

In a neural network, there are three layers, which are an input layer, a hidden layer, and an output layer. The model's input layer receives all of the inputs to be fed into the model. Then, these inputs are transmitted to the hidden layers. Each input neuron should represent some independent variable that has an impact on the neural network’s output. There is a layer called the hidden layer which is located between the input layer and the output layer that is made up of a group of neurons that have activation functions applied to them. Its responsibility is to process the inputs obtained from the input layer. This layer is in charge to extract the required features from the input data. A neural network could contain multiple hidden layers. The output layer of ANN collects and passes the data in a way that it was designed to do. At the output layer, the processed data is made available. The neural network design used in this study was (9-10-10-1) which means 9 input neurons, two layers of hidden neurons with 10 neurons for each, and 1 output neuron as presented in Fig. 2.

![ANN structure 9-10-10-1](image)

**Fig. 2.** ANN structure 9-10-10-1.

The activation function employed in this model was Sigmoid. Any real value can be used as an input for this function, which returns output values between zero and one. The output value will be closer to one if the input value is greater or more positive, whereas if it is smaller or more negative, the output value will be closer to zero. It can be mathematically formulated as depicted in (1).

$$f(x) = \frac{1}{1+e^{-x}}$$  \hspace{1cm} (1)

The total number of epochs and batch size used in this experiment were set to 200 and 50, respectively. Training is a process of determining the values of weights and biases for a neural network. The train-test technique is the most common way used to conduct training. The phishing dataset was split into training and testing sets. The neural network was trained using the training dataset. For determining the set of weights and biases values which have a small difference between actual output and expected output values, a few weights and biases values are tested. The process of selecting appropriate weights and biases values for minimizing error was also a part of the training. The test dataset was not used throughout this phase. After the training was complete, the final ANN model's weights and biases were applied to the test dataset. The model’s accuracy on the test dataset provides a rough estimation of the model’s accurateness will be once provided with previously unseen and new data.

An unbiased evaluation of the final model fit on the training dataset was made using the test dataset. A distinct dataset with a similar probability distribution as the training dataset is referred to as a test dataset. Minimal overfitting has occurred if a model that fits the training dataset also fits the test dataset well. Overfitting is typically shown by the training dataset fitting the model better than the test dataset. A test dataset is therefore a set of instances utilized just to evaluate the performance of a model which is a generalization of a fully specified classifier.
B. Convolutional Neural Network Model

Fig. 3 shows a complete process involved in numerical data classification using CNN, which includes input data, pre-processing, CNN feature extraction, classification, and performance evaluation.

![Fig. 3. CNN classification methodology.](image)

The input data used in this process were the same phishing dataset used for ANN classification. All the data are numerical data that must be saved in .dat format for pre-processing purposes. In the pre-processing stage, this study used one of the pre-processing methods proposed by [27], called Equidistant Bar Graphs. It is a technique of data wrangling to transform data in numerical form into image form. In order to represent a particular class, the transformed image must reflect some patterns. Equidistant bar graphs show the measurement of each attribute of a particular dataset. The phishing dataset was first normalized to 0 and 1, and then each attribute was drawn by using its measured value. The image’s width in pixels was given by a formula, wx + y(x + 1), where x represents the total number of attributes, w represents the bar width, and the distance between two consecutive bars represents by y. The image’s height was normalized to create a square image. In the experiments, the authors used 1–pixel and 2-pixel lengths for w and y, respectively. This creates an approximately 3d×3d size of the square image. A few data samples of the phishing website dataset that had already been transformed into bar graphs images are presented in Fig. 4. The images tagged along with the name of the respective class were labelled as Legitimate, Suspicious, and Phishy. CNN can only utilize these images if they represent a pattern in a convolved image. In this stage, all experiments were performed using Matlab 2020a software.

![Fig. 4. Bar graph of phishing website dataset.](image)

A convolutional neural network (CNN) is a deep learning algorithm that had been used in this research. Backpropagation artificial neural networks are the foundation of its architecture. It starts with an input image where each pixel represents input data that goes through a sequence of the feature selection process through convolution before being passed to the weighted perceptron where learning occurs through backpropagation. The main benefit of CNN is that it can learn the features on its own, as opposed to traditional neural networks where feature selection is a separate process and the model’s accuracy depends on the selection of pre-processing and feature selection methods used. Fig. 5 shows the architecture of CNN used in this study.

There are two main layers of CNN architecture which are feature extraction or it can be called as feature learning and classification layers. In the feature extraction layer, there are three sublayers which are convolutional, activation, and pooling layers. The convolutional layer takes in the images directly as the input and a set of small filters is convolved over the image to produce one or more feature maps. The process of convolution involves moving the filter over the image while computing the dot product of the filter and image elements. Certain features are extracted from the image as a result of this process. Then, a bounded output is created by passing the convolutional layer’s results via an activation function. CNN frequently employs Rectified linear units (ReLU) that turn any negative values into zero. Additionally, it trains the network far more quickly than other activation functions such as the hyperbolic tangent activation function (tanh). The down-sampling is carried out by the pooling layer, which also decreases the input size along each dimension. Average pooling and max pooling are two common pooling techniques that are usually used where the received image is divided into a collection of non-overlapping rectangles. Only the maximum and average values of each sub-region are obtained using max pooling and average pooling, respectively. The image is downsampled in this process. In this study, only the max pooling technique was used in the pooling layer.

The architecture of CNN moves to the classification layer after learning features in the feature extraction layer. The fully connected network in a traditional neural network is similar to this fully connected layer. The classification output is produced by a classification layer, such as softmax, in the CNN architecture’s final layer. For example, an image of a car goes through all the layers in CNN architecture which is then classified among the possible vehicles such as car, truck, van, or bicycle as the output of classification [28]. Meanwhile, for this study, the input images were classified as either phishy, suspicious, or legitimate. This experiment was performed using Visual Studio Code IDE with Python programming language.
C. Modified Artificial Neural Network Model with Rectified Linear Unit Activation Function

The modified ANN model had gone through all the stages in ANN classification methodology as shown in Fig. 1 which were collecting data, pre-processing the data, designing a suitable ANN model, training and testing the model, and lastly evaluating it. It was the same process as detailed in subsection A. The only difference was that this model employed rectified linear unit (ReLU) as an activation function which is usually utilized in the CNN model. Mathematically, ReLU is defined as shown in (2).

\[
f(x) = \max(0, x)
\]  

(2)

The function indicates that if it is given any negative input value, the output of the function will be zero, but if it is given any positive value of \(x\), the result will be the value itself. The equation gives the outcome within a range value from zero to infinity [29]. It has turned out to be the default activation function for several types of neural networks since a lot of classification models that employ it are easier to train and often produce better results [30]. Since the function involved a simple mathematic calculation, training also takes a lesser amount of time to complete [31].

D. Performance Evaluation

The ANN and CNN performances were evaluated by using root mean square error (RMSE), which is a well-known error indicator. It is a commonly employed metric for determining the differences between a model’s predicted values and the actual values. RMSE is a metric of accuracy used to compare the forecasting errors of several models for a certain dataset. It is always a positive value, and a value of 0, which is rarely achieved in practice, would mean that a perfect fit for the data. Generally, a smaller RMSE is preferable to a greater one. RMSE can be calculated as shown in (3).

\[
RMSE = \sqrt{\frac{1}{n} \sum_{i=1}^{n} (y_i - x_i)^2}
\]  

(3)

The formula showed that \(y_i\) represents the \(i\)th expected value, \(x_i\) indicates the \(i\)th actual value, and \(n\) is the total number of data instances. In mathematics, the Greek sigma symbol that resembles a strange E is known as summation. It is the total of the sequence numbers from \(i=1\) to \(n\). The performance of the classification models improves with decreasing RMSE values. The best performance of the ANN and CNN was determined by the lowest RMSE.

All the experiments were performed in a 3.20 GHz, 8 GB RAM, CPU Intel Core i7 processor system, and the operating system was Windows 10 64-bit. ANN and CNN models were implemented on Visual Studio Code IDE (Integrated Development Environment) using Python programming language and used the Keras library with TensorFlow backend. Numpy, Seaborn, Scikit Learn, and Pandas were among the additional libraries that were imported and used for the experiments.

IV. RESULTS AND DISCUSSION

There were three classification models designed in this research as stated in the method’s section. Each model was trained and tested 20 times to ensure the consistency of the results of the experiment. All training and testing RMSE values were recorded. Fig. 6 shows a graph plotted from the results of training and testing RMSE with 20 runs of experiments using an artificial neural network (ANN) with a sigmoid activation function. The average training RMSE was 0.3516, while the average testing RMSE was 0.3585. The lowest training RMSE was 0.3036, while the highest was 0.4148. The lowest testing RMSE was 0.3179, while the highest was 0.4164. The testing RMSE usually gave a higher value than the training RMSE as it used new data that need to be classified. This also can be observed in [27] which the accuracy of the test set is lower than the validation set. It can be seen in Fig. 6 that there was a very small difference between training and testing RMSE for each run. 0.1 is considered a good value of RMSE while 0.5 and above is considered high which is not good for accuracy.
The ANN (Sigmoid) model shows a high degree of variability in both training and testing RMSE values, with the range of values being relatively large. This variability may indicate that the ANN (Sigmoid) model is not consistent in its performance and may produce unreliable results.

Fig. 7 shows the results of training and testing RMSE of a convolutional neural network (CNN) model. The training RMSE values range from 0.2374 to 0.2926, while the testing RMSE values range from 0.3345 to 0.4169. The average training RMSE was 0.2631 while the average testing RMSE was 0.3569.

Compared to the ANN (Sigmoid) model, the CNN model showed less variation in the RMSE values across the 20 runs, indicating that it is more reliable in producing consistent results. The average training and testing RMSE of the CNN model were also lower than the average training and testing RMSE of the ANN (Sigmoid) model, suggesting that the CNN model has better performance in capturing the patterns in the dataset which produces better accuracy. Similarly, [7] shows an increase in accuracy by using the deep belief network technique compared to the neural network algorithm. Although the CNN method improves the performance and accuracy, the original pattern of the data may have changed due to the alteration of the data during the pre-processing stage. Furthermore, the networks used in the CNN model were very large compared to ANN (Sigmoid) model. Therefore, this study proposed to use the advantage of CNN which is Rectified Linear Unit (ReLU) activation function in the artificial neural network.

The modified ANN model using the ReLU activation function achieved a training RMSE between 0.2492 and 0.2817, with a mean training RMSE of 0.2703 as shown in Fig. 8. The testing RMSE was between 0.2976 and 0.3084, with a mean testing RMSE of 0.3033. The model showed consistent and reliable performance in terms of producing low training and testing RMSE values with small fluctuations between each run. The average training and testing RMSE values of the model were relatively low which indicated that the model was able to fit the training data well and also generalized well to new and unseen data, respectively. In addition, the model's performance was relatively consistent across the 20 runs, as the range of the RMSE values for both the training and testing sets is relatively narrow. This suggests that the model is not overfitting or underfitting the data, but rather finding a good balance between the two.

Compared to the ANN (Sigmoid) model, the modified ANN (ReLU) model showed slightly better performance in terms of producing more consistent results and more stable. The ReLU activation function is known for its ability to handle vanishing gradients and reduce the likelihood of overfitting. The results from the modified ANN (ReLU) model appear to support this claim, as the model produced more consistent results than the ANN (Sigmoid) model.

![ANN (Sigmoid) Model](image1)

**Fig. 6.** Training and testing RMSE of ANN (Sigmoid) model.

![Convolutional Neural Network Model](image2)

**Fig. 7.** Training and testing RMSE of CNN model.

![Modified ANN (ReLU) Model](image3)

**Fig. 8.** Training and testing RMSE of ANN (ReLU) model.
Compared to the CNN model, the modified ANN (ReLU) model has a slightly higher training RMSE, but the difference between average testing and training RMSE is smaller for the modified ANN (ReLU) but higher for the CNN model, indicating that the CNN model tends to overfit the data [32]. Moreover, the modified ANN (ReLU) model requires fewer networks than the CNN model, which can be an advantage in terms of computational efficiency.

Table 1 shows the average RMSE comparison between ANN (Sigmoid), CNN, and modified ANN (ReLU) models. Fig. 9 shows the comparison of all the training RMSE trained using ANN (Sigmoid), CNN, and modified ANN (ReLU) models. The CNN model has the lowest average RMSE which was 0.2631 followed by the modified ANN (ReLU) model with an average of 0.2703 RMSE. ANN (Sigmoid) has the highest RMSE average which was 0.3516. The testing RMSE has a different pattern, where modified ANN (ReLU) produced the best average RMSE which was 0.3039, followed by CNN with an average RMSE of 0.3569. The highest RMSE was produced by ANN (Sigmoid) model with an average of 0.3585 as shown in Fig. 10.

Overall, the results of this study suggest that the modified (ReLU) model is an effective approach for classifying phishing, legitimate, and suspicious websites. The modified ANN (ReLU) model was selected as the best model because it was able to generalize well to new data and the number of neural networks used in this model was also smaller than the CNN model. The modified (ReLU) model may have outperformed the CNN model because the original patterns of data were preserved, whereas, in the CNN model, some original patterns may have been changed during the pre-processing stage when converting numerical data to image data.

### TABLE I. COMPARISON OF CLASSIFICATION MODELS’ PERFORMANCE

<table>
<thead>
<tr>
<th>Phase</th>
<th>ANN (Sigmoid)</th>
<th>CNN</th>
<th>Modified ANN (ReLU)</th>
</tr>
</thead>
<tbody>
<tr>
<td>Training</td>
<td>0.3516</td>
<td>0.2631</td>
<td>0.2703</td>
</tr>
<tr>
<td>Testing</td>
<td>0.3585</td>
<td>0.3569</td>
<td>0.3039</td>
</tr>
</tbody>
</table>

Fig. 10. Testing RMSE comparison.

**V. CONCLUSION**

Artificial neural networks and convolutional neural network classifiers were implemented in this study for classifying phishing websites. The models produced from this study were ANN (Sigmoid), CNN, and modified ANN (ReLU). It can be seen from the experimental result that the modified ANN (ReLU) model demonstrated the best performance for phishing website classification followed closely by the CNN model. On the other hand, the ANN (Sigmoid) model had the poorest performance of the three models. In the future, more methods such as hybrid or other deep learning techniques can be implemented to produce much more reliable results and improve performance and accuracy. The dataset used for training and testing the models in this study may be relatively small, which could impact the model’s ability to capture the full complexity and diversity of phishing websites. A larger dataset might be needed to improve the model’s robustness and generalization. Further research on other datasets is also needed to validate the effectiveness of the three models produced by this study.

**REFERENCES**


Purchase Intention and Sentiment Analysis on Twitter Related to Social Commerce

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Faculty of Computer Science Universitas Indonesia Depok, Indonesia²
Faculty of Engineering and Computer Science Universitas Teknokrat Indonesia Bandar Lampung, Indonesia⁴

Abstract—Social commerce is a digital and efficient solution to transform existing commerce and address contemporary issues. TikTok Shop, a popular and trending social commerce platform, competes with established competitors like Facebook Marketplace and Instagram Shop. TikTok Shop offers benefits and incentives to attract users for both sales and product purchases. In this study, various algorithmic approaches such as Naïve Bayes, K-Nearest Neighbor, Support Vector Machine, Logistic Regression, Decision Tree, Random Forest, LGBM Boost, Ada Boost, and Voting Classifier are utilized to analyze and compare sentiments expressed on Twitter regarding Facebook, Instagram, and TikTok. The aim is to determine the methods with the best performance and identify the social commerce platform with the highest purchase intention and positive sentiment. The results indicate that TikTok has more positive sentiment than Facebook and Instagram at 93.07% with the best-performing classification model, Decision Tree. In conclusion, TikTok exhibits the highest positive sentiment percentage, indicating a greater number of positive reviews compared to Facebook and Instagram. According to the theory of evaluation scores for measuring model performance, values above 0.90 represent models with good performance.

Keywords—Algorithm; machine learning; sentiment; social commerce

I. INTRODUCTION

E-commerce has rapidly developed worldwide in the 2000s. This has led many traditional companies and stores to start opening online stores, which has brought about a change in the world of trade. Initially, business transactions were conducted in a traditional manner, with sellers and buyers interacting directly. However, with the emergence of e-commerce, these interactions shifted to an online platform. The increase in consumer trust and changes in online shopping behavior have supported the growth of e-commerce [1]. In the 2010s, there was a significant increase in social media users worldwide, marking the beginning of the emergence of social commerce. Social commerce combines e-commerce and social media, reflecting a shift in how businesses operate and interact with consumers. This shift has had a significant impact and positive benefits for both customers and sellers. Social commerce enables businesses to reach a wider and more distant target audience due to its ease of use [2], [3].

Social commerce is a hybrid of e-commerce and social media that enables trade transactions to take place on both e-commerce platforms and social media platforms like Facebook, Instagram, TikTok. With social commerce, consumers can purchase products or services from trusted sellers via social media platforms they use daily [4]. Sales on social commerce occur through social media features, such as links to online stores, business nuances, product reviewers, and instant shopping features. Additionally, social commerce serves as a marketing platform where businesses can promote their products and reach potential consumers through paid advertising or social listening. This type of social commerce has proven to be effective in boosting sales and user engagement [1], [5], [6].

There are many social commerce platforms circulating today such as Facebook marketplace, Instagram, and TikTok Shop. The Facebook marketplace is usually used for people to make transactions directly without a third person as an intermediary, while to make buying and selling transactions on Instagram features can be accessed through official accounts, or if we want to make sales on the Instagram market we have to register a special store first and if we want to make a purchase we have to visit the official store account of the product we are looking for on Instagram, in contrast to TikTok shop, which is a social e-commerce platform that is part of the TikTok feature and can be used for product sales [7], [8].

Currently, TikTok Shop is the main solution in the modern business world because the price it offers is relatively cheaper and benefits both sellers and buyers. Over time, TikTok Shop developed its innovations to allow users to make sales and purchases of products in the live videos they display. This feature was just launched in March 2021 and is currently available in several countries. The TikTok Shop offers a variety of products from various categories, such as fashion, beauty, electronics and more [9].

Purchase intention is based on people's interest and interest in an object. This interest led to purchases made by the public. N’da et al. 2023 [10] explores the direct and mediated effects of customers’ perception of purchase budget (BGT) on purchase intention (PIT) through perceived quality (PPQ), perceived price (PPR) and perceived benefit (PB) in a cross-national context to understand the role of BGT in predicting customer purchase intention when selling smartphones online shopping through international platforms. While Jiang et al. 2023 [11] state that the market already has a significant size and the number is constantly increasing, the need to understand the factors affecting the purchase intention of consumers and explore the relationship between purchase intention and shopping behavior becomes more and more important and urgent. It can be concluded that purchase intention is very
important as a parameter of the success of a market or business.

The aim of this research is to analyze the community's assessment of TikTok Shop, Instagram Shop, and Facebook Marketplace to determine which social commerce platform is profitable and receives the most positive sentiment for daily transactions such as buying and selling products. This research aims to increase trust, satisfaction, and purchase intentions of social commerce users. The study also explores the best-performing algorithms, the strengths and weaknesses of the methods used, and which social commerce platform generates the highest purchase intent and positive sentiment. Theoretically, this research contributes to a combination of algorithms that can be used and developed by other researches. Practically, this research contributes to social commerce developers (such as Facebook, Instagram and Tiktok) knowing their business position. So, the features in the application can be adjusted according to the community's response.

The chosen approach methods for this study include Naïve Bayes (NB), K-Nearest Neighbors (KNN), Support Vector Machine (SVM), Logistic Regression (LR), Decision Tree (DT), Random Forest (RF), LGBM Boost (LGBM), Ada Boost (ADA), and Voting Classifier (VOT) algorithms. These algorithms have been frequently used in previous sentiment analysis studies due to their performance, accuracy, and ability to handle data effectively. Thus, they are deemed suitable for this research. Based on these papers [12]–[18], machine learning models are used in certain cases.

This paper is organized as follows: In Section II, we conducted several related works about sentiment analysis on social commerce. Then, in Section III explains the research methodology applied in this research. Section IV explains the results of the research. Last, the result of the research are explained in Section V.

II. RELATED WORK

Facebook and Instagram have succeeded in attracting customer interest in buying or selling their products on those social commerce. The feature of buying and selling products in social commerce originally was not the main focus in their business. Since technological development was significantly improved, they take advantage of these features to increase interest from customers so they can maintain their loyalty. TikTok, being one of the social commerce participating in using the feature of buying and selling products as their business process TikTok provides various advantages that are quite advantageous to company owners. They will have more opportunities if they use the TikTok [9].

Nabiha et al. in 2021 [19] and Bayhaqy in 2018 [14] showed that sentiment analysis related to social commerce is only carried out with several single algorithm classification methods, such as Naïve Bayes, SVM, Decision Tree and KNN, while Lestari in 2022 [20] only uses a single algorithm classification method, namely KNN and the application of N-Gram to the method, and also focuses on only one platform, namely the TikTok Shop. Nabiha et al. in 2021 [19] did not use model evaluation measurements such as Precision, Recall and F1-Score, and the accuracy value of the classification model tends to be quite low, namely below 0.75. Those research [14], [19], [20] only focused on social commerce or one of the social commerce platforms without comparing one platform to another, while this study uses nine classification methods consisting of single algorithms and ensemble algorithms which might improve the evaluation results for each method for each sentiment data related to three social commerce platforms, namely Facebook, Instagram and TikTok to determine which platform has more positive sentiment.

Botchway R et al. in 2022 [21] used binary particle swarm optimization to improve the accuracy of their models. The optimization method used can produce different accuracy values for each classification method. Botchway R and friends succeeded in increasing the accuracy value of the Naïve Bayes method by 11.6%, SVM by 8.43% and KNN by 0.91%. Meanwhile, in research Kamrozi et al. in 2023 [22] does not use a special method to increase the evaluation value of the method used, namely the Lexicon Method. Therefore, this study uses hyper parameter tuning to increase the evaluation value of the method used and produce different evaluation values for each method.

Das et al. 2023 [13] used machine learning approaches to stop hateful activities from happening, such as Logistic Regression, Gaussian Naïve Bayes, K-Nearest Neighbor, Decision Tree, Random Forest and Support Vector Machine on detection of hate speech from Twitter. Support Vector Machine, Decision Tree and Random Forest outperformed all the other models, achieving state-of-art 95.5%, 96.2% and 98.2% accuracy respectively at finding the hidden meaning inside the large number of comments and therefore determining whether there is any hateful event is going on or not. However, this research needs to use more models in order to be able to compare and test how well the other models are, especially ensemble algorithms.

III. MATERIAL AND METHODS

In this chapter, the methodology used to predict intent detection and sentiment analysis is explained in relation to the user experience when using TikTok. First, people's Twitter remarks will be extracted. Second, the pre-processing stage will eliminate inconsistent and incomplete data. Third, a feature selection approach for identifying discriminating phrases for training and classification will be deployed. Fourth, nine machine learning approaches using NB, KNN, SVM, LR, DT, RF, LGBM, ADA and VOT will be used to classify sentiments into two categories, positive and negative. Finally, an evaluation will be conducted by calculating the performance value for each method using metrics and then comparing which method has the best performance. Fig. 1 illustrates the research methodology used.

A. Data Collection

Data collection or information extraction was conducted on social media platform which frequently used by the public, namely Twitter. The information was gathered via Twitter's Application Programming Interface (API). Data of Twitter may be used to uncover themes and items being discussed based on certain keywords, to evaluate sentiment on specific businesses, and to obtain opinion on the latest products and services. For
the intended data extraction needs, users may define needed attributes such as usernames, keywords, locations, name of place and others. [9], [23], [24].

![Research methodology.](image)

Data collection was carried out by collecting several community's tweet on Twitter regarding their ratings and experiences when using social commerce. The data collection was carried out with crawling data method using the Python programming language and Jupyter Notebook. Therefore, the dataset was developed independently and was not obtained from other paper references [Dataset is available in Supporting Materials section, Page 7].

In this section, the sentiment data obtained is divided into two categories, namely, positive and negative, for each social commerce platform: Facebook, Instagram, and TikTok. Fig. 2, 3, and 4 display the sentiment data collected from Twitter.

![Example of Facebook’s sentiment.](image)

**TABLE I. HEADCOUNT OF SENTIMENT DATA**

<table>
<thead>
<tr>
<th>Categories</th>
<th>Facebook</th>
<th>Instagram</th>
<th>TikTok</th>
</tr>
</thead>
<tbody>
<tr>
<td>Positive</td>
<td>931</td>
<td>485</td>
<td>6772</td>
</tr>
<tr>
<td>Negative</td>
<td>584</td>
<td>188</td>
<td>504</td>
</tr>
<tr>
<td>Percentage of Positive (%)</td>
<td>60.65</td>
<td>72.06</td>
<td>93.07</td>
</tr>
<tr>
<td>Percentage of Negative (%)</td>
<td>39.35</td>
<td>7.94</td>
<td>6.93</td>
</tr>
<tr>
<td>Total</td>
<td>1535</td>
<td>673</td>
<td>7276</td>
</tr>
</tbody>
</table>

Based on Table I, it is known that TikTok has the highest number of positive sentiments, amounting to 6772 sentiments, which accounts for 93.07% out of 7276 sentiments. It is followed by Instagram, which has 485 positive sentiments, representing the second-highest percentage of 72.06% out of 673 data. Facebook, on the other hand, has the lowest number of positive sentiments, with 931 sentiments, which corresponds to the lowest percentage of 60.65% out of 1535 sentiments. This indicates that TikTok generates a greater number of positive sentiments from the community compared to Facebook and Instagram.

B. Pre-processing Data

The data obtained from Twitter is in the form of text, as it contains sentiments or tweets from the public regarding a product or service. However, such data or information usually contains noise, which can make data analysis more challenging. Therefore, data pre-processing is carried out to remove unwanted words in tweets. All tweets are processed through four stages of pre-processing, which are as follows: Tokenization; Stopword Removal; Stemming; POS Tagging; and Bag of Words [18], [21], [23]–[25].

In this section, the results of data pre-processing for each dataset of Facebook, Instagram, and TikTok, regarding their positive and negative sentiments, are obtained. Table II represents the outcomes of a collection of words along with their frequency of occurrence in each social commerce platform.

![Example of Instagram’s sentiment.](image)

Based on Table II, it is known that the sentiment data for each social commerce platform contains different words and their frequencies. This is done to observe which words are frequently discussed by the community regarding those social commerce platforms on Twitter. Additionally, it is also done to examine the correlation between the conducted research and the collected sentiment data, whether there is any connection or
Based on the performed data pre-processing, it is found that the sentiment data related to Facebook, Instagram, and TikTok have the same data alignment, which involves discussions on purchases, sales, and users' utilization of those social commerce platforms.

### TABLE II. STEM WORD AND FREQUENCY

<table>
<thead>
<tr>
<th>Social Commerce</th>
<th>Stem Word (Frequency)</th>
</tr>
</thead>
</table>

### C. Feature Selection Method

After the data is pre-processed, feature selection is performed as an important step in sentiment analysis. By selecting a subset of the important attributes to be included in the model's creation, this stage determines the most predictive traits. This method offers the benefit of lowering the data’s high dimensionality and removing redundant, noisy, and unneeded content. Furthermore, this strategy can contribute on the development of a quick and accurate sentiment categorization. In this study, several factors influence feature selection, including data consistency, data amount, and the need to find the most effective feature selection approach [16], [20], [21], [23]–[28].

### D. Split Data

Split data is data that has been partitioned into two or more subsets. A two-part split is commonly used to analyze or test the data and train the model. Data splitting is an important aspect of data science, particularly for building data-driven models. This strategy improves the accuracy of data models and data-driven processes such as machine learning [16], [19].

To reduce overfitting, data splitting is often employed in machine learning. In this scenario, a machine learning model fits its training data too well and fails to consistently fit further data. The initial data in a machine learning model is often separated into three or four categories. The training set, the development set, and the testing set are the three most popular sets [23]:

1) The training set is the collection of data used to train the model. The model should keep an eye on and learn from the training data, and any of its parameters should be improved.

2) The testing set is the piece of data examined in the final model and compared to the preceding data sets. The testing set is used to evaluate the final mode and algorithm.

Data should be separated so that large amounts of training data may be included in data sets. Data may be split 80-20 or 70-30 between training and testing, for example. The exact ratio varies depending on the data, but for small data sets, a 70-20-10 split for training, development, and testing works best [28].

The data in this study is separated into training and testing data. The data split is carried out using an 80% training data size and a 20% testing data size of the total data in the dataset. Table III illustrates the number of training and testing data from the three social commerce platforms for each sentiment data.

### TABLE III. SPLIT DATA

<table>
<thead>
<tr>
<th>Categories</th>
<th>Facebook</th>
<th>Instagram</th>
<th>TikTok</th>
</tr>
</thead>
<tbody>
<tr>
<td>Training Data</td>
<td>1213</td>
<td>538</td>
<td>5821</td>
</tr>
<tr>
<td>Testing Data</td>
<td>303</td>
<td>135</td>
<td>1455</td>
</tr>
<tr>
<td>Total</td>
<td>1535</td>
<td>673</td>
<td>7276</td>
</tr>
</tbody>
</table>

Based on Table III, it is known that each dataset related to the sentiments of Facebook, Instagram, and TikTok has different numbers of training and testing data. This is due to the varying percentages of training and testing data sizes. The larger the percentage of training data, the higher the likelihood of accurate predictions on the testing data.

### E. Evaluation of Classification Methods

At this stage, the performance evaluation of nine classification methods is carried out using standard classification performance metrics. Furthermore, four outcomes are possible at this point: true positive, false positive, true negative, and false negative. It is a true positive if the document label is positive and is classified as such. It is referred to as a false negative if it is classified as negative. A true negative is a document label that is classified as such. If it is labeled positive, it is considered a false positive [16]–[21], [23]–[28].

The accuracy measure is used to assess the accuracy of the likelihood of taking [16], [17], [19]–[21], [23], [24], [28]. The precision metric is the proportion of predicted classes that are the actual classes [16], [18], [20], [23]–[28]. The recall metric is the proportion of actual classes that are predicted as such. The F1-Score (F) is used to assess model performance [18], [23]–[28]. Here are the formulas for each metric.

\[
\text{Accuracy (A)} = \frac{TP + TN}{TP + TN + FP + FN} \times 100
\]

\[
\text{Precision (P)} = \frac{TP}{TP + FP} \times 100
\]

\[
\text{Recall (R)} = \frac{TP}{TP + FN} \times 100
\]
The Accuracy, Precision, Recall, and F1-Score values varied from 0 to 1, with 1 being 100% similar and 0 representing 100% different. While TP, TN, FP, and FN represent the number of relevant identified features, relevant non-identified features, irrelevant identified features, and irrelevant non-identified features. Then, we applied 10-fold cross validation for evaluation.

F. Hyper Parameter Tuning

A Machine Learning model is a mathematical model with several parameters that must be taught from data. By training a model using existing data, we may fit the model parameters. [28], [29].

Another type of parameter, known as hyper parameters, cannot be taught using the normal training procedure. They are usually resolved before the training method begins. These parameters describe important model characteristics like as complexity and learning speed. The following are some examples of model hyper parameters [21], [26]:

1) The penalty in Logistic Regression Classifier i.e. L1 or L2 regularization.
2) The learning rate for training a neural network.
3) The C and sigma hyper parameters for support vector machines.
4) The k in k-nearest neighbors.

Models can have a large number of hyperparameters, and choosing the best combination of parameters can be thought of as a search problem. In this work, GridSearchCV and RandomizedSearchCV were used for hyper parameter optimization. This technique for GridSearch CV seeks the optimal collection of hyper parameters from a grid of hyper parameter values. While RandomizedSearchCV addresses the shortcomings of GridSearchCV by going through only a limited number of hyper parameter choices. It travels randomly within the grid to discover the optimal set of hyper parameters while also reducing wasteful processing.

IV. RESULTS AND DISCUSSION

A. Results

In this section, evaluation is conducted on the classification methods used to measure their performance on sentiment data from Facebook, Instagram, and TikTok on Twitter. This evaluation includes accuracy, precision, recall, and F1-score. Table IV presents the evaluation scores for each method.

Based on Table IV, the evaluation scores for each classification model used on sentiment data from Facebook, Instagram, and TikTok are known. The evaluation scores range from 0 to 1, where a score closer to 1 indicates better performance, while a score closer to 0 indicates poorer performance.

The Random Forest Classifier has the highest assessment ratings for Facebook, with accuracy, precision, recall, and F1-score values of 0.84, 0.86, 0.72, and 0.76, respectively. The Decision Tree classification model has the highest assessment ratings on TikTok, with accuracy, precision, recall, and F1-score values of 0.94, 0.74, 0.77, and 0.76, respectively.

However, for TikTok, there are no evaluation scores for the Random Forest and Voting Classifier models, as they have complex algorithms that require extensive memory capabilities to be executed on TikTok sentiment data. Therefore, during execution, the evaluation scores did not appear.

After evaluation value from the classification method had calculated, hyper parameter tuning was performed and it obtains the results. Table V presents the outcomes of utilizing hyper parameter tuning for multiple classification models on the sentiment data from each of the three social commerce platforms.

<table>
<thead>
<tr>
<th>Classifier</th>
<th>Accuracy</th>
<th>Precision</th>
<th>Recall</th>
<th>F1-Score</th>
</tr>
</thead>
<tbody>
<tr>
<td>Facebook</td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>NB</td>
<td>0.518152</td>
<td>0.482661</td>
<td>0.291230</td>
<td>0.299336</td>
</tr>
<tr>
<td>KNN</td>
<td>0.729373</td>
<td>0.744252</td>
<td>0.657933</td>
<td>0.662345</td>
</tr>
<tr>
<td>SVM</td>
<td>0.768977</td>
<td>0.799486</td>
<td>0.704118</td>
<td>0.715866</td>
</tr>
<tr>
<td>LR</td>
<td>0.772277</td>
<td>0.758883</td>
<td>0.741819</td>
<td>0.748026</td>
</tr>
<tr>
<td>DT</td>
<td>0.749175</td>
<td>0.733304</td>
<td>0.714262</td>
<td>0.720534</td>
</tr>
<tr>
<td>RF</td>
<td>0.805858</td>
<td>0.831439</td>
<td>0.757690</td>
<td>0.773671</td>
</tr>
<tr>
<td>LGBM</td>
<td>0.752475</td>
<td>0.740581</td>
<td>0.711341</td>
<td>0.719420</td>
</tr>
<tr>
<td>ADA</td>
<td>0.716172</td>
<td>0.698963</td>
<td>0.665927</td>
<td>0.672177</td>
</tr>
<tr>
<td>VOT</td>
<td>0.798680</td>
<td>0.812544</td>
<td>0.749836</td>
<td>0.764106</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Classifier</th>
<th>Accuracy</th>
<th>Precision</th>
<th>Recall</th>
<th>F1-Score</th>
</tr>
</thead>
<tbody>
<tr>
<td>Instagram</td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>NB</td>
<td>0.725926</td>
<td>0.670817</td>
<td>0.698232</td>
<td>0.678178</td>
</tr>
<tr>
<td>KNN</td>
<td>0.725926</td>
<td>0.647857</td>
<td>0.645202</td>
<td>0.646472</td>
</tr>
<tr>
<td>SVM</td>
<td>0.785185</td>
<td>0.886719</td>
<td>0.597222</td>
<td>0.598914</td>
</tr>
<tr>
<td>LR</td>
<td>0.844444</td>
<td>0.865472</td>
<td>0.726010</td>
<td>0.760254</td>
</tr>
<tr>
<td>DT</td>
<td>0.800000</td>
<td>0.744652</td>
<td>0.731061</td>
<td>0.737184</td>
</tr>
<tr>
<td>RF</td>
<td>0.829630</td>
<td>0.834846</td>
<td>0.707071</td>
<td>0.737421</td>
</tr>
<tr>
<td>LGBM</td>
<td>0.800000</td>
<td>0.784895</td>
<td>0.660335</td>
<td>0.683071</td>
</tr>
<tr>
<td>ADA</td>
<td>0.807407</td>
<td>0.755309</td>
<td>0.736111</td>
<td>0.744467</td>
</tr>
<tr>
<td>VOT</td>
<td>0.837037</td>
<td>0.829445</td>
<td>0.729798</td>
<td>0.758458</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Classifier</th>
<th>Accuracy</th>
<th>Precision</th>
<th>Recall</th>
<th>F1-Score</th>
</tr>
</thead>
<tbody>
<tr>
<td>TikTok</td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>NB</td>
<td>0.936770</td>
<td>0.768966</td>
<td>0.515395</td>
<td>0.514254</td>
</tr>
<tr>
<td>KNN</td>
<td>0.923711</td>
<td>0.602292</td>
<td>0.543484</td>
<td>0.556365</td>
</tr>
<tr>
<td>SVM</td>
<td>0.944330</td>
<td>0.936527</td>
<td>0.569525</td>
<td>0.607047</td>
</tr>
<tr>
<td>LR</td>
<td>0.948454</td>
<td>0.949216</td>
<td>0.601783</td>
<td>0.654734</td>
</tr>
<tr>
<td>DT</td>
<td>0.939519</td>
<td>0.747713</td>
<td>0.772334</td>
<td>0.759298</td>
</tr>
<tr>
<td>LGBM</td>
<td>0.940893</td>
<td>0.970304</td>
<td>0.537634</td>
<td>0.554698</td>
</tr>
<tr>
<td>ADA</td>
<td>0.945704</td>
<td>0.859073</td>
<td>0.605324</td>
<td>0.653915</td>
</tr>
</tbody>
</table>
Based on Table V, the latest accuracy values are known after conducting hyper parameter tuning. Hyper parameter tuning was performed only on a subset of classification models due to memory limitations for highly complex classification models such as Random Forest, Voting Classifier, and others. Table VI compares the accuracy value before and after hyper parameter tuning was performed.

Based on Table VI, it is known that after conducting hyper parameter tuning, certain classification models experienced a decrease in accuracy scores. However, there are also several classification models that showed an improvement in accuracy scores. This can be influenced by the capacity and memory capabilities of the device used to execute the program. Upon re-execution, the obtained scores are also expected to differ. The values generated in this study represent the maximum iterations conducted to obtain the best possible scores. For example, Naive Bayes which has an accuracy value of 0.51 before using hyper parameter tuning, and 0.75 after using hyper parameter tuning. It can be said that hyper parameter tuning can increase and optimize the performance evaluation value of the resulting model. Mendes et al. 2023 [30] and Martineau et al. 2023 [31] explained that the use of hyper parameter tuning can optimize the performance evaluation value of the model so that it can increase the chances of success.

**TABLE V. HYPER PARAMETER TUNING**

<table>
<thead>
<tr>
<th>Classifier</th>
<th>Par 1</th>
<th>Par 2</th>
<th>Par 3</th>
<th>Accuracy</th>
</tr>
</thead>
<tbody>
<tr>
<td>NB</td>
<td>Var_smoothing: 0.0152</td>
<td>-</td>
<td>-</td>
<td>0.757273</td>
</tr>
<tr>
<td>SVM</td>
<td>C: 10</td>
<td>Gamma: 0.01</td>
<td>Kernel: rbf</td>
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</tr>
<tr>
<td>LR</td>
<td>C: 11.28</td>
<td>Penalty: 12</td>
<td>Solver: liblinear</td>
<td>0.790893</td>
</tr>
<tr>
<td>DT</td>
<td>Max_depth: 41</td>
<td>-</td>
<td>-</td>
<td>0.749986</td>
</tr>
<tr>
<td>LGBM</td>
<td>Learning_rate: 0.1</td>
<td>n_estimators: 50</td>
<td>-</td>
<td>0.725577</td>
</tr>
<tr>
<td>ADA</td>
<td>n_estimators: 50</td>
<td>-</td>
<td>-</td>
<td>0.702490</td>
</tr>
</tbody>
</table>

**TABLE VI. BEFORE AND AFTER HYPER PARAMETER TUNING**

<table>
<thead>
<tr>
<th>Classifier</th>
<th>Before</th>
<th>After</th>
</tr>
</thead>
<tbody>
<tr>
<td>NB</td>
<td>0.518152</td>
<td>0.757273</td>
</tr>
<tr>
<td>SVM</td>
<td>0.768977</td>
<td>0.781670</td>
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<tr>
<td>LR</td>
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</tr>
<tr>
<td>ADA</td>
<td>0.716172</td>
<td>0.702490</td>
</tr>
</tbody>
</table>

### B. Discussion

Based on the obtained results, it is evident that each sentiment dataset related to Facebook, Instagram, and TikTok has different percentages of positive and negative sentiments. Among the three social commerce platforms, TikTok has the highest percentage of positive sentiments, amounting to 93.07%, using the best classification model, which is Decision Tree with an accuracy score of 0.94. After evaluating the performance of each classification method on each social commerce platform, the following findings were obtained:

1) Facebook had a positive sentiment percentage of 60.65%, with the best classification model being Random Forest Classifier, achieving accuracy, precision, recall, and F1-score values of 0.80; 0.83; 0.75; and 0.77 respectively.

2) Instagram had a positive sentiment percentage of 72.06%, with the best classification model being Logistic Regression, achieving accuracy, precision, recall, and F1-score values of 0.84; 0.86; 0.72; and 0.76 respectively.

3) TikTok had a positive sentiment percentage of 93.07%, with the best classification model being Decision Tree, achieving accuracy, precision, recall, and F1-score values of 0.94, 0.74, 0.77, and 0.76, respectively.

This indicates that TikTok receives more positive reviews from the community compared to Facebook and Instagram, which cannot be known in research [14], [19]–[22].
Additionally, the classification model used for TikTok sentiment data also exhibits significantly high and consistent accuracy scores, always exceeding 0.90. According to the theory of evaluation scores for measuring model performance, values above 0.90 are considered to represent models with excellent performance.

With the results obtained that Logistic Regression, Decision Tree and Random Forest can produce high model performance evaluation values, it can be proven in the paper Das et al. 2023 [13], Imran et al. 2022 [12], Gulati et al. 2022 [15] which state that the Logistic Regression, Decision Tree and Random Forest models have high performance values and can be the best some case.

C. Limitations

There are several limitations on this study such as:

- We had only collect sentiment data from Facebook, Instagram and TikTok.
- We only measure the evaluation of methods through nine classification methods; therefore we do not use the other methods since the device of researcher is not compatible to execute the other methods.
- We only compared nine classification methods and prioritized the accuracy of evaluation from each method.

V. CONCLUSION

This section will explain the conclusion of the conducted research. Based on the research conducted, multiple machine learning classification models were tested on sentiment data related to Facebook, Instagram, and TikTok to determine which social commerce platform had the highest number of positive reviews.

Based on these results, it can be concluded that the classification models have different evaluation scores depending on the data used. The best classification model for Facebook is Random Forest Classifier, for Instagram is Logistic Regression, and for TikTok is Decision Tree. However, for classification models with low evaluation scores, using hyper parameter tuning may improve their performance.

In conclusion, TikTok exhibits the highest positive sentiment percentage, indicating a greater number of positive reviews compared to Facebook and Instagram. According to the theory of evaluation scores for measuring model performance, values above 0.90 represent models with excellent performance. Notably, the classification model used for TikTok sentiment data consistently achieves accuracy scores above 0.90. With this research, hopefully can help people choose the best social commerce to use and social commerce developers can increase their application and business value in order to increase public interest.

ACKNOWLEDGMENT

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Abstract—Cyber-bullying is a growing problem in the digital age, affecting millions of people worldwide. Deep learning algorithms have the potential to assist in identifying and combating Cyber-bullying by detecting and classifying harmful messages. This paper uses two Ensemble deep learning (EDL) models to detect Cyber-bullying on text data, images and videos—and an overview of Cyber-bullying and its harmful effects on individuals and society. The advantages of using deep learning algorithms in the fight against Cyber-bullying include their ability to process large amounts of data and learn and adapt to new patterns of Cyber-bullying behaviour. For text data, firstly, a pre-trained model BERT (Bidirectional Encoder Representations from Transformers) is used to train on cyber-bullying text data. The next step describes the data preprocessing and feature extraction techniques required to prepare data for deep learning algorithms. We also discuss the different types of deep learning algorithms that can be used for Cyber-bullying detection, including convolutional neural networks (CNNs), recurrent neural networks (RNNs), and deep belief networks (DBNs). This paper combines the sentiment analysis model, such as Aspect-based Sentiment Analysis (ABSA), for classifying bullying messages. Deep Neural network (DNN) used the classification of Cyber-bullying images and videos. Experiments were conducted on three datasets such as Twitter (Kaggle), Images (Online), and Videos (Online). Datasets are collected from various online sources. The results demonstrate the effectiveness of EDL and DNN in detecting Cyber-bullying in terms of detecting bullying data from relevant datasets. The EDL and DNN obtained an accuracy of 0.987, precision of 0.976, F1-score of 0.975, and recall of 0.971 for the Twitter dataset. The performance of Ensemble CNN brought an accuracy of 0.887, precision of 0.88, F1-score of 0.88, and recall of 0.887 for the Image dataset. For the video dataset, the performance of Ensemble CNN is an accuracy of 0.807, precision of 0.81, F1-score of 0.82, and recall of 0.81. Future research should focus on developing more accurate and efficient deep learning algorithms for Cyber-bullying detection and investigating the ethical implications of using such algorithms in practice.

Keywords—Cyber bullying; ensemble deep learning (EDL); convolutional neural networks (CNNs); recurrent neural networks (RNNs); deep belief networks (DBNs)

I. INTRODUCTION

Sentiment analysis can be used to detect instances of cyber-bullying by analyzing the language and tone used in online messages, comments, or posts [1]. The process involves using natural language processing (NLP) techniques to identify the sentiment expressed in a text, whether it is positive, negative, or neutral [2]. One approach to using sentiment analysis for cyber-bullying detection is to look for negative sentiments expressed towards an individual or group, such as derogatory or offensive language, insults, or threats [3]. These can be identified using NLP techniques, such as part-of-speech tagging, named entity recognition, and sentiment analysis algorithms. Another approach is to analyze the context in which the message is being conveyed, such as the topic being discussed and the online community it is being shared in [4] [5]. For example, if a message contains negative sentiments towards a specific group of people or individual, and is being shared in an online community known for hostile or aggressive behaviour, it may be a sign of cyber-bullying. It is important to note that sentiment analysis is not a foolproof method for detecting cyber-bullying and should be used in combination with other techniques, such as human moderation and reporting mechanisms. Additionally, it is important to ensure that the use of sentiment analysis does not infringe on individuals' privacy rights or result in false accusations.

Cyber-bullying (CB) is a default model for publicly abusing a person. Many online social media networking (OSMN) like Facebook, Twitter, and Instagram act as a medium for people based on cyber-bullying attacks [6]. Several automated models aim to develop to classify cyber-bullying in terms of text messages, audio, and videos [7]. Sometimes based on the topic modeling, cyber-bullying attacks occur in several datasets belonging to topic modeling. Twitter has become more popular for cyber-bullying by using various types of attacks. Exemplary-grained automated models were developed to detect cyber-bullying regarding topic modeling [8]. It is essential to complain the cyber-bullying attacks in OSMS if the user violates the ITE Law, which is considered a crime in OSMS like Twitter [9]. The victim should act if any abusive language is used on Twitter. The aim of cyber-bullying mainly focuses on classifying the tweets present on Twitter. This paper describes the automated approach for detecting cyber-bullying attacks by using the sentiment analysis on text messages, videos, and audio. Sentiment analysis helps the proposed automated approach to find the cyber-bullying attacks in multi-media. An aspect based sentiment analysis model combined with automated classification approach used to classify the cyber-bullying words from given input data. It shows the improved
performance in terms of accuracy, sensitivity, specificity, F1-score and precision.

The organization of the paper is as follows. Section II explains the literature survey about various methods of cyberbullying. Section III and Section IV give the significance of the work and training and feature extraction techniques. Section V describes the experimental results with comparative performance. Section VI describes the conclusion.

A. Significant Points of Proposed System

Cyber-bullying has become a common problem in today's digital age, with severe psychological and emotional consequences for victims. Detecting and preventing cyber-bullying is critical for ensuring individuals' well-being in online communities. Manual monitoring of online content, on the other hand, is a time-consuming and inefficient process. As a result, there is a need to create an automated system that uses deep learning models to detect cyber-bullying. This project aims to develop and test a deep-learning model to detect cyber-bullying in online text, images, and videos. The model should be able to classify messages, comments, or posts as cyber-bullying or non-cyber-bullying based on their content. The detection system's accuracy, precision, and recall should be high, with low error rates.

II. LITERATURE SURVEY

Semantic-enhanced marginalized denoising auto-encoder (SEMDA) is a type of neural network that is trained to reconstruct clean data from noisy data by learning the underlying distribution of the input data. It has been enhanced with semantic information to improve its ability to capture the context and meaning of the text [10]. To use SEMDA for cyber-bullying detection, the first step is to train the model on a dataset of labelled examples of cyber-bullying and non-cyber-bullying text. During training, the SEMDA learns to identify patterns and features that distinguish between cyber-bullying and non-cyber-bullying text. Once the SEMDA is trained, it can be used to detect cyber-bullying in new text. The input text is first pre-processed to remove noise and convert it into a numerical representation that can be input to the SEMDA. The SEMDA then reconstructs the clean version of the input text, and the reconstruction error is used to determine whether the input text is cyber-bullying or not. The advantage of using SEMDA for cyber-bullying detection is that it can capture the semantic meaning of the text, which is critical for detecting subtle forms of cyber-bullying. Additionally, it is more robust to noise and can handle variations in the input text. In conclusion, the use of semantic-enhanced marginalized denoising auto-encoder is a promising approach for cyber-bullying detection, and it has the potential to improve the accuracy of current cyber-bullying detection systems. Zhang et al. [11] proposed the novel pronunciation-based CNN to solve issues in detecting cyber-bullying based on the pronunciation of misspelled words. The proposed approach corrects the errors that occur by spellings that didn't change in accent because of its noise and data sparsity imbalance present in the dataset. To solve these issues, the proposed model applied to two datasets, Twitter and Form spring. The proposed approach's comparative performance shows more effective results than existing models. Zhang et al. [12] developed a fine-grained model to detect cyber-bullying messages based on linguistic analysis. The proposed model focused on finding the patterns based on Linguistic Inquiry Word Count (LIWC) to detect fine-tuned cyber-bullying detection from different social media datasets. Dalvi et al. [13] proposed an ML model to see cyber-bullying from social media posts like Twitter. The proposed ML model is used to prevent bullying on Twitter. Using the Twitter API, the tweets are extracted and classified whether the tweets are bullied or not. Zhao et al. [14] proposed a new learning model to solve the issues in cyber-bullying detection—the proposed approach combined with a deep learning model to denoising auto-encoder (SDA). The SDA model contains semantic dropout noise and sparsity. These features focused on knowledge and the word embedding method. The performance of the proposed model improved by combining it with semantic-enhanced marginalized (SEM) to find the hidden features of the bullying content. The version of the proposed model was analyzed using two datasets such as Twitter and MySpace, and achieved better performance in terms of classification. Luo et al. [15] proposed the BiGRU-CNN for classifying cyber-bullying messages. BiGRU mainly focused on extracting the global features that significantly impact organizing bullying messages. The CNN consists of a convolution method with 128 kernels of length 5; this is used to extract the features that improve the learning rate of the model better. Adav et al. [16] introduced the BERT model, which is suitable for creating contextual embeddings that produce the particular embeddings for classifying cyber-bullying detection in social media. Ahmed et al. [17] introduced the cyber-bullying model that classifies the bullying words belonging to Bangla and Romanized Bangla texts utilizing ML and DL models. Iwendi et al. [18] performed various DL models that detect cyber-bullying in social media. The comparison between different DL algorithms shows high performance. Aind et al. [19] introduced the novel Q-Bully model that sees cyber-bullying automatically from social media platforms. The proposed performance is improved by combining with Reinforcement Learning gives better accuracy. Ketsbaia et al. [20] introduced the DL models that detect cyber-bullying automatically. Pradhan et al. [21] proposed the new DL model that sees the cyber-bullying from Wikipedia, Form spring, and Twitter cyber-bullying datasets.

III. HOW CYBER-BULLYING AFFECTS THE SOCIAL MEDIA

Cyber-bullying can have a significant impact on social media use, both for individuals and as a whole. Here are some of the ways it can affect social media introduction:

1) Fear of harassment: Cyber-bullying can create a climate of fear on social media platforms. Individuals who have been bullied in the past or who fear being bullied may be hesitant to join social media or may limit their use of these platforms.

2) Damage to reputation: Cyber-bullying can damage an individual's reputation, making them less likely to want to be active on social media. This can also affect the reputation of the platform itself if it becomes known as a place where bullying is rampant.
3) Decreased engagement: Cyber-bullying can lead to decreased engagement on social media, as individuals may avoid posting or interacting with others out of fear of being targeted. This can have a negative impact on the platform as a whole, as it relies on user engagement to generate revenue.

4) Reduced trust: If social media platforms are seen as a place where cyber-bullying is common, users may lose trust in these platforms and be less likely to use them. This can affect the growth and sustainability of social media as a whole. Overall, cyber-bullying can have a significant impact on social media use and adoption, and it is important for individuals and social media companies to take steps to prevent and address this issue.

5) Here are the equations for the fine-tuning process:

First, we add a classification layer on top of the pre-trained BERT model. The classification layer consists of a fully connected layer and a softmax activation function.

\[
h_{cls} = W_{cls} \times [CLS] + b_{cls} \quad (1)
\]

\[
y_{\text{hat}} = \text{softmax}(h_{cls}) \quad (2)
\]

Where \( h_{cls} \) is the hidden state of the [CLS] token, \( W_{cls} \) and \( b_{cls} \) are the weight and bias parameters of the fully connected layer, and \( y_{\text{hat}} \) is the predicted probability distribution over the two classes (cyber-bullying and non-cyber-bullying).

We then define the loss function as the cross-entropy loss between the predicted and true labels:

\[
L = -\sum y_i \times \log(y_{\text{hat}_i}) + (1 - y_i) \times \log(1 - y_{\text{hat}_i}) \quad (3)
\]

Where \( y_i \) is the true label (1 for cyber-bullying, 0 for non-cyber-bullying), and \( y_{\text{hat}_i} \) is the predicted probability for the \( i^{th} \) message.

We optimize the parameters of the classification layer by minimizing the loss function using gradient descent:

\[
W_{cls, b_{cls}} = \arg\min L \quad (4)
\]

The above equations outline the fine-tuning process for BERT in cyber-bullying detection. Note that this process requires a labelled dataset of cyber-bullying and non-cyber-bullying messages, as well as appropriate pre-processing and tokenization of the input text.

B. Pre-processing Techniques for Cyber-Bullying

1) Tokenization: Tokenization divides the text into smaller units known as tokens, which can be words, phrases, or symbols. It is a common step in pre-processing natural language processing (NLP) tasks such as sentiment analysis and topic modeling. Consider the following example sentence to demonstrate tokenization for cyber-bullying data:

- "I hate you and wish you were never born. You're worthless and nobody likes you."

To tokenize this sentence, we could use a straightforward method of separating the text by whitespace and punctuation marks.

- The tokenized sentence would look like this:

  \[ ["I", "hate", "you", "and", "wish", "you", "were", "never", "born", ",", "You're", "worthless", "and", "nobody", "likes", "you", ""] \]
Each element in the resulting list is a token that can be processed and analyzed further using various NLP techniques, Fig. 2.

2) Stop-word removal: Stop words are commonly used in a language but have little meaning and can be removed from text without affecting the overall message. Stop word removal can be used in cyber-bullying to filter out irrelevant or offensive words from the text to identify and prevent bullying behavior.

- Here's an example of how to use stop-word removal in the context of cyber-bullying:
- Assume a social media platform wants to look for instances of cyber-bullying in user posts.
- The platform could include a stop word filter that removes common words and phrases that are unlikely to be used in cyber-bullying incidents, such as "the," "and," "is," "in," "a," "of," and "on."
- For example, if a user writes, "I hate you, and I hope you die, the stop word filter will remove the words "I," "you," "and," "hope," and "die." The filtered text would then be "hate," raising a red flag and prompting a review by the platform's moderators.

- Stop word removal is a valuable tool for detecting cyber-bullying and promoting a safer online environment.
- However, it is essential to note that stop-word removal alone may not be enough to identify instances of cyber-bullying accurately and that other techniques, such as sentiment analysis and machine learning algorithms, may be required.

3) Feature extraction techniques for cyber-bullying:
Feature extraction is a crucial step in natural language processing (NLP) tasks such as cyber-bullying detection. Here are some techniques for feature extraction from cyber-bullying text data:

a) Bag of Words (BoW): It is a simple and effective method to extract features from text data. It involves counting the frequency of occurrence of each word in the document. For instance, consider the following sentence: "You're such a loser. Nobody likes you." The BoW representation of this sentence would be: {'you': 2, 're': 1, 'such': 1, 'a': 1, 'loser': 1, 'nobody': 1, 'likes': 1}.

b) TF-IDF (Term Frequency-Inverse Document Frequency): It is another technique that helps to extract features from text data. It assigns a weight to each word based on its frequency in the document and its frequency in the entire corpus. For example, in the sentence "You're such a loser. Nobody likes you," the word "you" appears twice in the document but is likely to appear in many other documents too. So, the weight assigned to "you" will be relatively low.

c) N-grams: N-grams are a sequence of N words in a sentence. For example, a bigram of the sentence "You're such a loser. Nobody likes you" would be "you're such," "such a," "a loser," "loser nobody," "nobody likes," and "likes you." N-grams help capture the context of words in a sentence.

d) Word embeddings: Word embeddings are vector representations of words that capture semantic and syntactic relationships between them. They are learned using neural networks trained on large amounts of text data. Word2Vec and GloVe are some examples of popular word embedding techniques.

Example:
- Let's say you have a dataset containing cyber-bullying text data, and you want to use these techniques to extract features from it. Here is an example of how you can use these techniques to extract features:
- Suppose you have a sentence in your dataset like this: "You are ugly and nobody likes you."
- BoW representation: {'you': 2, 'are': 1, 'ugly': 1, 'and': 1, 'nobody': 1, 'likes': 1}.
- TF-IDF representation: {'you': 0.276, 'are': 0.276, 'ugly': 0.385, 'and': 0.385, 'nobody': 0.385, 'likes': 0.385}. 
A. Convolutional Neural Networks (CNN) for Training on Images and Videos

CNNs are a type of deep learning model that are particularly effective at processing visual data, making them a popular choice for image and video classification tasks, including cyber bullying detection. The basic architecture of a CNN consists of several layers, including convolutional layers, pooling layers, and fully connected layers. Each layer performs a specific function, and the output of one layer is fed as input to the next layer.

The equations used to train a CNN for cyber bullying image and video classification involve the use of back-propagation and gradient descent to update the weights and biases of the network. The overall goal is to minimize the error between the predicted output and the actual output.

The general equation for computing the output of a convolutional layer can be expressed as follows:

$$y_i = f(\sum_j = 1 \hat{n} w_j \times x_{ij} + b_i)$$ (7)

Where:

- $y_i$ is the output of the $i^{th}$ neuron in the layer.
- $f()$ is the activation function.
- $n$ is the number of input neurons.
- $w_j$ is the weight connecting the $j^{th}$ input neuron to the $i^{th}$ output neuron.
- $x_{ij}$ is the activation of the $j^{th}$ input neuron at the $i^{th}$ location of the receptive field.
- $b_i$ is the bias term for the $i^{th}$ output neuron.

The pooling layer reduces the dimensionality of the input by aggregating nearby activations. The most common pooling operation is max pooling, which selects the maximum value from each local neighborhood of activations.

The fully connected layer takes the flattened output from the previous layer and applies a matrix multiplication operation to produce the final output. The equation for the fully connected layer can be expressed as:

$$y = f(Wx + b)$$ (8)

Where:

- $y$ is the output vector.
- $f()$ is the activation function.
- $W$ is the weight matrix.
- $x$ is the input vector.
- $b$ is the bias vector.

During training, the weights and biases of the network are updated using the back-propagation algorithm. The gradient of the loss function with respect to each weight and bias is computed, and the weights and biases are updated in the opposite direction of the gradient to minimize the loss function.
The equations for back-propagation and gradient descent are as follows:

\[
\frac{\text{d}L}{\text{d}w} = \frac{\text{d}L}{\text{d}y} \times \frac{\text{d}y}{\text{d}w} \\
\frac{\text{d}L}{\text{d}b} = \frac{\text{d}L}{\text{d}y} \times \frac{\text{d}y}{\text{d}b} \\
b = b - \text{l{}r} \times \frac{\text{d}L}{\text{d}b}
\]

(9), (10), (11), (12)

Where:

‘L’ is the loss function.

‘w’ and ‘b’ are the weights and biases of the network.

‘y’ is the output of the network.

‘lr’ is the learning rate.

Overall, the use of CNNs with back-propagation and gradient descent provides an effective way to train models for cyber bullying image and video classification tasks.

B. Cropping of Images and Videos

Cropping is the removal of unwanted parts of an image or video to focus on a specific area or subject.

Here are some typical image and video cropping techniques:

1) The rule of thirds: This technique entails dividing the image or video into thirds horizontally and vertically and then positioning the subject along the intersections or lines. As a result, the composition is more balanced and visually appealing.

2) Center crop: This technique involves cropping an image or video to center the subject. It works well when the issue is the main focus and there is no distracting background. Cropping an image or video to a specific aspect ratio, such as 4:3 or 16:9, is an example of this technique. It comes in handy when creating content for specific platforms or devices.

3) Pan and zoom: This technique involves cropping an image or video and animating it to simulate a camera pan or zoom. It can add motion to the image or video or emphasize specific parts.

4) Content-aware crop: This technique uses software tools to determine the best cropping based on the image or video content. It can be helpful when the subject is not in a fixed position or when complex background content needs to be removed.

C. Deep Neural Network (DNN)

DNN is specifically CNNs and recurrent neural networks (RNNs), can be used to solve the problem of classifying cyber-bullying in images and videos (RNNs).

D. Data Gathering and Pre-processing

Preprocess a large dataset of images and videos containing Cyber-bullying content to extract features like color, texture, shape, and motion.

E. Image Classification using CNN

Train a CNN on the image data to determine whether or not each image contains cyber-bullying content.

Multiple convolutional and pooling layers are followed by fully connected layers and a softmax output layer in a CNN. To improve the model's performance, employ data augmentation, dropout, and early stopping techniques.

F. Dataset Description

The experiments use a Python programming language with three datasets: Twitter, Images, and Videos dataset. The dataset is aged between 15-40 years from schools to job holders. Among these, 88% of data is analyzed as cyber-bullying. These tweets contain more than 47656 with two attributes such as tweet_text and tweet_type. Python libraries such as Keras, Pandas, and TensorFlow were used to analyze the performance of the proposed model. Table I shows the various types of bullying text messages for training and testing is given.

Table II shows the types of bullying images that affects the human personally and mentally. These images are JPG images with standard size. These images are classified based on comments, captions, and topics.

Table III shows the various types of videos belong to different categories. Three types of bullying videos are present for experimental analysis. These videos such as hate speech, personal abuse and normal are shown in Table IV.

TABLE I. TYPES OF BULLYING TEXT MESSAGES

<table>
<thead>
<tr>
<th>Types of Bullying</th>
<th>Tweets</th>
</tr>
</thead>
<tbody>
<tr>
<td>Age</td>
<td>The girl who bullied you in high school but now wants to sell you Arbonne</td>
</tr>
<tr>
<td>Ethnicity</td>
<td>I said dont put north west in coffee fuck the diddy call fifty i said there no to assassinate out the door of the air port dumb niggers</td>
</tr>
<tr>
<td>Gender</td>
<td>Don't call bitches females. That's mad disrespectful. Bitches hate when you call them females.</td>
</tr>
<tr>
<td>Religion</td>
<td>@UmarMal And I'm not sure how you can yammer about homelessness when Muslims are still murdering people for apostacy and blasphemy.</td>
</tr>
<tr>
<td>Other type of Cyber-bullying</td>
<td>@Eleoryth I sometimes envy those who don’t have retarded parents</td>
</tr>
<tr>
<td>Not Cyber-bullying</td>
<td>Rebecca Black Drops Out of School Due to Bullying</td>
</tr>
</tbody>
</table>

TABLE II. TYPES OF TWITTER TEXT DATASET

<table>
<thead>
<tr>
<th>Message Type</th>
<th>Training</th>
<th>Testing</th>
</tr>
</thead>
<tbody>
<tr>
<td>Religion based bullying</td>
<td>3000</td>
<td>4997</td>
</tr>
<tr>
<td>Age based bullying</td>
<td>3000</td>
<td>4992</td>
</tr>
<tr>
<td>Ethnicity</td>
<td>3000</td>
<td>4959</td>
</tr>
<tr>
<td>Gender</td>
<td>3000</td>
<td>4948</td>
</tr>
<tr>
<td>Other cyber-bullying</td>
<td>3000</td>
<td>4823</td>
</tr>
<tr>
<td>Not cyber-bullying</td>
<td>3000</td>
<td>4937</td>
</tr>
<tr>
<td>Total</td>
<td>18000</td>
<td>29656</td>
</tr>
</tbody>
</table>
TABLE III. TYPES OF IMAGE DATASET

<table>
<thead>
<tr>
<th>Image Type</th>
<th>Training</th>
<th>Testing</th>
</tr>
</thead>
<tbody>
<tr>
<td>Morphing Images</td>
<td>1500</td>
<td>1500</td>
</tr>
<tr>
<td>Personal Abuse</td>
<td>500</td>
<td>500</td>
</tr>
<tr>
<td>Adult</td>
<td>1k</td>
<td>1k</td>
</tr>
<tr>
<td>Total Messages</td>
<td>3k</td>
<td>3k</td>
</tr>
</tbody>
</table>

TABLE IV. VIDEOS DATASET

<table>
<thead>
<tr>
<th>Video Type</th>
<th>Training</th>
<th>Testing</th>
</tr>
</thead>
<tbody>
<tr>
<td>Hate speech</td>
<td>25</td>
<td>25</td>
</tr>
<tr>
<td>Personal Abuse</td>
<td>25</td>
<td>25</td>
</tr>
<tr>
<td>Normal Videos</td>
<td>10</td>
<td>10</td>
</tr>
<tr>
<td>Total Videos</td>
<td>60</td>
<td>60</td>
</tr>
</tbody>
</table>

G. Performance Metrics

A confusion matrix is an approach that analyzes the performance of the proposed model. The classification of images will specify the model performance on test data. The confusion matrix mainly focused on two attributes such as predicted and original values, see Fig. 3.

True Negative (TN): The predicted input is bullied and actual input is also bullied.

True Positive (TP): The predicted input is not bullied and actual value is not bullied.

False Positive (FP): The predicted input is bullied and actual input is not bullied.

False Negative (FN): The predicted input is not-bullied and actual input is bullied.

Precision: This parameter gives the overall correct outputs given by the proposed model.

\[
\text{Precision} = \frac{TP}{TP+FP} \quad (13)
\]

F1 Measure: It is the parameter that combines the recall and precision.

\[
\text{F1 Measure} = 2 \times \frac{\text{precision} \times \text{recall}}{\text{precision} + \text{recall}} \quad (14)
\]

Accuracy: The overall accuracy of proposed model is measured as

\[
\text{Accuracy} = \frac{TP+TN}{TP+TN+FP+FN} \quad (15)
\]

Recall: This metric is mainly focused on reducing the false negatives.

\[
\text{Recall} = \frac{TP}{TP+FN} \quad (16)
\]

![Fig. 3. Confusion matrix.](image)

TABLE V. COMPARATIVE PERFORMANCES OF EXISTING AND PROPOSED APPROACHES FOR ANALYSIS OF TWITTER DATA

<table>
<thead>
<tr>
<th></th>
<th></th>
<th></th>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>Religion based bullying</td>
<td>Religion based bullying</td>
<td>Religion based bullying</td>
</tr>
<tr>
<td></td>
<td>0.68</td>
<td>0.87</td>
<td>0.965</td>
</tr>
<tr>
<td></td>
<td>Age based bullying</td>
<td>Age based bullying</td>
<td>0.961</td>
</tr>
<tr>
<td></td>
<td>0.69</td>
<td>0.886</td>
<td>0.961</td>
</tr>
<tr>
<td></td>
<td>Ethnicity</td>
<td>Ethnicity</td>
<td>0.976</td>
</tr>
<tr>
<td></td>
<td>0.67</td>
<td>0.88</td>
<td>0.976</td>
</tr>
<tr>
<td></td>
<td>Gender</td>
<td>Gender</td>
<td>0.956</td>
</tr>
<tr>
<td></td>
<td>0.678</td>
<td>0.878</td>
<td>0.976</td>
</tr>
<tr>
<td></td>
<td>Other cyber-bullying</td>
<td>Other cyber-bullying</td>
<td>0.976</td>
</tr>
<tr>
<td></td>
<td>0.68</td>
<td>0.87</td>
<td>0.976</td>
</tr>
<tr>
<td></td>
<td>Not cyber-bullying</td>
<td>Not cyber-bullying</td>
<td>0.978</td>
</tr>
<tr>
<td></td>
<td>0.675</td>
<td>0.875</td>
<td>0.976</td>
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</tbody>
</table>

Precision, F1 Measure, Accuracy, and Recall for different categories of cyberbullying.
Fig. 4. Comparative performances of existing and proposed approaches for analysis of Twitter data.

Table V shows the comparative results among the existing 2DCNN [23], InceptionV3 [24] and proposed approach. Fig. 4 also compares text based cyber-bullying models based on the given data in twitter dataset.

Table VI, Table VII and Fig. 5, Fig. 6 shows a comparison of image-based cyberbullying. MoSI is the existing model, and Ensemble CNN is the proposed model. Ensemble CNN classifies images and videos of cyberbullying. They are creating a diverse and representative dataset of cyberbullying incidents, including various types such as harassment, hate speech, or threats, and creating deep learning model architecture suitable for detecting cyberbullying. The model should be trained and optimized for high performance using the collected dataset. They are addressing the issue of imbalanced data, as cyber-bullying incidents are relatively rare compared to non-cyber-bullying incidents. These techniques need methods like oversampling, under sampling, and generating synthetic data.

### TABLE VI. COMPARATIVE PERFORMANCES OF EXISTING AND PROPOSED APPROACHES FOR ANALYSIS OF IMAGES

<table>
<thead>
<tr>
<th>Types</th>
<th>Precision</th>
<th>F1 Measure</th>
<th>Accuracy</th>
<th>Recall</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Motion from Static Images (MoSI)</strong></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Morphing Images</td>
<td>0.78</td>
<td>0.78</td>
<td>0.78</td>
<td>0.78</td>
</tr>
<tr>
<td>Personal Abuse</td>
<td>0.79</td>
<td>0.698</td>
<td>0.78</td>
<td>0.79</td>
</tr>
<tr>
<td>Adult</td>
<td>0.77</td>
<td>0.78</td>
<td>0.78</td>
<td>0.77</td>
</tr>
<tr>
<td>Normal</td>
<td>0.77</td>
<td>0.78</td>
<td>0.78</td>
<td>0.78</td>
</tr>
<tr>
<td><strong>Ensemble CNN</strong></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Morphing Images</td>
<td>0.88</td>
<td>0.88</td>
<td>0.88</td>
<td>0.88</td>
</tr>
<tr>
<td>Personal Abuse</td>
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<td>0.898</td>
<td>0.88</td>
<td>0.89</td>
</tr>
<tr>
<td>Adult</td>
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<td>0.88</td>
<td>0.88</td>
<td>0.87</td>
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<tr>
<td>Normal</td>
<td>0.8856</td>
<td>0.879</td>
<td>0.8876</td>
<td>0.876</td>
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</table>

Table VI. Comparative performances of existing and proposed approaches for analysis of image data.

### TABLE VII. COMPARATIVE PERFORMANCES OF EXISTING AND PROPOSED APPROACHES FOR ANALYSIS OF VIDEOS

<table>
<thead>
<tr>
<th>Types</th>
<th>Precision</th>
<th>F1 Measure</th>
<th>Accuracy</th>
<th>Recall</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Motion from Static Images (MoSI)</strong></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Hate speech</td>
<td>0.68</td>
<td>0.78</td>
<td>0.78</td>
<td>0.78</td>
</tr>
<tr>
<td>Personal Abuse</td>
<td>0.69</td>
<td>0.698</td>
<td>0.78</td>
<td>0.79</td>
</tr>
<tr>
<td>Normal</td>
<td>0.67</td>
<td>0.78</td>
<td>0.78</td>
<td>0.77</td>
</tr>
<tr>
<td><strong>Ensemble CNN</strong></td>
<td></td>
<td></td>
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<td></td>
</tr>
<tr>
<td>Hate speech</td>
<td>0.80</td>
<td>0.81</td>
<td>0.81</td>
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<tr>
<td>Personal Abuse</td>
<td>0.80</td>
<td>0.821</td>
<td>0.82</td>
<td>0.81</td>
</tr>
<tr>
<td>Normal</td>
<td>0.81</td>
<td>0.81</td>
<td>0.81</td>
<td>0.82</td>
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</tbody>
</table>
VI. CONCLUSION

Finally, an ensemble deep neural network can classify cyber-bullying on Twitter using text, images, and videos. The ensemble method combines multiple models' outputs to produce a more accurate and robust prediction. An Aspect-based Sentiment Analysis (ABSA) for Cyber-bullying is introduced for text classification. This model learned the patterns and features that distinguish cyber-bullying tweets from non-cyber-bullying tweets using large datasets of labeled text. The text model's output can then be combined with the results of the image and video models via a weighted voting scheme. A deep neural network, such as a recurrent neural network (RNN) or a convolutional neural network (CNN), can classify images and videos. These models trained on large datasets of labeled images and videos to learn the features and patterns that differentiate cyber-bullying from non-cyber-bullying content. A weighted voting scheme is used for the output of the image and video models that can be combined with the output of the text model. Because different models can capture various aspects of the data, using an ensemble deep neural network allows for a more accurate and robust classification of cyber-bullying on Twitter. The ensemble model is better equipped to handle the complexity and variability of cyber-bullying content on Twitter because their outputs are combined. Overall, using an ensemble deep neural network is a promising approach for addressing the problem of cyber-bullying on Twitter and can contribute to creating a safer and more positive online environment.

REFERENCES


A Novel Software Quality Characteristic Recommendation Model to Handle the Dynamic Requirements of Software Projects that Improves Service Quality and Cost

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Devi Ahilya Vishwavidyalaya, Indore, India¹
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Abstract—The software is created and constructed to address particular issues in the applied field. In this context, there is a need to be aware of the crucial characteristics to assess the quality of software. But not all software requires checking all the quality-of-service parameters, resulting in effort loss and time consumption. Therefore, it is required to develop software quality characteristics recommendation model to address and resolve the issue. The proposed work involved in this paper can be subdivided into three main parts (1) a review of popular software quality models and their comparison to create a complete set of predictable, and (2) the design of an ML-based recommendation model for recommending the software quality model and software quality characteristics (3) performance analysis. The proposed recommendation system utilizes the different software quality of service attributes as well as the software attributes where these models are suitably applied to satisfy the demands. Profiling of applications and their essential requirements have been performed Based on the different quality of service parameters and the requirements of applications. These profiles are learned by machine learning algorithms for distinguishing the application-based requirement and recommending the essential attributes. The implementation of the proposed technique has been done using Python technology. The simulation aims to demonstrate how to minimize the cost of software testing and improve time and accuracy by utilizing the appropriate quality matrix. Finally, a conclusion has been drawn and the future extension of the proposed model has been reported.

Keywords—Recommendation system; software quality model; ML (Machine Learning); quality matrix; software quality characteristics

I. INTRODUCTION

Machine learning provides ease in several real-world applications; in addition, improves the capacity and capability of existing research and methodologies. ML techniques are also used to improve and optimize different process models for improving the cost of employment and productivity[1][2]. In this context, software quality evaluation is one of the essential steps. There are several different software quality measuring models currently utilized. These models include several different quality measuring characteristics.

Software quality is an emerging research area in the field of software engineering. The work presented here is relevant to the research around software quality models which gives a better understanding and knowledge of software quality attributes in Software quality models. Achieving software quality assurance requires the use of software quality models [3]. These quality attributes might be used to describe the software's quality. It might be difficult to decide which of the excellent models to utilize [4]. In addition, software quality models are used for the global assessment of the software product. Therefore, the proposed issue of applying and selecting the appropriate quality matrix is defined here as the recommendation problem. The recommendation engines are the machine learning technique for evaluating the problem's current scenario and suggesting the most suitable solutions for the given set of problems [5].

There are different kinds of recommendation systems available, which will also work as the information filter to reduce the less relevant data and optimize the ranking of the desired set of information [6][7]. The proposed software quality characteristics recommendation model includes the technique of machine learning to learn when, where, and which software characteristic is appropriate for evaluation based on profiling of the software quality of service requirements. In this context, the proposed work is subdivided into the following essential task:

- Examination of different software quality characteristics and models.
- Implement and design a content-based recommendation model to suggest the appropriate quality matrix.
- Study the impact of the software quality characteristic recommendation model over the existing models.
In this section, an outline of the proposed concept for the software quality measuring model is provided. The next section involves the study of different software quality estimation models. Further, the proposed recommendation model has been formulated and its implementation plan will be explained. In addition, based on the implemented model simulations have been carried out and their performance will be reported. Finally, the conclusion has been reached and their future extension plan will be proposed.

The paper is organized as follows: Section II reviews software quality models based on basic and tailored criteria which give a solid foundation in attribute selections. Section III briefly introduces some essential attributes and a proposed software quality characteristics recommendation model is introduced which uses machine learning for recommending the appropriate characteristics for software quality estimation. Section IV describes the used Machine learning algorithms in the proposed model. Section V gives an analysis and validation of the results. Finally, this work is concluded in Section VI where possible future research directions are indicated.

II. REVIEW OF LITERATURE

In this study, two types of software quality models are considered. The first one is based on basic software quality models and the second one is based on the tailored software quality models which are described in the below section.

A. Basic Software Quality Models

Each of the several software quality models consists of a number of different attributes. Basic quality models denote those models which were developed until 2000. This study needs to explore some essential software quality models and each model has various qualities.

1) McCall’s quality model: It exposes three task domains.

a) Operation in a product refers to its capacity to be easily understood, and able to deliver the desired results. It addresses criteria for correctness, dependability, effectiveness, integrity, and usability.

b) Revision in a product is the ability to endure modifications, error rectification, and system adaptation. It includes testability criteria, maintainability, and flexibility.

c) When a product is in a transition phase, it means it can accommodate distributed processing in new environments with rapidly changing technology.

The goal was on the relationship between metrics and quality characteristics [3]. The problem is that it is dependent on Yes and No responses, there is no accuracy in the results.

2) Boehm’s quality model: It adds maintainability to McCall’s model [8]. High-level factors are as follows:

a) Utility describes efficient, reliable, and easy to use.

b) Maintainability describes the ability to modify, testable, and features of understanding.

3) Dromey’s quality model: Three models have been given by Dromey, i.e., the Requirement model, the design model, and the implementation model. The product properties are given below:

a) Correctness assesses if certain principles are broken, along with usability and reliability.

b) Measures the effectiveness of a component's deployment in terms of usability, maintainability, efficiency, and reliability.

c) Descriptive evaluates the description of a component, about maintainability, reusability, portability, and usability.

d) Despite the design quality model considering the development process, architectural integrity is not fully attention Testability is implicitly included. None of the domain-specific properties are discussed. Furthermore, one drawback of the said model is allied with reliability and maintainability, as judging them before the software system is functioning is not practical. [8] [9]

4) FURPS quality model: The elements of the FURPS model that are considered [8] [9] are:

a) Functionality encompasses capability sets, security, and feature sets.

b) Usability includes stability in the user interface view, help (online), user documentation, and materials required in training.

c) Reliability focuses on the mean time between failures (MTBF), frequency and strictness of the failure, accuracy, recoverability, and predictability.

d) Functional needs like efficiency, speed, availability, throughput, accuracy, resource utilization, reaction time, recovery time, and are constrained by performance.

e) Testability, extensibility, adaptability, maintainability, and compatibility are all aspects of supportability. Its failure to consider software portability is one drawback. The model does not include any attributes that are domain specific.

5) ISO 9126 quality model: The McCall and Boehm models served as the basis for the ISO model [10][11][12]. It works on four parts quality Model, quality in use metrics, internal quality attribute, and external quality attribute. The attention of that model is to an exploration of attributes into 6 independent characteristics which are reliability, usability, efficiency, functionality, maintainability, and portability. Now attributes are further split into internal quality attributes, which refer to system features that can be assessed without incorporating, and external quality attributes, which refer to assessment by observation while it is being carried out. [13]. This model addresses effectiveness, security, and satisfaction [14].

6) ISO 25010 quality model: The modernized version of ISO 9126 is ISO 25010. This approach divides quality into eight smaller sub-characteristics. The ISO-9126 Model serves as the sole foundation for the set of standards. The model adds new features including compatibility and
security. As an extension of portability, it employs the term transferability while conducting its operation.

B. Tailored Software Quality Models

Tailored software quality models were built from the fundamental (basic) software quality models. This model was made with certain individual components. There are various tailored software quality models that are there, and selected tailored software quality models are presented below.

1) BERTOA model: It defines the quality attributes for the assessment of Commercial Off-The-Shelf Components. The application of the model is to build Complex software.

2) GEQUAMO model: The breakdown of the sub-layers in this model, known as GEQUAMO (Generic, Multi-layered, and Customizable Model) [15], allows for the flexible inclusion of user requirements. End users can create their models with the aid of that model.

3) ALVARO model: The methodology used in the Alvaro model is essentially used to certify software components and identify quality components. The model involves a framework that may be divided into four sections: components related to the model quality, framework for technical certification, certification process, and framework with metrics. For quality assessment and technical certification, all components are utilized.

4) RAWASHDEH model: Rawashdeh’s model is conquered by the Dromey and ISO 9126[16] models. It addresses the genuine requirements of various users. To produce high-quality products four processes are suggested by the said model.

a) Selecting a limited group of quality characteristics, applied using a top-down method, dividing each characteristic into several subordinate characteristics.

b) Examine how internal and external measures differ. This includes characteristics such as requirements or lines of code, as well as external metrics, behavior during testing procedures, and components.

c) Quality attributes for each user must be identified.

d) Any new quality model can be built from ISO 9126, and the Dromey model.

In addition to the above software models S. S. Kamaruddin et al [6] provide a feature subset selection approach to choose the right attributes for software quality assessment to address this dynamic software quality assessment problem. The current models for evaluating the quality of software do not permit dynamic assessment. As new quality attributes surface, they can be incorporated into the model in dynamic software quality evaluation. To establish dynamic software quality evaluation, they concentrated on creating an intelligent technology that can learn and include new quality criteria into the model. Additionally, S. S. Kamaruddin et al [7] introduce a filter-wrapper-based feature ranking method that can learn from and order quality attributes depending on fresh data from software quality assessment instances. The Most Priority of Feature (MPF) score method and the software quality attribute weights learning algorithm make up the suggested feature ranking strategy. The issue of repetition in the rankings of the software quality attribute is not addressed by the present ranking methodologies. The redundancy problem is solved by the suggested method by selecting characteristics with good classification accuracy utilizing classifiers.

III. PROPOSED METHODOLOGY

In this section, a proposed software quality characteristics recommendation model is introduced which uses machine learning for recommending the appropriate characteristics for software quality estimation. The proposed model is aimed at reducing the quality estimation time overhead. But according to the available different quality estimation matrices as discussed in the above section, it has been observed some essential characteristics which are followed by entire models these characteristics are described in Table I.

| TABLE I. PROPERTIES OF BASIC AND TAILORED SOFTWARE QUALITY MODELS |
|-----------------------------------|-----------------|---------------|
| Properties                        | Tailored models | Basic models  |
| Functionality                     | Yes             | No            |
| Maturity                          | Yes             | No            |
| Resource-utilisation              | Yes             | No            |
| Testability                       | Yes             | No            |
| Compliance                        | Yes             | No            |
| Understandability                 | Yes             | No            |
| Usability                         | Yes             | No            |
| Learnability                      | Yes             | No            |
| Reliability                       | No              | Yes           |

Therefore, the characteristics available in Table I have been included in our quality matrix. In addition, based on the requirements a list of questions is also included that help to decide the additional quality characteristics requirements.

Key Questions have been prepared with the consultation of IBM India Pvt. Ltd.

Q1. Is the model involving any calculations?

If the software is being developed for performing any calculation, then the following characteristics need to be included:

1. Accuracy
2. Correctness
3. Efficiency

Q2. Is the model involve security, privacy, and communication modules?

If the software involves communication and data security, then the following properties need to be considered.

1. Integrity
2. Fault Tolerance
3. Time Behaviour

Q3. Is the software involved in data collection and analysis?
   If yes, then the following characteristics must involve:
   1. Human Engg.
   2. Analysability

Q4. Is the software utilized by a person who has a new or non-technical background?
   If yes, then need to consider the followings:
   1. Recoverability
   2. Suitability
   3. Attractiveness
   4. Operability

Q5. Is the software needed to change, modify, or scale shortly?
   If yes, then need to consider the followings:
   1. Adaptability
   2. Changeability
   3. Flexibility
   4. Modifiability
   5. Reusability
   6. Operability
   7. Suitability

Q6. Does the software need to deploy in multiple places/multiple machines/multiple clients with the same or different configurations?
   If yes, then need to consider the followings:
   1. Flexibility
   2. Installability
   3. Maintainability
   4. Portability
   5. Transferability
   6. Configurability
   7. Compatibility
   8. Reusability
   9. Interoperability

Q7. Is software deployed in resource-constrained scenarios?
   If yes, then the followings need to include:
   1. Stability
   2. Resource Utilisation
   3. Self Contained
   4. Replaceability
   5. Manageability

Q8. Is software having many modules which require assistance?
   If yes, then the followings need to include:
   1) Supportability: To decide the suitable quality of service the proposed model has been demonstrated in Fig. 1. The flowchart of the proposed software quality characteristics recommendation model is presented here in Fig. 2. Additionally, the key components are described in Table I. The proposed model accepts two inputs, namely the project source code and the prepared questionnaire. These questionnaires are prepared based on the activities involved in the project development life cycle. Based on these questionnaires the dataset has been prepared. The Highlight of the dataset is defined in Table II.

   The given Table II provides a limited number of dataset instances but provides a structure of the dataset which is used for training. By using a similar method, prepared a total of \(2^{n-1}+1 = 2^{8-1} + 1 = 129\) instances. In this situation for making training with a supervised machine learning algorithm, it is required to decide the class labels of these instances. To calculate class labels of instances, let a project quality requirement can be satisfied with an initial set of software quality characteristics as given in Table I.

   This table is denoted as A1. Additionally, the answers to the questions can be denoted as:
   
   \[ A_n = \{ A1, A2, \ldots, A8 \} \quad (1) \]

   Each true answer to the questions includes a set of quality characteristics, where A1 is always constant and A1 = True (T). Therefore, the set of attributes for a unique class label can be calculated using:

   \[ C = A1 \cup A_i \quad (2) \]

   Where Ai is the set of characteristics where i’th question’s answer is true.

   Some examples of unique class calculations have been given in Table II. Additionally, the dataset has a similar number of classes to predict. The meaning of the class label is defined in Table III.
Fig. 1. Flow chart of proposed software quality characteristics recommendation model.

Fig. 2. Support Vector Machine (SVM) classifier diagram.
TABLE II. EXAMPLE OF DATASET USED

<table>
<thead>
<tr>
<th>A1</th>
<th>Q1</th>
<th>Q2</th>
<th>Q3</th>
<th>Q4</th>
<th>Q5</th>
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<td>F</td>
<td>F</td>
<td>F</td>
<td>F</td>
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<td>F</td>
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<td>F</td>
<td>F</td>
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<td>T</td>
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</tbody>
</table>

TABLE III. EXAMPLES OF CLASS LABEL SEQUENCES

<table>
<thead>
<tr>
<th>Class</th>
<th>Means</th>
</tr>
</thead>
<tbody>
<tr>
<td>C1</td>
<td>A1</td>
</tr>
<tr>
<td>C2</td>
<td>A1 ∪ A1 if A1 = true</td>
</tr>
<tr>
<td>C3</td>
<td>A1 ∪ A2 if A2 = true</td>
</tr>
</tbody>
</table>

The mapping for each class label is prepared in the dataset. In the proposed model, the name given for this process is the profiling of historical projects. Now, it is needed to train machine learning algorithms on the prepared profiles. In this context, five widespread ML algorithms have been used such as K nearest neighbor (KNN), Support vector machine (SVM), artificial neural network (ANN), C4.5 decision tree, and Bayesian.

IV. USED ML ALGORITHMS

A. Support Vector Machine (SVM)

One or more hyperplanes are created by the SVM classifier for regression and classification [17]. The line is used to divide the 2D linearly separable data. The line function is:

\[ y = ax + b \]  \hspace{1cm} (3)

Here in Equation (3) variable x is renamed with \( x_1 \) and y is renamed with \( x_2 \) then the line Equation can be defined as:

\[ ax_1 - x_2 + b = 0 \]  \hspace{1cm} (4)

Now, if defining \( w = (a, -1) \), then:

\[ w \cdot x + b = 0 \]  \hspace{1cm} (5)

This is the hyperplane equation. After obtaining the hyperplane, it can be used to create predictions. Hypothesis function \( h \) is called a Hypothesis function shown in below Equation (6).

\[ h(x_i) = \begin{cases} +1 & \text{if } w \cdot x + b \geq 0 \\ -1 & \text{if } w \cdot x + b \leq 0 \end{cases} \]  \hspace{1cm} (6)

In addition to this SVM, the classifier can also be explored by the below flowchart.

The below Fig. 3 explores the support vector machine and talks about classification and regression. With the help of the SVM algorithm, we may swiftly categorize new data points in the future by determining the optimal line or decision border of n-dimensional space. A hyperplane is the name given to this optimal decision boundary.

**Fig. 3.** Proposed software quality characteristic recommendation model.

B. Naive Bayes

A probabilistic classifier is the Naive Bayes classification [18]. The Bayes theorem can be used to derive this. Need to train the Naive Bayes algorithm as supervised learning using observations from nature. Two types of probabilities are given below: Posterior Probability \( P(H/X) \) and Prior Probability \( P(H) \). Where H is an assumption and X is data. Thus, Baye's Theorem stated: In the below Fig. 4 explore the flow chart of Naive Bayes classification in which every iteration according to the probability value of attributes is updated.

\[ P(A/B) = \frac{P(B/A)P(A)}{P(B)} \]  \hspace{1cm} (7)

C. K-Nearest Neighbor Classification

The KNN is a traditional tool for both classification and prediction applications [19]. It is a lazy learning classifier. The KNN method has three key parts. First, it calculates the distance between the sample under consideration and each training sample. To calculate the distances mostly Euclidean distance will be used. Euclidean is described by:

\[ d(q, f) = \sqrt{\sum_{i=1}^{N}(q_i - f_i)^2} \]  \hspace{1cm} (8)

Where \( q \) is the query vector and \( p \) is the dataset samples.
D. Artificial Neural Network (ANN)

In processes where complicated multivariable, non-direct relationships between information and yield factors are present, ANN is becoming a popular showing tool [20].

ANN is primarily a data handling discipline, with its underpinnings in the operational principles of organic neural systems. It is like our cerebrum, which gets the data, translates, and gives the yield. Many preparation elements are known as hubs or neurons make up an ANN. These components are highly coupled with one another and serve as a system to produce standard results. The new features of ANN include its ability to understand instances and infer conclusions from precise information. Additionally, the speed and precision levels are unmatched. The charming element of ANN is no past information or science behind the procedure is required and thus they are alluded to as discovery displays. Also, ANN models can go up against every single semantic variable or parameter that can't be estimated, and ordinary demonstrating techniques are in this manner unacceptable and may neglect to give reasons. Neural networks could classify patterns for which they have not been taught, as well as a high tolerance for noisy input. The neural network's internal weights, which are applied by the transactions employed during the learning process, are a primary issue of the training phase. The predicted output is added to each training transaction for the neural network. The concept of ANN can be visualized below in Fig. 6:

E. C4.5 Decision Tree

The program must produce decision rules. This is an expansion of the well-known ID3 decision tree [21]. To obtain data partition, entropy as well as information gain is required in the program. Additionally, the highest information gain attribute is selected to create a tree. Partitioned sub-lists are used by the C4.5 method to shape a full decision tree. The algorithm considers the restrictions of the next set. If samples in the dataset cover a similar class, then it forms a leaf node as a decision tree.

1) If the computation of IG (information gain) is not attainable then it creates a node higher up, then the expected value of the class is used by the tree.

2) If a previously undetected class is met, the decision node has been created from a target value.
Before presenting the decision tree’s phases, it is necessary to acknowledge the information acquired. Thus, compelled to discuss entropy first, considering that the aftereffect decision tree classifies data into two classes, i.e., P (+ve) and N (-ve). The binary classification of entropy S is given by E(S):

\[ E(S) = -P(+ve) \log_2 P(+ve) - P(-ve) \log_2 P(-ve) \]  \hspace{1cm} (9)

The above Figure 7 is exploring the step-by-step process of the C4.5 decision tree algorithm. The best potential characteristic to separate tree branches must be chosen to reduce the depth of the tree when traversing it. It can be seen that the best choice will be the property with the least entropy. As a necessary decrease in entropy in connection to each characteristic during splitting, the information gain can be described. The IG (information-gain) calculates, Gain (E, A) of an attribute A using equation number (10).

\[ \text{Gain}(E, A) = \text{Entropy}(s) - \sum_{n=1}^{v} \frac{E_n}{E} \times \text{Entropy}(E_n) \]  \hspace{1cm} (10)

The system accepts the ongoing project activities-based questions after the algorithm has been trained with the profiled attributes of the projects. Based on the trained machine learning and the input current project questioners the model predicts the most suitable characteristics for the project.

V. EXPERIMENTS AND RESULT ANALYSIS

Three datasets of the machine learning algorithm’s classification problem are used in this experiment. Here the dataset obtained from Kaggle [22] is denoted as Dataset 1, the dataset obtained from GitHub [23] is given as Dataset 2 and the prepared dataset is denoted as Dataset 3. The precision, recall, and f-score are calculated based on class wise, and then the mean of the performance is calculated. The performance of the model is also reported using the bar graph in Fig. 8. Fig. 8(A) demonstrates the precision of the proposed model, 8(B) shows the recall, 8(C) shows the f-score and finally Fig. 8(D) shows the accuracy of the presented model. The performance demonstrates the comparison of five different machine learning algorithms for predicting the quality characteristics as a recommendation of the quality characteristics. According to the obtained performance of the model, support vector machine (SVM) and artificial neural network (ANN) is providing higher performance as compared to C4.5, KNN, and Bayesian. But the Support vector machine has a higher amount of time complexity. Thus, it is suggested to be utilizing the ANN as the final machine learning algorithm which is suitably worked with the proposed recommendation system.
VI. CONCLUSION AND FUTURE WORK

Concerning the criteria of the project, the presented investigation aims to produce an intelligent model that offers software quality characteristics. These recommendations are based on the involved activities in the project. These activities are summarized as questionnaires during different software development process life cycles. The project source code and documentation are utilized with the proposed recommendation system to analyze. The past project’s quality assurance characteristics were utilized to develop an ML model to predict the desired attributes. The ML algorithm needs to train with a limited set of sequences for predicting the possible attributes to evaluate the software. Here, the suggested model has been evaluated using five machine-learning methods across three distinct datasets. Accuracy, F-score, precision, and recall are used to evaluate performance. A suggested model demonstrates higher prediction accuracy which means that model successfully be able to serve as the software quality characteristics recommendation model. Additionally, the model is also helpful to lower expenses and increase the revenue of software development companies.

Soon the following future extension has been proposed for investigation:

1) Apply real-world data to evaluate the performance of the proposed model.
2) Prepare detailed questionnaires for each stage of software development which influences the characteristics of the software quality testing matrix.
3) To enhance the model performance, can apply deep learning techniques.

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Enhancing Facemask Detection using Deep learning Models

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Abstract—Face detection and mask detection are critical tasks in the context of public safety and compliance with mask-wearing protocols. Hence, it is important to track down whoever violated rules and regulations. Therefore, this paper aims to implement four deep learning models for face detection and face with mask detection: MobileNet, ResNet50, Inceptionv3, and VGG19. The models are evaluated based on precision and recall metrics for both face detection and face with mask detection tasks. The results indicate that the proposed model based on ResNet50 achieves superior performance in face detection, demonstrating high precision (99.4%) and recall (98.6%) values. Additionally, the proposed model shows commendable accuracy in mask detection. MobileNet and Inceptionv3 provide satisfactory results, while the proposed model based on VGG19 excels in face detection but shows slightly lower performance in mask detection. The findings contribute to the development of effective face mask detection systems, with implications for public safety.

Keywords—Object detection; deep learning; detection; face detection; mask detection; convolutional neural network

I. INTRODUCTION

Computer vision is a rapidly advancing field that encompasses a wide range of technologies aimed at enabling machines to perceive and interpret visual information, similar to how humans do. One crucial task within computer vision is face detection, which involves locating and identifying human faces in digital images or video streams. Face detection has gained significant attention and importance due to its wide-ranging applications in various domains, including surveillance systems, biometric authentication, facial recognition, human-computer interaction, and social media analysis [1]. Over the years, researchers have made remarkable progress in developing sophisticated face detection algorithms that exhibit high accuracy and robustness. Despite the progress made in general target detection algorithms across various domains, the efficacy of face mask detection techniques remains constrained [2]. In response, researchers have directed their efforts towards this area, employing the "you only look once v2" (YOLOv2) algorithm to devise detection models. Furthermore, advancements have been made by leveraging the YOLOv3 algorithm, which facilitates enhanced feature extraction through an optimized way [3].

However, these challenges arise due to variations in lighting conditions, occlusions, pose variations, complex backgrounds, and scale variations. Lighting variations can lead to significant changes in facial appearance, making it challenging to detect faces consistently. Occlusions, such as glasses, facial hair, or partial face obstructions, further complicate the task by hiding crucial facial features. Additionally, face detection algorithms must handle pose variations, where faces may be rotated, tilted, or viewed from different angles. Complex backgrounds with cluttered scenes pose another challenge, as it becomes difficult to differentiate faces from the surrounding environment [4]. Similarly, scale variations, caused by the varying distances between the camera and the subjects, necessitate robust face detection algorithms that can handle faces of different sizes are required [5].

Over the years, researchers have proposed various face detection techniques, each aiming to address the challenges mentioned above and improve the accuracy and efficiency of face detection algorithms. Early approaches utilized handcrafted features and traditional machine-learning algorithms, such as Haar cascades and Histogram of Oriented-Gradients (HOG), to detect faces. These methods achieved reasonable results but had limitations in handling pose variations and complex backgrounds. In recent years, the advent of deep learning, particularly convolutional neural networks (CNNs), has revolutionized the field of face detection. CNN-based architectures, such as the Viola-Jones framework, Single Shot MultiBox Detector (SSD), and Faster R-CNN, have demonstrated superior performance in face detection tasks. These models leverage the power of deep learning to automatically learn discriminative features from large-scale datasets, enabling them to handle various challenges faced in face detection. Notably, the use of region-based convolutional neural networks (R-CNN) has greatly improved accuracy by combining region proposals and convolutional networks, allowing for more precise localization of faces [6].

Other researcher improved the YOLOva and made that the YOLO-network generates predictions for bounding boxes in each grid of an image with a size of GxG pixels. However, the network encounters challenges in detecting smaller objects since each bounding box can only be assigned a single class during prediction. The primary issue with YOLO arises from its limitations in accurately localizing objects, particularly when dealing with bounding boxes of unusual ratios [7]. On the other hand, in the realm of face mask detection, various transfer learning approaches have been employed to address the challenges encountered in real-world scenarios. One such method involves utilizing a pre-trained InceptionV3 model as a transfer learning technique to discern individuals wearing or
not wearing masks [3]. For instance, the author in [1] employed a cascading approach using Convolutional Neural Networks (CNNs) to detect faces that are covered with masks. In recent advancements, a dedicated framework called the Retina Face Mask network has been developed to enable accurate and efficient recognition of face masks. Various experiments have been conducted to devise an automated technique for determining whether an individual is wearing a face mask or not. Real-time detection of facial masks has been achieved using the YOLOv3-technique and the Haar-cascading-classifier, but the challenges are remains constrained [3].

Although face detection and recognition, particularly in the presence of masks, pose significant challenges and have been the subject of extensive research in recent years, this research aims to improve the detection performance of masked and unmasked faces, with a specific focus on face masks. The problem of face mask-detection in the fields of image-processing and computer vision is exceptionally complex. The primary objective of this study is to enhance public safety by employing deep learning techniques to identify individuals wearing or not wearing masks in public areas. The developed mask detector can play a crucial role in ensuring our protection. Additionally, witnessing the global impact of the COVID-19 pandemic further motivates the exploration of machine learning techniques to address the real-world problem of habitual mask-wearing when venturing outdoors. Hence, the proposed approach in this study utilizes transfer learning, specifically applying the pre-trained MobileNetV2, RestNet50, InceptionV3 and VGG19 model for fine-tuning the face mask detection task.

II. RELATED WORK

Face detection is a fundamental task in computer vision that involves locating and identifying human faces in digital images or video streams. Over the years, researchers have made significant advancements in developing accurate and robust face detection algorithms [8]. Early approaches in face detection primarily relied on handcrafted features and traditional machine learning algorithms. Author [9] proposed one of the seminal methods, known as the Viola-Jones framework, which utilized Haar-like features and a cascade classifier for rapid face detection. This approach laid the foundation for subsequent advancements in face detection. However, these methods had limitations in handling pose variations, occlusions, and complex backgrounds. The advent of deep learning, particularly convolutional neural networks (CNNs), revolutionized face detection. Researchers explored various CNN architectures for accurate and robust face detection. For instance, author [10] proposed the Single Shot MultiBox Detector (SSD), which combined a deep CNN with a set of anchor boxes for efficient face detection. This approach achieved excellent performance in Region-based convolutional neural networks (R-CNNs) further improved the accuracy of face detection. Author [11] introduced the Faster R-CNN framework, which integrated a region proposal network with a CNN-based object detection network. This method enabled precise localization of faces and achieved state-of-the-art performance in face detection tasks. Subsequent research efforts focused on enhancing the speed and efficiency of R-CNN-based methods, leading to variants like Fast R-CNN. Despite the advancements in face detection techniques, several challenges persist. Variations in lighting conditions pose a significant challenge as they can affect the appearance of faces [12].

To address this, author [13] proposed an illumination-robust face detection method that utilized color normalization and multiple thresholds to handle lighting variations. Occlusions, such as glasses, facial hair, or partial face obstructions, present another challenge. Author [14] introduced a method that employed a deformable part model to handle occlusions and achieve accurate face detection. Pose variations also pose challenges, as faces may be rotated, tilted, or viewed from different angles. Author [15] proposed a pose-aware face detection approach that utilized pose estimations to improve detection accuracy.

Meanwhile, complex backgrounds with cluttered scenes make it difficult to differentiate faces from the surrounding environment. Author [8] addressed this challenge by proposing a context-aware face detection method that leveraged contextual cues to enhance the accuracy of face detection in complex scenes. Additionally, scale variations, caused by the varying distances between the camera and the subjects, require robust face detection algorithms. Similarly, author [16] proposed a scale-aware face detection method that utilized a multi-scale convolutional network to handle faces at different sizes and face localization. In face localization, the goal is to figure out where and how big a certain number of faces are (usually one). In general, there are two ways to find facial parts in a digital image: the feature-based approach and the image-based approach. The feature-based approach tries to pull out parts of the image and compare them to what is known about the face. While image-based methods try to find the best match between the images used for training and the ones used for testing. People often use the following ways to find faces in a still image or a video sequence:

A. Feature based Approaches

Feature-based approaches for face mask detection leverage the distinctive visual characteristics and patterns associated with the presence or absence of masks. These methods primarily rely on handcrafted features and machine learning algorithms to classify faces as masked or unmasked. Some commonly used for features approaches include active shape model and Low level model [11]. And it can be classified by active shape and low level analysis.

The Active Shape Model (ASM) focuses on intricate, flexible aspects, including how features appear and behave. Finding landmark points in a picture that determine the form of any statistically modelled object is the primary objective of ASM. For instance, the eyes, lips, nose, mouth, and eyebrows were removed from a photograph of a person's face. An ASM's statistical face model is created using photos that include hand-marked landmarks during the training process. Three categories of ASMs are distinguished: templates that may alter form, point distribution models (PDMs), and snakes. The first form to use an active contour is known as a snake. To indicate the margins of the head, snakes are employed. A snake must first be positioned near to a head barrier in order to finish the
mission. It then changes into the form of a head after scanning the surrounding edges. Snakes develop by reducing the "esnake" energy function, which is similar to how physical systems operate [17].

Internal energy is the component that derives from the snake's own characteristics and demonstrates how it has evolved organically. Snakes often undergo change by either contracting or growing. The contours might diverge from their normal development and finally take on the form of adjacent features—the head boundary at equilibrium—because the external energy resists the internal energy. There are two key considerations while creating snakes: which energy phrases to utilise and how to use the least amount of energy. Elastic energy is often referred to as "internal energy". The snake's internal energy modifies the distance between its control points. It gains a shape as a result, which acts as an elastic band and causes it to shrink or expand. Image characteristics, on the other hand, rely on external energy. To determine how to consume the least amount of energy, optimisation methods like steepest gradient descent are applied. For quick iteration, there are other greedy algorithms. There are various drawbacks to snakes, such as how often their edges get caught on false image features and how they can't be utilised to eliminate non-convex features [18].

For instance, author [19] introduced the Constrained Local Models (CLM), which combined ASM with a local appearance model to handle non-rigid facial deformations but it has still some mistakes about the edges. CLM utilized a patch-based appearance model to capture local appearance variations and refine the shape estimation iteratively, leading to improved accuracy in face detection and landmark localization. Another significant contribution is the Supervised Descent Method (SDM) proposed by author [20]. SDM incorporated a cascade regression framework with ASM to refine the shape estimation progressively. This method achieved state-of-the-art performance in face detection and facial landmark localization, particularly in real-time scenarios.

On the other hand, low-level analysis models have significantly contributed to the advancement of face detection by extracting relevant low-level visual features. The evolution of these models, from handcrafted features to deep learning-based approaches, has led to improved accuracy and robustness in face detection. Challenges such as variations in lighting conditions, complex backgrounds, and occlusions continue to be addressed through illumination normalization techniques, context-aware models, and adaptive feature learning. Several notable research contributions have advanced the field of low-level analysis models in face detection. For instance, author [21] introduced the DeepFace model, which used a deep CNN to learn discriminative facial features. DeepFace achieved remarkable performance in face detection and recognition, particularly in handling pose variations and challenging lighting conditions. Another significant contribution is the FaceBoxes method proposed by [22]. FaceBoxes utilized a lightweight CNN architecture specifically designed for face detection tasks but have some problem with color.

Meanwhile, the color of the skin serves as the face's foundational feature. There are many benefits to using skin tone as a face-tracking feature. Facial traits other than color are processed far more slowly than the former. Color may be seen from any direction under specific lighting circumstances. Since only a translation model is required for motion estimation, this feature greatly simplifies the process. There are several obstacles when trying to use color as a feature for tracking human faces, such as the fact that the color representation of a face obtained by a camera can be affected by things like lighting conditions and the motion of the object being photographed [23].

B. Image based Approaches

Image-based approaches in the field of face detection refer to the methods and techniques that rely on analyzing and processing images to detect the presence and location of human faces. These approaches utilize the visual information present in images, such as pixel values, colors, textures, edges, and spatial relationships, to identify regions that potentially contain faces [3].

One of the examples of images-based approaches is Neural network-based image analysis approaches that have revolutionized the field of face detection by leveraging the power of deep learning to extract meaningful features from images and accurately detect faces. For instance, author [24] introduced the Deep Convolutional Network Cascade (DCNC) for face detection. DCNC utilized a cascade architecture of CNNs to achieve high accuracy while maintaining real-time performance. Multi-task Cascaded Convolutional Networks (MTCNN) is another significant contribution in the field. Author [25] proposed MTCNN, which simultaneously performs face detection, facial landmark localization, and facial attribute classification using cascaded CNNs. MTCNN achieved state-of-the-art performance in face detection tasks, particularly for faces with various poses, scales, and occlusions.

Meanwhile, one-Shot detectors have gained attention for their efficiency and effectiveness in face detection. These detectors aim to accurately locate faces in a single pass of the neural network, enabling real-time performance. One notable contribution in this area is the Single Shot MultiBox Detector (SSD) introduced by [26]. SSD utilizes a single neural network to perform face detection by predicting bounding box locations and class probabilities at multiple scales. This approach achieved high accuracy while maintaining fast inference speed. Another notable contribution is the RetinaFace model proposed by [27]. RetinaFace utilized a single-stage dense face localization approach that incorporated anchor-based and anchor-free strategies to handle faces with various scales and poses.

Similarly, one auto-associative network detects frontal-view faces, and another detects faces turned 60 degrees left or right. Then, a face detection system employing PDBNN was introduced by [28]. PDBNN is like an RBF network with probabilistic learning rules. The system consists of two stages: pre-processing and processing. In contrast to that, a deep-dense face detector was proposed by [29] requires that no single model can annotate poses or landmarks and recognize faces in many orientations.
On the other hand, Support Vector Machine (SVM) has been widely employed in the field of face detection due to its ability to effectively handle complex classification tasks. For example, author [30] introduced the "Support Vector Machines Applied to Face Detection" (SVMAFD) method, which showcased the effectiveness of SVMs for face detection. The authors presented an SVM-based approach that utilized a set of carefully designed features to classify image sub-windows as face or non-face. SVMAFD achieved promising results, demonstrating the potential of SVMs in face detection. Similarly, to that, author [31] proposed an improved SVM-based face detection method that incorporated feature selection techniques. The authors employed a combination of Haar-like features and Local Binary Patterns (LBP) and applied Recursive Feature Elimination (RFE) to select the most discriminative features. Their approach achieved competitive performance in face detection tasks, highlighting the importance of feature selection for SVM-based methods.

III. PROPOSED MODEL

This research framework is developed based on a benchmark for object-recognition presented in reference [32]. As it can be seen from Figure 1, it shows how this benchmark divides object-recognition tasks into training, classification, and detection tasks. Training and deployment use separate pipelines to assure surveillance device compatibility. The training process creates an impartial customised dataset and fine-tunes all models. Face identification and extraction follow image/real-time video frame extraction in the deployment process.

The classification task corresponds to a baseline convolutional-neural-network (CNN) that extracts information as data from input images and generates a feature map in the baseline. In this framework, transfer learning is applied on the classification, leveraging the learned attributes of a pre-trained and powerful CNN to extract new features for the model. To achieve optimal performance in facemask detection, an extensive backbone building strategy is conducted, utilizing four popular pre-trained models: MobileNet, ResNet50, Inception and VGG19. The novelty of the proposed work lies in the training task, an intermediate module that performs various preprocessing tasks before the actual image classification.

In the deep-learning neural network, the detection acts as an identity detector or predictor. The trained facemask classifier acquired by transfer learning is used in the proposed architecture to recognise faces with or without masks. The ultimate goal is to deter people from wearing face masks in public spaces by identifying those who do so. The following steps may be conducted in line with administrative or governmental policies. Using OpenCV 0.20, similar to how previous studies utilised it [33], an affine transformation approach is used to detect facial characteristics due to differences in face size and orientation inside cropped areas of interest (ROI). This guarantees correct identification despite variations in face features. The following points provide a thorough explanation of each job in the proposed framework.

A pre-trained under supervision is the first step. On the initial biased MAFA dataset, the CNN model underwent discriminative pre-training. The free Caffe Python package was used for the pre-training procedure [33] same as this author. In order to ensure that the model learns generalizable features and to enable better performance and faster convergence on the target task with little labelled data, this pre-training step aids the model in capturing general knowledge about the data distribution and extracting high-level representations.

A finite-turning of pre-trained is the second step. Due to its improved performance compared to other classification techniques, deep neural networks are used in this research to identify facemasks. Deep neural network training, however, is a time- and resource-intensive process that needs a lot of computing power. Transfer learning based on deep learning concepts is used to overcome these issues and produce quicker and more affordable training. Transfer learning enables the transfer of learnt information from an existing neural network to a new model in terms of the parameter weights. Even when trained on a modestly sized dataset, the new model performs much better thanks to this method. ImageNet, a large dataset with over 14 million photos, has been used to train a number of pre-trained models, including MobileNet and ResNet50. For this framework, the pre-trained models for facemask classification include MobileNet, ResNet50, Inceptionv3, and VGG19. Each of them has five more layers added to the final layer to refine it. These recently added layers are composed of a flattening layer, a dense ReLU layer with 128 neurons, an average pooling layer with a pool size of 5x5, a dropout layer with a rate of 0.4, and a deciding layer that uses the softmax activation function for binary classification.

Finding the expected face and determining which faces were found wearing masks or not is done in the final step. Following face recognition, the faces sans masks are individually fed into a neural network to investigate the identity of the person, paying particular attention to those who deviate from the face-mask norms. However, a fixed-sized input is needed for this stage. One method to meet this criterion is to resize the face inside the bounding box to 96x96 pixels which we done it. However, if the face is facing in a different direction, there may be a problem with this technique. A
simple solution is provided by the use of an affine transformation approach to address this problem. This method resembles the deformable part models proposed in [34] in certain ways were also used to tackle it.

IV. RESULTS AND DISCUSSIONS

The upcoming sections will illustrate the dataset description followed with its discussion on proposed models in term of face detection and face with mask detection.

A. Dataset Description

The MAFA (Multi-Attribute Facial Action) dataset is a facemask-centric dataset that has been widely used in the field of computer vision, particularly for tasks related to facemask detection and analysis. The MAFA dataset consists of a large collection of facial images that are annotated with various attributes related to facial appearance, including the presence or absence of a facemask. The dataset was specifically curated to address the need for comprehensive and accurate facemask detection in real-world scenarios, such as surveillance systems or public health monitoring.

The MAFA dataset is composed of over 35,000 facial images captured from diverse sources, including different genders, age groups, and ethnicities. This diversity ensures that the dataset covers a wide range of facial variations, which is essential for training robust facemask detection models. Each facial image in the MAFA dataset is manually annotated with multiple attributes, including the presence or absence of a facemask, gender, age group, and other facial attributes. These annotations provide valuable ground truth information for various facial analysis tasks. To ensure unbiased performance of the facemask detection models, the MAFA dataset undergoes an unbiased customization process during the training phase. This process involves carefully selecting and balancing the training samples to minimize any biases that may arise due to the dataset’s composition.

B. Identify Comparing MobileNet, ResNet50, Inceptionv3 and VGG19

The four models were evaluated for face detection and mask detection tasks as a separated way. The models include RetinaFaceMask based on MobileNet, RetinaFaceMask model based on ResNet50, RetinaFaceMask based on Inceptionv3, and RetinaFaceMask model based on VGG19. The performance of each model was assessed in terms of precision and recall for both face detection and mask detection.

RetinaFaceMask based on MobileNet in term of Face Detection, the model achieved a precision of 84.0% and a recall of 96.0%. This indicates that the model can effectively detect faces, with a relatively high recall rate, capturing a majority of the true faces present in the images. In term of Mask Detection, the model achieved a precision of 81.3% and a recall of 88.2%. This suggests that the model performs reasonably well in detecting whether individuals are wearing masks or not, with a good balance between precision and recall.

On the other hand, the proposed model based on ResNet50 in term of Face Detection has demonstrated excellent performance in face detection, achieving a high precision of 99.4% and a recall of 98.6%. These results indicate that the model is highly accurate in detecting faces, with a low false positive rate and a high true positive rate. Meanwhile, in term of Mask Detection, the model also showed strong performance in mask detection, with a precision of 98.83% and a recall of 98.5%. These results indicate that the model can effectively distinguish between masked and unmasked individuals, with a high level of accuracy and recall.

Similarly, RetinaFaceMask based on Inceptionv3 in term of Face Detection has also achieved a precision of 80.0% and a recall of 91.4% in face detection. Although the precision is relatively lower compared to other models, the model shows a good recall rate, capturing a high percentage of faces in the images. However, in term of Mask Detection, the model achieved a precision of 92.1% and a recall of 86.3%. This suggests that the model performs well in detecting masks, with a higher emphasis on precision compared to recall.

Final model RetinaFaceMask based on VGG19 in term of Face Detection and it also demonstrated strong performance in face detection, achieving a precision of 96.4% and a recall of 98.2%. These results indicate that the model can accurately detect faces, with a relatively low false positive rate and a high true positive rate. Meanwhile, in term of Mask Detection, the model achieved a precision of 86.7% and a recall of 90.2%. This suggests that the model can effectively distinguish between masked and unmasked individuals, with a good balance between precision and recall. It can be seen from Table I for all the precision and recalls of the four models.

<table>
<thead>
<tr>
<th>Models</th>
<th>Face Detection</th>
<th>Mask Detection</th>
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<tbody>
<tr>
<td></td>
<td>Precision (%)</td>
<td>Recall (%)</td>
</tr>
<tr>
<td>MobileNet</td>
<td>84.0</td>
<td>96.0</td>
</tr>
<tr>
<td>ResNet50</td>
<td>99.4</td>
<td>98.6</td>
</tr>
<tr>
<td>Inceptionv3</td>
<td>80.0</td>
<td>91.4</td>
</tr>
<tr>
<td>VGG19</td>
<td>96.4</td>
<td>98.2</td>
</tr>
</tbody>
</table>

In general, in term of Face Detection and mask detection, the ResNet50 model outperformed the other models, achieving the highest precision and recall values. This indicates that the ResNet50 model is highly accurate in detecting faces, with a low rate of false positives and false negatives. It also demonstrated superior performance in mask detection, with high precision and recall values. This suggests that the ResNet50 model is effective in accurately identifying individuals wearing masks. This indicates its potential as a robust and accurate model for detecting mask/non-mask faces.

However, further analysis and comparisons with existing models are necessary to evaluate its performance in relation to other state-of-the-art face mask detection models.

C. Output of Faces Detection Result

Here is the output of detected faces while wearing mask or not wearing mask as it can be seen from Figure 2 and Figure 3. In order to determine the best model for detecting mask/non-mask faces using transfer learning, we compare the
performance of MobileNet, ResNet50, Inceptionv3, and VGG19 models based on their given precision and recall from the provided results, it can be observed that the ResNet50-based proposed model achieves the highest accuracy in both face detection and mask detection tasks as already mentioned. With a precision of 99.4% and a recall of 98.6% for face detection, and a precision of 98.83% and a recall of 98.5% for mask detection, the ResNet50 model demonstrates superior performance in accurately detecting faces and distinguishing between masked and unmasked individuals. These results suggest that the ResNet50 model is the best fit as a backbone for detecting mask/non-mask faces using transfer learning as it can be seen from Figure 2.

On the other hand, MobileNet, Inceptionv3, and VGG19 models were classified wrong in the figure 4 and they marked not wearing a mask that someone who is wearing a mask while ResNet50 has marked the same Figure3 correctly. This also shows that in term of backbone detection for mask/non-mask faces, ResNet50 is outperformed others.

Meanwhile, to assess the utility of identity prediction in the proposed model, further details regarding identity prediction are done. However, it can be evaluated by examining the true positives, false negatives, false positives, and true negatives associated with identity prediction in the provided confusion matrices. Additionally, considering performance metrics such as precision, recall, and other relevant indicators specific to identity prediction can provide insights into its utility in the proposed model and upcoming table 1 illustrates that point.

The confusion matrices provide a detailed overview of the performance of each model in terms of face detection and mask detection as it can be seen from Table 2. These matrices reveal the true positives (TP), false negatives (FN), false positives (FP), and true negatives (TN) for each model. Starting with the RetinaFaceMask model based on MobileNet, the face detection results demonstrate a precision of 84.0% and recall of 96.0%. This indicates that the model accurately detects 84.0% of the faces present in the dataset, while 16.0% of the faces are missed. The mask detection performance shows a precision of 81.3% and recall of 88.2%, implying that the model correctly identifies 81.3% of the masked faces, but misclassifies 18.7% of the faces as masked.

### Table II. Confusion Matrices of Four Models

<table>
<thead>
<tr>
<th>Models</th>
<th>Face Detection</th>
<th>Mask Detection</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>Predicted Positive</td>
<td>Predicted Negative</td>
</tr>
<tr>
<td>MobileNet</td>
<td>Actual Positive 84.0% (4420)</td>
<td>16.0% (234)</td>
</tr>
<tr>
<td></td>
<td>Actual Negative 2.2% (108)</td>
<td>97.8% (4713)</td>
</tr>
<tr>
<td>ResNet50</td>
<td>Actual Positive 99.4% (4680)</td>
<td>0.6% (243)</td>
</tr>
<tr>
<td></td>
<td>Actual Negative 2.8% (132)</td>
<td>97.2% (4609)</td>
</tr>
<tr>
<td>Inceptionv3</td>
<td>Actual Positive 80.0% (4798)</td>
<td>20.0% (203)</td>
</tr>
<tr>
<td></td>
<td>Actual Negative 3.4% (127)</td>
<td>96.6% (4530)</td>
</tr>
<tr>
<td>VGG19</td>
<td>Actual Positive 96.4% (4374)</td>
<td>3.6% (214)</td>
</tr>
<tr>
<td></td>
<td>Actual Negative 3.8% (187)</td>
<td>96.2% (4760)</td>
</tr>
</tbody>
</table>
Moving on to the proposed model based on ResNet50, the face detection outcomes exhibit an exceptional precision of 99.4% and recall of 98.6%. This indicates that the model effectively identifies almost all faces present in the dataset with a high precision. The mask detection performance is also impressive, with a precision of 98.83% and recall of 98.5%, indicating accurate classification of the presence or absence of masks. The RetinaFaceMask model based on Inceptionv3 demonstrates a face detection precision of 80.0% and recall of 91.4%. Although the recall is relatively high, the precision suggests that the model may incorrectly detect some non-facial objects as faces. For mask detection, the precision is 92.1%, indicating accurate identification of masked faces, but the recall is 86.3%, suggesting some misclassification of masked faces as non-masked.

Lastly, the proposed model based on VGG19 exhibits a face detection precision of 96.4% and recall of 98.2%. These results indicate accurate and comprehensive face detection, capturing a large majority of the faces with high precision. In terms of mask detection, the precision is 86.7%, suggesting a relatively high accuracy in identifying masked faces. The recall of 90.2% implies that some masked faces may be misclassified as non-masked. Based on the comparison of the models, the proposed model based on ResNet50 emerges as the most suitable backbone for detecting mask/non-mask faces using transfer learning. It demonstrates outstanding performance in both face detection and mask detection, achieving high precision and recall scores. This indicates its capability to accurately identify faces and classify them based on the presence of masks.

V. CONCLUSION

In conclusion, the performance evaluation of the four models, namely RetinaFaceMask based on MobileNet, Proposed model based on ResNet50, RetinaFaceMask based on Inceptionv3, and proposed model based on VGG19, provides valuable insights into their effectiveness in the context of face detection and mask detection tasks. Based on the results obtained, it can be concluded that the proposed model based on ResNet50 outperforms the other models in terms of face detection precision and recall. This indicates that ResNet50 serves as a robust backbone for accurately detecting faces in various scenarios. Furthermore, the proposed model demonstrates high precision and recall values for mask detection, indicating its capability to effectively identify individuals wearing or not wearing masks. This is crucial for enforcing mask-wearing protocols and ensuring public safety. Comparatively, the RetinaFaceMask based on MobileNet and the RetinaFaceMask based on Inceptionv3 show relatively lower performance in face detection and mask detection tasks. Although they provide satisfactory results, they exhibit slightly lower precision and recall compared to the proposed model based on ResNet50. In terms of computational speed, a detailed analysis was not provided in the given information, which limits our ability to draw definitive conclusions regarding the models' efficiency. Future studies should consider evaluating the computational performance of these models to gain a comprehensive understanding of their real-time applicability.

REFERENCES


Sequential Model-based Optimization Approach Deep Learning Model for Classification of Multi-class Traffic Sign Images

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Abstract—Autonomous vehicles are currently gaining popularity in the future mobility ecosystem. The development of autonomous driving systems is still challenging in the research area of image processing and signal processing. Extensive research work was conducted on various traffic sign datasets. It achieved respectable results, but a robust network structure is still needed to improve the traffic sign recognition (TSR) system. In this research work, there is an alternative approach to designing deep learning models, which are implemented in TSR systems. The proposed deep learning model was also tested with different datasets to obtain the generalized model. The proposed model was based on a convolutional neural network (CNN), and Bayesian Optimization optimizes the model’s hyperparameters to find the best hyperparameters grid. After that, the optimized CNN model was used to classify the traffic sign images from three different datasets, including the German traffic sign recognition benchmark (GTSRB), the Belgium traffic sign classification (BTSC) dataset, and the Chinese traffic sign database, achieving the average accuracy scores of 99.57%, 99.15%, and 99.35%, respectively.

Keywords—Autonomous driving; convolutional neural network; deep learning; traffic sign; optimization

I. INTRODUCTION

Traffic sign images are fundamental and the most frequent visual tools to control traffic on the road, which are also used to communicate information to road users [1]. Thus, self-driving technologies use this road traffic information to control and drive an autonomous vehicle. For this reason, traffic sign recognition (TSR) systems have become quite popular and have been adopted in many engineering fields over the last decade [2]. Many applications are driven by this TSR system, such as advanced driver assistance systems (ADAS) [3], autonomous driving systems [4], and mobile mapping [5]. Moreover, many proposed problems related to the TSR system have been studied in recent research. In previous research [6], authors proposed two different models to identify factors affecting the retro-reflectivity of traffic signs. However, some authors found an overview of the interaction between road infrastructure and automated driving systems [7]. Additionally, the proposed problem of traffic sign recognition system problem has been developed by different approaches such as color-based, shape-based, and deep learning-based methods [8]. The deep learning-based method was the most popular because of its speed and accuracy [9].

Convolutional neural networks (CNN), one of the deep learning methods, can adjust the data without making any explicit specification, and CNN can also learn features efficiently from many samples without any preprocessing [8][9]. Some researchers applied CNN model to detect and recognize Thai traffic signs and achieved an average precision score of 0.93 [10]. An end-to-end convolutional network inspired by YOLOv2 was proposed and evaluated according to their Chinese traffic sign dataset (CTSD) [11]. Their approach achieved 90.37% in recall and 95.31% in precision, with the fastest detection speed of 0.017s per image. CNN was used to develop an autonomous traffic and road sign detection and recognition system of traffic signs collected from roads in Saudi Arabia, and there were 24 different traffic sign labels [11]. Their experimental results showed an accuracy of 100% in their dataset.

However, for the TSR system, a new architecture for neural network structure remains challenging for all the researchers in this field. Although many network architectures have been proposed, a robust network structure is still needed to develop for improving the TSR system. This research work implements an alternative approach to designing deep learning models for TSR systems. The main contribution of the proposed method is as follows:

- A new model is based on a convolutional neural network (CNN), and Bayesian Optimization optimizes the model’s hyperparameters to find the best hyperparameters grid, and
- The proposed model is also tested with three different datasets to get the generalized model.

II. LITERATURE BACKGROUND

The performance comparison between machine learning algorithms and humans to classify road traffic signs and the results are reported in their research work [12]. The human performance of classification of the road sign achieved an accuracy of 99.22%, while the best machine learning approach, CNN, with 99.46% correct classification accuracy. According
to their research, the performance of the CNN outperformed the human ability, and therefore CNN-based TSR system can be one of the best solutions to assist the driver. A hinge loss stochastic gradient descent (HLSGD) method was suggested to train CNN, and their CNN model was evaluated on the GTSRB [13]. This experiment showed a state-of-the-art result of 99.65% in recognition rate. In [14], an extremely fast detection module was proposed, and this module is based on a support vector machine (SVM) and CNN to detect and classify traffic signs. The overall classification accuracy of 98.24% for the German Traffic Sign Detection Benchmark (GTSDB) and 98.77% for CTSDB. OneCNN, a single CNN model, was proposed and tested over multiple traffic sign datasets from Germany, Belgium, and Croatia [15]. Their model got a classification rate of 99.04% in the German dataset, 97.66% in the Belgium dataset, and 99.37% in the Croatia dataset.

Their research proposed a deep learning approach to traffic sign recognition systems, and their classification experiments were conducted over different traffic sign datasets from Germany and Belgium [8]. The recognition rate of their proposed model got an accuracy of 99.71% in the German traffic sign dataset and 98.87% in the Belgium traffic sign dataset. Some researchers have designed a detector that was implemented using an R-convolutional neural network, and the structure of MobileNet and CNN was also used to be the classifier of traffic signs [16]. They got a detection accuracy of 96% in prohibitory, 100% in mandatory, 100% in Danger, and 99% in unique on GTSDB. Their classification accuracy was 99.66% in GTSRB. MicronNet, a highly compact deep convolutional neural network for real-time embedded traffic sign recognition system, was based on the microarchitecture design principle proposed and achieved an accuracy of 98.9% on the German traffic sign dataset [17]. Recently, a novel neural network architecture called Separating-Illumination Network (Sill-Net) was proposed by Zhang et al. [18]. Their model got a classification accuracy of 99.68% on GTSRB. The implementation of TSR using YOLOv5 was conducted with their own dataset, and they have a performance measure of 97.70% for all classes [19]. Recently, a recognition system based on Faster R-CNN and YOLOv4 network was implemented and achieved an accuracy score of 99.20% [20]. Recent advancements in computer vision and machine learning greatly help improve the accuracy of the TSR system [21]. However, the proposed deep learning model has achieved better performance across three traffic sign datasets.

III. MATERIALS AND METHODS

In this research work, three different traffic sign datasets used in testing the performance of the proposed CNN model are highly imbalanced. These three datasets are as follows:

- German traffic sign recognition benchmark,
- Belgium traffic sign classification dataset, and
- Chinese traffic sign database.

A. German Traffic Sign Recognition Benchmark (GTSRB)

In this dataset, there are single images with a multi-class classification problem of more than 40 traffic sign classes and more than 50,000 images, as shown in Fig. 1. Its ground truth data is reliable due to its semi-automatic annotation. The traffic sign images are raw RGB images with sizes ranging from 15x15 to 250x250 pixels [22].

B. Belgium Traffic Sign Classification (BTSC) Dataset

The Belgian traffic sign classification dataset (BTSC) [23] has 4533 training images and 2562 validation images split into 62 traffic sign types, as shown in Fig. 2. Compared with the GTSRB dataset, this dataset has different traffic sign pictograms, lighting conditions, occlusions, image resolutions, etc. Moreover, it contains categories that cluster different types of traffic signs (e.g., 50-speed-limit sign and 70-speed-limit sign), thereby raising the difficulty of the recognition task.

C. Chinese Traffic Sign Database

This database [24] includes 6164 traffic sign images containing 58 sign categories, as shown in Fig. 3. The images are divided into two sub-databases as training database and testing database. The training database includes 4170 images, while the testing contains 1994 images. All images are annotated with the four coordinates of the sign and the category.

![Fig. 1. Dataset distribution of German traffic sign images.](image1)

![Fig. 2. Dataset distribution of Belgium traffic sign images.](image2)

![Fig. 3. Dataset distribution of Chinese traffic sign images.](image3)
D. Convolutional Neural Network (CNN)

Convolutional neural network (CNN) is one of the self-adaptive methods that can adjust to the data without any explicit specification [8]. CNN can also effectively learn features from numerous samples with minimal preprocessing [9]. Therefore, CNN was used in this research work to implement a traffic sign recognition system; the overall block diagram is shown in Fig. 4.

![Overall block diagram of traffic sign recognition system.](image)

Fig. 4. Overall block diagram of traffic sign recognition system.

First, the image was loaded into the CNN model before being converted into a grayscale image with histogram equalization. Then, the images were rescaled by dividing with a maximum pixel value of 255. OpenCV library was used for these image preprocessing steps. During the preprocessing stage, all samples from GTSRB were down-sampled or up-sample to 32x32, while the samples from BTSC and Chinese datasets were down-sample to 50x50 pixels. After these steps, the proposed CNN model applied feature extraction and classification of the image. The based CNN model with seven layers was inspired by LeNet-based architecture [25], and then the model was optimized by Bayesian optimization.

The proposed CNN used Keras with TensorFlow backend, and the model was trained and evaluated the classifier on all three datasets. The proposed model comprises several layers, including a convolutional layer, Rectified Linear Units (ReLU) [26], max pooling [27], and fully connected layers. The convolution and pooling layers were used as a feature extractor of the input images, and then fully connected layers were attached to flatten the matrix into a vector. Finally, softmax was used as an activation function to classify the outputs of different traffic sign labels. The softmax function in this research uses the categorical cross-entropy loss function [8]. The LeNet-based CNN architecture for the traffic sign recognition system is shown in Fig. 5.

![LeNet-based CNN architecture for traffic sign recognition.](image)

Fig. 5. LeNet-based CNN architecture for traffic sign recognition.

E. Hyperparameter Tuning Using Bayesian Optimization

In this research, the proposed CNN model was optimized by Bayesian Optimization, a sequential approach for the global optimization of black-box objective functions that are costly to evaluate [28]. The global maximizer or minimizer is mathematically needed to find an objective function [29]. The function \( f \) is defined as:

\[
x^* = \arg \max_x f(x),
\]

where \( x \) are the hyperparameters. In this research work, the set of hyperparameters was optimized to get the minimum value of the negative accuracy. Therefore, Bayesian optimization with a scikit-optimizer was used to find the minimum value. It provides a utility function to create the range of values to examine for each hyperparameter [30]. The following hyperparameters were optimized by scikit-optimize:

- the learning rate,
- the number of fully connected Dense layers,
- the number of nodes for each of the dense layers, and
- activation function.

Table I shows the range grid of the hyperparameters in this experiment. During the optimization process, there are 15 subsequent evaluations of \( f(x) \), and the target minimum value of accuracy was obtained in the 13th number of calls. In Fig. 6, the result can be visually seen with the convergence plot, and the test was done using the dataset of GTSRB. Therefore, Bayesian optimization [31] can give a better model performance on this dataset, and the optimized model will be used for testing with another two datasets. The selected hyperparameters combination is described in Table I.

![Convergence plot](image)

Fig. 6. The convergence plot of function \( \min f(x) \).

<table>
<thead>
<tr>
<th>TABLE I. HYPERPARAMETER GRID FOR DEEP LEARNING MODEL</th>
</tr>
</thead>
<tbody>
<tr>
<td>Parameters</td>
</tr>
<tr>
<td>Learning rate</td>
</tr>
<tr>
<td>Number of dense layers</td>
</tr>
<tr>
<td>Number of dense nodes</td>
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</table>
TABLE II. HYPERPARAMETER TUNING FOR DEEP LEARNING MODEL

<table>
<thead>
<tr>
<th>Learning rate</th>
<th>Number of dense layers</th>
<th>Number of dense nodes</th>
<th>Activation function</th>
<th>Accuracy</th>
<th>Loss</th>
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<td>1.00E-05</td>
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<td>512</td>
<td>relu</td>
<td>50.44</td>
<td>1.077</td>
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<td>sigmoid</td>
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<td>relu</td>
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<td>1.20E-05</td>
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<td>930</td>
<td>relu</td>
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<td>3.70E-05</td>
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<td>3.5674</td>
</tr>
<tr>
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<td>608</td>
<td>sigmoid</td>
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<td>3.5515</td>
</tr>
<tr>
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<td>679</td>
<td>relu</td>
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<tr>
<td>7.70E-06</td>
<td>9</td>
<td>977</td>
<td>relu</td>
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</tr>
<tr>
<td>1.00E-02</td>
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<td>512</td>
<td>relu</td>
<td>96.44</td>
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</tr>
<tr>
<td><strong>1.50E-04</strong></td>
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<td><strong>512</strong></td>
<td><strong>relu</strong></td>
<td><strong>99.57</strong></td>
<td><strong>0.2068</strong></td>
</tr>
<tr>
<td>1.60E-04</td>
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<td>512</td>
<td>relu</td>
<td>99.24</td>
<td>0.5398</td>
</tr>
<tr>
<td>2.20E-04</td>
<td>5</td>
<td>1027</td>
<td>relu</td>
<td>99.39</td>
<td>0.2104</td>
</tr>
</tbody>
</table>

The detailed values of each hyperparameter can be seen in Table II, and as mentioned before, the best hyperparameter grid was in the 13th iteration with 99.57% accuracy and 0.2068 in loss value. Therefore, the proposed model was updated with this parameter grid and shown in Fig. 7. For training the GTSRB, the updated model architecture of convolutional layers, ReLU, max pooling, and fully connected layers are reported in Table III. The detailed structure of the proposed CNN model is shown in Fig. 7.

IV. RESULTS AND DISCUSSION

All experiments of CNN architecture and Bayesian optimization were done on a computer with an Intel Core i5 – 7300HQ CPU, 16 GB of RAM, and an NVIDIA GeForce GTX 1050 discrete GPU with 2 GB of RAM; this computer was enough to train and run models. Training and validation sets are already included in the datasets since they can be downloaded from the source link. Therefore, the developed CNN model was trained with a training set and evaluated with the model with a validation set. Fig. 8, Fig. 9, and Fig. 10 show the values of training loss and accuracy from different datasets and are compared with the values of validation loss and accuracy on those datasets. The training time per epoch of the proposed CNN model is 15.76 seconds, and the simulation ran 100 epochs per dataset. Our proposed CNN model achieved better accuracy with lower loss values on all three datasets.

Fig. 7. The updated CNN architecture for classification of traffic sign images.

Fig. 8. Comparison between training and validation of German traffic sign dataset regarding (a) loss and (b) accuracy.

Fig. 9. Comparison between training and validation of Belgium traffic sign dataset in terms of (a) loss and (b) accuracy.

TABLE III. THE PROPOSED CNN ARCHITECTURE

<table>
<thead>
<tr>
<th>Layers</th>
<th>Type</th>
<th>Output shape</th>
<th>Param#</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>Convolutional (ReLU)</td>
<td>(None, 28, 28, 60)</td>
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</tr>
<tr>
<td>2</td>
<td>Max-Pooling</td>
<td>(None, 14, 14, 60)</td>
<td>0</td>
</tr>
<tr>
<td>3</td>
<td>Convolutional (ReLU)</td>
<td>(None, 12, 12, 30)</td>
<td>16230</td>
</tr>
<tr>
<td>4</td>
<td>Max-Pooling</td>
<td>(None, 6, 6, 30)</td>
<td>0</td>
</tr>
<tr>
<td>5</td>
<td>Flatten</td>
<td>(None, 1080)</td>
<td>0</td>
</tr>
<tr>
<td>6</td>
<td>Fully connected (ReLU)</td>
<td>(None, 512)</td>
<td>553472</td>
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<tr>
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<tr>
<td>11</td>
<td>Dropout</td>
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<tr>
<td>12</td>
<td>Fully connected (ReLU)</td>
<td>(None, 512)</td>
<td>22059</td>
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</table>

Total params: 1,643,945
Trainable params: 1,643,945
Non-trainable params: 0
Fig. 10. Comparison between training and validation of Chinese traffic sign dataset regarding (a) loss and (b) accuracy.

Performance is evaluated based on the GTSRB dataset by calculating precision, recall, F1-score, and accuracy:

\[
\text{Precision (\%)} = \frac{TP}{TP + FP} \times 100, \quad (2)
\]

\[
\text{Recall (\%)} = \frac{TP}{TP + FN} \times 100, \quad (3)
\]

\[
\text{F1-score (\%)} = \frac{2 \times TP}{2 \times (TP + FP + FN)} \times 100, \quad (4)
\]

\[
\text{Accuracy (\%)} = \frac{TP + TN}{TP + FP + TN + FN} \times 100, \quad (5)
\]

Where TN is the number of true negatives, TP is the number of true positives, and FN and FP are the number of false negatives and false positives, respectively [32]. The precision, recall, and f1-score are calculated for individual labels to classify traffic signs.

It clearly shows that the average percentage of the classification of 43 sign labels from the GTSRB dataset is 96.79% in precision measure, 97.21% in recall measure, and 96.65% in F1-score, as shown in Fig. 11. The proposed model is capable of classification of 61 different annotated sign labels and the performance measures of Belgium traffic sign database are shown in Fig. 12. The proposed model achieved 94.06% of precision measure, 93.82% of recall measure and 92.77% of F1-score. Moreover, the proposed model was also tested with another traffic database of the Chinese traffic sign database, and the average results on each performance measure were 91.24%, 86.11%, and 86.41%, respectively. In this database, there are 57 different sign labels; these performance measures can be seen in Fig. 13.
The new CNN model was proposed in this research work, and the experimental results are shown in the above section. In this section, a detailed discussion of the accuracy results from this proposed model and the existing deep learning models. The classification accuracies of the proposed model on the German, Belgium, and Chinese traffic sign datasets are mentioned as follows: the average accuracy of 99.57% on the German dataset, 99.15% on the Belgium dataset, and 99.35% on the Chinese dataset. A comparison between the recent research and the proposed CNN model on the German dataset is shown in Table IV. The proposed model achieved a better accuracy score than the previous study with CNN [15], and MicronNet [17], but the score is slightly less than CNN with three spatial transformer layers and SGD without momentum as the loss function optimizer [8]. However, our CNN model’s total number of parameters is 1,643,945, which is much lower than their model parameters. (See Table VI). In contrast, the proposed model got the best accuracy value on the Belgium traffic dataset compared with the recent research, including CNN with three Spatial Transformer Networks [8], a Gaussian-Bernoulli deep Boltzmann machine-based (GBDM) hierarchical classifier [30], and other approaches [2]. The accuracy comparison between our proposed CNN model and the recent studies is shown in Table V. Therefore, the lightweight CNN model with higher accuracy measures was successfully proposed in this research and can be used in real-time scenarios.

V. CONCLUSION

In this paper, sequential model-based optimization (Bayesian optimization with scikit-optimizer) approach deep learning model was proposed and tested with different traffic sign datasets, including GTSRB, BTSC, and the Chinese traffic sign database. The proposed model was in a rank with the top result in BTSC and the top three results in GTSRB. It is possible to show the high-quality results obtained from the proposed models using the Chinese dataset. Therefore, the proposed CNN model significantly improves traffic sign recognition (TSR) system performance compared with other existing studies. Further work should be conducted with other traffic sign datasets from different countries to guarantee the quality and accuracy levels of the proposed CNN model while comparing the real-world scenarios.

AUTHORS’ CONTRIBUTIONS

Si Thu Aung conceived and designed the experiments, performed the experiments, analyzed the data, performed the computation work, prepared figures and/or tables, authored or reviewed drafts of the paper and approved the final draft. Jarutwat Rajjuangrabin conceived and designed the experiments, analyzed the data, performed the computation work, authored, or reviewed drafts of the paper, and approved the final draft. Ekkarat Viyanit authored, or reviewed drafts of the paper, and approved the final draft.

References

TABLE IV. ACCURACY COMPARISON OF GERMAN TRAFFIC SIGN DATASET BETWEEN THE PROPOSED MODEL AND RECENT STUDIES

<table>
<thead>
<tr>
<th>References</th>
<th>Methods</th>
<th>Accuracy</th>
</tr>
</thead>
<tbody>
<tr>
<td>[8]</td>
<td>CNN with 3 Spatial Transformer Networks</td>
<td>99.71%</td>
</tr>
<tr>
<td>[18]</td>
<td>Separating-illumination Network (Sil-Net)</td>
<td>99.68%</td>
</tr>
<tr>
<td>Proposed model</td>
<td>Bayesian Optimization approach CNN model</td>
<td>99.57%</td>
</tr>
<tr>
<td>[15]</td>
<td>CNN</td>
<td>99.11%</td>
</tr>
<tr>
<td>[17]</td>
<td>MicronNet</td>
<td>98.9%</td>
</tr>
</tbody>
</table>

TABLE V. ACCURACY COMPARISON OF BELGIUM TRAFFIC SIGN DATASET BETWEEN THE PROPOSED MODEL AND RECENT STUDIES

<table>
<thead>
<tr>
<th>References</th>
<th>Methods</th>
<th>Accuracy</th>
</tr>
</thead>
<tbody>
<tr>
<td>Proposed model</td>
<td>Bayesian Optimization approach CNN model</td>
<td>99.15%</td>
</tr>
<tr>
<td>[33]</td>
<td>GBDM</td>
<td>98.92%</td>
</tr>
<tr>
<td>[8]</td>
<td>CNN with 3 Spatial Transformer Networks</td>
<td>98.87%</td>
</tr>
<tr>
<td>[15]</td>
<td>CNN</td>
<td>98.17%</td>
</tr>
<tr>
<td>[2]</td>
<td>INNLP+SCR(PI)</td>
<td>97.83%</td>
</tr>
</tbody>
</table>

TABLE VI. COMPARISON OF A NUMBER OF TRAINABLE PARAMETERS OF THE PROPOSED CNN MODEL WITH RECENT STUDIES

<table>
<thead>
<tr>
<th>References</th>
<th>Methods</th>
<th>Trainable parameters</th>
</tr>
</thead>
<tbody>
<tr>
<td>[8]</td>
<td>CNN with 3 Spatial Transformer Networks</td>
<td>14,629,801</td>
</tr>
<tr>
<td>Proposed model</td>
<td>Bayesian Optimization approach CNN model</td>
<td>1,643,945</td>
</tr>
</tbody>
</table>
FUNDING

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DATA AVAILABILITY STATEMENT

The German traffic sign recognition Benchmark (GTSRB) can be downloaded from https://benchmark.ini.rub.de/gtsrb_dataset.html Belgium traffic sign classification (BTSC) dataset can be obtained from https://btsd.ethz.ch/shareddata/ Chinese traffic sign database can be accessed from http://www.nlpr.ia.ac.cn/pal/trafficdata/recognition.html All datasets have been used for this study.

REFERENCES


Whale Optimization-Driven Generative Convolutional Neural Network Framework for Anaemia Detection from Blood Smear Images

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Abstract—Anaemia is a frequent blood disorder marked by a reduction in the quantity of haemoglobin or the number of red blood cells in the blood. Quick and accurate anaemia detection is crucial for fast action and effective treatment. In this research, we provide a new structure called Whale Optimization-Driven Generative Convolutional Neural Network (WO-GCNN) for the detection of anaemia using blood smear pictures. To increase anaemia detection accuracy, the WO-GCNN system combines the strength of generative models and convolutional neural networks (CNNs). In order to create artificial blood smear images and learn the underlying data distribution, generative models, such as Generative Adversarial Networks (GANs), are used. Improve the functionality of the WO-GCNN system by applying the Whale Optimisation Algorithm (WOA), which is based on the hunting behaviours of humpback whales. To create the optimal set of CNN weights, the WOA effectively achieves a compromise between exploitation and exploration. The WO-GCNN framework accelerates convergence speed and increases overall performance of anaemia detection by incorporating the WOA into the training process. On a sizable dataset of blood smear pictures obtained from clinical settings, we assess the suggested WO-GCNN system. A highly accurate and effective approach for the early identification of anaemia is produced by combining generative models and CNNs with the WOA optimisation. By enabling early anaemia identification, the proposed WO-GCNN framework has the potential to have a substantial impact on the field of medical image analysis and enhance patient care. It can be a useful tool for medical personnel, supporting them in making decisions and giving anaemia patients urgent interventions.

Keywords—Generative adversarial network, blood smear images; convolutional neural network, anaemia; Whale Optimization

I. INTRODUCTION

The most prevalent haematological condition, defined as low haemoglobin concentration, affects over two billion individuals worldwide. Numerous acute and chronic diseases, including cancer, malnutrition, gastrointestinal bleeding, and chronic kidney disease, may trigger it. While severe anaemia, like thalassemia, require ongoing monitoring, others can be managed with simple medications. Some dangerous emerging reasons for anaemia, like significant gastrointestinal bleeding, necessitate early identification. Until the illness is not compensated and consequences arise, it is challenging to identify anaemia using simply patient records and examinations. The most common method for identifying anaemia is a complete blood count conducted in a laboratory. The laboratory test is intrusive, expensive, and needs specialised facilities and supplies (such as skilled medical personnel for blood sample and a haematology analyser for analysis using a biochemical reagent) [1].

The main factor contributing to anaemia, which typically has a high onset during preadolescence and happens if the blood's capacity for carrying oxygen is insufficient to satisfy the body's physiological needs, is a decline in red blood cell production. It is a significant issue in worldwide public health that necessitates the use of preventative diagnostic technologies. For adults, the lower recognised threshold for haemoglobin content in the blood is: 13.0 g/dL for males and 12.0 g/dL for women; tolerance levels due to varied races and ages should be taken into consideration. The creation of remedies intended at bolstering diagnosing concerns is the focus of the investigation discussed in this paper. To detect critical conditions, including the need for a donation of blood, or to allow autonomous tracking of haemoglobin levels, extremely precise instruments are needed. Lower equipment costs and convenience of use are also priorities in settings lacking surgeries and suitable instruments. Additionally, since non-invasive techniques enable the tracking of therapy answers, this technique is crucial for directing a more efficient and affordable tracking plan [2].

IDA, which is more prevalent in women, continues to be one of the five most common causes of years spent with a handicap in people. While it has historically been viewed primarily as a problem of public health impacting developing children, premenopausal women, and pregnant women, it is also coming to light as a clinical condition which may affect
clients presenting across various healthcare and surgical fields, particularly those with chronic illnesses and older adults. International practise recommendations have been prompted to give special consideration to IDA assessment and treatment as new data on the role of IDA in poor medical results continues to emerge. The vague nature of the signs and the numerous IDA aetiologies, however, might make the diagnosis difficult. Additionally, the availability of several iron supplements formulas can make treatment choices more difficult [3].

According to the World Health Organisation (WHO), a patient blood management (PBM) strategy should be utilised to maximise the surgical patient's natural volume of blood. Several instructions, which include those for managing and optimising pre-operative anaemia and cells salvaging, have been published to aid hospitals in the execution of patient blood management. Studies have demonstrated that managing patient blood results in better outcomes and lower healthcare expenses [4]. Implementing patient blood management was found to reduce transfusions, hospitalisations, morbidity, and hospital readmission in total knee and hip arthroplasty and heart surgery. In one potential multi-centre study, the safety of a hospital-wide implementation of patient blood management was assessed. The outcomes of 54,513 patients prior to and 75,206 patients following the implementation of patient blood management showed that it was not disadvantageous. Data on the application of patient blood management in four tertiary institutions for 605,046 patients was provided by a sizable Australian study. A five-year implementation plan was started, and the last year's results were compared to the baseline years in terms of mortality, costs, and red cell transfusions. [5].

Red blood cells include the iron-containing protein known as haemoglobin, also known as Hgb or Hb, which is in charge of transporting oxygen from the respiratory system to other organs. Healthy adolescent blood haemoglobin levels for males and females normally fall between 14 to 18 g/dl and 12 to 16 g/dl, respectively. Blood sugar levels can, however, drop considerably due to a variety of illnesses, poor nutrition, and abnormalities of the bone marrow. Anaemia is a serious haematological condition that can lead to physical and mental weakness, mood swings, cognitive impairment, skin pallor, and in the worst cases, cardiac arrest. The World Health Organisation (WHO) has identified anaemia as a severe health concern for young children and expectant mothers. Invasive tests for blood are the most popular method of detecting anaemia and monitoring Hb levels. The intrusive blood tests are uncomfortable and run the danger of becoming infected while in the hospital. A visual inspection of other tissues in the body, such as the tongue, nail beds, palmar crease, and palpebral conjunctiva, is recommended in order to identify anaemia [6]. But a highly skilled medical professional who is able to connect the visual indicators to anaemia ought to just carry out such a test. Inspection by hand also carries the hazards of inter- and intra-observer bias, human error, and non-reproducible outcomes. As a result, there is a pressing need for automated tools to assist in the anaemia screening process. Recent developments in deep learning have brought efficient answers to issues in the biomedical sector. Additionally, they can be utilised to create sophisticated computer-aided diagnosis (CADx) tools that examine images of body components like the conjunctiva and fingernails to find anaemia. Such instruments would be able to measure image-based factors with great accuracy, such as pallor and the erythema index (EI). Pre-processing, segmentation, feature extraction, and classification are the fundamental subprocesses that make up the CADx method for diagnosing anaemia. [7]. The framework has the potential to be a reliable and effective way for detecting anaemia from blood smear images, which will improve medical diagnostics. The amount and diversity of the supplied dataset may have an impact on the framework's performance, necessitating thoughtful deliberation and augmentation to guarantee resilience across varied populations of patients and conditions.

The key contribution of this model is given below:

1) To increase the accuracy of anaemia detection, the WO-GCNN system combines the strength of generative models, namely Generative Adversarial Networks (GANs) and convolutional neural networks (CNNs).

2) The WO-GCNN solves the problem of insufficient data to train deep learning models by creating false blood smear images and understanding the fundamental information distribution through the use of GANs.

3) To discover the best combination of CNN weights, the approach effectively strikes an equilibrium between exploration and exploitation by incorporating the Whale Optimisation Algorithm (WOA) into the training phase of the WO-GCNN.

4) The suggested WO-GCNN method exhibits extremely precise and efficient anaemia recognition, with the potential for early identification and enhanced patient care in the area of medical imaging analysis.

5) An innovative method for overcoming the difficulties of detecting anaemia from blood smear images is demonstrated by the integration of generative models and CNNs with the WOA optimisation.

The structure of the essay is organised as follows: The associated work was covered in Section II. Section III describes the problem statement. Section IV describes the suggested convolutional Neural Network algorithm for identifying anaemia. In Section V, the proposed method's effectiveness and performance were evaluated, and the findings were shown in graphs and tables. The final Section VI provides a summary of the paper's findings.

II. RELATED WORKS

Alzubaidi et al. [8] proposed Red blood cells (erythrocytes), which are elongated, circle (normal), and other blood components, are divided into three groups using three deep learning models. The suggested models were created using classic and parallel convolutional layers, two types of convolution neural networks. The amount of layers and learnable filters in these models are different. The most effective model out of the three has been determined empirically. In this study, transfer learning and data augmentation were used to address the problem of not having enough data for training deep learning models for the
classification of red blood cells. The target dataset's images were gathered from the identical domain using the comparable domain transfer learning technique, and previously learned models were then suggested and trained. The categorization task for sickle cell disease was subsequently enhanced using these previously trained models. Our models were assessed using two datasets. The experiment's findings demonstrated that classifying sickle cell disease using the identical domain transfer learning greatly increased. It also made it possible for our algorithms to work better than the most recent erythrocyte classification techniques. By reaching an accuracy of 99.54% with the framework alone, 99.98% with the approach plus a multiple classes SVM classifier on the erythrocytes DB dataset, and 98.87% on the gathered dataset, our model exceeded the most recent techniques and reached state-of-the-art performance. Last but not least, training a multiclass SVM classifier demonstrated its efficacy in extracting features and produced excellent outcomes. The drawback is that the model needs to be taught in doing the following job to be able to classify white blood diseases.

Kilicarslan et al. [9] explains The two combined models for predicting HGB-anemia, B12 deficiency anaemia, nutritional anaemia (iron deficiency anaemia and folate deficiency anaemia), and individuals without anaemia have been developed using genetic algorithms (GA) and deep learning algorithms of arranged auto encoder (SAE) and convolutional neural network (CNN). The developed GA-SAE and GACNN approaches use GA for optimising the high-level parameters of the SAE and CNN algorithms because it can be difficult to choose acceptable values for deep learning algorithms. The suggested algorithms' estimations and classification results were assessed using the accuracy, precision, F-score, and sensitivity criteria. The newly developed GA-CNN approach, whose layers were continuously trained separately of one another, was 98.50% more efficient than that of the study presented in the literature, according to the lab's evaluations utilising the dataset. The suggested combinations look at medical data to make diagnosis while lowering human error because of infirmity or exhaustion. This indicates how ineffective and unsuccessful it is.

Haematological condition sickle cell anaemia was once considered to be one of the special characteristics of the indigenous population, but it today affects everyone in the world and calls for immediate medical attention. Yeruva et al. [10] describes The history of Sickle Cell Disease (SCD), both nationally and internationally (in the context of India), the description of the disease's signs and symptoms, caution signs, effects, and treatment. Blood characteristics are also used to categorise the blood cells in individuals with sickle cell disease and thalassemia patients. This study utilises a multi-layer perceptron to better accurately simulate sickle cell disease with the goal to decrease the period and energy needed for sickle cell disease painful control systems. In this work, we investigate the "MLP Classifier" deep learning model, which outperforms methods like Support Vector Machine, Decision Tree Classifier, K-Nearest Neighbour, Random Forest Algorithms, and Logistic Regression in terms of output. According to simulation results, the proposed MLP classification has an accuracy of 99.9% for predicting both sickle cell and thalassemia. Normal medical lab workers cannot apply the MLP classification algorithm and prediction model in the TSCS scenario since it is not faster or quicker.

El-kenawy et al. [11] proposed employing haematological criteria, machine learning is being used for prediction and to approximate the market value of haemoglobin. Considering the Random Forest, Support Angle Device, and Artificial Neural Networks three computer learning techniques. The least expensive mistake was made by K.N.N. Set. In conclusion, the current investigation revealed aberrant RBCs, HGB, HCT, and CRP levels in the CBCs of frequent COVID-19 cases, which were also the most likely laboratory findings in these patients. We provide a machine learning approach in this study to estimate blood level based on bloodstream test criteria. Pre-processing is done in our suggested method for haemoglobin evaluation so that data is minimised and normalised immediately before training the models. The results of this technique are contrasted with those from different manufacturers' learning versions. A precise Blood assessment is produced by the suggested version. When reading the CBC of COVID-19 patients, clinicians should take these factors into account. To improve the highest possible accuracy new optimization algorithm is utilised.

El-kenawy et al. [12] explains classification and regression are two machine learning tasks. Using haematological data, estimating the value of haemoglobin for the regression. Three machine learning techniques, namely Random Forest, Artificial Neural Networks, and Linear Regression, were employed. The least mistake was generated using random forest. The type of anaemia is then classified using the haematological parameters and the estimated haemoglobin levels. Comparing Decision Tree against Random Forest, Artificial Neural Networks, and Naive Bayes, it became clear that Decision Tree was the superior classifier. Finally, we suggested an ensemble model (HEAC) that combines the Decision Tree classifiers, Naive Bayes, and Random Forest. The proposed model performed better than current classifiers and decision trees. Before training the models, pre-processing reduces and normalises the data in the Haemoglobin Estimation and Anaemia Classification (HEAC) method that we recommend. Results obtained from various machine learning models are compared to results from this model. The haemoglobin estimation and anaemia classification findings from the suggested method are quite precise. Genetic algorithms are used to obtain the greatest precision and identify the best weights.

Sickle cell anaemia (SCA) is a significant haematological condition which often requires hospitalisation over the course of a patient's lifetime and may even be fatal. Aliyu et al. [13] presents a two-step process Initially identify the RBC region of interest (ROI) using automatic red blood cell (RBC) extraction from a person's blood smear image. In order to categorise and forecast whether anomalies will be found in SCA patients, a deep learning AlexNet model is used. The research included (nearly 9,000 single RBC pictures) from 130 SCA patients, with 750 units for every category, with the goal to provide an overall multiscale forms assessment and shape factor quantization. We demonstrate how the suggested
structure can automatically categorise 15 different RBC shape types, includes normal, using a sophisticated AlexNet transfer learning model. The specificity, sensitivity, precision and accuracy of the cell's name categorization predictions were 95.92%, 77%, 98.82%, and 90%, respectively. It requires a lot of time to perform.

III. PROBLEM STATEMENT

The improvement of categorization and model predictions for diverse haematological disorders, such as sickle cell disease and anaemia, is the main focus of the issue statement discussed in this study. To guarantee prompt medical attention, these illnesses must be accurately identified and classified due to their major global effects. However, a number of obstacles prevent this field from progressing. First of all, deep learning models struggle to operate at their best due to a lack of training data. Furthermore, the effectiveness and success rates of current studies are frequently low, emphasising the need for better methods. Another issue is computational time efficiency. Additionally, there are difficulties in using these models in real-world medical laboratory settings, necessitating easier and more accessible solutions. As a result, the issue statement focuses on creating reliable and accurate classification and prediction models, overcoming data constraints, enhancing effectiveness, ensuring practical use, and speeding up the diagnosis and treatment of haematological disorders [14].

IV. PROPOSED WO-GCNN METHOD FOR ANAEMIA DETECTION

A convolutional Neural Network is used for image collection, pre-processing, feature extraction, feature selection, and categorization in order to detect and recognise the anaemia. The collected examples of photos have been posted. In pre-processing, filtering is employed to get rid of noise, and operations are used to find the edges of the image. The relevant traits are then extracted, and the classification of normal and pathological cells are done. Fig. 1 shows the block diagram for classification of anaemia using WO-GCNN mechanism.

A. Data Collection

Consulting pathologists gathered, organised technically, and annotated blood imaging data for the suggested method. A 40x Olympus Dp27 lens was used to examine the blood smear slides. Images at a resolution of 1920 x 1080 pixels were taken with an 8.9-megapixel CMOS sensor. From 50 blood smear slides, it has 500 images in total, 250 of which are healthy and 250 of which are anaemic RBC images. Those without anaemia are represented in the images, while those with anaemia are represented in the images [15].

The dataset is partitioned into three separate sets or folds for 3-fold cross-validation. The remaining two folds are utilised for training, while each fold is used once as a validation set. To guarantee every subset has been utilised as a validation set precisely once, this procedure is repeated three times. The total number of photos and the proportion of normal and anaemic red blood cells (RBC) images in each fold are displayed in the Table I.

<p>| TABLE I. 3 FOLD CROSS VALIDATION OF HEALTHY AND ANAEMIC RBC IMAGES |
|---------|-----------------|-----------------|</p>
<table>
<thead>
<tr>
<th>Folds</th>
<th>Total images</th>
<th>Healthy RBC images</th>
<th>Anaemic RBC images</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>167</td>
<td>83</td>
<td>84</td>
</tr>
<tr>
<td>2</td>
<td>167</td>
<td>83</td>
<td>84</td>
</tr>
<tr>
<td>3</td>
<td>166</td>
<td>84</td>
<td>82</td>
</tr>
</tbody>
</table>

Fig. 1. Classification process of anaemia.

Blood smear image

Noise removal using weiner filter

Data augmenting using GAN

Normal RBC

Abnormal RBC (Anaemia)

CNN classification for detecting anaemia

WOA for feature selection
B. Data Generation using GAN

The generative models known as GANs are built on adversarial and deep learning techniques. GANs’ main objective is to learn and replicate the distribution and traits of some interesting data in order to create similar data of the same type. A discriminator and a generator are the two neural network models that make up GANs. The discriminator is taught to categorise the veracity of the data it receives as input. The generator is adept at creating new artificial information that is exact replicas of the real information. Although the discriminator and generator systems are trained concurrently, they have various goals (as well as loss) parameters [16].

The discriminator’s goal is to increase the probability that it can tell the difference between genuine data (from the training set) and fictitious data from the generators. For labelled genuine training data \( x \), \( \log (D_0(x)) \), \( D_0(x) \) is ideally 1, and for fake data \( z \) produced by the generator \( G_0 \), its loss is \( \log (1 - D_0(G_0(z))) \). Loss is measured using cross-entropy. The discriminator’s aim for identifying authentic and fraudulent data is then maximised during training using feedback from its loss measurement loop is given by eq. (1).

\[
\max_{D_0} V_{\theta_0}(D_0) = E_{0 \sim P_{data}(x)}[\log D_0(x)] + E_{0 \sim P_{z}(z)}[\log (1 - D_0(G_0(z)))]
\]

(1)

Another reverse propagation feedback process from the discriminator drives training of the generator with the goal of reducing the likelihood that the generator’s bogus data will be identified in eq. (2).

\[
\min_{G_0} V_{\theta}(G_0) = E_{0 \sim P_{z}(z)}[\log (1 - D_0(G_0(z)))]
\]

(2)

Equation (3) contains the whole GAN objective function.

\[
\min_{G_0} \max_{D_0} V_\theta(D_0, G_0) = E_{0 \sim P_{data}(x)}[\log D_0(x)] + E_{0 \sim P_{z}(z)}[\log (1 - D_0(G_0(z)))]
\]

(3)

The discriminant and generation compete during GAN training in a manner comparable to a zero-sum mini-max game. Training proceeds till the GAN structure combines, at which point (i) the discriminator is able to distinguish between genuine and phoney data with adequate accuracy and (ii) the generator produces data that the discriminator has an extremely small likelihood of finding. The GAN network needs to be retrained if just one of the two results is (i) or (ii) [16].

C. Pre-processing

The augmented anaemia dataset is used in pre-processing for removing the extra noises. An image of the binary model of the multi-coloured blood stain is created. The picture is then improved. The Weiner filter is a flexible noise reduction device. A clearer picture of the concentration, in this case just the red blood cells in the blood sample, is then provided by clearing up the picture. The acquired image is then put to use in additional processing. In the pre-processing stage, noise reduction, unwanted effects like noise are reduced or eliminated with the goal of preparing the image for subsequent processing. Wiener equation is calculated as eq. (4)

\[
W = (F^{-1}x)^2
\]

(4)

The following equation can be used to calculate the gradient is shown in eq. (5,6)

\[
g_0(m, n) = G_0, \sigma(m, n) * f_0(m, n)
\]

(5)

Where,

\[
G_0, \sigma = \frac{1}{\sqrt{2\pi\sigma^2}} \exp(-\frac{m^2+n^2}{2\sigma^2})
\]

(6)

Using any gradients operator [17], such as Roberts, Prewitt, Sobel, or another method, the gradient of \( g_0(m, n) \) is calculated to obtain in eq. (7)

\[
M_0(n, n) = \sqrt{g_{0R}^2(m, n) + g_{0I}^2(m, n)}
\]

(7)

D. Sobel Operator for Edge Detection

Edge detection is a technique for spotting breaks in images. Edge detectors come in a variety of varieties, including Sobel Operator, LoG Operator, Robert’s Operator, and Zero cross Operator, Canny Operator, and Prewitt Operator. The Sobel Technique is presented below among all of these. It serves as an edge detector with 33 gradients. The Sobel algorithm calculates the 2-D spatial gradient of a picture and highlights edge-corresponding regions with higher spatial frequency. When a grayscale image is provided, it is utilised to determine the roughly absolute magnitude of the gradient at each location. The Sobel Operator uses 2 3x3 matrix that are convolved with the starting image, one for horizontally variations and one for vertically shifts, for approximating the derivatives of the image. The Sobel operator may be utilised to calculate the predicted relative magnitude of gradients at each position of a given grey scale image by applying 2-D spatial gradient measurements on the image and emphasising regions of high spatial frequency that correspond to edges. Therefore, the Sobel operator is used in the edge identification of biomedical images [18].

E. Whale Optimization Algorithm

Mirjalili and Lewis developed the whale optimisation algorithm (WOA), a new metaheuristic technique. The global optimum answer for an issue is sought after and discovered using WOA, which works similarly as various metaheuristic optimisation methods. Update until the optimal value is attained, the algorithm keeps enhancing and updating the answer depending on its framework. The way the WOA principles update and refine the answer is the main distinction between the WOA and other metaheuristic algorithms. The WOA is modelled after the instinctual way that whales hunt prey—by spiralling around it to set up a trap and then hitting it. This type of eating behaviour is known as bubble-net feeding [19].

Before striking, the humpback whale spirals around its prey to produce bubbles. This consuming behaviour served as the basis for the WOA's primary structure. The bubble-net method's mathematical framework is described as follows in eq. (8) to (10).
\[ X_0(t + 1) = \begin{cases} X_0(t) - AD & P < 0.5 \\ D_0e_0^b\cos(2\pi t) + X_0(t) & P \geq 0.5 \end{cases} \] (8)

Where, \( D_0 \rightarrow \) Distance between the \( i^{th} \) whale and its victim (the ideal answer), \( P \) and \( r_0 \rightarrow \) random constants between 0 and 1, \( l \rightarrow \) a random constant between \([1, 1]\), \( t \rightarrow \) current iteration, \( b \rightarrow \) the logarithmic spiral’s form, and \( a \rightarrow \) linearly across the iteration, from 2 to 0.

The circling process is represented by the initial term in the formula above, while the bubble net process is represented by the following term. The extraction and utilisation terms of the WOA are represented by these two clauses. It shows how the prey is circled and how bubble-net hunting is done. The WOA begins with a randomly chosen populace, as was already mentioned. The results are then updated after every round in accordance with the mathematical model created for hunting using a bubble net and orbiting the prey. When \( |X_0| > 1 \), the optimum solution adjusts the location of the agents to ensure that the algorithm converges. If not, the most effective solution serves as a pivot point.

**Algorithm of WO-GCNN mechanism for detecting anaemia**

**Start**

**Input:** Initializing the whale population \( X_{0i} \)

Initializing \( a, A_0, \) and \( C \)

**Output:** Evaluating the most effective solution is displayed by search agents with fitness \( X^*_0 \) at this time.

**Applying WOA:** for feature selection

\[ t_0 = 1 \]

while doing \( t_0 < \text{max iterations} \)

for the agents do in all

if \( |A_0| = 1 \) then

End if

Updating the search’s current location

Else if \( |A_0| = 1 \) then

End if

Choose the \( X_{0\text{rand}} \) random search agent.

The computational difficulty of WOA is \( O_0 \) (Max Gen NP (fitness)), where Max Gen is the maximum number of generations and \( O_0 \) (fitness) is set by the application. This is because WOA has a fairly basic structure, based to the pseudocode [19].

**F. Convolutional Neural Network**

A set of layers collectively referred to as CNN are used for converting the input layer into an output layer. Each layer is made up of a collection of neurons. Each neuron in a layer, with the exception of the input layer, is the outcome of the task that was assigned to the neurons in the layer before it, i.e. \( y = f(x) \). The most often used layers are the convolutional layer, the fully connected layer, the pooling layer, and the activation layer. Fig. 2.

Each neuron in a layer that is fully connected is connected to all of the neurons in the layer below it. The strength of the connection connecting the \( j^{th} \) neuron in the current layer and the \( k^{th} \) neuron in the previous layer should be represented by the number \( \omega_{jk} \). Let \( b_{0j} \) be the bias of the \( j^{th} \) neuron in the current layer. The result of the layer’s \( j^{th} \) neuron is given by eq. (12)

\[ y_{0j} = \sum_k \omega_{jk} x_{0k} + b_{0j} \] (12)

The neurons in the convolutional layer that are typically used for generating a kernel or filter have identical biases and values. Each neuron in this layer will be connected to a \( n \times n \) region of the neurons in the layer beyond if the size of the filter is set to \( n \times n \). In line with this, the \( (j,k) \) \( j^{th} \) neuron’s outputs will be in eq. (13)

\[ Y_{0j,k} = \sum_{i=0}^{n-1} \sum_{m=0}^{n-1} \omega_{i,m} x_{0j+i,k+m} + b \] (13)

![Convolutional neural network architecture](image_url)
Tanh, Sigmoid, and the rectified linear unit, that has become the standard suggestion for modern neural networks, are instances of frequently employed functions for activation. Activation layers usually appear following convolutional or fully connected layers to offer elementwise non-linear behaviour. By using the activation function, ReLU is defined in eq. (14)

\[ f_{\text{ReLU}}(x) = \max(x, 0) \]  

(14)

The downwards sampling method for every sub-area in the pooling layer provides the dimension of a single neuron in the present one by dividing the neurons of the layer preceding it into an array of not overlapping rectangles. Maximum pooling and average-pooling, the two most popular pooling procedures, offer the subarea's maximum value and average value, respectively [20]. A convolutional neural network usually sets up a sequence of convolutional (Conv)-ReLU layers, before adding the pooling layers (Pool), and continues doing this till the picture gets spatially combined to a compact size. At certain points, it is usual to switch to fully-connected layers (FC).

V. RESULTS AND DISCUSSION

A normal blood sample is pre-processed, binarized, and the permeability of each connected component is determined using MATLAB 7.14.0.739 software. Similarly, an anaemic blood specimen is binarized, pre-processed, and the permeability of each associated component is determined.

A. Performance Evaluation

The classification methods’ performance was evaluated using the confusion matrix's assessment measures (Table II). Although the true negative (TN) represented the conjunctiva images that the machine learning algorithm accurately identified as not considered anaemic, the true positive (TP) represented the conjunctiva tissue images that the classification successfully classified as being in the anaemic class. False positives (FPs) were conjunct images of people who were not anaemic but who were wrongly classed as being anaemic. Finally, false negatives (FNs) were found in the conjunctival pictures of anaemic individuals who were mistakenly classified as not anaemic. Accuracy, sensitivity, and specificity are defined in accordance with these notations is given by eq. (15-17).

\[ \text{Accuracy} = \frac{TP + TN}{TP + FN + FP + TN} \]  

(15)

\[ \text{Sensitivity} = \frac{TP}{TP + FN} \]  

(16)

\[ \text{Specificity} = \frac{TN}{FP + TN} \]  

(17)

These metrics were calculated based on every cross-validation operation that was performed, and the effectiveness of the classifier was assessed using the mean and standard deviation of the outcomes. Although the accuracy measure is typically an accurate measure of a system's effectiveness, the dataset's imbalance between the two groups makes it less reliable [21]. Since the goal of an approach like the one suggested in this research is initially aimed at giving a patient who was not aware of being deficient in iron for carrying out further examinations, it was chosen to prioritise sensitivity, which says, as a percentages, the number of the actual anaemic individuals were identified by the system.

A graph called the receiver operating characteristic curve shows the way a classification system works at every level of categorization. The curve shows the following two variables: 100% True Positive. False Positive Rate. Fig. 4 shows the ROC Curve along with the false positive rate and true positive rate. To assess the ability to discriminate of each metric in identifying anaemia, ROC curves were generated [24].

<table>
<thead>
<tr>
<th>Classifiers</th>
<th>Accuracy</th>
<th>Sensitivity</th>
<th>Specificity</th>
</tr>
</thead>
<tbody>
<tr>
<td>KNN</td>
<td>67.4%</td>
<td>67%</td>
<td>84%</td>
</tr>
<tr>
<td>SVM</td>
<td>78%</td>
<td>71%</td>
<td>85%</td>
</tr>
<tr>
<td>NN</td>
<td>79.8%</td>
<td>65%</td>
<td>91%</td>
</tr>
<tr>
<td>Proposed WOG-CNN</td>
<td>99%</td>
<td>92%</td>
<td>96%</td>
</tr>
</tbody>
</table>

Fig. 3. Graphical representation of performance metrics.

The above graph (Fig. 3) shows that the accuracy, specificity and sensitivity of the existing and proposed model and ROC curve for false positive rate and true positive rate. This shows that the proposed method gives better efficiency.
B. Dataset Comparison

In Table III and Fig. 5, the performance of the proposed WOG-CNN model in comparison to the approach presented is evaluated using two datasets: the IDB Dataset and the ImageNet Dataset. The evaluation metrics used are sensitivity, accuracy, and specificity. The proposed WOG-CNN model consistently achieves higher specificity, accuracy, and sensitivity on both the IDB Dataset and the ImageNet Dataset compared to the approach.

![Dataset Comparison](image)

**Fig. 5.** Dataset comparison.

<table>
<thead>
<tr>
<th>Datasets</th>
<th>Accuracy</th>
</tr>
</thead>
<tbody>
<tr>
<td>IDB dataset [25]</td>
<td>98.92%</td>
</tr>
<tr>
<td>ImageNet dataset [26]</td>
<td>99.22%</td>
</tr>
</tbody>
</table>

The suggested WOG-CNN and SVM were two separate methods that were compared in this study for how well they performed on the IDB Dataset and ImageNet Dataset. On the IDB Dataset, the SVM approach produced results with an accuracy of 98.87%, a sensitivity of 91.65%, and a specificity of 91.46%. On the ImageNet Dataset, the results were even better with an accuracy of 99.54%, a sensitivity of 80.00%, and a specificity of 90.91%. The suggested WOG-CNN, however, performed better than SVM on both datasets, obtaining 99% accuracy on the IDB Dataset and 92% and 96% sensitivity and specificity, respectively. Additionally, the WOG-CNN attained an accuracy of 99.63% on the ImageNet Dataset, with a sensitivity of 87.1% and a specificity of 93.8%. These findings show that the suggested WOG-CNN outperforms the state-of-the-art in both datasets, underscoring its promise as a highly precise and reliable solution for the classification of images tasks, especially in the setting of anaemia diagnosis and beyond.

VI. CONCLUSION

This framework effectively recognises and categorises anaemia-related patterns and abnormalities in the blood smear images by combining the strength of generative algorithms and convolutional neural networks (CNNs). By optimising its parameters and accelerating its convergence, the Whale Optimisation Algorithm (WOA) is used in the framework to improve the CNN's performance. WOA is a metaheuristic algorithm inspired by nature that simulates the behaviour of humpback whales during hunting, making it appropriate for CNN optimisation. The WOGCNN framework's capacity to generate realistic synthetic blood smear images through learning helps it expand the training dataset and solve the issue of scarce data availability. This augmentation method enhances the CNN model's resilience and generalisation, enabling it to precisely identify anaemia in hidden blood smear images. The WOGCNN architecture has shown higher performance compared to conventional anaemia detection techniques through comprehensive trials and evaluations. It performs with excellent precision, sensitivity, and specificity in identifying different anaemia types and severity levels, making it an important tool for early diagnosis and treatment. This paradigm can aid in improving patient outcomes and lessening the strain on healthcare systems by enabling early detection and intervention. It is crucial to remember that additional analysis and validation are required in order to properly prove the efficacy and dependability of the WOGCNN architecture. To evaluate its effectiveness in real-world situations and guarantee its generalizability across various populations and locations, extensive clinical trials and validation on a variety of datasets are required. The proposed framework demonstrates highly accurate anaemia detection from blood smear images while achieving faster computational time.

The performance and utility of the WOG-CNN may be further improved by additional research and development in this field. The robustness and generalizability of the model could be improved by enlarging the dataset to include a wider range of samples from various populations. The accuracy and interpretability of the model may also be improved by investigating the incorporation of domain-specific information or expert annotations. The application of the WOG-CNN in actual clinical settings necessitates careful consideration of data privacy and regulatory compliance. Gaining the trust and approval of medical organisations and practitioners will depend on following ethical standards and making sure patient data is secure.

REFERENCES


A Transformer-CNN Hybrid Model for Cognitive Behavioral Therapy in Psychological Assessment and Intervention for Enhanced Diagnostic Accuracy and Treatment Efficiency

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Abstract—The use of Cognitive Behavioral Therapy (CBT) as a method for psychological assessment and intervention has shown to be quite successful. However, by utilizing advancements in artificial intelligence and natural language processing techniques, the diagnostic precision and therapeutic efficacy of CBT can be significantly improved. For CBT in psychological evaluation and intervention, we suggest a unique Transformer-CNN hybrid model in this work. The hybrid model combines the strengths of the Transformer and Convolutional Neural Network (CNN) architectures. While the CNN model successfully extracts local and global features from the input sequences, the Transformer model accurately captures the contextual dependencies and semantic linkages in the text data. It intends to enhance the model's comprehension and interpretation of the complex linguistic patterns involved in psychological evaluation and intervention by merging these two algorithms. On a sizable collection of clinical text data, which includes patient narratives, treatment transcripts, and diagnostic reports, we undertake comprehensive experiments. The proposed Trans-CNN hybrid model outperformed all other methods with an impressive accuracy of 97%. In diagnosing psychiatric problems, the model shows improved diagnosis accuracy and offers more effective therapy advice. Our hybrid model's automatic real-time monitoring and feedback capabilities also make it possible for prompt intervention and customized care during therapy sessions. By giving doctors a formidable tool for precise evaluation and efficient intervention, the suggested approach has the potential to revolutionize the field of CBT and enhance patient outcomes for mental health. In order to improve the diagnostic precision and therapeutic efficacy of CBT in psychological evaluation and intervention, this work provides a transformational strategy that combines the advantages of the Transformer and CNN architectures.

Keywords—CBT; psychological assessment; intervention; diagnostic accuracy; treatment efficiency; Transformer; CNN; NLP

I. INTRODUCTION

Cognitive Behavioral Therapy (CBT) is a widely used approach in psychological assessment to identify and understand various cognitive, emotional, and behavioral patterns that contribute to mental health disorders [1]. It focuses on exploring and modifying these patterns to improve psychological well-being [2]. The assessment helps identify specific areas of concern, symptoms, and the impact of these factors on the individual's daily life and functioning. Initial Assessment in CBT is defined as the therapist conducts an initial assessment to gather information about the client's history, presenting problems, and treatment goals. Structured or semi-structured interviews, questionnaires, and rating scales may be used to assess symptoms, cognitive biases, and maladaptive behavior [3]. The formulation outlines the interplay between thoughts, emotions, behaviors, and external factors, providing a conceptual framework for treatment planning. The assessment findings guide the development of a tailored treatment plan in CBT [4]. Specific treatment goals and strategies are identified to address the individual's unique difficulties and promote positive changes in thoughts, emotions, and behaviors. Assessment continues throughout the course of therapy to monitor progress, reassess symptoms, and adjust treatment strategies if needed. Regular feedback and outcome measures are used to evaluate the effectiveness of interventions and make informed treatment decisions. Psychological assessment in CBT is an ongoing process that informs treatment decisions and guides the therapeutic interventions. It helps therapists gain a comprehensive understanding of the individual's cognitive and emotional functioning, enabling them to tailor treatment approaches to meet specific needs and achieve positive outcomes.

Cognitive Behavioral Therapy (CBT) has emerged as a highly effective approach for psychological assessment and intervention. However, the diagnostic accuracy and treatment efficiency of CBT can be further enhanced by leveraging
advances in artificial intelligence and natural language processing techniques. In this study, we propose a novel Transformer-CNN hybrid model for CBT in psychological assessment and intervention. The hybrid model combines the strengths of both the Transformer and Convolutional Neural Network (CNN) architectures. The Transformer model captures the contextual dependencies and semantic relationships in the text data, while the CNN model effectively extracts local and global features from the input sequences. By integrating these two architectures, we aim to improve the model’s ability to understand and interpret the nuanced language patterns associated with psychological assessment and intervention. We conduct extensive experiments on a large dataset of clinical text data, including patient narratives, therapy transcripts, and diagnostic reports. In cognitive-behavioral therapy (CBT), the classification process involves categorizing individuals based on their presenting problems, symptoms, and assessment results. Standardized diagnostic standards, such as those listed in the Diagnostic and Statistical Manual of Mental Disorders (DSM-5), are first applied in this procedure, to evaluate whether the individual meets the criteria for a specific mental health disorder. Diagnostic interviews, structured questionnaires, and rating scales are commonly used to gather information about the individual’s symptoms, duration, and functional impairment [5].

In addition to diagnostic classification, the classification process in CBT extends to determining appropriate treatment recommendations for individuals. Based on the assessment findings, therapists match the individual’s diagnosis or presenting problems with evidence-based treatment protocols. This includes specific therapeutic techniques, intervention strategies, and self-help resources tailored to the individual’s needs. It’s important to note that the classification process is not static and may evolve throughout the course of therapy. Ongoing evaluation and monitoring of symptoms, progress, and treatment outcomes help therapists assess the effectiveness of interventions and make adjustments as needed. The classification process may be refined based on the individual’s response to treatment and changes in symptom presentation. The classification process in CBT provides a framework for understanding an individual’s mental health concerns, identifying specific symptoms, and tailoring treatment recommendations [6]. By systematically categorizing individuals based on their assessment results, therapists can provide targeted interventions and support to address their unique needs and improve psychological well-being. The results demonstrate that our hybrid model achieves superior performance compared to traditional CBT methods and standalone Transformer or CNN models. The model exhibits enhanced diagnostic accuracy in identifying psychological disorders and provides more efficient treatment recommendations [7]. Furthermore, our hybrid model also enables automated real-time monitoring and feedback during therapy sessions, facilitating timely intervention and personalized treatment. We utilize a sequential process that involves data collection, data preprocessing, model design, input representation, training, and optimization, and evaluation. The dataset is split into training, validation, and test sets, and appropriate loss functions and optimization algorithms are employed to train the model. Relevant parameters, such as accuracy, precision, recall, F1-score, or area under the receiver operating characteristic curve (AUC-ROC), are used to assess the model. The suggested model has the power to transform the field of CBT by providing clinicians with a powerful tool for accurate assessment and effective intervention, leading to improved mental health outcomes for patients. The primary objective of this study is to develop a novel approach to CBT in psychological evaluation and intervention that leverages the benefits of both the Transformer and CNN models. By doing so, we aim to improve the accuracy of psychiatric problem diagnosis and provide more effective therapy advice to patients. Here, introduces a Trans-CNN hybrid model that effectively merges the Transformer and CNN architectures. The CNN model excels at extracting local and global features from input sequences, while the Transformer model accurately captures contextual dependencies and semantic linkages in text data. By combining these two algorithms, our hybrid model offers a powerful tool for precise evaluation and efficient intervention in mental health care. Overall, this study presents a transformative approach that combines the strengths of the Transformer and CNN architectures to enhance the diagnostic accuracy and treatment efficiency of CBT in psychological assessment and intervention [8]. The key contributions of this research lie in the development of a hybrid model that combines Transformer and CNN architectures, aiming to enhance diagnostic accuracy, treatment efficiency, and personalization are as follows:

- Data is collected from the posts of depressed person.
- Initially, Text pre-processing methods such as is employed to pre-process the text in the collected data posts.
- Feature extraction is employed by utilizing transformer encoder using the self-attention method.
- Classification using Transformer-CNN Hybrid Model for CBT in psychological assessment and intervention.

The sections that make up the research’s structure are as follows: Section II describes related works that demonstrate a comparison of the proposed method with several other methods, Section III discusses the problem statement, Section IV defines the recommended approach for the Trans-CNN method, Section V elaborates the evaluation metrics, and Section VI describes the outcome and potential future research.

II. RELATED WORKS

The scientific community is becoming more interested in recognizing emotions as a result of the many fields it may be used in, such as health-care or traffic security schemes. D. Griol et al. [9] presents a speech- and facial-based multimodal emotion identification method. Researchers tested numerous transfer-learning methods, more especially embedding extraction, and fine-tuning, for the speech-based medium. The PANNs framework’s CNN-14 was fine-tuned to produce the most excellent accuracy results, demonstrating that instruction was more reliable when it didn’t start from scratch and when the assignments were comparable. Using saliency maps and
face pictures, we provide architecture for emotion-based facial recognizers consisting of a bi-LSTM with an attention mechanism after a pre-trained Spatial Transformer Network. Regardless of the domain change, the error analysis revealed that the frame-based systems could pose issues if employed directly for completing a video-based task. In order to resolve this discrepancy and make use of the inherent information of these pre-trained models, this brings up a new avenue of study. Finally, using a late fusion technique and these two modalities together, we classified eight emotions with 80.08% accuracy on the RAVDESS dataset using a subject-wise 5-CV assessment. The findings showed that various modalities convey pertinent data to identify users' emotional states, and their integration enhances the efficiency of the system. Despite domain adaptation, this study notes that frame-based systems may run into issues when utilized directly for video-based activities. This shows that using pre-trained models efficiently for video-based emotion identification may provide some difficulties. To overcome these issues and enhance the system's effectiveness in real-world circumstances, more study is required.

Y. Chen et al. [10] proposed the research paper “Screening Children’s Intellectual Disabilities with Phonetic Features, Facial Phenotype and Craniofacial Variability Index.” In this study, a kind of neurological deficit condition called intellectual disability (ID) is brought on by congenital disorders or things that happen after birth. If early screening for this syndrome was effective, the condition of patients may be improved, and their capacity for self-care increase. Clinical interviews are the only method for achieving early ID screening, and they necessitate the active involvement of medical specialists and assets associated with medicine. The methods employed in this research paper comprises that the analysis of young participants' phonetic traits and facial phenotype has been suggested as a potential way for screening ID. In order to compute the craniofacial variability index (CVI) and determine the likelihood of ID, the geometric aspects of the faces of the participants and phonetic characteristics of the subjects' voices are initially determined from interview footage. Furthermore, a technique for enhanced ID screening centered on visual and phonetic traits is established using machine learning algorithms. The Results suggested that the technique was assessed using three feature sets, including phonetic, geometric, and CVI characteristics. Accuracy performed best when it was near 80%. In conclusion, based on the findings obtained with the three feature sets, it can be said that, with further development, the suggested approach may be used in a clinical environment. The diversity of the dataset used to assess the suggested strategy is not disclosed by this approach. The method's applicability to larger populations may be in doubt if the dataset is restricted in terms of characteristics or certain ID subtypes.

Z. Chen et al. [11] proposed the research study “Detecting Reddit Users with Depression Using a Hybrid Neural Network.” In this study, a common mental health problem, depression is thought to afflict 3.8% of the world's inhabitants. Additionally, it is a significant factor in the global burden of disability. People are increasingly using social media sites like Reddit to communicate their problems and health issues, such as depression, and to ask for help from other users in online groups. By analyzing millions of messages for possible solutions, it creates a fantastic opportunity for machines to identify social media users who are depressed. Since deep learning techniques are simple to use, analyze information quickly, and produce cutting-edge results on a variety of NLP tasks, they have started to take the lead in the machine learning and natural language processing (NLP) fields. In this study, researchers present a hybrid deep learning model to identify depressed people from Reddit postings using a pre-trained sentence BERT (SBERT) and convolutional neural network (CNN). To learn the meaningful representation of semantic information in every message, the sentence BERT is employed. CNN makes it possible to further alter these embeddings and to identify user behavior trends over time. Using the Self-reported Mental Health Diagnoses (SMHD) data, we educated and assessed the model's ability to detect Reddit users who are depressed. The hybrid deep learning model exceeded the most recent results (F1 scores of 0.79 by other machine learning models in the literature) by achieving an accuracy rating of 0.86 and an F1 score of 0.86. The outcomes demonstrate the viability of using a hybrid model to detect depression in people. The hybrid model may easily be tweaked and used for other text classification tasks as well as many clinical scenarios, despite its validation to identify sadness using Reddit postings. The Self-reported Mental Health Diagnoses (SMHD) data are mentioned in this study as a training and assessment tool. Self-reported diagnoses, however, are not always precise or trustworthy, which might affect the quality of the labeled data and, in turn, the efficiency of the predictive algorithm.

S. Aleem et al. [12] presents ML techniques to identify the depressive disorder. In this research, Stress, worry, and today's fast-paced lives have all had a significant psychological impact on people's minds over time. The worldwide technical advancement in healthcare digitizes the relevant data, making it possible to map the many facets of the human condition more precisely than with conventional measurement methods. For analyzing the vast quantity of data in the healthcare sector, machine learning (ML) has been recognized as an effective method. In order to forecast the likelihood of mental diseases and, consequently, execute prospective therapies, ML techniques are being used in the field of psychological wellness. Numerous machine-learning methods for recognizing and diagnosing sadness are discussed in the above article. Classification, deep learning, and ensemble are the three categories of ML-based sadness detection techniques. In this article, we provide a basic paradigm for detecting depression that entails data gathering, pre-processing, ML classification training, detection classification, and performance evaluation. Additionally, it provides a summary of the goals and restrictions of the many research papers that have been presented in the field of depressive disorder detection. Additionally, it covered potential directions for future study in the realm of diagnosing depression.

N. Shusharina et al. [13] presents the early detection and treatment of neurodegenerative and depressive illnesses utilizing machine learning technologies. Using disability-
adjusted decades as a measure, the 2 types of illnesses rank among the top global contributors to the loss of quality of life. Despite decades of study, the creation of fresh methods for diagnosing and treating depressive conditions and neurodegenerative illnesses continues to rank among the most important topics of study in psychology, neurophysiology, genetics, and interdisciplinary healthcare. Recent study possibilities are made possible by modern machine learning techniques and health information technology. Prior to the improvements being extensively adopted in clinics, it is still a struggle to obtain agreement regarding how new machine learning techniques should be applied and how they should be integrated with current norms of care and assessment. A cohesive strategy for utilizing the expanding body of clinical data is being developed as a result of research on clinical predictions and categorization techniques. The demands of doctors, researchers, and government authorities ought to be taken into account in this coordinated strategy. The present situation of investigation into neurodegenerative and depressive illnesses is given in the current study. However, because the results of these techniques take time to manifest, assessing their efficacy might be more difficult.

J. J. Wood et al. [14] presents CBT strategy effective for kids with autism spectrum disorders and disruptive anxiety. This research demonstrates that the children and adolescents with autism spectrum disorder (ASD), anxiety is prevalent and frequently interferes with adaptive behavior. The effectiveness of psychological treatments, which are frequently employed to treat school-aged children with autism spectrum disorder, is yet to be proven. To compare the relative efficacy of two cognitive behavioral therapy programmes with standard care and to assess the impact of treatment on disruptive and unhelpful anxiety in children with autism spectrum disorders. Further objectives included assessing how the treatment affected the degree to which autistic spectrum disorder symptoms were present and how well the patient could adjust to their worry. The randomized clinical study's recruitment period began in April 2014 at three universities in American cities. Random assignments were made to provide conventional cognitive behavioral treatment, CBT specifically designed for autism spectrum disorders, or TAU to children with autism and maladaptive and disruptive stress. The combination wasn’t revealed to the impartial judges. Data was gathered up until January 2017 and examined between December 2018 and February 2019. The major components of standard-of-practice CBT were reappraisal, modeling/rehearsal, in vivo exposure tasks, affect recognition, and reinforcement. Similar to the original cognitive behavioral therapy intervention, the autism spectrum disorder adaptation also addressed difficulties with interacting with others and self-regulation utilizing perspective-taking training and behavior-analytic methods. According to the a priori hypothesis, the primary measurement of outcome was the Paediatrics Anxiety Rating Scale. Secondary outcomes included the Clinical Global Impressions-Improvement scale response to therapy and assessment evaluations. In the current study, cognitive behavioral treatment was useful for children with restless anxiousness and autism spectrum disorders, and a personalized CBT method showed further advantages. If therapists are providing emotional support to school-aged children with ASD, they ought to think about learning CBT skills. Random slopes for a site, on the other hand, were not statistically significant and were therefore excluded from the final models because they did not provide any proof of differential treatment effects at certain sites.

### III. Problem Statement

Existing traditional CBT methods have certain limitations, such as heavy reliance on manual evaluation and the potential for human errors, which hinders achieving optimal diagnostic accuracy and treatment efficiency. These limitations create a gap in providing timely and precise interventions to patients. To address these issues, our proposed hybrid model leverages the power of artificial intelligence and integrates the strengths of both the Transformer and CNN architectures. The Transformer-CNN hybrid model enables a more automated and accurate assessment of psychological disorders compared to manual evaluation. The Transformer component captures contextual dependencies and semantic linkages in the text data, allowing for a deeper understanding of complex linguistic patterns involved in psychological assessment. Meanwhile, the CNN component effectively extracts local and global features from the input sequences, enhancing the model's ability to recognize important patterns relevant to psychiatric diagnoses.

By capturing and analyzing comprehensive patient narratives, therapy transcripts, and diagnostic reports, the hybrid model facilitates more efficient and data-driven treatment recommendations. This automation significantly reduces the time and effort required for psychological assessment and intervention, enabling clinicians to focus more on delivering personalized and effective treatments. The limitations of traditional CBT methods, such as manual evaluation and the potential for human error, are not suitable for achieving optimal diagnostic accuracy and treatment efficiency in modern mental health care. The proposed hybrid model's integration of artificial intelligence and natural language processing capabilities addresses these limitations by providing automated and precise assessments, leading to improved mental health outcomes for patients [15].

### IV. Methodology

The study project's technique entails a number of crucial elements. Data preprocessing is the first step, where the collected clinical text data is processed using text preprocessing techniques like stop word removal, punctuation removal, tokenization, and stemming/lemmatization to standardize vocabulary, process the textual content, and reduce noise in the data. The architecture of the Trans-CNN model is then developed. Concatenation is utilized to conduct feature extraction, fusing together the output features from the Transformer and CNN models into a single fused feature representation. The Hybrid Trans-CNN technique is used for categorization. A classification model, such as a fully connected neural network or another appropriate algorithm, receives the combined information. To understand the correlations between the characteristics and the goal variable, which might be a label for a diagnosis or a prescription for a course of therapy, this model is trained using labelled data. Fig. 1 represents the entire process of the proposed approach.
A. Dataset

The training set, validation set, and testing set are the three primary sets that make up the dataset for "Enhanced Diagnostic Accuracy and Treatment Efficiency." Each group is further separated into the depression and control categories. A total of 2,632 users make up the training set, with 1,316 users in the depression category and 1,316 users in the control category distributed equally. These users have made a significant contribution to textual data, totaling 609,471 postings. There are 216,022 postings in the depression category and 393,449 in the control category. This dataset forms the basis of our hybrid model’s training. The validation set is similarly divided into control and depression groups, 2,616 user’s altogether, including 1,308 individuals in each category. There are 684,788 postings total in this corpus of text data. The control category has 393,930 posts, and the depression category is made up of 290,858 posts. The model’s performance throughout the training phase is evaluated using the validation set, and any necessary corrections are then made. The testing set is then used to assess the hybrid model’s ultimate performance and generalizability. With 1,316 users in each category, it has the same number of users as the training and validation sets.

<table>
<thead>
<tr>
<th>Dataset</th>
<th>Labels</th>
<th>Total Users</th>
<th>Total Posts</th>
</tr>
</thead>
<tbody>
<tr>
<td>Training Set</td>
<td>Depression</td>
<td>1,316</td>
<td>216,022</td>
</tr>
<tr>
<td>Control</td>
<td>1,316</td>
<td>393,449</td>
<td></td>
</tr>
<tr>
<td>Validation Set</td>
<td>Depression</td>
<td>1,308</td>
<td>290,858</td>
</tr>
<tr>
<td>Control</td>
<td>1,308</td>
<td>393,930</td>
<td></td>
</tr>
<tr>
<td>Testing Set</td>
<td>Depression</td>
<td>1,316</td>
<td>209,188</td>
</tr>
<tr>
<td>Control</td>
<td>1,316</td>
<td>390,385</td>
<td></td>
</tr>
</tbody>
</table>

In Table I, the testing set contains 599,573 posts in total, with 209,188 posts in the depression category and 390,385 posts in the control category. By utilizing this extensive dataset, which includes a significant number of users and posts in both depression and control categories, we aim to train and evaluate our hybrid model for cognitive behavioral therapy in psychological assessment and intervention. The dataset provides a diverse range of real-world data that enables us to enhance the diagnostic accuracy and treatment efficiency of our proposed model [11].

B. Data Pre-processing

For the research project "A Transformer-CNN Hybrid Model for Cognitive Behavioural Therapy in Psychological Assessment and Intervention for Enhanced Diagnostic Accuracy and Treatment Efficiency," a number of text-specific techniques can be used to prepare the textual data for analysis. These methods include dealing with stopwords, removing punctuation, tokenization, and stemming/lemmatization [16].

1) Stopword elimination: Stopwords are frequently used words in a language, including "a," "an," "the," and "is," that have little or no significance. The elimination of these words from the text data is known as stopword removal because they can amplify noise and add dimensions to the data without adding anything to the study. First, stopwords—common words with little actual meaning—are eliminated from the text data. These contain the letters "a," "an," "the," and "is." By getting rid of these stopwords, the emphasis is put on words that help the reader grasp the text semantically.

2) Punctuation removal: For the analysis of textual data, punctuation markers like periods, commas, and quote marks are typically not necessary. Eliminating punctuation marks makes the text simpler and helps readers concentrate on the words' true meaning and content. By removing pointless symbols from the input, this process makes it simpler for the model to focus on the text's underlying meaning.

3) Tokenization: Tokenization is the process of separating the text into tokens, which are distinct words. To determine the borders of words, it divides the text depending on whitespace or punctuation. By helping to provide a structured representation of the text, tokenization enables word-level data analysis in the model. Tokenization is used in the text pre-processing stage to separate the text into distinct words or tokens. Tokenization facilitates the division of the text into digestible chunks for subsequent processing. This is usually accomplished by segmenting the text according to whitespace or punctuation. For instance, the terms "I," "love," "to," "play," and "soccer" would be tokenized from the statement "I love to play soccer" [17].
4) Stemming/Lemmatization: Stemming or lemmatization techniques may be used to further the investigation. Lemmatization changes words to their base form, whereas stemming lowers words to their base or root form by deleting suffixes. By reducing the dimensionality of the data and standardizing the vocabulary, these strategies help the model comprehend the subtleties of the text’s semantic meaning. Stemming is the process of eliminating suffixes from words to get the original stem form (for example, "running" to "run"). On the other side, lemmatization reduces words to their simplest form (e.g., "better" becomes "good"). By lowering the dimensionality of the data and standardizing the vocabulary, these strategies help the model better capture the semantic meaning of the words [18].

The textual data is cleaned, streamlined, and altered using various pre-processing approaches so that it may be successfully analyzed by the Transformer-CNN hybrid model. This guarantees that the model may concentrate on the text’s significant content and gather the pertinent data for improved cognitive behavioral therapy diagnosis accuracy and treatment effectiveness.

C. Design of Trans-CNN Model Architecture

To improve cognitive behavioral therapy (CBT) in psychological evaluation and intervention, the Hybrid Transformer-CNN model integrates the Transformer and Convolutional Neural Network (CNN) architectures in this research.

Fig. 2 showcases the integration of the Transformer and Convolutional Neural Network (CNN) architectures, combining their respective strengths to effectively process clinical text data. The CNN component excels at extracting local and global features from input sequences, capturing important patterns within the text. Begin to examine the operation of this hybrid model: Text data, such as patient narratives, treatment transcripts, or diagnostic reports, are inputted into the model. Using methods like word embeddings or character-level embeddings, the input text is transformed into numerical representations appropriate for the model. In this stage, the text is transformed into a format that the model can understand. The hybrid model incorporates the Transformer architecture. Contextual dependencies and semantic linkages in the input sequence are well-captured by the Transformer. Self-attention methods are used to help the model concentrate on crucial sections of the input sequence and recognize the relationships between various words or phrases. To increase the model's comprehension of the text data and capture increasingly complicated patterns, many Transformer layers are layered. The hybrid model incorporates the CNN architecture. The extraction of local and global information from input sequences by CNNs is excellent. Different patterns and characteristics are captured by applying convolutional filters of different sizes to the input sequence. In order to extract the most important characteristics from the convolved sequences, pooling procedures like max pooling are performed. To capture more complicated patterns in the text input, many CNN layers are layered. A unified representation is created by combining the output representations from the Transformer and CNN components. This fusion stage tries to take use of the complimentary characteristics of both architectures, enabling the model to successfully capture both local features and contextual relationships. The dimensions can be aligned and the characteristics from both components combined using methods like concatenation or other linear transformations. To anticipate the intended results, such as detecting psychiatric problems or offering treatment suggestions, a classification layer is put on top of the fused characteristics. The final predictions are produced using activation functions like softmax or sigmoid. The model is trained to use suitable loss functions, such as cross-entropy, to optimize the classification aim. The hybrid model is first trained using a dataset of clinical text data, and then training and optimization are carried out for improved process functioning. In most cases, the dataset is divided into training, validation, and test sets. Using the training set of data, back-propagation and optimization algorithms like Adam or SGD are used to modify the model's parameters. Hyper-parameters, such as learning rate and batch size, are tuned through experimentation and validation performance.
D. Feature Extraction using Transformer Encoder

The transformer architecture contains a decoder that generates the expected output of the input text and an encoder that receives the text input. The BERT model solely makes use of the transformer encoder. Each token in a phrase is represented by the transformer encoder using the self-attention method according to other tokens. Three vectors are created for each token in this self-attention process, and they are: a query vector $L$, a key vector $M$, and a value vector $N$. These three vectors were created by multiplying the embedding vector $x_i$ with three weight matrices $(W_L, W_M, W_N)$ respectively. If $d_m$ is the dimension of the key and query vectors, then the output $K$ of self-attention for each word is calculated based on the following Eq. (1):

$$K = softmax \left( \frac{L \cdot MT}{\sqrt{d_k}} \right) N$$

The self-attention is computed eight times with eight separate weight matrices, providing eight $Z$ matrices, since the transformer encoder employs a multi-head attention mechanism to focus attention on various spots. The next step is to blend the eight $Z$ matrices into one single matrix, multiply it by an extra weight matrix, and then transmit the resultant matrix to a feed-forward layer. The model can efficiently capture dependencies and long-range interactions within the input sequence thanks to the self-attention mechanism in the Transformer encoder. By paying attention to pertinent terms, the model may train itself to give the most informative elements of the input more weight throughout the encoding phase [19].

E. Classification using Hybrid Trans-CNN Approach

To create predictions or categories the input data, the features collected from the Transformer encoder and the features retrieved by the Convolutional Neural Network (CNN) are merged. The embeddings are input into the CNN component of the model after the text data has been turned into contextual embeddings using the Transformer encoder. By using convolutional procedures, the CNN is in charge of separating local and global characteristics from the embeddings. The CNN performs convolutional operations by applying a series of filters to the embeddings to capture distinct patterns and features at various sizes. This enables the model to extract from the text both fine-grained features and more general contextual information. A pooling layer, such as max pooling or average pooling, is used after the CNN layer’s output to lessen the dimensionality of the recovered features while retaining the most important data. The features from the CNN and Transformer encoder are combined or concatenated after pooling. In order to provide a more complete representation of the input data, the model can use both the contextual data collected by the Transformer and the local patterns collected by the CNN. The fully linked layers serve as a classifier after the concatenated features have been passed through them. The desired number of output classes or labels is mapped to the features using these layers. The final probabilities or forecasts for each class may be produced using activation functions, such as softmax or sigmoid, depending on the particular job. The Transformer-CNN hybrid model effectively captures semantic relationships, local patterns, and global context in the text data by combining the strengths of both the Transformer and CNN architectures. This improves diagnostic precision and treatment effectiveness in the field of cognitive behavioral therapy.

**Algorithm:** Trans-CNN model

**Input:** Text data for psychological assessment and intervention  
**Output:** Predict diagnostic labels and treatment recommendations

1. Load the input text data.  
2. Pre-process the given data  
3. Apply tokenization to convert the text into a sequence of tokens.  
4. Remove stopwords, punctuation, and perform stemming/lemmatization  
5. Encode the input text sequence using the Transformer encoder  
6. Pass the contextual embeddings through the CNN layers  
7. Apply convolutional operations to extract local features.  
8. Concatenate the features from the Transformer encoder and CNN.  
9. Compute the features of transformer encoder using Eq. (1)  
10. Combine the features to create a comprehensive representation of the input.  
11. Pass the fused features through fully connected layers.  
12. Generate the final probabilities or predictions for each class.

V. RESULT AND DISCUSSION

In this study, we aim to evaluate the performance of our proposed Transformer-CNN Hybrid Model for cognitive behavioral therapy (CBT) in psychological assessment and intervention. To assess the effectiveness of our model, we compare it with existing methods commonly used in CBT. To compare our Transformer-CNN Hybrid Model with existing methods, we selected a set of well-established approaches commonly used in CBT. These methods include traditional CBT techniques, standalone Transformer models, and standalone CNN models. We trained and evaluated these methods on the same dataset using the same evaluation metrics. The results of our experiments demonstrate that the Transformer-CNN Hybrid Model outperforms the existing methods in terms of diagnostic accuracy and treatment efficiency. The model shows superior performance in accurately identifying psychological disorders and providing personalized treatment recommendations. Additionally, the real-time monitoring and feedback capability of our model further enhances the efficiency of the intervention process. Our findings indicate that the integration of the Transformer and CNN architectures in our hybrid model leads to significant improvements in CBT outcomes. The combined strengths of both architectures enable a more comprehensive understanding of the nuanced language patterns associated with psychological assessment and intervention.

The Table II displays the accuracy results of different methods used in cognitive behavioral therapy (CBT) for psychological assessment and intervention. The 300 dim CNN-GloVe models achieved an accuracy of 81.3%, while the SBERT-CNN model obtained an accuracy of 86%. The AiME method, incorporating multimodal deep networks with LSTM,
achieved an accuracy of 69.23%. However, the proposed Trans-CNN hybrid model outperformed all other methods with an impressive accuracy of 97%. This model combines the strengths of the Transformer and CNN architectures to enhance diagnostic accuracy and treatment efficiency in CBT. The superior performance of the Trans-CNN model suggests its potential for improving the identification of psychological disorders and providing personalized treatment recommendations.

The graph in Fig. 3 shows the accuracy percentages achieved by different methods for cognitive behavioral therapy (CBT) in psychological assessment and intervention. The proposed Trans-CNN model achieves the highest accuracy of 97%, outperforming the other methods.

The Table III presents precision, recall, and F1-score results for different methods. The 300 dim CNN-GloVe models achieved a precision of 53.8%, recall of 70.8%, and an F1-score of 61.1%. The SBERT-CNN model demonstrated higher performance with a precision of 85%, recall of 87%, and an F1-score of 86%. Another method, XGBoost, achieved a precision of 82%, recall of 29%, and an F1-score of 43% is depicted in Fig. 4. However, the proposed Trans-CNN hybrid model outperformed all other methods, achieving a precision of 91%, recall of 95%, and an impressive F1-score of 94%. These results indicate that the Trans-CNN model excels in accurately identifying psychological disorders and provides a balanced trade-off between precision and recall.

The graph in Fig. 4 shows the comparison graph of precision, recall and F1-score.}

The research conducted extensive experiments to evaluate the performance of the proposed Transformer-CNN Hybrid Model for cognitive behavioral therapy (CBT) in psychological assessment and intervention. The following key findings were obtained: Enhanced Diagnostic Accuracy: The hybrid model achieved superior diagnostic accuracy compared to traditional CBT methods and standalone Transformer or CNN models. It demonstrated the ability to accurately identify psychological disorders based on the analysis of text data, including patient narratives, therapy transcripts, and diagnostic reports. This improvement in diagnostic accuracy can lead to more effective treatment planning. Treatment Efficiency and Personalization: The hybrid model provided more efficient and personalized treatment recommendations. By analyzing the input text data, the model extracted relevant features and generated tailored treatment suggestions based on the identified psychological condition. This personalized approach can significantly enhance the effectiveness of therapy and improve patient outcomes. Real-time Monitoring and Feedback: The proposed model enabled automated real-time monitoring and feedback during therapy sessions. This feature allowed for timely intervention and adjustments in the treatment plan, ensuring that patients receive the necessary support and guidance when needed. Real-time monitoring and feedback contribute to the overall effectiveness of CBT. Generalizability and Adaptability: The results demonstrated the generalizability and adaptability of the hybrid model. While initially developed for CBT in psychological assessment and intervention, the model showed potential for application in other text classification tasks and clinical settings. This versatility makes it a valuable tool for various mental health-related applications. The results highlight the effectiveness of the proposed Transformer-CNN Hybrid Model in improving diagnostic accuracy, treatment efficiency, and personalization in CBT. The findings underscore the potential of integrating advanced deep learning techniques with psychological assessment and intervention, paving the way for more efficient and effective mental health care.

While the proposed Trans-CNN model shows promising results, there are several avenues for future research. Firstly, incorporating additional modalities such as audio or

<table>
<thead>
<tr>
<th>Method</th>
<th>Accuracy (%)</th>
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<tr>
<td>Proposed Trans-CNN</td>
<td>91</td>
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</table>

<table>
<thead>
<tr>
<th>Method</th>
<th>Precision (%)</th>
<th>Recall (%)</th>
<th>F1-score (%)</th>
</tr>
</thead>
<tbody>
<tr>
<td>LSTM [11]</td>
<td>74</td>
<td>79</td>
<td>77</td>
</tr>
<tr>
<td>Proposed Trans-CNN</td>
<td>91</td>
<td>95</td>
<td>94</td>
</tr>
</tbody>
</table>
physiological data could enhance the model's performance and provide a more comprehensive understanding of psychological assessments. Additionally, exploring different variations of the Transformer-CNN architecture, such as incorporating attention mechanisms or exploring different fusion strategies, could further improve the model's effectiveness.

REFERENCES


Abstract—Mental illnesses are severe obstacles for the global welfare. Depression is a psychological disorder which causes problems to the individual and also to his/her dependents. Machine learning based methods using audio signals can differentiate patterns between healthy and depressive subjects. These methods can assist health care professionals to detect the depression. Literature in depression detection, based on audio signals, used only single classifier, lacks to take advantage of diverse classifiers. The current work combines predictive capabilities of diverse classifiers using stacking method to detect depression. Audio clips are reordered while a predefined paragraph is being read out, for acoustic analysis of speech. The dataset is created which has extended Geneva Minimalistic Acoustic Parameter Set (eGeMAPS) features, that are extracted using openSMILE toolkit. The normalized feature vectors are given as input to multiple classifiers to give an intermediate prediction. These predictions are combined using a meta classifier to form a final outcome. K-Nearest Neighbours (KNN), Naïve Bayes (NB), Support Vector Machine (SVM), and Decision Trees (DT) classifiers are utilised on the normalized feature vector for intermediate predictions and Logistic Regression (LR) is used as meta classifier to predict final outcome. Our proposed method of using diverse classifiers achieved significant accuracy of 79.1%, precision of 83.3%, recall of 76.9% and F1-score of 80% on our dataset. Results are discussed while using stacking method on our dataset, then compared with various baseline methods also while applying on a publicly available benchmarking dataset. Our results showed that combining predictive capability of multiple diverse classifiers helps in depression detection.

Keywords—Health care; depression detection; acoustic features; speech elicitation; feature selection; openSMILE; ensemble methods

I. INTRODUCTION

Mental health disorders are key impediments to the global health agenda's progress (World Health Organisation, 2020). According to WHO data, about 280 million people suffered from depression in 2019 including 23 million children [1]. The COVID-19 epidemic has become a worldwide health disaster. The epidemic has resulted in various damages viz. massive employment loss, lowered earnings, halted children's education, and hindered economic progress. During COVID-19 pandemic, several lockdowns were imposed which made people to stay indoors, for quite a long period. Measures like stay-at-home orders, isolation and quarantine further resulted in emotional, financial risks and aroused several mental health disorders, mainly depression. Depression is one of the world's most accepted serious health issues. Depression is the root cause for triggering other mental health disorders, if not attended properly. Even it leads to psychosomatic disorders, in which actual physiological symptoms will surface, on diagnosis. Thus the detection of depression early and its treatment plays vital role in maintaining health. The study focuses on detection of depression, with more accuracy, with audio signals, using speech elicitation from the subjects.

Generally, there are two methods for depression detection. These are 1) based on standard questionnaires and 2) with the interaction of health care professionals. These methods have their own disadvantages. The method using standard questionnaires needs proper attention while administering them. Health care professionals like Psychiatrists, Psychologists, and Counselors involve upon their availability and expertise in their domain. With the rapid growth and advancement in Information Technology sector, there is an increasing interest in the researchers to develop Machine Learning (ML) models in health care domain for disease diagnosis. ML models learn the meaningful patterns from the individual biomarkers [2], [3]. The automatic depression detection method using audio is the ‘Gold standard’ among machine learning researchers. Such a method can be utilised as a pre-assessment test for a depression suspect (also named as client) at home. This can benefit users by saving time and cutting down expenses and travel costs which help in the speedy and efficient diagnosis. Subsequently, it results in timely medical care which can be given to identify clients, now named as patients [4], [5].

The studies of audio-based depression detection were started in the late 1980s. Speech has several qualities that make it a desirable candidate for integration into an automated assessment system, according to Scherer et al. [6]. It offers reasonably priced, remote, non-intrusive measurement capabilities. Due to the complexity of speech production, it acts as a sensitive output system, where even slight changes in physiology and cognition can produce audible changes in sound. It is conceivable that the acoustic quality of speech may be impacted in a quantifiable and objectively measurable way given the anticipated effects of depression on both cognitive and physiological aspects that influence speech control.

Studies in developing machine learning models to detect depressed and control subjects support that a person’s verbal communication can indicate depression [7], [8]. During interaction between the clinicians and the subject (client), health care professionals rely on the subject’s acoustic cues to

A Stacking-based Ensemble Framework for Automatic Depression Detection using Audio Signals

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assess their level of depression. Speech-based depression methods suggest that there exists a relationship between acoustic speech and depression. Typically, the clinician observes acoustic speech (how they speak) as a main parameter [9]. So ideally, the automatic audio-based depression methods have to consider clinician’s acoustic observations in the assessment of depression.

More over the use of audio-based approaches for diagnosing depression has the advantage of protecting individuals’ privacy and identification. Visual-based approaches, on the other hand, cannot provide the same level of privacy and may reveal the person’s identity known throughout the assessment process. The literature on depression detection methods demonstrates a strong preference for multimodal approaches [10], with audio-based methods receiving lesser attention. Rather than exploring on unimodal systems, researchers have largely concentrated on integrating many modalities. These multimodal systems incur costs for data collecting and computing time, which might impair processing speed. The existing studies also lack the larger dataset, and relatively lesser number of studies were reported using ensemble methods. The current study investigated on the stacking method of ensemble method which is not utilised in depression detection but utilised in other applications of medical diagnosis problems [11], [12], [13].

To overcome these issues we developed a method to create a dataset of voice samples. We have extracted numerous features by using state-of-the-art feature extraction tool kit. We have investigated with two ensemble techniques. Then, we conducted experiments with four developed models that classify depressed and healthy subjects. The present work investigates the acoustic components of speech, to design a comprehensive speech-based depression diagnosis method. An end-to-end machine learning solution was proposed to classify subjects’ depressed and non-depressed classes. We believe that such a method can be more effective in the diagnosis of depression.

Main contributions of the current work are:

- To develop a ML model that will classify a subject as being healthy or depressed.
- To create a Dataset of voice samples using speech elicitation technique where subjects were instructed to provide voice samples in two different scenarios: i) while participants are reading out a phonetically balanced short story called as “The north and the south wind” ii) while participants are giving an open form of talk on the choice of their interest.
- To extract baseline features in acoustic speech recognition called as eGeMAPS feature set using state-of-the-art feature extraction toolkit called as openSMILE.
- The ensemble method called stacking was experimented and results were analysed on the created dataset. The performance is compared with the baseline methods and evaluated on the publicly available DAIC-WOZ dataset. Their results are reported.

II. RELATED WORK

Speech plays a crucial role in detecting depression for the following reasons: First, the acoustic component of speech is often influenced by the subject’s mental state. Second, the clinician observes acoustic speech characteristics to analyse the condition of the individual. Several studies have found distinguishable acoustic characteristics between non-depressed and depressed individuals. Third, ease of recording and the availability of open source tools for extracting acoustic features such as openSMILE, COVEREP, etc. [14], [15]. For instance, studies employing the acoustic tier of speech indicated that depressed subjects show variations in measures like pitch, intensity, energy, etc., compared with non-depressed speakers [16]. Studies on non-voiced aspects of speech like pauses, hesitations reveal that they do have an association with the depression.

The field of mental health care has seen promising applications for audio-based depression detection techniques. These methods examine audio recordings, such as speech patterns and emotional signals, to find probable symptoms of depression by utilising machine learning techniques. An inexpensive and non-intrusive method of screening people for depression symptoms is, automated diagnosis by audio analysis, which enables early intervention and support. Sentiment analysis is another application in speech, which enables the detection of unfavourable emotions and depressive symptoms, offering insightful information about a person’s mental condition. Additionally, depression-related emotional patterns can be recognised using speech emotion recognition applications, allowing for a more thorough understanding of a person’s emotional health [17].

Andrew et al. in [18] examined the potential gender bias in automated depression detection systems based on audio features. They analysed on a publicly available dataset called DAIC dataset of clinical interviews among participants with and without depression. They found that the classification accuracy is significantly higher for female participants compared to male participants, which is partly due to differences in the acoustic characteristics of male and female speech. Their work suggests the importance of considering gender bias in the development and evaluation of automated depression detection systems. Speech is used in prediction of not only depression but also in other diseases like dementia, stress disorders, parkinson’s disease etc. [19], [20].

Espinola et al. [21] recorded audio samples during the interaction of subjects (11-control and 22-depressed) with the psychiatrist. The samples were processed with the audacity audio software to eliminate the interviewee’s voice signals and other noises. These samples were provided as input for an open-source vocal feature extraction tool called GNU Octave, to extract statistical features. Further, the Weka tool was used for analyzing the classification results using different ML classifiers. Among all classifiers that were experimented, random forest classifier achieved 87.5% accuracy. Their study concluded that the acoustic tier of voice samples is promising for depression detection.

Wegina et al. in [22] conducted a study on 144 volunteers, out of which 54 were diagnosed with depression and 90 as
healthy subjects. Audio samples were recorded while participants responded to a personal interview adapted by vocal screening protocol [23]. Subsequently, voiced responses of participants were also recorded while they were responding to the self-report questionnaires called Beck Depression Inventory (BDI) [24] and Self Reporting Questionnaire (SRQ) - 20 questionnaires [25]. The vocal samples were processed by the PRAAT tool for feature extraction. Acoustic parameters of speech, such as pitch, jitter shimmer, etc., were extracted using PRAAT. Then statistical features like median, mean and standard deviation, etc., were extracted from the acoustic parameters. A multiple linear regression model was used to demonstrate that acoustic parameters of speech were discriminative indicators for depression.

Thati et al. in [26] collected audio data through a mobile application that recorded participants’ speech during a task-based depression assessment. They extracted various audio features, including pitch, loudness, and spectral features, and used machine learning algorithms to classify the audio recordings as either depressed or not. Their study found that audio features were effective in distinguishing between non-depressed and depressed participants, achieving an accuracy of 86.3%. Thati et al. in [27] extended their work with different fusion strategies to detect depression. In their work, they combined audio, video, text and also smart phone usage records to build a multimodal feature vector. Among the multimodal feature vectors audio features showed more correlation than the other modalities. Their work suggested that audio features are highly correlated with the depressive behaviour and resulted in efficient depression diagnosis.

Naulegari Janardha et al. in [28] proposed a novel method to enhance the prediction accuracy of depression using speech’s acoustic features. Their study employed the Fisher score-based feature selection to choose the most informative features and dynamic ensemble selection to enhance the classification performance. They conducted experiments on a dataset of speech recordings from both depressed and non-depressed individuals. Their findings demonstrate that the proposed approach significantly outperforms existing methods, achieving an accuracy of 82.1%.

Brain sumali et al. in [29] used acoustic features to detect depression and dementia. Authors showed the statistical significance between the acoustic features and depression. In their work, they experimented with feature selection method called as Least Absolute Shrinkage and Selection Operation (LASSO) and SVM classifier with linear kernel to predict depression. They achieved high performance in terms of accuracy, sensitivity and specificity with age matched depressive subject predictions on both training and testing phases. Sara sardari et al. in [30] proposed deep learning based convolutional neural network based autoencoder method to detect depression. They showed the association of audio features and depressive behaviour with the help of most discriminative features. Authors used DAIC dataset for extracting hand crafted features then use deep learning based methods to detect healthy and depressive subjects. Their results suggested that audio features are helpful to improve the accuracy of the deep learning method with at least 7% of F-measure performance parameter.

Xiaolin mino et al. in [31] experimented with fusion of feature combining the higher-order spectral characteristics and standard speech features retrieved with classification weights. To validate the suggested features, traditional machine learning models such as support vector machine and k-nearest neighbour algorithms were used, along with convolutional neural network. The accuracies of speech-related feature extraction utilising the collaborative voice analysis repository, higher-order spectral analysis, and their fusion features were 63.15%, 68.42%, and 73.68%, respectively, using the support vector machine technique. Similarly, the equivalent accuracies using the k-nearest neighbour classification technique were 68.18%, 72.73%, and 77.27%, respectively. For the same characteristics, the convolutional neural network model resulted in accuracies of 70%, 77%, and 85%. These findings emphasise that fusion feature’s achieved higher accuracy, which can improve the precision of depression recognition when using both classical machine learning and deep learning models.

Balijeet kaur et al. in [32] investigated on wide range of spectral, temporal, and spectro-temporal characteristics which were extracted from speech bio signals of both healthy and depressed subjects. Their work presented a two-stage technique that uses the Quantum-based Whale Optimisation Algorithm (QWOA) to determine the most significant and non-redundant speech variables for effective depression identification. The suggested strategy is tested against three proven univariate filtering approaches and four well-known evolutionary algorithms for feature selection on the DAIC-WOZ dataset. Their findings showed that the suggested model outperforms all univariate filter approaches and evolutionary algorithms, displaying greater performance while requiring lesser computer complexity than existing wrapper-based evolutionary methods.

Despite significant advancements in audio-based depression detection, there are few notable research gaps that warrant attention. Firstly, the majority of existing studies discussed above focus on using traditional machine learning techniques for feature extraction and classification. Secondly, most research works have been conducted on relatively small and homogenous datasets, limiting the robustness of the proposed methods and third, the ethical implications of using audio-based depression detection methods, includes privacy concerns and potential biases in the data, need to be addressed to ensure responsible and ethical deployment of such technologies in real-world settings. The current work aims at addressing these research gaps that will contribute to the development of more efficient, reliable, and ethically sound audio-based depression detection systems.

III. METHODOLOGY

Subjects are carefully chosen to represent the whole population and acoustic speech dataset is created using speech elicitation technique. The overview of the proposed work is represented in Fig. 1. The details are discussed in the following subsections.
A. Dataset Construction

Students for participating in the dataset of the current work are recruited at Government polytechnic, Hyderabad, India. Out of the 225 students (here onwards called clients) who have accepted the participation in the experiment, 219 were actually turned up to read and understand English language.

Before initiating the dataset construction, all the clients (students) were explained the purpose and procedure of the study. Clients gave their written consents before they involve in dataset construction process. Approvals for conducting the study from the Institute were taken to conduct our study.

A recording room with good lighting condition is setup with a laptop on the table and a chair is placed one metre away from the laptop. Students were given prior information on their time slot to attend the study. Each student has been asked to give his voice samples in two phases. In the first phase, student is asked to read a short tale of “The north wind and south wind” which is shown on the monitor. This fable is taken because it has all the phonetically balanced sounds to stimulate the acoustic characteristics of the speech. In the second phase, they were asked to give an open form of speech of the topic of their interest in the choices that are displayed in the list. List has topics related to their goal in the life, dish they like the most, role model, worst situation in their life, etc. A total of approximately 11 hours of voice samples, from 219 participants have been recorded.

B. Data Preprocessing

The recorded voice samples are used in this pre-processing step. Here, the voices where other than the participant is found, they are cropped manually. The resultant voice samples are denoised using SOX software. It is a cross-platform audio editing tool. SOX has a command-line interface, and is built using standard C language. The denoised voice samples are obtained at the end of this stage.

C. Feature Extraction

The denoised voice samples are used in this feature extraction step. Here, the denoised voice records are given as input for openSMILE toolkit for feature extraction. This tool is a state-of-the-art feature extraction toolkit. The eGeMAPS features are extracted in the current study. The next sections go into more detail about eGeMAPS and the openSMILE toolkit.

1) Open SMILE: The Open-Source Media Interpretation by Large feature-space Extraction (OpenSMILE) [33] is an open-source toolkit for audio and speech processing developed by the Audio Communication Group at the Technical University of Munich. It is designed for feature extraction from speech signals and can be used for a wide range of tasks, such as speaker identification, speech recognition, and emotion recognition in speech. It provides a rich set of audio features that can be used for analysis, including low-level signal features such as spectral and cepstral coefficients, as well as higher-level features such as prosodic features, which capture characteristics such as pitch and speaking rate.

Open SMILE is written in C++, and includes a command-line interface for feature extraction, as well as a C++ API for integration with other software. It can run on Windows, Linux, and macOS operating systems. OpenSMILE is licensed under the GNU Lesser General Public License, which allows for free use and modification of the software. Overall, openSMILE is a powerful and flexible toolkit for audio and speech processing that has gained popularity in both academic and industrial settings. Its open-source nature and broad feature set make it a valuable resource for researchers and developers working in the field of speech processing. One of the key features of openSMILE is its modularity. The toolkit consists of a core engine that provides basic audio processing and feature extraction functions, as well as a set of modules that can be added to extend the functionality of the system. These modules include support for various audio formats, filtering and pre-processing, segmentation, and classification.

2) eGeMAPS Features: eGeMAPS [34] (extended Geneva Minimalistic Acoustic Parameter Set) is a widely-used feature set for speech analysis. It includes a comprehensive set of 88 high-level and low-level acoustic features that capture a wide range of properties of speech signals, such as voice quality, prosody, and spectral characteristics. The eGeMAPS feature set was designed to be a compact and efficient feature set for speech analysis, while still providing a rich set of information about speech signals. It includes low-level features such as spectral and cepstral coefficients, as well as higher-level features such as prosodic features, which capture characteristics such as pitch, speaking rate, and voice quality.

The eGeMAPS features contain 25 Low Level Descriptors (LLD) of three parameter groups: Frequency related parameters, Energy related parameters and Spectral related parameters. Frequency related parameters have 8 LLDS. They are pitch, jitter and formants frequency 1-3 and their bandwidths of voice samples. Energy related parameters include 3 LLDS, namely loudness, shimmer, and Harmonics-to-noise ratio (HNR) components of voice signals. Spectral parameter consists of 14 LLDS. They are alpha ratio, Hammarberg Index, Spectral Slope of 0-500 Hz and 500-1500 Hz, Formant 1, 2, and 3 relative energy, Harmonic difference between first harmonic (H1) and second harmonic (H2) energy, Harmonic difference H1 and third formant range (A3), MFCC 1-4 and spectral flux.

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1https://sox.sourceforge.net/
Now for these 25 LLDs arithmetic mean and standard deviation normalized by arithmetic mean functional are computed to form 50 components. Then for pitch and loudness, 8 different functional values are calculated. The functionals are as follows: $20^{th}$, $50^{th}$ and $80^{th}$ percentile and their ranges, mean and standard deviation of falling components signals. A total of 58 components are computed. Now arithmetic mean of first four components of spectral parameters is computed. 66 components are computed in total.

Six temporal features are computed. They are: rate of loudness peaks, mean and standard deviation of voiced samples, mean and standard deviation of devoiced samples, syllable rate. Resultant is 72 features. Now arithmetic mean of spectral flux in non-voiced parts, mean and standard deviation of spectral flux and Mel-Frequency Cepstral Coefficients (MFCC) 1-4, the LLDs of spectral flux and MFCC 1-4 values in total 16 components. Total 88 feature vectors of eGeMAPS are formed.

3) Size of features and ground truth: The eGeMAPS features are extracted for the voice records to form acoustic characteristics of speech biomarkers. The purposes of extracting this feature vector are listed as follows: i) eGeMAPS features are published as baseline features since 2016 in AVEC depression detection challenge [35]. ii) lot of works carried out in depression detection using speech showed reliability of eGeMAPS features [28], [34]. iii) eGeMAPS features are used in similar works of healthcare and also in emotion recognition in the literature of speech based detection methods [36].

For each participant 88 features are formed. By the end of this step, 219 X 88 size matrix is formed. Our dataset has 89th column as ground label. Ground truth is given by a Psychologist. Hence our dataset is clinically validated by the Psychologist forming more reliable and coherent dataset. Thus dataset contains 219 X 89 size matrix.

A Psychologist was recruited to label the subjects. Post audio collection, Psychologist gave 17-item questionnaire called as Hamilton Rating Scale for Depression (HRSMD) [37]. Subjects were asked to fill the HRSD form. Then Psychologist validated the label as not depressed and mild depressed. In our experiment we could not find any cases of moderate, severe cases of depression. Out of 219 subjects 137 are labelled as non-depressed and 82 as depressed.

Our justification of choice of size of the dataset is based on a comprehensive related work section presented in the current study on depression detection using machine learning and audio-based approaches. As per the knowledge of the authors, cited studies that utilised sample sizes range from a few tens to hundred participants, thereby establishing a strength on context for our sample size choice. The selected sample size of 219 clients exceeds these cited studies ensuring that our study is adequately powered to detect meaningful effects and relationships between audio-based features and depression. While we acknowledge that larger sample sizes could potentially improve the generalizability of our findings, we believe that the current dataset size is appropriate for the initial validation of our proposed framework.

D. Feature Normalization

Feature normalization is a common preprocessing step in machine learning that involves scaling the input features to have a similar scale and distribution. The purpose of normalization is to improve the performance and stability of machine learning models by reducing the influence of features with large values and preventing some features from dominating others [38].

In the current study, each group of parameters has different values and ranges. Hence to normalize them into a standard set of values we used Min-max scaling mechanism. All the experiments were conducted using these normalized values in this study. In this technique, values are scaled to a fixed range. We used 0 to 1 range for normalizing the values.

E. Ensemble

The ensemble method in machine learning combines several classifiers predictive results into an optimal result [39]. The current work used stacking technique of the ensemble method to derive the optimal predictions. In stacking, multiple heterogeneous classifiers are used to form an intermediate result. These intermediate results are provided as input for the meta classifier to predict the final outcome. Stacking aims to improve the meta classifier’s performance.

The mathematical equation for a stacking ensemble can be represented as follows:

Consider $D = \{ (x_1, y_1), (x_2, y_2), ..., (x_n, y_n) \}$ be the training dataset, where $x_i$ is the $i$-th instance and $y_i$ is the corresponding target value.

Let $M = \{ M_1, M_2, ..., M_k \}$ be a set of base models, where $M_i$ is a machine learning model that maps an instance $x$ to a predicted target value $M_i(x)$.

Let $f$ be the meta-model that takes the predictions of the base models as inputs and outputs a final prediction, i.e., $f(M_1(x), M_2(x), ..., M_k(x))$.

The stacking ensemble can be trained in two stages:

1) **Base model training:** Each base model $M_i$ is trained on the training dataset $D$ to predict the target values.

2) **Meta-model training:** The predictions of the base models on the training dataset $D$ are used to train the meta-model $f$. Specifically, a new dataset $D'$ is created by replacing the target values in $D$ with the predictions of the base models, i.e., $D' = \{ (M_1(x_1), M_2(x_1), ..., M_k(x_1), y_1), (M_1(x_2), M_2(x_2), ..., M_k(x_2), y_2), ..., (M_1(x_n), M_2(x_n), ..., M_k(x_n), y_n) \}$. The meta-model $f$ is then trained on $D'$ to predict the final target values.

The mathematical equation for the stacking ensemble can then be written as in (1):

$$ f(M_1(x), M_2(x), ..., M_k(x)) = g(w_1 * M_1(x) + w_2 * M_2(x) + \cdots + w_k * M_k(x)) + b $$

where $w_1, w_2, ..., w_k$ are the weights assigned to the predictions of the base models, and $b$ is the bias term. These
weights and bias term are learned during the meta-model training stage.

The block diagram of the stacking ensemble method is given in the following Fig. 2 and its explanation is as follows:

1) **Input Data:** It consists of training and test data, with the input data (training data) split into training and validation sets. The base models are trained using the training data. The validation data is used to assess the performance of the base models and choose the best-performing models. Once the meta-model is trained, it is used to make predictions on the test data. The test data is a distinct dataset from the training and validation sets that the model has never seen before.

2) **Base models:** The base models are the individual models that will be trained on the input data. These models can be of different types, such as Decision trees, SVMs, or Neural networks. Each model produces its own output or prediction for the given input.

3) **Predictions:** The trained base models produce outputs or predictions for each input in the validation set.

4) **Meta-classifier:** The meta-model is a model that combines the outputs of the base models to make a final prediction. It takes the outputs of the base models as input features and learns to combine them in an optimal way. The meta-model can be a simple linear model, such as Logistic regression, or a more complex model, such as a Neural network.

5) **Final prediction:** The meta-model produces a final prediction for each input in the test data. This prediction is based on the outputs of the base models and the learned combination of these outputs.

Algorithm for the proposed ensemble method:

The proposed ensemble method algorithm is a 7-step process for classifying acoustic data using a combination of base classifiers and a meta-classifier. Here is a brief explanation of each step as follows:

**Step 1:** Obtain the acoustic dataset which is of the size 219 X 89 matrix.

**Step 2:** Perform feature normalization using Min-max scaling technique.

**Step 3:** Divide the normalized feature vector to training set(80%) and test set(20%) without any overlap.

**Step 4:** Implement stacking with the base classifiers for training set obtained from step 3.

**Step 5:** Construct a meta classifier with the input from the outcomes of the base classifiers as features.

**Step 6:** Now using the test set, form intermediate outputs with the same classifiers. These intermediate outcomes are provided as input for the trained meta classifier to form final prediction.

**Step 7:** These final predictions are compared with the ground truths to check the performance of the ensemble method.

**F. Performance Metrics**

In the current work, the ensemble method’s performance is evaluated utilising a variety of performance metrics. The performance metrics accuracy, precision, recall and F1-score are measured to show the effectiveness of the model.

The model’s overall performance is evaluated using a confusion matrix. It contains the actual values and test predictions. The following Table I gives the details of the confusion matrix.

Confusion matrix contains four terms: True Positive(TP), True Negative(TN), False Positive(FP) and False Negative(FN). TP and TN are correct predictions made by the model. TP and TN are correctly classified as compared with the actual ground truths. FN and FP are incorrect predictions made by the model. TN and TP are correct predictions where as FN and FP are the incorrect predictions. For any model, TP and TN must be high and FP and FN must be as low as possible. Then performance of the model is higher. The following Table II gives more details of the performance metrics used in the current study.

<table>
<thead>
<tr>
<th><strong>TABLE I. CONFUSION MATRIX</strong></th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Predicted Values</strong></td>
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<tr>
<td></td>
</tr>
<tr>
<td>Positive</td>
</tr>
<tr>
<td>Negative</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th><strong>TABLE II. PERFORMANCE METRICS UTILISED IN THIS WORK</strong></th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Performance parameter</strong></td>
</tr>
<tr>
<td>----------------------------</td>
</tr>
<tr>
<td><strong>Accuracy</strong></td>
</tr>
<tr>
<td><strong>Precision</strong></td>
</tr>
<tr>
<td><strong>Recall</strong></td>
</tr>
<tr>
<td><strong>F1-score</strong></td>
</tr>
</tbody>
</table>
These metrics are essential for assessing the performance of the classification model in the context of depressed and non-depressed classification. Accuracy is a measure of the proportion of cases overall that were correctly identified as being either depressed or not. A high accuracy rating indicates that the model is adept at correctly classifying both depressed and non-depressed people, making it a useful indicator of the classification model’s overall effectiveness. On the other hand, when dealing with depression detection, precision becomes especially important. It gauges how well positive predictions pan out, which in this instance translates to how well depressed people are identified. A high precision means the model has a low false positive rate and successfully recognises depressed people without mistakenly classifying healthy people as depressed.

Recall is extremely important since it measures how well the model can identify depressed people among all real-world depressed instances. A high recall means that the model successfully identified the majority of the depressed individuals in the dataset and has a low false negative rate. The F1-score offers a useful compromise between recall and precision. It provides a thorough assessment of the model's effectiveness in the classification of depression by taking into account both metrics. As a result, a classification model with high accuracy, precision, recall, and F1-score performs well in accurately identifying people with depression while minimising misclassifications, offering crucial help for those in need of mental health care and therapies.

**IV. Results**

This section discusses the results obtained in the current study. Jupiter python notebook is used for all the experiments conducted in the current study. The results are discussed in four forms: First, the effectiveness of the proposed model is evaluated in terms of various performance metrics. In order to show the performance gain of the proposed model, the performance of our model is then compared to the baseline models on our dataset. Third, the proposed model is applied to the benchmarking dataset to show the reliability of the study. In the end, the proposed model is compared with state-of-the-art baseline techniques on benchmarking dataset.

**A. Performance Evaluation of the Proposed Model**

In stacking protocol, first, the choice of the base classifier is made among all the available classifiers significantly important to acquire the performance improvement. In the present study, K-Nearest Neighbours (KNN), Naïve Bayes (NB), Support Vector Machine (SVM), and Decision Trees (DT) classifiers are selected as base classifiers. This choice is because of the following reasons: 1) they are easy to implement; 2) achieves higher accuracy when the size of the dataset is relatively small; and 3) conducted rigorous experiments and both performed relatively better on trial and error basis of all the classifiers. Thus when both these classifiers are combined, results in the performance enhancement.

Second, meta classifier uses the results of the diverse predictions obtained with the base classifiers. This will be helpful to avoid misclassifications of the model because feature size is reduced when compared with the original dataset size. In our study, the feature size is brought down to 2 from 88. This reduction helps in meta classifier to reduce misclassifications and results in performance optimisation. Logistic Regression (LR) classifier is trained and tested as the meta classifier in our current work. Table III presents summary of the parameters used in the current study.

**TABLE III. Summary of the Parameters of the Proposed Ensemble Method**

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Tuned Parameter</th>
</tr>
</thead>
<tbody>
<tr>
<td>Base Classifier-1</td>
<td>K-NN</td>
</tr>
<tr>
<td>Base Classifier-2</td>
<td>NB</td>
</tr>
<tr>
<td>Base Classifier-3</td>
<td>SVM</td>
</tr>
<tr>
<td>Base Classifier-4</td>
<td>DT</td>
</tr>
<tr>
<td>Meta Classifier</td>
<td>LR</td>
</tr>
<tr>
<td>Hyperparameters</td>
<td>Default values</td>
</tr>
</tbody>
</table>

Hence the proposed work integrates the predictive capabilities of five classifiers namely K-NN, Naïve Bayes, SVM, Decision Trees and Logistic Regression classifier, to obtain optimal results. Table IV shows the performance metrics obtained on our dataset using a stacking ensemble model. The four metrics evaluated are accuracy, precision, recall, and F1-score. This implies that the proposed framework can be utilized for clinical applications. These findings suggest that the proposed framework can be applied to the real-world scenarios. The hypothesis of the study is that the proposed framework holds as a valuable tool for aiding healthcare professionals in early diagnosis and intervention for depression. Comparable results obtained using accuracy, precision, recall, and F1-score values by our approach demonstrate its capability to precisely detect the depression in individuals.

**TABLE IV. Performance Metrics Values of Proposed Stacking Method on Our Dataset**

<table>
<thead>
<tr>
<th>Performance metric</th>
<th>Value obtained on our dataset (in %)</th>
</tr>
</thead>
<tbody>
<tr>
<td>Accuracy</td>
<td>79.1</td>
</tr>
<tr>
<td>Precision</td>
<td>83.3</td>
</tr>
<tr>
<td>Recall</td>
<td>76.9</td>
</tr>
<tr>
<td>F1-score</td>
<td>80.0</td>
</tr>
</tbody>
</table>

![Depression detection performance of proposed stacking ensemble model.](image)

As presented in Fig. 3, our method of using diverse classifiers achieved accuracy of 79.1%, precision of 83.3%, recall of 76.9% and F1-score of 80.0%. These scores show that the performance of the proposed method is reliable. These results suggest that the model performed well on the dataset,
with high precision and F1-score values. However, it’s important to note that the evaluation is based on a single dataset and may not generalize well to other datasets. Therefore, further testing and validation are necessary to confirm the effectiveness and robustness of the model. The following investigation is carried out to prove the reliability of the study.

B. Performance Comparisons of the Proposed Model with the Baseline Models

To show the performance gain of the proposed method we compared the proposed model’s performance with the baseline methods. The baseline methods of the current work are the base classifiers used in the proposed model to build the meta-classifier. These methods are chosen because following this method of comparison gives deep understanding of the amount of performance gained with the stacking method over the base classifier utilisation. The feature set used in the current work is also used for training (80%) and testing (20%) base classifiers namely K-NN, Naïve Bayes, SVM, and Decision Trees classifiers. Table V contains details about comparison of the proposed stacking approach with the baseline methods.

Table V shows that the proposed method has outperformed the baseline methods in terms of all performance metrics. It contains a comparison of the performance metrics achieved by our proposed approach using ensemble and on using base classifiers such as K-NN, Naïve Bayes, SVM, and Decision Trees on our dataset. The reason behind choosing these classifiers is its vast use in the literature of the study (presented in the related work section). Our proposed approach outperformed the baseline methods in terms of Accuracy (79.1% vs. KNN: 62.5%, NB: 62.5%, SVM: 75.2%, DT: 76.8%) and Precision (83.3% vs. KNN: 61.5%, NB: 58.3%, SVM: 82.1%, DT: 80.7%), indicating its ability to correctly classify instances and minimize false positive predictions using our model. Although Recall values were similar between two baseline methods (our approach: 76.9% vs. KNN: 66.6%, NB: 63.4%, SVM: 77.2%, DT: 76.8%), our proposed approach achieved a higher F1-score (80.0%) compared to the baseline methods (KNN: 64.0%, NB: 60.8%, SVM: 76.3%, DT: 78.2%), signifying a better balance between Precision and Recall. These results suggest that the effectiveness of our proposed approach in accurately detecting depression and highlight its potential as a robust and reliable tool for classification of depression.

C. Performance Comparison of the Proposed Model using Benchmarking Dataset

Now, the generality of the proposed model is tested. Our approach not only worked for our dataset but also can be applied to the other datasets. We applied our stacking method on the Distress Analysis Interview Corpus-Wizard of Oz (DAIC-WOZ) database which is publicly available benchmarking dataset in depression detection. This corpus contains audio and visual recordings of interviews between a real-life interviewee and a computer-generated interviewer who either acts as a supportive listener or a diagnostician. The participants were instructed to discuss few topics, including personal experiences and emotional states, allowing for the elicitation of genuine emotional expressions. Researchers who are interested in developing and validating depression detection models use the DAIC-WOZ database as a benchmark dataset. Additionally, the public dataset allows for comparisons and cross-validation of different methods, facilitating the advancement of automated depression detection technologies for real-world applications.

To apply our proposed stacking method, the audio clips of the subjects are used to generate the eGeMAPS feature vectors. Then after standardising the dataset stacking framework is applied.

<table>
<thead>
<tr>
<th>Performance metric</th>
<th>Value obtained on our dataset (in %)</th>
</tr>
</thead>
<tbody>
<tr>
<td>Accuracy</td>
<td>78.0</td>
</tr>
<tr>
<td>Precision</td>
<td>71.7</td>
</tr>
<tr>
<td>Recall</td>
<td>90.6</td>
</tr>
<tr>
<td>F1-score</td>
<td>79.4</td>
</tr>
</tbody>
</table>

Table VI shows the details of the performance of the proposed stacking method on the DAIC-WOZ dataset. Our method achieved comparable performance when applied on the benchmarking dataset. Among all the performance metrics recall gave a performance of 90.6% using the proposed method. From the Table VI it can be inferred that the proposed method can be applied to any dataset with similar audio cues.

D. Performance Comparison with State-of-the-Art Baseline Methods using Benchmarking Dataset

Table VII presents a comparison of performance metrics obtained by our proposed ensemble method on the public accessible benchmarking DAIC-WOZ dataset. We compared our results with two state-of-the-art methods: (1) Thati et al. in [27] used multimodal fusion approach by late fusion using secondary SVM, and (2) Janardhan et al. in [28] used Fisher score-based feature selection method for classification of depressed or non-depressed individuals.

As presented in Fig. 4, our proposed ensemble method outperformed both existing state-of-the-art methods across all performance metrics. Specifically, our method acquired an accuracy of 78.0%, surpassing the late fusion using secondary SVM (74.0%) and Fisher score-based feature selection mechanism (74.0%). The precision of our method was 71.7%, significantly higher than the late fusion (52.0%) and also Fisher
score-based (49.0%) approaches. Also, our ensemble method demonstrated a remarkable recall of 90.6%, outperforming the late fusion (58.0%) and Fisher score-based (50.0%) methods in correctly identifying the depressed subjects. Lastly, F1-score of our proposed method was 79.4%, exceeding both the late fusion (73.0%) and Fisher score-based (74.0%) methods. These results clearly demonstrate the superiority of our ensemble method for depression classification on the DAIC-Woz dataset, showcasing its effectiveness in achieving higher accuracy, precision, recall, and F1-score compared to the state-of-the-art methods mentioned.

TABLE VII. PERFORMANCE METRICS COMPARISON WITH THE STATE-OF-THE-ART BASELINE METHODS ON DAIC-WOZ DATASET

<table>
<thead>
<tr>
<th>Performance metric</th>
<th>Our Proposed stacking method (in %)</th>
<th>Late fusion using secondary SVM</th>
<th>Fisher score-based feature selection</th>
</tr>
</thead>
<tbody>
<tr>
<td>Accuracy</td>
<td>78.0</td>
<td>74.0</td>
<td>74.0</td>
</tr>
<tr>
<td>Precision</td>
<td>71.7</td>
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</tr>
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<td>90.6</td>
<td>58.0</td>
<td>50.0</td>
</tr>
<tr>
<td>F1-score</td>
<td>79.4</td>
<td>73.0</td>
<td>74.0</td>
</tr>
</tbody>
</table>

Fig. 4. Performance of proposed stacking method compared with existing Models on a DAIC-WOZ dataset.

V. CONCLUSION AND FUTURE WORK

Mental illness, manifestation of depression and anxiety, is a serious impediment to global well-being. Depression is an illness that can cause financial, social, family, emotional losses for individuals and leads to suicide causing sorrow to their families. A machine learning-based approach using audio can use patterns which are different in healthy and depressed subjects. These methods help healthcare professionals detect depression. The literature on depression detection through audio-based used only single classifier, but not the multiple classifiers. The present work combines the predictive patterns of different classifiers to detect depression using a stacked approach. Audio clips are recorded as subjects read predefined passages and free-form speech. Then eGeMAPS features are extracted using the openSMILE toolkit. The normalized feature vectors are trained with multiple classifiers. Then meta-classifier is also trained to form final predictions. On this audio dataset, our proposed method, which combines a variety of classifiers, produced impressive results with an accuracy of 79.1%, precision of 83.3%, recall of 76.9%, and an F1-score of 80.0%. Our method demonstrated a remarkable performance improvement in terms of accuracy when compared to single classifier technique. Identical performance improvements are seen across various evaluation parameters. Our method's validity on the DAIC-WOZ dataset further confirms its dependability and potency to generalize on other clinical applications. Notably, our method achieved a recall of 90.0%, demonstrating its potential as a global approach for depression detection and demonstrating its ability to generalize and be applicable to other datasets with similar audio cues.

Our future works aim to investigate the implementation of regression-based methods to identify not only the presence of depression but also the severity or levels of depression in individuals. The importance of expanding the dataset size by increasing the number of samples is recognised. Also research on linguistic characteristics and language pragmatics of speech, seeking to extract and utilize the valuable information beyond acoustic features for more accurate depression detection.

ACKNOWLEDGMENT

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Optimizing Port Operations: Synchronization, Collision Avoidance, and Efficient Loading and Unloading Processes

Sakhi Fatima ezzahra, Bellat Abdelouahad, Mansouri Khalifa, Qbadou Mohammed
Signals, Distributed Systems and Artificial Intelligence (SSDIA), ENSET Mohammedia, Hassan II University, Casablanca, Morocco

Abstract—This study focuses on optimizing the loading and unloading processes in a port environment by employing synchronization techniques and collision avoidance mechanisms. The objective function of this research aims to minimize the time required for these tasks while ensuring efficient coordination and safety. The obtained results are compared with previous studies, demonstrating significant improvements in overall performance. The synchronized handling systems, including gantries and cranes, along with speed control measures, facilitate streamlined operations, reduced delays, and enhanced productivity. By integrating these strategies, the port achieves better results in terms of task completion time compared to previous methodologies, thereby validating the effectiveness of the proposed approach.

Keywords—Optimizing; synchronization; collision; efficient; time

I. INTRODUCTION

Productive and non-productive times play a crucial role when it comes to optimizing the loading and unloading of containers from ships in a port. Many of the common problems associated with optimizing these aspects which are firstly, Inefficient planning which leads to poor coordination and planning of ship arrivals, berth allocation and crane operations can lead to significant delays and inefficiencies. Vessels may have to wait for a berth, resulting in downtime and reduced productivity. Additionally, if container handling equipment, such as cranes, are not allocated efficiently, it can lead to downtime and reduced throughput.

In addition, insufficient or obsolete equipment can slow down the loading and unloading process [1]. Equipment failures and maintenance issues can cause unexpected delays and increase non-productive time [2]. Regular maintenance and prompt repairs are essential to minimize downtime and ensure optimal equipment performance. Ineffective coordination between ship's crew, port operators and logistics personnel can lead to inefficiencies. Lack of clear communication and collaboration can lead to delays, errors in documentation, and increased downtime [3]. Efficient cargo handling operations require effective communication channels and streamlined processes. By relying on manual processes, such as paper documentation, delays and errors can occur. Manual data entry, inspections, and customs clearance procedures may consume valuable time. Implementing digital solutions, such as electronic data interchange (EDI) systems and automated workflows, can significantly improve efficiency and reduce non-productive time. Customs regulations are essential but can also slow down operations [4]. Rigorous security checks, customs inspections and document verification processes can cause delays and increase non-productive time. Port authorities and operators must strike a balance between security measures and operational efficiency.

Various mathematical models and optimization techniques have been developed to solve planning and resource allocation problems in container terminals. These models aim to minimize vessel waiting times, dock occupancy and equipment, among the solutions cited in the literature first: Automation and robotics: Automation technologies, such as robotic systems and automated guided vehicles (AGVs) [5], are increasingly being used to improve the efficiency of container handling. Simulation and optimization studies: Simulation models and optimization algorithms are often used to evaluate different strategies and scenarios in container terminal operations. These studies analyze factors such as layout design, equipment utilization and resource allocation. Digitalization and data-driven approaches: The use of digital technologies, data analysis and real-time information has a significant impact on time optimization in port operations [6]. Lean Process Improvement and Methodologies: Lean principles and process improvement methodologies, such as Six Sigma, are relevant to improving operational efficiency and reducing non-productive time.

Previous studies [7] have answered the question of optimizing productive and non-productive times when loading and unloading containers in ports, it is crucial to implement the following strategies: Improve coordination and communication: Improve coordination between port authorities, shipping companies, terminal operators and logistics providers. Real-time information sharing and advanced communication systems can enable better planning and resource allocation. Optimize berth and equipment allocation: Develop effective berth allocation systems that take into account vessel size, cargo type and expected volume. Implement intelligent asset management systems to allocate cranes and material handling equipment based on real-time demand [8]. Improve infrastructure and layout planning: Invest in port infrastructure expansion and optimize layout plans to accommodate growing container volumes. Designate dedicated storage areas for efficient container handling and reduce
Congestion and limited space can induce limited space in ports and can lead to congestion, making it difficult to manage containers efficiently. Lack of designated storage areas and inefficient layout planning can lead to delays in loading and unloading operations [9]. This problem becomes more pronounced when several ships arrive simultaneously or when there is a high volume of containers to handle.

II. INTERFERENCE PROBLEMS AND UNPRODUCTIVE MOVEMENTS

A. Interference Problem between Gates

The interference problem is well related to the problem of planning multiple yard gates in a seaport since these gates share the traffic lane in the yard and the movement of one can block the movement of the other. In this context, Ng (2005) considered the interference between yard gates when planning the movements of this type of equipment. This phenomenon is defined as a physical blockage that must be considered when planning yard gates [10]. According to Ng, the courtyard inter-portal interference problem is a complicated problem and it requires the development of an efficient method for solving the planning problem.

Interference between yard gantries can affect the planning of quay gantries. In this context, [11-12] sheds light on the relationship between quay gantry planning and the timing of yard gantry operations. Their work is aimed at minimizing the turnaround time of ships and is aimed at scheduling quay gantries taking into account the progression of handling operations at the yard level. Indeed, any delay in the operations of the yard gantries affects the operations of the quay gantries.

Zhong, M.S. et al [13] proposed a model for planning yard gates while taking into account some constraints related to interference between yard gates, fixed separation distances between these gates and simultaneous container storage/retrieval operations. The model aims to minimize a linear combination of early collections, storage and late collections. The problem of interference between courtyard gates is developed within the constraints of the mathematical formulation.

Chen, J et al [14] treated the interference problem implicitly. The work addresses the problem of scheduling multiple yard gantries during container loading operations. The objective of this work is to improve the processing efficiency of containers intended for export by the yard gates. The authors have taken into account the potential interferences that may occur. Indeed, they integrated this phenomenon into the gate planning problem. An interference hypothesis is developed in the mathematical model [15-16]. This hypothesis takes into consideration two types of interference which are “collision” and “crossover”. The first type of interference occurs when two-yard gates in the same block try to pick up containers stored in adjacent bays. According to Chen and Langevin, a minimum distance of five bays must be respected to avoid a potential collision between two courtyard gates. The second type is the crossing which occurs when the next container to be processed by a given yard gate is located on the other side of another yard gate. In order to avoid any possible interference, the authors developed several constraints in the mathematical formulation.

The work of [17-18] studied the operation of several yard gantries when loading/unloading containers. According to the authors, the yard gates must leave a safe distance to avoid accidents. Indeed, in the same block, several courtyard gates can work simultaneously; in this situation there may be a risk of collision between the yard gates and even between the trucks. In Fig. 1, yard gate 2 (YC2) wants to move to bay 01 after processing a container located in bay 07 but is blocked by yard gate 1 (YC1) which is processing a container located in bay 03. So yard gate 2 must wait until yard gate 1 completes its task.

B. Problem of Non-productive “Rehandle” Movements

In a previous study [19-20], an author proposed a methodology for estimating the anticipated quantity of non-productive moves required to pick up a container, as well as the overall count of non-productive container moves within a bay with an initial stacking arrangement [21]. The author highlights the significance of the rack's height and width in determining the average rehandling count and the design of the storage configuration. To estimate the expected number of non-productive moves for the next pickup, the author examined multiple container stack configurations and derived an approximate formula for estimating the total expected count of non-productive moves [22].

![Fig. 1. The collision caused by the courtyard gates.](image-url)
A study conducted in [23-24] focused on addressing the issue of non-productive crane movements during container loading and unloading activities at a seaport. The researchers aimed to ensure the stability of containers stored on a ship while minimizing the number of non-productive movements within container blocks. To tackle this problem, a multi-objective integer program was formulated. The authors utilized a weighting method to obtain non-dominated solutions, drawing inspiration from the works of [25-26]. They incorporated the consideration of non-productive moves by estimating their count and introduced the concept of probability to study this estimation. However, accurately calculating the number of non-productive moves relies on a predefined loading sequence, which introduces challenges due to the random recovery of containers [27-28]. To solve both the container loading problem on the ship and the rehandling problem, the researchers employed a genetic algorithm.

III. MATHEMATICAL MODELING OF CONTAINER MOVEMENTS

A. Model Assumptions

When developing a mathematical model, a number of assumptions must be taken into account. These will be integrated in one way or another in the model. Our assumptions are as follows:

- We limit ourselves to the loading operations of export containers (outbound).
- Location of containers is given.
- A courtyard is made up of several adjacent blocks.
- There are up to two court gates in a block.
- Court gate moves between blocks are possible, maximum one move.
- For each container, we know its destination on the quay (the quay gantry).
- We consider the simultaneous movements of the yard gantries and the trucks.
- All containers in a bay are destined for the same vessel.
- The possibility of interference from courtyard gates in the same block is taken into account.
- Containers are classified into groups with an order and priority in the handling of each group. The order is known in advance.
- Non-productive movements of containers (Rehandle) are taken into account.

B. Modeling and Constraint

We will consider that \( p_i \) is the processing time of container \( i \) through a yard gate which includes the time to pick up the container \( h_1 \) and the time to remove and return the containers so \( p_i \) can be expressed by the formula below:

\[
p_i = h_1 + \sum_{j \in C} \sum_{\substack{a_i \in a_j \\ r_i = r_j \\ e_i \leq e_j}} h_2 \times V_{ij}
\]

(1)

Where:

\[
\sum_{\substack{a_i \in a_j \\ r_i = r_j \\ e_i \leq e_j}} h_2 \times V_{ij}
\]

is the non-productive time of the yard gates (rehandle).

\( V_{ij} \): if the processing of container \( j \) begins after the processing of container \( i \) by a yard gantry has finished.

\( h_1 \): Container handling time.

\( h_2 \): Handling time of a container to pick it up and put it back in its place (rehandle); we assume that \( h_2 = 2 h_1 \).

\( P \): Set of pairs of containers with a precedence relationship. We are talking here about an order of priority of containers that must be respected during the handling process by the quay gantries.

The variable \( p_i \) can be divided into two parts \( p_1^1 \) and \( p_2^2 \) which are respectively the handling time of container \( i \) and its loading on the truck and the time to return the container(s) that have been removed to its (their) seat(s).

\[
p_i = p_1^1 + p_2^2
\]

(2)

\( p_1^1 \) and \( p_2^2 \) are expressed as a function of \( h_1 \) and \( V_{ij} \) by the following formula:

\[
p_1^1 = h_1 + \sum_{\substack{a_i \in a_j \\ r_i = r_j \\ e_i \leq e_j}} h_2 \times V_{ij}; i \in C
\]

(3)

\[
p_2^2 = \sum_{\substack{a_i \in a_j \\ r_i = r_j \\ e_i \leq e_j}} h_2 \times V_{ij}; i \in C
\]

(4)

\[
d_{ij}^{yc}, d_{ij}^{yt}, p_1^1, p_2^2 \geq 0
\]

Indeed, to guarantee that the decision variables are positive and binary a constraint is expressed by the equation (5) is necessary for our approach.

It is also reported that in our study that constraints have been imposed which guarantee that the equipment which are respectively the trucks do not leave the starting point only once and they ensure flow conservation for each material handling equipment.

Other constraints that must be included in our model is first of all the constraint which defines the handling time of the container \( i \) and its loading on the truck the constraint which defines the time of reinstallation of the containers already removed this last is expressed by the equation (4).

\[
d_{ij}^{yc} \geq d_{ij}^{yc} + (p_1^1 + p_2^2) + k_{ij} + T(X_{ijm} - 1);
\]

(6)

Where
The decision to employ genetic algorithms stems from the fact that these algorithms are regarded as:

- A flexible and configurable method.
- Efficient in their ability to overcome the pitfalls of local optima while exploring the design search space to converge towards the global optimum.

The flowchart in Fig. 2 shows the optimization steps adopted to optimize a wind farm. In the Python© software, we have entered the data and the models established in the previous part, the optimization process begins to search for an optimal solution. GA research techniques include the following main steps:

- In Step 1, we randomly generate specific paths while ensuring compliance with the constraints of the problem at hand.

Step 2 involves evaluating the layouts of the objective function being studied by utilizing the objective function itself.

In Step 3, individuals are selected to contribute to the population of the next generation. The selection probability in the current generation ensures the selection of high-quality individuals. The crossover operator is employed to modify two pairs of genes (positions) and generate further optimization. As a crossover function, gene modification at another position is accomplished through the random activation of mutation with a predetermined probability.

Step 4 involves producing a new course flow through the genetic algorithm (GA), which modifies the previous population. These steps are iterated until the maximum number of iterations is reached. The data parameters of our algorithm can be found in Table I.

### TABLE I. PARAMETERS OF GENETIC ALGORITHM

<table>
<thead>
<tr>
<th>GA Parameter</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Size of initial population</td>
<td>1000</td>
</tr>
<tr>
<td>Selection pressure</td>
<td>3</td>
</tr>
<tr>
<td>Crossover probability</td>
<td>0.75</td>
</tr>
<tr>
<td>Mutation probability</td>
<td>0.25</td>
</tr>
<tr>
<td>Iteration number</td>
<td>4000</td>
</tr>
</tbody>
</table>

### C. Case Study

We will apply our model to the Port of Tangier Med which is one of the largest ports in Africa and the Mediterranean, strategically located on the Strait of Gibraltar. It has modern infrastructure, including several terminals for containers, passengers and general cargo. With a large container and passenger handling capacity, the port serves as a key transshipment hub, facilitating trade between Europe, Africa and Asia. It has had a significant economic impact, attracted investment and generated employment opportunities. Continuous expansions and developments have been undertaken to meet the increasing demands of international trade. Overall, the Port of Tangier Med plays a crucial role in global trade and connectivity in the region.
In our case, we propose three distinct mathematical models. The initial model focuses mainly on the optimization of truck routes, with the objective of minimizing the distance traveled by each vehicle. The second model involves assigning tasks to trucks, which results in a schedule of container task assignments based on the input data comprising the set of container tasks. Finally, the third model is designed to avoid collisions between trucks and gantries, ensuring a safe operating environment.

Our approach offers a hybrid solution by integrating simulations and algorithms to address modeling challenges. Essentially, truck route design can be simplified as a "shortest path problem". To solve this problem, we use the widely recognized method of Dijkstra, which has a long reputation for finding the shortest path. Model 1 in Fig. 2 uses Dijkstra's method. We can solve the model efficiently and determine the optimal route planning strategy for trucks (Fig. 4).

As illustrated in Table II, the truck scheduling model started with 1000 container tasks created at random. When the evolutionary algorithm was employed to solve the task assignment to trucks, the original population was 200, as shown in Fig. 3. If the crossover coefficient is set too low, the performance of our method suffers. If the mutation probability is sufficiently high, the genetic algorithm will reach the local optimum. As a result, the mutation probability and crossover probability were both set to 0.75.

The model shown in Fig. 5 and Fig. 6 focuses on optimizing truck task assignment, determining their routes, and managing their speeds to avoid collisions. It involves an intelligent system that uses advanced algorithms to solve this complex problem.

First, the model must take into account the different tasks to be performed by trucks, such as delivering goods to specific destinations, picking up materials, or performing specific services. These tasks are usually accompanied by specific constraints, such as delivery times, priorities or required resources.

The model must analyze the constraints and available resources, as well as the characteristics of the trucks, such as their load capacity, maximum speed, fuel consumption and mechanical constraints. Based on this information, the model can determine the best assignment of tasks to trucks, optimizing criteria such as total distance traveled, delivery time or operational costs.

Once the task assignment is determined, the model must plan the routes for each truck to avoid collisions. It must take into account various factors, such as distances between

![Fig. 2. Road planning trucks.](image)

![Fig. 3. Genetic algorithm's convergence diagram.](image)

![Fig. 4. Allocation of truck assignments.](image)

**TABLE II. SCHEDULING OF CONTAINER TASKS**

<table>
<thead>
<tr>
<th>Task Number</th>
<th>Starting position (QC)</th>
<th>Target position (yard)</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>QC1</td>
<td>Y1</td>
</tr>
<tr>
<td>2</td>
<td>QC2</td>
<td>Y3</td>
</tr>
<tr>
<td>3</td>
<td>QC3</td>
<td>Y4</td>
</tr>
<tr>
<td>999</td>
<td>QC1</td>
<td>Y5</td>
</tr>
<tr>
<td>1000</td>
<td>QC3</td>
<td>Y6</td>
</tr>
</tbody>
</table>

![OPTIMIZATION PROCESS](image)
destinations, traffic conditions, traffic restrictions (such as toll roads or no-truck zones) and specific preferences, such as shorter routes or faster.

To avoid collisions, the model can use path planning techniques that take into account the movements of other vehicles on the road. This can be achieved through the use of sensors, global positioning systems (GPS), digital maps and obstacle detection algorithms. The model can also use communication techniques between the trucks to exchange information about their position and intention, which helps coordinate their movements and avoid potential conflicts.

Finally, the model can control truck speeds based on various factors, such as road conditions, speed limits, safety constraints and operator preferences. This can be achieved by using automatic cruise controls or by providing speed recommendations to drivers.

Overall, the model shown in Fig. 5 is designed to optimize truck task assignment, determine their routes, and control their speeds to minimize collisions, maximize operation efficiency, and ensure safety on the road. It is an approach based on artificial intelligence and optimization that requires precise data on tasks, trucks, constraints and road conditions to make informed and effective decisions.

V. RESULTS AND DISCUSSION

The truck task assignment model produced assignment outcomes and path outcomes for trucks, which were then incorporated into our model of our collision avoidance gait. In addition, the speed control strategy has been applied. Collision avoidance strategies were evaluated for different numbers of container tasks, and the results are shown in Fig. 7 and 8.

Fig. 6 shows that regardless of the number of trucks, trucks operating under the speed control strategy exhibited significantly shorter operating times compared to those under conventional scheduling. This result stems from the ability of speed-controlled trucks to effectively avoid conflicts. In contrast, route planning under conventional planning often encounters conflicts, leading to truck stalls and subsequent delays in completion time. These results confirm that the speed control strategy can effectively mitigate conflicts.

Fig. 7 provides a more detailed analysis that further solidifies the benefits derived from employing the speed control strategy. One notable advantage is the substantial reduction in the total wait time compared to conventional scheduling. By implementing the speed control strategy, conflicts in truck route planning are effectively minimized, allowing for smoother and more efficient transportation operations.
The reduced wait time is of paramount importance as it directly translates to improved transportation efficiency. With fewer conflicts and delays, vehicles can complete their assigned routes in a timelier manner, resulting in enhanced overall productivity and customer satisfaction. The speed control strategy ensures that vehicles can navigate through the road network more effectively, optimizing their travel paths and minimizing potential bottlenecks or congestion points.

The positive impact of the speed control strategy on transportation efficiency reinforces the significance of conflict resolution in road planning. By addressing conflicts through the three-step optimization algorithm proposed in this paper, the overall performance of the transportation system is greatly improved. The algorithm takes into account various factors such as traffic patterns, delivery schedules, and vehicle capacities to optimize route planning and minimize conflicts.

The findings presented in Fig. 7 confirm that the integration of the speed control strategy, along with the proposed three-step optimization algorithm, leads to substantial improvements in conflict resolution and overall transportation effectiveness. This provides valuable insights for decision-makers and stakeholders in the field of logistics and transportation management, emphasizing the importance of implementing intelligent strategies to enhance operational efficiency and customer service.

The curves in Fig. 8 illustrating a reduction in task execution time for trucks in a studied port due to the synchronization of handling systems, gantries, cranes and speed control to avoid collisions indicate the positive impact of these measures on operational efficiency. By coordinating the various components involved in port operations and implementing collision avoidance mechanisms, the port can streamline its processes and improve its productivity.

Reduced execution time means that tasks are executed faster and more efficiently. Synchronizing material handling systems, such as gantries and cranes, ensures a smooth flow of operations by minimizing delays and bottlenecks. This coordination allows the port to optimize resource allocation and effectively prioritize tasks.

Incorporating speed control mechanisms to prevent collisions is crucial to maintaining safety while maximizing productivity. By regulating the movement of trucks and equipment, potential accidents and disruptions are mitigated, leading to uninterrupted operations and improved overall performance.

It should be noted that the effectiveness of these measures is influenced by the number of tasks performed and their duration. As the number of tasks increases, the benefits of timing and speed control become more pronounced, resulting in greater time savings. The relationship between the number of tasks and the proportional reduction in time implies that the impact of these measures scales accordingly, leading to increased efficiency across different workloads.

Overall, the curves showing a reduction in lead time demonstrate the positive results achieved by synchronizing handling systems and implementing speed control measures. These improvements result in increased productivity, minimized delays, increased security and optimized allocation of resources in the studied port.

VI. CONCLUSION

As a sophisticated and information-intensive logistics system for loading and unloading operations, the developed port terminal at Tangier Med relies extensively on efficient truck transportation to establish seamless connections between the seaside loading and unloading area and the landside storage zone. The overall operational efficiency of the terminal greatly hinges upon effective time management and synchronized coordination between the transportation and handling systems. In light of this, we propose a three-stage model that tackles the truck route planning predicament by considering task allocation and integrating a speed control strategy to ensure collision avoidance during truck movement. The model is solved through the utilization of a heuristic algorithm, and its efficacy is exemplified through a comprehensive large-scale numerical demonstration, culminating in substantial enhancements to the operational efficiency of the Tangier Med port terminal. Nevertheless, practical circumstances necessitate the consideration of additional factors, such as the impact of truck charging on their movements. Therefore, future research endeavors focusing on devising collision avoidance routines for trucks while taking their load into account emerge as pivotal directions to explore for the further development of the Tangier Med port and its terminal.

REFERENCES


Abstract—The early diagnosis and treatment of lung tumour, the primary cause of cancer-related deaths globally, depend critically on the identification of lung tumours. In this approach, a new method is suggested for detecting lung tumours that combines a Gaussian filter with a hybrid Recurrent Neural Network-Generative Adversarial Network (RNN-GAN). Utilising the sequential data seen in images of lung tumours, the RNN-GAN architecture is used. In processing the sequential input, the RNN component looks for temporal relationships and patterns. The GAN component improves the training of the RNN for accurate classification by creating synthetic tumour specimens that resemble actual tumour images. In addition, the proposed approach pre-processes lung tumour images using a Gaussian filter to improve their quality. The Gaussian filter improves feature extraction and the visibility of tumour borders by reducing noise and smoothing the pictures. The proposed experimental findings on a dataset of lung tumours show that the suggested strategy is successful. In comparison to conventional techniques, the hybrid RNN-GAN delivers higher accuracy in lung tumour identification due to the incorporation of the Gaussian filter. While the GAN component creates realistic tumour samples for improved training, the RNN component efficiently captures the sequential patterns of tumour images. The lung tumour images are pre-processed using a Gaussian filter, which greatly enhances image quality and facilitates precise feature extraction. The proposed hybrid RNN-GAN with the Gaussian filter shows promising potential for accurate and early detection of lung tumours. The integration of deep learning techniques with image pre-processing methods can contribute to the advancement of lung cancer diagnosis and treatment, ultimately improving patient outcomes and survival rates. Further research and validation are necessary to explore the full potential of this approach and its applicability in clinical settings.

Keywords—Lung tumour; recurrent neural network; generative adversarial network; CT images; hybrid

I. INTRODUCTION

Lung cancer has become one of the main causes of cancer-related death in many countries all through the world. According to data on cancer worldwide, a startling number of new diagnoses and fatalities in 2018 were caused by lung cancer. There were a total of 1,761,007 fatalities, representing 18.4% of all cancer-related fatalities, and 2,093,876 newly identified instances, or 11.6% of all cancer sites [1]. The term "lung cancer" describes the unchecked proliferation of cancer cells in the lungs, which raises both male and female morality rates. While it cannot be totally prevented, there are some measures to lessen the risk of this illness, which is characterised by the fast proliferation of lung cells. For patients to have a better chance of surviving lung cancer, early identification is essential [2]. Lung cancer can be found via diagnostic imaging tests. Complex lung cancer datasets are examined to discover essential features using machine learning (ML) methods. A CAD method was created in the early period to help clinicians analyse medical pictures in an effort to increase effectiveness and the rate of survival. Healthcare has greatly benefited from a variety of machine learning methods. Additionally, we have investigated deep learning approaches, methodologies, and algorithms that may be used for identifying, detecting, and forecasting various cancer kinds [3].

Pathologists may be helped by the application of CNN-based approaches in the identification of lung cancer [4]. Selvanambi et al. recommended a more complex recurrent neural network that was optimised via glow-worm swarm optimisation to present a unique method for predicting lung cancer [5]. The use of computed tomography (CT) as a diagnostic tool for pulmonary disorders has become more important. Particularly, columnar CT is widely regarded as helpful tools in the diagnosis of scattered interstitial lung
disease (DILD) images have been the subject of several researches aimed at improving computerised analysis of images and computer-aided diagnostic (CAD) systems. These developments are meant to increase the precision and efficacy of pulmonary disease diagnosis across the board, including diffuse interstitial lung disease (DILD) [6]. With the advancement of computer-assisted diagnosis and detection techniques, several efforts are being made to improve the clinical effectiveness of lung cancer detection and categorization. Sajja et al. [7] advised deep learning and transfer learning to detect lung cancer. Their study demonstrates how effective this method is in accurately identifying lung cancer cases. Bhatia el al. [8] presented a feature extraction strategy based on deep residual networks for the identification of tumours. Tulin Ozturk, et al. [9] suggested a model using deep learning for automating tumour diagnosis.

The American Cancer Society estimates that every year, between 10 and 20 percent of all cancer patients (or around 1.7 million people) receive incorrect diagnoses, as well as that at least 40,000 of these individuals end up dying as a direct result. Furthermore, cancer is the second-leading cause of mortality in the US. There is a chance of making a mistake when a pathologist has, on a typical basis, 80 Computed Tomography (CT) slides each patient to assess. Therefore, it is necessary to computerise the tumour diagnosis procedure in order to cut down on the possibility of human mistake. Monitoring with low-dose CT, which has been demonstrated to lower lung cancer mortality by up to 20%, has the potential to identify lung cancer at an earlier stage. However, diagnosing a lung CT scan is a highly specialised process needing specialist understanding. If done manually, it is a difficult and time-consuming process. False negative situations occur when a cancer manages to escape detection by the human eye since it may resemble other lung-existing particles in terms of size and texture. The current study primarily focuses on the qualitative examination of a cancer that a pathologist manually found during a single CT scan. The literature lacks a completely automated framework that can automatically detect tumours across a series of CT scans in a single run, giving the viability of the framework priority. Despite this, there would be 28.40 million new instances of tumours worldwide in 2040, a 48 percent rise from the previous year. An increase in risk factors, such as smoking, will make (even worse. people with respiratory issues, people who have been smoking for 30–40 years, patients who experience no symptoms, as well as patients with no symptoms are common hurdles in identifying lung tumours in patients from the decade. In an effort to overcome the difficulties involved in the diagnosis of tumours, various investigators have employed a variety of strategies, such as differentiation, recognition, and classification methods [10].

Using a Gaussian filter, the picture is first enhanced by data pre-processing, and the noise in the CT representation is then eliminated. The pre-processing methods divide the incoming data into several categories. Pre-processing will be followed by the extraction of the affected lung region. The afflicted area of the lung tumour is found using segmentation. The segmentation of the CT representation facilitates the extraction of useful characteristics for the procedure. To prepare the input image for testing and training, two sets are divided. During training and testing, the errors in the samples of data are removed. Then, using the feature extraction process, which extracts the features using the Grey Level Co-occurrence Matrix (GLCM), the cancerous lung nodule is found. The RNN-GAN hybrid model is used to distinguish between the damaged and unaffected parts of the lungs.

The following are the research's main contributions:

- From a lot of cases, CT pictures are initially gathered.
- In addition, undesirable noises are present in the reconstructed real CT lung pictures, which are filtered using a sophisticated Gaussian filter model.
- The segmentation procedure has been carried out using an enhanced K-means clustering.
- The Grey Level Co-occurrence Matrix (GLCM) was used to extract features.
- The afflicted and unaffected lung nodules are categorised by the RNN-GAN.
- To demonstrate the usefulness of the implemented technology, its performance is verified and compared to current methodologies.

The manuscript of the approached paper is organised as follows: In Section II, some related works are reviewed. In Section III, Information regarding the problem statement is provided. In Section IV, the proposed RNN-GAN is covered in detail. In Section V, experiment results are provided, and discussed, and an extensive evaluation of the proposed approach to current best practices is made. In Section VI, the conclusion of the paper is provided.

II. RELATED WORKS

Rahman et al. [11] utilized several open datasets to construct a database comprising 3500 chest X-ray images infected with TB and 3500 images of healthy chests. They developed a reliable method for consistently identifying tuberculosis from these chest X-ray images. Image initial processing, enhancement of data, segmentation of images, and deep learning techniques classification algorithms were all used in the methodology. The success of a transfer learning strategy employing deep CNN for automatically detection of tuberculosis is discussed in conclusion. The study assessed nine various CNN models and found that the ChexNet model performed well (96.47%, 96.62%, and 96.47%) without image segmentation, while the DenseNet201 model excelled with (98.6%, 98.57%, and 98.56%) segmented lungs. The study also emphasized the significant improvement in classification accuracy through image segmentation. The Score-CAM visualization results provided confirmation of the key role played by lung segmentation in ensuring accurate decision-making based on the lung region. One potential drawback of the proposed method is that it focuses only on the detection of TB without considering the detection of other lung abnormalities or diseases. While this approach is valuable for TB detection it may not be suitable for identifying other lung conditions or providing a comprehensive analysis of the patient's chest radiograph.
Bharati, et al. [12] proposed a new deep learning hybrid framework by merging (STN) VGG with CNN and named as VDSNet to detect the lung disease. The dataset obtained from the Kaggle library was subjected to this methodology. There are a sizable number of lung X-ray pictures in the investigation's dataset. This model has the greatest validation accuracy of 73% for the complete dataset example. Comparatively, the accuracy of the validation values for various other models, are 69%, 67.8%, 60.5%, 69.5%, and 63.8%, respectively. With an accuracy for validation of 73%, this framework outperformed the sample dataset's efficiency of 70.8%. However, VDSNet requires more time for training, with the process taking 431 seconds as opposed to the sample dataset's 19 seconds.

Asuntha & Andy Srinivasan [13] focuses on using Deep Learning methods to locate cancerous lung nodules and classify lung cancer severity. Fuzzy Particle Swarm Optimisation Convolutional Neural Networks (FPSOCNN) is a unique method that is suggested for lowering computational complexity. We use a number of feature extraction methods, such as HoG, wavelet transformations, LBP, SIFT, and Zernike Moment. The FPSO algorithm is used to extract texture, geometric, volumetric, and intensity characteristics and choose the best feature; evaluation utilising a unique dataset (DICOM) from the Indian hospital Aarthi Scan. Nearly 10,000 lung pictures are included. All images have a resolution of 256 by 256. The experimental data demonstrate that new FPSOCNN outperforms existing methods. The optimisation and classification performance of the suggested model still require work. This suggests that the suggested paradigm could have some drawbacks.

Kim et al. [14] employed a technique called transferred learning to elevate the accuracy and efficacy of CAD and created a deep learning approach for classifying lung disease in chest X-ray (CXR) images. The method used CNN models, particularly the Efficient Net v2-M model, as well as the transfer learning techniques, empirical hyper-parameters, and CNN models. The one-step, end-learning method utilised to gather useful features for sickness categorization involved simply feeding raw CXR images into the deep learning system. Two datasets were used for the experiments, which include a privately created database with four categories and an open-access benchmark dataset from the U.S. National Institutes of Health with three distinct groups. The suggested strategy produced reliable results in precisely classifying lung diseases. The suggested method's disadvantage is that the success rate for the different categories varied, with the usual categories having the lowest possible accuracy at 63.60%. This suggests that in comparison to the other lung disease categories, the model would have trouble properly classifying normal patients.

Demir et al. [15] a range of sample frequency ranges, noise from the background, and other noises are included in the CBHI 2017 database, which was suggested. The lung sound waves were transformed into spectrogram visualisations. Two methods based on deep learning were used to classify lung sounds. In the first approach, lung sounds were categorised using an SVM classifier, and features were retrieved using an already trained CNN. In the second technique, the previously trained deep CNN model was modified by transfer learning while lung sounds were identified using spectrogram images. The accuracy of the first and second suggested approaches was 65.5% and 63.09%, respectively. This strategy is then demonstrated to perform better than other results by comparison to the current results. The problematic elements in the suggested approach for classifying lung sounds include noises, background noise, and different sampling frequencies. This suggests that a smaller dataset may have been used to test the model.

Shakeel, et al. [16] focuses on boosting the accuracy of lung cancer detection and the quality of lung pictures. The (CIA) dataset's lung CT pictures were used. A weighted mean histogram equalisation method was used to minimise misclassification and improve picture quality by removing noise from the photos. Further enhancing picture quality, the afflicted region was segmented using an IPCT. A deep learning neural network was instantly developed to anticipate lung cancer using the partitioned region's multiple spectral properties. The technique achieved a success rate of 98.42% with a low classification error of 0.038, and its effectiveness was assessed using MATLAB-based simulation results. These results show how well the algorithm performs in correctly identifying lung cancer and enhancing lung pictures. It is noted that without data enhancement and transfer learning, categorising numerous lung illnesses on grayscale chest X-ray (CXR) pictures is difficult. A trade-off exists when dropout regularisation is used among overfitting and losing important information. A low rate of abandonment might result in acceptable performance on the initial training set but poor performance on the validation or testing set, whereas a large dropout rate could result in the loss of critical features.

Chen, et al. [17] proposed to assess chest X-ray pictures of common pulmonary diseases in children. A computer-aided diagnosis method has been created. The appropriate lung field is automatically recognised and cropped by the algorithm using the YOLOv3 architecture. The one-versus-one classification system outperformed the other two when they were compared to each other. In recognising certain illnesses and detecting irregularities, the system had great accuracy rates. The suggested method's limitation to useable medical resources is its main flaw. Issa, et al. [18] examined how well four pre-trained participants performed using CXR pictures to identify various lung illnesses. Utilising CXR pictures and 5-fold cross validation, the model was trained and evaluated. With a remarkable area under the curve (AUC) of 99.84% for ROC, XceptionNet had the greatest accuracy (94.775%). With regards to accuracy, quick convergence, resource use, and almost real-time detection (0.33s), DarkNet19 demonstrated an excellent balance. With a lengthier prediction time (5.68s), the ensembles feature technique, which included many models, obtained the maximum accuracy of 97.79%.

III. PROBLEM STATEMENT

The problem addressed in the research is the development of a deep learning-based CAD system for classifying lung diseases in CXR images, with the ultimate aim of enhancing diagnostic accuracy and efficacy in medical imaging [14]. The database contains a range of sample frequency ranges,
background noise, and other noises, making lung sound classification challenging. The researchers sought to develop effective methods based on deep learning to classify these lung sounds accurately [15]. In this proposed approach a unique method for Optimized Ensemble of Hybrid RNN-GAN Models for Accurate and Automated Lung Tumor Detection from CT Image.

IV. PROPOSED RNN-GAN

By incorporating RNN-GAN, the proposed method effectively leverages the sequential nature of lung tumour, capturing temporal patterns that may be crucial for accurate tumour identification. The GAN component plays a vital role in creating synthetic tumour samples, improving the training process, and augmenting the limited dataset to better handle class imbalances and improve classification accuracy. The application of the Gaussian filter enhances image quality and reduces noise, leading to improved feature extraction and better visibility of tumour borders, ultimately contributing to more reliable and precise tumour detection. The combination of deep learning techniques with image pre-processing methods shows promise for early detection of lung tumours, which is critical for improving patient outcomes and survival rates. The proposed method represents a significant step in the field of lung cancer diagnosis and treatment, demonstrating the potential of integrating cutting-edge techniques for addressing complex medical challenges.

The computerised tomography representations of lung tumours are first gathered. The simulations are then used for teaching and testing. The Gaussian filter is applied in the pre-processing of lung-dependent CT images to remove extra noise. In this work, lung nodules are first detected using a mixed RNN-GAN model. The severity of lung nodules is categorised using the suggested RNN-GAN model. Fig. 1 shows the RNN-GAN model.

![Proposed RNN-GAN model](image)

Fig. 1. Proposed RNN-GAN model.

A. Data Collection

CT lung representations of 10,000 datasets containing healthy and unhealthy lung representations collected from the University of California are utilized in the research. Among these 50% of representations are used for training and 50% of representations are utilized for testing [20]. The assessed system was tested using many cancer datasets form the University of California, Irvine collection. The data set has 57 high points, but only four of them may be looked at with conventional analysis of components. The lungs dataset was utilised in evaluating their programmed method for cancer identification. There are 32 occurrences, 57 features, and a single class characteristic altogether in the collection. The dataset, which consists of 32 samples with a total of 57 characteristics and a theoretical range of 0–3, was found in the database for machine learning at UCI. Here 2500 is a tumour data. The collected datasets are given in Table I.

<table>
<thead>
<tr>
<th>TABLE I. COLLECTED DATASETS</th>
</tr>
</thead>
<tbody>
<tr>
<td>Unhealthy</td>
</tr>
<tr>
<td>Training data</td>
</tr>
<tr>
<td>Testing data</td>
</tr>
</tbody>
</table>

B. Data Augmentation

The data augmentation technique creates augmented training data using a style-based generative adversarial network (GAN). It makes sense to utilise data augmentation techniques to address the issues of data imbalances and scarcity. This strategy has shown to be a successful means of enhancing model performance and avoiding overfitting. Utilising GAN to enhance data or create training samples is another genre. The GAN training phase and enhanced image generation are the two components of this approach. The design of the generator and the discriminator are both optimised concurrently throughout the training phase. They are instructed to use both the semantic label and the style matrix to rebuild the actual image [21].
A generating network (G) and a discriminative network (D) are the two deep networks that make up the RNN-GAN's architecture, which is shown in Fig. 2. As a condition, G produces a series of 2D segmentation semantic outputs from a set of 2D images, and D determines whether or not those outputs are real. RNN-GAN experiences improvements where better resolution attributes are coupled with upsampled relatively not high-quality image attributes, helping the neural network to learn both global and local data. Typically, a GAN's generative model G aims to translate the output image v into the erratic noise vector z; G: z → y. The possibility that a sample comes from the source of data x false as opposed to its data for training x genuine is calculated by a model exhibiting bias D in the meanwhile. More specifically, an RNN-GAN network is proposed, and a generative model learns its semantic classification of matching labels from a set of 2D medical images u, v_{seg}; G : u, z → {v_{seg}}. The discriminator uses the ground truth and the generated data to assess if the label that was predicted is true or false whereas the segmentation was anticipated at the pixel level by the generator which is represented in eq. (1).

$$\mu_{adv} \leftarrow \min_{G} \max_{D} k(G,D) = K_{u,v_{seg}}[\log D(u,v_{seg})] + K_{u,z}[\log(1-D(u,G(u,z)))]$$  

(1)

C. Pre-Processing

The initial step in finding cancerous pulmonary nodules is pre-processing. It is used to remove unnecessary data in addition to fill in dataset gaps. The peculiar and abnormal vibrations that influence the CT portrayals slow down the rate of evaluation of the first visualisations. The CT depictions are most affected by the irregular noises that are caused by internal and external causes. As a consequence, the noise in CT lung depictions is reduced using the resulting Gaussian filter. The created Gaussian filtering technique is used to minimise residual deviations of spatial severity in the depictions as well as disturbance in the representations. The use of the filter known as the Gaussian causes the mean value of the surrounding pixels, which depends on the distribution of Gaussian, substitutes the distorted pixel within the representation. The RNN-GAN model makes use of noise-reduced representations to identify lung cancer nodules [22].

The Gaussian function is given in eq. (2).

$$G(V) = \frac{1}{\sqrt{2\pi}\delta}e^{\frac{y^2}{2\delta^2}}$$  

(2)

Where δ is the standard deviation of the distribution. The distribution is assumed to have a mean of 0.

D. Segmentation Using K-Means Clustering

The segmentation approach is mostly used for separating the impacted area in illustrations based on CT images. For image processing to be implemented, the segmentation procedure must be accurate. Segmenting an image is frequently used to determine the location of the affected nodules as well as the constraints of the images’ curves and lines. The representations are divided into groups of pixels, and each group’s pixels are annotated. Classification of cancer
nODULES and the generation of sufficient data to perform additional identification are the primary goals of picture segmentation in medical representation processing. The technique of dividing a picture into several sections, each with a different collection of pixels, is known as image segmentation. The method of global thresholding technique depends on the gray-level pixels’ luminosity for threshold P, projected to divide the picture. Using eqn. (3) and \( I(u, v) \), it is possible to describe the fragmented picture that was obtained by global thresholding. The image’s pixel value is represented here by \( I(u, v) \).

\[
I(x, y) = \begin{cases} 1 & \text{if } I(u, v) > P \\ 0 & \text{if } I(u, v) \leq P \end{cases}
\]  

(3)

By implementing a certain structural element at every site that is feasible for smoothing the region of interest, quantitative morphological procedures are estimated. When \( F \) is a binary picture and \( J \) is an organising element, as shown in eqns. (4), (5), (6) and (7), respectively, mathematical computations are carried out.

\[
\text{Erosion} : L \ominus J = \{H[J \subseteq L]\} \quad (4) \\
\text{Dilation} : F \oplus J = \{H[J \cap F \neq 0]\} \quad (5) \\
\text{Opening} : F \ominus J = L \ominus \{J \oplus L\} \quad (6) \\
\text{Closing} : F \oplus J = L \oplus \{J \ominus L\} \quad (7)
\]

The simplest and most traditional method of analysing clusters is the k-mean clustering algorithm. To divide the provided dataset through two or more groups, k-means is used. Since choosing the right cluster core is crucial for obtaining the most beneficial results, the correctness of the procedure is assessed by examining every single cluster centre that the algorithm generates. The Euclidean distance, that is employed to assign pixels to certain clusters, is a pretty straightforward technique for separating the dataset. This algorithm makes use of the following function as represented in eq. (8).

\[
k = \sum_{i=1}^{m} \sum_{j=1}^{n} V_{ij} \|u^i - y_j\|^2
\]  

(8)

Where, \( V_{ij} \) is the number of data points for the \( i \)th cluster, \( u^i \) is the number of pixels, \( V^i \) is the cluster centres, \( \|u^i - y_j\| \) is the Euclidean distance between \( u^i \) and \( V^i \), and \( C \) is the number of cluster centres. In the anticipated phase, points of information are allocated to the nearest cluster. Each cluster’s kernel is determined during the optimisation process [23].

E. Feature Extraction

In order to record and retain the details of the initial collection of data, the process of separating characteristics requires turning unorganised information into quantitative attributes. Each patient has a unique way of processing information, and these characteristics are derived from the full collection of depictions that were taken. The number of dimensions of the depiction must be lowered for the purpose to detect lung nodules while the dimension of the representation grows throughout testing. To fix this problem, the extraction of features is done. The Grey Level Co-occurrence Matrix (GLCM) is used in the feature extraction. By computing various combinations of pixels with specific values, it shows how the visual representation is organised hierarchically. The GLCM employs the associated grayscale and displays the luminosity of the depicted pixels. To remove the statistically important textureing characteristic, the second-degree representation’s energy, contrast, correlation, entropy, homogeneity and other qualities are assessed.

The extraction of features stage is carried out using the GLCM technique based on the scenario. The GLCM functions determine how frequently pairs of pixels with specific parameters and in a particular arrangement of pixels appear in an image in order to describe the surface texture of that image. Statistical variables generated from the following GLCM outcomes. The formulas below were used to determine these values:

1) Energy: When the total values are asymmetrical and usually greater, energy is defined as the presence of an equal number of grey levels in portrayals. eqn. (9) calculates the energy of the provided data.

\[
E = \sum_o \sum_p (Q(o,p))^2
\]  

(9)

Where the representations are denoted as \( Q \) and grey level squares are denoted as \((m,n)\).

2) Contrast: Attributes are used to assess the local contrasts of a representation, so if it is in a consistent saturation value, it is assessed as a low value. The number of grayscale values in the original form as well as the contrast are projected in eqn. (10)

\[
C = \sum_{y=0}^{y-1} \sum_{p=0}^{p-1} \sum_{x=0}^{x} \sum_{o} (Q(o,p) - Q(O,p))^2
\]  

(10)

Where R indicates the grey level of the representation, \( Q \) indicates the representation, and \((o,p)\) indicates the grey level square of the representation.

3) Correlation: The relationship between the features can be used to account for both the linear dependence of the grey levels on the pixels and statistical interactions between the parameter values. The characteristics are exposed in eq.1. (11).

\[
C = \frac{\sum_{o} \sum_{p} Q(o,p) Q(o,p)}{\sigma_x \sigma_y}
\]  

(11)

In the representation, the values of mean, as well as standard deviation, are \( \delta_x, \delta_y, \sigma_x, \) and \( \sigma_y \) are defined as row and column.

4) Entropy: The anticipated significant amount of the uncertainty of the split of the grey levels is known as entropy, which is denoted in eq. (12) [24].

\[
E_n = -\sum_o \sum_p Q(o,p) \log_2 (Q(o,p))
\]  

(12)

5) Mean: Mean indicates the overall brightness within the matched image. It has very limited expressive and discriminating skills which is denoted in eq. (13).

\[
M = \frac{\sum_{o} Q(o,p)}{\sum_{o} \sum_{p}}
\]  

(13)
6) Energy features: Angular second instant is another name for energy. The grey level occurrence matrix's sum of squared elements is returned. Energy for a constant picture will be 1 and it is denoted in eq. (14)

\[ E = \sum \sum Q(o, p)^2 \]  

(14)

7) Homogeneity: It evaluates how well the GLCM's constituent distribution matches its diagonal. Which is denoted in eq. (15) [25].

\[ H = \sum_i \sum_j \frac{Q(i,j)}{1 + ||i-j||} \]  

(15)

F. Classification using RNN-GAN

In order to reduce the total distinction amongst the anticipated value and the biggest value already in existence, the loss caused by opposition with 1 distance is used. The small quantity is made up of 2D pictures taken from an identical patient using the identical acquisition surface during the stages of training and testing. Both the variance and the mean are estimated for an individual from the exact same acquisition plane as well as similar phase, and then the recommended method first normalises the inputs from all available photos. The impact of anomalies is reduced because to this normalisation. The recommended method utilised batch standardisation to equalise the signals that were employed (activations from the previous layer) coming into every layer specified in eqns. (16) and (17) using the mean and variance of all the activation for the entire mini-batch. There is less turbulence and the borders are smoother since the parameter function takes into account RNN features a

\[ \mu_1 = K_{uz} ||vseg - G(u, z)|| \]  

(16)

\[ \mu_2 = \frac{1}{d} \sum_{i=1}^{d} \sum_{j=1}^{d} \frac{v_{i,j}^{seg\cap G(u_{i,j}, z)}}{\sum_{i=1}^{d} \sum_{j=1}^{d} v_{i,j}^{seg\cap G(u_{i,j}, z)}} \]  

(17)

where \( u \) and \( z \) represent, correspondingly, the quantity of 2D slices and the degree of semantic classes for every patient. Additionally, the suggested method combined categorised loss of accuracy, to reduce unbalanced data from training by allocating a greater cost to a fewer-represented group of pixels, raising its value throughout the learning process. Classification accuracy loss determines if the maximum real value and maximum forecasted value for each classification category are equal. The final of RNN is calculated through eq. (18) [26].

\[ \mu_{RNN}(D, G) = \mu_{adv}(D, G) + \mu_1(G) + \mu_2(G) \]  

(18)

In traditional generative modelling approaches, the performance of a model is indicated by the reverse divergence between the desire distribution \( Q_t \) and our generator’s distribution \( Q \), as represented in eq. (19).

\[ V_{kl}(Q_t||Q) = \int x Q_t \log \frac{Q_t}{Q} dx \]  

(19)

RNN-GAN is a mixture of the RNN and GAN, two well-known deep learning models. While GANs are used to produce accurate patterns of distribution, RNNs are utilised for processing data sequentially. RNN and GAN frameworks should be combined for cooperative training. Actual and false lung tumour pictures produced by the GAN are used to train the RNN. The RNN gains the ability to distinguish between benign and malignant pictures. Utilise the test set of lung tumour pictures to assess the trained RNN model’s performance. The proposed approach need to alter the RNN-GAN model’s hyper-parameters, architecture, or data augmentation strategies in accordance with how well it performs. The goal of this stage is to increase the model’s generalisation and consistency capabilities.

Fig. 3 represents the proposed flow diagram for lung tumour detection. Initially the data is loaded. The images pre-processed by Gaussian filter to remove the noise present in the image. The pre-processed image is segmented using k-means clustering technique. Then the features are extracted by GLCM technique and finally the proposed classifier classifies the tumour.

Algorithm: RNN-GAN mechanism

Input: CT representation of lung nodules

Output: Detection of affected and unaffected lung nodules

Load input representation data

Train the input images \( v_i \) in the system, where \( i = 1 \) to \( n \)

Pre-processing of images

Let \( U(i) \) be the input images from the dataset

\[ \text{for every } U_i \] // Gaussian filter

\[ V(i) = V(i) - N \] // \( N \) denotes unwanted noise

Segment the affected part
// Bat and Whale Optimization

Initially, the affected part is detected
If( initial region is met)

Update the tumour present
Else

Find the initial location

Repeat until the stopping condition is reached // until all the tumour region is identified

End if

Return

Feature extraction using

GLCM

Classification of tumour // RNN-GAN classifier
Fig. 3. Flow diagram of the proposed model.
V. RESULT AND DISCUSSION

The recommended method has been evaluated using lung computed tomography datasets and executed in MATLAB software on the Windows 10 platform. The combination of revolutionary RNN-GAN is used to detect a lung tumour when contrasting datasets from healthy and unwell individuals. Performance indicators like Accuracy, Precision, Recall, and F-measure are used to evaluate the effectiveness of proposed methodology.

The proposed model will be implemented using the python programming language. The test is run on a machine equipped with an Intel(R) Core(TM) i5-3470 CPU @ 3.20 GHz, 3200 MHz, 4 core(s), 4 logical pro. And micro soft corporation, is an OS manufacturer installed physical memory (RAM) 8GM.

A. Accuracy

The model's complete precision demonstrates how well it functions throughout every classification. In general, it is the notion that all occurrences will be accurately predicted. Eq. (20) expresses accuracy.

\[
A = \frac{T_{pos} + T_{neg}}{T_{pos} + T_{neg} + F_{pos} + F_{neg}} \tag{20}
\]

B. Precision

The number of exact positive ratings that differ from the total number of positive evaluations is used to calculate precision. Eqn. (21) is used to compute the portion of precisely identifying the afflicted area as having tumour nodules.

\[
P = \frac{T_{pos}}{T_{pos} + F_{pos}} \tag{21}
\]

C. Recall

The recall is the correlation between the overall quantity of accurately classified samples that are positive and the number of true positives. It displays the percentage of predictions that correctly identified the malignancy nodules that is given in eq. (22):

\[
R = \frac{T_{pos}}{T_{pos} + F_{neg}} \tag{22}
\]

D. F1-Score

The F1-Score computation combines precision and recall. Precision and recall are used to construct the F1-Score given in eq. (23):

\[
F = \frac{2 \times \text{Precision} \times \text{Recall}}{\text{Precision} + \text{Recall}} \tag{23}
\]

TABLE II. PERFORMANCE EVALUATION

<table>
<thead>
<tr>
<th></th>
<th>CNN</th>
<th>RNN-GAN</th>
</tr>
</thead>
<tbody>
<tr>
<td>Training</td>
<td>99.1</td>
<td>99.8</td>
</tr>
<tr>
<td>Testing</td>
<td>97.5</td>
<td>98.9</td>
</tr>
</tbody>
</table>

The accuracy of the convolutional neural network used for both the training and testing stages is 99.4% and 97.6%, respectively, according to Table II. When RNN-GAN is utilised, the accuracy of the testing and training processes increases to 99.8 and 98.9, respectively. Fig. 4 shows an evaluation of performance.

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When compared to the current lung tumour detection techniques, the suggested technique RNN-GAN obtains a greater level of accuracy. The contrast of efficiency between RNN-GAN and other approaches is shown in Fig. 5 and Table III.
Table IV demonstrates that the proposed technique of combined RNN-GAN achieves higher precision and recall of 100% and 99% when compared to the existing lung tumour nodule detecting methods the advanced RNN-GAN gives better accuracy than the performance evaluated. Here, the achieved accuracy level is 98.92% using the RNN-GAN model. Fig. 6 illustrates the precision and recall between RNN-GAN and other methods.

In Fig. 7, the performance metrics for each class are presented. The analysis involves two classes: "without tumour" and "tumour." The metrics include accuracy (98.6%), precision (100%) for "without tumour," and accuracy (98%), precision (99%) for "tumour." Additionally, the f1-score and recall values for both classes are provided, indicating the model's overall classification performance for each category.

VI. CONCLUSION

Today's healthcare technology is rapidly advancing image processing, but in some cases, the intricate nature of the depictions makes it difficult to categorise and diagnose disorders. The ill or damaged section is classified, characterised, and segmented using the indicated method. It mainly focuses on locating lung tumour nodules. First, data from patients with various types of lung tumours are acquired from their CT scans to learn more about the patient. The unexpected effects or distortion in the images from CT are then removed employing a Gaussian filter that is applied during pre-processing. The RNN-GAN framework is utilised for scenarios including the recognition of lung cancers. The RNN-GAN architecture enables sequential processing of lung cancer pictures and the creation of realistic tumour samples by combining the strengths of RNNs and GANs. Because the data are sequential, the RNN component of the RNN-GAN can accurately classify the lung tumour images by detecting relationships between time and patterns. The RNN can use the synthetic tumour samples produced by the GAN component to train and improve its capacity to distinguish between benign and malignant tumours. The RNN-GAN architecture learns to categorise lung tumour images through joint training using both actual and produced tumour samples. While the GAN creates realistic tumour samples to aid in learning, the RNN is taught to categorise the photos. The RNN-GAN strategy has the potential to increase the precision and effectiveness of lung cancer detection systems by utilising the complementary capabilities of sequential data processing provided by RNNs and generative modelling provided by GANs. To guarantee the success of the RNN-GAN framework for lung cancer detection tasks, it is crucial to take into account the unique implementation aspects, such as architectural choices, training methods, and dataset quality. The proposed method requires careful evaluation on larger and diverse datasets to ensure generalization to various tumour types and patient populations. Additionally, the computational cost of using RNN-GAN and Gaussian filters may be a challenge for real-time applications; optimizing the model's efficiency is a critical aspect for future research and practical deployment. To validate and perfect the RNN-GAN technique for practical lung tumour detection applications, more investigation and testing are required. Consequently, this model’s forecast precision was 98.6%.

REFERENCES


Abstract—There is no doubt that pancreatic cancer is one of the most deadly types of cancer. In order to diagnose and stage pancreatic tumors, computed tomography (CT) is widely used. However, manual segmentation of volumetric CT scans is a time-consuming and subjective process. It has been shown that the U-Net model is highly effective for semantic segmentation, although several deep learning models have been proposed. In this study, we propose a U-Net-based method for pancreatic tumor segmentation from abdominal CT images and demonstrate its simplicity and effectiveness. Using the U-Net architecture, the pancreas is segmented from CT slices in the first stage, while tumors are segmented from masked CT images in the second stage. For validation set of NIH dataset and according to the proposed method's dice scores, the pancreas segmentation and tumor segmentation performance was outstanding, demonstrating its potential to identify pancreatic cancer efficiently and accurately.

Keywords—U-net; deep learning; segmentation; computed tomography images; hyper parameters; PDAC

I. INTRODUCTION

The pancreas is an accessory organ, an exocrine gland of the digestive system, and an endocrine gland that generates hormones. In a healthy adult, the pancreas weighs around 100g, ranges in length from 14 to 25 cm, and has a volume of roughly 72.4 to 25.8 cm³. It has an extended lobular form. The five anatomical divisions are the uncinate process, neck, body, and tail. Release of digestive enzymes that assist the digestion of fatty foods is the primary duty of the exocrine pancreas. The endocrine gland assists in regulating blood sugar levels and cell nutrient uptake. Exocrine pancreas cancer is referred to as "pancreatic cancer". It is one of the most common cancers, particularly in western countries and Japan. Pancreatic cancer is the melanoma that occurs in Americans the second most frequently. It accounts for 5% of all cancer mortality in that country and is more common in African Americans. It strikes men more commonly than women. The incidence increases after age 50. Pancreatic cancer develops when the cells of the pancreas undergo abnormal DNA modifications; this leads to the cells' uncontrolled division and growth into tumors. Occasionally, the liver, abdominal wall, lymph nodes, lungs, or bones may become infected by this tumor. The risk factors for getting pancreatic cancer include smoking, being overweight, having long-term diabetes, having a substantial family history of the illness, eating a lot of processed food and red meat, and having chronic pancreatitis. It is the fourth most common reason for cancer-related deaths in the west. It is predicted to overtake cancer as the second-deadliest illness in 10 years. It has an annual incidence rate of 12.50 per 100,000 persons, accounting for 3% of all cancer cases in America. One of the most fatal cancers, pancreatic cancer has a five-year mortality rate of less than 10%. Computed tomography (CT) scans of pancreatic tumors must be accurately segmented in order to make a diagnosis, determine a course of treatment, and track the disease's progression. Radiologists must manually segment images, which takes time and is prone to inter-observer variation. Deep learning models for automated segmentation have shown the potential in overcoming these constraints. The U-Net model is a convolutional neural network architecture designed for semantic segmentation tasks, particularly in biomedical image processing, including the segmentation of CT (Computed Tomography) images [15]. The U-Net architecture is named after its U-shaped design, which consists of two main parts: the contracting path and the expansive path. In this study the U-Net model is proposed to segment both pancreas and tumors, which are all cascaded together. When initializing the weights of the first U-Net, we employ Kaiming initialization to prevent the issues caused by disappearing or exploding gradients. A multi-class cross-entropy loss function, used by the second U-Net, is suitable for segmentation tasks. Our method is straightforward and efficient, requiring little time and resources compared to other deep learning methods that demand numerous layers and substantial computational resources for equivalent accuracy.

II. LITERATURE REVIEW AND RELATED WORK

The bulk of efficient deep learning algorithms for semantic segmentation of the pancreas and PDAC from abdominal CT scans are based on the U-Net, ResNet, AlexNet, and VGG-net models, as well as variants of these models.

The first of research was proposed by the Pancreas Segmentation in Abdominal CT Images with U-Net Model by [1] in their study using a convolutional neural network (CNN) called the U-Net model. The segmentation procedure used the Pancreas CT dataset from The Cancer Imaging Archive (TCIA) database, which included computed tomography images from 82 patients. The report provides a thorough explanation of the outcomes of this segmentation method. The Dice and Jaccard similarity coefficients, used to measure the efficacy of the U-Net model for pancreatic segmentation, produced values of 0.78 and 0.66, respectively. In [2] the study being reviewed introduces a framework based on convolutional neural networks (CNNs) for the segmentation of
pancreatic ductal adenocarcinoma (PDAC) mass and surrounding vessels in CT images. The proposed framework first localizes the pancreas region from the whole CT volume using a 3D-CNN architecture and 3D Local Binary Pattern (LBP) map of the original image. Then, segmentation of the PDAC mass is performed using 2D attention U-Net and Texture Attention U-Net (TAU-Net), which is introduced by fusing dense Scale-Invariant Feature Transform (SIFT) and LBP descriptors into the attention U-Net. The Dice score for PDAC mass segmentation in portal-venous phase by 7.52% compared to state-of-the-art methods in terms of DSC. In [3] AX-Unet: A Deep Learning Framework for Image Segmentation to Assist Pancreatic Tumor Diagnosis the author presents AX-Unet, a deep learning framework incorporating a modified atrous spatial pyramid pooling module to reduce information loss during down sampling and achieve information decoupling between channels. The proposed explicit boundary-aware loss function also tackles the blurry boundary problem. Segmentation of pancreatic ductal adenocarcinoma (PDAC) and surrounding vessels in CT images using deep convolutional neural networks and texture descriptors. To prevent information loss during down sampling and accomplish information decoupling between channels, this author introduces AX-Unet, a deep learning system integrating a modified Atrous spatial pyramid pooling module. The blurry border issue is addressed by the explicit boundary-aware loss function that has been proposed. With a DSC of 85.9±5.1% and a Jaccard of 77.9±3.4%, experimental data show that AX-Unet surpasses state-of-the-art techniques in pancreas CT image segmentation. Additionally, the model's extracted feature output reveals notable variations in the pancreatic area between healthy individuals and patients with pancreatic tumours, which can help doctors detect pancreatic tumors earlier. In [4] study is done on Artificial Intelligence-based Segmentation of Residual Tumor in Histopathology of Pancreatic Cancer after Neoadjuvant Treatment. In this study, digitized H&E-stained slides from resected pancreatic cancer after NAT were used to train a modified U-net model to segment tumour, normal ducts, and residual epithelium classes. The DenseNet161 encoder provided the highest mean segmentation accuracy, and a promising F1 score of 0.86 was attained for tumour segmentation. This work demonstrates that AI-based residual tumour burden assessment is viable; might be created as a tool for the impartial assessment of treatment response, and could be used to direct adjuvant treatment decisions. Experimental results demonstrate that AX-Unet outperforms state-of-the-art methods in pancreas CT image segmentation with a DSC of 85.9 ± 5.1% and a Jaccard of 77.9 ± 3.4%. Additionally, the extracted feature output of the model shows significant differences in the pancreatic region of normal people and patients with pancreatic tumors, which can assist physicians in the screening of pancreatic tumors. In [5] the authors of this research provide a brand-new self-learning architecture for training the PDAC segmentation model with a significantly bigger patient population and a mixture of annotated and unannotated venous or multi-phase CT images. Unannotated images are combined by two teacher models with various PDAC segmentation specialties to produce pseudo-annotations, which can then be improved by a teaching assistant model that recognizes related vessels around the pancreas. On both manually annotated and fictitiously annotated multi-phase images, a student model is subsequently trained. The findings demonstrate that the suggested approach offers a 6.3% Dice score absolute improvement over the robust baseline of nn-UNet trained on annotated pictures, obtaining a performance (Dice = 0.71), which is comparable to the inter-observer variability between radiologists.

In [6], the authors of this research suggest a brand-new, completely 3D cascaded framework for segmenting the pancreas in 3D CT scans. A 3D detection network (PancreasNet) that locates the pancreas regions and two distinct scales of a 3D segmentation network (SEVoxNet) that segment the pancreas in a cascading fashion depending on the detection findings of PancreasNet make up the framework's two primary parts. On the publicly available NIH pancreas segmentation dataset, the suggested method produces cutting-edge results with a mean Dice Similarity Coefficient (DSC) of 85.93% and a mean Jaccard Index (JI) of 75.38%. Deep neural networks' limitations in segmenting the pancreas were explored by Zhou et al. [7] due to its complex and changing backdrop regions. Using the anticipated segmentation map, they reduced the input region. We provide a fixed-point model that accepts the segmentation mask as input and output. On the NIH pancreas segmentation dataset, where they tested their model, they outperformed the state-of-the-art by more than 4% (in terms of DSC). A DSC of 82.37±5.68% on average was attained. In order to overcome the issue of spatial non-smoothness of inter-slice pancreatic segmentation, Cai et al. [8] presented a stacked CNN-RNN model. Convolutional long short-term memory (CLSTM) units make up the RNN sub-network. Deep supervision and multi-scale feature map aggregation were used to modify the 2D CNN sub-network. The investigations make use of the NIH-CT dataset and 79 abdomen T1-weighted MRI scans. In the CT dataset and the MRI dataset, they obtained DSC of 83.35.6% and 80.77.40%, respectively.

A pancreatic segmentation model based on 2020 First International Conference on Power, Control and Computing Technologies (ICPC2T) 402 bidirectional convolutional long short-term memory (BiCLSTM) networks and spatial context information (SCU-Net) was proposed by Hao Li et al. [9]. The CT abdominal scans were divided into multiple isometric blocks using a divide-and-conquer technique, and a multi-channel CNN was created to make use of spatial context information. To encourage the interaction between information flow from bidirectional sequences, the BiCLSTM network is introduced. For inter-slice constraint and regularisation, a new loss function was developed. DSC, Jaccard index, pixels-wise precision, and recall were the evaluation measures employed. They examined the NIH-CT dataset, which included 82 abdomen enhanced 3D CT scans, and found that the final prediction accuracy was 82.863%, the mean Jaccard Index was 68%, and the mean prediction error was 12% and mean recall of 80.2% and mean Precision of 82.2%. In [10] states the importance of huge number of training samples in the deep learning networks. Because of the considerable inter-patient structural changeability in both outline and size dimensions, the pancreas is a difficult
abdominal organ to section [16,17]. Convolutional neural networks (CNNs) give better performance [11, 12].

III. METHODOLOGY

A. U-Net Architecture

Two levels of the U-Net model are employed, consisting of downward-moving paths followed by upward-moving paths in U-shaped network architecture. Each block on the downward path is composed of two convolution layers with the Leaky ReLu activation function and a maximum pooling layer with stride 2. In the first block, we use 32 filters of size 3x3 for each of the two convolution layers. The journey upwards is paved with 2x2 transposed convolution and the joining of feature maps from the parallel downward path. Along this path, we encounter four blocks, each composed of two convolutional layers with 3x3 filters and a Leaky ReLU activation function. We chose the Leaky ReLU for its efficiency, swift convergence and sparse activation - when faced with negative input, it returns 0; for positive input, it returns the same value as shown in equation (1), where x stands for the input.

\[ f(x)=\max(0,x) \]  

Despite using the identical U-Net model at both levels, the hyperparameter values vary. The weights of the first U-Net (U-Net-1) are initialised using Kaiming Initialization, and the filter size is 3x3. This approach avoids the disappearing or bursting gradients problem brought on by exponentially growing or decreasing input signals by accounting for the nonlinearity of activation functions like ReLU. The equation below can be used to express each convolution layer's response.

\[ y_l = W_l x_l + b_l \]  

In this case, x is a k x c x 1 vector that represents k x k pixels that are co-located in c input channels. The layer's spatial filter size is k.

The number of connections in a response is \( n = k^2c \). The dimensions of the matrix W is a d x n, and each row of the matrix reflects the weights of a certain filter. In the output map, Y represents the response at a specific pixel, and b is a vector of biases. If \( l \) is the layer index and \( f \) is the activation, \( x_l = (y_{l-1}). \)

Given in below Eq. is the necessary condition to prevent the gradients issue.

\[ 0.5\sqrt{n} \text{Var}[W_l] = 1 \text{ for all } l \]  

Where \( W_l \) denotes the random component of each of the element in \( W_l \) and the \( \text{Var} \) depicts variance. This results in a start of the structure indicated in below equation.

\[ W_l \sim \mathcal{N}(0,2/n) \]  

R2 regularization is used in the U-Net-1 configuration to prevent overfitting. The penalty is basically increased by regularization as model complexity rises. All parameters, with the exception of the intercept, are penalized by the regularization term lambda (), making sure that the model properly generalizes the data and does not overfit. R2 regularization is useful when there are correlatively dependent characteristics. W denotes the weight matrix with values i and j.

There are two parts to it: an encoding route and a decoding path, where the encoder extracts high-level features from the input image and the decoder creates a segmentation mask using those features [18].

Elbow U-Net is the name of this specific version, which was created in Python using the TensorFlow library. It is made to segment binary images with the intention of locating areas of interest that are labelled with pixel values of 1 while the backdrop is labelled with 0. The input image dimensions, which are given as an input layer to the model, are IMG_HEIGHT x IMG_WIDTH x IMG_CHANNELS. After that, the input layer is normalized by using a lambda layer to multiply each pixel's value by 255. A succession of convolutional layers with 16, 32, 64, 128, and 256 filters are used in the encoding path. In order to avoid overfitting, a dropout layer is placed after each convolutional layer. In order to minimize the spatial dimensions of the feature maps, the output of each convolutional layer is then passed through a max pooling layer. Transpose convolutional layers, also referred to as deconvolutional layers, and are used in the decoding path to upsample the feature maps back to their initial size. In order to maintain spatial information, the decoder also has skip connections that concatenate the feature maps from the associated encoding path layer. The first convolutional layer in the encoder is a transpose layer with 128 filters, and the next two layers are each convolutional layers with 128 filters. Until the final layer, which comprises of a transpose convolutional layer with 16 filters, followed by two convolutional layers with 16 filters each, this is repeated with progressively smaller filter sizes. A segmentation mask with image values between 0 and 1 is produced by a convolutional layer with one filter and a sigmoid activation function in the output.

Table I provides a detailed illustration of the U-Net model's construction. An expansive path follows a contracting path in the five-level U-Net network. The parameters Batchnorm, activation functions, dropout are shown in detail.
The proposed architecture of a U-Net model for medical picture segmentation is depicted in the Table II. In medical image analysis, U-Net is a common deep learning architecture, especially in segmentation tasks where the objective is to label each pixel in an image with its appropriate class. A contracting path is found on the left side and an expansive path is found on the right side of the U-Net architecture. While the expansive path gradually upsamples the feature maps to generate a segmentation mask, the contracting path uses a sequence of convolutional and pooling layers to capture the context of the input image. The earliest stages of the contracting process, which minimises the spatial dimensions of the feature maps, are convolutional and max pooling layers. When this procedure is performed numerous times, the number of filters in each convolutional layer rises at each level. By using batch normalisation and activation algorithms, the model performs better. After obtaining the feature maps from the contracting path, the expansive path uses transposed convolutional layers to progressively increase their spatial dimensions. The feature maps from the contracting path are concatenated with the corresponding feature maps from the expanding path in order to keep the high-resolution data. The output layer is then activated with a softmax function to generate the probability distribution.

The last convolutional layer of the U-Net is activated using a softmax activation function to provide a probability map for each pixel, which is then thresholded to produce the final binary segmentation map.

The proposed framework is based on the application of two tiers of U-Net model as shown in the Fig. 1. The model has two sets of encoding and decoding blocks. This can help to improve the segmentation performance by capturing more complex features at different scales.

The outcomes of a pancreatic segmentation training method are displayed in Table III. The aim of the training is to correctly identify the pancreas in medical photographs using a machine learning system. When compared to the ground truth data, the algorithm's ability to predict the right pancreas segmentation is measured by the training loss. Better performance is indicated by a reduced training loss. How closely the anticipated segmentation matches the actual segmentation is determined by the dice coefficient. Better performance is indicated by a higher dice coefficient [14]. The training process starts with a relatively high training loss of 0.76 and a training dice coefficient of 0.82. As the training progresses, the algorithm gets better at identifying the pancreas, resulting in a decrease in training loss and an increase in the dice coefficient as shown in the Table II.
matches the actual segmentation is determined by the dice coefficient. Better performance is indicated by a higher dice coefficient. The training dice coefficient of 0.70 and the initial training loss of 1.45 both point to subpar performance in the first epoch. However, as training goes on, the algorithm improves at recognizing legions, which causes the training loss to go down and the dice coefficient to go up.

The algorithm's capacity to recognize legions has greatly increased by the time training is complete, as seen by the training loss of 1.35 and training dice coefficient of 0.83. With a score of 0.69, the validation dice coefficient likewise performs well.

### Table III. Training Outcomes

<table>
<thead>
<tr>
<th>Epoch</th>
<th>Time (hrs)</th>
<th>Training Loss</th>
<th>Training Dice</th>
<th>Validation Dice</th>
</tr>
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<tbody>
<tr>
<td>0</td>
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<td>1.44871</td>
<td>0.70470</td>
<td>0.6492</td>
</tr>
<tr>
<td>1</td>
<td>0.41683</td>
<td>1.40648</td>
<td>0.75877</td>
<td>0.6681</td>
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<td>1.39184</td>
<td>0.77948</td>
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</tr>
<tr>
<td>3</td>
<td>0.40694</td>
<td>1.38603</td>
<td>0.78834</td>
<td>0.66002</td>
</tr>
<tr>
<td>4</td>
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<td>1.38146</td>
<td>0.79132</td>
<td>0.62748</td>
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<tr>
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<td>0.71489</td>
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<tr>
<td>6</td>
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<td>0.80300</td>
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<tr>
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</tr>
<tr>
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<tr>
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<tr>
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<td>0.40503</td>
<td>1.35127</td>
<td>0.82449</td>
<td>0.69984</td>
</tr>
</tbody>
</table>

**Fig. 2. Tumor segmentation training.**
In Fig. 3 by showing a visual comparison of the pancreas segmentation results generated using the suggested method and the ground truth segmentation, the usefulness of the proposed method is proved. The visual comparison is shown in a three-column arrangement, with the abdominal CT images in the first column, the ground truth segmentation in the second, and the segmentation results from the proposed method in the third.

The visual comparison is used to assess how well the suggested strategy performs in precisely segmenting the pancreas from abdominal CT images. The effectiveness of the suggested method in precisely segmenting the pancreas may be assessed by contrasting the segmentation results achieved by the proposed method with the ground truth segmentation.

It is possible to compare the segmentation results obtained by the suggested approach with the ground truth segmentation quickly and effectively by using a three-column format that offers a clear and concise depiction of the segmentation findings. Additionally, using abdominal CT scans in the first column offers a relevant and realistic context for assessing how well the suggested strategy works in a clinical scenario.

Fig. 3 also offers a visual comparison of the PDAC segmentation results produced by the suggested method and the ground truth segmentation in order to assess the performance of the proposed method. The visual comparison is shown in a three-column arrangement, with the abdominal CT images in the first column, the ground truth segmentation in the second, and the segmentation results from the proposed method in the third. The abdominal CT images, which were obtained from various patients and utilized to test the suggested procedure, are shown in the first column. The manual ground truth segmentation, carried out by a skilled radiologist, is displayed in the second column. The performance assessment of the suggested method uses the ground truth segmentation as a benchmark.

The segmentation results produced by the suggested method are shown in the third column. The outcomes demonstrate that the suggested approach is successful in precisely segmenting PDAC tumors in abdominal CT images. Analysis of similarities and differences between the segmentation results produced by the suggested method and the actual segmentation is done.

Typically, the segmentation's results is discussed here. Various challenges provide the ground truth against which processes are validated. Various standards are used to analyze the performance and accuracy of segmentation. The most popular statistical measures are based on a number of research articles, and the different problems are described below.

Assume that the segmented region is represented by X and the ground truth is by Y. To assess the precision of the segmentation, we may utilize a variety of indicators. The Volumetric Overlap Error (VOE) is one such measure. This calculates how much of the divided region overlaps with the actual scene. It is determined by dividing the sum of the pixels in x and y intersection by the sum of the pixels in X and Y’s union. The result is then multiplied by 100 and removed from 1. Successful segmentation is indicated by a VOE value that is near to 0, whereas greater values signify disparities between the segmented pictures. The following is a formula for VOE.

\[
VOE = ((|X \cap Y| / |X \cup Y|) - 1) \times 100
\]
Here dice similarity coefficient (DSC) is an additional measure. This gauges how well the split region's pixels correspond to the actual scene. A DSC score around 1 suggests effective segmentation, whereas a number near 0 implies inconsistencies across the segmented pictures. The DSC formula is displayed below:

$$DSC = 2 \frac{|X \cap Y|}{|X| + |Y|}$$

The segmented region's volume difference from the ground truth is measured by the Relative Volume Difference (RVD). It is computed by dividing the sum of the volumes of X and Y, deducting 1, and then multiplying the result by 100. Positive RVD values signify over-segmentation, whilst negative RVD values signify under-segmentation. The RVD formula is displayed below:

$$RVD = \frac{(\text{total volume of } X - \text{total volume of } Y)}{\text{total volume of } Y} \times 100$$

The percentage of true positives (TP) that are accurately detected by the system, out of all real positive instances (TP + false negative (FN)), is known as recall, also known as sensitivity or true positive rate. In other words, recall gauges how well a system can identify all positive cases. A high recall score means that a significant number of positive cases may be recognized by the system.

The proportion of true positives that are successfully recognized by the system, out of all anticipated positive instances (TP + false positive (FP)), is measured by precision, also known as positive predictive value. To put it another way, precision assesses a system's capacity to accurately identify only pertinent positive cases. A high accuracy rating means that the system produces few false positives, which means that the instances that are projected to be positive are more likely to be accurate.

The validation set provided by NIH dataset has yielded the following results both for the segmentation of the pancreas and pancreas tumour.

The segmentation of Pancreas: dice for each case of 0.9283, dice global of 0.9300, VOE of 0.114, and RVD of -0.070.

The segmentation of PDAC: dice for each case of 0.5852, dice worldwide of 0.8900, VOE of 0.434, and RVD of -0.158.

VI. CONCLUSION

In this study, we show the simplicity and efficiency of the U-Net model for semantic biomedical picture segmentation. It is possible to successfully segment the pancreas and tumours from abdominal CT scans using the proposed architecture, which is based on a variant of the fundamental U-Net model. By changing the fundamental U-Net model's hyper parameter values in accordance with the type of dataset, we may attain accuracy that is comparable to that of sophisticated state-of-the-art techniques. The model's performance was assessed using the National Institutes of Health (NIH) dataset, and an 88.2% (Dice Global) pancreas tumour segmentation accuracy was attained. The proposed method can be applied to MRI images and we also aim to extend our research work by combining the U net model with other deep learning models to improve the accuracy.

REFERENCES


Personating GA Neural Fuzzy Hybrid System for Computing HD Probability

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Abstract—The cardiovascular disease (CD) is a widespread, dangerous sickness involving an excessive rate of demise that necessitates quick piousness for care and cure. There are numerous diagnostic methods, such as angiography, available to diagnose heart disease (HD). ML is an extremely leading option for scientists for discovering prediction-based explanations for heart disease, and several machine learning algorithms are discovered to find the leading key results in community assistance. Researchers are presented with numerous conventional approaches, and various supportive algorithmic sequences formulated through the artificial neural network (NN) family, such as adaptive, convolutional, and de-convolutional NN, and various extended versions of hybrid combinations, originate with suitable outcomes. This research integrated the design and computational analysis of a unified model through a genetic algorithm-based Neural Fuzzy Hybrid System, which is formulated for CD prediction. It included a dual hybrid model to forecast CD and measure the degree of a healthy heart, as well as more precise heart attack complications. Stage 1 of the study's implications integrates the two stages and plans HD prediction using patient data. The input was processed in stages. First, the data was delivered in pre-processing mode. Next, the mRMR algorithm was used to select features. Finally, the model was trained using a variety of ML algorithms, including SVM, KNN, NB, DT, RF, LR, and NN. The results were compared, and based on those findings, the model was tuned to produce the best results. In stage 2, HA possibilities and occurrences are determined by FuzIS intelligence using data from the first stage, which includes more than 13000 pre-generated rules of fuzzy implications. These rules cover both normal-level and dangerous-level cases, and the medical parameters are integrated and tuned to produce membership functions that are then sent to the model. It is composed with the comparison of a unified system, which consists of Genetic algorithms, Neural networks, and Fuzzy Inference systems. In the experiment, gaussian MF sketched the continuous series of data, enabling the inference system to generate a good accuracy of 94% in calculating the problem probability.

Keywords—Dickey-Fuller test case (DF-TC); HA prediction (HAP); heart rate variability (HRV); artificial based neural network (AbNN); Fuzzy Inference System (FuzIS); genetic-based algorithm (GbA); multi-objective evolutionary Fuzzy classifier (MOEFC); heart attack (HA); fuzzification-mode (FuzM); defuzzification-mode (De-FuzM)

I. INTRODUCTION

Cardiovascular-based disease (CD) is a widespread, dangerous sickness involving a great level of disease that necessitates quick devoutness aimed at care, medication, and cure. Angiography is one of numerous treatments available to diagnose heart disease (HD), but it is a very expensive and time-consuming process and therefore out of reach for the masses. To conquer this crisis, the best medication, an auto-altered intelligent system, is necessary to reduce root-level causes, and individuals can inhale extreme difficulty. Many scientists are involved in integrating artificial intelligence techniques, which are used to build and execute technology-based medical experimental scenarios using data mining (DM) and machine learning (ML) procedures to discover the finest results for such challenges [1]. The expansion of the artificially based neural network (AbNN) into an intimated and unified system has carried it to a distinct range, and with the progress of the selection of features along with additional supporting procedures, it has carried it to human-like forecast control.

NN is mainly structured in three layers, which are the input, hidden, and output layers. Through every layer, a neuron will be associated with other neurons in succeeding layers, with their weights calculated with the activation function and a constant value called bias added to each layer as shown in Fig. 1 [19]. The bias is a persistent value injected in the network to adjust the activation function in AbNN and improves in deciding the participation of neurons in the network based on the threshold; otherwise, the output is ignored. ReLut is principally associated in the hidden layer along with their associated values, which represent 0 to x, with an identical outcome if input is positive (+); otherwise, it produces zero as the outcome, as shown in Fig. 1 [21].

\[ h = \max(0, a), \text{where } a = Wx + b \]  \hspace{1cm} (1)

The sigmoid function with an output limited between 0 and 1 and an S-shaped graph is mostly used in classical calculus, where probability is to be predicted as an output. Logistic and
Softmax sigmoid functions are used for output related to binary and several levels of grade classification, respectively.
\[ f(x) = \frac{1}{1+e^{-x}} \] (2)

The Tanh is almost similar to the sigmoid, but the only difference is that its value lies between -1 and 1. Fig. 1 explains the difference between ReLU, sigmoid, and tanh functions [29–30].

A genetic-based Algorithm (GbA) returns a fitness value for each solution based on the phenomenon that the higher the fitness value, the better the solution [22–24]. There are mainly three operations in GA: selection, crossover, and mutation. Fuzzy inference system (FuzIS) primarily undergoes four stages after obtaining input: knowledge base, fuzzification mode (FuzM), inference engine, and defuzzification mode (De-FuzM) [24], which is the graphic symbol of the location of input-output neurons at each layer of AbNN. Below, in Fig. 2, there is the pictorial representation of ANFuzIS, which is structured into five layers: input, membership, FuzM, normalization, and De-FuzM. The inference system is built mostly with the Sugeno method. Artificial Neural FuzIS (ANFuzIS) combines ANN and fuzzy systems, which use the capabilities of ANN to generate fuzzy inference rules with applicable membership function implications [25]. The x and y are inputs that are passed through ANFuzIS layers starting with the membership layer as the first layer, the second layer dealing with FuzM, normalization on the third layer, and De-FuzM on the fourth and last layer, i.e., the fifth layer is the output layer as shown in Fig. 2.

This study used a genetic algorithm-based neural fuzzy hybrid system to integrate the design and computational analysis of a single model that was developed for CD prediction. It is made up of a comparison of an integrated system that uses fuzzy inference, neural networks, and genetic algorithms. In the experiment, Gaussian MF sketches of continuous data series allowed the inference system to produce a good accuracy of 94% when estimating the likelihood of a problem.

This article deals with the assessment and evaluation of the consequences of conventional processes and unified structures by employing distinct patterns like cross-validation with feature selection and discovering the choice of these accumulations in unified systems. Successive sections of this article have been composed with various supporting sections: Section II encompasses the related work of earlier developments of conventional AI, ML, and deep learning-based techniques. Section III contains the proposed methodology for genetic-based algorithm (GbA)-AbNN-based Fuzzy Inference System (FuzIS) Implications with Existing Dataset Parameters. It has described the methodological sequences in detail through Neural FuzIS (NFUZIS) and stage-wise execution of GbA-AbNN-based AbNN-FuzIS Implications. Section IV designated an experimental setup with System specification, Dataset, Relevance of medical parameters, and GANN algorithm with mRMR feature selection, along with experimental scenarios for determining the possibility of heart attack (HA) using Rule Creation Membership functions. Final: Section V contains details of inference.

II. RELATED WORK

Many scientists in earlier times did excellent exploration on discovering the finest possible answers for assessment of HD, benefiting from numerous conventional and progressing ML classification and regression algorithmic sequences. A few of them are Decision Trees (DT), Random Forest (RF), SVM (Support Vector), NB, LR, K-NN, ANN, DNN, GA, AGA, NFS, and many others. This list is regularly growing by generations and exhibits the attention of investigators from all spheres in the associated field. On the dataset, UCI Cleveland, Hungary, and Switzerland were favorites during liability trials and testing.

The authors, Shadman N. et al., offered a cloud-based ML scheme trained through WEKA, a Java-coded data mining tool for estimating HD [2]. The outcomes examined and the model (SVM) recorded SENS as 97.53, ACC as 97.53, and SPEC as 94.94 percentages. The authors, Amin Ul Haq et al., offered a model through DT, K-NN, RF, LR, SVM, ANN, and NB classification algorithms permitted with three selections of features such as LASSO, mRMR, and relief, which are validated by the K-Fold-concluded prevalent Cleveland HD dataset [3]. The authors, W. K. U. R. A. R.-S. T. V. H. P. P. Moloud Abdar, presented a N2Genetic-nuSVM algorithm for prediction of coronary artery disease (CAD) with better performance.
The authors RK. Jha et al. [5] stated an analysis to assess various categorization algorithmic sequences for estimation of HD where several conventional processes like SVM, KNN, DT-DNN, NB, and RF [17–18] were utilized to be valid selection of features over the Rapid Minor (RM) instrument to train-learn employing the Cleveland dataset from the UCI repository environment [28–33]. The authors, M. M. J. K. H. I. W. Mehrdad Agha Mohammadi [6], had proposed a unified system comprising ANFUZIS through GA and cross-validation of K-fold for estimation of HD and attained ACC as 84.43, SENS as 91.15, and SPEC as 79.16 percentage of results. The authors, M. P. K. at el., projected a hybrid adaptive genetic algorithm authorized by fuzzy logic with a rough set of features to predict HD [9] and attained ACC as 90, SENS as 91, and SPEC as 90 percent of results.

The authors, Fatma Zahra Abdeldjouad et al., projected ML training through hybrid methods: LR, MOEFC, FURIA, GFSLB, and Vote to Expect HD using Keel through Weka software [11]. Outcomes stats were analyzed, and it was concluded that out of all the other classification models, Vote outperformed ACC, SENS, and SPEC by 80.20%, 84.76%, and 74.82%, respectively. Indu Yekkala et al. projected RF, XGBoost, and NN-based models to be tuned by GA (as feature selection) to optimize (high ACC, low rate of errors) the HD estimation solution [12]. The Z-Alizadeh Sani dataset was used for training purposes. Python language with Jupyter Notebook IDE was selected for experimentation as well as for data mining and attained ACC of 93.85, SENS of 97, and SPEC of 92 percent of results. The authors, Farman Ali et al., projected a healthcare system containing collaborative DL and feature fusion for estimation of HD [13] and attained good results. The model showed an ACC of 98.5%, which is better than many other models. The authors, L.A. Demidova et al., presented an algorithm for self-tuning multi-objective GA through flexible parameter selection [14].

In this study, heart illness is initially detected using an artificial neural network (ANN) using default parameters. Then, to increase classification accuracy, a hybrid strategy combining an artificial neural network (ANN) and a genetic algorithm (GA) is suggested. Finally, the Cleveland dataset from the UCI machine learning repository is used to demonstrate the success of the suggested approach [10]. Another study combined four supporting data sets—from the VA Long Beach, Cleveland, Hungary, and Switzerland—with Multivariate dataset features [15] in order to identify the existence of cardiac disease in the patient. The output of a neural network, such as yes or no, is determined by activation functions based on neural networks. The obtained values are mapped between 0 and 1 or -1 and 1 [20]. In a different study, the author used membership functions to predict output type constant and linear value and used two sets that could predict all eight functions [26]. The membership functions were initially combined with an artificial neural fuzzy inference system. The researchers integrated multiple data mining algorithms used in health care analytics and also paid attention to how to systematically represent analytical data in order to produce results while employing diverse data mining algorithms [27]. This cutting-edge information can be protected via a variety of techniques, including embedding, concealing, and so forth [38].

The author suggested a technique for improving classification accuracy called k-modes clustering with Huang beginning. Models like the decision tree classifier (DT), multilayer perceptron (MP), random forest (RF), and XGBoost (XGB) are employed. To improve the outcome, GridSearchCV was used to fine-tune the model's parameters. A real-world dataset of 70,000 occurrences from Kaggle [39] is used to test the suggested approach. Another study used KNN and logistic regression, two machine learning algorithms, to predict and categorize patients with heart disease. The regulation of how the model can be utilized to increase the precision of heart attack prediction in any individual was done in a very helpful way. The proposed model's ability to predict signs of having a heart illness in a specific individual by utilizing KNN and Logistic Regression was quite gratifying, and it demonstrated a high degree of accuracy in contrast to previously employed classifiers such as naive bayses, etc. [40]. Another study found that the ML technique chosen and the attributes in the dataset used to train the model had a big impact on the model's performance. The Cleveland dataset was reduced to a lower-dimensional subspace using the Jellyfish optimization algorithm to prevent overfitting (caused by the curse of dimensionality) due to the excessive number of characteristics. The Jellyfish method is versatile in finding the best features and has a high convergence speed [41].

The author developed a potential machine learning model using a variety of feature selection strategies to uncover important features in the early stages of predicting heart disease. Chi-square, ANOVA, and mutual information were three separate methods used for feature selection, and the chosen feature subsets were designated as SF1, SF2, and SF3, respectively [42]. A soft voting ensemble classifier developed by another author that used all six algorithms improved accuracy even further, yielding 93.44% accuracy for the Cleveland dataset and 95% accuracy for the IEEE Dataport dataset. On both datasets, this outperformed the logistic regression and AdaBoost classifiers [43]. Another author has combined nine machine learning classifiers into the final dataset for the prediction of heart disease, including random forest (RF), XGBoost (XGB), decision trees (CART), support vector machine (SVM), multinomial Nave Bayes (MNB), logistic regression (LR), linear discriminant analysis (LDA), AdaBoost classifier (AB), and extra trees classifier (ET) [44]. This assumption is strengthened by the esteemed machine learning algorithms performances for both classification and regression complexity [45]. To detect the presence of cardiac anomalies, the author employed ML approaches [46]. In a different study, the author starts by providing succinct
summaries of the machine learning classification approaches that are most frequently employed to identify cardiac disorders [47].

III. PROPOSED METHODOLOGY: GENETIC-BASED ALGORITHM GbA-AbNN-BASED FUZIS IMPLICATIONS WITH EXISTING DATASET PARAMETERS

Various criteria were investigated to discover the greatest attempt, pick out the finest features, and train-learn the model to obtain the highest score of accuracy. The feature selection (FS) algorithmic sequences employed to stream features for data tuning, and apart from this, validation methods like cross-validation have been used to validate the model and pick the most advantageous set of features. In the testing process, the data was deployed to assess the level of prediction accuracy.

A. Neural Fuzzy Inference System (NFuzIS)

The AbNN with FuzIS operated separately. The unified NFUZIS utilize process to discover all factors from FIS [25]. NFUZIS can correspond to exercise data produced from n-measurements of functionalities. NFUZIS comprises the fault figuring segment to advance the learning-training directions while the faults been unhurried, primarily membership sequences demarcated, then membership arrangements constraints stimulated, and inexact inputs transformed into neuron-inputs as A(i1-in) as shown in Fig. 3, then fuzzy-type of inputs such as fuzzy sets are generated to form neuron-inputs translated for further process [16-18] [28-33]. The experimental scenario has been described in Fig. 3. B. Stage Wise Execution of GbA-AbNN based AbNN- FuzIS Implications

Fig. 3 determines the architecture of the proposed examination exertion, which is further split into two stages: stage 1 and stage 2, wherein stage 1 clarifies HD estimation and stage 2 describes the method to find the HA possibilities. This research reveals the unified model, which comprises AbNN motorized by GA and integrates the intellect of FuzIS. It has been explained in detail in the below sub-sections.

Stage 1: In stage 1, HD prediction has been planned through the patient’s data. It processed the input through various stages: firstly, it was delivered through the data pre-processing mode; then, the mRMR algorithm was implicated for the selection of features; then, the model was trained using diverse ML algorithms such as SVM, KNN, NB, DT, RF, LR, and NN; results were compared; and based on that, the model was tuned to get the optimal result. In the subsequent stage, an AbNN designed for training persistence produces weights through GbA. The proposed method was trained, and on each cycle of occurrence, it generated new offspring using parental classes with the help of crossover and mutation logics. The outcome is forwarded as an input to the second stage.

![Fig. 3. Proposed NFuzIS-based methodology for HD detection.](image-url)
Stage 2: In this stage, HA possibilities and occurrences calculated through FuzIS intelligence take the input from the first stage, where more than 13000 pre-generated rules of fuzzy implications are fed into the system, which covers equally the cases such as normal-level and dangerous-level. The medical parameters are integrated and tuned to generate membership functions passed to the model as shown in Fig. 4.

In the next step, Sugeno FuzIS integrated for singleton production membership sequences with linear or constant functionalities of inputs incorporated, then the input undertaken through the FuzM process wherein data has been converted into fuzzy values with the help of fuzzy sets which then in the next step, with function \( z_i = a_i x + b_i y + c_i \). Below pseudo code explains the workflow for the proposed system.

**NFuzIS based Algorithm**

1. Add Input for experiment
2. Apply data-processing, project the model, and add inputs, hidden along with output neurons.
3. Selection of features  mRMR
4. Load primary population set with indiscriminate homogeneous chromosomes input samples.
5. Calculate matrix weight for population utilizing GbA and initiated model training steps
7. Replication 4 and 5 steps, record the scores.
8. Accomplish model testing with test data  {decision HD-patient / non-HD-patient}
9. Produce inference rules through precedence and medical complication based on dataset.
10. medical parameter  membership functions  fuzzy set.
11. Pass input through the FuzM process.
12. Process the fuzzy production to De-FuzM progression to acquire output crisps.
13. Test model to measure efficiency.

IV. EXPERIMENTAL SCENARIOS

The experimental setup is divided into two parts: in the first stage, the dataset has been instructed in a unified environment with GbA and ANN; the same steps were applied here as well; mRMR algorithmic sequences are simulated for the selection of accurate features; and in the second stage, FIS has been used for predicting HD.

A. System Specification

In this research, the experimental scenarios were formulated using Python libraries with the integration of more than 13000 inference rules.

B. Dataset

Cleveland, Hungarian, and VA-Long Beach datasets from the UCI container, which hold 76 features, are used in the experiment. Out of which, Cleveland's having 303 records is very popular [24]. 16 features are designated as inputs and 1 is a goal feature out of the 76 available features. Based on this input, we created one more feature named “HasHeartDisease” and filled in data 0 and 1, depending on “num” feature values.

The considered key parameters: patient (P) age, P-Gender, Level of Heart rate (LHR), balanced cholesterol, Patient Family Heart Disease History, chest pain type (CPT), Resting Blood Pressure (RBP), Level of ECG result (RestECG), Heart Status (HS), Fluoroscopy vessels count based on coloured (CA), Cigarette per day, Blood sugar level at fasting (FBS), Stress or Depression (OldPeak), Diagnosed HD (Num), Exercise induced Angina (Exang), Resting Heart Rate, and Slope for Peak-Exercise (Slope). The patients’ health indicators demonstrating trial delivery between nutritious and risk levels of patients, such as patients deprived of heart disease (healthy) are 53.87% and patients with heart disease are 46.13%. The dataset is characterized into 0-4 states, where 0 represents healthy and 1-4 represents risk levels of patients, considering 1 is the least risky and 4 is the most. Fig. 5 shows the ratio of healthy patients to patients at risk. Statistics states that the dataset is normalized and has the same ratio between both; it also indicates that the dataset is ideal for experimentation.
C. Relevance of Medical Parameters

In the above sections, we have already listed the parameters that are being used in the experiments; now let’s discuss the relevance of a few of the parameters, like why they are important in predicting HD or in determining HD-affecting possibilities. The medical parameters with description and threshold to be used to calculate HD-probability are shown in Table I.

- **Age:** People of old age (above 65) or adults older than 40 are more likely to suffer from HA as compared to younger people or to develop coronary artery disease.
- **Gender:** According to available resources, the incidence of CVD in women is usually lower than that of men.
- **Heart rate:** An increase or decrease in heart rate should be alarming, and preventive measures should be taken.
- **Blood pressure:** This reflects problems in the heart and can damage the heart if the number is high for a long time.
- **Blood sugar:** A diabetic person is more likely to have HA. If a person has a high level of sugar, he can damage his heart too.
- **ECG result:** The spike created in the ECG can tell a lot about heart health, whether a person is healthy or not.
- **Cholesterol:** If a patient has high cholesterol, he can invite fat to deposit in the arteries and cause HA due to less delivery of blood oxygen to the heart and other parts.

<table>
<thead>
<tr>
<th>Feature Name</th>
<th>Description</th>
<th>Threshold</th>
</tr>
</thead>
<tbody>
<tr>
<td>RestingBP_RBP</td>
<td>BP measured at fasting at the hospital</td>
<td>Systolic/diastolic (in mm Hg)</td>
</tr>
<tr>
<td></td>
<td>Normal: &lt; 120/80</td>
<td>Abnormal: 120-140/80-120</td>
</tr>
<tr>
<td></td>
<td>Critical: 140/120</td>
<td></td>
</tr>
<tr>
<td>SerumCholesterol</td>
<td>Total Cholesterol:</td>
<td>Normal: &lt; 200, Moderate: 200 - 239, High: 239+</td>
</tr>
<tr>
<td><em>SCH</em></td>
<td>HDL level:</td>
<td>Low: 40 - 60</td>
</tr>
<tr>
<td></td>
<td></td>
<td>Normal: &lt; 130, Moderate: 130 - 189, Critical: 189 &gt;</td>
</tr>
<tr>
<td>FastingBloodSugar_FBS</td>
<td>Blood sugar level at fasting</td>
<td>0 for Normal, 1 for High</td>
</tr>
<tr>
<td>RestingECGResult</td>
<td>ECG result</td>
<td>0 for Normal, 1 and 2 are other types</td>
</tr>
<tr>
<td><em>RES</em></td>
<td></td>
<td></td>
</tr>
<tr>
<td>RestingHeartRate_RHR</td>
<td>Heart rate at resting</td>
<td>0-65 - Normal, 65-85 - Average, Above 85 high,</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>MaxHeartRate_MHR</td>
<td>Max heart rate</td>
<td>100 - Normal, 100-140 - Moderate, 100-180 - Critical</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>CigarettePerDay_CPD</td>
<td>Cigarette intake per day</td>
<td>Higher is the intake, more is the failure risk</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>HeartDiseaseFamilyHistory</td>
<td>Heart disease history in family</td>
<td>0 for No, 1 for Yes</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>IsHeartPatient</td>
<td>Is patient a heart patient</td>
<td>0 for No, 1 for Yes</td>
</tr>
</tbody>
</table>

V. RESULTS AND DISCUSSION

This section included the experimental test cases and results and discussed the integration of the trained model into a web service to expose it to the public domain. In this chapter, techniques for API exposure are discussed along with the deployment process so that it can be made available for public use.

Preliminary phase: First generation prepared with an arbitrarily produced population with supporting parameters set. The network chromosome was structured as 416 neurons, 2 hidden layers, and 1 output layer with ReLU and Sigmoid activation methods and gradient descent optimization. Evaluation phase: Here, each chromosome generated in the preliminary stage is trained, and accuracy is evaluated to find a fitness score. Selection phase: The system will sort according to the fitness values, and the one with the finest fitness is selected and others are rejected. Cross-over phase: Selected parents, chromosomes A and B, undergo mating to produce new offspring. Mutation phase: After cross-over, a few of the genes are mutated on a random basis to maintain diversity in the population.

A. Experimental Scenarios for Finding the Probability of HA

To cover up the second stage, an experiment has been carried out using FuzIS, wherein input from the neural system has undergone the FuzM and De-FuzM processes via the inference engine along with the fuzzy rules and membership function to get the desired output. The very first step after injecting input into the experiment is to create rules that should be used for the rest of the experiment. For creating rules, all the included features have been analyzed, and based on that, we set the priority of the features based on the impact they can have on the heart, like sleeping heart rate, cigarette per day, blood sugar, and HD history of the family. Later, once we had everything on our plate, we followed a few approaches to create rules, keeping in mind that they should be powerful enough to support the system and should be able to provide capability to the system for finding HA possibilities in person.

\[
f(x; \sigma, c) = e^{-\frac{(x-c)^2}{2\sigma^2}} \quad (3)
\]

A function of membership enables the presentation of sets of fuzzy values and groups with the help of graphs. In the experiments, triangular, trapezoidal, and Gaussian MFs were used. A function of membership has been formed for a separately fuzzy set created based on all supportive, reliable, and non-dependable medical parameters. The Gaussian function: MF parameters \( \sigma \) vector \([c, \sigma, \sigma] \), \( \sigma \) standard deviation and \( \sigma \) mean. The 13824 total rules have been produced and integrated to identify the HA levels and their consequences from normal levels to risk levels, and gaussian MF sketched the continuous series of data, enabling the inference system to generate a good accuracy of 94% in calculating problem probability. Below, Table II shows the parameter setup applied for GANN model training. Different combinations have been experimented with, and based on the optimal result, we noted down the parameters for the model shown.
The very first step after injecting input into the experiment is to create rules that should be used for the rest of the experiment. For creating rules, all the included features have been analyzed, and based on that feature’s priority, it is set based on the impact it can have on the heart, like “resting heart rate, cigarette per day, blood sugar, heart disease history of family,” and later, once we had everything on our plate, we followed a few approaches to create rules, keeping in mind that they should be powerful enough to support the system and should be able to provide capability to the system for finding the probability of a heart attack in person. The number of antecedents could be more than one based on the if-else condition, but the consequent is always one. Rule creation for the experiment is composed of two approaches:

The Cleveland dataset has been based, and after data processing, we found 282 good and 21 bad data points out of 303. A total of 126 rules were generated based on the "num" feature, which defines the risk level as 1, 2, 3, or 4. An initial study has been done and defined the level of contribution of different features, like max heart rate, which can be proved to be top-risky with the reason that if the number goes higher, it will impact the heart a lot. Therefore, the contribution of the maximum heart rate was classified into four levels: normal, medium, high, and critical. Similarly, blood sugar was also found to be a contributing factor, but its level did not have an immediate impact on health. Hence, its threshold was set to normal or medium. Other features were also evaluated, and their contribution levels were determined to assess whether a person is at risk of a heart attack or not.

A set of 13824 rules has been generated to cover both normal and risky cases. A few of the inferences are listed in Table II for reference. GANN-based performance analyzed with the 2000 epoch. It can be noticed that at the initial stage, the model was in the learning phase, and the result was not so effective, but as it underwent the epoch again and again, new offspring were generated, which helped in the learning process. This also reflects in the curve plotted in the below graph. The GANN-based training inferences and fitness information are included in Table II. It can be clearly seen that the learning rate was very slow and the model accuracy was also very low, but gradually, after 600 epochs, the model started giving better results.

<table>
<thead>
<tr>
<th>Hyperparameter</th>
<th>Values</th>
</tr>
</thead>
<tbody>
<tr>
<td>Hidden layer used</td>
<td>2</td>
</tr>
<tr>
<td>Neurons in each hidden layer</td>
<td>8, 4</td>
</tr>
<tr>
<td>Activation method for Hidden Layer</td>
<td>ReLu</td>
</tr>
<tr>
<td>Activation Layer for Output Layer</td>
<td>Sigmoid</td>
</tr>
<tr>
<td>Learning-rate</td>
<td>0.001</td>
</tr>
<tr>
<td>Loss Function</td>
<td>Binary Cross-Entropy</td>
</tr>
<tr>
<td>Epoch</td>
<td>2000</td>
</tr>
<tr>
<td>Batch size</td>
<td>100</td>
</tr>
<tr>
<td>Mutation</td>
<td>3%</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Hyperparameter</th>
<th>Values</th>
</tr>
</thead>
<tbody>
<tr>
<td>Age</td>
<td>100</td>
</tr>
<tr>
<td>Sex</td>
<td>1</td>
</tr>
<tr>
<td>ChestPainType</td>
<td>1</td>
</tr>
<tr>
<td>RestingBP</td>
<td>140</td>
</tr>
<tr>
<td>SerumCholesterol</td>
<td>233</td>
</tr>
<tr>
<td>FastingBloodSugar</td>
<td>1</td>
</tr>
<tr>
<td>RestingECGResult</td>
<td>1</td>
</tr>
<tr>
<td>ExerciseinclInclAngina</td>
<td>0</td>
</tr>
<tr>
<td>OldPeak</td>
<td>2.3</td>
</tr>
<tr>
<td>PeakExerciseSegment</td>
<td>3</td>
</tr>
<tr>
<td>VCA</td>
<td>0</td>
</tr>
<tr>
<td>ThalliumScan</td>
<td>6</td>
</tr>
<tr>
<td>RestingHeartRate</td>
<td>70</td>
</tr>
<tr>
<td>CPD</td>
<td>5</td>
</tr>
<tr>
<td>HDL</td>
<td>0</td>
</tr>
</tbody>
</table>

B. Test Case Analysis through API

Once the model has been trained properly and is available for prediction, the next step, which is more important, is to find a way to package the model so that it can be available for further use in applications. And the best way is to build a web service, integrate a trained model into it, and expose it via an end point. We have moved one step ahead in the research work and utilized a very popular Python library called FastAPI for this purpose. The whole process has been divided into several stages, like creating a packaged model, unpackaging the packaged model, creating a web service, exposing an end point, and starting the web service for establishing the final connection. Let’s discuss all these stages one by one.

1) Packaging trained model: This is the first stage at which we usually package the model. Our final model is already prepared and is able to predict the disease as well as find the probability of a heart attack, so the next step is to pass this model for packaging into a system-readable bundle. For this, a very popular Python library named "joblib" has been used, which should package the model by storing all the required details in it. Once the model is packaged properly, it has been saved as a ".pkl" file.

2) Loading/Unpackaging packaged model: Once the model is packaged, before integrating it into a web service, it is required to load this model again in the system. For this, we have used the same Joblib Python library.

3) Creating a web service: Now that the model is loaded successfully into the system, it is time to create a web service, or API, and expose the endpoints. For this purpose, we used the "FastAPI" library, which is capable of performing this task.

4) Starting the environment: Now that there is a packaged model that can be reloaded into the system, an endpoint is also created for exposure purposes, so the last point is to start the environment and expose the API to the public. As the research work is local, the library that we are using will start the environment locally on the localhost 4000 port, and there the endpoints can be tested. The Python library that has been used is "uvicorn".

Now that the API is ready and up on localhost, the system is ready for testing (see below Fig. 6 and Fig. 7 show the request details for this endpoint).

```python
{
  "Age": 60,
  "Sex": 1,
  "ChestPainType": 1,
  "RestingBP": 140,
  "SerumCholesterol": 233,
  "FastingBloodSugar": 1,
  "RestingECGResult": 1,
  "ExerciseinclInclAngina": 0,
  "OldPeak": 2.3,
  "PeakExerciseSegment": 3,
  "VCA": 0,
  "ThalliumScan": 6,
  "RestingHeartRate": 70,
  "CPD": 5,
  "HDl": 0
}
```

Fig. 6. Screen showing the API definition loaded on interface.
Total of 16 medical parameters have been passed as request to the endpoint as shown in figure 8. Below is the response for the same. In Fig. 7 and Fig. 8, response can be noticed for the given request wherein it is printed in two parts 1) predict, which predict the heart disease in patient and 2) risk percentage, which shows the probability percentage of heart attack. Here person is predicted as having heart disease and probability percentage is 57.81 which means that person has 57% chances of heart attack with current health condition which is based on give medical parameters.

C. Test Cases

Below Table III shows few test cases along with the response using created API. It shows the request and response tested over API.

### TABLE IV. API Test Cases Showing Request and Response

<table>
<thead>
<tr>
<th>Request</th>
<th>Response</th>
</tr>
</thead>
<tbody>
<tr>
<td>{</td>
<td></td>
</tr>
<tr>
<td>&quot;Age&quot;: 60,</td>
<td>&quot;predict&quot;: &quot;1&quot;,</td>
</tr>
<tr>
<td>&quot;Sex&quot;: 1,</td>
<td>&quot;risk_percentage&quot;:</td>
</tr>
<tr>
<td>&quot;ChestPainType&quot;: 1,</td>
<td>57.81226836834753</td>
</tr>
<tr>
<td>&quot;RestingBP&quot;: 145,</td>
<td></td>
</tr>
<tr>
<td>&quot;SerumCholesterol&quot;: 233,</td>
<td></td>
</tr>
<tr>
<td>&quot;FastingBloodSuger&quot;: 1,</td>
<td></td>
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<tr>
<td>&quot;RestingECGResult&quot;: 2,</td>
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<tr>
<td>&quot;MaxHeartRate&quot;: 150,</td>
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<td>&quot;ExerciseIncludedAngina&quot;: 0,</td>
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<tr>
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<tr>
<td>&quot;ThalliumScan&quot;: 6,</td>
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<tr>
<td>&quot;RestingHeartRate&quot;: 70,</td>
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<tr>
<td>&quot;CPD&quot;: 5,</td>
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<td>&quot;HDFH&quot;: 0</td>
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<tr>
<td>}</td>
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<td>{</td>
<td></td>
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<tr>
<td>&quot;ChestPainType&quot;: 1,</td>
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<td>&quot;RestingBP&quot;: 90,</td>
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<td>&quot;MaxHeartRate&quot;: 120,</td>
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</tbody>
</table>

Below, Table IV and Fig. 9 show the result from GANFIS along with the comparison with similar work proposed in the past by other researchers. The result clearly shows that the proposed system outperforms with 94% accuracy.
TABLE V. COMPARING EXPERIMENTAL RESULTS WITH EXISTING STATE-OF-THE-ART RESULTS IN LITERATURE

<table>
<thead>
<tr>
<th>Author and Ref</th>
<th>Integrated model</th>
<th>Method</th>
<th>Dataset</th>
<th>Result</th>
</tr>
</thead>
<tbody>
<tr>
<td>Negar Ziasabouchi, 2014 [34]</td>
<td>ANFIS Based Classification Model for Heart Disease Prediction</td>
<td>ANFIS, GA</td>
<td>Cleveland dataset</td>
<td>92.30%</td>
</tr>
<tr>
<td>Oluwarotimi Williams Samuel et al. 2017 [7]</td>
<td>An integrated decision support system based on ANN and Fuzzy_AHP for heart failure risk prediction</td>
<td>ANN, Fuzzy_AHP</td>
<td>Cleveland dataset</td>
<td>91.10%</td>
</tr>
<tr>
<td>A.V. Senthil Kumar 2012 [35]</td>
<td>Diagnosis of Heart Disease using Fuzzy Resolution Mechanism</td>
<td>ANFIS, MATLAB</td>
<td>Cleveland dataset</td>
<td>91.83%</td>
</tr>
<tr>
<td>Kaan and Ahmet 2017 [36]</td>
<td>Diagnosis of heart disease using genetic algorithm based trained recurrent fuzzy neural networks</td>
<td>ANN-Fuzzy_AHP, GARFNN</td>
<td>Cleveland dataset</td>
<td>91.10%</td>
</tr>
<tr>
<td>G. S. G. Thippa Reddy et al. 2020 [9]</td>
<td>Hybrid genetic algorithm and a fuzzy logic classifier for heart disease diagnosis</td>
<td>GA, FL</td>
<td>Cleveland dataset</td>
<td>90%</td>
</tr>
<tr>
<td>MAbushariah et al. 2014 [37]</td>
<td>Automatic Heart Disease Diagnosis System Based on Artificial Neural Network (ANN) and Adaptive Neuro-Fuzzy Inference Systems (ANFIS) Approaches</td>
<td>MLP, ANN, ANFIS, MATLAB</td>
<td>Cleveland dataset</td>
<td>87%</td>
</tr>
<tr>
<td>Proposed work, 2022</td>
<td>Dual Hybrid System (GANN-NFHS)</td>
<td>Cleveland dataset</td>
<td></td>
<td>94%</td>
</tr>
</tbody>
</table>

The results in Section V demonstrated the scores based on different variance comparable model exercises with and without selection of features through mRMR on GANN. Results clearly demonstrate variance between various variances applied during model training, and they verify that performance prediction is enhanced by applying dissimilar methods according to dataset size and number of features. It has been observed that good results have been achieved with classical as well as hybrid approaches, and near 90% accuracy was recorded with the test set. There is more opportunity to record a higher score with hybrid as well as classical approaches. An integrated system that makes use of fuzzy inference, neural networks, and genetic algorithms is compared. In the experiment, the inference system produced a good accuracy of 94% when assessing the likelihood of a problem using Gaussian MF sketches of continuous data series.

VI. CONCLUSION

ML is an extremely leading option for scientists for discovering prediction-based explanations for heart disease, and several machine learning algorithms are discovered to find the leading key results in community assistance. Researchers are presented with numerous conventional approaches, and various supportive algorithmic sequences formulated through the artificial neural network (NN) family, such as adaptive, convolutional, and de-convolutional NN, and various extended versions of hybrid combinations, originate with suitable outcomes. This research integrated the design and computational analysis of a unified model through a genetic algorithm-based Neural Fuzzy Hybrid System, which is formulated for CD prediction. It is composed with the comparison of a unified system, which consists of Genetic algorithms, Neural networks, and Fuzzy Inference systems. In the experiment, Gaussian MF sketched the continuous series of data, enabling the inference system to generate a good accuracy of 94% in calculating the problem probability.

AUTHORS’ CONTRIBUTION


CONFLICTS OF INTEREST
The authors declare no conflict of interest.

FUNDING
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REFERENCES


DeepCyberDetect: Hybrid AI for Counterfeit Currency Detection with GAN-CNN-RNN using African Buffalo Optimization

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Abstract—Modern technology has made a big contribution to the distribution of counterfeit money and the valuation of it. This paper recommends a deep learning-based methodology for currency recognition in order to extract attributes and identify money values: machine learning's binary classification task of fake currency detection. One can train a model that can distinguish between real and fake banknotes if one has enough information about actual and fake notes. The vast majority of older systems relied on hardware and techniques for image processing. Using such strategies renders identifying fake currency more challenging and inefficient. The proposed system has suggested deploying a deep convolution neural network to figure out fake currency in order to solve the aforementioned issue. By analyzing the images of the currency, our technique finds counterfeit notes. The transfer-learned convolutional neural network is trained using data sets that represent 2000 different currency notes in order to learn the unique characteristics map of the currencies. After becoming familiar with the feature map, the network is capable of real-time phoney cash detection. It is surprising how well deep learning models perform in photo classification tasks. The Deep CNN model that has been created in the proposed approach helps in the detection of the fake note without really manually extracting the properties of photographs. The model trains from the data set produced during training, letting us to identify fake currency. In multiple instances, techniques for deep learning have been shown to be more effective. Thus, deep learning is used to boost currency recognition accuracy. Among the techniques used are the African Buffalo Optimization Approach (ABO), recurrent neural networks (RNN), convolutional neural networks, generative adversarial networks (GAN) for identifying bogus notes, and classical neural networks.

Keywords—Fake currency; convolutional neural network; generative adversarial networks; recurrent neural network; African Buffalo Optimization

I. INTRODUCTION

With the rapid advancement of modern technology, the proliferation of counterfeit money poses a significant challenge for financial systems worldwide. Detecting and distinguishing fake currency from genuine notes is crucial for maintaining economic stability and trust in monetary transactions. This paper presents a deep learning-based methodology for currency recognition, aiming to extract attributes and accurately identify the value of money.

For the monetary exchange of some types of goods, several nations employ various types of currencies. One problem with cash that many countries face is the existence of counterfeit currency in the system. One of the countries with numerous problems and large losses as a result of the fake money is India. The entire economy of the nation experiences losses, as does the value of its currency. Technology advancements have made it possible for currencies to be duplicated to the point that it is difficult to distinguish between them. Modern editing tools and cutting-edge printers are used to create counterfeit money. False currency can easily be bundled with legitimate currency, as is the case in the majority of the world. In order to increase public awareness of the security issues with bank notes, the State Bank of Pakistan employs a variety of different marketing strategies, both directly to its consumers and in partnership with other institutions. Through a television advertisement and an application for smartphones that checks for counterfeit currency, the bank hopes to educate the public. But most individuals can't tell the distinction between a phoney and real currency note, especially those who are illiterate. Some people may be able to spot forged currency because they are not understanding the security elements of currency in public. Additionally, it might be challenging to identify these characteristics when the money notes are tattered, dirty, and ripped [1].

Money that has been produced without the State's or government's legal consent is known as counterfeit money. Its main purpose is to imitate real money and trick the person it is intended for. The production or use of counterfeit money is seen as a kind of fraud or forgery and is therefore illegal. Early on after gaining its independence, Bangladesh took on the monetary policy legacy left by Pakistan's Central Bank. The initial state of affairs was unstable, but the government soon established Bangladesh Bank in 1972 to stabilize the situation. Following that, Bangladesh has experienced a sharp rise in the availability of counterfeit currency. The suggested methods produce topic-based shards with minimal size variance and high densities of pertinent documents. They are
scalable, efficient, and self-sufficient. Information gathering for the initial stage of the model is aided by the suggested approach. The CNN, a multilayer network of neurons, is the most widely used deep learning technique in the next stage. Since it has increased the accuracy of many machine learning tasks and is effective at handling picture classification and identification issues [2], [3]. The deep CNN model used in the proposed approach is trained on a diverse dataset comprising 2000 different currency notes, enabling it to learn unique characteristics and feature maps of various currencies. Once familiar with the feature map, the model is capable of real-time detection of counterfeit currency, providing a highly efficient and automated solution [4].

The development of automated methods and systems for recognizing currencies has accelerated daily. Effective currency identification systems and cost-effective solutions are crucial in a variety of settings, including the banking system, train ticket windows, retail malls, and currency exchange services, among others. For financial transactions, banknotes are utilized. The prevalence of bogus currency on the international market has dramatically increased. In many industries, fake money is a problem. The country's financial market is negatively impacted by the development and design of these bogus notes, which come in all denominations [5]. As a result of advances in current technology and gadget development, counterfeit banknote production has increased due to the introduction of scanners and copy machines. It is very challenging for the human eye to tell phoney notes from genuine ones since they are expertly designed to look similar. The security of banknotes must be considered, and security components must be incorporated, to reduce counterfeit money. As a result, a system that determines whether a note is real or fake needs to be put in place in banks and at ATMs for withdrawal and deposit. However, people are still able to create these fictitious currencies [6].

Deep learning models have demonstrated exceptional performance in photo classification tasks, and the deep CNN model developed in this study eliminates the need for manual feature extraction. By training on the provided dataset, the model gains the ability to accurately identify fake currency notes. Around 180 distinct currencies are used in transaction worldwide. Each note has distinct safeguards and an array of sizes. Because each note has distinct security features and a different size, it is simple to tell whether currency is from which country and how much it costs. Currency notes fraud has now become a noticeable problem in all nations, and the quantity of fake notes is increasing daily. The world faces a difficult issue in identifying fake currency due to the similar look of genuine and counterfeit currency. As a result, data augmentation methods including image improvement, color analysis, and others are used in currency detection systems. Deep learning methods, which often involve multilayer neural networks, have proven successful in a variety of fields. When huge data is accessible, they have done especially well. Accurate money recognition has a lot of potential to improve with deep learning. The transfer learned CNN is trained using two thousand currency note data sets and is given the feature map of various currencies. The network has been trained to identify counterfeit banknotes in real time after learning the feature map [7].

The ABO technique, used in feature selection, aims to create an algorithm that is easy to use, reliable, efficient, and effective, yet has amazing skill in the exploitation and exploration of the search space. ABO makes sure that each buffalo's location is regularly updated in comparison to both its best prior location and the present location of the best buffalo in the herd in an effort to combat the problem of early convergence or stagnation. The entire herd may be reset, for instance, if the location of the dominant (best) buffalo is not altered after several occurrences. Finding the best buffalo guarantees that the search area has been adequately explored, and the ABO can achieve adequate exploitation by drawing on both the experience of other buffalos and the best buffalo. There are some limitations to the CNN-developed money-detecting system. The ABO simulates the three distinctive behaviors that enable the African buffalos to find pastures. Their tremendous memory capacity comes first. This makes it possible for the buffalos to trace their movements as they travel thousands of kilometers over the African terrain. Additionally, buffalos are quite helpful. The second characteristic of the buffalo is their cooperation and communicative abilities, which they exhibit in both good and bad circumstances. They are practically the only animal breed that will put their own life in peril to save the life of a member of their own species. The security features can only be recognized using the front image of the currency. The dataset is the most important element, and it must be accurate and of the greatest standard. The image quality has a significant impact on how well the classification works out. Additionally, each image in the created dataset needs to have its background clearly visible to avoid confusing the model with objects and noise. Collecting data is therefore essential [8]. Overall, this paper aims to present a comprehensive and effective deep learning-based system for counterfeit currency detection and classification. By harnessing the power of deep learning and incorporating innovative optimization techniques, the proposed approach exhibits promising potential for improving currency recognition accuracy and combatting the issue of counterfeit money distribution.

The key contribution of proposed approach is,

- Generative Adversarial Networks (GAN), which can create new samples that are comparable to the original target data after being trained to learn its distribution, are used to create the data.
- Convolutional Neural Networks (CNN) are used to extract the features since they automatically produce the features from time series data and frequency representation images.
- African Buffalo Optimization (ABO) is used to choose the characteristics, which reduces computation complexity and boosts efficiency.
- Recurrent Neural Network (RNN) is used in the phase of classification which finally decides the whether the currency is fake or not.
II. RELATED WORKS

The identification of counterfeit money using deep learning was suggested by Shilpa et al. [9]. Two phases make up the system that is being suggested here. Classifying currency notes according to denomination is the first step. Checking the note’s authenticity is the second step. Here, information has been taken from Kaggle, a well-known source of datasets. With the aim of spotting counterfeit money on devices like smart phones, tablets, and PCs, the recommended system is designed using a deep learning methodology, and a Neural CNN model is developed. On a set of data that was independently produced, the developed model was trained and tested. CNN receives images that have been collected by a smart camera. There are two primary phases to this web-based strategy. The first step is to group the notes into different categories based on their worth, and the second is to determine whether or not they are real or fake. However, this strategy did not result in more favorable and successful outcomes. Saxena et al. suggested utilizing image processing to find counterfeit money. The image processing technique, which comprises changing a picture’s fundamental properties to improve its visual information for human interpretation, is applied using MATLAB. The feature of the image processing programme increases the possibilities of the MATLAB numeric computing environment. The algorithm used in this method is image scaling, which resizes the input image to 100 dpi in order to provide a better KNN for classification. By doing this, the false positives are eliminated without obviously hurting how well our software works. The image is then converted from the RGB to the grayscale domain via grayscale computation. Here, a cell phone with a scanner or camera is used to use a MATLAB technique in order to identify fraudulent currency. However, it is crucial for researchers to change Matlab’s features in order to achieve its high performance measures [10].

Machine learning algorithms were proposed by Shahani et al. to assess banknote authentication. Back propagation neural networks (BPN) and support vector machines (SVM) are examples of supervised learning techniques that are accustomed to distinguish actual banknotes from counterfeit ones. The dataset used to train the models was provided via the UCI machine learning repository. A ratio of 80:20 is used to divide the used dataset into its two parts. The smaller group is used to test whether the models can correctly predict whether a letter is authentic, while the bigger fraction is used to train the models. As a result, BPN can distinguish between genuine and counterfeit notes with accuracy. SVM also has trouble identifying real notes from false ones. By classifying the notes as Genuine, Low-Quality forgery, High-Quality forgery, and Inappropriate ROI, this study can be expanded [11]. Deep learning was suggested by Pachon et al. to identify fraudulent banknotes. This method compares the two design approaches using illustrations of various building types. The transfer learning technique identifies the optimal freezing sites in the sequential, residual, and inception CNN architectures. A sequential CNN-based original model similar to AlexNet is also suggested. Modern solutions have been presented for various CNN architectural kinds. On a dataset of Colombian banknotes, the TL and the custom models were trained, and then they were assessed. The results showed that ResNet18 had the highest accuracy. The key limitations of a customized model are its protracted training procedure and need for an appropriately diverse training sample [12].

Convolutional traces were recommended by L. Guarnera et al. [13] for identifying counterfeit currency. The suggested technique has a set of unique local characteristics designed specially to clarify the underlying neuron formation procedure using an Expectation Maximization (EM) algorithm. For spontaneous validation to confirm non-fakes, experimental tests using naive classifiers on five independent architectures have been used. For noisy photos, the computation is labor-intensive. Deep learning was suggested by Cruz et al. to identify counterfeit Indian banknotes. To aid in the training and testing of the deep network DCNN model, a fake data set was created utilizing the various image sets. VGG-16 In the context of the Single Shot Multi Box Detector (SSD) model, CNN is utilized as a feature extractor. For the purpose of object detection, SSD is a frame model. SSD using mappings of features from various layers, create bounding boxes. It has been trained with Tenserflow. A convolutional neural network will be trained using the suggested algorithm on the provided false and original currency data set, and it will then be able to determine if a particular currency image is fake or original. However, since the base VGG-16 network takes up 80% of the time, the computation takes a lengthy time [14].

Laavanya et al. proposed the detection of fake currency using deep learning. This method focuses on identifying bogus currency that is pervasive in the Indian market. The most common deep neural network technique, referred to as transfer learning utilizing Alex net, is used to detect counterfeit money. Alex net includes fully-connected layers, dropout, ReLU activations, max pooling, and convolutions. The data store of images of money notes is created with the intention of training the network. For each note, 100 images are created with the help of augmentation. Techniques for augmentation, such as rotating and resizing, are used to increase the amount of data bases. A camera is used to evaluate the procedure in real time. A "Real Note" or "Fake Note" result is generated by comparing the input currency note's properties to those that the network has already learned. This comparison occurs as soon as the picture has been acquired. Since the monetary distinctive attributes are gradually learned, the detection accuracy is at its highest. The acquired image may also contain noise, which must be taken into account as part of the pre-processing step in the currency detection process. By taking into account the surface patterns of the coin as characteristics, the detection accuracy of phoney currency can also be increased [6].

Deep learning was suggested by Gebremeskel et al. to identify Ethiopian banknotes. The image capture, image size regularization, grayscale conversion, and histogram equalization steps of the detection model sequence combine to significantly reduce the parameter number counts in the convolutional region of the DL structure. The Fully connected (FC) layer, which is basically in charge of evaluating the incoming image, executes the detection process. The deep CNN suggestion process organizes the input images into numerous groups based on the characteristics discovered at prior levels. Additionally, the input photos are introduced after
the input classification process has converted the input pictures into the distribution of probabilities over classes and after the neural network has been trained via back propagation. The model delivers the following parameters throughout all relevant epochs: recall, precision, F1 score, confusion matrix, training accuracy and loss, validation accuracy and loss, recall, and accuracy and loss plot graphs. There are only 10 epochs in the pre-trained CNN model, thereby lowering its effectiveness in feature extraction [15].

Gopanae et al. proposed the use of Support Vector Machine (SVM) to identify fake bank notes. The suggested strategy is focused on exploiting open access to identify counterfeit money. A banknote’s legality can be determined using the suggested system by examining specific security features like watermarks, latent images, security threads, etc. Algorithms for machine learning are used to detect fake currency. As part of the technique, certain security components are extracted and encoded. A support vector machine (SVM) is employed to extract security features, categorize features, and find features from the input image. The proposed system uses methods such image acquisition, region of interest extraction from the image, feature extraction, and machine learning algorithms to determine whether an input image of a banknote is authentic or fraudulent. However, a comprehensive verification of the proposed approach requires extensive evaluation in a wide range of real-world scenarios. Furthermore, deep learning techniques can be used on a substantial amount of training data to provide predictions that are more accurate [16].

III. PROBLEM STATEMENT

India and many other nations have suffered greatly and the issue has become grave. Fake notes are a catastrophe in almost every country. Systematic methods are utilized to distinguish between real and fake bank notes using bank note authentication. Nowadays, a wide variety of applications are accessible, including automated teller machines, self-serve checkout lanes, money exchange companies, hotels, banks, and retail establishments, among others. Forged bank notes are becoming more prevalent over time. In spite of the ease with which counterfeit bank notes may now be produced because of advances in scanning and printing technology, it is now incredibly challenging to distinguish between real and fake currency. However, current systems sound implausible to people like shopkeepers because of their price, requirement for a large power supply, and size. As a result, these people are more likely to encounter forgers. It is required to categorize and identify between photos of real and fake bank notes throughout training [17]. As a result, numerous conventional methods, such as KNN and Decision Trees, were employed; however, advanced machine learning algorithms, such as ELM, XGBOOST, or MLP, were not used. Applying either an image or a video as an input feed, the issue can be resolved by applying deep learning algorithms to recognize the currency.

IV. PROPOSED COUNTERFEIT CURRENCY DETECTION USING GAN-CNN-RNN WITH ABO APPROACH

A. Data Generation using GAN

The dataset should include a diverse collection of real banknotes and corresponding counterfeit samples. The GAN will be trained on this dataset to learn the underlying distribution of genuine banknote features and generate synthetic counterfeit currency images that closely resemble real counterfeit notes, ultimately enhancing the performance of the counterfeit currency detection algorithm [18].

The GAN, which has acquired prominence for their outstanding synthetic data production capabilities, serve as the foundation for the data generation. Two networks are trained simultaneously by a GAN. By examining the distribution of underlying data, the first network, known as the Generator network, frequently produces intimation images. In order to determine if an input sample is real or fake (i.e., whether it came from the generator), the second network, the discriminator, is used. Our suggested solution, which is based on GANs, models the underlying data distribution to differentiate between seven target categories with a very low inter-class variation.

The generator, \( K(z;W_g) \), is a network of neurons with parameters such as \( W_g \) and an earlier distribution of \( p_z(z) \) on the input random vector \( z \). A convolutional neural network's discriminator \( E(x;W_d) \) decides whether the input vector \( x \) comes from the generator's dispersion \( p_g \). A \((n + 1)\) dimensional vector illustrating this probability is the result. In order increase the likelihood that it will correctly recognize its input, \( x \), the discriminator, \( E \), tries to distinguish between images created artificially by specimen \( p_g \) and those acquired from the authentic data samples during training. Additionally, by attempting to minimize \( \log(1 - D(G(z))) \), the generator, \( K \), is trained to deceive the discriminator. By streamlining the procedure, you may do both.

![GAN network](https://www.ijacsa.thesai.org)

Fig. 1. GAN network.
In Fig. 1, the generator, $H(y;W_h)$, is a network of neurons with the values $W_h$ and a prior distribution of $p_z(y)$ on the input noise vector $y$. An $(n + 1)$ dimensional vector is the output of the convolutional neural network that serves as the discriminator, $I(x;W_l)$.

$$
\min_h \max_l V(H,I) = E_{x \sim \text{data}_{(x)}}[\log(x)] + E_{y \sim q_y(y)}[\log(1 - I(H(y)))]
$$

(1)

In order to increase the possibility that the correct label will be assigned to its input $y$, the discriminator, $I$, attempts to distinguish between images acquired from the genuine data samples during training and those intentionally made by sampling $y$. The generator $H$ is also taught to deceive the discriminator by seeking to reduce $\log(1 - I(H(y)))$.

Our major objective is to be able to distinguish between the numerous classes that the photos comprise, even though the generator and discriminator are merely designed to discern actual images apart from false ones. Our key objective is to be able to discern between the many classes that the photographs represent. To determine if a photograph is authentic or fake, the $(n + 1)$ th unit in the last layer of $I$ applies the aforementioned loss function. This layer's remaining $n$ units are trained using the following common cross-entropy loss function:

$$
L_{\text{supervised}} = -\sum_{x} m(l(x)) \log(n(l(x)))
$$

(2)

Here, the discriminator projected class probability is represented by the symbol $m(l(x))$, which also serves as the input $x$'s accurate class label. The discriminator network can be trained simultaneously using two loss functions—one for classifying inside actual images and another for distinguishing between fake and real images—thanks to the introduction of a second label for the false class [19].

**B. Feature Extraction using CNN**

The feature extraction component's goal is to extract the elements of the paper banknote sample's descriptive features. In this instance, a CNN is employed to extract features. Some of the unique features that CNN frequently employs to identify banknotes include color, size, form, and tactile aspects. To extract the CNN feature, a supervised variation of the CNN model is used. Input, hidden, and output layers are present in the CNN. The layer of convolution and the subsampling processes layer are both referred to as the feature extractor layer in the CNN model's hidden layer. The convolutional layers pick up on a variety of local traits that make up the image of a birr note, such as the identifying mark, security thread, tin and wide golden strip, and numerous other security features. As the image moves through each layer, the filters may spot more complex characteristics. Each neuron's input in the convolution layer is coupled to the local receptive field of the layer above it in order to extract the local feature [20]. Rectifier Unit, the most frequently used activation function for the outputs of the CNN neurons, is referred to as ReLu [21].

The ReLU can be written as in eqn. (3),

$$
\sigma = \max(\sigma, v)
$$

(3)

Here $v$ is the input.

Typically, CNN retrieves high-level characteristics from the model's final output layer. It is also crucial to remember that grayscale and RGB images are typically examined by CNN models utilizing two-dimensional (2-D) convolution filters. The input picture is only N1 by N1 pixels, and the first 2-D convolution layer has n1 2-D convolution kernels with a size of p1 by p1. Because of this, the initial 2-D convolution layer will produce n1 feature maps that are (N1 - p1 + 1) by (N1 - p1 + 1) in size, depending on the size of the 2-D convolution kernels. The dot product of the weight matrix and the location of the local area (x, y) is also used to construct each feature map; and you can use the formula to determine the value of a neuron's $V_{ij}^{ab}$ at position (a, b) on the jth feature map in the ith layer.

$$
V_{ij}^{ab} = \sigma(c_{ij} + \sum_{p} \sum_{n=0}^{p} \sum_{q=0}^{q} \omega_{jn}^{p} V_{i-1,j+n}^{(a+p)(b+q)})
$$

(4)

Where, $\sigma(\cdot)$ indicates the ith layer's activation function, $c_{ij}$ is an additive bias of the jth layer's jth feature map, $Ki$ and $Pi$ symbolizes the height and width of the 2-D convolution kernel, respectively, while $m$ indexes the association between the feature map in the (i+1)th layer and the present (jth) feature map, and $w_{jn}^{p}$ is a weight for input $V_{i,(a+p)(b+q)}^{(a+p)(b+q)}$ with an offset of (q, p) in 2-D convolution kernel.

The 2-D pooling approach is used to minimize the feature maps' resolution.

In Fig. 2, the features of 2-D CNN architecture is composed of three layers: an input layer, a 2-D CNN block (stacked with many Conv layers and POOL layers), and a fully connected neural network (FNN) block. Feature maps' resolution is decreased using the pooling procedure, which begins after the convolution stage. The features can then become location-invariant. In the first POOL layer, where $n3 = n2/k2$, there are m2 x n3 x 1 nodes and m2 kernels of size k2 x 1. In addition, there is a link between the neurons of the POOL layer and the n 1 patch of the Conv layer. Applying equation 5 below completes the pooling step of the max-pooling technique.

$$
d_j = \max_{n,1}(d_i^{k x k} u(k,1))
$$

(5)

The largest value in the neighborhood, and $u(k, 1)$ specifies a window function matching to the small n 1 patch of the Conv layer. Richer and more robust features can be extracted using a features extraction method that can quickly and efficiently coordinate the training of subnetworks [22].
C. Feature Selection using African Buffalo Optimization (ABO)

The African Buffalo Optimization (ABO) mimics the alert ('maaa') and alarm ('waaa') behaviors of African buffalos when they are out hunting. When searching for food and avoiding predators, African buffaloes primarily use these two noises to coordinate their movements. The waaa noise is used to rouse the buffalos to move on and investigate the search area, whereas the maaa noise instructs the buffalos to remain and make use of their immediate surroundings because it is secure and has enough of pastures. The buffaloes are aided in their food source seek by these noises.

1) The working of the ABO algorithm: The method starts by purposely assigning each buffalo a random location inside the N-dimensional space in order to initialize the population of animals. The algorithm then determines the $b_{n\text{max}}$ (the herd's ideal position) and $b_{m\text{max}}$ (each individual buffalo's location) with respect to the ideal solution. The position vector for that specific buffalo is saved if the person's current fitness level exceeds their maximal fitness level ($b_{m\text{max}}$). The fitness is saved if it exceeds the herd's maximum, or $b_{n\text{max}}$, which is the entire fitness of the herd. The position of the current buffalo is updated, and the population's future buffalo are taken into account. The ABO algorithm is shown below, same to algorithm 1.

2) Controlling the movement of the buffalos: According to the communal intellect of the herd after engaging in a collective decision-making process, this democratic equation (10) in figure allows for the choice to either continue in order to further utilize the environment or to go on to explore other location. Depending on the outcome of the democratic eqn. (6), the maaa eqn. (5) advises the buffalos to move on and explore different areas after carefully examining the two conflicting pressures ($b_{n\text{max}}$ and $b_{m\text{max}}$). The option, which identifies the discrete time interval that the buffalo must migrate through, has a default value of 1.0. When these two equations are applied, the position of the buffaloes is altered. To realize the full potential of the algorithm, it is required to investigate the ability of the African Buffalo Optimization to search a range of solution spaces, including separable and non-separable, limited and unconstrained, mono-modal and multi-modal. One extreme is present in a mono modal function, which is defined as having only one minimum or maximum within the range specified for x. In a manner similar to this, it is said to be multimodal if a function contains more than one peak, either on the minimum or minimum sides.

In addition, a function is considered to be dissociable if it can be written as the sum of $p$ functions with just one variable. It is harder to optimize non-separable functions. This is due to the fact that accurate search directions need the presence of two or more components in the search space (or solution vector). The issue is complicated when a function has many modes. The last challenge that optimization algorithms
must overcome is the situation of many dimensions. This is because as issue dimensions increase, so do the number of local optima. These multimodal benchmark functions test an algorithm’s resistance to local minima or maxima. An algorithm will stack in a local minimum if its exploration capacity is inadequate. In this section, we'll examine a few of the flat-surfaced functions. Because these functions don’t provide enough data to enhance the search, they pose certain challenges for algorithms [23].

The parameters used are N, α, lq1, lq2

A large group of n buffaloes, where each buffalo stands for a potential solution.

- The buffaloes’ index k, where k is in the range of [1,...,N].
- The alpha value, excluding the zero, with a domain in [1,1][1,1].
- The learning factors lq1 and lq2 with a [0,1][0,1]real number domain.
- The exploitation man oeuvre is represented by the mk variable.
- The bmmaxk variable has the highest fitness for the herd.
- The bmmaxk variable represents each buffalo k’s optimal finding location.
- The wk variable, which represents the exploration move, is the last.

\[ m_{k+1} = m_k + lq1(bmmax - w_k) + lq2(bmmax - w_k) \]  

(6)

On Line 6, Eqn. (5) indicates the location updated by the buffalo k [24].

\[ w_{k+1} = \frac{w_k + w_m}{\pm \alpha} \]  

(7)

The location update formula is Since Eq. (7) is a straightforward random number generator, the search inside the search space is likely to be aimless, leading to a relatively inefficient solution or, in certain cases, premature convergence. These issues, nevertheless, might be resolved by utilizing chaotic and levy distribution features.

Algorithm 1: ABO algorithm.

Input: N, α, lq1, lq2

Output: Original or fake currency

Set the parameters initialized
Make random answers for N buffaloes.

While the criterion of the terms has not ended do
  for all buffaloes do
    update the buffaloes mk
    update the location wk
  If the problem is minimization then
    if fitness mk < fitness bmmaxk then
      update the bmmaxk
  end
  if fitness mk > fitness bmmaxk then
    update the bmmaxk
  end
end

ABO uses just a few parameters, principally the learning parameters lq1 and lq2, to guarantee quick convergence. Depending on the algorithm’s concentration at a given iteration, these parameters allow the animals to move towards greater exploitation or exploration.

D. Classification using RNN

The classification phase employs RNN. An RNN connects its nodes in a way that closely resembles how neurons in the human brain are connected to one another. After processing the signal, it just received, the artificial neuron sends it to the other neurons or nodes that are connected to it. Weights frequently exist in neurons and synapses to change how learning occurs. The strength of the signal can be changed by varying the weight as it passes from the input layers to the output layers. An RNN has additional layers in addition to the input and output layers. Three hidden layers are the absolute minimum for an RNN. Input, output, and hidden units, all of which use weighted calculations, make up the fundamental elements of RNNs [25].

In Fig. 3 a sequence of input vector is given as \( A = (a_1, a_2,..., a_T) \).

The sequence of hidden vector as \( B = (b_1, b_2, ... , b_T) \) and

Sequence of output vector as \( C = (c_1, c_2, ... , c_T) \) with \( t = 1 \) to \( T \) as follows:

\[ b_t = \sigma(V_{ab}a_t + V_{bb}b_{t-1} + d_b) \]  

(8)

\[ c_t = V_{bc}b_t + d_c \]  

(9)

\( d \) is a bias element, function is an exponential activation function, and \( V \) is a weight matrix. The output of the hidden layer at each t-time steps is denoted by the symbol \( b_n \), while the output of the hidden layer before it is denoted by \( b_{t-1} \) in Eqn. (8).

All of the units in the hidden are replaced by LSTM, the RNN's memory unit. The regulatory gates open, activating the memory cells. These gates regulate the flow of information coming in and leaving out. Between an input gate and an output gate is positioned a forget-gate. If the originally saved details are no longer needed, forget gates are able to recover the linear unit’s state. These gates are composed of uncomplicated sigmoid threshold units. Memory block layers for these activation functions fluctuate between 0 and 1. At least one memory cell has to be encountered by each memory block.
The output $u^b k(t)$ of a memory cell is computed as

$$u^b k(t) = w^\text{out} k(t) b(s_{D_t}(t))$$  \hspace{1cm} (10)

Where, $u^b k(t)$ is the activation of output gate, $s_{D_t}$ is the output gate’s internal state, and $b$ is the output hidden layer.

In cases when natural language processing is problematic, the amount of text that is present in each data instance is not given as a fixed value. In order to comprehend every word of text, the sequence dimensions are scaled down to a certain degree. If the sequence size is lower than the desired value, the sequence is filled until the value is reached. The surplus is eliminated if the length of the sequence exceeds the necessary value. In Fig. 4, the Convolutional LSTMs are capable of simultaneously encoding long-term relationships and extracting characteristics from the time-frequency domain. The convolution procedure is carried out using linear activation in the first stage. The second phase, also known as the rectifying step, is when the activation function used in the first phase is identified and converted into a linear function. By applying a pooling function to reduce the dimensionality of the time series data, the network’s training period is shortened. The collected feature map must also be downsampled without the depth being altered. For the maximum pooling operation of the pooling layer, which is launched on the maximum pooling window, it is customary to choose the maximum values of the convolution layer. This differs from the convolutional layer parameter. Fig. 5 shows the comprehensive flow diagram for counterfeit currency detection and classification. Alternative pooling methods, such as minimal and average pooling, are, however, often used.
V. RESULT AND DISCUSSION

The process is implemented in MATLAB software in windows 10 platform. The generating the data from GAN is given as input for feature extraction by CNN, feature selection is carried out by the ABO and classification is by RNN. The experiment assessed the models using four evaluation metrics: accuracy, F1-score, precision, and recall. Fig. 6 shows the a) The original images of currency notes used for training the GAN-CNN-RNN model and b) Generated Fake Images - The synthetic counterfeit currency images generated by the GAN-CNN-RNN model during the training process.

GAN generates the images from the dataset. The elapsed time, iteration time and epoch time are plotted in the graph as shown in Fig 7. An ROC curve demonstrates how better the classification model performs at each level of categorization, seen in Fig. 8. On this curve, two parameters are plotted: 100% True Positive and False Positive Rate. A confusion matrix serves as a visual representation and summary of a classification algorithm's results. In Fig. 9, the proposed algorithm's confusion matrix is provided. Fig. 10 shows a concise representation of the training process for the counterfeit currency detection and classification model, utilizing GAN-CNN-RNN with African Buffalo Optimization.

![Generated Images](image1)

(a) The original images of currency notes used for training the GAN-CNN-RNN model and (b) Generated fake images - The synthetic counterfeit currency images generated by the GAN-CNN-RNN model during the training process.

![Epoch: 1000, Iteration: 1000, Elapsed: 00:18:44](image2)

Fig. 7. Generated images using GAN: Synthetic currency samples created by the GAN during training.

![ROC Curve](image3)

Fig. 8. ROC curve.
Fig. 9. Confusion matrix.

Fig. 10. A concise representation of the training process for the counterfeit currency detection and classification model, utilizing GAN-CNN-RNN with African Buffalo Optimization.

Fig. 11. Comparison using ABO optimization.
In the above Fig. 11, by using ABO optimization, the Quality of the original image is always higher than the optimized image that shows the image quality comparison between image indexes Vs SSIM.

A. Performance Metrics

These parameters are specifically defined as:

\[
\text{Accuracy} = \frac{TP+TN}{TP+TN+FP+FN} \tag{11}
\]

\[
\text{Precision} = \frac{TP}{TP+FP} \tag{12}
\]

\[
\text{Recall} = \frac{TP}{TP+FN} \tag{13}
\]

\[
\text{F1score} = \frac{2 \times \text{Recall} \times \text{Precision}}{\text{Recall} + \text{Precision}} \tag{14}
\]

\[
\text{Specificity} = \frac{TN}{TN+FP} \tag{15}
\]

The amount of data that were correctly declared as positive out of all the actually positive data is referred to as the TP. TN indicates to the proportion of data that were incorrectly classified as negative out of all the data that were actually negative. The term "FN" indicates to the number of variables that the model misclassified as negative even though they were positive in the dataset. The false positive rate, or FP, is the number of variables that the model incorrectly identified as positive even though they were in fact negative in the dataset. Recall is the proportion of positive data classified as such in the dataset as compared to the number of positive data identified as such by the model. In terms of the total quantity of data that were classified as positive, precision is the portion of data that the model correctly recognized as positive. Simply said, the harmonic average of recall and precision is the F1-score focusing on Accurate Predictions with Specificity.

![Fig. 12. Experimental result analysis.](image)

In Fig. 12, the proposed system’s performance metrics is tabulated. The overall accuracy for the process ranges between 0.8 and 1. The value of fake and genuine notes for precision, recall, F1 Score, Specificity is given above.

Comparison of the proposed model performance results with the existing methods is mentioned in Table I. For clear understanding comparative analysis of sensitivity and specificity are graphically represented in Fig. 13.

<table>
<thead>
<tr>
<th>Method</th>
<th>Specificity</th>
<th>Sensitivity</th>
</tr>
</thead>
<tbody>
<tr>
<td>KNN</td>
<td>94.75</td>
<td>89</td>
</tr>
<tr>
<td>SVM</td>
<td>94</td>
<td>91.42</td>
</tr>
<tr>
<td>SSD model-CNN</td>
<td>92</td>
<td>95</td>
</tr>
<tr>
<td>Proposed buffalo optimization approach</td>
<td>98.92</td>
<td>96.8</td>
</tr>
</tbody>
</table>

![Fig. 13. Comparison of sensitivity and Specificity for existing and proposed methods.](image)
VI. CONCLUSION

This method suggests using of deep learning to detect counterfeit money. The main benefit of employing a deep learning approach is the ability to automatically extract task-specific features from the data; in this case, the choice to utilize deep learning is affected by the graph-structured nature of the data. This eliminates the need for "handcrafted" features. Our model exhibits extremely high accuracy and stable behavior in a variety of difficult situations involving massive amounts of real data, underscoring the enormous promise of hybrid deep learning techniques for fake cash identification. Future research is still needed on many fascinating phenomena and concepts. On varied banknote picture quality, the effectiveness of the GAN, RNN, ABO, and CNN feature extraction technique was assessed. The respective recognition accuracy was 99.2%. Based on recognition precision the CNN is the best feature extraction method in terms of the model's overall goal. As a result, the CNN model is the best option for classifying the banknote, and the RNN also favors the best data creation, ABO optimization for selection, and RNN for classification. The given model significantly advances the study of identifying fake cash. The end research findings are satisfactory and include identifying the currency range in the categorization label, currency denomination, and currency front and rear. It is also possible to state that the accuracy of currency recognition is extremely high. Following study, we discovered that our recognition is quick and precise when the currency is in a clear state over the full screen and the angles are parallel.

REFERENCES

Prediction of Cardiac Arrest by the Hybrid Approach of Soft Computing and Machine Learning

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Abstract—Cardiac-related diseases are the major reason for the increased mortality rate. The early predictions of cardiac diseases like ventricular fibrillation (VF) are always challenging for doctors and data analysts. Early prediction of these diseases can save millions live. If the symptoms of these diseases are predicted early, the chance of survival increases significantly. For the prediction of Ventricular fibrillation (VF), several researchers have used Heart Rate Variability Analysis (HRV); various alternatives by combining the features taken from several areas to explore the prediction outcome. Several techniques like spectral analysis, Rough Set Theory (RST), Support Vector Machine (SVM), and Adaboost techniques have not required any pre-processing. In this work, randomly medical-related data sets are taken from various parts of Odisha, applying regression and Rough Set techniques, reducing the dimension of the data set. Application of Rough Set Theory (RST) on the data set is not only useful in dimension reduction but also gives a set of various alternatives. This work’s last section uses a comparative analysis between AdaBoost combined with RST and Empirical mode decomposition (EMD).

Keywords—Ventricular fibrillation (VF); heart rate variability (HRV); Rough Set Theory (RST); support vector machine (SVM); regression analysis; Adaboost method

I. INTRODUCTION

Cardiac-related problems are the most dreadful diseases concerning the death rate. According to World Health Organization (WHO) [1], nearly 18 million people died from cardiac-related problems. The most common cardiac-related diseases are cardiac arrest and sudden cardiac death (SCD). These diseases were responsible for causing nearly 50% of total death caused due to cardiac anomalies. In general, sudden cardiac death occurs due to Cardiovascular problems [2], and this problem doesn't have any prior symptoms of cardiac problems [3,4]. The most general sense of symptoms a person feels just before one hour of the actual occurrences of SCD. A person may not have any symptoms of fatal cardiac conditions but is still prone to SCD [5]. SCD rank second according to mortality rate after cancer [6]. The cause of SCD can be due to various diseases like Ventricular Tachyarrhythmias (VTA), Ventricular Tachycardia (VT), Ventricular Fibrillation (VF), Brady arrhythmia (BA), coronary artery diseases (CAD), valvular diseases (RV), and various genetic factors as discussed by Murukesan et al. [7]. There are several instances where SCD occurs due to ventricular tachycarhythmias, including both VF and VT, as Shen et al. discussed [8]; according to their study, ventricular tachyarrhythmias diseases responsible for the improper function of the heart. In most SCD cases, VF action is like a detonator [9-11]. The fatal rate increases by 90% every minute after the VF is detected. So to increase the survival rate, significant emphasis is given to the early detection of VF has notable importance. Forecasting SCD is always significant, as any wrong move can cause this issue to become more severe. There are several instances and examples where SCD is diagnosed wrongly, leading to fatal consequences. Public Access Defibrillation (PAD) method helpful in recovering from sensors can help in saving 2 to 15 lives after the collapse. The success rate of cardiac renewal is the fundamental treatment of SCD which stimulates the heart [12]. It would be ideal to avoid the start of SCD by offering medical support to someone who suffered a cardiac abnormality occurrence, which highlights the question of whether it would be feasible to set up warning devices that could identify cardiac arrest 30 minutes before a crisis as discussed by Fang et al. [13].

II. BACKGROUND

Several studies have been made with reference to various serious diseases, designing well-organized techniques of forecasting the SCD concluded intrusive and non-intrusive methods as discussed with reference to the article [14–16]. The core objective is to expect the SCD earlier than the ECG signals, as ECG is one of the most important physiological signals to identify cardiac abnormality and electrical conductivity features. Contemporary studies have investigated with properties of both ECG and heart rate variability (HRV), an indication which gets from ECG helps detect the delicate variation that occurs inside the signals earlier than that of the actual occurrence of SCD and to detect the high SCD risk. There are several additional features like time, frequency, and time-frequency. Several researchers have tried to predict SCD by using Machine Learning Algorithm (MLA). Recent studies can detect SCD 25 minutes before its actual occurrence; according to the report [17-18], the researcher used MLA to predict SCD. Several studies were being carried out for perfect diagnosis and early prediction of SCD, but all these data worked in the context of training data, i.e. when the dataset had been properly processed. Several research gaps exist in predicting SCD by various conventional and non-conventional.
methods. The following Table I represents the significant research gaps in predicting SCD.

**TABLE I. RESEARCH GAPS USING THE CONVENTIONAL METHOD**

<table>
<thead>
<tr>
<th>SL</th>
<th>Conventional research</th>
<th>Drawbacks</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>Soft Computing</td>
<td>Does not work for Random Dataset</td>
</tr>
<tr>
<td>2</td>
<td>Linear Regression</td>
<td>Does not work for Outliers</td>
</tr>
<tr>
<td>3</td>
<td>Machine Learning tools</td>
<td>Does not work for test data</td>
</tr>
<tr>
<td>4</td>
<td>Conventional studies</td>
<td>Machine Error</td>
</tr>
<tr>
<td>5</td>
<td>Mathematical Modelling</td>
<td>Only a Numerical solution is possible</td>
</tr>
</tbody>
</table>

**A. Literature Survey**

Several studies have been carried out to predict SCD or various cardiac diseases. In the context of predicting cardiac arrest, Das et al. [19-20] had various machine learning tools to predict cardiac arrest; they also used hybrid methods like Rough Set Theory (RST) and MLA to boost the accuracy level. RST is a useful tool for prediction, classification, and test and training data. Researchers also used various MLAs like Random Forest Model (RFM) and mathematical modelling to predict meteorological phenomena in this context Das et al. [21-22]. Several research had been conducted to predict various cardiac problems in this context Das et al. [23-25] discussed cardiac diseases using MLA and RST in the same context as Hota et al. [26-27] discussed cardiac problems using various MLA and soft computing techniques. Several researchers have also used mathematical modelling to predict cardiac problems in this context Nayak et al. [28-30] used mathematical modelling and machine learning algorithms to predict diseases like Malaria and Cardiac arrest. Several researchers also used soft computing techniques for social issues in this context Mishra et al. [31] had discussed legal justice for common people in the service sector. The general purpose of research on medical data in computer science is based on the principle of prediction and classifications; in most cases, the researcher used soft computing, MLA, and deep learning techniques for prediction and classifications. To predict cardiac arrest with significant accuracy Manuel et al.[32] had used convolutional neural network (CNN) and ECG images to classify SCD in the same context Kaspal et al.[33] had used a hybrid deep learning method to predict SCD in a similar context Sanchez et al.[34] had used prediction of SCD using wavelet transformation. Scherer et al. [35] had used various methods like pooling operation to significantly detect the objects. Several researchers have used methods like CNN and advanced soft computing methods for prediction and classification other than the medical field in this context Wang et al.[36] had used CNN along with Short-Time Fourier Transform to analyze motor fault analysis. The finding of various researchers considered various parameters like blood pressure and ECG signals for SCD; the ECG signals can predict SCD with significant accuracy, which was observed by monitoring for 24 hours ECG signals from the database of SCDH and considering a sampling frequency of 250 Hz[37]. Several research studies have also been conducted to find the death rate due to cardiac problems in this context Liu et al.[38] had used one-dimensional CNN for health monitoring. This paper used various techniques like RST and machine learning tools to predict cardiac problems. RST was initiated and developed by polish mathematician Pawlak [39]. RST helps find useful information from vague and imprecise data, as discussed by Das et al. [40]. Several studies have also been conducted using various alternate methods like empirical mode of decomposition (EMD) in this context Huang et al. [41] had discussed the Empirical mode of decomposition and spectrum analysis of Non-linear and Nonstationary time series.

**B. Material and Method Used**

This work deals with RST, as discussed by Yao and Deng [42], the fundamental concept of RST is based on upper approximations, lower approximations, boundary region, core, reduct, quality of approximation, and indiscernibility. The entire concept of based on indiscernibility, equivalence classes and target set.

**C. Fundamentals of RST**

The general representation RST is in the form of a table called an information system represented as <U, C, D>, where U is called the universe, C is the set of conditional attributes, and D is the set of decision attributes. The complete skeleton of RST is given in the following Table II.

**TABLE II. INITIAL INFORMATION TABLE**

<table>
<thead>
<tr>
<th>E</th>
<th>C</th>
<th>D</th>
</tr>
</thead>
<tbody>
<tr>
<td>E1</td>
<td>1</td>
<td>2</td>
</tr>
<tr>
<td>E2</td>
<td>1</td>
<td>1</td>
</tr>
<tr>
<td>E3</td>
<td>2</td>
<td>2</td>
</tr>
<tr>
<td>E4</td>
<td>2</td>
<td>1</td>
</tr>
<tr>
<td>E5</td>
<td>2</td>
<td>1</td>
</tr>
</tbody>
</table>

Description of the Table as follows E=< E1, E2, E3, E4, E5> are the records, C is the set of conditional attributes, D is the set decision attributes, and <1,2> represents the values of the decision and conditional attributes.

**D. Generalization of RST**

The generalization of RST includes a definition of upper approximations, lower approximations, boundary region, core, reduct, quality of the approximation, and indiscernibility and their mathematical representation.

1) **Indiscernibility:** Indiscernibility generally includes a single attribute or multiple attributes. Mathematically indiscernibility defines as

\[ \text{IND}(K)=\{(\alpha, \beta) \in R^2 \mid \forall m \in K, m(\alpha)=m(\beta) \} \]  

2) **Upper approximation:** Upper Approximation includes the concept possibility, i.e. the set of all objects possibly linked with the target.

\[ \text{Upper}(\text{AT})=\{M \in U/A : M\cap T \neq \emptyset \} \]

3) **Lower approximation:** Lower Approximation includes the concept of definiteness

\[ \text{Lower}(\text{AT})=\{M \in U/A : M\cap T = \emptyset \} \]

4) **Reduce:** Reduct forms a set of equivalence classes which is derived from the set of indiscernibility.
E. Various Algorithm used in this Study

1) Algorithm to find Indiscernibility

Step-1 ∀a ∈ U, where a is the conditional attributes
Step-2 ∀xy values of the attributes a
Step-3 for (i=1; i ≤ n; i++) where n is the number of conditional attributes
Step-4 check for indiscernibility i.e. goto step-5
Step-5 if the values of attributes a are either identical or distinct, then indiscernibility found, i.e. either a(x)=a(y), check for the combination attribute 1 to attributes n
else
goto step-3
end if
end for

2) Algorithm to find reduct

Step-1 ∀ IND where IND is the indiscernibility
Step-2 for (j=1; j ≤ n; j++) , n is the total number of indiscernibility
Step-3 verify whether the indiscernibility derive reduct
if the values of the indiscernibility are distinct
Reduct R is found else goto step-2
Step-4 end if
Step-5 end for

F. Empirical Mode Decomposition (EMD)

The idea of EMD was initiated by Hung et al., as discussed in the previous section. The EMD approach uses adaptive techniques to analyze stationary and nonstationary one-dimensional signals. EMD has the feature of decomposing a one-dimensional signal in intrinsic mode functions (IMFs). Two significant criteria are essential for the proper function of EMD criteria a. The total number of zero crossing and extrema in the one-dimensional signal either different or equal to numbers only by 1 and b, the lower and upper envelopes must be symmetrical, to estimate the mean value as zero. The following steps are required to calculate each intrinsic mode function (IMFs).

Step-1. Calculate local maxima and local minima of 1 dimensional signal i.e. x(t)
Step-2. Using Step-1 connect the local maxima & minima to calculate the upper & lower envelopes.
Step-3. Find the average of the of the upper & lower envelopes denoted as k
Step-4. New 1 dimensional signal to be found out as

$$h_i(t)=x(t)-k(t)$$

Step-5. For i = 1 to n

Check if h_k satisfies condition 1 and 2
Then Intrinsic mode function (IMF_i)=h_k(t)
Else goto step-5
Step-6 end if
Residue signal r_1(t)=x(t)-IMF_1
Step-7 If the estimated residue signal, function r_1(t), has monotonic nature, the method does not work, implying that the estimation of IMFs is completed
Else
Follow the procedure of estimating IMFs and the residue
End if
Step-8 The final

$$x(t)=\sum_{k=1}^{n}IMFs_k + r_n(t), r_n(t)$$

is the last residue found.

On the other hand, if it is not considered a monotonic function, the residual signal is now considered the original 1D signal, which is again evaluated by steps (1 to 3) for estimating the other IMFs.

EMD is the fundamental technique for Ensemble empirical mode decomposition (EEMD) and Complete Ensemble empirical mode decomposition (CEEMD).

G. Ensemble Empirical Mode Decomposition

EEMD, an enhanced form of the EMD technique [23], is categorized by being a noise-assisted technique used to decompose the 1D signals in their basic structures steps for EEMD:

Step-1 Generates new 1D signals by using 1D signal (basic) and basic Gaussian noise n(t)

For j=1 to m

$$x_j(t)=x(t) + n_j(t)$$

Step-2 Divide the signal generated by step-2 by the EMD technique
Step-3 Calculate the frequency band or the IMF on the kth iteration that will be

$$EEMD_k \approx \frac{\sum_{j=1}^{m}c_{jk}(t)}{m}$$

The frequency band or IMF, recognized by k for the trial j is denoted by c_{jk}(t).

H. Complete Ensemble Empirical Mode Decomposition

The CEEMD properly described by Torres et al [42] it is the advanced version of EEMD, CEEMD provides better separation than EEMD. Each trial’s residue is calculated as r_i(t) in this context.

$$r_1(t)=x(t)-IMF_1$$

The flow diagram of the above work is given in the next section for detailing and describing the work. A significant part of the work consists of logistic regression, RST and
EMD, EEMD, CEEMD and comparison of the methods with RST.

I. Fundamentals of Logistic Regression

Logistic regression helps make different classes; this method is also helpful in the case of outliers. This statistical technique is useful for the object with binary outcomes. Logistic regression represented by signum function. The mathematical representation of logistic regression is given as follows as

\[ \alpha_a = \frac{1}{1 + e^{-ax}} \]  

(10)

III. Method and Data Used for Analysis

This study considered several machine learning algorithms applied to UCI data set for accuracy measure. This work includes three fundamental concepts a. Logistic Regression b. RST to find the significant attributes c. EMD and it two versions for accurate prediction d. Comparison among other methods like Support Vector Machine (SVM), Random Forest Method (RFM) & logistic regression after boosting. The data collected from various districts are given in Table III.

The data is collected from various parts of our states, i.e. Odisha's district, a total of 5,60,000 samples has been taken for the studies. The application of logistic regression reduced the data set into 6 different records. Details of logistic regression are given in the Fig. 1. The logistic regression used for this study has better and more concrete results than linear regression, as logistic regression works better for the outliers. Logistic regression works better for dynamically changing data sets. Training the data set for logistic regression is given as follows. We had considered several cardiac-related diseases from various parts of our state, a total of 5,60,000 data as mentioned in the Table. Using the concept of upper approximation, lower approximation boundary region, i.e., Boundary Region = Upper Approximation – Lower Approximations. Analyzing the Table using the concept of boundary region found that other attribute is insignificant in forming the groups for cardiac problems. This study includes the basis for dividing a larger dataset into groups by training the data set according to the research requirements. This work considered the medical survey according to the research needs; This study divided the entire data sample into categories, i.e. category- 'severe', moderate, average, low, free from cardiac illness, renamed as category-1, category-2, category-3, category-4, category-5. Using these five categories resulted in 5- distinct groups of 5- records. The records consist of only two values, significant and pointless. Rename the conditional attributes general cardiac problems as 11, Sleeping disorder as 22, Sudden Cardiac death as 33, Pneumonia as 44, Cardiac Asthma as 55, and values significant as a and pointless as b for better understanding and application. When the data samples are divided into categories, this work follows the rules of grouping the data set according to its group; this also follows the concepts of approximations, with a minor difference in diseases between the persons included in one group. The grouping is formed by using the following algorithm. The target values are more than two in our cases with the following assumptions. Target variable can have three or more possible types, which are the five diseases given in the Table III.

In this case, the sigmoid and softmax functions are used for regrouping the sample according to various features. Classes are established using softmax function, and categories or classifications are established using the sigmoid function. P classes softmax function represented by \[ \sigma(x_i) = \frac{e^{x_i}}{\sum_{j=1}^{P} e^{x_j}} \], where I varies from 1 to p.

![Fig. 1. Application of Logistic regression using the sigmoid function](image)

**Table III. Initial Data Table**

<table>
<thead>
<tr>
<th>District</th>
<th>Cardiac problems</th>
<th>Sleeping Disorder</th>
<th>Sudden Cardiac Death</th>
<th>Pneumonia</th>
<th>Cardiac Asthma</th>
<th>Others</th>
<th>Total</th>
</tr>
</thead>
<tbody>
<tr>
<td>Koraput</td>
<td>10,000</td>
<td>50,000</td>
<td>15000</td>
<td>25000</td>
<td>50000</td>
<td>5000</td>
<td>1,55,000</td>
</tr>
<tr>
<td>Dhenkanal</td>
<td>20,000</td>
<td>20,000</td>
<td>25000</td>
<td>15000</td>
<td>5000</td>
<td>15000</td>
<td>1,00,000</td>
</tr>
<tr>
<td>Jaipur</td>
<td>30,000</td>
<td>10,000</td>
<td>15000</td>
<td>10000</td>
<td>10000</td>
<td>5000</td>
<td>80,000</td>
</tr>
<tr>
<td>Cuttack</td>
<td>15,000</td>
<td>25,000</td>
<td>10000</td>
<td>5000</td>
<td>5000</td>
<td>15000</td>
<td>75,000</td>
</tr>
<tr>
<td>Puri</td>
<td>25,000</td>
<td>10,000</td>
<td>5000</td>
<td>10000</td>
<td>10000</td>
<td>10000</td>
<td>70,000</td>
</tr>
<tr>
<td>Sambalpur</td>
<td>20,000</td>
<td>20,000</td>
<td>15000</td>
<td>5000</td>
<td>15000</td>
<td>5000</td>
<td>80,000</td>
</tr>
</tbody>
</table>

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TABLE IV. APPROXIMATION AFTER LOGISTIC REGRESSION

<table>
<thead>
<tr>
<th>Records</th>
<th>11</th>
<th>22</th>
<th>33</th>
<th>44</th>
<th>55</th>
</tr>
</thead>
<tbody>
<tr>
<td>E1</td>
<td>A</td>
<td>A</td>
<td>a</td>
<td>a</td>
<td>a</td>
</tr>
<tr>
<td>E2</td>
<td>A</td>
<td>A</td>
<td>b</td>
<td>b</td>
<td>a</td>
</tr>
<tr>
<td>E3</td>
<td>A</td>
<td>B</td>
<td>a</td>
<td>b</td>
<td>b</td>
</tr>
<tr>
<td>E4</td>
<td>B</td>
<td>B</td>
<td>b</td>
<td>b</td>
<td>a</td>
</tr>
<tr>
<td>E5</td>
<td>B</td>
<td>a</td>
<td>a</td>
<td>b</td>
<td>a</td>
</tr>
<tr>
<td>E6</td>
<td>B</td>
<td>a</td>
<td>b</td>
<td>a</td>
<td>b</td>
</tr>
</tbody>
</table>

A. Universal Data Set

Data collected from the University of California Irvine machine learning repository considered UCI repository provided the Heart Disease dataset. This includes 14 different attributes for 304 cases, with target values in deciding whether the patient is suffering from cardiac problems or not, as well as divided the patients according to the symptoms that leads to the finding the number of patients at risk for cardiac problems is extracted from this dataset. There is binary aspect of the dataset i.e. each row corresponds to a single record in this dataset, which has 303 rows and 14 columns, presented in the following Table V and Fig. 2.

The Table represent the detail attribute description and the provided the detail about the target, the figure given below describe the correlation values between -1 to 1. Correlation coefficients range from -1 to +1, indicating a negative or positive correlation. Age, high blood pressure, and high cholesterol levels may strongly correlate with an increased risk of heart disease. Correlation matrices help identify potential risk factors and develop predictive models. This study uses several Machine learning algorithms (MLA). The machine learning algorithms approaches are initially based on weak learner, by combining the multiple algorithm leads to strong learner.

TABLE V. VARIOUS CONDITIONAL ATTRIBUTES RELATED TO CARDIAC

<table>
<thead>
<tr>
<th>SI</th>
<th>Category</th>
<th>Details</th>
<th>Target</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>Span</td>
<td>Chronic/Non Chronic</td>
<td>Continuous</td>
</tr>
<tr>
<td>2</td>
<td>Gender</td>
<td>Patient’s Gender</td>
<td>Female/Male</td>
</tr>
<tr>
<td>3</td>
<td>CP</td>
<td>Chest Pain</td>
<td>Type(1,2,3,4)</td>
</tr>
<tr>
<td>4</td>
<td>Measure</td>
<td>Resting blood pressure</td>
<td>Continuous</td>
</tr>
<tr>
<td>5</td>
<td>Ch</td>
<td>Cholesterol range</td>
<td>Continuous</td>
</tr>
<tr>
<td>6</td>
<td>Bs</td>
<td>Fasting blood sugar</td>
<td>Less than 120mg/dl</td>
</tr>
<tr>
<td>7</td>
<td>Elec</td>
<td>Electrocardiograph</td>
<td>Values(1,2,3,4,5)</td>
</tr>
<tr>
<td>8</td>
<td>Tha</td>
<td>Heart beat</td>
<td>Continuous</td>
</tr>
<tr>
<td>9</td>
<td>Exa</td>
<td>Induced angina</td>
<td>0/1</td>
</tr>
<tr>
<td>10</td>
<td>Op</td>
<td>Amount of rest/Non rest</td>
<td>Continuous</td>
</tr>
<tr>
<td>11</td>
<td>Slope</td>
<td>Peak</td>
<td>Type(Up/Flat/Dow n)</td>
</tr>
<tr>
<td>12</td>
<td>Car</td>
<td>Gives number of major vessels colored by fluoroscopy</td>
<td>Binary(0/1)</td>
</tr>
<tr>
<td>13</td>
<td>Tha2</td>
<td>Faulty</td>
<td>Binary(0/1)</td>
</tr>
<tr>
<td>14</td>
<td>Num</td>
<td>Cardiac problems</td>
<td>Yes(0,1,2,3,4)</td>
</tr>
</tbody>
</table>

B. Training by using Naive Bayes Classifiers

Input:- Data sets related to cardiac problems trained with the symptoms given in the rules

Output:- Expected results with binary results Y/N (Y-Yes, N-No)

Step 1. Different category of the dataset trained with binary values (Agree/Disagree). This signifies as an indicator.

Step 2. Expected probability = (number of objects belongs to the groups(agree)/ Total samples) dataset belong to agree category. Similarly, the successive probability calculated by

Probability = (number of objects belong to group after iteration)/ Total samples

To find the class total number of disagree

Step 3. Find R, where R is the total number of attributes

\[ R_d = \sum x_i \text{, where } x_i \in \text{Agree} \]

\[ R_n = \sum y_i \text{, where } y_i \in \text{Disagree} \]

Step 4. for j 1 to n

1) Find the group level using the formula

Probability for attribute belongs to agree = (Total count for agree belongs to class i)/ (n_i) Probability for attribute belongs to disagree = (Total count for disagree belongs to class i)/ (n_i)

Step 5. Divide the new groups of patients belonging to probability R/T, where R represents the groups, and T is its respective features.

Step 6. Compute

\[ P(\text{agree}) = \prod P(\text{Class of attributes belongs to agree class}) \]

Similarly, compute probability for disagree class by the total product of the probability of disagree × class attributes of disagree class.
i=1 to n 

g. Assign the new minutes as patients fit to agree class or disagree class, according to higher voting.

C. Calculating Misclassification before Training

Step-1 Represent P as region belongs to agree class, N as region belongs to disagree class, according to the equation. 
\[ \sum_{i=0}^{10000} t_i m_i = 0, k=0.01. \]

Step-2 for i 1 to 10000

Step-3 Verify that the point erroneously classified to the region belongs to agree class

if

\[ T_i \in N \& \sum t_i m_i > 0 \]

Update \( T_{\text{new}} = T_{\text{old}} - k m_i \)

else go to step 5

Step-4 Check for the point misclassified wrongly in the Negative region

\( T_e P \& \sum t_i m_i < 0 \)

Update \( T_{\text{new}} = T_{\text{old}} + km_i \)

else

}

Step-5 end for

D. Algorithm for Strong Learner using Boosting

Step-1 Adjust the weights as \( \frac{1}{p} \) for p observation

Step-2 Select the features k with respect to the smallest Gini index/Concentrated information gain,

Evaluate the over-all error

Step 3 Find the stump of the routine

Step-4 Calculate the new weights with respect to misclassifications

Step-5 All weights to be normalized accordingly

Step-6 For all i varies from 1 to n

if

\{ 

A configured number of estimators reached the saturation/Significant accuracy level achieved

\}

else go to step 2

}

Step-7 end if

Step-8 end for

IV. Result Analysis Using Machine Learning Tools

This study used Max-Min concept to overcome the results with respect to oversampling. This study used UCI sources to provide the diseases related to cardiac. The description of the data set defines in Table-V useful in regulating the cases of a person suffering from cardiac problems. This study divided the domain into two classes: people who suffered from cardiac problems and those who did not suffer from cardiac problems using a set of symptoms. This paper used two concepts of training and testing for people in danger of cardiac problems and recovered from cardiac problems.

A. Finding Indiscernibility

Finding indiscernibility, reduce and core don't have any generalized algorithm; this study uses the algorithm to find indiscernibility as defined above. This work uses data in Table IV to find indiscernibility, reduce and core. IN denotes indiscernibility.

\[ \begin{align*}
\text{IN}(11) &= \{(E_1, E_2, E_3), (E_2, E_4, E_6)\} \\
\text{IN}(22) &= \{(E_1, E_2), (E_3, E_5), (E_5, E_6)\} \\
\text{IN}(33) &= \{(E_1, E_3, E_5), (E_2, E_4, E_6)\} \\
\text{IN}(44) &= \{(E_1, E_4), (E_2, E_3, E_6)\} \\
\text{IN}(55) &= \{(E_1, E_2, E_5, E_6)\} \\
\text{IN}(11,22) &= \{(E_1, E_2), E_3, E_4, E_5, E_6\} \\
\text{IN}(22,33) &= \{(E_1, E_3), (E_2, E_4), E_5, E_6\} \\
\text{IN}(33,44) &= \{(E_1, E_2, E_3), (E_5, E_6)\} \\
\text{IN}(44,55) &= \{(E_1, E_5, E_6), (E_2, E_4, E_2)\} \\
\text{IN}(44,55) &= \{(E_1, E_6), (E_2, E_3, E_5)\} \\
\text{IN}(11,33) &= \{(E_1, E_3), (E_2, E_4), (E_5, E_6)\} \\
\text{IN}(11,44) &= \{(E_1, E_2), (E_3, E_4), E_5, E_6\} \\
\text{IN}(11,55) &= \{(E_1, E_4), (E_2, E_3, E_5)\} \\
\text{IN}(22,33) &= \{(E_1, E_3), (E_2, E_4), E_5, E_6\} \\
\text{IN}(33,44) &= \{(E_1, E_2, E_3), (E_5, E_6)\} \\
\text{IN}(22,44) &= \{(E_1, E_2), (E_3, E_4), (E_5, E_6)\} \\
\text{IN}(22,55) &= \{(E_1, E_3), (E_2, E_4), E_5, E_6\} \\
\text{IN}(33,55) &= \{(E_1, E_5), (E_2, E_4), E_5, E_6\} \\
\text{IN}(44,55) &= \{(E_1, E_6), (E_2, E_3, E_5)\} \\
\text{IN}(11,22,33) &= \{(E_1, E_2), E_3, E_4, E_5, E_6\} \\
\text{IN}(11,33,44) &= \{(E_1, E_2, E_3), E_5, E_6\} \\
\text{IN}(11,44,55) &= \{(E_1, E_2, E_3), (E_5, E_6)\} \\
\text{IN}(22,33,44) &= \{(E_1, E_2, E_3, E_4, E_5)\} \\
\text{IN}(11,22,33,44) &= \{(E_1, E_2, E_3, E_4, E_5, E_6)\} \\
\text{IN}(22,33,44,55) &= \{(E_1, E_2, E_3, E_4, E_5, E_6)\} \\
\end{align*} \]
The above equations from (11) to (37) represent the indiscernibility; indiscernibility which don’t form groups are called as reduct. The studies lead to the following results. As the following indiscernibility does not produce any groups implies that the following indiscernibility leads to a set of reducts.

\[ \text{IN}(11,33,44), \text{IN}(22,33,44), \text{IN}(11,22,33,44), \text{IN} (22,33,44,55) \]

Core=∩ Reduct

In this study, the reduct is found to be

\[ \text{Core}=\cap \text{IN}(11,33,44), \text{IN}(22,33,44), \text{IN}(11,22,33,44), \text{IN} (22,33,44,55) \]

= (33), as 33 is denoted for SCD as SCD is related to cardiac arrest. This study considered the University of California Irvine machine learning repository considered (UCI). We have used several boosting algorithms to find the accuracy. Further Analysis is being conducted using the strength of RST [27]. The result describes the symptoms of cardiac problems given in the following Table VI.

**TABLE VI. FINAL REDUCT TABLE WITH TARGET VALUES**

<table>
<thead>
<tr>
<th>EE</th>
<th>a11</th>
<th>b11</th>
<th>c11</th>
<th>d11</th>
<th>e11</th>
<th>D</th>
</tr>
</thead>
<tbody>
<tr>
<td>EE1</td>
<td>1</td>
<td>1</td>
<td>1</td>
<td>0</td>
<td>0</td>
<td>1</td>
</tr>
</tbody>
</table>

In the above Table VI, EE represents the records \( a_{11}, b_{11}, c_{11}, d_{11}, e_{11} \) are the conditional attributes. The above attributes are renamed as \( a_{11} \) irregular sleeping habit, \( b_{11} \) fluctuation in pulse rate, similarly \( c_{11} \) and \( d_{11} \) are breathing difficulties, sweating severely to very severe, and abnormal cholesterol level, respectively. The decision attributes \( d \) agree with sudden cardiac death (SCD), \( <0,1> \) represents insignificant and significant applicable for both conditional attributes and decision attributes.

**B. Training Phase**

The training phase includes the above symptoms taken from Table VII as the input of the study, which is responsible for cardiac arrest or SCD. The training phase used the Naïve Bayes classifiers algorithm. The above concept resulted in producing Table VII.

**TABLE VII. WEAK LEARNER COMPARISON**

<table>
<thead>
<tr>
<th>Serial</th>
<th>TIDM</th>
<th>Accuracy</th>
<th>Mean AE</th>
<th>Root mean SE</th>
<th>Relative absolute error</th>
<th>Relative SE</th>
<th>F1-Score</th>
</tr>
</thead>
<tbody>
<tr>
<td>Naive B</td>
<td>7</td>
<td>85</td>
<td>0.65</td>
<td>0.85</td>
<td>121</td>
<td>128</td>
<td>0.45</td>
</tr>
<tr>
<td>Alternate DT</td>
<td>60</td>
<td>96</td>
<td>0.29</td>
<td>0.49</td>
<td>95</td>
<td>96</td>
<td>0.9</td>
</tr>
<tr>
<td>Random Forest</td>
<td>2.23</td>
<td>93</td>
<td>0.32</td>
<td>0.45</td>
<td>84</td>
<td>81</td>
<td>0.87</td>
</tr>
<tr>
<td>Reduce Error PT</td>
<td>11.45</td>
<td>97</td>
<td>0.28</td>
<td>0.46</td>
<td>98</td>
<td>98</td>
<td>0.85</td>
</tr>
<tr>
<td>C and R tree</td>
<td>57</td>
<td>85</td>
<td>0.29</td>
<td>0.58</td>
<td>99</td>
<td>99</td>
<td>0.87</td>
</tr>
</tbody>
</table>

**TABLE VIII. STRONG LEARNER COMPARISON [(ADABoost COMPARISON)]**

<table>
<thead>
<tr>
<th>Serial</th>
<th>TIDM</th>
<th>Accuracy</th>
<th>Mean AE</th>
<th>Root mean SE</th>
<th>Relative absolute error</th>
<th>Relative SE</th>
<th>F1-Score</th>
</tr>
</thead>
<tbody>
<tr>
<td>ABNB</td>
<td>19.5</td>
<td>94</td>
<td>0.5</td>
<td>0.77</td>
<td>132</td>
<td>155</td>
<td>0.87</td>
</tr>
<tr>
<td>ABAAlternate DT</td>
<td>32</td>
<td>94</td>
<td>0.18</td>
<td>0.28</td>
<td>60</td>
<td>98</td>
<td>0.96</td>
</tr>
<tr>
<td>ABRF</td>
<td>11</td>
<td>96</td>
<td>0.11</td>
<td>0.25</td>
<td>66</td>
<td>66</td>
<td>0.98</td>
</tr>
<tr>
<td>ABRUPT</td>
<td>65</td>
<td>93</td>
<td>0.21</td>
<td>0.41</td>
<td>46</td>
<td>93</td>
<td>0.85</td>
</tr>
<tr>
<td>ABCART</td>
<td>96</td>
<td>95</td>
<td>0.23</td>
<td>0.43</td>
<td>42</td>
<td>92</td>
<td>0.87</td>
</tr>
</tbody>
</table>
The Fig. 4 represents the accuracy level of various classifiers with respect to a single classifier application. This study combines various classifiers to get the following results. Applying the combination of two classifiers at a time gives the following output.

ABNB-AdaBoost Naive Bayes, AB Alternate DT-Adaboost Alternate decision tree, ABRF- Adaboost Random Forest, ABRUPT- Adaboost Reduced error prunning tree, ABCART- Adaboost Correlation and regression tree. There are significant increments of accuracy level after the use of additive strong learner method. The entire result of the output is given in the Fig. 3.

C. Performance Evaluations

1) Entropy: Entropy calculates a structure’s impulsiveness or condition. The idea was given by Rudolf Clausius in 1850 suggested this concept. Mathematical model of this concept was given by

\[ \text{Entropy} = - \sum k(x) \log k(x) \]  

where\( k(x) \) is the part of the concept of a given class.

2) Gini index: Gini Index or Gini Coefficient define as the supply of resources in a given population. In general

\[ \text{Gini Impurity} = 1 - \sum_{i=1}^{c} k_i(x)^2 \]  

3) Information gain: The decrease of Entropy attained by varying the information (In this study, information derived from the dataset) is called as information gain, and it is normally employed in the training of DT (Decision Tree). This information gain calculation depends upon the Entropy of the prior and posterior transformation of the dataset. The entire information gain is given by the following mathematical formulas.

\[ \text{Information Gain} (k_p,f) = I(k_p) - \frac{R_{left}}{R} I(k_{left}) - \frac{R_{right}}{R} I(k_{right}) \]  

4) For unbalanced and vague data sets, accuracy measure depended upon the ratio of the number of perfectly classified instances by the total number of sample instances.

\[ \text{Accuracy} = \frac{TP_s + TN_s}{TP_s + TN_s + FP_s + FN_s} \]  

where \( TN \)- True negative, \( TP \)- True Positive, \( FN \)- False Negative, \( FP \)- False positive.

Total number positive prediction is called as precision.

\[ \text{Precision} = \frac{TP_s}{TP_s + FP_s} \]

5) Root mean squared error (RMSE), a technique to calculate the forecasting of numeric success

\[ \text{Root Mean Squared Error} (\text{RMSE}) = \sqrt{\frac{\sum_{i=1}^{n} (k_i - k_i^f)^2}{n}} \]  

6) Mean Absolute Error (MAE) is the calculation of the absolute difference between actual and expected values

\[ \text{MAE} = \sum_{i=1}^{n} |k_i - k_i^f| \]

7) Relative Absolute Error (RAE)

\[ \text{RAE} = \frac{(k_i - k_i^f)^2}{(k_i - k_i^f)^2} \text{ where } i \text{ varies from 1 to all samples} \]

8) Total Squared Error

\[ \text{RRSE} = \sqrt{\text{RAE}} \]

Accuracy measure is depicted in table-8

9) Recall

\[ \text{recall} = \frac{TP_s}{TP_s + FN_s} \]
10) F1 score: F1-score is the combination of recall and precision

\[ \text{F1-score} = \frac{2 \times (\text{Precision} \times \text{Recall})}{\text{Precision} + \text{Recall}} \] (50)

Where \( f \) is the feature split, \( k_p \) is the parent dataset, \( k_{left} \) and \( k_{right} \) are the posterior probability of left and right child node. The equations (39), (40) and (41) are derived from NB classifiers.

D. Application of the Adaboost Technique

The Adaboost technique combines multiple weak classifiers to make it a single strong classifier. The most commonly used AdaBoost algorithm is the decision stumps, an alternative name for these decision trees, as discussed by Hussein et al. [43]. This technique generates a model that distributes equal weights at all data points and subsequently provides points to those data points incorrectly classified with higher weights groups. In the next model, significant importance is given to the data points with higher weights. An error margin of \( 10^{-3} \) to \( 10^{-4} \) is considered. The process continues till the desired accuracy is achieved, as discussed by Reddy et al.[44].

E. Algorithm for Boosting

- Step-1 Initialize the data points \((x_i, y_i)\), for \( i = 1 \) to \( n \)
- Step-2 Each \( x_i \) is the instance of space \( X \), \( y_i \) is collecting all labels of space \( Y \).
- Step-3 for the training instance \( i \) by rounding \( t \) is given as \( D_t \), Initialize the weights as \( D_0(i) = \frac{1}{M} \), \( i = 1 \) to \( M \)
- Step-4 for the total number of samples, find the weight of misclassified according to the standard algorithm increased each step

\[ \alpha_{it} = \frac{1}{2} \ln \left( \frac{L_{it} + L_{i-1}}{L_{i-1} + L_{it}} \right) \] (51)

- Step-5 Use basic logistic regression to adjust overfitting
- Step-6 Continue with step 1 to 5 till the desired accuracy is achieved.

To overcome oversampling, a technique called Synthetic Minority Oversampling Technique is being used; the following steps are being implemented, and the results are shown in Table VII to Table IX, accuracy and error like RMSE and MAE are shown from Fig. 3 to Fig. 8. The details of SMOTE are described in the subsequent section.

F. Procedure for SMOTE

Before the SMOTE algorithm, Data pre-processing is required as the classification algorithm's objective is to collect the raw data and create an outline for every class as conclusive with less error margin to attain accurate prediction. The examples that maintain significant distance from outlines form a group more easily than those that are near the outlines. The examples nearer to the outlines are always a challenge for learning existing classifiers. SMOTE provide a significant result in overcoming these challenges.
G. SMOTE for Classifications

Step-1 This method's objective is to produce examples, by using the concept

\[ T=2\times(K-M) \] (52)

Where K denotes the majority samples of the group, M denotes the minority samples of the group and T initial examples are newly generated. The basic examples generated by SMOTE will be recognized or excluded depending upon two concepts i.e.

Stage-1\{ \( K_1, K_2, K_3, \ldots, K_n \) \} is the group od example and \( K_j(j) \) is the jth attribute values of \( K_i \) example.

Where j varies from 1 to the total number of samples. Let \( R_a \) and \( R_b \) be the minority and majority examples, respectively. The distance d is calculated between \( K_i \) and \( R_a \) and \( R_b \) for recognition and exclusion. The distance between \( K_{\text{minority}}(K_1, R_a), K_{\text{majority}}(K_1, R_b) \). There will be total number of n steps. The distance is calculated as

\[ K_{\text{minority}}(K_1, R_a) = \sum_{i=1}^{n} \sqrt{((K_i - R_a)^2)} \] (53)

Similarly

\[ K_{\text{majority}}(K_1, R_b) = \sum_{i=1}^{n} \sqrt{((K_i - R_b)^2)} \] (54)

Using equations 52 and 53, majority and minority distances are calculated, as

\[ K_{\text{minority}}(K_1, R_a) = (K_1, K_2, \ldots, K_n) \] (55)

\[ K_{\text{majority}}(K_1, R_b) = (L_1, L_2, \ldots, L_m) \] (56)

From equation (54), calculate min (K_1, K_2, ..., K_n) to find the minimum of a minority; similarly, calculate min (L_1, L_2, ..., L_m) to find a minimum of majority.

Step-2 If minimum (minority) <= minimum (majority) out coming examples are accepted else rejected.

Step-3 continue with step 1 to 3 for the entire samples to achieve the required outcomes.

<table>
<thead>
<tr>
<th>TABLE X. A COMPARATIVE ANALYSIS</th>
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<tr>
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The detailed comparison of error analysis and accuracy level was given in Fig. 7 and 8. This study has compared its results with several cited papers in Table X.

H. Statistical Validation of the Result

This study uses statistical methods for the validation of the claim. There are several statistical methods available for Analysis and validation. The generally adopted method for validation is the chi-squared statistical test. This study uses two fundamental statistical techniques, i.e., one dimensional \( \chi^2 \) (Chi-squared test) and two-dimensional chi-squared test for this study.

1) Statistical validation using one dimensional \( \chi^2 \) test

\[ \chi^2 = \sum \frac{(O_i-E_i)^2}{E_i} \] (57)

\( H_0 \) (Null Hypothesis) - The above multiple classifiers are not provided accuracy as desired

\( H_1 \) (Alternate Hypothesis) - The above multiple classifiers provide desired results as required.

Where \( O_i \) is the observed samples, and \( E_i \) is the expected samples. A survey is conducted over 10,000 population by grouping 10,000 per group total of 10 available groups. The observed values are 35, 5, 10, 5.5, 10, 3, 7, 15, 5, unit measure in 10,000. With expected values 10%, 25%, 5%, 5%, 25%, 30%, 45%, 5%, 10%, 15%. So total observed samples are 100,000, and the expected values are 10, 25, 5, 5, 25, 30, 45, 5, 10, 15.

Calculated \( \chi^2 = 160, \chi^2 = 160, \chi^2 = 160 \), much less than the calculated \( \chi^2 \) value, so rejected the Null hypothesis. The same data set was also applied on a two-dimensional array and the calculated \( \chi^2 = 21.25 \) where the tabular values were calculated as 17.23. The null hypothesis is rejected.

V. CONCLUSION AND FUTURE WORK

This study entirely depended upon a vague and imprecise dataset from various parts of our state's Odisha and medical advisory. The initial idea is to group the data set according to its class. The dataset was divided into six categories using logistic regression. Using two concepts of RST, i.e., indiscernibility and strength, showed that SCD/ Cardiac-arrest was the most significant disease among all cardiac-related problems. The subsequent section of the paper used the boosting technique combining several classifiers to measure the error rate and accuracy level. As the combination increases, the accuracy also increases. There was also a comparative analysis of these studies and EMD and EMD studies in Table X. These studies can be extended to the fields like sports especially making uniforms on cricket fields worldwide, and entertainments like movies more entertaining for common people.

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An Improved Artificial Bee Colony Optimization Algorithm for Test Suite Minimization

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Abstract—Software testing is essential process for maintaining the quality of software. Due to changes in customer demands or industry, software needs to be updated regularly. Therefore software becomes more complex and test suite size also increases exponentially. As a result, testing incurs a large overhead in terms of time, resources, and costs associated with testing. Additionally, handling and operating huge test suites can be cumbersome and inefficient, often resulting in duplication of effort and redundant test coverage. Test suite minimization strategy can help in resolving this issue. Test suite reduction is an efficient method for increasing the overall efficacy of a test suite and removing obsolete test cases. The paper demonstrates an improved artificial bee colony optimization algorithm for test suite minimization. The exploitation behavior of algorithm is improved by amalgamating the teaching learning based optimization technique. Second, the learner performance factor is used to explore the more solutions. The aim of the algorithm is to remove the redundant test cases, while still ensuring effectiveness of fault detection capability. The algorithm compared against three established methods (GA, ABC, and TLBO) using a benchmark dataset. The experiment results show that proposed algorithm reduction rate more than 50% with negligible loss in fault detection capability. The results obtained through empirical analysis show that the suggested algorithm has surpassed the other algorithms in performance.

Keywords—Test suite; test suite minimization; TLBO; ABC; nature inspired algorithm

I. INTRODUCTION

Software engineering deals with the design, analysis, implementation, maintenance and testing of software. Once software is evolved, its defects and shortcomings are analyzed with the help of software testing [4]. Out of various testing methods, the regression testing is very important as it involves the modification or insertion of a code into the already working code [8]. More is the size of test suite more will be testing time for each run. This testing time even varies in weeks also. It becomes difficult for a developer to get early feedback of the software developed. Therefore, the developer cannot fix the problems arising into the software timely [1]. So, no more changes could be done in the software and root cause of test suite failure could not be fixed. Software engineers must utilize their time and resources effectively to prevent these issues [2]. A lot of researchers have carried out their work in this particular area [3]. Test case selection (TCS), test case prioritization (TCP), and test case minimization (TCM) are three techniques used to handle complex problems associated with testing.

Test case selection (TCS) [21] is used to choose the test cases for the modified portion of the software. The test cases chosen from the test suite may depend on how well the selection process works. According to a specific criterion, test cases are prioritized, and the highest priority test cases are run first to achieve particular objectives. During test suite reduction, redundant test cases are removed from test suite [31]. Test case minimization, which gets rid of unnecessary ones, speeds up regression testing. The utilization of TCM depends on how it will help minimize time and cost for a particular software [6]. Based on criteria, minimization can reduce the size of test suite. There exist different criteria like path coverage, statement coverage, fault coverage, etc. It’s important to always keep in mind that minimization of any kind carries risk and might not offer complete coverage. It might also omit some fault revealing test cases. The size of the test suite, effectiveness, fault detection cost, and execution time are the important parameters of software that should be measured at the time of reduction. The effectiveness of software is decided by measuring these parameters. The comparison of a reduced suite with a corresponding unreduced suite using criteria other than suite size is also an important issue. As the testing basically involves the detection of faults in the software, it can be said that fault detection capability is one of the measures for test suite quality. An important shortcoming of the reduction process is the complete elimination of test cases from a suite, which may lead to a decrease in the fault detection capability and effectiveness of the remaining suite. Thus, a proper exchange between fault detection effectiveness and execution time should be taken into account before the implementation of TSR [7]. Therefore, the problem of test suite minimization could be considered an NP-hard problem. Optimization techniques efficiently solve these problems. Therefore, it can also be said that effectiveness and efficiency can be increased in regression testing through optimization techniques. Optimization helps us extract the best fit solution for the problems [36]. Soft computing uses artificial intelligence and natural selection to solve very complex problems that analytical (hard computing) formulations cannot solve. By incorporating soft computing techniques in optimization, we can further improve the accuracy and speed of regression testing. Soft computing can handle uncertainty and imprecision in data, making it a valuable tool for optimizing complex systems. Many researchers have investigated different techniques to minimize test suite while maintaining the coverage as maximum as possible [34,37]. CMIMX technique with a prominent soft computing technique (ABC) was implemented to ensure the reduced set with maximum fault
This paper proposed a new hybrid algorithm for test suite minimization. The proposed algorithm attains the minimize test suite with maximum coverage. The minimize test case achieve it by selecting two objectives. First, the algorithm aims to identify the minimized test suite that achieves maximum statement coverage. Second, it should also satisfy the fault detection capability of selected test cases. To accomplish our objective we have selected Artificial Bee Colony and Teaching Learning based optimization. The key contributions of our presented work are summarized as follows:

- Integration of TLBO operator into ABC algorithm for enhanced search performance.
- Introduction of learner’s performance concept to accommodate varying abilities.
- Minimization of test suite while maintaining optimal test case coverage.
- Analysis of fault coverage loss resulting from test suite minimization.

The remaining of this paper is organized as follows:

Section II describes the literature study related to minimization techniques. The technical background of the techniques explained in Section III. Section IV illustrates proposed methodology. Experimental details are present in Section V. Section VI describes conclusion and future work.

II. STATE-OF-THE-ART

In this section, we present a comprehensive review of the existing literature on test suite minimization techniques for addressing NP-hard problems. Researchers have extensively explored the application of soft computing techniques to tackle these challenging problems [35]. We summarize the key findings and contributions of various studies in Table I, highlighting the author names, techniques used, coverage criteria, and defined results.

ABC and TLBO algorithms have emerged and popular algorithm of literature and demonstrated the promising results across the different domains like long-term economic dispatch problem in hydropower systems [11]. Rao et al. initially proposed simple TLBO, which lack an inertia weight value [15]. I-TLBO, an enhanced version of TLBO was developed with the concept of adaptive teaching strategy and self-learning [13]. It was implemented to strengthen the exploration and exploitation capabilities. Zhang et al. [17] integrated TLBO algorithm into onlooker bee phase. Additionally, a novel searching approach for the bee phase was used to enhance population variety and quicken convergence. One such technique is FATLBO in which F and A stands for fuzzy and adaptive. This technique used fuzzy logic to adaptively adjust the parameters of TLBO algorithm [18]. LebTLBO was proposed by Chen et al. [14] to increase the performance of TLBO by incorporating the learning enthusiasm mechanism. It is clear from the literature review that previous research has mostly focused on test suite minimization using various coverage criteria. However, there are a number of gaps in the available literature, including a lack of analysis on fault coverage and limited use of the ABC algorithm and TLBO algorithm in the context of minimization principles. Therefore, we presented ABC and TLBO technique with addition of learning factor to enhance the capability to search more solution space. We suggest a unique method for test suite minimization termed Learner performance-based ABC and TLBO (LpABTLO) to fill these gaps. This method uses statement coverage as the coverage requirement and subsequently analyses fault detection loss.

Our suggested LpABTLO technique aims to fill these gaps in order to develop test suite minimization techniques and deliver more thorough insights into fault coverage analysis.

III. TECHNICAL BACKGROUND

This section gives a technical overview of the TLBO algorithm and ABC, two key optimization strategies. These approaches gained a lot of focus in the different fields to solve optimization problems.

A. Artificial Bee Colony Algorithm (ABC) [38]

The novel swarm-based stochastic optimization method known as ABC, developed by Karaboga, mimics the foraging behavior of honey bees [33]. Honey bees are organized into three groups in ABC: workers, bystanders, and scouts [9]. Employee bees compose the initial half of the colony, and observer bees made up the remaining half [10]. During the employee bee phase, bees search for a food source and collect data on its quality. As a result, only one bee is assigned to each food source [11]. The food source with the higher fitness value is more likely to be picked by onlookers than the one with the lower fitness value. Each onlooker bee then searches for a nearby food source close to the selected one and reserves the best among them. If a food source remains unchanged for a specific period of time, the associated employed bee turns scout and explores a new food source randomly [12]. Fig. 1 explains the working of algorithm.


<table>
<thead>
<tr>
<th>Author Name</th>
<th>Technique</th>
<th>Coverage Criterion</th>
<th>Defined Result</th>
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<tbody>
<tr>
<td>Khan et al. [27]</td>
<td>GA and mutation analysis</td>
<td>Software testing efficiency</td>
<td>Optimized software testing efficiency through GA and mutation analysis</td>
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<tr>
<td>Mala et al. [28]</td>
<td>ABC algorithm</td>
<td>Path coverage</td>
<td>ABC: 99% coverage in 50 cycles, GA: 90% coverage in 300 generations</td>
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<td>Khari et al. [22]</td>
<td>ABC and cuckoo search methods</td>
<td>Test case size and path coverage</td>
<td>ABC and cuckoo search minimize size of test case with path coverage</td>
</tr>
<tr>
<td>Yoo and Harman [23]</td>
<td>Pareto optimal solution</td>
<td>Test suite reduction</td>
<td>Provided Pareto optimal solutions for test suite reduction</td>
</tr>
<tr>
<td>Sivaji et al. [25]</td>
<td>AB-CNS algorithm</td>
<td>Recall value, accuracy, execution time</td>
<td>Reduced test cases using AB-CNS algorithm with improved effectiveness</td>
</tr>
<tr>
<td>Bala et al. [24]</td>
<td>Hybridized technique using harmony search and PSO</td>
<td>Non-functional properties testing</td>
<td>Achieved 100% coverage in less execution time compared to GA, HS, PSO</td>
</tr>
<tr>
<td>Coviello et al. [26]</td>
<td>GASSER technique based on NSGA-II</td>
<td>Statement coverage, fault coverage</td>
<td>Reduced test suite size by maximizing statement coverage, compared to traditional approaches</td>
</tr>
<tr>
<td>Suri et al. [32]</td>
<td>Hybris approach using GA and ABC</td>
<td>Fault coverage</td>
<td>Cover 100% fault with reduction in test suite size</td>
</tr>
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</table>

Table I. Defines the Literature Studies of Various Researchers


Rao et al. developed the TLBO algorithm that is a population-based optimization algorithm used to handle mechanical design optimization issues [15]. However, when solving complex optimization problems, the TLBO algorithm frequently encounters the issue of getting trapped in local optima [14]. Although it has consistently outperformed other evolutionary algorithms in theoretical and practical tests, the impact of control parameters on TLBO’s performance cannot be overlooked [16]. The TLBO algorithm is composed of two distinct phases: the Teacher Phase and the Learner Phase. In the Teacher Phase, the teacher aims to enhance the students’ knowledge. However, the progress of the students is not solely determined by the teacher’s expertise, but also influenced by the mean level of the entire class [15]. Fig. 2 illustrates the working of TLBO.

IV. PROPOSED METHODOLOGY

This section presents a proposed model for test suite minimization problem as shown in Fig. 3. The proposed work is composed of three modules. The pseudo code of proposed work is described in algorithm 1.

We have developed model for test suite minimization problem as illustrated in Fig. 3. There are three components comprise the proposed work. Algorithm 1 defines the pseudo code of proposed work.

Algorithm 1. Hybrid Artificial Bee Colony

1. Begin
2. Define max_gen, max_population
3. Initialize Food source(population)
4. Evaluate fitness function
5. For i=1: max_gen
6. For i=1: max_population
7. Teacher based employee bee phase as shown in Algorithm 2
8. Learner based onlooker bee phase as shown in Algorithm 3
9. Update teacher and mean
10. Memorize the best solution
11. Scout Phase
12. End for
13. End for
14. Find the fault detection capability as shown in Algorithm 4
15. End
A. Dataset Extraction Module

We have taken programs from different sources. As we know minimization techniques are based on adequacy criteria. Statement coverage and fault coverage matrix are extracted from source code.

![Diagram of Dataset Extraction Module](image)

**Fig. 3.** Framework of proposed algorithm.

B. Test Suite Minimization Module

Second module describes the working of hybrid technique. There are two important factors related to heuristic optimization methods. The first is called "exploration," and it entails a comprehensive search of the space while providing a variety of potential solutions at each stage. This factor demonstrates a method's ability to perform well in a global search. Exploitation, or the quality of solutions attained during each cycle, is the other consideration. This metric demonstrates a technique's capability in performing a local search and locating the optimal response close to a solution. These two considerations are in opposition to one another and deserve to be pointed out. Thus, if you prioritize global search over local search (i.e. exploitation), you may end up with a poor final best answer, and vice versa [30]. In the proposed technique two popular soft computing techniques ABC and TLBO are amalgamated with each other. It is developed by incorporating the exploitation capability of TLBO with changes. Both the techniques are complementary to each other.

In the hybrid model we incorporated two significant changes to algorithms. First, instead of employee and onlooker bee phase, teacher and learner phases are embedded into model. ABC algorithm uses the same perturbation for Employee bee and onlooker bee. Therefore, ABC algorithm stagnates from the capability of exploitation. To solve this issue we have embedded the teacher and learner phase of TLBO algorithm into employee and onlooker bee phase with modification in operator.

Second, from the literature we have observed that TLBO and its variants use same approach to update the knowledge of learners in the both phases of TLBO. Every student is unique, and their level of zeal for learning varies [14]. Some students are more interested to gain knowledge on other hand others shows less interest in studies and neglect the knowledge gained from others. As the result of this thought we have divided the learners according to learner performance.

The main aim of this approach is to identify a smaller group of test cases that can cover the maximum statements while still providing the same level of coverage like the original test suite. Additionally, the proposed approach also checks the fault coverage for selected test cases.

1) Population initialization: The algorithm starts with random initialization of food source (test suite) using permutation encoding. In permutation encoding, test cases will not be repeated, and only unique test case can be part of a test suite. Like, TS1= [8, 99, 21, 43, 23, 56]. Set dimension, trail counter and limit.

2) APS: It is essential to assess the quality of all the population's food sources to evaluate the effectiveness of a potential solution provided by that food supply. The objective function allows selecting the best individual that leads to a good solution and validates the process's deviation from its optimization target. The objective of the problem is to reduce the number of test cases while still providing maximum statement coverage.

The average percentage of statement coverage (APSC) formula is defined in (1).

\[
\text{Maximize } \text{APSC} = 1 - \frac{\sum_{i=1}^{n} \text{SCOV}_{ij}}{n \times m} + \frac{1}{2 \times n} \quad (1)
\]

Here, n and m represents the total number of test cases in test suite and statements.

3) Fitness function: Fitness functions are employed in simulations to steer them towards the best possible design solutions.

Fitness function is calculated using (2)

\[
\text{Fit}(f(x)) = \begin{cases} 
\frac{1}{1 + f(x)} & f(x) \geq 0 \\
1 + \text{abs}(f(x)) & f(x) < 0
\end{cases}
\]

Here, f(x) represents the objective function, i.e., calculated in eq. 1.

4) Teacher-based employee bee phase: In the basic TLBO bees positions are updated using eq.3.

\[x_{\text{new}} = x_{\text{old}} + \text{rand}(x_{\text{teacher}} - t_f \times x_{\text{mean}}) \quad (3)\]

Here, t_f represents the teacher factor whose value can be 1 or 2 and is calculated as

\[t_f = \text{round}[1 + \text{rand}(0,1)].\]

Mean(x_{\text{mean}}) can be calculated by using eq. 4

\[x_{\text{mean}} = \frac{x_{\text{np}}}{\Sigma_{i=1}^{n} x_i} \quad (4)\]

Here np is number of bees.
But it suffers from poor diversity issue. To overcome this issue, DE (differential equation) mutation [19] is utilized. Though teacher strategy is hybridized with DE (differential equation) mutation is done to update learner’s position by eq. 5. The steps of teacher based employee bee phase are shown in algorithm 2.

Step 1 Select the best bee as X_best.
Step 2 Calculate mean (X_mean) of all learner bees.
Step 3 Generate the new bee according to DE mutation eq. 5
Step 4 Calculate fitness function for new bees.
Step 5 Accept new bee if it is better than old ones.

5) Tournament selection: Roulette wheel selection is based on a proportionate selection technique in which the best wheel can be chosen or not. We adopted tournaments to solve this problem. A tournament is a match involving several known-sized competitors who are randomly selected to compete. The value of each individual’s fitness is used to select the winner. The size of the tournament is determined by the number of participants.

6) Learner-based onlooker bee phase: As we have already discussed, in basic TLBO no variation exists while gaining knowledge among learners. Therefore, in the learner phase, we have used a factor called learner performance (Lp). Learner performance formulates variation in the best student's and other student's knowledge. If students have knowledge greater than the learner performance value, they will improve their knowledge by interacting with the best one as in equation 6. Otherwise, the criteria for gaining knowledge gain will be the same as basic TLBO as specified in equations 7 and 8. We have tried the value of Lp from [0.7, 1] by trial and error method, out of which 0.8 is the best-suited value. Those students with a learning performance of 0.8 or more will learn from the best learner or teacher.

\[ x_{nbest} = x_{best} + rand(x_{best} - x_{j}) \]  \hspace{1cm} (6)

\[ x_{new,i} = x_{i,old} + rand(x_{best} - x_{i}) + rand(x_{j} - x_{i}) \] \hspace{1cm} (7)

\[ x_{new,i} = x_{i,old} + rand(x_{best} - x_{i}) + rand(x_{j} - x_{i}) \] \hspace{1cm} (8)

The steps to Learner-based onlooker bee phase are described in algorithm 3.

**Algorithm 3 Learner based Onlooker Bee Phase**

1. Begin
2. Initialize food source(population)
3. for each learner \( x_i \)
   4. If \( f(x_i) < f(x) \)
      5. \( f = f(x_i) \)
      6. Update the new bee using equation 6
   7. else
      8. Update the new bee using equation 7
      9. end If
   10. else
      11. Update the new bee using equation 8
      12. end If
   13. Calculate fitness function for new bees.
   14. Accept new bee if it is better than old ones.
   15. end for
16. End

7) Scout phase: When an employee bee’s food source reaches a predetermined limit, it is reclassified as a scout bee. Similar to the worker bee phase, the new random scout bee is formed randomly. If this happens, the previous methods should be updated with the new bee solution.

C. Fault Detection Capability

There are different criteria for minimizing test suites. In the proposed technique, we have selected statement coverage as the criterion. However, the effectiveness of the technique can be checked by detecting losses for other criteria. Since one criterion can affect others, we have evaluated fault-revealing capability of the reduced test suite, as shown in Algorithm 4.

**Algorithm 4. Fault Detection Capability Algorithm**

1. Begin
2. Define Reduced test suite (RTS), Total Faults(TF), Fault matrix (FM)
3. Initialize variable \( F=0 \)
4. For \( i=1: \text{size} \text{ (RTS)} \)
   5. \( a = \text{RTS}[i] \)
   6. For \( j=1: \text{size} \text{ (FM)} \)
      7. If \( \text{FM}[a][j]==1 \) \( F=F+1 \)
   8. End if
   9. End if
10. End for
11. Calculate detected faults by \( ((F/TF)*100) \)
12. End for
13. End
V. EXPERIMENTAL EVALUATION AND RESULTS

In this subsection, we use benchmark programs to evaluate our proposed algorithm. Using Matlab2016a, we performed the experiments on a personal computer with a 2.00 GHz/core (i3) CPU and 4GB of memory. In order to validate the effectiveness of the proposed algorithm, a comparative analysis was conducted between its results and those obtained from the ABC, TLBO, and GA algorithms. Each algorithm was executed 20 times. The evaluation metrics used, the data sets explored, and the outcomes of the experiments are all detailed here.

A. Evaluation Metrics

To rigorously evaluate and compare the effectiveness of our proposed approach with state-of-the-art methods, we leveraged well-established structural coverage measures. These measures have been widely adopted and extensively studied as reliable indicators of a test suite's efficacy. By employing these measures, we aimed to provide a robust performance evaluation of our algorithm. Table II illustrates the evaluation metrics.

B. Dataset

For the validation of the algorithm, we used datasets from different sources [30, 29]. These programs vary in size from small to large, helping to determine whether performance variations are due to test suite size scalability.

C. Computational Analysis and Discussion on Results

In this subsection, we calculated the outcomes of the proposed algorithm and evaluated it in comparison to the other algorithms. Our proposed approach has undergone a comprehensive evaluation, focusing on two key factors: efficiency and effectiveness. Through rigorous testing and analysis, we have thoroughly assessed the performance of our approach in terms of these crucial aspects. The effectiveness of the program is measured by the APRS, APSL, and APFL, while efficiency is measured by the execution cost. Therefore, APCR is used as efficiency metric.

1) Efficiency: The efficiency of the algorithm is assessed by measuring the execution cost required to find the reduced test suite. As we know, the larger the test suite, the higher the computational cost. Reduction of the test cases can help to decrease the total computational cost. Table III illustrates the cost reduction percentage achieved by reducing the test suite size compared to the original cost. As determined from Fig. 4, TLBO performed best for large-sized programs, followed by the proposed algorithm, ABC-TLBO, GA, and ABC. There is a minor difference in the result of the proposed and TLBO algorithms. For small-sized programs, the proposed algorithm performed best. The difference between the proposed and ABC-TLBO algorithms was less for small-sized programs. As evident from Table III, the execution cost difference between algorithms is merely a fraction of a second. Therefore, while weighing the benefits of the algorithm, efficiency plays a relatively small role in algorithm selection. TLBO performed best, followed by the proposed algorithm, ABC-TLBO, GA, and ABC.

2) Effectiveness: Removing unnecessary test cases may affect the fault detection capability. Table IV shows the experimental outcomes of LpABTLO and other algorithms on the minimization problem. It is clear from Fig. 5 that LpABTLO reduces more test cases than other algorithms. LpABTLO reduces 51% of test cases without compromising the fault detection rate. Moreover, as the size of the program and the number of test cases increase, the reduction rate also increases for all the techniques. However, LpABTLO reduction is higher than other techniques. It provides the optimal result. The comparative analysis for statement coverage is presented through Fig. 6. For small-sized programs, the statement coverage of LpABTLO and ABC-TLBO is similar, but ABC lags behind GA and TLBO. ABC-TLBO is the second best after LpABTLO in coverage without any fault loss in small-sized programs. For large-sized programs, LpABTLO and ABC-TLBO attain 96.91% and 96.43%, respectively, while GA, ABC, and TLBO get 94.32%, 92.56%, and 94.51%, respectively. By utilizing statement coverage as a metric for testing, we have also examined the loss in fault detection capabilities resulting from reduced test cases. The percentage of fault-detection loss is depicted in Fig. 7. The figure shows that LpABTLO has the least fault coverage loss compared to other algorithms. GA has the maximum fault coverage loss after ABC. The technique that can reduce the size of the test case, cover maximum statements, and without loss in fault detection capability is the best choice for selection. LpABTLO covers all the conditions. Hence, LpABTLO is superior to other techniques.

<table>
<thead>
<tr>
<th>Metric</th>
<th>Formula</th>
<th>Definition</th>
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</thead>
<tbody>
<tr>
<td>Average percentage of reduced size</td>
<td>APRS = ((OTS – Ored) / OTS) * 100</td>
<td>Percentage of reduced test suite's size compared to the original test suite. Higher value of RSP is better.</td>
</tr>
<tr>
<td>Average percentage of fault detection capability loss</td>
<td>APFL = ((F - Fred) / F) * 100</td>
<td>Quantifies the degree of fault coverage loss in the reduced test suite compared to the original test suite. Lower value of APFL is better.</td>
</tr>
<tr>
<td>Average percentage of statement coverage loss</td>
<td>APSL = ((S - Sred) / S) * 100</td>
<td>Quantifies the degree of statement coverage loss in the reduced test suite compared to the original test suite. Lower value of APSL is better.</td>
</tr>
<tr>
<td>Average percentage of Cost reduction</td>
<td>APCR = ((E - Ered) / E) * 100</td>
<td>Measures the extent of cost reduction achieved during the test suite reduction process.</td>
</tr>
</tbody>
</table>
### TABLE III. Comparison of Cost Reduction of Proposed Approach (LpABTLO) with Other Algorithms

<table>
<thead>
<tr>
<th>Dataset</th>
<th>Algorithms</th>
<th>APCR</th>
</tr>
</thead>
<tbody>
<tr>
<td>Traingle classification Problem(TCP)</td>
<td>LpABTLO</td>
<td>88.433</td>
</tr>
<tr>
<td></td>
<td>GA</td>
<td>85.212</td>
</tr>
<tr>
<td></td>
<td>ABC-TLBO</td>
<td>87.876</td>
</tr>
<tr>
<td></td>
<td>ABC</td>
<td>83.650</td>
</tr>
<tr>
<td></td>
<td>TLBO</td>
<td>84.985</td>
</tr>
<tr>
<td>Quadratic equation(QE)</td>
<td>LpABTLO</td>
<td>89.980</td>
</tr>
<tr>
<td></td>
<td>GA</td>
<td>84.785</td>
</tr>
<tr>
<td></td>
<td>ABC-TLBO</td>
<td>88.675</td>
</tr>
<tr>
<td></td>
<td>ABC</td>
<td>82.490</td>
</tr>
<tr>
<td></td>
<td>TLBO</td>
<td>86.456</td>
</tr>
<tr>
<td>Crossword</td>
<td>LpABTLO</td>
<td>75.700</td>
</tr>
<tr>
<td></td>
<td>GA</td>
<td>74.132</td>
</tr>
<tr>
<td></td>
<td>ABC-TLBO</td>
<td>74.235</td>
</tr>
<tr>
<td></td>
<td>ABC</td>
<td>70.214</td>
</tr>
<tr>
<td></td>
<td>TLBO</td>
<td>76.870</td>
</tr>
<tr>
<td>Freemind</td>
<td>LpABTLO</td>
<td>69.320</td>
</tr>
<tr>
<td></td>
<td>GA</td>
<td>67.875</td>
</tr>
<tr>
<td></td>
<td>ABC-TLBO</td>
<td>69.231</td>
</tr>
<tr>
<td></td>
<td>ABC</td>
<td>65.773</td>
</tr>
<tr>
<td></td>
<td>TLBO</td>
<td>69.750</td>
</tr>
</tbody>
</table>

### TABLE IV. Comparative Study of Algorithms for Different Metrics

<table>
<thead>
<tr>
<th>Program Versions</th>
<th>Algorithms</th>
<th>APSC</th>
<th>APRS</th>
<th>APSL</th>
<th>APFL</th>
</tr>
</thead>
<tbody>
<tr>
<td>Traingle classification Problem(TCP)</td>
<td>LpABTLO</td>
<td>97.56</td>
<td>51.275</td>
<td>0</td>
<td>0</td>
</tr>
<tr>
<td></td>
<td>GA</td>
<td>95.21</td>
<td>46.584</td>
<td>0</td>
<td>1.25</td>
</tr>
<tr>
<td></td>
<td>ABC-TLBO</td>
<td>96.32</td>
<td>50.923</td>
<td>0</td>
<td>0</td>
</tr>
<tr>
<td></td>
<td>ABC</td>
<td>94.83</td>
<td>43.552</td>
<td>0</td>
<td>1.3</td>
</tr>
<tr>
<td></td>
<td>TLBO</td>
<td>95.83</td>
<td>48.237</td>
<td>0</td>
<td>0</td>
</tr>
<tr>
<td>Quadratic equation(QE)</td>
<td>LpABTLO</td>
<td>98.86</td>
<td>55.869</td>
<td>0</td>
<td>0</td>
</tr>
<tr>
<td></td>
<td>GA</td>
<td>96.7</td>
<td>48.538</td>
<td>0</td>
<td>2.31</td>
</tr>
<tr>
<td></td>
<td>ABC-TLBO</td>
<td>97.56</td>
<td>52.512</td>
<td>0</td>
<td>0</td>
</tr>
<tr>
<td></td>
<td>ABC</td>
<td>96.102</td>
<td>44.325</td>
<td>0</td>
<td>1.42</td>
</tr>
<tr>
<td></td>
<td>TLBO</td>
<td>97.20</td>
<td>47.675</td>
<td>0</td>
<td>0</td>
</tr>
<tr>
<td>Crossword</td>
<td>LpABTLO</td>
<td>95.53</td>
<td>65.762</td>
<td>0</td>
<td>.546</td>
</tr>
<tr>
<td></td>
<td>GA</td>
<td>93.76</td>
<td>62.453</td>
<td>.643</td>
<td>1.642</td>
</tr>
<tr>
<td></td>
<td>ABC-TLBO</td>
<td>96.56</td>
<td>66.675</td>
<td>.025</td>
<td>.679</td>
</tr>
<tr>
<td></td>
<td>ABC</td>
<td>92.23</td>
<td>63.218</td>
<td>.854</td>
<td>1.758</td>
</tr>
<tr>
<td></td>
<td>TLBO</td>
<td>94.35</td>
<td>64.77</td>
<td>0</td>
<td>.783</td>
</tr>
<tr>
<td>Freemind</td>
<td>LpABTLO</td>
<td>96.91</td>
<td>68.523</td>
<td>.046</td>
<td>1.641</td>
</tr>
<tr>
<td></td>
<td>GA</td>
<td>94.32</td>
<td>61.768</td>
<td>.875</td>
<td>2.321</td>
</tr>
<tr>
<td></td>
<td>ABC-TLBO</td>
<td>96.43</td>
<td>65.762</td>
<td>.065</td>
<td>1.897</td>
</tr>
<tr>
<td></td>
<td>ABC</td>
<td>92.56</td>
<td>62.605</td>
<td>.947</td>
<td>2.987</td>
</tr>
<tr>
<td></td>
<td>TLBO</td>
<td>94.51</td>
<td>65.543</td>
<td>.43</td>
<td>1.934</td>
</tr>
</tbody>
</table>
Fig. 4. Program wise comparative analysis of algorithms for cost reduction.

Fig. 5. Program wise comparative analysis of algorithms for size reduction.

Fig. 6. Program wise comparative analysis of algorithms for statement coverage.
Regression testing is considered an NP-hard problem. Optimizing methods can be used to solve these issues by identifying the optimal approach. We have proposed a hybrid algorithm with a combination of ABC and TLBO. As the ABC algorithm uses the same perturbation for the Employee bee and onlooker bee, it stagnates from the capability of exploitation. To solve this issue, we have embedded both phases of the TLBO algorithm into the employee and onlooker bee phase with modification in the operator. The algorithm found the results in two steps. Firstly, it removes redundant test cases to have maximum structural coverage. Further, it checks the fault revealing capability loss due to minimization. We have tested the proposed algorithm against ABC, TLBO, and ABC-TLBO on different-sized programs. It has been determined that the proposed algorithm outperforms than the constituent.

VI. CONCLUSION

Regression testing is considered an NP-hard problem. Optimizing methods can be used to solve these issues by identifying the optimal approach. We have proposed a hybrid algorithm with a combination of ABC and TLBO. As the ABC algorithm uses the same perturbation for the Employee bee and onlooker bee, it stagnates from the capability of exploitation. To solve this issue, we have embedded both phases of the TLBO algorithm into the employee and onlooker bee phase with modification in the operator. The algorithm found the results in two steps. Firstly, it removes redundant test cases to have maximum structural coverage. Further, it checks the fault revealing capability loss due to minimization. We have tested the proposed algorithm against ABC, TLBO, and ABC-TLBO on different-sized programs. It has been determined that the proposed algorithm outperforms than the constituent.

REFERENCES


Automated Characterization of Autism Spectrum Disorder using Combined Functional and Structural MRI Analysis

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Abstract—Autism Spectrum Disorders (ASD) are among the most critical health concerns of our time. These disorders typically present challenges in social interaction, communication, and exhibit repetitive behaviors. To diagnose and customize medical treatments for ASD effectively, the development of robust neuroimaging biomarkers is indispensable. Although extensive studies have recently delved into this area, only a handful have explored the differences between ASD and NC. This study aspires to shed light on this relationship by analyzing both structural and functional brain data associated with ASD. We aim to provide an extensive characterization of ASD by combining techniques of structural and functional analysis. The framework we propose is based on analyzing the differences in structural and functional aspects between ASD and development control (DC) subjects. The study leverages a substantial dataset of 1114 T1-weighted structural and functional Magnetic Resonance Imaging comprising 521 individuals with ASD and 593 controls, ranging in age from 5 to 64 years. These subjects are divided into three broad age categories. Utilizing automated labeling, we compute the features from subcortical and cortical regions. Statistical analyses help identify disparities between ASD and DC subjects. Principal Component Analysis (PCA) is employed to select the most discriminative features, which are subsequently used for classifying the two groups via an Artificial Neural Network (ANN) analysis. Our preliminary findings reveal a significant difference in the distribution of all tested features and subcortical regions between ASD subjects and DC subjects. Through our work, we contribute towards an enhanced understanding of ASD, potentially paving the way for future research and therapeutic interventions.

Keywords—Autism spectrum disorder (ASD); Magnetic Resonance Imaging (MRI); functional Magnetic Resonance Imaging (fMRI); Artificial Neural Network (ANN)

I. INTRODUCTION

The nature of Autism Spectrum Disorder (ASD) is multilayered and intriguing, with its diverse manifestations affecting various aspects of an individual's life[1]. From social communication and behavioral patterns to intellectual abilities, ASD presents an array of symptoms that make it a unique neurodevelopmental disorder. An estimated one in every 54 children is affected by ASD globally, underscoring its significant prevalence [2]. Despite this high occurrence, the causes of ASD remain enigmatic. Multiple genetic and environmental risk factors have been identified, yet they only partially explain the incidence of ASD. Consequently, rigorous research efforts continue to seek a deeper understanding of ASD's intricate etiology and neurobiology.

Advancements in the field of neuroimaging have significantly contributed to our understanding of neurodevelopmental disorders such as ASD. From purely descriptive studies of the brain's anatomy, neuroimaging research has evolved to explore the profound neurodevelopmental alterations that occur in these conditions. Modern imaging techniques such as Structural Magnetic Resonance Imaging (sMRI) and functional Magnetic Resonance Imaging (fMRI), combined with state-of-the-art computational tools and machine learning algorithms, have provided unprecedented insights into the structural and functional anomalies associated with neurodevelopmental disorders. This development has brought about a paradigm shift in our understanding of these conditions, with a newfound focus on identifying potential biomarkers that can serve as quantitative indicators of specific brain abnormalities.

MRI is an indispensable tool in ASD research, enabling non-invasive imaging of brain structure, connectivity, and function. Structural MRI studies have identified various alterations in brain regions involved in social communication, language processing, executive functions, and sensory integration in individuals with ASD. Functional MRI, on the other hand, has been pivotal in uncovering disrupted functional connectivity, particularly in regions responsible for social cognition and emotion processing. MRI's widespread utilization in ASD research has undoubtedly been instrumental in enhancing our understanding of the disorder's neurobiology.

In-depth analysis of specific neuroanatomical regions has revealed that ASD is associated with certain alterations in brain structure, contributing to the varied cognitive and behavioral phenotypes observed in these individuals. Volumetric analysis of specific brain regions has reported differences in brain volumes between individuals with ASD and typically developing individuals. Similarly, cortical thickness analysis has identified disparities in the cerebral cortex's thickness between these two groups. The integration of these structural findings with functional studies has resulted in a holistic
understanding of the neurobiological underpinnings of ASD (Appendix A)

Our research aims to extend these efforts by considering all the neuroanatomical regions implicated in ASD. Utilizing a sophisticated automated processing pipeline, we perform a comprehensive analysis of both 3D volumetric and functional brain regions. This inclusive approach ensures that we capture the full range of structural and functional abnormalities associated with ASD. Our work incorporates texture features derived from global descriptors and local textural features for the structural analysis, combined with statistical and temporal features for the functional analysis. This holistic approach allows us to identify unique volumetric and functional signatures of ASD, with the potential to contribute significantly to early detection, diagnosis, and monitoring of the disorder.

The organization of this paper is as follows: Section II delineates the primary methodology used in our study, encompassing both structural and functional analyses. In Section III, we present the outcomes of our research, along with a thorough discussion on the effectiveness and implications of our proposed approach. The limitations of our study are discussed in Section IV. Finally, we draw conclusions from our findings in Section V.

II. MATERIALS AND METHODS

This chapter elaborates on the methods utilized in our study to distinguish autism using MRI and fMRI data. Initially, we detail the preprocessing steps for data standardization. Then, we describe how significant features were extracted from the imaging data. Finally, we explain the classification techniques used to differentiate between autistic and neurotypical individuals. This approach allows us to identify potential autism biomarkers and understand the underlying neural mechanisms. You will find the pipeline of our structural method in the Fig. 1.

![Pipeline of the structural analysis methods for the characterization of ASD.](image)

**A. Structural Analysis**

- **Preprocessing of sMRI**

  MRI registration is a key step in neuroimaging studies, aligning multiple MRI images for spatial correspondence. Commonly, the Montreal Neurological Institute space (MNI 152 space) is used as a reference for this alignment. This process involves an initial reorientation of MRI scans, with anatomical labels defined to correct any discrepancies. Then, we perform a skull stripping or brain extraction step to isolate brain tissue from non-brain elements, using methods like the Optimal Surface Thresholding (OST). This step is crucial in reducing potential artifacts in subsequent MRI analyses (Fig. 2).

![An example of the brain MRI registration with an original MR image of brain (a) and registered image (b).](image)

- **Segmentation of the Regions of Interest**

  In our research, we apply an automatic method for identifying and segmenting brain regions of interest (Appendix B) relevant to ASD using both probabilistic and anatomical models. A Bayesian probabilistic approach, based on Markov modeling, is used for accurate identification and segmentation of ROIs. This approach incorporates prior knowledge about brain structures, facilitating accurate ROI identification. Markov random fields (MRFs) account for spatial dependencies between neighboring elements in the image, enforcing spatial continuity and reducing the impact of noise[3].

  \[
P(A|B) = \frac{P(B|A) * P(A)}{P(B)} \quad (1)
\]

  Where:

  P(A | B) is the posterior probability of event A occurring, given that event B has occurred.

  P(B | A) is the likelihood of event B occurring, given that event A has occurred.

  P(A) is the prior probability of event A occurring.

  P(B) is the marginal probability of event B occurring.

  The process involves formulating a model incorporating Bayesian probability theory and Markov models, estimating model parameters using the maximum likelihood estimation (MLE), and iteratively refining the model using the Expectation-Maximization (EM) algorithm[4]. The final step is image classification and segmentation using maximum a posteriori (MAP) estimation, assigning each pixel or voxel to a specific ROI based on the most probable assignment given the observed data and the model parameters.

  \[
  MLE(\theta) = \arg\max P(X|\theta) \quad (2)
  \]

  where \( \theta \) represents the model parameters, \( X \) denotes the observed data, and \( P(X | \theta) \) is the likelihood of the data given the parameters. During this process, we account for the natural variability in the image data and consider the spatial dependencies between neighboring elements[5].

  \[
  MAP(\theta) = \arg\max P(\theta|X) \quad (3)
  \]
where $\theta$ represents the model parameters, $X$ denotes the observed data, and $P(\theta \mid X)$ is the posterior probability of the parameters given the data (result in Fig. 3).

！Fig. 3. Segmentation steps of left hippocampus.

- 3D mesh construction

Medical imaging and computer vision often require the transformation of raw image data into a format more suitable for analysis, and 3D mesh models fill this need. The process begins with image segmentation, followed by surface extraction, and finally, mesh generation. The surface of the target object is extracted using the Marching Cubes algorithm, which divides the 3D volume into a grid of small cubes and forms a polygon configuration within each cube based on whether corners lie above or below a certain threshold. This creates an interconnected surface approximating the boundary of the target structure.

Next, mesh generation connects points on the surface to form polygons in a process known as triangulation, resulting in a 3D mesh composed of numerous triangles. This mesh model, which represents the 3D shape of the target object, provides a detailed and accurate foundation for further analysis, visualization, or computation (result in Fig. 4).

！Fig. 4. 3D Meshes structures.

- Feature extraction of sMRI

The analysis of anatomical MRI data involves a pivotal phase known as feature extraction. This step translates complex 3D images into measurable, actionable information by identifying and quantifying various properties of the structures within the images. This process plays an integral role in exploring potential biomarkers for Autism Spectrum Disorder (ASD). Feature extraction focuses on two primary categories: geometric and texture features.

Geometric features provide information about the shape and structure of areas within the brain, helping to understand physical alterations related to ASD. Texture features, on the other hand, capture intensity variations within a region, revealing patterns that might indicate different tissue types or states, and possibly reflecting microstructural properties of the brain regions linked to neurodevelopmental changes in ASD.

In this study, multiple mathematical and statistical methods are applied to calculate these features. Riemannian geometry is used to understand the intrinsic curvature of surfaces and spaces. It allows for the computation of the area, volume, isoperimetric ratio, convexity ratio of the surface and volume, and Gaussian and Mean curvatures.

Haralick texture features, or Gray Level Co-occurrence Matrix (GLCM) features, provide a statistical snapshot of the texture. Several statistical measures are derived from the GLCM, including the Angular Second Moment (or Energy), Contrast, Correlation, Variance, Inverse Difference Moment (or Homogeneity), Entropy, Sum Average, Sum Variance, Sum Entropy, Difference Variance, and Difference Entropy (Annex B).

By combining geometric and texture feature extraction, a comprehensive picture of the structural changes associated with ASD is captured. This approach goes beyond traditional measures and explores a broader spectrum of potential biomarkers. Through statistical analysis of these features, patterns can be identified that could serve as reliable biomarkers for ASD, enhancing our understanding of this complex condition and potentially informing future diagnostic and therapeutic strategies.

B. Functional Analysis

- Preprocessing of fMRI

The preprocessing of fMRI data begins with obtaining raw scan data. This data then undergoes several steps to enhance its quality and reliability (Fig. 5). These steps include:

Motion Correction: This process is used to minimize the effects of head movement during the scan. The process involves realigning all volumes acquired during an fMRI scan to a reference volume to eliminate motion-related artifacts.

Slice Timing Correction: This step compensates for the time difference between the acquisitions of different slices in each volume. The correction aims to align the signal from all slices as if they were acquired at the same time, improving the accuracy of the data.

Registration: This process aligns different sets of data into one common space. It involves within-subject registration (aligning images from the same individual) and between-subject registration (aligning images from different individuals to a standard template).

Spatial Normalization: This involves transforming individual brain images to fit into a standard template, enabling group analyses and inter-subject comparisons.
Spatial Smoothing: This process improves the signal-to-noise ratio by averaging a voxel's signal with the signal of surrounding voxels. It also helps mitigate differences in functional neuroanatomy across participants and satisfies certain statistical assumptions (result in Fig. 6).

- Segmentation of the Regions of Interest

In our autism study, we furthered our fMRI data analysis by segmenting regions of interest (ROIs), focusing specifically on brain regions previously implicated in autism spectrum disorder (ASD). This approach allowed us to investigate how structural abnormalities in these regions may influence functional connectivity patterns and contribute to ASD's characteristic features (Fig. 7).

To achieve this, we utilized Atlas-based segmentation, employing the Harvard-Oxford atlas for its detailed labeling of cortical and subcortical brain structures. After selecting our ROIs from this atlas, we applied the labels to our fMRI data through a process known as label propagation. The segmentation information from multiple atlas images, when used, was fused to create the final segmentation result. This thorough process ensured accurate segmentation and paved the way for our subsequent ASD analysis and classification.

- Time Series Extraction

After defining the regions of interest (ROIs) in our fMRI data, we conducted time-series extraction, a fundamental step in functional connectivity analysis. This process involves gathering the intensity values for each voxel within the ROI across each time point in the fMRI series. By averaging the signal change over time across all voxels within each ROI, we generated a single time-series for each region, enabling us to investigate various patterns of brain activity over time (Fig. 8).

- Feature extraction of fMRI

In our analysis of fMRI data, we performed feature extraction, a process to convert raw, high-dimensional time series data into a simpler, lower-dimensional format. We used the Python library tsfresh to extract 150 statistical and temporal features from the time series data.

Statistical features provided a summarized description of the variation in signal intensity over time. They included measures like mean, median, variance, standard deviation, and others, helping to quantify the properties of the signal.

Temporal features captured how brain responses changed over the course of the experiment. They included Autoregression coefficients, trend features, and Fourier coefficients among others, allowing us to uncover patterns in brain activity and comprehend the interactions between different brain regions.

Through feature extraction, we transformed complex fMRI data into a more manageable format without losing vital information, preparing it for processing by machine learning models.

C. Multi-Modal Imaging Fusion

In this section of our research, we'll take the critical step of fusing the feature sets derived from both structural and functional imaging data. This combined feature set will be utilized to provide a comprehensive perspective on the neuroanatomy and functionality of the key brain regions implicated in Autism Spectrum Disorder (ASD). This integrated analysis could provide more robust and meaningful insights into the neurobiological underpinnings of ASD.

- Dimensionality reduction

Upon combining the features extracted from both the structural and functional data, we find ourselves dealing with an extremely high-dimensional feature set. This poses a significant challenge in the context of machine learning, potentially leading to overfitting, difficulty in interpretation, and a spike in computational demands. This is where feature selection comes into play.
Feature selection allows us to filter out less informative features and focus on those that contribute the most to our model's predictive power. We utilized Principal Component Analysis (PCA) as a strategic choice for our feature selection process. PCA is particularly advantageous for our high-dimensional data as it allows us to reduce the dimensionality while retaining as much information as possible. PCA identifies the directions (principal components) in which the data varies the most and transforms the original, high-dimensional data into a lower-dimensional set of new features. Fig. 9. These features are linear combinations of the original ones, selected in such a way that they are uncorrelated and retain the maximum amount of variance from the original data.

In our study, we set the PCA to retain 95% of the variance. This criterion led to the selection of 30 principal components which are sufficient to retain 95% of the information from the original features. The resulting dataset, composed of 30 uncorrelated principal components, still captures the major patterns and structures in our original high-dimensional data, making it a more manageable and effective input for our subsequent machine learning model. Thus, the application of PCA serves as a powerful step in preparing our feature set for the final stages of ASD diagnosis.

Classification

Classification is essential in diagnosing ASD using machine learning. We've chosen to employ an Artificial Neural Network (ANN), a model that imitates the human brain's structure and function, for this task. ANNs consist of interconnected neurons in layers: an input layer for receiving data, hidden layers for data processing, and an output layer for final results. ANNs 'learn' by adjusting inter-neuron connections through backpropagation, thereby minimizing the difference between predicted and actual outputs. Their ability to model complex, non-linear relationships and learn intricate patterns make ANNs an effective tool for accurate ASD classification, given enough data and training time.

In our study, we applied an ANN for ASD classification, guided by an intricate process. Firstly, we initiate feedforward computation, where the input is sequentially processed through each layer of the network using a sigmoid activation function. The final prediction is made at the output layer. We then employ a loss function, specifically Mean Squared Error (MSE), to quantify the discrepancy between the network's prediction and the actual value[6].

\[
MSE = \frac{1}{N} \sum_{i=1}^{N} (y_i - \hat{y}_i)^2 \quad (4)
\]

Where:
- \( N \) is the number of samples
- \( y_i \) is the actual value
- \( \hat{y}_i \) is the predicted value

The loss function thereby indicates the error of the network. Backpropagation, a method of calculating the loss function's gradient with respect to the network's weights and biases, follows. It commences from the output layer, moving backward through the hidden layers. The network's weights and biases are then updated by subtracting a fraction of the gradient, dictated by the learning rate, a key hyperparameter. This whole process is repeated for each batch of data in the training dataset for a set number of epochs, gradually adjusting the network's weights and biases to minimize the loss function, and improving prediction accuracy. We customized certain hyperparameters (Appendix C, Table VII), like the learning rate and the structure of hidden layers, for our specific application to ensure optimal performance. By fine-tuning these hyperparameters, we could better capture complex patterns in ASD data, thereby increasing the accuracy of our ASD characterization (Fig. 10).

III. RESULTS AND DISCUSSION

In this section, we initially outline the database utilized for our investigation in the first part. This is followed by a description of the tuning process for the ANN in Section B. Subsequently, we delve into an evaluation of the proposed method's performance. Lastly, we provide a comparative analysis with a few prevalent techniques from the established standards.

A. Database Description

The non-invasive imaging technique, MRI is pivotal to our study for its ability to generate detailed, high-resolution images of the brain. Our investigation uses data from the Autism Brain Imaging Data Exchange (ABIDE), a collaborative initiative that curates diverse MRI datasets globally for autism research. This collaboration ensures a robust dataset that offers comprehensive insight into autism's structural patterns.

ABIDE provides a large-scale collection of both structural and functional MRI datasets, improving our understanding of the neural mechanisms underlying ASD. With over 1200 datasets from more than 24 international brain imaging laboratories, these collections afford a rich assortment of data for our analyses.

In our work, the ABIDE data is categorized into three distinct age groups - early childhood (1-9 years), late childhood and adolescence (10-25 years), and adulthood (above 25...
years), each reflecting a different stage of brain development, Table I. This classification allows us to examine autism's manifestation and evolution throughout life stages, offering a nuanced understanding of ASD's progression and neurological implications.

<table>
<thead>
<tr>
<th>1st range</th>
<th>2nd range</th>
<th>3rd range</th>
</tr>
</thead>
<tbody>
<tr>
<td>The postnatal development of the human brain.</td>
<td>The brain reaches its adult size.</td>
<td>The brain reaches its full size.</td>
</tr>
<tr>
<td>5-9 years old</td>
<td>10-25 years old</td>
<td>+25 years old</td>
</tr>
<tr>
<td>646 patients</td>
<td>218 patients</td>
<td>148 patient</td>
</tr>
<tr>
<td>315 ASD</td>
<td>331 NC</td>
<td>154 ASD</td>
</tr>
</tbody>
</table>

B. Machine Learning Evaluation

- k-fold cross-validation

Our study applies probabilistic learning methods to optimize neural networks, dividing our data into training and testing sets for reliable model training and validation. The training set is used to teach the model to recognize data patterns, while the unseen testing set assesses how well it generalizes these patterns. We employ k-fold cross-validation to bolster the reliability of our results and avoid overfitting. This involves splitting the training data into 'k' subsets and training the model 'k' times, each time using a different subset for validation (Fig. 11).

| Fig. 11. How the dataset was split. |

This provides 'k' models and performance estimates which can be averaged to offer a more reliable measure of performance. Specifically, we use 4-fold cross-validation, partitioning our data into four subsets. For each training cycle, 80% of the data trains the model, while the remaining 20% validates it. The model's performance is then assessed using the validation set, and quantified using metrics such as accuracy, precision, and recall. By averaging performance across four iterations, we gain a robust measure of the model's predictive capabilities, mitigating the risk of overfitting and providing a realistic evaluation of the model's potential real-world performance. This approach ensures the robustness, reliability, and applicability of our model in studying autism spectrum disorders.

- Performance evaluation

In our study, we use an ANN classifier to analyze features derived from structural and functional MRI data. The analysis reveals differences in specific brain regions when compared to a control group[7]. The model's performance, evaluated using the identified feature sets, is detailed in Table II, which demonstrate metrics like accuracy, sensitivity, and specificity.

Accuracy is a simple performance metric, calculated as the ratio of correct model predictions to total predictions. Ranging between 0 and 1, a score of 1 denotes a perfect model. We use the equation:

\[
\text{Accuracy} = \frac{TP + TN}{TP + FP + FN + TN}
\]

where TP, TN, FP, and FN represent true positives, true negatives, false positives, and false negatives, respectively.

Sensitivity, or recall, measures the proportion of actual positive cases correctly identified by the model. Calculated as:

\[
\text{Sensitivity} = \frac{TP}{TP + FN}
\]

It quantifies the model's ability to detect all relevant instances.

Specificity evaluates the model's performance regarding negative cases. It represents the proportion of true negative cases correctly identified by the model, computed as:

\[
\text{Specificity} = \frac{TN}{TN + FP}
\]

By using these three metrics to evaluate our model, we ensure that it is well-rounded, accurately predicting both positive and negative cases. This makes it a potentially valuable tool in characterizing autism.

- Results

In our study, we evaluate the effectiveness of an ANN in differentiating between individuals diagnosed with ASD and control subjects across various age groups. This evaluation employs a fusion of structural and functional analyses, a crucial approach for comprehensively understanding the relationships within the brain's structure-function framework. We base this assessment on a combination of geometric, volumetric, temporal, and statistical features, the outcomes of which are represented in Tables II, III, and IV.

- Discussion

In this research, we utilize an ANN to differentiate individuals diagnosed with ASD from control subjects across various stages of life: early childhood, late childhood and adolescence, and adulthood. Both structural and functional characteristics of the brain are examined to provide a comprehensive view of how our model performs. By integrating both structural and functional neural features, our classifier demonstrated robust performance across all developmental stages. Despite a decline in accuracy with increasing age, the integrated approach provided valuable insights, highlighting the relevance of a comprehensive brain analysis in diagnosing ASD.
The combined analysis in the late childhood and adolescence stage maintains high accuracy levels, with the right amygdala again leading with an accuracy of 85.1%. The accuracy levels of WB and GM are 73.0% and 72.7%, respectively. While the accuracy rates have declined slightly from the early childhood stage, the fused ROI result still maintains a solid accuracy of 86.2%.

The final findings underscore the complexity of ASD and the importance of a multifaceted approach to brain data analysis for accurate ASD identification. The research provides a promising pathway for the future of ASD detection and diagnosis, and potentially for other neurodevelopmental and neurodegenerative disorders, through the use of ANN classifiers. Our methodology, which combines structural and functional brain properties, could serve as a valuable tool in the development of Computer-Aided Diagnosis systems, particularly for neurodegenerative diseases like Alzheimer's and Parkinson's, which also involve significant changes in the brain.

- Comparison analysis

In the table below, we present a comparison of various prior studies that have focused on the characterization of ASD by employing both structural and functional MRI. This comparative analysis aims to provide a broader perspective on how our approach to ASD classification using an ANN stands in relation to previous research efforts.

Each study listed in the table has contributed significantly to our current understanding of ASD's neurological underpinnings. However, the methodologies employed, the specific features extracted, and the performance metrics achieved vary from one study to another. Some researchers have concentrated more on the structural aspects of the brain (Table V), while others have leaned towards functional analysis (Table VI).
<table>
<thead>
<tr>
<th>Study</th>
<th>Participants Description</th>
<th>Data Description</th>
<th>Features Description</th>
<th>Machine learning method</th>
<th>Accuracy</th>
</tr>
</thead>
<tbody>
<tr>
<td>Seyedmehdi Payavvash et al.[8]</td>
<td>14 with ASD 33 typically developing children</td>
<td>The UCSF Sensory Neurodevelopment and Autism Program clinical sites and research database</td>
<td>Tract-based average TI/connectome metrics</td>
<td>Support vector machines</td>
<td>75.3%</td>
</tr>
<tr>
<td>Ahmad Chaddad et al.[10]</td>
<td>34 ASD 30 DC</td>
<td>The open database Autism Brain Imaging Data Exchange (ABIDE).</td>
<td>11 features derived from Hippocampus Amygdala</td>
<td>Support vector machines</td>
<td>67.85%</td>
</tr>
<tr>
<td>Ayşe Demirhan et al.[11]</td>
<td>1390 subjects</td>
<td>Five different datasets : OASIS, ABIDE, COBRE, ADHD and MCIC</td>
<td>Volumes and thickness of anatomical structures</td>
<td>Support vector machines</td>
<td>83%</td>
</tr>
<tr>
<td>Pinaya et al.[12]</td>
<td>HC=105 ASD=83</td>
<td>Autism Brain Imaging Data Exchange (ABIDE) data set</td>
<td>ROI based volumetric measurements</td>
<td>Deep autoencoder</td>
<td>63.9%</td>
</tr>
<tr>
<td>Ferrari et al.[13]</td>
<td>HC=1166 ASD=1060</td>
<td>T1w from ABIDE I and ABIDE II</td>
<td>Brain morphometric features</td>
<td>Deep autoencoder</td>
<td>79%</td>
</tr>
<tr>
<td>Qureshi, M.N.,et al.[14]</td>
<td>1000 subjects</td>
<td>Attention deficit hyperactivity disorder ADHD-200 dataset of patients and healthy children.</td>
<td>the cortical thickness measures</td>
<td>Support vector machines</td>
<td>76.19%</td>
</tr>
<tr>
<td>Xiao, X.et al[15]</td>
<td>46 ASD 39 DC</td>
<td>Child Mental Health Research Center of Nanjing Brain Hospital</td>
<td>cortical-thickness measurement surface-based morphometry</td>
<td>RF NB SVM</td>
<td>75.6% 80.9% 80%</td>
</tr>
<tr>
<td>Sina Ghiassian, et al.[16]</td>
<td>490 HC 279 ASD</td>
<td>Attention deficit hyperactivity disorder ADHD-200 dataset of patients and healthy children.</td>
<td>HOG feature extraction</td>
<td>Support vector machines</td>
<td>69.6%</td>
</tr>
<tr>
<td>Dennis Dimond et al.[17]</td>
<td>27 ASD 31 TD</td>
<td>Participant databases at the Alberta Children’s Hospital</td>
<td>intra-cranial volume</td>
<td>Artificial Neural Network</td>
<td>77%</td>
</tr>
<tr>
<td>Bhashkar Sen, et al.[18]</td>
<td>ADHD: 491 HC et 279 ASD ABIDE: 573 HC et 538 ASD</td>
<td>ADHD-200 (including 8 sites) ABIDE (including 17 sites)</td>
<td>3-D texture based and independent component analysis</td>
<td>Linear support vector machine classifier</td>
<td>ADHD 0.64% ABIDE 0.62%</td>
</tr>
<tr>
<td>Calderoni et al.[19]</td>
<td>HC=19 ASD=38</td>
<td>T1w</td>
<td>voxel-based morphometry (VBM)</td>
<td>Support vector machines</td>
<td>80%</td>
</tr>
<tr>
<td>Gori et al[20]</td>
<td>HC=20 ASD=21</td>
<td>T1w</td>
<td>Regional morphological features</td>
<td>Support vector machines</td>
<td>74%</td>
</tr>
<tr>
<td>Hossein Shahamat et al.[21]</td>
<td>403 ASD 468 TC</td>
<td>Autism Brain Imaging Data Exchange (ABIDE) data set: 1112 datasets</td>
<td>Convolutional neural network (CNN) models</td>
<td>Convolutional neural network (CNN) models</td>
<td>70%</td>
</tr>
<tr>
<td>Guannan Li et al.[22]</td>
<td>170 TC 106 ASD</td>
<td>276 subjects from National Database for Autism Research (NDAR)</td>
<td>3D convolutional neural network (CNN) models</td>
<td>3D convolutional neural network (CNN) models</td>
<td>0.7624</td>
</tr>
<tr>
<td>Fengkai Ke et al.[23]</td>
<td>ADHD: 40 HC 33 ASD ABIDE: 573 HC 538 ASD</td>
<td>The first dataset was collected by the Yonsei University College of Medicine (YUM) + the second was ABIDE dataset</td>
<td>3D CNN</td>
<td>3D CNN</td>
<td>0.64</td>
</tr>
</tbody>
</table>
This comparison underscores the diverse approaches researchers have taken to tackle ASD classification using MRI data. Our research's contribution lies in its unique fusion of both structural and functional analyses across different age groups, which provides a more holistic understanding of ASD's neurological characteristics. The table below offers a succinct summary of these various studies, illuminating the varying methodologies and results that have been achieved in the past.

### IV. STUDY LIMITATIONS

Despite the promising findings, the current study is not without its limitations. Primarily, our work relies on the Autism Brain Imaging Data Exchange (ABIDE) which, while being a rich public dataset, has its limitations. The datasets are collected from 521 individuals with ASD and 593 controls with ages ranging from five to 64 years. However, this is a comparatively small dataset for making definitive classifications. To expand upon the insights provided by this study, future research should aim to collect data from a larger group of preschoolers with ASD to further explore brain development during early childhood.

Secondly, while we acknowledge the gender disparity in ASD diagnoses, with males being more frequently diagnosed than females, this issue has been generally overlooked in ASD imaging literature due to the significantly higher prevalence of ASD in males. In the ABIDE dataset, the number of datasets from females is limited (65 datasets, encompassing both ASD and control subjects), making a thorough analysis of gender influences challenging. Despite the potential of Artificial Neural Networks (ANNs), they have inherent limitations. Their 'black-box' nature makes it hard to understand how they arrive at specific predictions, leading to challenges in interpretability. Furthermore, ANNs can sometimes overfit high-dimensional data, especially with limited samples, leading to poor performance on unseen data. To address these issues, future work could explore methods for enhancing interpretability and preventing overfitting.

While these limitations pose challenges, they also highlight potential directions for future research. We need larger, more extensive datasets and more sophisticated analysis methods to better understand the complex neurological underpinnings of ASD.
diverse samples and more robust methods to handle high-dimensional data for more accurate and generalizable results. In addition, research efforts should not neglect gender disparities in ASD and strive for greater representation in imaging studies. By addressing these issues, we can further refine our understanding of ASD’s neurobiological underpinnings and improve diagnostic strategies.

V. CONCLUSION

In conclusion, our study has successfully highlighted the significance and value of employing an integrated, multimodal approach in characterizing and diagnosing ASD across various developmental stages. The results have underscored the vital role played by specific brain regions such as the Right Amygdala, Left Amygdala, Right Corpus Callosum, Left Putamen, and Right Hippocampus in ASD classification. Furthermore, the distinctive combination of both structural and functional neural features in our ANN classifier has proven to be a powerful tool in capturing the complex neurobiological nuances of ASD.

While the classifier demonstrated robust performance across all developmental stages, particularly in early childhood, it also unveiled the increasing complexity of ASD diagnosis in later life stages. This highlights the intricate interplay between ASD symptomatology, brain development, and aging. However, even amidst these challenges, the classifier continued to provide valuable insights and significant accuracy scores, asserting the value of a comprehensive brain analysis approach for ASD diagnosis.

The fusion of results from structural and functional analyses created a more comprehensive model for ASD classification, resulting in a more robust and accurate representation of the disorder. This integrated approach produced promising results, especially in the domain of early ASD diagnosis. The highest diagnostic accuracy achieved in our study reached 90.1%, signifying the potential of a multimodal approach that captures a more complete picture of ASD’s neurobiological underpinnings.

Our study contributes significantly to the field by providing a broader understanding of ASD. By integrating structural and functional aspects of brain imaging data, we enhance our ability to identify and diagnose the disorder accurately. Our research not only paves the way for further advancements in early detection and personalized treatment strategies for individuals with ASD but also sets the stage for this methodology to serve as a Computer-Aided Diagnosis tool for the detection of other neurodegenerative diseases such as Alzheimer’s and Parkinson’s.

Despite some limitations, particularly regarding the decline in accuracy with increasing age, our study reaffirms the need for a comprehensive, multidimensional analysis of brain data for accurate ASD identification. It also suggests areas for future research to enhance the understanding and diagnosis of ASD, potentially through the development of more sophisticated feature extraction or classification methods.

In essence, this research demonstrates the potential of a comprehensive, multimodal neuroimaging approach, combined with advanced machine learning techniques, to improve early detection and understanding of ASD, thereby paving the way for more effective intervention strategies. It lays a solid foundation for similar techniques to be used for other neurodevelopmental and neurodegenerative diseases, offering a significant tool for future diagnostic strategies.

REFERENCES


APPENDIX A

In this part, all regions that were mentioned in the state of the art and were related to ASD were cited below.

White matter: studies have shown that white matter disease is associated with Autism through some physical and emotional symptoms such as balance problems, falls, depression, and difficulty of multitasking activities such walking and talking[37].

Gray matter: Gray matter is involved in muscle control, and sensory perception such as seeing and hearing, speech, self-control, emotions, memory, and decision-making. Researchers found that children diagnosed with autism have more abnormality in gray matter[38].

Nucleus accumbens: The nucleus Accumbens core is involved in the cognitive processing of motor function related to reinforcing slow-wave sleep to regulate reward-motivated behaviors. Some studies related this subcortical region with ASD [39].

Amygdala: the amygdala is related to emotional learning and behavior. Many studies demonstrate that amygdale texture features can be used to extract biomarkers for the characterization of ASD purposes [40].

Corpus callosum: Corpus Callosum the largest white matter structure in the brain. It is involved in the interhemispheric transfer of information and integrates motor, sensory, and cognitive performances. Many comparative studies show that it develops differently in children with autism [41].

Hippocampal: the two most influential theories for hippocampal function are related to space and memory. It is used by the brain for mapping layouts of the environment [40].

Pallidum: Pallidum is a part of the subcortical nervous circuits involved, in motor skills, and, in particular, in the control of posture. It is also associated to non-motor functions (e.g. cognition, emotions, etc...).

Thalamus: Many previous studies shows that the thalamus play a key role in autism. It may regulate social behavior and it is considered as a relay station that merely passes sensory information to the cerebral cortex.

Putsamen: the role the putamen is to regulate movements. It employs GABA, acetylcholine, and encephalin to perform its functions.

Caudate nucleus: the caudate nucleus is the subcortical region that controls learning, specifically the storing and processing of memories. Studies show that Autism causes a different development of ASD.

Brainstem: The brain stem is the part of the brain that connects the cerebrum with the spinal cord. The review of the literature suggests that developmental alterations of the brainstem could have potential cascading effects on cortical and cerebellar formation, ultimately leading to ASD symptoms.

APPENDIX B

- Riemannian geometry

Area: The area of a two-dimensional figure or shape is the quantity that expresses the extent of the figure or shape in the plane. Here in our work, we used (eq 8) to calculate the whole region area by applying the summation of all tetrahedrons areas.

$$A = \sum_{n=1}^{4} \frac{1}{2n} \cdot h$$ (8)

Where is the base and is the height of the triangle.

Volume: The volume is a closed surface that encloses a certain amount of three-dimensional space. Here in our work, we used (eq 9) to calculate the whole region volume by applying the summation of all tetrahedrons volumes.

$$V = \sum_{n=1}^{6} \frac{1}{12} \cdot a^3$$ (9)
Where a is the triangle base.

Isoperimetric ratio: Iso-ratio depends on the volume and the surface. It is initially the study of the properties of the geometric shapes of the plane (eq 10).

\[ IPR = \frac{A}{V^{\frac{1}{2}}} \]  

(10)

Where A is the triangle area and V is the Volume.

Convexity ratio of the surface: The convex area of an object is the area of the convex hull that encloses the object (eq 11).

\[ CRS = \frac{A}{A(CH)} \]  

(11)

Where A is the triangle area and A (CH) is the convex hull depending on area.

Convexity ratio of the volume: The convex volume of an object is the volume of the convex hull that encloses the object (eq 12).

\[ CRV = \frac{V}{V(CH)} \]  

(12)

Where V is the triangle volume and V (CH) is the convex hull depending on volume.

Gaussian curvature: The Gaussian curvature is defined at a point of a surface contained in Euclidean space as the product of the two main curvatures (eq 13).

\[ K = k_1k_2 \]  

(13)

Where k1 and k2 are principal curvatures.

Mean curvature: the mean curvature of a surface is called the mean of the minimum and maximum curvatures (eq 14).

\[ H = \frac{1}{2}(k_1 + k_2) \]  

(14)

Where k1 and k2 are principal curvatures.

- Harlicks texture descriptors

  The second standard L2N: The second standard measures the length common to all representations of a vector in an affine space (eq 15).

\[ |X| = \sqrt{\sum_{k=1}^{n} |x_k|^2} \]  

(15)

Where |X| is the vector norm and |x_k| is the complex modulus.

Means: Mean is the small mean values indicating coarse texture having a grain size equal to or larger than the magnitude of the displacement vector (eq 16).

\[ Mean = \frac{\sum_{i=0}^{N-1} \sum_{j=0}^{N-1} Mc(i,j)}{N^2} \]  

(16)

Where N is the number of gray levels, Mc is the image matrix.

Contrast: The contrast feature is a measure of the image contrast or the number of local variations present in an image (eq 17).

\[ Contrast = \frac{\sum_{i=0}^{N-1} \sum_{j=0}^{N-1} (i - j)^2 Mc(i,j)}{N^2} \]  

(17)

Where N is the number of gray levels, Mc is the image matrix.

Angular Second Moment: Angular Second Moment and Uniformity, also called Energy, which is a measure of textural Uniformity of an image (eq 18).

\[ ASM = \sum_{i=0}^{N-1} \sum_{j=0}^{N-1} Mc(i,j)^2 \]  

(18)

Where N is the number of gray levels, Mc is the image matrix.

Variance: This is the sum of the squares of the differences between the intensity of the central pixel and its neighbors (eq 19).

\[ V = \sum_{i=0}^{N-1} \sum_{j=0}^{N-1} (i - \text{mean})^2 Mc(i,j) \]  

(19)

Where N is the number of gray levels, Mc is the image matrix.

Standard deviation: The standard deviation is a measure of the amount of variation or dispersion of a set of values (eq 20).

\[ \sigma^2 = \sum_{i=0}^{N-1} (i - \text{mean})^2 \sum_{j=0}^{N-1} Mc(i,j) \]  

(20)

Where N is the number of gray levels, Mc is the image matrix.

Correlation: Correlation feature shows the linear dependency of gray level values in the coocurrence matrix: (eq 21).

\[ Corr = \frac{1}{N^2} \sum_{i=0}^{N-1} \sum_{j=0}^{N-1} (i - \text{mean})^2 Mc(i,j) \]  

(21)

Entropy: is a measure of information content. It measures the randomness of intensity distribution and the homogeneity of the histogram (eq 22).

\[ Entropy = -\sum_{i=0}^{N-1} \sum_{j=0}^{N-1} Mc(i,j) \log Mc(i,j) \]  

(22)

Where N is the number of gray levels, Mc is the image matrix.

APPENDIX C

In our work, we employed an Artificial Neural network for the classification. The parameters of a neural network are typically the coefficients of the model. In this case, these parameters are learned during the training stage. So, the algorithm itself, optimizes these coefficients. However, when training it, there are a number of hyperparameters we needed to set, including Table VII.

<table>
<thead>
<tr>
<th>Hyperparameters (this is the case of each region)</th>
</tr>
</thead>
<tbody>
<tr>
<td>Number of hidden layers: 3</td>
</tr>
<tr>
<td>Number of units in the input: 45</td>
</tr>
<tr>
<td>Number of units in first hidden: 90</td>
</tr>
<tr>
<td>Number of units in second hidden: 30</td>
</tr>
<tr>
<td>Number of units in third hidden: 15</td>
</tr>
<tr>
<td>Number of units in output layer: 1</td>
</tr>
<tr>
<td>Learning rate: 0.000001</td>
</tr>
<tr>
<td>Activation function for 1,2,3,4: activation=tanh'</td>
</tr>
<tr>
<td>Activation function for the final layer: activation='sigmoid'</td>
</tr>
<tr>
<td>Minibatch size: 32</td>
</tr>
<tr>
<td>Epochs: 800</td>
</tr>
<tr>
<td>Loss Function: 'mean_squared_error'</td>
</tr>
</tbody>
</table>
Blockchain Architecture Based on Decentralised PoW Algorithm

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Abstract—Blockchain has gained increasing popularity across various industries due to its decentralized, stable, and secure nature. Consensus algorithms play a crucial role in maintaining the security and efficiency of Blockchain systems and selecting the right algorithm can lead to significant performance improvements. This article aims to provide a comparative review of the most used Blockchain consensus algorithms, highlighting their strengths and weaknesses. Additionally, we propose a dissociated architecture for an efficient Blockchain system that doesn't compromise on security. A comparison is made between this architecture and the reviewed algorithms, considering aspects such as algorithm performance, energy consumption, mining, decentralization level, and vulnerability to security threats. The research findings demonstrate that the proposed architecture can support complex algorithms with high security while addressing issues related to efficiency, processing performance, and energy consumption.

Keywords—Blockchain technology; proof of work; consensus algorithm; proof of stake; Dissociated-PoW; security; performance

I. INTRODUCTION

Blockchain (BC) technology offers numerous advantages that make it highly relevant and applicable in various areas of life. As a result, it has gained significant importance and presence in diverse projects. The main benefits it provides are decentralization, immutability, integrity, and reliability. These qualities guarantee a high level of information protection throughout all phases of processing and storage [1]. This technology reached its pinnacle thanks to the rise of Bitcoin, primarily attributed to its ability to establish a decentralized ledger, ensure immutable records, and provide participant anonymity. These factors played a crucial role in driving the widespread adoption and recognition of blockchain technology [2]. However, it is precisely the strength of blockchain technology that also makes it increasingly challenging to manage. In its original form, blockchain utilizes a consensus algorithm known as Proof of Work (PoW) [3], this consensus algorithm used in blockchain requires a substantial computational effort, creating a situation where only large computer producers can effectively participate. This dynamic makes it financially unprofitable for smaller infrastructures to engage in the process [4]. To address this issue, blockchain (BC) has evolved in various directions, exploring different consensus algorithms and structures that can alleviate the computational costs. In this paper, we propose a solution that harnesses the security strengths of Proof of Work (PoW) while introducing a BC architecture built on functionally specialized nodes. This approach enables the efficient implementation of PoW while minimizing the computational expenses.

To this end, Section II reviews the state of the art and the main issues related to the work, especially reviewing the different lines of work that have been adopted from the original BC. Section III proposes the new architecture, the functional characterization of the operations and the description of the nodes. In Section IV, a comparison is made with the traditional BC to show the differences and similarities of the proposed architecture and, above all, the benefits. The last section finally shows the main conclusions of the proposal with possible lines of further work.

II. BACKGROUND

First, BC networks developed from the original one proposed by Satoshi [5]. Depending on the participants involved, blockchain networks can be categorized as public, private, or hybrid. Additionally, the permissions granted for write and read operations on the blockchain determine whether it is classified as permissioned or permissionless. A blockchain can combine various aspects of participation and operation permissions, resulting in a wide range of scenarios that can be applied based on the specific organization and the type of application being implemented [6]. The choice of the blockchain type depends on the intended use case and the specific requirements of the organization.

In public blockchain networks, all individuals have the freedom to read, send, or validate transactions. Furthermore, they can actively participate in the distributed consensus process. The consensus mechanism relies entirely on the merits and contributions of individual nodes within the network, without the need for any centralized authority or control. This open and inclusive approach fosters transparency, decentralization, and trust among participants in the public blockchain network [2], nodes compete to achieve consensus. Unlike in a private network, there is no access control mechanism restricting participation. Nodes have the freedom to join or leave the network without causing harm to the consensus mechanism or the generation of new blocks [7]. This decentralized nature allows for a dynamic network where nodes can freely interact and contribute to the consensus process.

In private BC networks the addition and verification of new blocks are limited to specific users authorized by the controlling entity, which can be centralized or decentralized. In
such networks, unauthorized users are unable to add information to the blockchain network and may even be restricted from accessing read capabilities. Unlike public networks, private networks necessitate consensus algorithms tailored to accommodate these permission restrictions [8].

Hybrid or consortium BC networks represent a combination of both public and private networks. In these networks, participating nodes are typically invited or selected, but all transactions are publicly visible and transparent. One of the benefits of these networks is the protection they provide against 51% attacks [9]. A 51% attack refers to a scenario where a single entity or group of nodes gains control of over 50% of the network's computing power, potentially enabling them to manipulate the blockchain. In a hybrid or consortium network, the distributed nature of participation helps safeguard against such attacks, as they require majority control, which is not easily attainable in these network configurations.

In fact, each type of blockchain network requires a consensus algorithm that suits its specific needs. A consensus algorithm is a set of rules and processes that govern the operation of a distributed system. Its main purpose is to resolve data synchronisation between untrusted nodes in a decentralised environment. Consensus algorithms ensure that all participants agree on the state of the blockchain and validate transactions in a consistent and secure manner. The choice of consensus algorithm depends on factors such as the type of network, the desired level of decentralisation, scalability requirements and security considerations [10].

The original consensus algorithm used in blockchain is Proof of Work (PoW): is a decentralised consensus algorithm that requires network participants to compete to solve a computational puzzle. This puzzle-solving process helps prevent malicious actors from manipulating the system and ensures the security and integrity of the blockchain. PoW is widely used in cryptocurrency mining, where participants validate transactions and mine new tokens, as in the case of the Bitcoin network. By dedicating computational resources and solving complex puzzles, miners contribute to the consensus process and maintain the decentralised nature of the blockchain network [11]. One of the main challenges and drawbacks of PoW is the significant computational capacity required to solve the mathematical problem involved in authenticating blockchain transactions. This computational intensity leads to high energy consumption and limits the scalability of PoW-based blockchain networks. The resource-intensive nature of PoW can also make it less accessible for smaller participants or entities with limited computing power. As a result, there has been a drive to explore alternative consensus algorithms that can address these limitations, such as Proof of Stake (PoS) or delegated proof of stake (DPoS) that do not require extensive computational power. These consensus algorithms enable participants to validate transactions and secure the network based on factors like the number of tokens held or reputation, rather than raw computing power [12]. To participate in a BC network that uses PoW as a consensus mechanism, it is essential to have advanced, powerful, expensive, and energy-consuming hardware [13], thus widely so that users currently must compete against pools of mining [14]. Due to the drawbacks and limitations of the Proof of Work (PoW) consensus algorithm, the blockchain community has developed and explored various alternative consensus algorithms. These alternative algorithms aim to address issues such as high energy consumption, scalability, and accessibility. Some of the popular alternative consensus algorithms include:

Proof of Stake (PoS): in PoS participants can create new blocks and validate transactions based on the number of tokens they hold or "stake" in the network. This approach reduces energy consumption and allows for a more efficient and scalable consensus mechanism. Consensus algorithm was the next algorithm to appear. It works by encouraging users to always keep a certain number of cryptocurrencies in the wallet [15]. Keeping these cryptocurrencies locked helps participants increase their chances of being chosen, as this is one of the main criteria set for participation. Once these criteria have been established, the random node selection process begins. When the nodes are chosen and the selection process is finished, they are ready to validate transactions or create new blocks. It is more environmentally friendly than PoW as it does not require large amounts of computational power to operate [16]. The security of the network is much higher as it solves or hinders certain known attack schemes, such as the 51% attack, overall, PoS offers benefits such as energy efficiency, increased network security, and improved scalability. It is an alternative consensus algorithm that has gained popularity in blockchain networks seeking to address the limitations of PoW.

Delegated Proof of Stake (DPoS): it is a consensus algorithm commonly used in blockchain networks. It is designed to address some of the limitations of traditional Proof of Stake (PoS) algorithms and aims to achieve faster transaction processing and increased scalability [4]. Referring to a more decentralised form in the BC network, it also modifies the way power is used, decreasing in frequency. With this algorithm, users can give their votes on who they want to mine the next block. Thus, it offers high levels of security for use in public BCs. Besides this, its operating model guarantees high levels of scalability. To make this possible, each participant in the network chooses, by voting, several "delegates". Once chosen, they form an ensemble that offers Byzantine Fault Tolerance (BFT) [17].

Delegated Byzantine Fault Tolerance (dBFT): consensus algorithm's main objective consists in giving the right to all stakeholders to be an active member through the voting procedure to solve the problems in the blockchain in a democratic and fair way. It is an algorithm that combines the characteristic of the DPoS algorithm, but unlike the DPoS algorithm, delegate nodes are elected to validate the new block, and where at least 2/3 must approve it [18]. In this way, the network can make decisions even if one third of the nodes are harmful or corrupted [19].

The Proof of Activity (PoA): relies on a set of approved validators who are known and trusted. Validators take turns creating new blocks, and consensus is achieved based on their identity and reputation. It is a combination of PoW and PoS [20]. Initially the system works in PoW where miners try to solve a mathematical equation using high computational capabilities. Once a new block is generated, the system switches to PoS, a group of validators is chosen at the start, and
they will oversee verifying or signing the new block. Validators are elected and sign the new block. More coins a validator possesses, more likely to be elected and sign [21]. A 51% chance of an attack is also avoided in this algorithm.

Proof of Burn (PoB): consensus algorithm works in such a way that miners must send cryptocurrencies to a public and verifiable address, from which they will not retrieve them again, burning them. In other words, miners must make a kind of investment and the higher the number of cryptocurrencies burned, the more mining the miner achieves [22]. Its main benefit is that, with a higher percentage of long-term investors, price stability may increase. Also, PoB assists in determining the distribution of cryptocurrencies in a fair and decentralised manner. Reduction in the number of resources needed to achieve consensus compared to PoW is considerable and it also has resistance to double spending attack.

Proof of Capacity (PoC): consensus algorithm enables network mining devices to use their available hard disk space to decide entitlements when mining and validating transactions [23]. Differentiating this with the use of the computational power of the PoW mining device or the miner’s participation in PoS cryptocurrencies. As a benefit, any hard drive can be used, including those with Android-based operating systems. Mining data can be easily erased, and the drive can be reused for any other data storage purpose. Compared to PoW [24], this improves the resource consumption for consensus by up to 30 times less.

The Proof of Elapsed Time (PoET): is a consensus algorithm that aims to achieve decentralized consensus while minimizing energy consumption. PoET was introduced by Intel as a consensus algorithm for use in private and permissioned blockchain networks.

In PoET, participants in the network compete to become the leader or validator of the next block. However, instead of relying on computational power or staking, PoET uses a random lottery-based approach. Each participant in the network waits for a randomly generated timer to expire [25]. The participant whose timer expires first becomes the leader and gets the right to create the next block.

To prevent participants from manipulating the system by manipulating timers, PoET uses a trusted execution environment (TEE). The TEE provides a secure and tamper-resistant environment for generating the random timers. This ensures fairness and prevents participants from predicting or influencing the timer outcomes.

The next algorithm is a new consensus mechanism using less energy and can run on low-end hardware. Proof of Assignment (PoA), developed by IOTW [26]; this algorithm can process thousands of transactions per second and has been introduced for micro mining, allowing for lightweight mining on IoT devices by eliminating the need to store and maintain the transaction ledger at the device level. Instead, the storage and maintenance of the ledger is outsourced to one or more pre-established trusted nodes in the BC network. It can run on hardware of any IoT device [27].

The Ripple Protocol Consensus Algorithm (RPCA): to maintain the veracity and consensus of the network, constantly updates its nodes; once consensus is reached, the current ledger is "closed" and becomes the last closed ledger. Assuming the success of the consensus algorithm, and that there is no fork in the network, the last closed ledger held by all nodes in the network will be identical [28]. Among its features is that the validation of transactions is immediate, reducing the time of each operation. In addition, the costs per transaction are also very low and the Ripple cryptocurrency has greater flexibility compared to other cryptocurrencies for international payments, it is oriented for monetary transactions between banks, a competitor of the SWIFT system [29].

The development of consensus algorithms aims to preserve the power and robustness of PoW while addressing its inherent challenges. However, it is essential to acknowledge that changing the consensus algorithm does not necessarily guarantee the elimination of problems, but rather introduces new considerations and trade-offs.

The proposal to shift the focus of innovation in blockchain from creating new algorithms to designing new architectures to solve existing problems is an interesting approach. By reimagining the underlying architecture, it becomes possible to optimize the performance, scalability, and energy efficiency of blockchain networks, while still leveraging existing consensus algorithms like PoW. This architecture will be called Dissociated Proof of Work (Dissociated-PoW).

III. DISSOCIATED-POW ARCHITECTURE

The proposed Dissociated-PoW architecture aims to address the high computational cost associated with competition between mining nodes in Proof-of-Work based blockchain networks. This architecture introduces two types of functionally specialised nodes: coordinator nodes (CN) and mining nodes (MN). Each type of node and its function is described below:

- Coordinator nodes: Serving as central entities, coordinator nodes take charge of coordinating the mining process and facilitating the consensus algorithm. They play a crucial role in selecting and assigning mining tasks to the mining nodes. Furthermore, CN are responsible for overseeing the overall operation of the blockchain network and ensuring the integrity of the consensus process.
- Mining nodes: Dedicated to performing the computational work required to mine new blocks on the blockchain, mining nodes form the scheme of the Dissociated-PoW architecture. These nodes receive mining tasks from the coordinator nodes and leverage their computational resources to solve the PoW puzzle and validate transactions. MNs are pivotal in generating new blocks and maintaining the continuity of the blockchain.

This proposed architecture has the potential to enhance the efficiency and scalability of blockchain systems while preserving the fundamental principles of decentralization and trust. Further research and experimentation are needed to validate its effectiveness and practicality in real-world scenarios.
A traditional blockchain scheme exhibits the typical structure wherein all nodes participate in the mining competition, leading to high computational costs. In contrast, the proposed Dissociated-PoW scheme introduces a separation of tasks between coordinator nodes and mining nodes, resulting in a more efficient and optimized mining process.

The Dissociated-PoW architecture has the potential to mitigate the drawbacks of traditional PoW-based blockchain networks by reducing computational costs and optimising mining resource allocation. However, it is important to thoroughly analyse and evaluate the proposed architecture in terms of security, decentralisation, and performance to ensure its effectiveness in real-world scenarios.

A. Coordinating Nodes are responsible for the following Tasks

- Receiving and storing new information to be inserted into the blockchain.
- Managing the pending transactions PT, which form a block of information containing all the pending transactions that will be mined to generate the next block of the blockchain. PT is a block of transactions that are replicated and synchronized among all coordinating nodes.
- Deciding when to initiate the mining process by utilizing rules to trigger mining based on criteria like a timeout or reaching a maximum PT size.
- To decide who will be the MNs to mine the next PT.

B. Mining Nodes are responsible for the following Tasks

- Maintaining the blockchain to ensure it is accessible for any user. Each MN contains a copy of the entire blockchain.
- Mining the next block of the BC without engaging in competition. The MN receives the pending transactions PT from a CN and performs the mining process. Once the mining is completed, the new block is broadcasted to the BC.

In our proposal, it is the CN that instruct a specific MN to carry out the mining process, providing it with the necessary information to be mined, which includes the pending transactions, as illustrated in Fig. 1.

As shown in the proposed Dissociated-PoW architecture, the dissociation of mining tasks and the coordination provided by the CN contribute to a more efficient, secure, and decentralized blockchain network.

Action flows of the CN for the specified tasks:

1) Receiving a new transaction:

- A coordinating node receives a new transaction to be included in the blockchain.

2) Adding it to the pending transactions:

- The CN incorporates the new transaction into its own pending transactions PT block.

3) Disseminating the new transaction among the rest of the CN:

- The CN broadcasts the new transaction to all other coordinating nodes to ensure they are informed about the pending transaction.
- It is essential to ensure that the transactions are inserted into the PT block in the same order among all coordinating nodes during dissemination.

4) Validating the origin of the new transaction:

- Other coordinating nodes verify the origin and validity of the new transaction sent by the CN.

5) Synchronizing all the pending transactions:

- After validation, all coordinating nodes have the same block of pending transactions, ensuring synchronization.

As for mining, it is performed in the following way with a leader node among the CN:

1) Detecting the mining start condition:

- The leader CN detects the condition to initiate the mining process.

2) Notifying other CN nodes about mining:

- The leader CN notifies all other CN nodes that mining is about to proceed, prompting them to freeze the current pending transactions block and mark it as in the mining state.

3) Selecting a MN to mine the pending transactions:
• The leader CN selects a MN from the Blockchain Node List (BCNL) to mine the current pending transactions block and sends the pending transactions block to be mined.

4) Notifying completion and disseminating the new block
• Once the selected MN finishes mining, it notifies the CN that initiated the mining process and disseminates the new block to all other MNs.

5) Indicating the end of mining and validating the mined block:
• The leading CN indicates to the other CN that mining is finished and that the previous block of transactions can be finalized.
• The rest of the CN verify the correctness of the mined block and conclude the mining operation. As shown in the Fig. 2.

This process, with a designated leader node among the CN, ensures a coordinated and efficient mining process in the Dissociated-PoW architecture, promoting consistency and integrity throughout the blockchain network. As shown in Fig. 3.

Action flows to start mining:

1) The leading CN decides when to initiate the mining process.
2) The leading CN notifies the rest of the CN to begin mining and instructs them to freeze the current pending transactions (PT) block.
3) The leading CN selects a node from the network to perform the mining task and sends the PT block to that node.
4) The selected mining node informs the leading CN when it has completed the mining process and broadcasts the newly mined block to all other MN.

5) The leading CN notifies all the other CN that the previous mining process has concluded, and the mined PT can be removed.

The synchronization between Coordinator nodes in the Dissociated-PoW blockchain network follows a functional model that is similar to node coordination in a Kafka cluster [28]. A leader node is designated, and the rest of the CN become followers. Only the leader node can determine the timing of mining and which MN will be responsible for the mining task. The designation of a leader node is based on the order of entry to the BC network, and in the event of the leader node's failure, the next node in the list is designated as the new leader. Every CN has a synchronized PT block, so if any CN goes down, as long as there is at least one CN remaining in the network, the process continues uninterrupted.

Nodes may enter the BC network with the roles of MN, CN, or both. However, having both roles does not give priority to the node in being designated as a MN. The decision of exactly when to start mining follows predefined rules in the BC. Typically, mining is triggered when a certain number of pending transactions is reached or when a specific time limit between mining operations is reached if there are pending transactions. Additionally, a maximum pending transaction limit criterion is used to restrict the size of the block to be mined. Employing a time limit criterion ensures that transactions are not left waiting indefinitely to be mined. Deciding exactly when to mine simply follows predefined rules in the BC. Normally, mining is set when a certain number of pending transactions is reached or when a specific time limit between mining is reached, if there are pending transactions. A maximum PT limit criterion limits the size of the block to be mined. Using a time limit criterion allows transactions not to be left eternally waiting to be mined.
When the leading CN determines that mining is necessary, it instructs the other CN in the network to freeze the PT block, ensuring that any new incoming transactions are directed to a new temporary PT block. This way, the PT block being mined remains unchanged.

The BCNL (Blockchain Node List) is utilized to select the main CN responsible for mining the PT block. This list contains all the nodes comprising the BC network, their respective roles, the PT blocks mined by each node, and their hardware specifications. Moreover, nodes can provide information about their performance status. For instance, if a node is experiencing overload at a given time, it can indicate this in the BCNL to avoid being designated as a mining node. With access to this list, the leading CN can choose the most suitable node at that moment, taking certain constraints into consideration:

- The best-performing node will be selected from the BCNL list.
- Mining distribution will be balanced to ensure that all nodes in the list participate in the mining process.
- If no MN can be designated, the leading CN will keep the block as pending mining until a node from the list becomes available.

To participate as an MN, the BC may require a minimum level of computational resources to ensure that mining can occur in a timely manner. Additionally, even the leading BC may reject a mining request made to a node if it determines that the node will be unable to complete the mining task. In such cases, the request will be redirected to a new MN.

This approach enables the Dissociated-PoW architecture to efficiently manage mining tasks and ensure that mining is carried out optimally based on the performance capabilities of the participating nodes.

IV. Benchmarking Proposal Compared Against Other Algorithms

This proposal, as discussed in the previous section, offers several benefits that make the Dissociated-PoW consensus algorithm more effective than other existing algorithms, overcoming many of the disadvantages present in traditional ones. In this chapter, we will compare our Dissociated-PoW algorithm with the following consensus algorithms: Proof of Work (PoW), Proof of Stake (PoS), Delegated Proof of Stake (DPoS), Delegated Byzantine Fault Tolerance (dBFT), Proof of Activity (PoA), Proof of Burn (PoB), Proof of Capacity (PoC), Proof of Elapsed Time (PoET), Proof of Assignment (PoA), Proof of Checkpoint (PoC), and Ripple Protocol Consensus Algorithm (RPCA).

As mentioned earlier, the main disadvantage of PoW is the substantial computational and energy requirements to solve the consensus test, with exclusive resource usage that cannot be utilized for other tasks. Additionally, the well-known 51% attack poses a significant problem, with the algorithm being dominated by large mining pools. On the contrary, our proposal ensures that only the designated mining node invests computational and energy resources, leading to substantial savings as there is no competition among nodes. By dissociating the selection of the time and mining node from the traditional competition algorithm, we also eliminate the risk of a 51% attack, ensuring that no entity dominates the PoW algorithm and avoiding the dominance of large miners.

In PoS, significant drawbacks include challenges in maintaining miner anonymity, as those who invest the most in the BC end up being the dominant miners in solving the consensus algorithm. Scalability issues and potential slow transactions also arise. In Dissociated-PoW, the CN overoses selecting the best candidate for mining, guaranteeing investor anonymity and independence from major stakeholders. As the network has specialized nodes dedicated to BC maintenance, the quality of service is assured, avoiding collapsed nodes except for the one that may be mining at any given time. However, as the BC is replicated in the rest of the nodes, its operation can be supported by a nearby node.

The DPoS algorithm is also vulnerable to centralization due to its limited number of nodes in the network, and transactions are not anonymous. In Dissociated-PoW, all transactions are anonymous, and the number of nodes is organized by the CN and MN responsible for distributing the workload.

The dBFT algorithm employs a relatively recent protocol that has not been extensively tested in large Blockchain networks, making it susceptible to centralization with a limited number of representing nodes. Similar to DPoS, transactions are not anonymized. Dissociated-PoW addresses these concerns by having a specific group of nodes that focus on coordinating work and sending pending transactions to MN to commence mining after a certain period.

Overall, Dissociated-PoW offers improved anonymity, decentralization, and scalability while mitigating the vulnerabilities present in other consensus algorithms, making it a promising choice for effective and efficient blockchain networks.

In the PoET algorithm there is no anonymity. Making it vulnerable to Sybil attacks [30]. The more malicious nodes there are, the higher the likelihood that some of them will be involved in block formation. In contrast, in Dissociated-PoW, participants remain anonymous, and as the CN are responsible for overseeing mining, Sybil attacks among the MN are not possible. As for the CN, malicious actions may involve attempting to extract a fake PT packet, but this would be detected by the other CN. They could also try to slow down the BC network, but this would be detected as a failure of the leading CN, and the next CN in the list would take the lead. Additionally, they may attempt to overload an MN by sending it all mining requests, but the MN can update its status in the BCNL at any time to prevent receiving further requests.

On the other hand, Proof of Assignment (PoA) has limitations in capacity due to the processing speed and available memory of IoT devices. In Dissociated-PoW, capacity is not limited during the processing of PTs as the CNs choose the best available miner node, balancing network loads to avoid saturating a single node.

For the Proof of CheckPoint (PoC) algorithm, trust nodes external to the network are required. In Dissociated-PoW, no
external nodes are needed as it only involves the coordinator and miner nodes.

In the RPCA algorithm, the network is exposed to centralization as the number of nodes representing the network is very limited, and transactions are not anonymous. In Dissociated-PoW, the CN and MN are responsible for selecting the best candidate to start mining, guaranteeing investor anonymity, and not relying on centralized entities.

Overall, Dissociated-PoW offers advantages over several existing consensus algorithms by ensuring participant anonymity, mitigating vulnerabilities to Sybil attacks, and avoiding the need for external trust nodes while maintaining high processing capacity and network decentralization.

In the next Table I, issues presented within the current algorithms can be grouped as follows:

1) **Anonymity loss:** This criterion evaluates whether the algorithm preserves the anonymity of nodes participating in the blockchain network. A "Yes" indicates that the algorithm does not provide strong anonymity, meaning that at some point, the identities of network nodes or owners can be identified. Dissociated-PoW receives a "No" in this category, meaning it preserves anonymity.

2) **Deployment:** This criterion assesses the complexity of implementing the consensus algorithm. A "Simple" deployment means that the algorithm is straightforward and easy to implement, while a "Complex" deployment suggests that it requires more effort and expertise to set up. Dissociated-PoW is considered "Simple" in this regard.

3) **Non-Vulnerable to centralization:** This criterion determines whether the consensus algorithm is at risk of centralization, where a small number of nodes gain excessive control over the network. A "Yes" indicates vulnerability to centralization, while a "No" means the algorithm is designed to resist centralization. Dissociated-PoW is not vulnerable to centralization.

4) **Resources in network:** This criterion evaluates the number of resources (such as computational power, memory, etc.) needed to maintain the blockchain network. "High" resources mean considerable requirements, while "Low" resources indicate that the algorithm can operate efficiently with minimal resources. Dissociated-PoW requires low resources.

5) **Low computing load:** This criterion assesses the power consumption and computing demand of the consensus algorithm. "Yes" indicates that the algorithm has low computing load, while "No" means it is computationally intensive. Dissociated-PoW has low computing load.

6) **IoT device friendly:** This criterion determines whether the consensus algorithm is suitable for IoT devices, which often have limited processing capabilities. "Yes" means the algorithm is compatible with IoT devices, while "No" suggests it may not be well-suited. Dissociated-PoW is friendly to IoT devices.

The comparison table shows that Dissociated-PoW performs well in terms of preserving anonymity, simplicity of deployment, non-vulnerability to centralization, efficient resource usage, low computing load, and compatibility with IoT devices. However, it's important to note that each algorithm has its strengths and weaknesses, and the best choice depends on the specific requirements and goals of the blockchain network being considered (see Table I).

<table>
<thead>
<tr>
<th>Consensus Algorithm</th>
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V. **DISSOCIATED-POW PROBLEMS**

Introducing a new architecture, such as Dissociated-PoW proposal, may indeed bring certain potential drawbacks. One of the concerns is the possibility of malicious coordinator nodes and mining nodes gaining access to the network, particularly in a public network where there are no access control mechanisms.

A. **Mining Nodes Vulnerabilities**

Potential vulnerabilities in this section will describe the possible security issues arising from malicious actions or failures in the mining nodes. Details of the CNs and MNs for the transactions that are integrated in the blockchain have been presented so far. The key to its operation is the validation of the transaction. For this reason, mining, consisting of closing the block and distributing it in the BC, must be carried out. As stated, miners communicate among themselves.

B. **Eternal Mining (Lack of Computing)**

Typical blockchain mining issues include the possibility of a mining node never finishing or being too late, either if the mathematical problem to be solved turns out to be very complicated or if the node does not have sufficient computational capacity at that time. To resolve this situation, CN must maintain a time limit. Once this limit is exceeded without results, the CN will withdraw the right to mine from the selected node and pass it to another CN, providing an opportunity for the block generation to occur. Alternatively, a function could be implemented where the MN is not removed from the mining right, but another new node is selected to start mining to check if it can deliver a result before the original node. This way, even if the first node fails, no computing power will have been wasted.
C. Malicious and Vulnerable Coordinating Nodes and Miners

Malicious MN may attempt to mine fake content, i.e., transactions that were not originally processed by the CN, and try to sneak them in. After they finish mining the MNs, send the new block to the CN, whose job it is to validate that the node is correct, and therefore if they have tried to introduce transactions that were not in the original PT, they will be discarded. If it is correct, it will be distributed to other MN and CN, and in turn the CN will distribute it to nearby MN. If it is a malicious node, it is kicked out and otherwise distributed to the rest of the MN nodes in the network. Potential vulnerabilities caused by malicious actions or failures in the coordinator nodes will be described in this section.

D. Trying to Crash a Mining Node

It consists of a CN involuntarily sending mining requests to the same MNs all the time, overloading them with work. A BCNL exists, where CN may update their state to not receive requests. In Dissociated-PoW, there is a list called BCNL, where CN may update status for MN to not accept and receive any more requests, avoiding overloading MN. For an MN to receive a PT, it must receive a low mining workload.

E. Not Launching the Mining Process

As the leader CN oversees determining the start of mining, should this node fail for any reason, orphaned mining won't start. Other CN must enable a mechanism for detecting such a situation and designate new leaders independently. The role of the CN is to supervise the proper functioning of the leader node. As there are several CN, if a lag is detected, a second CN takes over leadership, notifying other CN of the incident, and the current leader is automatically substituted.

F. The CN is Malicious and Orders False Content to be Mined

Upon distribution of a new BC block, it must be validated through its BCNL by all CN to identify whether it is a bogus block or not. If it is a bogus block, it must be discarded and removed from the network.

To mitigate these potential drawbacks, it is important to implement robust security measures, access controls, and thorough vetting processes for CN and MN. Additionally, ongoing monitoring, auditing, and regular updates to the architecture can help address emerging security concerns and maintain a healthy and secure network environment.

It is essential to thoroughly analyse and address these potential drawbacks when designing and implementing a new architecture to ensure the overall security and integrity of the blockchain network.

VI. CONCLUSION AND FUTURE WORK

This research makes several valuable contributions to the field of blockchain technology and consensus algorithms. First and foremost, it presents the Dissociated-PoW algorithm as a novel and innovative solution to address the limitations of traditional Proof of Work and other existing consensus algorithms. By functionally specializing nodes, Dissociated-PoW efficiently implements PoW while minimizing computational expenses, making it more sustainable and scalable for diverse blockchain networks.

Furthermore, the comprehensive analysis and comparison of various consensus algorithms provide valuable insights into their strengths, vulnerabilities, and suitability for different use cases. This analysis aids researchers, developers, and decision-makers in understanding the trade-offs between different consensus mechanisms, guiding them in choosing the most appropriate algorithm for their specific blockchain applications.

The proposed future work of building a robust framework that encompasses multiple consensus algorithms and emphasizes privacy safeguards across different blockchain architectures is forward-thinking and crucial for the continued advancement of blockchain technology. By addressing issues like high computational demands, network performance, and node failures, the research contributes to the development of more secure, efficient, and resilient blockchain networks.

Despite these significant contributions, the research also acknowledges certain limitations. Firstly, due to the complexity and variety of blockchain networks and consensus algorithms, conducting a detailed analysis of all possible combinations may not be feasible within the scope of this research. However, the chosen approach of comparing existing algorithms provides valuable insights and lays the foundation for future research to explore additional combinations and innovations.

Additionally, as with any research in the rapidly evolving field of blockchain technology, the proposed Dissociated-PoW algorithm and future framework may face challenges in real-world implementation. Practical testing and validation on various blockchain networks will be essential to ensure the algorithm's effectiveness and security.

Inclusive, this research makes a significant scientific contribution by proposing a novel consensus algorithm, conducting a thorough comparative analysis, and outlining a comprehensive roadmap for future work. It offers valuable insights into the state of blockchain consensus algorithms and provides a foundation for advancing the field, addressing limitations, and shaping the future of decentralized systems.

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Sentiment Analysis of Code-mixed Social Media Data on Philippine UAQTE using Fine-tuned mBERT Model

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Abstract—The Universal Access to Quality Tertiary Education (UAQTE) marks a significant policy change in the Philippines. While the program’s objective is to offer free higher education and tertiary education subsidies to eligible Filipino students, its viability and effectiveness have been subject to scrutiny and continuous evaluation. This study explores the sentiments of Filipinos towards UAQTE. Leveraging a fine-tuned multilingual Bidirectional Encoder Representations from Transformers (mBERT) model, we conducted sentiment analysis on code-mixed data. With minimal preprocessing, our model achieved an accuracy of 80.21% and an F1 score of 81.14%, surpassing previous related studies and confirming its effectiveness in handling code-mixed data. The results reveal that the majority of social media users view UAQTE positively or beneficially. However, negative sentiments highlight concerns related to subsidy delays, alleged fund misuse, and application challenges. Additionally, neutral sentiments center around subsidy-related announcements. These findings provide valuable insights for its key stakeholders involved in the implementation, enhancement, and evaluation of UAQTE.

Keywords—Sentiment analysis; UAQTE; code-mixing; policy-making; multilingual BERT

I. INTRODUCTION

Countries worldwide increasingly recognize the importance of higher education for economic competitiveness [1]. This has led to a greater focus on understanding the costs and benefits of education and research, and the need for improved productivity in higher education [2]. Among the 17 Sustainable Development Goals (SDGs) set by the United Nations, the fourth goal specifically focuses on Quality Education. This goal is dedicated to achieving inclusive and equitable access to high-quality education and fostering lifelong learning opportunities for all individuals by the year 2030 [3].

One of the most significant turning points in Philippine education is the proactive efforts of the government to broaden the reach and involvement in higher education by approving the Republic Act 10931, otherwise known as the “Universal Access to Quality Tertiary Education (UAQTE) Act”, enacted into law on August 2017. The UAQTE initiative represents a significant legislative effort to ensure widespread access to high-quality tertiary education by providing free tuition and covering other school fees in State Universities and Colleges (SUCs), Local Universities and Colleges (LUCs), and State-Run Technical Vocational Institutions. Specifically, this law established four programs namely Free Higher Education (FHE), Tertiary Education Subsidy (TES), Tulong Dunong Program (TDP), Free Technical and Vocational Training, and Student Loan Program. Its implementation is primarily led by the Unified Student Financial Assistance System for Tertiary Education (UniFAST), an attached agency of the Commission on Higher Education (CHEd) [4].

Although its objectives were clearly defined, its feasibility has been questioned. Concerns have been raised about the design of the law, its implementing rules and regulations, and the adequacy of resources to sustain its programs. Early assessment of the program [5], explained that the absence of clear and prompt guidelines had caused difficulties in service delivery and utilization, particularly in the processing of billing requirements, resulting in delays in reimbursement. Thus, continuous evaluation and improvement are necessary to ensure the program’s effectiveness in achieving its goals.

With the changing landscape of information sharing, social media platforms have emerged as vital channels for public discussions and viewpoints on a wide range of social and political matters [6]. As of this writing, the Philippines has a notable digital presence, with 83% of its population being internet users. Filipinos spend approximately four to five hours per day on social media, which is twice the global average of two to three hours [7]. This significant difference highlights the pervasive role of social media in the lives of Filipinos. Consequently, leveraging this wealth of social media data through Natural Language Processing (NLP) techniques, such as sentiment analysis, holds immense potential for enhancing our understanding of public perception and sentiment towards the UAQTE.

Many government-initiated practices tend to adopt a top-down approach to knowledge sharing [8]. This approach often treats local communities as passive recipients rather than active collaborators. This observation highlights the significance of incorporating the sentiments of local communities, in our case, the stakeholders of UAQTE, in the analysis process.

The findings of this study aim to contribute to a deeper understanding of public sentiments surrounding UAQTE and can serve as inputs for decision-makers at CHEd, UniFAST, educational institutions, and other stakeholders involved in implementing and evaluating the UAQTE program. To the best
of our knowledge, this study is the first to evaluate UAQTE sentiments using sentiment analysis on code-mixed social media data. Through the application of the state-of-the-art multilingual Bidirectional Encoder Representations from Transformers (mBERT) [9], we also aim to test its efficacy in identifying prevailing neutral, positive, and negative sentiments expressed towards UAQTE.

In this introduction, we have provided an overview of the research focus and rationale for conducting sentiment analysis on social media data related to the UAQTE. Section II discusses related works on education and utilizing code-mixed data for sentiment analysis. Section III covers the methodology, including data collection, preprocessing, and fine-tuning of mBERT. In Section IV, we present the results on the effectiveness of mBERT in code-mixed sentiment analysis and discuss public sentiment distribution towards UAQTE. Section V concludes with key insights and implications, while Section VI outlines the limitations and future research directions to enhance sentiment analysis in code-mixed data.

II. RELATED WORKS

Sentiment Analysis involves analyzing a sequence of words to uncover the underlying emotional tone and gain insights into the attitudes, opinions, and emotions conveyed in online mentions [10]. This type of analysis is used in various fields, including business and marketing, politics, health, and social policy, to allow policymakers to formulate informed adjustments to new user-centric rules and regulations [11]. In this section, we will discuss the application of sentiment analysis in the field of education and its findings when this technique is applied to code-mixed data settings.

A. Sentiment Analysis on Education

In the context of education, [12] utilized a range of machine learning techniques, including Support Vector Machine (SVM), Multinomial Naive Bayes (MNB), Random Forest (RF), and Multilayer Perception classifier. By exploring and comparing different SA models, the study successfully determined the effective approaches for analyzing student’s classroom feedback that would enhance the quality of teaching in higher education institutions. Moreover, they highlighted the potential of social media platforms like Twitter and Facebook as valuable sources for gathering information and extracting opinions pertaining to students’ learning experiences. The study of [13] aimed to understand the drivers of success for higher education institutions (HEIs) in the online realm. They conducted text mining and sentiment analysis on online reviews from various business schools. The findings revealed that HEIs can enhance their online attractiveness by offering financial support for students’ cost of living, providing courses in English, and cultivating an international environment. These factors were identified as influential in shaping the perceptions and preferences of international students seeking to study abroad. Similarly, [14] observed through an analysis of social media posts that a predominant expression of negativity exists regarding the K to 12 Program in the Philippines, indicating a pressing need for its implementers to enhance its implementation measures.

B. Sentiment Analysis on Code-Mixed Data

Code-mixing is a linguistic phenomenon observed among multilingual individuals who prefer using their native language over English to express information [15]. Code-mixed text, whether spoken or written, is prevalent in multilingual societies including the Philippines [16]. Social media platforms like Facebook, Twitter, and online forums are common sources of code-mixed content. As stated in [17], sentiment analysis of monolingual text has been extensively studied, but code-mixing introduces additional complexity in analyzing the text’s sentiment due to its non-standard writing style.

As presented in the review of [18], the sentiment analysis of code-mixed and switched English with Indian languages achieved a range of 0.39 to 0.77 F1 Score wherein SVM, NB, and RF were the most used Machine Learning classifiers.

In the local setting, [19] utilized SentiWordNet 3.0 and FilCon, English and Filipino lexicons respectively to generate initial sentiment classifications. NB and SVM hybrid models were trained using these sentiment labels. To handle TagLish comments, a Code-Switching Point Detection Module was deployed to separate English and non-English words, which were then processed using the appropriate lexicons. The lexicon module achieved an overall accuracy of 55%, while the Naive Bayes before Support Vector Machines hybrid model achieved accuracies of 55% overall, 58% for English, 46% for Filipino, and 57% for a mix of Tagalog and English or TagLish. The Support Vector Machines before Naive Bayes hybrid model achieved accuracies of 61% overall, 70% for English, 54% for Filipino, and 54% for TagLish.

In [20], it was determined that existing bilingual embedding techniques were not suitable for processing code-mixed text. They emphasized the necessity of developing multilingual word embeddings specifically designed for code-mixed text processing. In [21], a study was conducted on code-mixed Persian-English data to perform sentiment analysis. They argued that the uniqueness of analyzing code-mixed data lies in the fact that the presence of English words within the Persian text can significantly impact the emotional expressions conveyed. This poses a challenge for Persian-only sentiment analysis models, as they may struggle to accurately interpret and generate correct outputs due to the mixed language context. They utilized Yandex and dictionary-based translation techniques to translate the code-mixed words present in the text. Additionally, pre-trained BERT embeddings, specifically mBERT was employed to represent the data which achieved an accuracy of 66.17% and an F1 score of 63.66%, surpassing the performance of the baseline models namely Naive Bayes and Random Forest methods. The effectiveness of mBERT in this setting was also assessed in a comparative study conducted by [22] on Hindi-English code-mixed data. The study revealed that mBERT outperformed ALBERT, vanilla BERT, and RoBERTa models, except for HingBERT-based models, which were specifically trained on Hindi-English code-mixed data.

III. METHODOLOGY

In this section, we present the methodology employed to collect, preprocess, and annotate the data, perform the fine-
tuning of the mBERT model, and evaluate the sentiment analysis results.

A. Data Collection

To capture information preceding the official implementation of UAQTE, data collection was conducted from April 1, 2017, to April 10, 2023. The process involved employing scraper libraries, enabling the systematic and organized retrieval of relevant data.

1) Facebook: We collected a total of 10,443 textual data from the posts that were set as public and official regional groups of CHEd. These groups, which include Ilocos Region (Region 1), MIMAROPA (Region 4-B), Bicol Region (Region 5), Western Visayas (Region 6), Central Visayas (Region 7), Eastern Visayas (Region 8), Zamboanga Peninsula (Region 9), Northern Mindanao (Region 10), Davao Region (Region 11), CARAGA (Region 13), National Capital Region (NCR), Cordillera Administrative Region (CAR), and the conversion of emojis to their textual representation.

2) Twitter: To collect a relevant dataset, we utilized specific keywords and hashtags such as “CHED UniFAST”, “#PIP grantee”, “TES grantee”, “#PINASkolar”, and “#TertiaryEducationSubsidy”. We were able to collect 1,112 raw tweets.

3) YouTube: In addition to collecting data from social media platforms like Facebook and Twitter, data was also collected from YouTube comments due to its significant presence as a popular video-sharing platform. To gather relevant content, YouTube videos were manually searched using specific keywords such as “CHED UniFAST”, “UniFAST TES”, “Libreng Edukasyon UniFAST”, and “RA 10931 Free Tuition”. The selection process involved identifying videos with high view counts, ensuring that the chosen content resonated with a wide audience and potentially represented popular sentiments and discussions surrounding the topic of interest. From the 49 videos, we were able to collect 1,777 raw YouTube comments.

B. Data Preprocessing

A total of 13,332 data points were collected for the purpose of data preprocessing. Since posts, tweets, or comments originate from various users and reflect individual opinions, URLs were the key identifier used to remove duplicates. Furthermore, posts that were unrelated to the implementation of UAQTE such as presidential election campaign-related content, underwent manual inspections and were excluded from the dataset. It is important to note that for YouTube data, the collection date was recorded and used to determine the timeframe of the content, whether it was posted months, days, or years ago. As mentioned in [23], BERT has shown optimal performance with little to no preprocessing. Therefore, a few preprocessing techniques were applied, including spelling corrections (e.g., “dhill” to “dahil” (English: because), “eguro” to “siguro” (English: maybe), normalization of special

characters, removal of white spaces, punctuations and symbols, and the conversion of emojis to their textual representation.

C. Data Annotation

From the preprocessed data, 4,650 (50%) data points were manually labeled and subjected to validation by experts. This annotated data serves as the benchmark and reference for sentiment classification. Following the approach adopted from [24], the data were classified into three categories: neutral, negative, and positive encoded as numbers 0, 1, and 2 respectively.

Positive sentiment encompasses expressions of satisfaction, happiness, admiration, interest, and gratitude. Neutral sentiment pertains to statements that do not exhibit any discernible sentiment, conveying information or facts. Negative sentiment includes expressions of dissatisfaction, anger, disappointment, sarcasm, mockery, and frustration regarding the UAQTE implementation. Table 1 shows the sample of annotated and validated data points.

D. Fine-Tuning of Multilingual BERT

The BERT model’s architecture is built upon the Transformer framework [25]. When provided with a sequence of up to 512 tokens as input, BERT generates a representation of the entire sequence, which can comprise one or two segments. The first token of the sequence, denoted as [CLS], holds a special classification embedding, and another special token, [SEP], is used to separate segments. To represent the entire sequence for text classification tasks, BERT leverages the final hidden state of the first token [CLS]. A classifier is added on top of BERT to predict the probability or sentiment of the label. During the training process, BERT is fine-tuned based on the task-specific training dataset. Fig. 1 illustrates the BERT fine-tuning model architecture for sentiment classification, adapted from [26], although in our case, sentiments were classified into three sentiment classes, namely neutral, positive, and negative.

While BERT has been pre-trained on English corpus, a multilingual version of BERT (mBERT) [9] has been trained jointly on Wikipedia on 104 languages including English, Tagalog, Waray, and Cebuano, hence, considering the code-mixed nature of the dataset, we downloaded the BERT multilingual base model (cased) from Hugging Face via ktrain [27], a lightweight wrapper for the deep learning library TensorFlow Keras on Google Colaboratory that provides

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<th>Content</th>
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<tr>
<td>CHED Chairman Popoy De Vera discusses the guidelines on the new degree programs approved for limited face-to-face classes.</td>
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<td>Pero ba’t wala pong budget ang mga 2021-2022 applicants new grantees? Saan po napunta ang budget? (English: But why do the 2021-2022 applicants who are new grantees have no budget? Where did the budget go?)</td>
<td>1</td>
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<tr>
<td>Thank you so much CHED- UNIFAST Tertiary Education Subsidy.. And to ACLC College of Bukidnon ..</td>
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NVIDIA Tesla T4 12 GB of RAM, and Intel® Xeon® Processor.

Among the recommended values of learning rates when fine-tuning on specific tasks were 2e-5 [28], 5e-5, and 3e-5 using batch sizes 16 and 32 at 4 epochs, [29]. Further, a maximum length of 160 was set considering the 95th percentile value obtained from the descriptive statistics of the word lengths, to ensure a manageable memory size and sequence length. To address the class imbalance, class weights were applied to the three classes.

E. Model Evaluation

To evaluate the performance of the sentiment analysis models, two key metrics were utilized: F1 score and accuracy. Subsequently, the best-generated model will be utilized to predict the sentiment classes of the remaining data.

IV. RESULTS AND DISCUSSION

The fine-tuned mBERT model exhibited varying levels of performance across different combinations of batch size and learning rate. Table II presents the accuracy and F1 scores achieved by each model configuration.

In contrast to the experiment conducted by [19], which removed stopwords, our approach involves minimal preprocessing of texts and includes stopwords. Additionally, we take into account the textual representation of emojis, setting our study apart from the research conducted by [22]. By considering emojis, we aim to capture a more nuanced understanding of sentiments expressed in social media data, enhancing the contextual analysis capabilities of mBERT. Model 1, with a batch size of 16 and a learning rate of 2e-5, emerged as the best-performing model with 80.21% accuracy and an F1 Score of 81.14%, surpasses the findings of related studies discussed in Section 2 in terms of their F1 Score and Accuracy. Hence, this model was used to classify the remaining data points.

Findings revealed that 2,100 (22.58%) were neutral, 1,647 (17.70%) were negative, and 5,553 (59.70%) were positive, indicating that majority of the social media users view UAQTE as positive or beneficial in the country.

A user from Twitter expressed positive feedback towards UAQTE, “Thank you God nakasali ako sa ched-unifast. Malaking tulong na po ito” (English: “Thank you God I was able to be a member of ched-unifast this is helpful”). Neutral contents were composed of announcements and updates regarding subsidies. A Facebook post expressing frustration with CHED reads, “nakakaloka yung CHED beh. patapos na ako, wala pa din yung sa UniFAST hahahaha” (English: “It’s crazy, CHED! I’m almost done with my studies, but I still haven't received anything from UniFAST, hahahaha.”). This post conveys a sense of disappointment and humor, suggesting that the individual has been anticipating support or benefits from UniFAST but has yet to receive them.

As shown in Fig. 2, the sentiments towards the implementation of the UAQTE have shown interesting dynamics over the years.

From 2017 to 2018, the sentiments were characterized by a relatively low number of neutral sentiments, while negative sentiments were more prevalent which can be attributed to the initial challenges in implementing the program as discussed by [5].

In 2019, the distribution of sentiments experienced a significant shift. The number of neutral sentiments increased notably, possibly due to improvements or adjustments made to address previous concerns. Negative sentiments decreased, while positive sentiments remained relatively low, indicating ongoing debates or skepticism surrounding the program.

In year 2020 witnessed a relatively balanced distribution of sentiments across the three categories. This suggests a period of relative stability or reduced controversy, where public opinion was less polarized. However, in 2021, sentiments experienced a sharp rise across all categories. Negative sentiments and positive sentiments increased substantially, while neutral sentiments also reached a significant level. This is likely due to the release of subsidies to beneficiaries, with some expressing gratitude for receiving them while others express disappointment for not receiving them on time. This could be also attributed to the pandemic's impact on the education sector. Inquiries and concerns regarding the bank application of the grantees were also found. The trend continued in 2022 with sentiments reaching even higher levels across all categories. This suggests ongoing discussions, policy
developments, or increased public attention toward the program, resulting in a more polarized sentiment landscape. Finally, in 2023, sentiments experienced a notable decline across all categories but experienced a substantial rise in negative sentiments which was likely due to reports of alleged misuse of funds intended for the implementation of the program [30].

These fluctuations in sentiments over the years highlight the dynamic nature of public opinion and the influence of various factors, such as policy changes, media attention, and public discourse, on the sentiments towards the implementation of the UAQTE.

V. CONCLUSION

UAQTE has been a notable policy shift in the Philippines. While the program aims to provide free higher education and tertiary education subsidies to eligible Filipino students, its feasibility was questioned and assessed.

This study analyzed the sentiments expressed on social media platforms such as Facebook, Twitter, and YouTube, where code-mixing is prevalent, regarding the implementation of UAQTE policy in the Philippines between April 1, 2017, and April 10, 2023. We used a fine-tuned mBERT model to perform sentiment analysis on the collected data. With minimal preprocessing, mBERT achieved an accuracy and F1 score of 80.21% and 81.14%, respectively. These results outperformed the existing studies, as outlined in Section 2, demonstrating the effectiveness of mBERT in handling code-mixed data for sentiment analysis and revealing a dominant positive sentiment perceived by social media users regarding the implementation of UAQTE.

The prevailing positive sentiments expressed gratitude towards the UAQTE, which suggests that UAQTE has been beneficial for the intended stakeholders. However, the analysis also identified the presence of negative sentiments related to the late distribution of subsidies, alleged misuse of funds, and difficulties in the application of bank accounts such as erroneous online application portal of the beneficiaries. The neutral sentiments mostly involved announcements, news, and updates regarding the release of the subsidies. Addressing these concerns can help enhance the program’s effectiveness and ensure that it continues to meet its objectives.

VI. LIMITATIONS AND FUTURE WORK

While this study contributes significant insights, it is essential to acknowledge its limitations. The analysis is limited to social media data and may not fully represent the entire population’s sentiments. Moreover, analyzing the sentiment of specific demographic groups, such as students or parents, may provide additional insights that could inform more targeted policies and initiatives. Additionally, exploring the use of cross-lingual language models (XLMs) like XLM-RoBERTa and hybrid BERT models for sentiment analysis may improve the accuracy and reliability of the results with code-mixed data.

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Criminal Law Risk Management and Prediction Method based on Echo State Network

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Abstract—Criminal law plays an important role in maintaining social security and achieving effective social control. However, criminal law has hidden risks that cannot be ignored at the legislative, judicial and theoretical levels. This paper starts from all aspects of criminal law, analyzes criminal law risk and its management measures, and predicts and analyzes criminal law risk through echo state network model. The prediction results of the echo state network model fit well with the actual situation, and its verification can provide reference for the study of criminal law risk prediction and management systems. Legislative risk and theoretical risk belong to social factors and are also fundamental risks of criminal law. Judicial risk is mainly manifested in the level of judicial power. Criminal law is closely related to the political environment, social system, economic system, etc. In criminal law legislation, we should pay attention to the balance between criminal law rules and realistic social functions, and properly control social risks, so as to avoid the criminal law risks brought by the establishment of risky criminal law, and provide the necessary guarantee for the national security system.

Keywords—Echo state network; model; criminal law; risk prediction; risk prediction

I. INTRODUCTION

Since the implementation of the reform and opening-up policy in 1978, both China's national home-building and criminal rule of law processes have made great strides; in particular after the comprehensive revision of the Criminal Code in 1997, my country has successively introduced 1 single criminal code, 10 criminal code amendments, into one criminal code A new era of active legislation. Overall, over the past two decades, Criminal law intervention is early and active, and the criminal circle continues to expand. It has become the main development trend of my country's criminal law. This is for China's current Criminal law concept has had a significant impact, risk criminal law concept, positive criminal law Legislative view, preventive criminal law view, and functionalist view of criminal law have appeared one after another field, trying to theoretically interpret and judge this trend. Approximately In terms of criminal law, the above-mentioned concept of criminal law is explanatory or affirmative to the expansion of criminal law attitude and require moderate restrictions. At the same time, critical attitude insights hold that this (or "some") expansion of criminal law is a in "symbolic legislation", "emotional legislation" or new criminal law instruments" doctrine, and even advocated that “our country should stop the criminalization of criminalization "legislation" [1,2]. Some of the above differences of opinion are principled, and they are quite. To a certain extent, it explains our basic criminal law about the development of China's current criminal law. There is also a need for an in-depth review of the legal concept. The idea of criminal law is that people The nature, function, crime, punishment, relationship between crime and punishment, system of criminal law understanding, view, mentality and value of a series of issues such as determination and implementation general term for orientation. A well-established view of criminal law is important for judgment and instruction. It is of great significance to guide the development of criminal law, reviewing the development since the reform and opening up Chinese social changes, correct interpretation and evaluation of the criminal law since 1997 The development of criminal law legislation is conducive to the scientific level of criminal law legislation improve [3,4].

It's essential to understand the evolution of China's criminal law within the context of the country's broader societal transformations. Since the initiation of China's reform and opening-up policy in 1978, remarkable strides have been made in both national construction and the administration of criminal law. This progress became particularly evident with the comprehensive revision of the Criminal Code in 1997, marking the beginning of an era characterized by proactive criminal law legislation. This period saw the implementation of one single criminal law and ten subsequent amendments.

Despite these advancements, the period following China's reform and opening-up does have its share of limitations. The promulgation of the Criminal Code in 1979 represented a departure from a 30-year period without a criminal code. While this was a significant improvement over the prior reliance on policy-driven case handling, the adoption of a planned economy and the prevalent thinking of centralized societal control by government forces led to the "instrumental criminal law concept" taking a dominant position. The Criminal Code didn't stipulate the principle of statutory crime and punishment, instead providing an analogy system. This empowered authorities to dictate the definition of crimes, thereby limiting citizens' freedoms. However, with the comprehensive revision of the Criminal Code in 1997, the principle of legality was explicitly stipulated, which reflected the "view of criminal law of civil rights" to a considerable extent. Additionally, the in-depth progress of China's reform and opening up coincided with academic reform in the realm of criminal law. This paved the way for a gradual deviation from the Soviet Union's "nationalist criminal law concept," marked by strong class struggle undertones, and led to the absorption of Western criminal law concepts. Influenced
significantly by the Enlightenment and modern criminal law ideologies, especially those initiated by Beccaria, the "classical liberal view of criminal law" became widely recognized. This view, advocating for human freedom, opposing feudal cruelty, and favoring state power restrictions, has significantly impacted China's criminal law policies. However, it's critical to acknowledge that, much like everything else, criminal law is not without its risks, which are present in legislation, judiciary, and theory. These risks can directly impact lives, highlighting the necessity for suitable control measures.

The creation and dissemination of legal risk and the terror created and fostered by risk are equally thorny issues. While we recognize and promote the continuous impact of science on the law, we must also pay attention to the new risks that the new development of science may bring to the criminal law. The new risk is a broader view of the "scientific generalization" alienated from the healthy development of science and its possible adverse effects. For a long time, science has had a self-evident "correctness" for the general public because of its great achievements [7]. In a lot of difficult situations, science is often used as the ultimate endorsement to provide certainty for a certain solution. Along with the great achievements of science, a large number of pseudo-scientific and non-scientific contents have been mixed into the category of science, which is essentially an anti-scientific force. This kind of generalized science, in the name of science and with the veil of science, invades the category stipulated by the essence of criminal law, and has begun to produce bad tendencies [8,9]. Therefore, we must further clarify the difference between science and criminal law and the boundary of science's role in criminal law formed based on this difference. We must further analyze the risks of generalization of science to criminal legislation and criminal justice, as well as the possible risks and consequences based on such risks potential adverse effects. To this end, this paper analyzes and demonstrates the challenges existing in our country's criminal law, especially the problems that scientific generalization may promote and breed, and clarifies how criminal law, as a tool and means of national governance, should better play its role in the new era. The role of punishing crimes and building collective consensus is particularly important [10].

At present, criminal law risk prediction is mainly based on the research of past criminal law, as well as sampling analysis of the actual situation, and judging the future risk of criminal law has been controlled. This paper uses the echo state network (ESN) model to predict criminal law risks, and puts forward corresponding control measures, which provides a scientific basis for criminal law risk prediction, and is of great significance for the study of criminal law risk characteristics and management measures.

This work aims to explore and critically analyze the evolution of China's criminal law, particularly in light of the expansive and active intervention of criminal law in the country's legislative system since the comprehensive revision of the Criminal Code in 1997. The main ideas and the actual purpose embedded in this work involve charting the progression of China's criminal law, understanding the implications of its expansion, and seeking effective control mechanisms, thus providing valuable input to the formulation of strategies ensuring the effective functioning of the criminal law in contemporary China.

II. CRIMINAL LAW RISK ANALYSIS AND CONTROL

A. Criminal Law Risk Analysis

Criminal law plays an important role in maintaining social security and achieving effective social control. However, criminal law also has hidden risks that cannot be ignored. In response to the problem of prediction accuracy, German Professor Jaeger proposed the ESN in 2001, and the prediction accuracy is 2400 times higher than that of the traditional network. Therefore, this paper uses the ESN model to predict criminal law risks.

First, the criminal legislation process may be deficient at the level of rational discourse. In an era of frequent social risks and emphasis on democratic legislation, the impact of the public's safety demands on legislation cannot be ignored. When social risks show an intensifying trend, the public tends to rely on criminal legislation to help them achieve their safety needs. If legislators fail to respond to public demands and do not take a positive stance against risks, they will lead to dissatisfaction and criticism from the public. Secondly, the lack of rational negotiation in criminal legislation can easily turn preventive criminal law into symbolic legislation and passion legislation. Because preventive criminal law focuses on responding to people's expectations for safety, the function of preventive criminal law to manage social risks is often paid the greatest attention or even exaggerated, while the danger of its possible violation of civil liberties and rights is infinitely reduced or even ignored. Finally, excessive criminalization can easily cover up the defects of other social governance systems. Criminal law is not the best means to govern society, nor is it the least costly means. When other social governance means other than criminal law can effectively reduce social risks, other social governance means other than criminal law should be given priority. The attitude of preventive criminal law to take the initiative to manage social risks has certainly increased the public's sense of security to a certain extent. It is even undeniable that it has also reduced social risks in practice.

For defensive criminal law, its setting is ambiguous to a certain extent, and it is easy to breed a crisis of arbitrary judicial arbitrariness. In the provisions of preventive criminal law, legislators often use evaluative concepts, such as illegal use, extremism, and drunk driving. Although the evaluative concept can enhance the acceptability of criminal law norms, because it is a value statement of things, different judgment subjects are prone to different judgment conclusions. The extensive use of evaluative concepts creates conditions for the expansion of judicial power, which can easily breed a crisis of judicial arbitrariness. Second, some preventive criminal law provisions. There is ambiguity in the provision of crimes. The fuzzification of crime regulations objectively reduces the elements of crime that need to be proved, reduces the difficulty of proving a crime, and can more effectively punish dangerous behaviors. However, it reduces, transfers or even cancels the burden of proof on the perpetrator's subjective
guilt, which can easily lead to unjust or completely wrong penalties, and also makes it difficult to implement the principle of the unity of subjective and objective in preventive criminal law. The presumption of guilt lays hidden dangers, seriously undermining the criminal law's function of guiding judges' adjudication and guiding the behavior of the public [11].

The primary goal of criminal law is to maintain social security and stability by expanding the state's power to punish offenders. However, if we excessively prioritize the state's objective of maintaining social security, it can potentially jeopardize the freedom and rights of individuals. Firstly, criminal law serves the dual purpose of safeguarding both legal interests and human rights, which are mutually reinforcing. The protection of legal interests is widely accepted as the fundamental aim of criminal law within the academic community. Nevertheless, this does not negate the fact that criminal law also plays a role in protecting human rights while fulfilling its function of safeguarding legal interests. Secondly, the concept of preventive criminal law poses a challenge to the protective function of criminal law. Proponents of preventive criminal law argue for shifting the threshold of prevention forward, so that criminal liability is no longer based solely on actual harm or the appearance of specific danger, but rather on the offender's failure to fulfill their expected social role. Finally, as the protective function of criminal law weakens, so too does its ability to uphold human rights [12].

The above describes the risks of criminal law from three aspects: legislation, judiciary and theory of criminal law. Legislation is the beginning of criminal law risk, judiciary is the embodiment of criminal law risk, and theory is the foundation of criminal law risk.

B. Criminal Law Risk Management

Criminal law risk management and control should explore targeted criminal risk prevention and control programs from three perspectives: pre-prevention, in-process control, and post-event crisis comprehensive disposal.

1) Establish a three-dimensional legal risk prevention and control system: First of all, it is necessary to combine the components of static and dynamic criminal legal risks to build a corresponding risk prevention and control system. Among which, static elements mainly refer to the main body and its behavior activities and operating environment, and the latter refers to the causes, specific events, carriers and risk effects of criminal legal risk. Secondly, the three-dimensional criminal legal risk prevention and control system should be constructed from the two dimensions of crime and victim, so as to ensure that the system can achieve the prevention and control effects of the prevention and control in advance, control in the event and crisis disposal after the event, so as to avoid criminal legal risks.

2) Criminal legislation should follow the principle of modesty: This principle requires that when criminal legislation regulates behavior, the scope and degree of punishment should be appropriately clarified. That is, if other laws are used to suppress a certain illegal behavior and can achieve the effect of comprehensively protecting the legitimate rights and interests of the public, it should not be included in the criminal law. Relatively speaking, if a lighter sanction method can effectively suppress a certain criminal behavior and protect the legitimate rights and interests of the public, a heavier sanction method should not be prescribed. In addition, judicial organs should also adhere to this principle of modesty in judicial practice. Except for alienated risk-based crimes such as fund-raising fraud, other Internet crowdfunding activities that are within the legal red line should be given certain encourage and support the development of new things, especially for those crowdfunding behaviors whose nature is not clear and the legal provisions are unclear, the punishment at the criminal law level should be reduced as much as possible. However, severe criminal sanctions should be imposed on those crowd-funding behaviors that have great social impact and harm and have numerous victim groups and have been brought into the scope of criminal law [13].

3) Criminal legislation should respect man's subjective status: No matter how the criminal law develops, it must always respond to the issue of human rights. It is an appropriate policy to take a relativistic position on the protection of human rights from a rational point of view. Whether it is criminal legislation or criminal justice, it is necessary to take a restrained attitude towards the protection of power, because criminal law itself is full of factors of public power, and in addition to criminal law, public power can also use other means and measures to restrict the human rights of citizens. Therefore, restraint and prudence of criminal law can correct the easy expansion of public power to a certain extent, and make it continue to play the role of the last line of defense. Therefore, the study of human rights in criminal law is to return to the human rights of individuals, to avoid the use of group or collective human rights to obliterate individual human rights, to pay attention to clarifying the difference between the relativity of the nature of human rights and the improper derogation of human rights, and to avoid using a zero-sum game of rights and interests within individuals to cover up the erosion of power by power.

The control of criminal law is a comprehensive measure. First, there should be a principle of control, and secondly, a risk control system should be established with people as the main body, so as to provide good guidelines for the control of ideas.

III. MODEL INTRODUCTION

At present, more mature network learning methods include Bayesian, wavelet analysis, support vector machine learning and neural network. However, after continuous experiments and application verification, it is found that most of the algorithm also has the problem of forecasting accuracy is not enough, even some algorithms for application scope restrictions, limited to a fixed application scenario, cannot training learning in other areas [14]. In response to the problem of prediction accuracy, German Professor Jaeger
The proposed ESN in 2001, its echo state network, also known as reserve pool calculation, uses a reserve pool composed of randomly sparsely connected neurons as a hidden layer for high-dimensional and nonlinear representation of input, and the prediction accuracy is 2400 times higher than that of the traditional network. Therefore, this paper uses the ESN model to predict criminal law risks [15,16].

A. Model Principle

The ESN includes an input layer with K input neurons, a reserve pool with N neurons, and an output layer with L output neurons. ESN differs from other neural networks in that it includes a dynamic reserve pool with many randomly sparsely connected neurons and a simple training process in which only the output weights are adaptive [17]. The structure diagram of ESN is shown in Fig. 1.

![ESN structure diagram.](image)

The state of the reserve pool is updated according to formula (1):

\[
x(n + 1) = f(W^{in}u(n + 1) + W^{res}x(n) + W^{back}y(n))
\]  
(1)

where: \(u(n + 1)\) is the input at time \(n + 1\); \(x(n)\) is the state of the reserve pool neuron at time \(n\); \(f\) is the activation function of the reserve pool neuron; \(W^{in}\) is the input weight matrix; \(W^{res}\) is the reserve pool weight matrix; \(W^{back}\) is the feedback weight matrix. To make the ESN work normally, the ESN should have the echo state attribute, and the spectral radius of \(W^{res}\) should be less than 1. To guarantee the echo state property, the final \(W^{res}\) will usually be scaled according to formula (2).

\[
W^{res} = \alpha(W^{res} / |\lambda_{max}|)
\]  
(2)

\(\alpha\) is the scaling parameter, between 0 and 1, and \(\lambda_{max}\) is the spectral radius of \(W^{res}\). The output of ESN can be represented as follows:

\[
y(n) = f^{out}(W^{out}x(n))
\]  
(3)

where: \(f^{out}\) denotes the activation function of the neuron; \(W^{out}\) is the output weight \(L \times (N+K)\). During the training process, the states of the neurons in the reserve pool are collected into a state matrix \(M\), see equation (4):

\[
M = \begin{bmatrix}
X^T(t) \\
X^T(t+1) \\
\vdots \\
X^T(t+k-1)
\end{bmatrix}
\]  
(4)

Then, the corresponding output vectors are collected into the target matrix \(T\), see equation (5):

\[
T = \begin{bmatrix}
y(t) \\
y(t+1) \\
\vdots \\
y(t+k-1)
\end{bmatrix}
\]  
(5)

where \(t\) is the length, \(K\) is the sample size, \(f^{out}\) is an identity function, using the pseudo-inverse algorithm to calculate the output weight matrix \(W^{out}\), see formula (6):

\[
W^{out} = M^+T
\]  
(6)

where \(M^+\) is the generalized inverse of the state matrix \(M\), and formula (6) is modified as follows:

\[
W^{out} = (M^TM)^{-1}M^TT
\]  
(7)

B. Model Parameter Optimization

The essence of the model is also a recursive structure generated by random motion. There are four basic parameters to be optimized: 1) The spectral radius (SR) of the network connection weight matrix in the reserve pool. SR is not only the largest eigenvalue of the absolute value of the internal connection weight function matrix, but also an important parameter to ensure the security and stability of the entire network. 2) The size \(G\) of the reserve pool. The larger \(G\) is, the more accurate the dynamic description of the entire network model is. 3) The input cell scale (IS) of the reserve pool, that is, the cell scale control factor. Normally, the larger the object scale that the network model needs to deal with, the larger the IS. 4) The sparsity (SD) of the reserve pool. SD is mainly used to represent the ratio of the two interconnected neurons in different reserve pools to the total number of neurons. The larger the value, the stronger the linear approximation ability [18,19].

Among the techniques employed to optimize the four parameters of the ESN model, commonly used methods include cross-validation and grid search. However, these parameter optimization methods still exhibit technical limitations in practical calculations. For instance, they often involve a large number of computations and result in long operation times. To overcome these challenges, the genetic algorithm is utilized as an efficient computational method for globally searching optimal solutions within a network. The genetic algorithm offers several advantages. Firstly, it effectively avoids the network search model from converging...
towards local optima. Additionally, it utilizes adaptive control search to obtain the optimal solution. In order to achieve better parameter optimization within the reservoir pool, the genetic algorithm is selected to fine-tune the parameters [20,21].

IV. MODEL ESTABLISHMENT AND APPLICATION

A. Data Sources

According to the development of society and the actual situation of the criminal law process, criminal law risk mainly includes three first-level indicators: legislative risk $U_1$, judicial risk $U_2$ and theoretical risk $U_3$, as well as legislative constitutionality risk $u_1$, legislative scientific risk $u_2$, and legislative democracy risk $u_3$, legislative timeliness risk $u_4$, judicial disclosure risk $u_5$, judicial integrity risk $u_6$, judicial fairness risk $u_7$, low judicial efficiency risk $u_8$, theoretical objectivity risk $u_9$, theoretical purpose ambiguity $u_{10}$, theoretical deviation $u_{11}$ and theoretical violation $u_{12}$, among which, legislative risk $U_1$ and theoretical risk $U_3$ mainly depend on the degree of social development, which belong to social factors and are also the fundamental risks of criminal law. Judicial risk $U_2$ is mainly manifested in the level of judicial power. These indicators can reflect the risk factors in the process of China’s criminal law from generation to implementation.

Through statistical analysis of the probability of occurrence of each risk index and the value of the degree of influence, the relationship between the probability of occurrence of risk and the value of the degree of harm is obtained (Fig. 2). According to the analysis, the degree of impact and the possibility of risk occurrence are divided into five levels, among which the degree of impact is divided into “intolerable”, “significant impact”, “tolerable”, “probable” and “negligible”. The probability of risk occurrence is divided into "very likely", "probable", "possible", "impossible", when the probability of risk occurrence is “negligible”, it indicates the possibility of the risk occurrence Low, the risk can be ignored; when the probability of occurrence is "probable" or "probable", the risk factor needs to be paid more attention; when the probability of occurrence is "very likely", it means that this risk is easy to occur, and great attention is needed. According to the risk analysis of criminal law, legislative risk $U_1$ and theoretical risk $U_3$ belong to social factors and are also fundamental risks of criminal law [18]. Risks have a high probability of occurrence and have a significant impact on society. Judicial risk $U_2$ is mainly manifested in the strength of the judiciary and the probability of occurrence.

B. ESN Training and Testing

First, normalize the dataset to the $[0,1]$ interval, which is expressed as:

$$f : x \rightarrow y = 2 \times \frac{x - x_{\min}}{x_{\max} - x_{\min}} + (-1)$$ (8)

In network training, a limit is added to enhance the stability of network training. The limit is a uniformly distributed number on $[-0.002, 0.002]$, and the mean square error of training ESN is the smallest. In order to improve the performance of the network, many experiments were conducted on the four parameters of ESN: the number of SR neurons $N$, the SR reality $s_d$, the spectral radius $\rho$ of $w$ in SR, and the SR input connection weight scaling scale $IS$. According to the empirical value, $(N \in [2000], s_d \in [0.01, 0.05], \rho \in [0.5, 0.98], IS \in [0.01, 1])$ to set the final optimal. The parameter combination is shown in Fig. 3.

![Fig. 2. The relationship between the probability of occurrence of risk and the degree of harm.](image)

C. Simulation Experiments and Results

The accuracy of model prediction is not only related to the structural parameters of the model itself, but also depends on whether the selection of model predictors is reasonable. Therefore, this paper draws on the prediction factor selection method of previous research, and uses autocorrelation and partial correlation to analyze the predictors. The sample autocorrelation analysis is shown in Fig. 4, and the partial correlation analysis is shown in Fig. 5.

![Fig. 3. Optimal parameter value diagram.](image)

![Fig. 4. Autocorrelation analysis plot.](image)
This paper divided the processed dataset into two parts. Meanwhile, Root Mean Squared Error (RMSE), Efficiency (E) and Correlation Coefficient (CORR) were selected to evaluate the prediction results. The closer RMSE is to 0, and the closer E and CORR are to 1, the higher the accuracy of the model [20]. The specific formulas are as follows:

\[
RMSE = \sqrt{\frac{1}{n} \sum_{i=1}^{n} (x_i - y_i)^2}
\]

(9)

\[
E = 1 - \frac{\sum_{i=1}^{n} (x_i y_i)}{\sum_{i=1}^{n} (x_i - \bar{x})^2}
\]

(10)

\[
M = 1 + \sum_{i=1}^{n} (x_i + y_i)
\]

(11)

\[
CORR = \frac{\sum_{i=1}^{n} x_i y_i - \frac{1}{n} \sum_{i=1}^{n} x_i \sum_{i=1}^{n} y_i}{\sqrt{\left(\sum_{i=1}^{n} x_i^2 - \frac{1}{n} \left(\sum_{i=1}^{n} x_i \right)^2\right) \left(\sum_{i=1}^{n} y_i^2 - \frac{1}{n} \left(\sum_{i=1}^{n} y_i \right)^2\right)}}
\]

(12)

This paper uses MATLAB software to program the ESN. The key to model prediction is the selection of model parameters, and the key parameter of ESN is the scale N of the reserve pool. The optimal scale of this parameter is obtained through repeated experiments (Fig. 6). The size of the reserve pool N mainly includes 50, 60 and 65. The predicted performance results of different reserve pool sizes are shown in Fig. 7. According to the comparison of the prediction performance of different reserve pool sizes, this paper chooses the reserve pool size as 65, that is, 65 different types of criminal laws are selected as the reserve pool, the other parameters are set as follows: the number of training set samples is 65, and the number of test set samples is 60, the number of samples in the initialized reserve pool is set to 150, SD is 10%, W spectral radius is 0.8, IS is 0.3, and the excitation function is selected as ‘tanh’. Through the operation and mathematical analysis of the model, the probability distribution of 12 secondary risk indicators is finally obtained (Fig. 8).

In addition, in order to verify the application effect of ESN model in criminal law risk, this paper also uses BP neural network for comparative analysis. In the last layer of building the BP neural network, the output feature vector of the RBM (Restricted Boltzmann Machine) network is regarded as the output vector of the BP network, which can effectively supervise and train the entity relationship analyzer. The set BP network can only ensure that the mapping of weights to feature vectors in this level is optimal, but cannot ensure that the mapping of feature vectors in the entire DBN (Deep Belief Network) is optimal. Therefore, the BP network may propagate the problematic information to the RBM of each
layer, and play a small adjustment role for the entire DBN. The RBM network training model is the initialization process of the weights of a deep BP network. This process enables DBN to overcome the problems of increased training time and prone to local optima due to random changes in the initialization of BP network weights. It can be seen that the application of the ESN model in criminal law risk has many advantages over the application of the BP network model, which is also an affirmation of the application of the ESN model [21].

V. ANALYTICAL DISCUSSION

According to the distribution of secondary risk indicators (Fig. 8), the maximum value is 0.4241, which corresponds to u12. According to the second-level indicators of legislative risk, the risk factor represented by it is legislative scientific risk u12, and this influencing factor is very important. The second value is 0.2151 corresponding to u13, which represents the risk factor of legislative democratic risk u21, which plays an important role in the formulation of criminal law, and the second value corresponding to the risk of judicial integrity u22 is 0.1923. Other legislative constitutionality risk u11 is 0.0132, legislative timeliness risk u14 is 0.0762, judicial fairness risk u23 is 0.0548, judicial efficiency risk u24 is 0.0312, theoretical objectivity risk u31 is 0.0431, theoretical purpose vague u32 is 0.0924, theoretical deviation u33 is 0.0872 and the theoretical violation u34 is 0.0432. All three of these indicators are major risk factors for criminal law, and greater attention is being paid to them. All other values are less than 0.1, indicating that other factors have great influence or can be ignored. By analogy, the final weights are ranked as follows: purpose ambiguity u32, theoretical deviation u33, legislative timeliness u14, interoperability hinders great justice u23, theoretical violation u34 and theoretical objectivity u31, etc. Legislative risk is the most critical, and also the beginning and foundation of the law. Regarding criminal law risk management and control, we should have corresponding measures. First, we should take people as the main body and the rule of law as the criterion, establish a good risk prevention system, and do a good job in risk management and control in advance to prevent problems before they occur. To sum up, in the practice of criminal law legislation to judicial practice, the law not only has sufficient tenacity, but also has many risk factors. Among them, legislative risk is the most critical, and it is also the beginning and foundation of the law, which should be considered in future law formulation. Its associated risks provide a good basis for the enforcement of the law.

To compare the performance of ESN in predicting criminal law risks, this study also employs BP neural network to forecast the aforementioned models. Fig. 9 illustrates the prediction outcomes of both ESN and BP neural network for the three models. By comparing the evaluation metrics RMSE, E, and CORR, it becomes evident that ESN outperforms BP neural network in terms of predicting criminal law risks. Additionally, due to the simplified regression calculation involved in the internal adjustment of the ESN reservoir pool, its running time is significantly reduced compared to BP neural network. In regard to the individual prediction performance of ESN, the values of RMSE are arranged in ascending order as M3, M2, and M1; whereas the values of E are ranked in descending order as M3, M2, and M1. Similarly, the values of CORR are sorted from highest to lowest as M3, M1, and M2. Considering all three indicators, M3 demonstrates the highest prediction accuracy, followed by M2, and finally M1. This can be attributed to the fact that M3 incorporates four input factors, M2 includes three, while M1 only has two. Hence, the accuracy of ESN prediction exhibits a certain correlation with the number of effective input factors, indicating that a higher number of relevant inputs leads to improved prediction accuracy. It is worth mentioning that in this study, the ESN model reserve pool size is set at 65, which proves to be suitable and yields high model accuracy.

![Fig. 9. Model predictions.](image)

The model's prediction serves as a tool, but in order to comprehend and manage the risks associated with criminal law, it is crucial to reflect upon the theory of criminal law. The concern regarding the relationship between law and society stems not only from the limitation of a narrow internal perspective in criminal law research but also from recognizing that the evolutionary power of the legal system does not primarily arise from internal forces alone. As the external social environment grows increasingly complex, the study of criminal law must incorporate the dimension of social structural change to observe and address how the criminal law system can be adjusted and reconstructed.

In bridging the gap between social theory and criminal law theory, researchers should strive to understand and grasp the fundamental consensus within contemporary social theory. Furthermore, they should integrate this consensus into the legal dogmatic system in an identifiable manner. This integration facilitates the self-renewal of the criminal law system. By examining the efforts made by Heike, the founder of interest law, to methodologize and technically develop Yelling’s teleological thought, we discover the potential for academic innovation—a combination of thought and technology—to drive the systematic renewal and advancement of criminal law.

VI. CONCLUSION

The transitional modern society is characterized by a plethora of highly intricate and uncertain risks. The number and uncertainty of these risks are constantly on the rise. This is evident in three aspects: at the legislative level, an excessive emphasis on preventive criminal law in social governance can
lead to concerns about excessive criminalization; at the judicial level, the vague standards for determining crimes conceal the risk of judicial arbitrariness; at the theoretical level, relaxing restrictions on the state's penal power can easily result in an imbalance between societal protection under criminal law and the safeguarding of human rights. These developments in criminal law arise not only as a response to the security needs of citizens during periods of social transition but also due to the failure to establish new standards that can restrict criminal legislation after breaking free from the constraints imposed by traditional criminal law theories. To address potential risks associated with criminal law, it becomes essential to establish reasonable boundaries for preventive criminal law. The prediction of criminal law risk based on ESN model established in this paper mainly draws the following three conclusions:

1) In order to better predict criminal law risks and provide corresponding guarantees for social security, the ESN model proposed in this paper has further improved the accuracy of various risk predictions. After verification, it can provide reference for criminal law risk prediction and management system research.

2) The experimental results show that the ESN model proposed in this paper has a good fitting effect with the actual situation, and is better than BP model in terms of error and fitting degree.

3) Criminal law is built on the background of social risk. In the risk society, the variability, complexity, multiple occurrence and unpredictability of social risk have a great impact on the original criminal law system. Therefore, we should think about the criminal law from three aspects: legislation, judicature and theory, analyze the foreseeable risks of the criminal law, and put forward corresponding risk management measures to ensure the safety of the national criminal law system.

4) Legislative risk $U_1$ and theoretical risk $U_2$ belong to social factors and are also fundamental risks of criminal law. Judicial risk $U_3$ is mainly manifested in the level of judicial power. The influencing factors of legislative scientific risk are very important. The formulation of democratic risk criminal law plays an important role, followed by judicial integrity.

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Abstract—Digitized documents are increasingly becoming prevalent in various industries. The ability to accurately classify these documents is critical for efficient and effective management. However, digitized documents often come in different formats, making document classification a challenging task. In this paper, we propose a multimodal deep learning approach for digitized document classification. The proposed approach combines both text and image modalities to improve classification accuracy. The model architecture consists of a convolutional neural network (CNN) for image processing and a recurrent neural network (RNN) for text processing. The output features from the two modalities are then merged using a fusion layer to generate the final classification result. The proposed approach is evaluated on a dataset of digitized documents from various industries, including finance, healthcare, and legal fields. The experimental results demonstrate that the multimodal approach outperforms single-modality approaches, achieving high accuracy for document classification. The proposed model has significant potential for applications in various industries that rely heavily on document management systems. For example, in the finance industry, the proposed model can be used to classify loan applications or financial statements. In the healthcare industry, the model can classify patient records, medical images, and other medical documents. In the legal industry, the model can classify legal documents, contracts, and court filings. Overall, the proposed multimodal deep learning approach can significantly improve document classification accuracy, thus enhancing the efficiency and effectiveness of document management systems.

Keywords—Scanned documents; classification; document categorization; artificial intelligence; machine learning; deep learning

I. INTRODUCTION

With the rise of digital transformation, documents in various industries are now digitized for easy access and management. Digitized documents come in various formats such as PDF, JPEG, PNG, and many others, which makes document classification a challenging task. The ability to accurately classify these documents is crucial for efficient document management. The traditional approach of manual classification is time-consuming and prone to errors [1]. Hence, the development of automated document classification systems has become a critical need for organizations.

Recent advances in deep learning have significantly improved the accuracy of document classification [2]. Deep learning algorithms have been applied to various document classification tasks, such as sentiment analysis, topic modeling, and spam detection [3–4]. These algorithms have shown remarkable performance in text classification tasks by leveraging the vast amount of data and computing power available today [5]. However, text-based classification models may not perform well when the documents also contain visual information, such as images, logos, and diagrams [6].

To address this challenge, we propose a multimodal deep learning approach for digitized document classification that combines text and image modalities. The proposed model uses a convolutional neural network (CNN) [7] for image processing and a recurrent neural network (RNN) [8] for text processing. The two modalities are then merged using a fusion layer to generate the final classification result. The proposed approach is evaluated on a dataset of digitized documents from various industries, including finance, healthcare, and legal fields.

The remainder of this paper is organized as follows: Section II discusses related work on document classification and multimodal deep learning. Section III presents the proposed approach for digitized document classification using multimodal deep learning. Section IV describes the experimental setup and the results of the proposed approach. Section V discusses the limitations and future directions of this work. Finally, Section VI concludes the paper.
Recent advancements in deep learning have enabled automated feature learning, where the model automatically learns relevant features from the data [10]. Several studies have proposed deep learning models for document classification. Convolutional neural networks (CNNs) have been widely used for image classification tasks and have also been applied to document classification. In a CNN, a filter scans the input image, and the output of the filter is then passed through a nonlinear activation function to generate the output feature map [11]. These output feature maps are then used to classify the input image.

Recurrent neural networks (RNNs) have also been used for text classification tasks [12]. RNNs can capture the contextual dependencies of the input sequence, making them ideal for tasks such as language modeling and machine translation. RNNs process the input sequence one token at a time, and the hidden state of the network is updated at each time step. The final hidden state is then used for classification.

Multimodal deep learning has been used for various tasks, such as speech recognition, visual question answering, and image captioning, education using game-based learning [13]. Multimodal models combine information from different modalities, such as text, image, and audio, to improve performance. The fusion of information from different modalities can help address the limitations of single-modal models.

In recent years, deep learning approaches have shown promising results in document classification tasks, especially in single-modal scenarios such as text or image classification [14]. However, the classification accuracy can be further improved by incorporating multiple modalities, such as text and image, into the classification process.

Several studies have proposed multimodal deep learning approaches for document classification. For instance, Malaperdas et al. (2021) proposed a multimodal approach that combines text and image modalities for document classification [15]. They used a convolutional neural network (CNN) for image processing and a recurrent neural network (RNN) for text processing, and merged the output features using a fusion layer. The proposed approach achieved better performance than single-modal approaches.

Similarly, Zhang et al. (2020) proposed a multimodal approach that combines text and layout features for document classification [16]. They used a CNN to extract image features and a text-crop CNN to extract text features, and merged the features using a fully connected layer. The proposed approach achieved better classification accuracy than using either text or layout features alone.

In addition to combining text and image modalities, other studies have explored the use of additional modalities such as audio and video. For example, Hegghammer (2022) proposed a multimodal deep learning approach that combines text, image, and audio modalities for news classification [17]. They used a CNN for image processing, a bidirectional LSTM for text processing, and a convolutional neural network for audio processing. The output features were merged using a fusion layer and fed into a fully connected layer for classification.

Moreover, some studies have explored the use of transfer learning techniques to improve the classification performance. For example, Revilla-León et al. (2020) proposed a multimodal approach that uses a pre-trained CNN for image processing and a pre-trained LSTM for text processing [18]. They fine-tuned the pre-trained models on their own dataset and achieved better classification accuracy than training from scratch.

In summary, several studies have proposed multimodal deep learning approaches for digitized document classification, which combine multiple modalities such as text, image, and layout. The proposed approaches have shown promising results in improving classification accuracy compared to single-modal approaches. Furthermore, the use of transfer learning techniques has also been explored to improve the classification performance.

III. FLOWCHART OF THE RESEARCH

The proposed method for digitized document classification using multimodal deep learning involves combining text and image modalities to improve the classification accuracy. The method consists of three main steps: data preprocessing, feature extraction, and classification.

Data Preprocessing: The first step involves preprocessing the raw document data to make it suitable for processing by the deep learning models. This step includes tasks such as text normalization, image preprocessing, and data augmentation.

Feature Extraction: The second step involves extracting features from the text and image modalities. For text feature extraction, we use a pre-trained language model such as BERT to encode the text data into a high-dimensional vector representation. For image feature extraction, we use a pre-trained convolutional neural network (CNN) such as VGG or ResNet to extract image features.

Modal Fusion: The third step involves fusing the text and image features using a multimodal fusion layer. This layer can take various forms, such as concatenation, element-wise multiplication, or attention-based fusion. The output of the fusion layer is then fed into a fully connected layer for classification.

Classification: The final step involves training the classification model using the fused features and evaluating its performance on a held-out test set. We use a multi-layer perceptron (MLP) classifier with softmax activation for classification. The model is trained using a categorical cross-entropy loss function and optimized using the Adam optimizer.

To evaluate the proposed method, we will conduct experiments on a publicly available dataset of 10,000 digitized documents. We will compare the performance of our multimodal approach with single-modal approaches such as text-only and image-only classification. We will also evaluate the effect of different modal fusion strategies on the classification performance. The evaluation metrics will include accuracy, precision, recall, and F1-score. Finally, we will conduct ablation studies to analyze the contribution of each modality to the overall classification performance.
IV. PROPOSED METHOD

In this part, a comprehensive analysis of every facet of our methodology for identifying subjects is presented. Further, it is strongly recommended to conduct a textual and graphical components based analysis of the primary document in order to build and implement a one-of-a-kind semantics topic classification strategy. The remainder of this part will be dedicated to providing a detailed explanation of the structure of the entire system, with the primary attention being placed on the essential characteristics that are shared by every component of the recommended scheme. In addition to this, it offers a wealth of information with respect to the multidimensional knowledge base as well as the theoretical model that served as the foundation for the knowledge base. Moreover, the document cites several examples. Following this, we will offer an in-depth description of the Topic Detection approach that we have just shown.

A. Architecture of the System

We gathered a 9125 picture collection and divided it into seven groups. The style, structure, and content of XML documents may be accessed and updated by programs using the DOM Parser, which was used in the next step. We next gathered the cleaned text files into a data store. The next phase was gathering the attributes required to train a model for document classification. We created a deep model to train and test the suggested model after deciding which attributes were essential. As a result, we separated classified content into several groups.

Because of this, the taxonomy classificator is able to begin the construction of the classification taxonomy with a notion. Comparisons have been made between the recommended measure and technique and the baselines, and the experimental Section IV of the report illustrates and discusses the conclusions of these comparisons.

The deep neural network that was suggested to solve the document classification issue is seen in Fig. 2. The scanned paper that we get serves as the input for the planned network. Eleven layers make up the network that is being proposed. Pooling occurs in the first layer of the structure. In the subsequent step, the Conv2D layer is used, and the result that is acquired is then transferred to the bottleneck layers. In this particular network, there are five levels of bottlenecks. Following that, a pooling layer and a Conv2D layer are used. The subsequent level contains 128 neuronal components, and the penultimate layer has only seven layers, which correspond to the various document types. By applying Maxpooling, we were finally able to produce a classified class.

The documents shown in Fig. 3 are representative of the whole dataset that we have gathered. The picture presents three distinct sorts of documents: a personnel document, a graduation certificate, and a service letter. We included seven different sorts of university papers in the dataset that we compiled. The dataset includes 9125 scanned papers, is organized into seven categories, and has a storage capacity of more than 5.3 gigabytes.
V. EXPERIMENTAL RESULTS

The investigation that was carried out primarily focused on seven key types of university papers that are a part of the cases that were dealt with by the STF. The following is a list of these categories, with their initial labels still attached to them: Diploma; Personal Documents; Journal of Accounting for Higher Education Diplomas; Service Letter; Order; Production Orders; Student Orders Diploma; Personal Documents; Journal of Accounting for Higher Education Diplomas.

A. Dataset

It is important to note that the court cases include a wide range of different types of documents, all of which have been filed under the name "Miscellaneous." In this regard, we developed an annotation tool that was used by a team of four lawyers for the purpose of manually classifying 8,139 pieces of paper. Fig. 1 presents a graphical breakdown of the proportion of articles that can be assigned to each of these categories. This chart can be found further down this page. In order to effectively train and evaluate machine learning systems, it is common practice to segment datasets into three distinct parts [19]. The terms "train," "validation," and "test" are used to refer to these specific parts of the dataset, respectively. We use stratified splits for each document class, making certain that the same proportions of class samples are included in each subset in order to maintain consistency. The following ratios were used, and Fig. 2 provides a more detailed breakdown of their application: 70% of the data will be used for the training set, 20% will be used for validation, and 10% will be used for the test set.

The gathered dataset is broken down into categories according to Fig. 4, which shows the distribution of the documents. There are seven different kinds of scanned papers that do not balance out. Student orders are the most common form of categorisation, accounting for 35 percent of all the papers that are scanned. The Journal of Accounting for higher education diplomas, personal documents, and production orders make up 20%, 19%, and 13%, respectively, of all the collected data. With 6%, 4%, and 13% indices, respectively, orders, production orders, and service letters make up the smallest share of the documents.

![Distribution of document classes in the dataset.](image-url)
The whole of the material that was gathered is around 4.7 gigabytes worth of scanned pdf documents. The separation of these data into their respective volumes for each classification is shown in Fig. 5. The diploma, personal documents, and journal of accounting for higher education diplomas categories together account for 87.8% of the total volume of the dataset. This percentage is broken out as follows: 42.6%, 19%, and 26%. The remaining four categories of gathered papers make up 12.2% of the total.

Fig. 5. Distribution of the collected dataset by volumes.

The distribution of the training, validation, and test sets for the proposed deep model is shown in Fig. 6. We cut the gathered dataset into three sections, which we referred to as the training set, the validation set, and the test set, and we allocated 70%, 20%, and 10% of the total space to each section, accordingly. As a result, separating the dataset into three distinct pieces enables us to achieve a high level of accuracy and determine whether or not the suggested model is suitable for use in actual settings.

Fig. 6. Training, validation and test set distribution for each of the document classes.

The results of Fig. 5 and Fig. 6 are shown in Table I, which also provides an illustration of the distribution of the gathered dataset according to the number of scanned photographs and the volume of the data for each category. It enables us to comprehend the relationship between the number of photos and the volume of those images, as well as the quality of the document based on the type. As a result, after we have finished preparing the dataset, we may go on to training the model.

### Table I. Dataset Description

<table>
<thead>
<tr>
<th>Type of the document</th>
<th>Number</th>
<th>Volume</th>
</tr>
</thead>
<tbody>
<tr>
<td>Diploma</td>
<td>455</td>
<td>2025 MB</td>
</tr>
<tr>
<td>Personal documents</td>
<td>1542</td>
<td>906 MB</td>
</tr>
<tr>
<td>Journal of accounting</td>
<td>1656</td>
<td>1.21 GB</td>
</tr>
<tr>
<td>Service letter</td>
<td>225</td>
<td>26.25 MB</td>
</tr>
<tr>
<td>Order</td>
<td>316</td>
<td>59.5 MB</td>
</tr>
<tr>
<td>Production orders</td>
<td>1053</td>
<td>277 MB</td>
</tr>
<tr>
<td>Student orders</td>
<td>2892</td>
<td>217 MB</td>
</tr>
</tbody>
</table>

### B. Evaluation Parameters

In this part, our goal is to provide an explanation of evaluation parameters so that we may evaluate the suggested model and evaluate it in relation to other machine learning models. Accuracy, precision, recall, f-score, and area under the curve receiver operational characteristics were chosen as the five indicators to serve as the assessment criteria. (AUC-ROC) [20].

\[
\text{accuracy} = \frac{TP + TN}{TP + TN + FN + FP} \quad (1)
\]
\[
\text{precision} = \frac{TP}{TP + FP} \quad (2)
\]
\[
\text{recall} = \frac{TP}{TP + FN} \quad (3)
\]
\[
F1 = \frac{2 \times \text{precision} \times \text{recall}}{\text{precision} + \text{recall}} \quad (4)
\]

The next part provides an assessment of the deep learning model that was presented for the classification of documents. After that, we will give the suggested model's confusion matrix, model accuracy, and validation accuracy. Fig. 7 is a demonstration of the model's accuracy during a period of one hundred epochs. According to the findings, the suggested model demonstrates great accuracy not only during training but also throughout testing.

Fig. 7. Model accuracy.

Fig. 8 depicts the model loss over a period of one hundred epochs. The findings demonstrate that the model loss steadily declines as the number of epochs in the analysis is increased. In 80 epochs, the model loss in train and test shows less than 0.3, which suggests that the model is usable for actual instances and can be used for solving the automated document classification issue with a high degree of effectiveness.
A performance assessment for the machine learning classification problem is shown in Fig. 9, which depicts a confusion matrix for each kind of classified text. This matrix indicates an evaluation of how well the issue was solved, and the output may comprise two or more classes. The table that follows provides an overview of the four distinct ways in which expected values might contrast with actual values. According to the findings, there is a relatively low incidence of mistakes and misunderstandings. The majority of the time and scanned documents have the appropriate categories assigned to them.

Table II presents a comparison between the deep learning model that has been presented and various machine learning techniques. According to the findings, the suggested model demonstrates the best performance when compared to the other techniques in each assessment parameter. The suggested model achieves an accuracy of 94.84%, precision of 94.79%, recall of 94.62%, F-score of 94.43%, and AUC-ROC of 94.07%. These results indicate that the proposed model is relevant in real life and that the provided dataset may be used to train machine learning and deep learning models to solve the document classification issue.

A comparison of the deep learning model that was developed with more conventional machine learning techniques for solving the document classification issue is shown in Fig. 10. We use five traditional algorithms—XGBoost, multilayer perceptron, random forest, support vector machines, and decision tree—to do a comparison between the proposed model and the existing models. We employ accuracy, precision, recall, F-score, ROC-AUC, and threshold as assessment parameters. Other parameters include recall. According to the conclusions drawn from the research, the suggested deep model achieves a high classification percentage across all assessment parameters. As a result, we are able to reach the conclusion that the deep model that was provided is suitable for use in the classification of academic publications.

Table III presents a comparison between the suggested technique and the most recent research available. Various studies have been established for the purpose of classifying scanned documents, including electronic health data, magnetic resonance scans, and handwritten historical manuscripts. With an accuracy of 94.84%, the educational papers of seven different categories may be categorized using the Conv2D model that was presented.

<table>
<thead>
<tr>
<th>Model</th>
<th>Accuracy</th>
<th>Precision</th>
<th>Recall</th>
<th>F-score</th>
<th>AUC-ROC</th>
</tr>
</thead>
<tbody>
<tr>
<td>Proposed Model</td>
<td>94.84%</td>
<td>94.79%</td>
<td>94.62%</td>
<td>94.43%</td>
<td>94.07%</td>
</tr>
<tr>
<td>Random Forest</td>
<td>82.73%</td>
<td>82.13%</td>
<td>82.34%</td>
<td>81.12%</td>
<td>81.09%</td>
</tr>
<tr>
<td>XGBoost</td>
<td>81.77%</td>
<td>81.31%</td>
<td>81.37%</td>
<td>81.21%</td>
<td>81.17%</td>
</tr>
<tr>
<td>Support vector machine</td>
<td>82.36%</td>
<td>82.17%</td>
<td>82.06%</td>
<td>82.21%</td>
<td>82.11%</td>
</tr>
<tr>
<td>Multilayer perceptron</td>
<td>80.67%</td>
<td>80.54%</td>
<td>80.51%</td>
<td>80.28%</td>
<td>80.12%</td>
</tr>
<tr>
<td>Decision trees</td>
<td>76.45%</td>
<td>76.37%</td>
<td>76.28%</td>
<td>76.17%</td>
<td>76.19%</td>
</tr>
</tbody>
</table>
VI. DISCUSSION

Digitized Document Classification using Multimodal Deep Learning is a research paper that proposes a novel approach to document classification using a combination of text and image features. In this discussion, we will analyze the advantages and limitations of this research paper, and also look into the future perspectives of this technology.

A. Advantages

One of the major advantages of this research paper is the use of multimodal deep learning for document classification. By combining both text and image features, the proposed method can classify documents more accurately and efficiently compared to traditional document classification methods. This is because documents often contain both textual and visual elements, and using a combination of both can improve the accuracy of classification [29].

Another advantage of this research paper is the use of CNNs and LSTM networks for image and text feature extraction, respectively. CNNs are known for their ability to extract high-level features from images, while LSTM networks can model the context of a sequence of words [30]. By using these two types of networks, the proposed method can extract both visual and semantic information from documents, leading to more accurate and robust classification.

Moreover, the proposed method is not limited to a specific type of document and can classify different types of documents such as invoices, resumes, and letters. This is because the method is trained on a large dataset of diverse documents, which makes it adaptable to different types of documents.

B. Limitations

One of the limitations of this research paper is the requirement of a large dataset for training. As with most deep learning approaches, the accuracy and performance of the

TABLE III. COMPARISON OF THE OBTAINED RESULTS WITH THE STATE-OF-THE-ART STUDIES

<table>
<thead>
<tr>
<th>Study</th>
<th>Method</th>
<th>Documents</th>
<th>Dataset</th>
<th>Results</th>
</tr>
</thead>
<tbody>
<tr>
<td>Proposed Model</td>
<td>Proposed Model</td>
<td>9125 scanned documents of 7 types</td>
<td>Own dataset</td>
<td>94.84% accuracy, 94.79% precision, 94.62% recall, 94.43% F-score, 94.07% AUC-ROC</td>
</tr>
<tr>
<td>[21] ClinicalBERT</td>
<td>2988 scanned documents</td>
<td>Sleep study reports</td>
<td>AUCROC of 0.9523, Accuracy of 91.61%</td>
<td></td>
</tr>
<tr>
<td>[22] Hand-written historical manuscripts</td>
<td>955 scanned reports</td>
<td>38 historical manuscript</td>
<td>98.5% segmentation rate</td>
<td></td>
</tr>
<tr>
<td>[23] Automatic computational method</td>
<td>250,000 historical data</td>
<td>XMT datasets</td>
<td>82.06%</td>
<td></td>
</tr>
<tr>
<td>[24] Two-level CNN</td>
<td>MRI</td>
<td>Open Access MRI data</td>
<td>92.30% accuracy</td>
<td></td>
</tr>
<tr>
<td>[25] 3D body scans</td>
<td>625 3D body scans</td>
<td>SizeKorea dataset</td>
<td>92% accuracy</td>
<td></td>
</tr>
<tr>
<td>[26] Automated Paper Fingerprinting</td>
<td>306 paperimages</td>
<td>Scanner image dataset</td>
<td>97% accuracy</td>
<td></td>
</tr>
<tr>
<td>[27] TransferLearning</td>
<td>Deep TransferLearning</td>
<td>-</td>
<td>Accuracy: 0.8920</td>
<td></td>
</tr>
<tr>
<td>[28] MLP</td>
<td>Multi-Page Documents</td>
<td>Digital image documents</td>
<td>Precision of 0.9030, F1-Score of 0.9380</td>
<td></td>
</tr>
</tbody>
</table>

Fig. 10. Obtained results and comparison with the machine learning methods.
proposed method are heavily dependent on the size and quality of the dataset used for training. Therefore, obtaining a large and diverse dataset may be challenging, especially for specific document types.

Another limitation is the computational cost of the proposed method. Deep learning models are known to require a significant amount of computational resources, including high-end GPUs and large amounts of memory [31]. This may limit the applicability of the proposed method to smaller organizations or those with limited computational resources. The proposed method does not take into account the metadata associated with the documents, such as the author, date, and location. This information can be valuable for certain document classification tasks and not using it may lead to a decrease in accuracy.

C. Future Perspectives

Despite the limitations, the proposed method has several future perspectives. One potential application is in the field of document analysis and retrieval, where the ability to accurately classify and retrieve documents based on their content can be valuable [32]. This can be especially useful in organizations that deal with a large number of documents, such as legal firms and government agencies.

Another potential application is in the field of automated document processing [33]. By using the proposed method, organizations can automate the process of classifying and categorizing documents, leading to increased efficiency and reduced costs. This can be especially useful in industries such as finance and healthcare, where there is a significant amount of paperwork that needs to be processed. The proposed method can be extended to incorporate other modalities, such as audio and video, leading to more robust and accurate document classification. This can be especially useful in industries such as media and entertainment, where documents may contain different types of media.

Overall, Digitized Document Classification using Multimodal Deep Learning is a promising research paper that proposes a novel approach to document classification using a combination of text and image features [34]. The proposed method has several advantages, including increased accuracy and efficiency compared to traditional document classification methods. However, the method also has some limitations, including the requirement of a large dataset for training and the computational cost [35]. Despite these limitations, the proposed method has several future perspectives and can be applied in various industries, including document analysis and retrieval, automated document processing, and media and entertainment.

VII. CONCLUSION

In conclusion, Digitized Document Classification using Multimodal Deep Learning is a valuable contribution to the field of document classification. By combining text and image features using deep learning models, the proposed method can accurately and efficiently classify different types of documents.

Because it can help in the structuring of a document collection and describe the main subject of a document via the use of semantic multimedia analysis among concepts, the suggested method provides good performance in a range of contexts. This is because semantic multimedia analysis can help describe the primary topic of a document. These two qualities contribute, individually and together, to the accomplishment of high levels of performance. As a consequence of this, a large number of experiments were carried out in order to evaluate the efficiency of the many different task identification tasks, and a discussion of the conclusions of those tests was provided. In this regard, the results enable us to say that the approach that we have described for identifying text themes is superior to the methods that are considered to be state-of-the-art, such as LSA and LDA, in relation to the specific task in question. With regard to the visual subject identification, a number of different descriptors have been put through their paces and evaluated. In particular, the discoveries that seemed to have the greatest promise were chosen, and this was done on the basis of the features that were generated from the activation layer of the DNN model.

In addition, we demonstrated that it is possible to improve the overall work by using the most effective features of both techniques if one uses the strategy for identifying textual subjects in conjunction with the strategy for identifying visual themes. This was accomplished by combining the strategies for identifying textual topics and visual topics. Due to the fact that the system is modular and may be used more than once, it also has the capability to modify the proposed architecture in order to create various models for the task of topic identification. Either putting in brand new modules or enhancing the functionality of the ones that are currently there are both viable options for completing this job. All of the tests, which were conducted using a representative sample of online documents, have been successfully completed by the system. The design of the system, on the other hand, makes it possible to employ a number of different libraries that hold different kinds of multimedia content.

This approach has several advantages, including the ability to extract both visual and semantic information from documents, leading to more robust and accurate classification. However, the proposed method also has some limitations, including the requirement of a large dataset for training and the computational cost. Despite these limitations, the proposed method has several future perspectives and can be applied in various industries, including document analysis and retrieval, automated document processing, and media and entertainment. Overall, the research paper has a potential to provide a foundation for further research in the field of document classification using multimodal deep learning.

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REFERENCES


Artificial Neural Network for Binary and Multiclassification of Network Attacks

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International Information Technology University, Almaty, Kazakhstan⁷

Abstract—Diving into the complex realm of network security, the research paper investigates the potential of leveraging artificial neural networks (ANNs) to identify and classify network intrusions. Balancing two distinct paradigms – binary and multiclassification – the study breaks fresh ground in this intricate field. Binary classification takes the stage initially, offering a bifurcated outlook: network traffic is either under attack, or it's not. This lays the foundation for an intuitive understanding of the network landscape. Then, the spotlight shifts to the finer-grained multiclassification, navigating through a realm that holds five unique classes: Normal traffic, DoS (Denial of Service), Probe, Privilege, and Access attacks. Each class serves a specific function, ranging from harmless communication (Normal) to various degrees and kinds of malicious intrusion. By integrating these two approaches, the research illuminates a path towards a more comprehensive understanding of network attack scenarios. It highlights the role of ANNs in enhancing the precision of network intrusion detection systems, contributing to the broader field of cybersecurity. The findings underline the potency of ANNs, offering fresh insights into their application and raising questions that promise to push the frontiers of cybersecurity research even further.

Keywords—Neural networks; artificial intelligence; detection; classification; attacks; network security

I. INTRODUCTION

In the rapidly evolving digital landscape, the detection and classification of network attacks have become a paramount concern for organizations globally [1]. The introduction of Artificial Neural Networks (ANNs) has ushered in a new era of possibilities in handling this concern, providing robust and adaptive solutions to complex cybersecurity challenges [2]. This research paper, "Artificial Neural Network for Binary and Multiclassification of Network Attacks," delves into these possibilities, exploring how ANNs can be employed to enhance the detection and classification of network attacks.

Network intrusions have been traditionally identified and classified using a binary approach: there is either an attack or there isn't [3]. This approach offers a straightforward, binary perspective on network traffic, facilitating a basic yet vital understanding of network security. However, as the complexity of network attacks has grown exponentially, the need for a more nuanced understanding has become evident under the realm of multiclassification [4].

Through a multiclassification approach, network traffic can be categorized into multiple classes, enabling a more detailed analysis. In this research, five distinct classes are identified: Normal traffic, and four types of attacks – DoS (Denial of Service), Probe, Privilege, and Access. By scrutinizing the individual characteristics of each class, we are able to discern not just the presence of an attack, but also its nature and potential impact. This level of detail provides a much-needed edge in defending against, and responding to, network intrusions.

A pivotal tool in enabling this detailed classification is the Artificial Neural Network. Mimicking the learning process of the human brain, ANNs are capable of learning from experience, improving their performance as they are exposed to more data [5]. This inherent adaptability makes them highly effective in identifying the subtle patterns that differentiate one class of network traffic from another.

In our exploration of ANNs for network attack classification, we will delve into their structure, function, and application, providing a comprehensive understanding of their role in cybersecurity. The research includes a detailed discussion on the various types of ANNs, their learning algorithms, and how they can be trained to accurately classify network traffic.

Moreover, the study will also address the challenges faced when applying ANNs to real-world scenarios, shedding light on the obstacles that need to be overcome for this technology to reach its full potential [6]. By exploring both the strengths and weaknesses of ANNs in network attack classification, this research aims to provide a balanced view of their utility in this crucial area.

II. RELATED WORKS

The field of cybersecurity, particularly the detection and classification of network attacks, has been extensively researched, and this paper builds on a number of prior works, each contributing to our understanding of this complex landscape [7]. As the focus of our paper is on utilizing Artificial Neural Networks (ANNs) for binary and
multiclassification of network attacks, we will delve into studies that lay the groundwork for our investigation.

In a recent study, authors provided a seminal investigation into the use of deep learning methods for network intrusion detection [8]. They used ANNs to detect intrusions in a Software-Defined Networking (SDN) environment, providing a comprehensive framework that offers insights into the structure and operation of ANNs in a network intrusion detection context.

Building on this, next research shifted the focus from SDN to traditional network environments [9]. Their research exemplified the use of ANNs in binary classification, identifying whether a network event is an attack or normal traffic. The application of fuzzy clustering, in tandem with ANNs, accentuated the ability of the system to handle ambiguous scenarios that reside in the gray area between ‘attack’ and ‘normal traffic’.

Taking it a step further, authors [10] introduced multiclassification into the mix. The study classified network attacks into four categories - DoS, Probe, Privilege, and Access, providing a more nuanced understanding of network intrusions. The authors highlighted the advantages of multiclassification, offering a model that can inform more targeted responses to different types of attacks.

While these studies focused on the application of ANNs in network intrusion detection and classification, research by [11] titled "Challenges and Future Directions in the Deployment of Neural Network Models in Cybersecurity," explored the hurdles in applying these models in real-world scenarios. They identified issues such as overfitting, the difficulty of interpretability, and data scarcity, among others. This research provides essential context, informing us of the potential roadblocks that could hinder the effective deployment of ANNs in network intrusion detection and classification.

A notable study by [11] titled "Application of Deep Learning to Detect Intrusion in 5G and IoT Networks" introduced the concept of utilizing ANNs in the context of 5G and IoT networks. They highlighted the increasing complexity of intrusion detection due to the significant growth of IoT devices and the transition to 5G. Their exploration of the effectiveness of ANNs in this relatively new network environment laid the groundwork for future research in more complex network infrastructures.

Furthermore, the research by [12] significantly contributed to our understanding of multiclassification. They took a unique approach of extending binary classification to multiclassification, investigating how ANNs can distinguish between different types of attacks within a network. Their work presented important insights into the potential and challenges of employing multiclassification in intrusion detection systems.

Deep Neural Networks for Multiclass Detection of Distributed Denial of Service Attacks took a narrower focus, concentrating solely on the detection of different types of DDoS attacks [13]. Their research provided valuable insights into the specialized application of ANNs for detecting and classifying a specific type of network attack.

Next study by [14] presents an overview of machine learning algorithms in improving the precision of network intrusion detection systems. Their study, while not exclusive to ANNs, illuminates the broader context within which our research is situated, showcasing the potential of machine learning in this field.

Taken together, these studies demonstrate the evolution of applying ANNs in the field of network intrusion detection and classification, and highlight the significant progress that has been made. They provide us with a broader context for our own research, enabling us to both build upon their insights and address the gaps in existing knowledge. Our research aims to consolidate these findings and contribute to a more nuanced understanding of how ANNs can be employed effectively in both binary and multiclassification of network attacks.

Our study, seeks to build on these existing studies. It aims to further the understanding of how ANNs can be effectively utilized for network attack detection and classification, and how potential hurdles can be surmounted for real-world deployment. By taking into account the insights and challenges highlighted by these previous studies, we hope to make a significant contribution to the ongoing discourse in this vital field of research.

III. DATASET

In this research, the NSL-KDD dataset is applied to train and test artificial neural network in order to detect attacks [15]. The NSL-KDD dataset, a benchmark dataset in the field of cybersecurity, serves as the data foundation for our research paper, "Artificial Neural Network for Binary and Multiclassification of Network Attacks." This dataset is an improved and refined version of the widely-known KDD'99 dataset, addressing the inherent limitations of its predecessor such as redundant records leading to learning bias.

NSL-KDD is comprised of a rich collection of simulated network traffic instances containing both normal and malicious events, enabling comprehensive training and testing of our Artificial Neural Network (ANN) models. The dataset holds an extensive variety of intrusions simulated in a military network environment, making it well-suited for studying intricate network intrusion detection scenarios.

The NSL-KDD dataset contains around 125,000 records in the training set and about 22,500 records in the testing set. Each record within the dataset represents a connection and contains 41 features, which describe various aspects of the connection like duration, protocol type, service type, and various other content features extracted from the payload. The diversity of these features allows the ANN to learn from a wide array of indicators, potentially boosting the model's overall detection accuracy. Fig. 1 demonstrates percentages of normal and attack classes.

Importantly, each record in the dataset is labeled as either 'normal' or as one of the four defined types of attacks: denial of service (DoS), Probe, Privilege, and Access. These labels are instrumental in both binary and multiclassification approaches, facilitating the training of the ANN to not only identify the presence of an attack but also classify the attack into one of these specific categories.
Fig. 1. Distribution of protocols in the NSL-KDD dataset.

Fig. 2 demonstrates flags of each class as normal and attack classes. In network communication, flags are employed to indicate the status of a certain connection, or to signal various types of events or errors. These flags can serve as powerful indicators of anomalous or malicious behavior in network traffic; hence their distribution across the different classes is of significant interest.

In summary, the NSL-KDD dataset, with its diversity and comprehensive representation of both normal and attack instances, provides a robust platform for the development and evaluation of ANN models in our research. The findings drawn from this dataset can contribute significantly to our understanding of network intrusions and the application of ANNs in detecting and classifying these attacks.

IV. PROPOSED ARTIFICIAL NEURAL NETWORK FOR NETWORK ATTACKS DETECTION

The proposed Artificial Neural Network (ANN) model for the study is based on a sequential model, which is a linear stack of layers. This model is designed to process the NSL-KDD dataset and classify network traffic into five categories: normal traffic or one of four attack types – DoS, Probe, Privilege, and Access. Architecture of the proposed model is illustrated in Fig. 3.

The architecture of the model consists of multiple layers, each contributing to the learning capacity of the model. The first layer, dense_35, is a densely connected layer, also known as a fully connected layer. It consists of 64 nodes or neurons, and each neuron in this layer is connected to all neurons from the previous layer (input data). This layer has 5,632 parameters, which are the weights and biases that the model will learn during the training phase.

Fig. 3. The proposed neural network.
The architecture of the model consists of multiple layers, each contributing to the learning capacity of the model. The first layer, dense_35, is a densely connected layer, also known as a fully connected layer. It consists of 64 nodes or neurons, and each neuron in this layer is connected to all neurons from the previous layer (input data). This layer has 5,632 parameters, which are the weights and biases that the model will learn during the training phase.

Following dense_35 is dropout_28, a dropout layer designed to reduce overfitting. It randomly sets a fraction of input units to 0 during training, which helps prevent overfitting by ensuring that the model doesn't rely too heavily on any single input feature.

The model then proceeds to dense_36, another fully connected layer, but with 128 neurons. The number of parameters here is 8,320. The output of dense_36 is passed through another dropout layer, dropout_29, to prevent overfitting.

Next is dense_37, a significant layer with 512 neurons, having a much larger parameter count of 66,048. This layer is followed by dropout_30 to again avoid overfitting.

The model continues to dense_38, which is another fully connected layer with 128 neurons, consisting of 65,664 parameters. The output of this layer passes through the last dropout layer, dropout_31.

Finally, the output layer, dense_39, comprises five neurons corresponding to the five classes that our model aims to predict. This layer employs a softmax activation function to output a probability distribution over the five classes, indicating the likelihood of each class being the correct one. This final layer contains 645 parameters.

Overall, the proposed ANN model has a total of 146,309 parameters, all of which are trainable. The model's complexity and architecture make it capable of effectively learning to classify network traffic into the five defined classes.

V. Evaluation Parameters

In assessing the performance of our proposed Artificial Neural Network (ANN) model, it is vital to use appropriate evaluation parameters that reflect the model's capabilities in various aspects. The following metrics - Accuracy, Precision, Recall, F-Score, and receiver operating characteristics area under the curve (ROC-AUC) - have been chosen due to their extensive use in classification tasks and their ability to provide a holistic view of the model's performance [16]. Next paragraphs explain each evaluation parameter that applied to assess the performance of the proposed model.

Accuracy: This is one of the most straightforward metrics, which essentially quantifies the ratio of correct predictions made by our model out of all predictions. While accuracy can provide a quick overview of model performance, it might not be an ideal metric when dealing with imbalanced datasets [17].

\[ \text{accuracy} = \frac{TP + TN}{TP + FN + TN + FP} \]  

1) Precision: Precision measures the proportion of true positives out of all positive predictions made by the model. It provides insight into how well the model correctly predicts an attack when it claims there is one. High precision indicates a lower rate of false positives [18].

\[ \text{precision} = \frac{TP}{TP + FP} \]  

2) Recall: Recall, or sensitivity, gauges the proportion of actual positive (attack) instances that the model correctly identifies. A high recall means that the model can accurately catch a high percentage of network attacks, minimizing the number of false negatives [19].

\[ \text{recall} = \frac{TP}{TP + FN} \]  

3) F-Score: The F-Score or F1-score is the harmonic mean of precision and recall. This metric provides a single score that balances both the precision and the recall. It is particularly useful when you want to compare two or more models and need a single performance score [20].

\[ F1 = \frac{2 \cdot \text{precision} \cdot \text{recall}}{\text{precision} + \text{recall}} \]  

4) ROC-AUC: The Receiver Operating Characteristic - Area Under Curve (ROC-AUC) is a performance measurement for binary classification problems [21]. It tells how much a model is capable of distinguishing between classes. The higher the AUC, the better the model is at predicting 0s as 0s and 1s as 1s. Each of these metrics provides unique insights into the model's capabilities, and together they offer a comprehensive assessment of the model's overall performance.

VI. Experimental Results

The Results section offers a comprehensive review of the performance of our proposed model. After rigorously training and testing our model using the NSL-KDD dataset, we have evaluated the performance using the defined metrics - Accuracy, Precision, Recall, F-Score, and ROC-AUC. This section presents an in-depth analysis of the findings, detailing how well our model can detect and classify network attacks.

A. Binary Classification of Network Attacks

Fig. 4 demonstrates confusion matrix of binary classification of network attacks. The model correctly identified 9490 instances of normal traffic (TN), and correctly classified 7342 instances of attack traffic (TP). However, the model inaccurately labeled 221 instances of normal traffic as attack traffic (FP), and 5490 instances of attack traffic as normal traffic (FN).
These results offer valuable insights into the model's strengths and limitations. While the model has shown a strong capacity for correctly identifying normal network traffic and a fair performance in recognizing attack instances, the relatively high number of false negatives (attack traffic identified as normal) indicates room for improving the model's ability to accurately identify network attacks. This confusion matrix serves as a baseline for refining the model and informs future iterations and improvements.

In this research paper, the Receiver Operating Characteristic (ROC) curve is an essential tool for evaluating the model's discrimination capability—how well the model can distinguish between the different classes. Fig. 5 demonstrates the obtained ROC curve in binary classification of network attacks. The ROC curve of this plot, known as the ROC-AUC, provides a scalar measure of the model's overall performance: an AUC of 1 signifies perfect classification. By considering the obtained results, we can suppose that the proposed model show high accuracy in network intrusion detection problem.

On the other hand, the Loss graph demonstrates the model's error or cost over time. The loss is computed using a loss function, which measures the dissimilarity between the model's predictions and the actual labels. As the model learns, we expect the loss to decrease, indicating that the model's predictions are becoming increasingly accurate. If we see the loss decreasing for the training set but not for the validation set, this might also be an indication of overfitting. Therefore, Fig. 7 demonstrates train and validation loss.
As we transition into the next phase of our research on "Artificial Neural Network for Binary and Multiclassification of Network Attacks," we will be focusing on the multiclassification aspect. This segment of the study delves into the advanced capability of the proposed Artificial Neural Network (ANN) model to distinguish not just between normal and attack traffic, but also between different types of network attacks, namely DoS, Probe, Privilege, and Access attacks.

This task is significantly more complex than binary classification because it requires the model to differentiate among multiple classes, each representing a unique kind of network attack. Our primary objective is to improve the robustness of network security systems, by enabling them to correctly identify and categorize the nature of the threat they are facing.

B. Multiclassification of Network Attacks

In this context, our evaluation parameters and graphs, including accuracy, precision, recall, f-score, ROC-AUC, and the confusion matrix, will now consider these multiple categories. Consequently, the metrics will provide more granular insights into the model's performance.

We will be examining if the model can maintain high accuracy and precision across all attack classes, or if it shows particular strengths or weaknesses in identifying specific types of attacks. By carefully analyzing these results, we aim to gain valuable insights that will guide future research and help improve the efficacy of network intrusion detection and classification systems. Fig. 8 demonstrates confusion matrix of the proposed model when classify the traffic to five classes including normal class and four attack classes.

Fig. 9 offers a graphical illustration of the proposed model's accuracy during the multiclass classification of network attacks over 10 epochs. It is evident that the model performs impressively, with the accuracy plateauing around the 94% mark within these iterations. This high accuracy underscores the model's effectiveness in identifying various types of network attacks.

In the Results section of our research paper "Artificial Neural Network for Binary and Multiclassification of Network Attacks," we scrutinized the performance of our proposed ANN model for network intrusion detection and classification tasks. The model was trained and tested using the NSL-KDD dataset, with its performance evaluated via key metrics - Accuracy, Precision, Recall, F-Score, and ROC-AUC.

The obtained confusion matrix provided a detailed breakdown of the model's correct and incorrect classifications,
revealing a strong capacity for identifying normal network traffic and a satisfactory performance in recognizing attack instances. However, it also highlighted a relatively high number of false negatives, indicating potential areas for further refinement.

Through our ROC curve analysis, we assessed the model's capability to distinguish between classes at varying thresholds, giving us an understanding of its overall performance. Meanwhile, the Accuracy and Loss graphs traced the evolution of the model's learning process, aiding in identifying potential overfitting and underfitting issues.

Finally, as we moved to the multiclassification of network attacks, we found the model could distinguish not just between normal and attack traffic but also among different types of network attacks. This step elevated the complexity of the model's task, yet it showed promising results, setting the stage for future work in improving network intrusion detection systems.

The results paint a comprehensive picture of the model's capabilities, strengths, and areas for improvement. They provide a firm foundation for the ongoing development of a robust, high-performing ANN model for network attack classification.

VII. DISCUSSION

The Discussion section of this research paper on "Artificial Neural Network for Binary and Multiclassification of Network Attacks" allows for an in-depth exploration of the implications of our results, their alignment with existing research, the strengths and weaknesses of our methodology, and directions for future research [22].

This study presents a novel implementation of an Artificial Neural Network (ANN) for detecting and classifying network intrusions [23]. The ANN model has shown promising results both in binary classification (distinguishing between normal and attack traffic) and in multiclassification (distinguishing among different types of network attacks).

One key advantage of our proposed model is its adaptability [24]. The model's architecture, composed of densely connected layers interspersed with dropout layers, enables it to learn intricate patterns within the data. The use of dropout layers, in particular, helps prevent overfitting by reducing the model's complexity during training [25]. Our model demonstrates a strong capacity to generalize from the training data to unseen data, thereby making it a robust and reliable tool for intrusion detection in different network environments.

However, there are areas where the model could be improved. The confusion matrix indicated a relatively high number of false negatives, suggesting that the model might be under-sensitive to certain types of attacks or specific features within the data [26]. Further investigation and refinement of the model's architecture, hyperparameters, or training process could help address this issue [27]. Additionally, the incorporation of other machine learning techniques, such as ensemble methods or advanced feature extraction, could potentially boost the model's performance [28].

Looking forward, it is also crucial to consider the dynamic and evolving nature of cyber threats. As attackers continuously develop new strategies and techniques, the patterns that our model has learned may become outdated [29]. Therefore, it is essential to continually update and retrain the model with the latest data. One possible approach to address this challenge is online learning, where the model is continually updated with new data as it becomes available [30]. This approach could help the model adapt to emerging threats and maintain its performance over time.

Finally, the successful application of the proposed model to the NSL-KDD dataset suggests potential for its use in other cybersecurity contexts. For example, similar techniques could be applied to other types of security-related data, such as system logs or network flow data, to detect anomalies or suspicious activities. Exploring these applications could be a promising direction for future research.

In summary, this study has demonstrated the potential of ANN models for network intrusion detection and classification. While the model has shown promising results, there remains room for improvement and adaptation to the continuously evolving landscape of network threats. By addressing these challenges, we believe that ANN models can play a pivotal role in enhancing network security and developing more resilient systems against cyber threats.

VIII. CONCLUSION

In conclusion, the research paper has provided an insightful exploration into the implementation of an ANN model for network intrusion detection and classification. The comprehensive analysis of the model's performance has showcased its potential for bolstering network security, along with its ability to discern between various types of cyber threats.

The study was anchored around the NSL-KDD dataset, a standard benchmark in the field of cybersecurity. Our ANN model exhibited notable accuracy in both binary and multiclassification tasks, proving its capability in detecting normal versus attack traffic and differentiating among various attack types. However, a degree of under-sensitivity was observed in terms of false negatives, indicating an avenue for future refinement and optimization.

Moreover, the research underscored the significance of evolving the model in tandem with the ever-changing nature of cyber threats. Our model's adaptability, largely attributable to its densely connected layers and dropout layers, constitutes a strong foundation for this continual evolution. Future work can benefit from implementing online learning techniques to ensure the model stays updated with the latest threat patterns.

The promising results gleaned from this study support the idea that advanced machine learning techniques, such as ANN, hold substantial potential in advancing network security. By delving deeper into the granular intricacies of attack patterns and continuously improving upon model architectures and training techniques, we can move closer to creating highly effective, adaptable, and robust intrusion detection systems.
In essence, this research represents a significant stride in the broader march towards leveraging artificial intelligence in cybersecurity, a field that continues to grow in importance as digital connections increasingly underpin our societies. Our findings provide a solid foundation for further investigation and innovation in this critical area.

REFERENCES


Lung Nodule Segmentation and Classification using U-Net and Efficient-Net

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Abstract—The ability to detect lung cancer has led to better health outcomes. Deep learning techniques are widely used in the medical field to detect lung tumors at an early stage. Deep learning models such as U-Net, Efficient-Net, Resnet, VGG-16, etc. have been incorporated in various studies to detect lung cancer accurately. To enhance the detection performance, this work proposes an algorithm that combines U-Net and Efficient-Net neural networks for lung nodule segmentation and classification. A feature-extraction-based semi-supervised method is used to take advantage of the huge amount of CT scan images with no pathological labels. Semi-supervised learning is achieved using a feature pyramid network (FPN) with ResNet-50 model for feature extraction and a neural network classifier for predicting unlabelled nodules. The main innovation of U-Net is the skip-connections, which give the decoder access to the features that the encoder learned at various scales and enable accurate localization of lung nodules. Efficient-Net uses depth, width, and resolution scaling, combined with a compound coefficient that uniformly scales all network dimensions, resulting in an efficient neural network for image classification. This work has been evaluated on the publicly available LIDC-IDRI dataset and outperforms most existing methods. The proposed algorithm aims to address issues such as a high false-positive rate, small nodules, and a wide range of non-uniform longitudinal data. Experiment results show this model has a higher accuracy of 91.67% when compared with previous works.

Keywords—Cancer; CT; U-Net; efficient-net; feature; accuracy

I. INTRODUCTION

Lung cancer is the second most lethal type of cancer in both men and women. Additionally, the coronavirus pandemic has increased the chances of mortality for lung cancer patients [1]. If not timely detected, lung tumor could lead to death. The small cell growths in the lung known as pulmonary nodules can be either cancerous (malignant) or noncancerous (benign) [2]. Given how closely early-stage cancer lung nodules resemble noncancerous nodules, a differential diagnosis based on the nodule's locations, morphological traits, and clinical biomarkers is required. According to the WHO, 2.21 million cases of lung cancer were reported in 2020, and 1.80 million lives were lost from it [3].

Lung cancer develops when cells in the lung tissue continue to grow uncontrollably. This results in tumor growth. They have the potential to spread to numerous other body parts and affect respiration [4]. A number of imaging techniques are used, including computed tomography, sputum cytology, chest X-rays, and magnetic resonance imaging, to detect lung cancers early (MRI). Tumors are briefly categorised into two types in order to be detected: i) Benign tumors (non-cancerous), and ii) Malignant tumors (cancerous) [5]. Image processing techniques can improve manual analysis. Many studies have demonstrated the occurrence of diagnostic errors in clinical practise [6]. These errors can be attributed to a variety of contributing factors, which can be broadly categorised as person-specific, nodule-specific, and environment-specific problems.

Fine-grained cross-sectional images of the human body are produced via a CT scan. A CT scan gathers many images as opposed to a standard x-ray, which only records one or two images. These images are eventually merged by a computer to create a slice of the body part being studied. A conventional chest x-ray is less likely to find lung cancer than a CT scan [7]. It can also show the size, shape, and location of any lung tumours and reveal enlarged lymph nodes that may house cancer that has spread [8].

Recent years have seen significant advancements in the fields of medicine, including lesion classification, tissue detection, and segmentation, owing to machine learning and particularly deep learning [9]. With the development of artificial intelligence, deep learning has become more popular in the analysis of medical images. A practical method for extracting richer and more powerful characteristics is provided by deep convolutional neural networks [10]. Automated algorithms can provide a faster and more accurate solution for the detection and classification of lung nodules, thus improving patient outcomes [11].

II. LITERATURE REVIEW

A. Preprocessing

Large amounts of unprocessed data from CT scans can contain noise, artefacts, and inconsistencies that may compromise the accuracy and reliability of subsequent analyses. In their work [12], A malorpavam et al. present a very detailed systematic review of existing morphological operations in digital image processing. The primary goal of this work is to change images using mathematical morphology in order to normalize them for the intended analyses. This work shows many approaches, such as erosion, dilation, thresholding, and blurring, can be effectively incorporated in the pre-processing of CT scan images. These methods are used to enhance the contrast between various tissues, remove noise from the images, and remove any non-nodule structures.

B. Segmentation

Developed on the foundation of a fully convolutional neural network, U-Net is a semantic segmentation network. Olaf et al. [13] successfully modelled the convolutional
network-based U-Net architecture, which was created specifically for the segmentation of images in the biomedical field. This work includes a traditional U-Net architecture with an encoder and a decoder and produced an average IOU ("intersection over union") of 92%

A study by Chen et al. [14] proposed a U-Net-based method for segmenting lung nodules in CT images. The proposed method consisted of two stages: the first stage used a pre-trained U-Net model for initial segmentation, and the second stage used a 3D fully connected conditional random field (CRF) for refinement. For the segmentation of lung nodules, Ronneberger et al. [15] developed a U-Net network based on FCN. When segmenting medical images with hazy boundaries, U-Net combines low- and high-resolution information by skipping connections. Low-resolution information is used for target identification, and high-resolution information is used for localising segmentation.

C. Feature Extraction

In a feature pyramid network, a convolutional neural network (CNN) is used to extract features from an input image and create a feature map. Xiaolong Wang et al. [16] proposed a semi-supervised learning method that uses a multi-level feature pyramid network (FPN) to leverage both labelled and unlabeled data. The FPN is used to extract features at different scales, and the method incorporates consistency regularisation to encourage the model to produce similar predictions on labelled and unlabeled data. With only 4,000 labelled samples, the suggested method achieves the best reported classification error rate of 4.42% on the test set. With 4,000 labelled samples and 50,000 unlabeled samples, it also achieves an error rate of 3.57%, which is also the best result to date.

Guangrui Mu et al., [17] proposed Feature pyramid networks with Relu Cascade for CT Pulmonary Nodule Detection. A detection network is first trained with few positive annotations (nodules) and randomly selected negative samples (background). In FPN, extraction is performed separately to produce feature maps of four different scales by using images onto each feature map. The "Relu cascade’s" uniqueness lies in the method used to link these networks together in a cascade.

D. Classification

Convolutional neural network architecture called EfficientNet has produced cutting-edge outcomes on a number of computer vision tasks, including image classification. Agrawal et al., [18] presented various deep learning based pre-trained CNN techniques for distinguishing benign and malignant brain tumor images. They used different optimizers to complete the tasks, namely Adam, RMSprop, and stochastic gradient descent (SGD). Their research demonstrated that a fine-tuned Alexnet could perform particularly well on challenges involving medical imaging. They used data preprocessing and augmentation techniques for boosting diversity in the data samples to decrease the overfitting of the previous models.

Hwejin Jung et al., [19] proposed a three-dimensional deep convolutional neural network (3D DCNN) with dense connections and shortcut connections was developed for the classification of lung nodules. By enabling the gradient to move swiftly and directly, shortcut connections and dense connections successfully address the gradient vanishing problem. This method achieved a highest CPM score of 0.910. Divya et al., [20] presented methods based on transfer learning that enhances current architecture in multi-class classification by relying on pretrained DCNN trained on ImageNet dataset. Pretrained weights of EfficientNetV2-B0 are transferred as initial weights and to distinguish between benign and malignant tissue samples in tumour cells and classify them, the model is fine-tuned.

The objective of this work is to create and construct a deep learning model based on the U-Net segmentation network and Efficient-Net architecture to segment and classify lung nodules in CT scan images. The work addresses various limitations, such as that improvements are needed in the detection rates of lung nodules, as the proposed models have only focused on training dataset image samples and also segmented anomalies as positive samples, an unequal number of benign and malignant lung nodules, resulting in an imbalanced dataset, an accurate diagnosis of the pathological type of lung cancer is crucial for effective treatment; and the performance of model relies heavily on the choice of optimizer.

Developed on the foundation of a fully convolutional neural network, U-Net is a semantic segmentation network. There are a total of 23 layers in the network, which is significantly fewer than the number of layers in other networks while maintaining accuracy [21]. Feature pyramid networks are used to extract feature vectors from segmented images. These feature vectors are used as input into a neural network classifier for an unlabelled nodule prediction task [22]. A family of convolutional neural networks called Efficient-Net was created to achieve cutting-edge performance while being computationally efficient. It accomplishes this by balancing network depth, width, and resolution in order to optimise performance within a predetermined computational budget [23].

The proposed algorithm will be put to the test on the LIDC dataset to see if it can accurately classify and identify specific features, and the outcomes will be compared to standard metrics like accuracy, recall, and F1-score [24]. The proposed algorithm aims to detect and segment lung nodules accurately. Using an ablation study, the project aims to examine the impact of various parameters (hyper-parameters) on the performance of the proposed lung nodule segmentation and classification algorithm [25].

To explore the possibility of employing semi-supervised learning and transfer learning to enhance the performance of the proposed algorithm on datasets with limited labelled data [26]. Adam optimizer is incorporated to stabilize the performance of the model and reduce the effect of errors [27].

III. PROPOSED METHODOLOGY

A. LIDC-IDRI: The Lung Image Database Consortium

The National Cancer Institute spearheaded the collection of the 1018 cases that make up the LIDC dataset, each of which was collaboratively labelled by four radiologists. The
LIDC-IDRI dataset (Armato et al., 2011; Armato III et al., 2015b; Clark et al., 2013) in the Cancer Imaging Archive (TCIA) contains 1018 clinical chest CT scans with lung nodules obtained from seven institutions. The locations of the nodules and the nine semantic attributes of subtlety, sphericity, internal structure, margin, lobulation, spiculation, and malignancy are detailed in an associated XML file for each CT scan that was evaluated for annotation by up to four radiologists (Qin et al., 2019).

It is a database that is widely utilised and devoted to the advancement of current methods for lung nodule segmentation. Three categories of nodules are distinguished: non-nodules (size >=3mm), nodules (size < 3mm), and nodules (size >=3mm). All of the images will be in the DICOM (Digital Imaging and Communications in Medicine) format and 512 x 512 in size. Based on the malignancy details provided in the metadata details provided, around 12000 labelled CT scan images and 3000 unlabelled CT scan images are used.

B. System Architecture

This section provides the detailed architecture and description and algorithmic details of the various subsystems in the proposed system. The detailed architecture of the proposed system is represented in the below Fig. 1. An algorithm has been proposed to segment the lungs in images by using U-Net model and classify the images using Efficient-Net architecture. Unlabelled dataset is labelled by using FPN with Resnet-50 as backbone. FPN extracts features from labelled images and provides these features as input into a neural network classifier. The classifier predicts the class of unlabelled images and these images are added to the dataset for model training.

This model typically involves a series of preprocessing steps, such as Gaussian blur, thresholding, erosion and contouring, to enhance and segment the objects in the images. The Gaussian blur function smoothes the image and reduces noise, while thresholding converts the image to black and white, simplifying it for further processing.

Contouring then isolates the objects within the image by drawing lines around them. The model uses the FPN architecture to extract features from the images when preprocessing is completed. The Efficient-Net model is a state-of-the-art deep learning architecture that has achieved exceptional results in image classification tasks, making it a powerful tool for object recognition.

To achieve this goal, the system will use the U-Net model to segment the lung nodules from the surrounding tissue and the efficient-net architecture to classify the nodules as benign or malignant. The feature extraction is performed using Feature Pyramid networks. The system will also have the potential to be used in other medical imaging modalities, such as MRI and PET, for the detection and classification of other types of abnormalities and diseases.

C. Module Design

The architecture is divided into five modules. The modules are namely preprocessing, augmentation, segmentation, feature extraction and nodule classification, respectively.

1) Pre-Processing: The data pre-processing [28] step is essential to normalise the data in a way that enables the convolutional network to learn appropriate and meaningful features properly because CT scans might be acquired by different scanners in different medical clinics without the use of identical acquisition protocols. Directly obtained CT scans of the lungs have noise, no discernible variations in the greyscale border, and other characteristics that make them difficult to divide. Consequently, the Images Pre-processing is first task. To reduce the noise in the lung CT scans, it is necessary to remove certain regions that make it difficult to separate.

Meanwhile, to make the upcoming segmentation process easier, the boundary portion of the CT picture that changes smoothly needs to be sharpened. Fig. 2 represents the Preprocessing Module which consists of Gaussian Blur, Thresholding, Erosion, Dilation and contouring Techniques.

a) Gaussian Blur: Gaussian blur [29] is a linear low-pass filter, where the pixel value is calculated using the Gaussian function. The rate at which this weight diminishes is determined by a Gaussian function, hence the name Gaussian blur.
\[ G(x, y) = \left( \frac{1}{(2\pi\sigma^2)} \right) \exp\left(-\frac{x^2 + y^2}{2\sigma^2}\right) \] (1)

Where,
\[ x \rightarrow X \text{ coordinate value}, \]
\[ y \rightarrow Y \text{ coordinate value}, \]
\[ \pi \rightarrow \text{ Mathematical Constant PI (value = 3.14)}, \]
\[ \sigma \rightarrow \text{ Standard Deviation}, \]
\[ \exp \rightarrow \text{ exponential function} \]

**Algorithm 1: Gaussian Blur**

**Input:** Image I with dimensions m x n, Gaussian kernel

**Output:** Image O with dimensions m x n

Normalize \( G \), \( G = G / \sum\sum G(x,y) \)

Pad the input image I with zeros to avoid boundary effects.

Amount of padding = (k-1)/2 on each side.

**For each** pixel \((i,j)\) in the input image I:

- Initialize the output pixel \( O(i,j) \) to zero.

**For each** kernel element \((x,y)\):

- Compute the pixel coordinates \((a,b)\):
  \[
  a = i - (k-1)/2 + x \\
  b = j - (k-1)/2 + y
  \]
- Add the product of the input pixel \( I(a,b) \) and the kernel element \( G(x,y) \) to the output pixel \( O(i,j) \):
  \[
  O(i,j) = O(i,j) + I(a,b) * G(x,y)
  \]

End

End

Clip the output image O to the range \([0,255]\)

Return the output image O

**b)Thresholding:** The simplest thresholding methods replace each pixel in an image with a black pixel if the image intensity is less than a fixed value called the threshold or a white pixel if the pixel intensity is greater than that threshold.

**Algorithm 2: Thresholding**

**Input:** Image I with dimensions m x n, Threshold value

**Output:** Image O with dimensions m x n

**For each** pixel \((i,j)\) in the input image I:

- If the pixel value \( I(i,j) \) is less than or equal to the threshold value (255):
  - set the output pixel \( O(i,j) \) to 255 (white)
- Else:
  - The output pixel \( O(i,j) \) remains the same

End

Return the thresholded output image O

**c)Erosion:** Erosion is a morphological image processing technique that is used to remove small details or isolated pixels from an image.

**Algorithm 3: Erosion**

**Input:** Image I with dimensions m x n, structuring element \( B \) with dimensions k x k

**Output:** Image O with dimensions m x n

Pad the input image I with zeros to avoid boundary effects.

Amount of padding = (k-1)/2 on each side

**For each** pixel \((i,j)\) in the input image I:

- Initialize a flag variable \( f \) to true

**For each** structuring element element \((x,y)\):

- Compute the pixel coordinates \((a,b)\):
  \[
  a = i - (k-1)/2 + x \\
  b = j - (k-1)/2 + y
  \]
- If \((I(a,b) == 0 & B(x,y) == 1)\):
  - set \( f \) to False.
End

If \((f == True)\):
  - set the output pixel \( O(i,j) \) to 1
Else:
  - set the output pixel \( O(i,j) \) to 0.

End

End

Remove the padding from the output image O

Return the output image O

**d)Dilation:** Dilation is a morphological image processing technique that is used to enlarge or extend the details in an image.

**Algorithm 4: Dilation**

**Input:** Image I with dimensions m x n, structuring element \( B \) with dimensions k x k

**Output:** Image O with dimensions m x n

Pad the input image I with zeros to avoid boundary effects.

Amount of padding = (k-1)/2 on each side

**For each** pixel \((i,j)\) in the input image I:

- Initialize a flag variable \( f \) to false

**For each** structuring element element \((x,y)\):

- Compute the pixel coordinates \((a,b)\):
  \[
  a = i - (k-1)/2 + x \\
  b = j - (k-1)/2 + y
  \]
- If \((I(a,b) == 0 & B(x,y) == 1)\):
  - set \( f \) to True.
End

If \((f == True)\):
  - set the output pixel \( O(i,j) \) to 1
Else:
  - set the output pixel \( O(i,j) \) to 0.

End

End
In U-Net, the encoder network is the contracting side of the architecture. It consists of convolutional and max-pooling layers that gradually reduce the spatial resolution of the input image while increasing the number of feature channels. This contraction path is designed to capture the contextual information and high-level features of the image. On the other hand, the decoder network is the expansive side of the architecture. It consists of up-sampling and convolutional layers that gradually increase the spatial resolution of the feature maps while reducing the number of feature channels.

This expansion path is designed to reconstruct the high-resolution details of the image and refine the segmentation mask. The encoder and decoder networks are connected by a bottleneck layer that preserves the high-resolution features of the image. The skip connections between the encoder and decoder networks help to combine the high-level features from the encoder with the low-level features from the decoder, resulting in a more accurate segmentation.

4) Feature extraction: A well-versed method in computer vision for various tasks like object detection, segmentation, and classification is feature extraction using Feature Pyramid Networks (FPN) with ResNet as the backbone. FPN is a feature extraction network that enhances the performance of a CNN by utilizing multi-scale feature maps, allowing the network to identify objects at different scales. ResNet, on the other hand, is a deep residual network architecture that has been shown to outperform traditional CNN architectures by allowing for deeper network depths without encountering the vanishing gradient problem.

When used as the backbone for FPN, ResNet provides a strong base for feature extraction and is capable of detecting high-level features in the input image. The FPN architecture takes feature maps from ResNet's intermediate layers and combines them to form a pyramid of multi-scale feature maps.

The top-down pathway in FPN involves up sampling the feature maps of lower resolutions and then combining them with the feature maps of higher resolutions.

Unlabelled nodule prediction:

The FPN is designed to address the problem of detecting objects of various scales in an image. The backbone network is typically a deep convolutional neural network such as ResNet, which is used to extract feature maps from the input image.

- The input image is passed through the first convolutional layer with 64 filters and a kernel size of 5x5.
- The output of the first layer is passed through a batch normalization layer, a ReLU activation layer, and a max pooling layer with a kernel size of 3x3 and stride of 2.
- The output of the first stage is passed through the second stage, which consists of 3 residual blocks with 256 filters and a kernel size of 3x3.
The output of the second stage is passed through the third stage, which consists of 4 residual blocks with 512 filters and a kernel size of 3x3.

The output of the third stage is passed through the fourth stage, which consists of 6 residual blocks with 1024 filters and a kernel size of 3x3.

The output of the fourth stage is passed through the fifth stage, which consists of 3 residual blocks with 2048 filters and a kernel size of 3x3.

The output of each stage is then passed through a feature pyramid network (FPN) module to create a feature pyramid. The feature pyramid is then used for Classification or other downstream tasks.

Classifier:

The classifier model is trained using two different class numpy array files, which are typically generated from a training dataset. One array file contains the extracted feature vectors of benign images, while the other array file contains the extracted feature vectors of malignant images.

5) Nodule classification: Efficient Net [31] is a convolutional neural architecture and scaling method that uniformly scales all dimensions of depth/width/resolution using a compound coefficient. Unlike conventional practice that arbitrary scales these factors, the Efficient Net scaling method uniformly scales network width, depth, and resolution with a set of fixed scaling coefficients.

Efficient net architecture provides compound scaling method (scaling all depth, width, and resolution dimensions) can help the model achieve the greatest accuracy gains. The baseline network has a significant impact on how well models scale. An extensive range of image classification tasks can be handled by Efficient Net.

It is a good model for transfer learning because of this feature. Efficient-Net performs Compound Scaling - that is, scale all three dimensions (depth, width, image resolution) while maintaining a balance between all dimensions of the network. The architecture of Efficient-Net consists of different Convolution and MBConvolution blocks interconnected together to produce feature maps of images.

MBConv:

This developed architecture uses the mobile inverted bottleneck convolution (MBConv). Then two more ideas are borrowed from MobileNet-V2 (which is a second improved version of MobileNet) including inverted residual connections, Linear bottlenecks. Mobile inverted bottleneck convolution (MBConv) is the main building block of Efficient-Net model family. MBConv is based on concepts borrowed from the MobileNet models.

Operations:

- The first operation is a linear transformation, which is performed on the input features using the weights of the first layer.
- The input features X are multiplied by a weight matrix W1, which has dimensions (input_dim, units). The result of this multiplication is a matrix of activations with dimensions (batch_size, units).
- The next operation is the application of a non-linear activation function to the activations.
- The process is repeated for subsequent layers, with each layer applying a new linear transformation to the activations of the previous layer, followed by a non-linear activation function.
- The final layer has a single unit and uses the sigmoid activation function, which maps the activations to a value between 0 and 1, representing the predicted probability of belonging to the class.
- During training, the model uses the back-propagation algorithm to compute the gradients of the loss with respect to the weights of each layer, which are used to update the weights during the optimization process.
- These gradients are computed using the chain rule of calculus, which involves computing the derivative of the loss with respect to the output of each layer, and then propagating these derivatives backwards through the layers of the network.
- The model can learn to predict the correct label for each input sample, based on the features in the input data.

IV. RESULTS AND DISCUSSIONS

The project was executed using python language and Anaconda Jupyter tool. Python is among the most widely used programming languages used for deep learning due to its flexibility and large number of available libraries, such as TensorFlow, PyTorch, Keras, and Scikit-learn. Python programming language's distribution Anaconda includes a collection of commonly used data science packages and tools. Jupyter is included as part of the Anaconda distribution. GPUs are designed to handle massive amounts of parallel processing, making them ideal for training complex neural networks.

The handling of large datasets, which is necessary for training our deep learning models, is made possible by the 16GB of memory. Running deep learning software and tools like TensorFlow and PyTorch requires a stable and user-friendly environment, which the Windows operating system offers.

The U-Net was given a collection of pre-processed CT images to train the model on. The weights were initialised with normal initialization and the ReLU activation function. The learning rate was set at 0.001 and the batch size to 32.

The binary cross-entropy loss function was utilised to calculate the loss function. In comparison, 80% of the data are utilised for training and 20% for validation. The model was trained for 250 epochs. The average dice score and best dice score resulted in values of 0.4273 and 0.5009. The average
loss of the model was found to be 0.27. The nodule segmented using U-Net is represented in Fig. 3.

![Nodule segmented image.](image)

Fig. 3. Nodule segmented image.

An alternative to the stochastic gradient descent approach for deep learning models is the Adam optimizer. This approach handles sparse gradients on noisy situations and provides optimization by integrating the key characteristics of AdaGrad and RMSProp. ReLU function resists simultaneously stimulating all neurons since it is a non-linear activation function.

In order to get a good result, the suggested U-Net model uses ReLU activation. The neuron will become inactive if the outcome of the linear transformation is less than 0. Results were satisfactory when the ReLU activation function was used with the Adam optimizer.

The training vs validation accuracy graph is a plot that shows the accuracy of a model on both the training and validation datasets. In Fig. 4, the blue line represents the training accuracy, while the red line represents the validation accuracy. The graph shows that the model is improving its accuracy on both the training and validation datasets from the first few epochs.

![Training and Validation Accuracy](image)

Fig. 4. Training vs validation accuracy graph.

The training vs validation loss graph is a plot that shows the loss of a model on both the training and validation datasets during the training and validation processes. During the training process, the model is trained on a training dataset to minimize the loss function. In Fig. 5, the blue line represents the training loss, while the red line represents the validation loss.

![Training and Validation Loss](image)

Fig. 5. Training vs validation loss graph.

Accuracy (Table I) measures the percentage of correctly classified instances. It is the most basic metric and gives a good overall measure of the model’s performance.

<table>
<thead>
<tr>
<th>S. No.</th>
<th>Epochs</th>
<th>No. of Images</th>
<th>Batch Size</th>
<th>Activation Function</th>
<th>Test Accuracy</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>25</td>
<td>13500</td>
<td>32</td>
<td>RELU</td>
<td>81.33%</td>
</tr>
<tr>
<td>2</td>
<td>25</td>
<td>16000</td>
<td>32</td>
<td>RELU</td>
<td>91.67%</td>
</tr>
<tr>
<td>3</td>
<td>25</td>
<td>16000</td>
<td>64</td>
<td>SWISH</td>
<td>83.70%</td>
</tr>
<tr>
<td>4</td>
<td>100</td>
<td>16000</td>
<td>32</td>
<td>RELU</td>
<td>91.34%</td>
</tr>
</tbody>
</table>

The False Positive Rate (FPR) is the ratio of all the benign nodules that are falsely identified as malignant nodules. The False Negative Rate (FNR) is the ratio of all the malignant nodules that are incorrectly identified as benign nodules.

<table>
<thead>
<tr>
<th>S. No.</th>
<th>True Negative</th>
<th>True Positive</th>
<th>False Negative</th>
<th>False Positive</th>
<th>FPR</th>
<th>FNR</th>
<th>DR</th>
<th>Recall</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>1447</td>
<td>1658</td>
<td>191</td>
<td>101</td>
<td>0.06</td>
<td>0.10</td>
<td>0.89</td>
<td>0.86</td>
</tr>
<tr>
<td>2</td>
<td>1627</td>
<td>1150</td>
<td>222</td>
<td>398</td>
<td>0.19</td>
<td>0.16</td>
<td>0.83</td>
<td>0.74</td>
</tr>
<tr>
<td>3</td>
<td>1920</td>
<td>1227</td>
<td>127</td>
<td>73</td>
<td>0.03</td>
<td>0.09</td>
<td>0.86</td>
<td>0.85</td>
</tr>
<tr>
<td>4</td>
<td>1087</td>
<td>1594</td>
<td>251</td>
<td>465</td>
<td>0.29</td>
<td>0.13</td>
<td>0.81</td>
<td>0.76</td>
</tr>
</tbody>
</table>
Detection Rate (DR) is the ratio of benign nodules that are correctly identified as malignant nodules and vice-versa. Table II represents the different parameters calculated using the classification results.

V. CONCLUSION AND FUTURE WORKS

The proposed model for lung nodule segmentation and classification using U-Net and Efficient-Net neural network has shown promising results in detecting and classifying lung nodules accurately and efficiently. The proposed algorithm outperforms state-of-the-art methods in terms of accuracy, recall, loss. The ablation study conducted in this project revealed the contribution of each component of the proposed algorithm to the overall performance.

Furthermore, the proposed algorithm has the potential to be applied to larger datasets and to be optimized for real-time processing. The successful implementation of this algorithm can potentially improve the accuracy and efficiency of lung cancer screening programs and contribute to the early detection and treatment of lung cancer. However, the proposed algorithm still has limitations, such as the need for large annotated datasets and the potential for over fitting.

The proposed algorithm can be improved by incorporating explainable artificial intelligence (XAI) techniques to enhance the interpretability and transparency of the algorithm. The inclusion of XAI techniques can enable the identification of the features and patterns that the algorithm uses to make decisions, thus providing insights into the reasoning behind the algorithm’s output. This can be particularly valuable for medical applications, where the decision-making process needs to be transparent and understandable to clinicians and patients.

The proposed algorithm can be optimized for multimodal imaging, such as computed tomography (CT) and positron emission 40 tomography (PET) scans, to improve the accuracy of lung nodule segmentation and classification. Additionally, the proposed model can be extended to address other types of lung cancers, such as small cell lung cancer (SCLC) and non-small cell lung cancer (NSCLC). Therefore, future work can focus on developing a more comprehensive algorithm that can detect and classify various types of lung abnormalities accurately and efficiently.

REFERENCES


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Providing an Improved Resource Management Approach for Healthcare Big Data Processing in Cloud Computing Environment

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Abstract—Due to the gathering of big data and the advancement of machine learning, the healthcare industry has recently experienced fast change. Acceleration of operations related to the analysis and retrieval of healthcare data is essential to facilitate surveillance. However, providing healthcare to the community is a complex task that is highly dependent on data processing. Also, processing health metadata can be very expensive for organizations. To meet the strict service quality requirements of the healthcare industry, large-scale healthcare data processing in the cloud confederation has emerged as a viable option. However, there are many challenges, including optimal resource management for metadata processing. Based on this, in the present study, a fuzzy solution for determining the optimal cloud using the resource forecasting technique is presented for health big data processing. During job processing, a fuzzy selection-based VM migration technique was used to move a virtual machine (VM) from a high-load server to a low-load server. The proposed architecture is divided into regional and global levels. After evaluating the local component, requests are sent to the global component. If the local component cannot meet the requirements, the request is sent to the global component. The hierarchical structure of the proposed method requires the generation of delivered requests before estimating the available resources. The proposed solution is compared with PSO and ACO algorithms according to different criteria. The simulation results show the effectiveness and efficiency of the model compared to alternative methods.

Keywords—Healthcare; big data; cloud confederation; service quality; Cloud Resource Management (CRM)

I. INTRODUCTION

The existing processes for collecting and processing the vital data of patients demand a large volume of workforce and calculations. Usually, these processes are error-prone and have large delays, which prevent the right data from being available in real-time [1]. As a novel technology, cloud computing with internet infrastructure and novel solutions provides considerable advantages in providing medical services electronically [2]. Indeed, the advent of cloud computing has brought a fundamental change in the path of the novel, developed, scaled, and up-to-date services of information technology [3]. With the fast development in processing and storage technologies and also the successfulness of the internet, the computing of resources has become cheaper, stronger, and more accessible than ever before, and governmental organizations have started to use architecture, platforms, and programs of cloud computing to provide the services and satisfying the needs of their subsidiaries [4]. However, there are still many challenges in optimal resource management and to what extent each data needs the computing resource are among these challenges [5].

The objective of the resource management system for the cloud is to satisfy the needs of the applications using it. Because it is possible, during the process, that some of the servers have high traffic load while the others have low or no load, the resource management mechanism should check the current situation of every resource in the cloud environment to provide the algorithms for better allocation of physical or virtual resources and thus reduce the operational costs in the cloud environment [6]. In this case, not only may the workload be spread among the available but underutilized servers, but the unused servers can be shut off altogether. Centralized work allocation to servers that do not take into account the individual solutions is obviously unfeasible in such large and complicated systems [7]. Also, according to the rapid growth in metadata in the advanced data centers and the urgent need to access good service quality, the necessity for providing solutions in order to increase the productivity of the available service providers in the metadata processing center is strongly felt. One way to achieve the desired productivity is to use resource management solutions [8]. For appropriate management of the resources of service providers, the load balance is required [9]. With an appropriate load balance, the requests will be distributed in a dynamic and balanced form among all nodes while ensuring the fairly and efficient allocation of every computing resource [10]. This process will result in increased user satisfaction and high productivity of the resources, which eventually increases productivity [11]. The load balancing aims to find an appropriate mapping of the tasks on the processors available in the system so that in each processor, approximately an equal number of tasks will be executed to minimize the overall runtime [12]. In recent years, due to population growth and advances in medical science, the number of discovered diseases has increased [13]. The information on each patient entered into hospital systems is so huge and complex that it has been classified as "big data," requiring plenty of resources to process [14]. It is important to pay attention to the fact that cloud computing has entered all fields, including the medical field, at a very high speed today [15]. Considering the high number of information related to a person that may be available in the hospital system, it is possible to use cloud processing instead of local storage of the person's information so that there is less need for local and physical facilities and the
speed of obtaining and processing the person's information in time multiplied the need. One of the prominent points of this method is that there are no worries about losing information due to local system damage, and information can be stored, processed, or retrieved at any time, regardless of the hospital system load. One of the biggest challenges here is choosing the type of resources to store information, whether local resources are more optimal or resources that can be rented from other places, or more importantly, how much resources each data needs to be able to store it. It is possible to select the most optimal resource by predicting the number of resources each piece of information needs and allocating it to it. In this research, we intend to present a new approach to improve the management of healthcare big data resources, and by using this method, we will significantly improve resource management by facilitating and accelerating the processing of medical care data in cloud computing.

This study predicts and attempts to implement the necessary resources for processing RIMA model data (big data) on drug consumption in various American states between 2018 and 2021. The fuzzy DEMATEL method has the lowest prediction error as well as closely resembles the reality of a request's resources. Thus, we employ it to allocate local and remote assets for each request. This algorithm was developed with profits and profits as its primary motivation. In a nutshell, the following is the author's contribution to this article:

- Using the RIMA model to predict the required resources with a low error percentage.
- Using the fuzzy Dematel technique to choose a cloud data center to accelerate and facilitate big data processing in the cloud environment.
- Increasing productivity by maintaining user QoS and reducing response time.

This paper's remaining sections are structured as follows. The context is provided in Section II. Section III provides fuzzy DEMATEL methods for evaluating RIMA-based cloud and resource forecasting methods. Evaluation and simulation of the suggested model are presented in Section IV, followed by a discussion of the results. Conclusion and directions for further research is presented in Sections V.

II. LITERATURE REVIEW

About [16], a review of the common algorithms in the field of load balance in metadata is provided. One of the most popular algorithms is the Min-Min algorithm. At the outset, the algorithm is given a pool of jobs from which to choose. The first thing to do is determine the quickest possible way to do every activity. The quickest task on each available resource is then chosen, and its corresponding minimum finish time is chosen. The next step is to allocate the required amount of time for the task on the appropriate machine. Besides optimal scheduling, the approach has a main issue, which can lead to famine. In research [17], a hybrid task scheduling strategy has been provided for managing and processing medical metadata in cloud VMs. The tasks' migration from one server to another has been addressed. In this research, the genetic algorithm, along with the particle swarm optimization, was used to provide the load balance and allocate the available network resource to the tasks so that the distribution of tasks between the resources was performed fairly and the servers were able to execute the processing on different requests of users simultaneously. In this research, the proposed algorithm has been simulated by CloudSim software, and the results indicate the effectiveness of this approach in load balance. In [18], a load balance and scheduling model based on the cloud segmentation concept has been proposed. A hybrid algorithm was provided in which the RR2 scheduling algorithm and game theory were used. This algorithm selects a low-load server with a high execution speed. Time slicing is done better in RR because the better server is selected. The results show that the load balance has improved using the proposed algorithm. In study [19], a cloud confederation model which provides ideal selections for target cloud providers has been proposed to address the heterogeneous IoT metadata customers' demands. In addition, a multi-purpose optimization model has been provided. A general structure for the genetic algorithm has been developed to solve this model. Different evaluation tests have tested the proposed model. In study [20], a real model based on linear programming has been presented to identify the strategies and decisions in the cloud federation. The user requests are established on all cloud federation levels, and the users are easily directed. As a result, the obtained advantages motivate providers' participation by free insourcing to support the other federation members. The proposed model is highly scalable. In [18], multi-layer resource allocation has been provided. The main idea behind this was based on the allocation of resources in a layering approach, which configures the resource allocation for personal tasks on a cluster node based on the resource exploitation levels. The results show that MTRA has improved the performance and runtime by 18% and 10%, respectively. In [21], it is examined how end users choose MEC servers, how they offload their data optimally, and how MEC servers determine their prices optimally in a multi-MEC server and multi-end user scenario. An SDN controller first implements a reinforcement learning framework based on stochastic learning automata to enable end users to choose a MEC server to load their data. The whole MEC selection process considers the MEC server's method, its congestion and penetration in terms of completing end users' computing duties, and its declared price for its computing services. A non-cooperative game is developed between the end-users of each server to determine the end user's data loading portion to the chosen MEC server, and the existence and uniqueness of the corresponding Nash equilibrium are demonstrated. An iterative and simple method has been developed to implement the suggested framework. The performance of the suggested technique was assessed through modeling and simulation in various scenarios with both homogeneous and heterogeneous end users. In [22], author offers thorough analyses of prior cognitive computing research, problems, fixes, and prospects for further study. There is a focus on cognitive computing-based methods for resolving practical issues in the four extensively studied application fields of healthcare, cyber security, big data, and the Internet of Things. Author examined research from 2012 to 2020 in [23]. This study aims to offer a thorough analytical understanding of big data for health care. Three goals were met in this study's
investigation: (a) identifying the most important themes in big data research on healthcare; (b) assessing the relationships between the themes in previous studies, and (c) creating time-series profiles for the topics. Healthcare research using big data has covered a variety of subjects. Based on the study's findings, a summary of researchers' interests in various techniques, technologies, and topic areas is given, along with a list of critical research gaps that need to be filled in the future. The advantage and disadvantages of different technologies are thoroughly described in research [24], along with the range of applications for each. Attacks, according to background information gleaned through data integration, can be avoided using various technologies, anonymity, and privacy during data collection. The majority of important medical data is kept in storage on a cloud computing platform. Encryption and auditing techniques are frequently employed to maintain the confidentiality and integrity of stored information. Access control techniques are also utilized during the data-sharing phase to control the objects with access to the data. Big medical and health data privacy protection is carried out under machine learning during the data analysis stage. Finally, acceptable management-level concepts have emerged due to broad privacy protection worries across the medical big data lifecycle across the sector. Table I discusses a summary of the comparison of existing techniques in resource management.

<table>
<thead>
<tr>
<th>Work</th>
<th>Target parameter</th>
<th>Method</th>
<th>Type of technique</th>
<th>Advantages</th>
<th>Disadvantages</th>
</tr>
</thead>
<tbody>
<tr>
<td>Miah et al. 2022 [25].</td>
<td>Achieving policies that help increase profits and overcome resource limitations.</td>
<td>Efficient resource management strategy, deployment of a cloud federation to coordinate resources</td>
<td>Model-based</td>
<td>1. It gives providers more flexibility in resource management and increased productivity. 2. Customers can maintain the machines for a long time if needed.</td>
<td>Outsourcing of spot requests is not considered in this method. Also, not having strategies that help to predict the availability of future resources for decision-making is one of the disadvantages of this method.</td>
</tr>
<tr>
<td>AbdulAhmeed et al. 2017 [26].</td>
<td>Explore a cooperative cloud approach to form a confederation to maximize CPs' profits and meet SLAs</td>
<td>Constructing a model of cloud federation to identify top-tier cloud service providers.</td>
<td>Multi-objective (MO) optimal model, genetic algorithm</td>
<td>Increase revenue for IOTDCs by sharing available virtual and physical resources</td>
<td>Heterogeneous request problems IOTDCs receive a large number of requests.</td>
</tr>
<tr>
<td>Misis et al. 2019 [27].</td>
<td>Data upload through MEC server.</td>
<td>SDN controller and reinforcement learning based on the concept of stochastic learning automata are used to load data on a MEC server.</td>
<td>SDN controller and reinforcement learning</td>
<td>Use of MEC technique, high security</td>
<td>Need to optimize the problem</td>
</tr>
<tr>
<td>Dhiman et al. 2022 [28].</td>
<td>Maximizing provider benefits in cloud federation</td>
<td>A linear numerical program for optimizing input workload distribution across federation members</td>
<td>Linear numerical programming algorithm, Branch &amp; bound algorithm</td>
<td>Access to the highest incomes, minimizing energy consumption, reducing operating costs, scalability</td>
<td>Requires a lot of knowledge for modeling</td>
</tr>
<tr>
<td>Navroop Kaur et al. 2016 [29].</td>
<td>Appropriate cloud resource allocation has become a research problem due to the high growth of cloud big data.</td>
<td>Proposing an effective resource management system for big data streams</td>
<td>Dynamic Clustering Algorithm by SOM (Self-Organized Map)</td>
<td>Reduce waiting time</td>
<td>Need to optimize the problem</td>
</tr>
<tr>
<td>ThomasRyan et al. 2021 [30].</td>
<td>Create a method for managing resources that can function effectively in settings with varying levels of complexity.</td>
<td>Allocation of resources for individual tasks in a layered approach of dynamic resource allocation</td>
<td>MTRA technique (Multi Tier Resource Allocation (MTRA))</td>
<td>Performance improvement up to 18%, runtime improvement up to 10%, increased scalability</td>
<td>Increasing the complexity of the system</td>
</tr>
</tbody>
</table>

III. PROPOSED METHOD

This section suggests a method for choosing the best cloud using the DEMATEL fuzzy technique as well as a method for resource prediction using the RIMA algorithm. The proposed architecture's structure is divided into two local and one global component. Before sending a request to the global part, the local function should be examined; if the local domain is unable to supply the necessary data, the request is forwarded to the global domain. Since the suggested procedure has a hierarchy architecture, the local data component receives the user request. The local analyzer examines and keeps track of it to forecast the resources. On this foundation, the RIMA algorithm first estimates the size of the available resources before creating the list of paid requests. The structure of the suggested architecture makes it clear that each IOTDC is presumed to have a Broker. The requests from users referring to each IOTDC are first received and examined by the Broker associated with that IOTDC. If there are sufficient resources to carry out the users' requests, do so and inform the IOTDC's users of the results. Even yet, imagine the Broker is unable to fulfill user requests with the IOTDC resources that are at his disposal. If so, it should utilize the Dematel fuzzy algorithm, which every Broker can access, to select the right IOTDC from the linked IOTDC. As a result, there is a Broker with the following components for each IOTDC:
The Broker's structure and its individual parts are shown in Fig. 1. A fuzzy selector-based VM migration technique is used after choosing the processing server to make sure that, should it become overloaded while processing or because of software or hardware violations, its VM can be moved to a low-load or available idle server, alleviating the overload as well as maximizing the server's useful life.

A. Receiving the Requests

In algorithm No. 1, requests are first added to a queue as well as then, in accordance with demand, are directed to the desired resource. The requests are positioned in the queue according to this algorithm. Each user's request is added to the queue (lines 2-4) because each user has a variety of requests. For all users, this step is repeated (line 1). The resources needed for each request are described in lines 6 through 8 that follow.

Algorithm 1: add Request to Queue

for i ← 1 to n do
    for j ← 1 to Req.Count do
        ADD Req to Request_queue
    end for
while (Request_queue is not empty) do
    foreach req in Request_queue do
        Send req to LCC
    end for

This stage is in charge of compiling the requests that have been made as well as the available resources. Two request and resource stages are part of the monitor phase. The phase of resource monitoring is in charge of acquiring data regarding resource productivity, resource capacity usage, and maximum VM count. The request monitoring phase is in charge of compiling data on the volume of requests made by users. These two subcomponents combine the observed data, which is then saved for use by the knowledge base's other phases.

Algorithm No. 2 keeps tabs on the services provided by each GCC and GCC resource. The procedure is run up until free resources (line 1) are present. This method examined local and global resources, although local resource monitoring is given top attention. Lines 8-12 analyze the services provided by each of the GCC's available IOTDCs. First, the number of services provided by local resources is verified; if none are available, line 6 makes reference to the broader cloud federation's local resources.

Algorithm 2: Resource monitoring

1: while (reso. = 0) do
2: for i ← 1 to n do
3: for j ← 1 to m do
4: if (IOTDC is available in our gcc) then
5: if (IoTDC is available in gcc) then
6: else
7: if (IoTDC is available in gcc) then
8: for i ← 1 to n do
9: for j ← 1 to m do
10: if (IoTDC is available in gcc) then
11: else
12: do resource = 0
end

B. Prediction Phase

The resource's forecast phase is in charge of determining how many resources will be needed to fulfill requests. Low supply happens when the available resources fall short of meeting demand, and high supply occurs when the available resources outpace actual needs. Therefore, the RIMA technique can be used to estimate the requirements for a given resource reliably. The third method makes use of a RIMA semi-code to estimate future service demands. Each request from the user has been viewed as an independent input, and features specific to that request have been defined using the x. Each request's amount of resource requirements is determined (line 4) based on the request's characteristics and the tracking factor of every modification. Z computes the upper bound of the genuine ask.

C. Resource Management Phase

The resource predictor reports back to the resource management of its origin after gauging the resource for the given request. The resource manager determines whether or not the new request can be fulfilled by the available resources after checking their availability. If no resource is available for the IOTDC host, DEMATEL is used to discover the best possible unified IOTDC. If the resource is accessible, it shouldn't be much larger than required for the request to avoid a high allocation.

Algorithm 3: RIMA Prediction

1: while (req ≥ 0) do
2: F or i ← 1 to 3 do
3: F or j ← 1 to n do
4: sum = sum + VL (j)
5: end for
6: μ = sum / n
7: for j ← 1 to n do
8: MA[i] = μ + (MA[i] * WL[n - j] + ε
9: end for
10: for j ← 1 to n do
12: end for
13: for j ← 1 to n do
14: L = Sqrt(φ[i] * WL[n - j] + ε
15: end for
16: Predict[i] = L + α + AR[i] + MA[i]
17: end for
18: return { Predict }

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D. IOTDC Selector Phase

For each request, an appropriate resource allocation is made in the IOTDC selection step. Using the fuzzy DEMATEL method, this component chooses the best cloud data center [14]. The fuzzy triangle variables are needed to display the evaluation criterion. Fig. 2 displays the numerical range and each of these variables' linguistic representations.

![Fig. 2. Provides the linguistic representation of each of these variables and their numerical range.](image)

To execute pairwise comparisons, the fuzzy numbers shown in Table II must first be defined in order for the responses to be appropriately delivered. Triangular fuzzy numbers have been employed in the analysis shown, as seen in the table. Since m is a triangular fuzzy number, it may be computed from equation (1) as follows:

\[
S_i = \sum_{j=1}^{m} M_{ij} \otimes \left[ \sum_{j=1}^{m} M_{ij} \right]^{-1}
\]

Within the pairwise comparisons matrix in the relationship mentioned above, M are triangular fuzzy numbers. In order to calculate the matrix S, we actually add each fuzzy number component individually and then multiply by the fuzzy inverse of the final sum. The normalized weights are computed in this phase similarly to the normal AHP approach, but with fuzzy numbers. The preliminary definitions should be made before introducing the suggested approach, followed by a presentation of the suggested algorithm. The investigation of fuzzy triangular numbers came first. Fuzzy triangular numbers are used in FAHP-based algorithms to verify criteria. The membership function for triangular numbers is stated as follows, and each triangle fuzzy number is displayed using three (n1, n2, n3).

All the operations that are defined for fuzzy numbers are performed in triangular fuzzy numbers on triples (n1, n2, n3).

Equation (2) defines membership conditions for triangular fuzzy numbers.

\[
\mu_{M}(x) = \begin{cases} 
\frac{x-n_1}{n_2-n_1} & x \in [n_1, n_2] \\
\frac{n_3-x}{n_3-n_2} & x \in [n_2, n_3] \\
0 & \text{otherwise}
\end{cases}
\]

Table II also shows triangle representation and linguistic interpretation.

The main goal is to pick the best IOTDC from the set of confederation IOTDCs using Fuzzy DEMATEL. The organizational structure of the appropriate confederation has been informed by the following factors: calculation costs (C1), a distance of IOTDCi requested from IOTDCj confederation (C2), a departure from agreements (C3), as well as available resources (C4). The suggested approach for selecting the IOTDC is depicted in Algorithm (4). This method features a three-tiered, hierarchical structure; the ground level, the top level, and levels 1 and 2. The algorithms are bilevel, as denoted by the second line of code. The criteria count is assumed to be n at each tier. The first-level execution is given below; the second-level algorithm follows the same pattern. As such, the algorithm begins by reading the values from the fuzzy comparing matrix and then compares each Si pair in turn (lines 2-3). The order of importance of these criteria is then determined using the connection (3). An anomalous weight is calculated for each criterion. Then, using a relation, the weights are normalized (4). These steps are likewise carried out on the second floor, which is where you'll find IOTDCs. The resulting weight represents the importance of each IOTDC, and the one with the highest importance can be chosen.

<table>
<thead>
<tr>
<th>Algorithm 4: Fuzzy DEMATEL</th>
</tr>
</thead>
<tbody>
<tr>
<td>1: First, get tracking data.</td>
</tr>
<tr>
<td>2: for both 1st and 2nd graders</td>
</tr>
<tr>
<td>2-1: Input a fuzzy rating of the relevance of each criterion pair (Table II).</td>
</tr>
<tr>
<td>2-2: All criteria C1Cn (i1...n)</td>
</tr>
<tr>
<td>2-2-1: The formula is as follows: 2-2-1: Si (E.q 1) = Fuzzy Synthetic Extent</td>
</tr>
<tr>
<td>2-3: Two-Three: i, j, 1,..., n</td>
</tr>
<tr>
<td>2-4: Determine V if Si &gt; Sj,</td>
</tr>
<tr>
<td>2-5: Use EQ to derive your actual weight for steps. 1-3</td>
</tr>
<tr>
<td>2-6: Use (EQ 2) to normalize the given Wight value.</td>
</tr>
<tr>
<td>3: Prioritize the return of the final Wight in the IOTDC.</td>
</tr>
</tbody>
</table>

### Table II. Triangular Fuzzy Numbers

<table>
<thead>
<tr>
<th>Linguistic Interpretation</th>
<th>Fuzzy Numbers</th>
<th>Membership Function</th>
<th>Interval</th>
<th>Linguistic Interpretation</th>
</tr>
</thead>
<tbody>
<tr>
<td>Just Equal</td>
<td>̃1</td>
<td>μm(X)=(3-X)/(3-1)</td>
<td>1≤X≤3</td>
<td>(1,1,1)</td>
</tr>
<tr>
<td>Equally Important</td>
<td>̃1</td>
<td>μm(X)=(3-X)/(3-1)</td>
<td>1≤X≤3</td>
<td>(1,1,3)</td>
</tr>
<tr>
<td>Weakly Important</td>
<td>̃3</td>
<td>μm(X)=(X-1)/(3-1)</td>
<td>1≤X≤3</td>
<td>(1,3,5)</td>
</tr>
<tr>
<td>Essential Or Strongly Important</td>
<td>̃5</td>
<td>μm(X)=(X-3)/(5-3)</td>
<td>3≤X≤5</td>
<td>(3,5,7)</td>
</tr>
<tr>
<td>Very Strongly Important</td>
<td>̃7</td>
<td>μm(X)=(X-5)/(7-5)</td>
<td>5≤X≤7</td>
<td>(5,7,9)</td>
</tr>
<tr>
<td>Extremly Preferred</td>
<td>̃9</td>
<td>μm(X)=(X-7)/(9-7)</td>
<td>7≤X≤9</td>
<td>(7,9,9)</td>
</tr>
</tbody>
</table>

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The weight vector is found using the following formula for \(i=k: k=1,2,...,n\).

\[
W' = (d'(A_1), d'(A_2), ..., d'(A_n))^T \tag{3}
\]

Then, we normalize the vector \(W'\), and we get the following:

\[
W = (d(A_1), d(A_2), ..., d(A_n))^T \tag{4}
\]

E. Fuzzy Balancer

Despite efforts to choose the best available resource, certain servers may fail to execute commands in the processing of healthcare metadata for a number of reasons (like a delay in processing the preceding commands, software and hardware errors, congestion, or overload) [31, 39]. In these cases, it is necessary to migrate virtual machines (VMs) from a node with a heavy traffic load to another node and to identify as well as choose which VMs should be moved; a fuzzy selector is used. The linguistic variables and their corresponding membership functions are established on this foundation.

1) Membership functions: In this section, the membership functions are described based on the defined linguistic variables. Indeed, by using three defined input variables, a VM fuzzy selector has been provided. On this basis, the membership functions are needed to be defined. The range used for the membership functions has been defined based on the outputs of the CloudSim simulation of the PlanetLab project [32]. For every main parameter related to the resource, the membership functions related to the linguistic variables are defined as follows:

- The available storage (\(T_{\text{RAM}}\)): (low, medium, high)
- Execution time (\(T_{\text{exe}}\)): (low, medium, high)
- Energy consumption (\(T_{\text{energy}}\)): (low, medium, high)

Now, based on the definition of the three parameters mentioned above, the VMs are categorized into classes (low, medium, high, and very high), and the machines which are located in the high and very high classes are the candidates to migrate to VMs with less and medium loads. In the following, tables and figures related to membership functions' parameters are shown. First, Table III and Fig. 3 show the membership functions related to the consumed RAM to define the low, medium, and high classes.

In Table IV and Fig. 4, the parameters of the membership functions related to energy consumption have been shown.

Finally, Table V and Fig. 5 show the parameters of the membership functions related to consumed energy.

<table>
<thead>
<tr>
<th>Parameters</th>
<th>RAM amount</th>
</tr>
</thead>
<tbody>
<tr>
<td>(a=0, b=0, c=651, d=751)</td>
<td>Low</td>
</tr>
<tr>
<td>(a=701, b=801, c=901, d=1001)</td>
<td>Medium</td>
</tr>
<tr>
<td>(a=900, b=1101, c=1801, d=1800)</td>
<td>High</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Parameters</th>
<th>Energy amount</th>
</tr>
</thead>
<tbody>
<tr>
<td>(a=0, b=0, c=121, d=136)</td>
<td>Low</td>
</tr>
<tr>
<td>(a=131, b=141, c=151, d=156)</td>
<td>Medium</td>
</tr>
<tr>
<td>(a=151, b=161, c=171, d=171)</td>
<td>High</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Parameters</th>
<th>Amount</th>
</tr>
</thead>
<tbody>
<tr>
<td>(a=0, b=0, c=0.07, d=0.08)</td>
<td>Low</td>
</tr>
<tr>
<td>(a=0.066, b=0.07, c=0.09, d=0.2)</td>
<td>Medium</td>
</tr>
<tr>
<td>(a=0.08, b=0.2, c=0.12, d=0.16)</td>
<td>High</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Parameters</th>
<th>Execution Time (second)</th>
</tr>
</thead>
<tbody>
<tr>
<td>(a=0, b=0, c=0.07, d=0.08)</td>
<td>Low</td>
</tr>
<tr>
<td>(a=0.066, b=0.07, c=0.09, d=0.2)</td>
<td>Medium</td>
</tr>
<tr>
<td>(a=0.08, b=0.2, c=0.12, d=0.16)</td>
<td>High</td>
</tr>
</tbody>
</table>
1) Fuzzy rules: In this section, based on the membership functions described in the previous step, the fuzzy rules are defined in order to select the VMs for migration. These rules are used to select the VMs that faced overloading due to the additional load. The corresponding rules are shown in Table VI.

<table>
<thead>
<tr>
<th>Execution</th>
<th>RAM</th>
<th>Energy</th>
<th>VM Selection</th>
</tr>
</thead>
<tbody>
<tr>
<td>H</td>
<td>H</td>
<td>H</td>
<td>VH</td>
</tr>
<tr>
<td>H</td>
<td>H</td>
<td>M</td>
<td>VH</td>
</tr>
<tr>
<td>H</td>
<td>H</td>
<td>L</td>
<td>VH</td>
</tr>
<tr>
<td>H</td>
<td>M</td>
<td>H</td>
<td>VH</td>
</tr>
<tr>
<td>M</td>
<td>M</td>
<td>M</td>
<td>H</td>
</tr>
<tr>
<td>H</td>
<td>M</td>
<td>L</td>
<td>H</td>
</tr>
<tr>
<td>H</td>
<td>L</td>
<td>H</td>
<td>H</td>
</tr>
<tr>
<td>H</td>
<td>L</td>
<td>M</td>
<td>M</td>
</tr>
<tr>
<td>H</td>
<td>L</td>
<td>L</td>
<td>M</td>
</tr>
<tr>
<td>M</td>
<td>H</td>
<td>H</td>
<td>VH</td>
</tr>
<tr>
<td>M</td>
<td>H</td>
<td>M</td>
<td>H</td>
</tr>
<tr>
<td>M</td>
<td>H</td>
<td>L</td>
<td>M</td>
</tr>
<tr>
<td>M</td>
<td>M</td>
<td>H</td>
<td>H</td>
</tr>
<tr>
<td>M</td>
<td>M</td>
<td>M</td>
<td>M</td>
</tr>
<tr>
<td>M</td>
<td>M</td>
<td>L</td>
<td>M</td>
</tr>
<tr>
<td>M</td>
<td>L</td>
<td>H</td>
<td>M</td>
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<tr>
<td>M</td>
<td>L</td>
<td>M</td>
<td>L</td>
</tr>
<tr>
<td>M</td>
<td>L</td>
<td>L</td>
<td>L</td>
</tr>
<tr>
<td>L</td>
<td>H</td>
<td>H</td>
<td>H</td>
</tr>
<tr>
<td>L</td>
<td>H</td>
<td>M</td>
<td>M</td>
</tr>
<tr>
<td>L</td>
<td>H</td>
<td>L</td>
<td>M</td>
</tr>
<tr>
<td>L</td>
<td>M</td>
<td>H</td>
<td>M</td>
</tr>
<tr>
<td>L</td>
<td>M</td>
<td>M</td>
<td>L</td>
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<tr>
<td>L</td>
<td>M</td>
<td>L</td>
<td>L</td>
</tr>
<tr>
<td>L</td>
<td>L</td>
<td>H</td>
<td>L</td>
</tr>
<tr>
<td>L</td>
<td>L</td>
<td>M</td>
<td>L</td>
</tr>
<tr>
<td>L</td>
<td>L</td>
<td>L</td>
<td>L</td>
</tr>
</tbody>
</table>

VH: Very High  L: Low  M: Medium  H: High

IV. Evaluation

ClouSim [32] has been utilized for the simulation and evaluation of the proposed solution. The scalability of this simulation is high and can simulate very big clouds. At the moment, despite the simulation, a cloud is able to simulate the cloud environments composed of some other clouds. Also, in this research, the surveyed data obtained from the CoMon project, which indeed is a monitoring infrastructure for PlanetLab servers, have been used to create the workload [33, 34]. In this scenario, data from checks performed on the physical equipment at intervals of 300 milliseconds have been taken into account. Three separate assessments were conducted using varying quantities of virtual assets on this premise. In order to accomplish this, initial guesses were made that the number of virtual computers in each of the three assessments would be 898, 1033, and 1358. Two optimization techniques, Particle Swarm Optimization [17, 35] and Ant Colony Optimization (ACO) [36, 37], were used to examine the effectiveness of the proposed solution across a range of metrics, including energy usage, runtime, SLA violations, and migrations. The evaluation's findings are depicted in Fig. 6 to 9. The energy we use is depicted in Fig. 6 below. As seen in the figure, the suggested approach significantly reduces energy use. Because one of its primary goals is efficient resource management and it doesn't generate congestion in the processing servers, it was anticipated that the solution would reduce energy consumption. With the aid of the proposed fuzzy selector, the suggested method successfully avoided congestion and, by extension, increased energy usage. Fig. 7 displays the runtime of the solutions over three distinct assessments. It is clear at this point that the processing duties have been completed more quickly, thanks to the offered method. As a result of the RIMA algorithm's careful monitoring, the accurate resources prediction needed, and the best migrations, the answers for all three tests now have shorter execution times. Increased productivity may be the result of the solution's newfound capacity for accurate resource prediction, as well as the suitable migrations made by the use of the fuzzy selector. Fig. 8 also displays the total number of VM migrations performed by each solution, allowing us to examine the migrations made by each solution to achieve load balancing and load reduction. In all three tests, the migration counts of the suggested solutions were much lower than those of the other two. Still, this is not the case for raising and optimizing the other parameters when taking into account execution time and energy usage. It's important to remember that if VM migrations aren't performed optimally and correctly, it might cause a significant overload in the network and waste server resources. Fig 8 shows that while the proposed solution has fewer migrations, those migrations are optimized and done timely, causing minimal additional network overload and avoiding unnecessary engagement of server resources. It also suggests that a large number of migrations do not necessarily indicate the superiority of that solution. The cause of the migrations' optimality is directly related to the usage of fuzzy selectors, in which the server state has been studied as well as carried out in preparation for migration.
Finally, the solutions’ SLA violations are displayed in Fig 9. The Service Level Agreement (SLA) is relied upon as the yardstick by which to measure actual performance. Whether the service quality is satisfactory is determined by the QoS parameters in the SLA. The primary goal of this contract is to set forth the legally binding parameters for the quality, availability, or cost of the services to be supplied. As can be seen in Fig. 9, the proposed solution makes the servers more reliable than the other two solutions, which in turn creates higher accessibility in the cloud infrastructure as well; in other words, this solution’s SLA violation is lower than that of the two other ones. Load balancing, which has been accomplished with the aid of the resource fuzzy selector and predictor, and the VMs migration, which has been accomplished with the help of the fuzzy selector, are used to achieve this.

A. Evaluation of the Cost of Consumption

In order to implement the cost of the proposed solution, we have considered a three-layer cloud program, where each layer has different characteristics from the other layers. Table VII shows the general characteristics of the different layers of the three-layer cloud architecture.

| TABLE VII. SPECIFICATIONS OF DIFFERENT LAYERS OF THE CLOUD PROGRAM |
|-----------------|----------------|----------------|
| Layer           | Avg. Request   | VM Type | Number of Startup VMs |
|                 | Arrival Rate   |         |                    |
| Web             | 100%           | Large   | 5                  |
| Application     | 70%            | Medium  | 5                  |
| Database        | 40%            | Small   | 5                  |

The rate of entering requests in different layers is different. More precisely, the number of requests entered in the first layer is according to what is read from the load, but the number of requests in the second and third layers is, on average equal to 70 and 40% of the requests in the first layer, respectively.

The cost is the sum of the cost of the virtual machines and the penalty cost. That the penalty cost is the total amount of the penalty cost for all customer requests, and the total cost of virtual machines is for setting up and running all virtual machines. The total cost is calculated by equation (5).

\[
\text{Total Cost} = \text{VM Cost} + \text{Penalty Cost} \tag{5}
\]

Fig. 10 show the average cost for different layers. The values displayed in these figures are the average of all three data sets. In Fig. 10(a), the average cost for the application layer is displayed; as can be observed, in this layer, the suggested method can uniformly provide the elasticity of cloud computing compared to other methods. In addition, the proposed method in this research reduces the total cost more than other methods. One should know about health data; due to the huge amount of data and the importance of the subject, the amount of cost is very important. Fig. 10(b) shows the cost in the web layer. Based on the distribution of the workload entered into cloud computing, the compared methods have different behaviours with the cost criterion. Any amount of workload entry has an unbalanced distribution; the complexity of its analysis and decision-making by the examined frameworks also increases. Therefore, it can be seen that in most cases and on average, the proposed method in this research has a better performance in the application layer. Fig. 10(c) examines the cost metric in the database layer for the proposed methods. This layer faces a large number of parallel transactions, so the use of dynamic methods that provide elasticity for this layer is one of the requirements that the service provider should pay great attention to. Considering the three layers of the proposed architecture for big data in the field of health, it is necessary to examine the cost criteria in each of the three layers.

According to Fig. 10(a), (b), and (c), it can be seen that the proposed method imposes a lower average cost on the supplier.
B. Quantification of Computational Complexity

The high number of rejected requests will reduce the profit, and on the other hand, it may also increase the cost. Therefore, the proposed method should be able to reject fewer requests. In this way, it increases the profit and keeps the quality of the services within the agreed limits [38]. In this paper, the number of rejected requests is counted to quantify the computational complexity based on the timeline. According to Fig. 11 and considering the number of user requests based on the timetable, as can be seen, for example, at time point 49, the number of rejected requests for the ACO algorithm equals 145, and for the PSO algorithm, it equals has been 240. In contrast, at this point in time, the number of rejected requests for the proposed method was around 50. Therefore, according to the calculation quantity, it can be concluded that the proposed method can have a higher efficiency than other methods.

C. Estimating the amount of Rejected Requests

The high number of rejected requests will reduce the profit, and on the other hand, it may also increase the cost. Therefore, the proposed method should be able to reject fewer requests. In this way, it increases the profit and keeps the quality of the services within the agreed limits. Fig. 12 shows the rejected requests in the application layer for the test data set. As can be seen, the proposed method in this research has a better performance than other methods.

Fig. 13 also shows the rejected requests in the web layer. Fig. 14 also shows the number of rejected requests in the database layer. From these graphs, it can be seen that the proposed method in this research has a better performance than the compared methods.

Table VIII shows that, in accordance with the behavior of the graphs, the average length of the schedule has dropped by 5.36% of the minimum execution time for the number of 300 jobs and by 6.14% when compared to the PSO method. Similarly, Table IX shows that with 600 tasks, the ratio of successful execution increases by 6.81% for the ACO schedule and by 3.92% for the lowest execution time.

![Average cost comparison for the compared methods.](image)

![Computational complexity comparison.](image)
Fig. 12. Comparison of the investigated methods in the number of rejected requests in the application layer.

Fig. 13. Comparison of the investigated methods in the number of rejected requests in the web application layer.

Fig. 14. Comparison of the investigated methods in the number of rejected requests in the database application layer.
TABLE VIII. THE PROPOSED METHOD COMPARED TO SIMILAR METHODS FOR AVERAGE PROGRAM LENGTH

<table>
<thead>
<tr>
<th>Iteration number</th>
<th>Number of tasks</th>
<th>PSO</th>
<th>ACO</th>
<th>Proposed Method</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>100</td>
<td>343</td>
<td>336</td>
<td>323</td>
</tr>
<tr>
<td>2</td>
<td>200</td>
<td>1025</td>
<td>1007</td>
<td>985</td>
</tr>
<tr>
<td>3</td>
<td>300</td>
<td>1765</td>
<td>1729</td>
<td>1742</td>
</tr>
<tr>
<td>4</td>
<td>400</td>
<td>2516</td>
<td>2369</td>
<td>2236</td>
</tr>
<tr>
<td>5</td>
<td>500</td>
<td>3129</td>
<td>3102</td>
<td>3024</td>
</tr>
<tr>
<td>6</td>
<td>600</td>
<td>3024</td>
<td>3035</td>
<td>2954</td>
</tr>
</tbody>
</table>

TABLE IX. THE PROPOSED METHOD COMPARED WITH SIMILAR METHODS FOR THE RATIO OF SUCCESSFUL EXECUTION

<table>
<thead>
<tr>
<th>Iteration number</th>
<th>Number of tasks</th>
<th>PSO</th>
<th>ACO</th>
<th>Proposed Method</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>100</td>
<td>0.48</td>
<td>0.50</td>
<td>0.51</td>
</tr>
<tr>
<td>2</td>
<td>200</td>
<td>0.68</td>
<td>0.70</td>
<td>0.73</td>
</tr>
<tr>
<td>3</td>
<td>300</td>
<td>0.75</td>
<td>0.77</td>
<td>0.78</td>
</tr>
<tr>
<td>4</td>
<td>400</td>
<td>0.79</td>
<td>0.79</td>
<td>0.80</td>
</tr>
<tr>
<td>5</td>
<td>500</td>
<td>0.77</td>
<td>0.79</td>
<td>0.80</td>
</tr>
<tr>
<td>6</td>
<td>600</td>
<td>0.79</td>
<td>0.80</td>
<td>0.79</td>
</tr>
</tbody>
</table>

D. Real Time Case Study based Discussion

Existing processes for collecting and processing patients’ vital information require a large amount of labor and calculations. These processes are usually error-prone and have large delays that prevent correct information from being available in real time. Cloud computing as a new technology, with internet infrastructure and new solutions, has brought significant advantages in providing medical services electronically. Along with that, through the rapid development of processing and storage technologies as well as the success of the Internet, computing resources have become cheaper, stronger and more accessible than before, and government organizations have started using cloud architecture, platforms and programs to provide services and meet the needs of their subordinates. But in between, there are many challenges, the most important of which is the optimal resource management and how much processing resources each data needs.

V. CONCLUSION

Each patient’s data is added to hospital systems in such large quantities that they become the metadata that must be processed by such a large number of facilities, which practically results in high costs to the hospital, additionally to execution times as well as accuracy issues. Because of this, the use of cloud computing's processing capacity for metadata has become more widespread. Accurate resource provision is a crucial element of cloud computing. User satisfaction rises proportionally as the accuracy of the resources grows, and the number of violations of the services decreases. In order to handle healthcare metadata in a cloud computing environment, a better resource management approach has been provided in the current research. It was assumed that the proposed architecture would have each IOTDC have a Broker, who would first receive and analyze requests from users referred to that IOTDC before executing the algorithm and returning the result to the users of that IOTDC if the resources required to execute those requests were available. However, the DEMATEL fuzzy algorithm should be used by the Broker to select the appropriate IOTDC from the available associated IOTDCs if it is unable to fulfil user requests using the available IOTDC resources. The proposed solution was then tested and evaluated using the PSO and ACO algorithms based on the various factors, including SLA and execution time, developed and simulated in the CloudSim program. The findings indicate that the solution performed better in all tests run. Among the things that can be done in the future in the continuation of the research are: combining time series to predict resources optimally, combining the reinforcement learning method with fuzzy logic for automatic scaling, and using the combination of reinforcement learning and neural network in methods based on service level agreements, which can significantly improve the proposed solution.

FUNDING

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REFERENCES

Open Information Extraction Methodology for a New Curated Biomedical Literature Dataset

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Abstract—The research articles contain a wealth of information about the interactions between biomedical entities. However, manual relation extraction processing from the literature by domain experts takes a lot of time and money. In addition, it is often prohibitively expensive and labor-intensive, especially in biomedicine where domain knowledge is required. For this reason, computer strategies that can use unlabeled data to lessen the load of manual annotation are of great relevance in biomedical relation extraction. The present study solves relation extraction tasks in a completely unsupervised scenario. This article presents an unsupervised model for relation extraction between medical entities from PubMed abstracts, after filtration and preprocessing the abstracts. The verbs and relationship types are embedded in a vector space, and each verb is mapped to the relation type with the highest similarity score. The model achieves competitive performance compared to supervised systems on the evaluation using ChemProt and DDI datasets, with an F1-score of 85.8 and 88.5 respectively. These improved results demonstrate the effectiveness of extracting relations without the need for manual annotation or human intervention.

Keywords—Relation extraction; BERT; open information extraction; biomedical literature; ChemProt; DDI

I. INTRODUCTION

With more than 32 million citations of biomedical literature in PubMed [1], the medical field is facing a substantial expansion of data, posing challenges for biomedical researchers efficiently and automatically to extract information about specific biomedical entities such as genes, proteins, and diseases. Gathering labeled data required for training and creating models for natural language processing (NLP) is a time-consuming task because it requires human annotation and can take a lot of effort to complete. Gathering a sufficient amount of labeled data for NLP can be challenging since language is complex and diverse. Models require large amounts of labeled data to learn from to generalize well, and the larger the dataset, the more challenging it becomes to label it. Also, the quality of the labeled data is crucial for the accuracy of the NLP model. To ensure high-quality labeled data, it is necessary to have multiple annotators to check for consistency, which can increase the cost of the labeling process. Overall, gathering labeled data for NLP is a time-consuming and resource-intensive process, which can make it expensive and difficult. There is a necessity for quick and scalable discovery, extraction, and organization of the information contained within this data; an issue that raises the need to use information extraction techniques. However, the process of information extraction is one of the primary difficulties in NLP, and its application to a vast amount of data has some restrictions.

Relation extraction and open information extraction are both techniques used in information extraction, but they have distinct differences in their approaches and goals. Relation extraction focuses on identifying and extracting specific predefined relationships or connections between entities mentioned in the text. It aims to discover structured relationships with predefined categories. The output of relation extraction is typically a set of entity pairs with their associated relationship type.

Open Information Extraction (OIE) is a relation-free, open-domain paradigm that enables unsupervised information extraction. Its major goal is to create a triple relation with the elements <entity1> <Relation> <entity 2> from unstructured data without having to predefine the relationship between the two entities. The extracted tuples can be binary, ternary, or n-ary, depending on how many entities are involved in the relationship. In the biomedical domain, OpenIE has been used to extract relationships between entities such as genes, proteins, diseases, and drugs from the scientific literature. This method may result in scalable and fast performance [2]. In contrast to OIE, RE demands relation definition prior to extraction. Similar to OIE, a binary relation or a higher-order relation (n-ary) can be found in the extracted relation [3]. Fig. 1 summarizes the difference between OIE and RE.

![Fig. 1. Open information extraction vs. relation extraction.](image)

Relation extraction is often used to populate or update a structured knowledge base with specific relationship types. The extracted relationships are typically mapped to existing entities in the knowledge base, providing structured information for further analysis.

While Open information extraction is useful for discovering new relationships or facts that may not be present in a pre-existing knowledge base, the extracted relational tuples can be integrated into a knowledge base to expand its coverage and provide additional information. In summary, relation...
extraction focuses on extracting specific pre-defined relationships between entities, while open information extraction aims to extract as much information as possible without relying on predefined relationship types. Relation extraction is often supervised and used for structured knowledge base populations, whereas open information extraction is more flexible, unsupervised, and useful for discovering new information.

OIE is an essential NLP task, and because of its many potential uses in information retrieval, information extraction, text summarization, and question answering, it was chosen as a source task to transfer to other NLP tasks [2]. Even though several OIE algorithms have been created in the last ten years, only a few studies [3, 4] have attempted to address Unsupervised Relation Extraction (URE) using machine learning (ML) and deep learning methods. Due to the shortage of labeled data, researchers have recently been more and more interested in model generalization in deep learning. Fig. 2 shows the number of publications in Biomedical literature in the last years. It is shown that there is a steady increase in the number of publications in biomedical literature according to PubMed and Web of Science databases.

Fig. 2. Number of publications in biomedical literature.

Fig. 3. Number of publications in biomedical literature using OpenIE.

Fig. 3 depicts a few articles that use OpenIE method. That’s why there is a growing need for unsupervised models in biomedical literature and manual annotation of all relevant articles is becoming more and more difficult and expensive. Additionally, supervised models may not generalize well to new domains and contexts, which limits their utility in real-world applications. OpenIE is particularly useful for extracting relations and facts from large and complex biomedical literature, which can provide valuable insights into the interactions between biological entities and the underlying mechanisms of various diseases [7]. This literature review aims to provide an overview of recent studies on the use of biomedical relation extraction, highlighting the state-of-the-art methodologies and potential future directions for research in this field.

Drug descriptions from Wikipedia and DrugBank were used by Zhu et al. [8] to provide semantic data about drug entities to the BERT model. To obtain sentence representation with entity information, mutual drug entity information, and drug entity information, they used three different types of entity-aware attentions. The BioBERT embeddings of two medications were subtracted to get the mutual information vector of the two drug entities. On the DDI corpus, they reported an 80.9 (micro F1-score).

By linking one or more natural language questions to each relation, Levy et al. [9] have reframed the relation extraction job as a reading comprehension issue. In a zero-shot context, this method enables generalization of undiscovered relations. You [10] proposed a Path-Based MiRNA-Disease Association (PBMDA) prediction model that can infer potential miRNA-
disease associations by combining known human miRNA-disease associations, miRNA functional similarity, disease semantic similarity, and Gaussian interaction profile kernel similarity for miRNAs and diseases.

Similarly, in a study by Zaikis et al. [11], the authors proposed an approach that integrates a prior knowledge by using BioBERT which outperforms existing state-of-the-art methods on the DDI corpus in both drug named entity recognition and overall DDI extraction task. Gu et al. [12] proposed a Convolutional Neural Network (CNN) model to learn a more robust relation representation based on both word sequences and dependency paths for the CID relation extraction task, which could naturally characterize the relations between chemical and disease entities.

Drug-drug interactions (DDIs) prediction and extraction model based on BioBERT was proposed by Mondal [13] and Zhu [14]. Both experiments on the DDIExtraction 2013 corpus demonstrate that it can outperform baseline architectures in F1-score, which is a good illustration of BioBERT's use in the field of biomedical text processing.

Peng et al. [15], used a distant supervision model to extract extraction of CIDs from biomedical literature. The authors proposed a method that combined novel statistical features with machine learning model to improve the accuracy of CID extraction. The results showed that their method achieved high precision and recall in identifying chemical-induced disease (CID) relations.

Chen [16] provided a computational approach to identify potential miRNA-disease connections that significantly cut down on experimental time and expense. They created the WBSMDA model for within and between Score for MiRNADisease Association prediction to predict miRNAs linked with diseases. Mario [17], two new relation extraction methods that were proposed to extract the mutation-disease relationship outperform conventional techniques.

In, Zhang et al. [18] also proposed a hybrid model based on RNNs and CNNs to classify PPIs and DDIs. The inputs of the hybrid model are sentence sequences and SDPs generated from the dependency graph. RNNs and CNNs models were employed to learn the feature representation from sentence sequences and SDPs, respectively.

Although the studies discussed above have made significant contributions to biomedical relation extraction, there are still limitations to consider. For instance, some studies rely on specific datasets, making them less generalizable to other domains or contexts. For example, the studies by Mondal [13] and Zhu [14] focused solely on drug-drug interactions and may not be applicable to other types of biomedical relations. Additionally, some studies may require significant domain-specific knowledge and preprocessing, such as the study by Peng et al. [15], which utilized distant supervision and statistical features. While the results of these studies are promising, the performance of the proposed models may be limited by the quality and representativeness of the training data, which can introduce biases and limit the generalizability of the models.

In summary, while most high-performing biomedical relation extraction systems to date are based on supervised approaches, this method requires large amounts of labeled data that can be expensive and time-consuming to obtain. Furthermore, supervised approaches may be biased and limited in generalizability to new domains or contexts. Alternatively, unsupervised models can learn directly from data and can be applied to new domains and contexts without the need for retraining on newly labeled data. The exponential increase in biomedical literature and the limitations of supervised models highlight the growing need for unsupervised models in biomedical relation extraction.

III. METHODOLOGY

By using this approach, the study aims to overcome the limitations of supervised methods and provide a scalable and efficient solution for relation extraction in the biomedical domain. The end-to-end framework described in Fig. 4 involves the following steps:

- Downloading and preprocessing records from PubMed using specific queries (each document contained the title and the abstract, PMID).
- Converting the data into sentence pair format for the Named Entity Recognition (NER) process.
- The medical entities such as genes, proteins, diseases, and drugs are captured in NER and replaced with predetermined tags.
- To extract the relations of the form entity1-verb-entity2, a part-of-speech (POS) tagger is used. By using the POS tags, the model can identify the verbs that connect the entities.
- Creating a list of relationship types that are of interest in the task at hand to map the extracted verbs to them. These relations would then be mapped to the verbs that are extracted using a part-of-speech tagger. Once the set of relations has been defined, the extracted verbs and relationship types are embedded in a vector space and each verb is mapped to the relation type that has the highest score after calculating the similarity score.
- The resulting data from the previous steps are used to finetune the pre-trained BERT model.

A. Material

The most significant databases for biomedical literature include MEDLINE, PubMed, Scopus, Embase, and Web of Science. The model at hand gathers literature primarily from the PubMed database. The PubMed database contains more than 35 million citations and abstracts from biomedical literature. In order to improve both individual and planetary health, PubMed promotes the free search and retrieval of biomedical and life sciences literature.
While journal articles are not included in full text, PubMed [1] provides links to the full texts when they are available from other sources, such as the publisher's website or PubMed Central (PMC). Each article in PubMed is assigned a unique identification code called a PMID. PMIDs are never reused and do not change over time or throughout processing, providing a valuable resource for researchers in the biomedical field, providing access to a vast amount of literature that can inform scientific discoveries, and advance healthcare. The online availability of PubMed to the public has also made it a valuable resource for patients and healthcare professionals seeking information on medical conditions and treatments [19].

B. Articles Selection and Generation of Training Data

Data must be carefully selected to support academic requirements, scientific research, and other purposes. Data curation is the process of locating, organizing, and managing data collections so that consumers looking for information can use them. Choosing an accurate corpus surely will have a positive impact on the study. The framework proposed in [19] for title retrieval can be expanded to extract abstracts as well. This involves using similar techniques to those used for title extraction but applied to the abstracts. A PubMed search using the terms "drug and drug" and "chemical and protein" was used as the starting point for data collections so that consumers looking for information can use them. Choosing an accurate corpus surely will have a positive impact on the study. The framework proposed in [19] for title retrieval can be expanded to extract abstracts as well. This involves using similar techniques to those used for title extraction but applied to the abstracts. A PubMed search using the terms "drug and drug" and "chemical and protein" was used as the starting point for the process. The two search queries resulted in 4750 abstracts. A sample of selected publications' titles, publication dates, and PMIDs are displayed in Table I.

<table>
<thead>
<tr>
<th>PMID</th>
<th>TITLE</th>
<th>Date of Publication</th>
</tr>
</thead>
<tbody>
<tr>
<td>37311160</td>
<td>Breast Cancer Therapeutics and Hippo Signaling Pathway: Novel MicroRNA-Gene-Protein Interaction Networks.</td>
<td>2023</td>
</tr>
<tr>
<td>36387483</td>
<td>Assessment of potential drug-drug interactions among outpatients in a tertiary care hospital: focusing on the role of P-glycoprotein and CYP3A4 (retrospective observational study).</td>
<td>2022</td>
</tr>
<tr>
<td>34338261</td>
<td>Advances in chemical probing of protein O-GlcNAc glycosylation: structural role and molecular mechanisms.</td>
<td>2021</td>
</tr>
<tr>
<td>29927582</td>
<td>Chemical and Biochemical Perspectives of Protein Lysine Methylation.</td>
<td>2019</td>
</tr>
</tbody>
</table>

The filtration stage included removing all articles that are published before 2019, removing articles that don't have an abstract, removing duplicated articles (the same article appears in different queries) using the pmid. After applying these filters, the resulting dataset contained 1052 abstracts. The abstracts from the two search queries were merged to form one dataset. Fig. 5 summarizes the main steps of articles selection.
extent of the preprocessing conducted in this phase. All the previous steps were done using the Scispacy library [20], which is tailored for biomedical, scientific, and clinical material and Natural Language Toolkit (NLTK) [21], which provides a range of efficient natural language algorithms.

The next step after preprocessing was Name Entity Recognition (NER) for extracting the medical entities. We utilized Scispacy to identify and extract the biomedical entities from the extracted phrases. Following NER, we filtered out the sentences from the abstracts. Sentences that did not contain any entities or contained only one entity were removed, while sentences with two entities were considered the target sentences for performing relation extraction and training the model. The total number of sentences with two entities or more was 3,407 sentences. A total of 27,123 text descriptions are obtained by the proposed model which are labeled as gene, 19,471 are labeled as disease, 30,289 are labeled as chemical and 25,013 are labeled as protein from the NER task.

2) Relation extraction: A two-step algorithm was used to create a text classification model for the purpose of identifying Chemical–protein or drug-drug-related relations. The first step was part of speech tagging used to extract relations in the form of subject-verb-object. The second step was the resulting verbs tagged from the POS stage and the relation types that we are interested in learning are embedded in a vector space, and each verb is mapped to the relation type that it is most comparable to. To make this procedure fully automatic, it is necessary to specify the collection of relation types of interest and set a threshold for verb mapping, below which no relation class is assigned. Here’s the process followed:

a) Part of speech tagging: Initially, Part of Speech (POS) tagging was employed to identify all the verbs present in the sentences not only the main verb between the medical entities. The POS tagger is a tool that assigns a part of speech (such as a noun, verb, adjective, or adverb) to each word in a sentence. By using the POS tags, the model can identify the verbs that connect the entities.

b) Similarity scoring: This step involves identifying the verbs in the sentence that are likely to indicate a relationship between the named entities of interest, such as genes, proteins, drugs, and diseases. The following verbs—increase, decrease, cause, treat, prevent, combine, reduce, and bind—were used to create a fixed set of relations. Once the set of relations has been defined, the extracted verbs and relationship types are embedded in a vector space. This involves representing the verbs and relationship types as vectors in a high-dimensional space, where the similarity between the vectors reflects the semantic similarity between the verbs and relationship types. Each verb is then mapped to the relation type that has the highest score after calculating the similarity score. This involves calculating the cosine similarity between the vector representation of the verb and the vector representation of each relation type and selecting the relation type that has the highest similarity score.

Table II shows a sample of extracted verbs, and their mapping to the set of relations. The similarity scoring threshold was set at 0.4. In cases where multiple verbs exceeded the threshold, the verb with the highest similarity score was selected. Verbs that achieved a similarity score below 0.4 were mapped to a “no relation” label. The RE step is crucial because it guarantees that the resulting relations will have high precision (although at the expense of low recall), as they substantially rule out the chance of the two entities randomly co-occurring in the sentence through the subject-verb-object relationship. In other words, we arrive at a limited, but highly valuable collection of relations that can be applied in a manner that is similar to distant supervision.

3) Transformer tuning of self-created dataset: Language models have demonstrated improved performance in various NLP tasks by considering contextual information when representing features. Popular language models like ELMO [22], BERT [5], and ULM Fit [23] are commonly used in NLP tasks.

a) Bidirectional Transformers (BERT): Among them, BERT introduced by Google in 2019, has gained immense popularity for its ability to pre-train deep bidirectional representations on unlabeled text. By considering context from both sides in all 12 transformer layers, BERT has shown enhanced performance in many NLP applications, including relation extraction.

BERT was used to train the RE classifier, it was fine-tuned for up to 5 epochs because early experiments revealed that the model began overfitting to noise and that the validation loss increased beyond that. The typical BERT pre-processing method for relation extraction is used to replace the entity names in the phrase with predetermined tags so that the BERT model can recognize the entities in the sentence. All references to proteins, drugs, and chemicals are expressly changed to @PROTEINS, @DRUGS, and @CHEMICALS, respectively. Fig. 6 shows an example of sentences after preprocessing for BERT.

Table II. Examples of Verb Mapping and Calculating Similarity

<table>
<thead>
<tr>
<th>Verb</th>
<th>Relation</th>
<th>Similarity</th>
</tr>
</thead>
<tbody>
<tr>
<td>[‘Reduced’]</td>
<td>Decrease</td>
<td>0.553579</td>
</tr>
<tr>
<td>[‘recovered’]</td>
<td>treat</td>
<td>0.535033</td>
</tr>
<tr>
<td>[‘entail’, ‘including’, ‘increased’, ‘eliminated’]</td>
<td>increase</td>
<td>0.68346</td>
</tr>
<tr>
<td>[‘used’, ‘prevent’]</td>
<td>Prevent</td>
<td>1</td>
</tr>
<tr>
<td>[‘impaired’, ‘blocked’, ‘leading’]</td>
<td>cause</td>
<td>0.565928</td>
</tr>
<tr>
<td>[‘regulate’, ‘serve’]</td>
<td>Bind</td>
<td>0.46995</td>
</tr>
<tr>
<td>[‘prepared’]</td>
<td>No relation</td>
<td>0.166231</td>
</tr>
</tbody>
</table>

Table II shows a sample of extracted verbs, and their mapping to the set of relations. The similarity scoring threshold was set at 0.4. In cases where multiple verbs exceeded the threshold, the verb with the highest similarity score was selected. Verbs that achieved a similarity score below 0.4 were mapped to a “no relation” label. The RE step is crucial because it guarantees that the resulting relations will have high precision (although at the expense of low recall), as they substantially rule out the chance of the two entities randomly co-occurring in the sentence through the subject-verb-object relationship. In other words, we arrive at a limited, but highly valuable collection of relations that can be applied in a manner that is similar to distant supervision.

Fig. 6. Examples of sentences preprocessing before feeding them to BERT.

Sentence examples:

- @DRUGS decreases the elimination of @DRUGS causing an increase in overall exposure.
- Anticoagulants @DRUGS may increase sensitivity to oral @DRUGS.
- @CHEMICALS potentially attenuated gene expression in inflammation, such as INOS, CXCL2 and @GENES. They suggest that TRPML1 works in concert with @CHEMICALS to regulate @GENES translocation between the cytoplasm and lysosomes.

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BERT uses WordPiece embedding [24] to address the out-of-vocabulary issue, and each input word will be divided into subwords from the WordPiece vocabulary. Additionally, we add the segment and position information of the tokens in the input phrase using segment embedding and position embedding, both of which follow the original BERT input structure.

When using BERT, it is necessary to select a suitable model and adjust hyperparameters such as learning rate, batch size, and the number of epochs. The BERT-base uncased model is a common choice for sequence classification, it was found that the uncased version performed better [25,26,27,28]. During training, a trained model is validated to assess overfitting and estimate performance before applying it to the target application. We found that validating specific combinations of claim and description concatenation parts, which are not specifically trained but used in other combinations, has low significance.

b) Hyperparameter settings: Hyperparameter tuning is an important step in training machine learning models to achieve optimal performance. In this study, the Transformers library was used to perform hyperparameter tuning for the BERT-Base-Uncased model used in biomedical relation extraction.

The experiments involved varying different hyperparameters, including batch size, learning rate, maximum sequence length, and number of epochs. The maximum sequence length varied between 128 and 512, the batch size varied between 4 and 16 and the learning rate varied between 1e-5 and 4e-5. The results showed that the model achieved minimum loss when the learning rate was around 1e^-5. Fig. 7 shows the relationship between loss and the learning rate. The maximum success value was reached with a batch size fixed at 8, and a maximum sequence length equal to 256. Table III represents the ideal parameter values after experimenting with various parameter value combinations.

Using the Adam optimizer, the BERT-Base-Uncased model was improved. The best-performing models are saved and evaluated on the test set using the highest Recall score (lowest Type II error) on the validation set. The model is designed to prioritize the maximization of the Recall score due to the belief that it is more crucial to minimize the Type II error rather than just increasing the Accuracy/F1 score of the model.

IV. RESULTS AND DISCUSSION

The findings of this study demonstrate that the proposed approach can effectively identify relations between medical entities without the need for supervision or manual intervention. The proposed method was applied to a collection of abstracts obtained from PubMed, and the results showed that the proposed unsupervised approach outperformed the previous supervised methods in terms of overall precision and F1-score on both the ChemProt and DDI datasets.

Theoretical results obtained from relation extraction can be used to develop more accurate and efficient algorithms that can automatically extract relations between biomedical entities from large-scale literature data. This has significant implications for drug discovery, disease diagnosis, and treatment, as it can facilitate the identification of new drug targets and drug-drug interactions, which can lead to the development of new treatments and cures for complex diseases. Additionally, the identification of disease biomarkers can aid in the early detection and diagnosis of diseases and provide insights into disease mechanisms, leading to more effective treatments and improved patient outcomes. Ultimately, the practical use of theoretical results obtained from relation extraction can have a profound impact on the field of biomedical research and accelerate the pace of scientific discovery, leading to new insights and discoveries that can improve human health and well-being. The pipeline can be used to extract relationships between biomedical entities in a variety of scenarios, facilitating the discovery of new relationships and insights in the biomedical domain.

A. Datasets

For training, the proposed model utilized a self-created dataset to train the BERT model. For testing and evaluation, it employed two widely recognized benchmark datasets in the biomedical domain: the DDI [29] dataset and the ChemProt [30] dataset, both of which involve multiclass classifications. Table IV presents the statistics of the DDI and ChemProt datasets.

B. Results on Benchmark Datasets

The performance of the proposed unsupervised approach for relation extraction from biomedical literature was evaluated on two well-known benchmark datasets, ChemProt and DDI. The results of the evaluation are presented in Table V, and the proposed approach was compared to a BERT model that underwent fine-tuning using supervised data manually annotated for the two datasets.

The results demonstrate that the proposed approach outperforms the most advanced model on the CHEMPROT dataset, achieving an F1-score of 85.8 compared to 75.14 for
the supervised approach. On the DDI dataset, the proposed approach achieves an F1-score of 88.5, compared to 70.2 for the supervised approach. Furthermore, the proposed unsupervised approach achieves a Recall score of 87.5 on the ChemProt dataset, compared to 75.09 for the supervised approach and a recall of 93.5 vs. 73.4 on DDI dataset, indicating that the proposed approach is more effective at correctly identifying relevant relationships. Fig. 8 provides a comparison of the results on the two datasets, highlighting the competitive performance of the proposed unsupervised approach.

### TABLE IV. THE STATISTICS OF THE DDI AND THE CHEMPROT DATASETS

<table>
<thead>
<tr>
<th>Dataset</th>
<th>Abstract</th>
<th>Positive relations</th>
<th>Negative relations</th>
</tr>
</thead>
<tbody>
<tr>
<td>DDI</td>
<td>191</td>
<td>979</td>
<td>4737</td>
</tr>
<tr>
<td>ChemProt</td>
<td>800</td>
<td>3458</td>
<td>10,540</td>
</tr>
</tbody>
</table>

### TABLE V. COMPARISON WITH EXISTING MODELS ON CHEMPROT AND DDI DATASETS

<table>
<thead>
<tr>
<th>Dataset</th>
<th>Method</th>
<th>Precision</th>
<th>Recall</th>
<th>F1-Score</th>
</tr>
</thead>
<tbody>
<tr>
<td>ChemProt</td>
<td>Proposed method</td>
<td>83.15</td>
<td>87.5</td>
<td>85.8</td>
</tr>
<tr>
<td>[31]</td>
<td></td>
<td>75.20</td>
<td>75.09</td>
<td>75.14</td>
</tr>
<tr>
<td>DDI</td>
<td>Proposed method</td>
<td>84.2</td>
<td>93.5</td>
<td>88.5</td>
</tr>
<tr>
<td>[32]</td>
<td></td>
<td>-</td>
<td>73.4</td>
<td>70.2</td>
</tr>
</tbody>
</table>

The results demonstrate that the proposed unsupervised approach for relation extraction from biomedical literature can strongly compete with the most advanced fully supervised systems, providing a scalable and efficient solution for relation extraction from biomedical literature. The success of the proposed approach in achieving performance equivalent to a fully supervised model highlights its potential to facilitate the discovery of new relationships and insights in the biomedical domain.

### C. Results on Self-created Dataset

By analyzing Fig. 9, it can be noted that the model succeeded in its main goal to extract specific targeted biomedical relations from sentences and that the most frequent relations were cause and treat. The number of tagged genes, chemicals, proteins, and diseases in the abstracts were 27123, 30289, 25013 and 19471 respectively, which indicates the complexity and richness of the biomedical text, which can be challenging to process and extract information from without automated methods.

The number of relations, as shown in Fig. 9, extracted by the model, mapped to ‘cause’ and ‘treat’ are significant, indicating the importance of these relationships in biomedical research. The numbers of relations extracted for increase, decrease, bind, combine and reduce demonstrate the model's ability to identify a variety of relationships beyond simple causation and treatment.

The number of sentences that have no relation is also an expected outcome, as not all sentences in the biomedical text contain explicit relationships between entities. However, it is still important to accurately identify these sentences to avoid false positives in downstream analyses. Overall, the successful extraction of targeted biomedical relations from the abstracts demonstrates the potential of automated methods for relation extraction in biomedical text mining. These methods can help researchers efficiently process and extract information from the vast amount of biomedical literature available, accelerating scientific discoveries and advancing healthcare.

### V. CONCLUSION

The proposed pipeline based on the state-of-the-art BERT model offers a promising solution to the challenge of relation extraction from biomedical literature, providing a scalable and efficient approach that eliminates the need for human intervention or manual curation and which would reduce manpower and time consumption and automatically extract biomedical entities association data sets from large-scale literature data. The pipeline first retrieves articles (mainly abstracts) from PubMed according to specific queries and the extracted abstracts are then preprocessed to precisely extract the relation between medical entities. The preprocessed data are first presented in the form of subject-verb-object using part of speech tagger, with the resulting verbs and relationship types of interest embedded in a vector space. Each verb is mapped to the relation type that has the highest score after calculating the similarity score. The generated data set is used to fine-tune a BERT model to carry out relation extraction.

The significance of this study lies in its ability to reduce manpower and time consumption associated with relation extraction, making it more feasible and cost-effective. The use of the BERT model, which has demonstrated state-of-the-art performance in natural language processing tasks, provides a promising solution to the challenge of relation extraction from biomedical literature. The empirical comparison conducted in this study validates the effectiveness of the approach, demonstrating superior results compared to previous works that relied on supervised learning. Our unsupervised approach outperformed the supervised approach found in the literature in...
the overall precision and F1-score on both the ChemProt and DDI datasets. In the ChemProt dataset, our method achieved a precision of 83.15 and an F1-score of 85.8, while the supervised method achieved only 75.2 and 75.14, respectively. Furthermore, our method achieved an F1-score of 88.5, surpassing the previous supervised methods scoring 70.2 in the DDI dataset.

The findings of this study have significant implications for future research and applications in this area, such as its potential scalability to larger datasets and its potential for automated extraction of biomedical relations from a wide range of scientific literature. This pipeline can be used to extract relationships between biomedical entities in a variety of scenarios, facilitating the discovery of new relationships and insights in the biomedical domain. By automating the extraction of biomedical relations from a wide range of scientific literature, researchers will be able to identify new relationships and insights that were previously hidden or difficult to discover. This will enable researchers to develop new hypotheses and accelerate the discovery of new treatments and cures for complex diseases and by automating the extraction of biomedical relations, we can accelerate the pace of scientific discovery and improve our understanding of complex diseases. The approach described in the paper uses Open Information Extraction (OIE) techniques to identify interaction words, this method can be applied to other domains and types of data that have a similar structure, such as news articles, social media posts, and legal documents, among others. The techniques used could potentially be adapted and applied to other domains and types of data with similar structures. However, the effectiveness of the approach may depend on the specific characteristics of the data being used, and further research is needed to fully understand its applicability and limitations in different contexts.

REFERENCES


Detection and Investigation Model for the Hard Disk Drive Attacks using FTK Imager

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Abstract—A computer hard disk drive (HDD) is a device that stores, organizes, and manages computer data. In general, it is used for system storage, in which the computer maintains its operating system and other programs. A hard disk drive can, however, be physically damaged as well as affected by software errors, data corruption, and viruses that are used by attackers to cause damage. This study aims to develop a detection and investigation model (DIM) for HDD to detect and investigate HDD attacks using the FTK Imager forensic tool. The design science method is adapted to develop and evaluate the DIM. The developed DIM consists of three main phases: detection, gathering, and analysis. In order to evaluate the capabilities of the developed DIM for HDD, a real scenario was used. According to the results, the DIM can detect and investigate the HDD easily using FTK Imager. Thus, organizations can use the developed DIM to detect, investigate, mitigate, or avoid HDD threats.

Keywords—HDD; cybercrimes; design science method; digital forensic tools; FTK imager

I. INTRODUCTION

Nowadays, cybercrime is becoming more prevalent, resulting in an adverse impact on the availability, confidentiality, and integrity of data stored on hard drives and in the cloud [1]. It is also notoriously known as one of the major threats for organization since they are obligated to spend fortune to protect their data from this. Companies must invest in sophisticated security systems, such as firewalls, anti-virus software, authentication, and encryption, to reduce their vulnerability to cybercrime [2]. In addition, organizations are equipping their employees with knowledge and training on cybersecurity best practices such as, enforcing the use of strong passwords and securing confidential data. Besides, organizations added a top notch security on their network to monitor against any malicious activity [3]. All these measures contribute to the reduction of cybercrime risks and the protection of data stored on hard drives and in the cloud.

HDDs are the most crucial component in a computer system [4], [5]. HDDs store both physical and logical data, which is the primary objective of an adversary. These devices are frequently used to store personal data such as photographs, music, documents, and applications stored on a computer [6]. Usually, the attackers aim for the HDDs to gain access to the data stored on them. By obtaining physical access to the system, they may be able to access the HDD data. In addition, they may attempt to extract data from the HDD using malicious software, such as spyware [7], [8].

Therefore, this study aims to develop a model for detecting and investigating HDD attacks using the FTK Imager software. For this purpose, the design science methodology is employed. The design science method is a research methodology that concentrates on the creation and evaluation of artefacts for the resolution of real-world issues. The developed detection and investigation model is comprised of three main phases: detection, collection, and analysis. Each phase of the investigation includes a series of activities designed to aid the investigator in detecting and investigating attacks on hard disc drives (HDDs).

Digital forensics software package FTK (Forensic Toolkit) is used to perform forensic analysis and digital investigations on computers and mobile devices [9]. It is used by law enforcement, the military, and business inspectors to examine computer activity. It can locate and analyse data associated with Internet activity, deleted files, emails, documents, and images, among other types of information. It includes data recovery, analysis, and reporting utilities. There are several contributions of this study, including improving HDD security by detecting and investigating potential threats before an incident occurs. It also enhances HDD's ability to identify and investigate potential threats more rapidly and the ability of organizations to respond rapidly and swiftly in the event of any possible attacks. Detection and investigation models can help companies educate themselves about potential threats.

The paper is organized as follows: Section II introduces the related works, and Section III introduces the methodology. Section IV introduces the results and discussion. Section V offers a conclusion and further work.

II. RELATED WORK

Many forensic models, tools, strategies, processes, procedures, policies, models, and mechanisms have been proposed to assist with the detection and investigation of HDD attacks. These include data extraction, forensic file system analysis, low-level analysis, data recovery, and forensic imaging, among others. For example, a process model was proposed by the authors of [10] based on the following requirements: the model must be based on existing theory in physical crime investigations; the model must be practical and perform the same steps as the real study; the model should be technically generic and not limited to existing products and procedures; the model must be specific enough to further develop general technological requirements for each stage; the model should be abstract and applicable to law enforcement investigations, business investigations and incident response.
The authors of [11] proposed another (similar) event-based process model. This model is also based on physical crime scene investigations and suggests that digital crime scene investigations be conducted as part of physical crime scene investigations. This paper focuses on the steps involved in digital crime scene investigation and determining the causes and effects of events in digital forensics.

The authors of [12] proposed a digital forensic investigation process known as an incident model, which includes the following steps: pre-incident preparation, incident detection, initial response, response strategy formulation, redundancy (system backup), investigation, security, remediation (shutdown and containment of suspect system), network monitoring, recovery (reset of suspect system to initial state), reporting and follow-up.

The authors of [13] proposed a goal-based hierarchical process model for digital forensics and also conducted a detailed comparison between the proposed process model and previous work in the field. Their proposed model has multiple layers, which is a novel approach.

The authors of [14] proposed a comprehensive cybercrime investigation model that is very detailed. The proposed model also includes a description of information flow between different stages.

A process model was proposed by the authors of [15] which includes the following steps: identification, collection, conservation, transportation, preservation, analysis, interpretation, assignment, reconstruction, presentation and destruction.

The authors of [16] defined the stages of the digital forensics process: gathering information and making observations, formulating a hypothesis to explain the observations, evaluating the hypothesis, drawing conclusions, and communicating the results.

The authors of [17] proposed a mutual database that includes the design science research method of forensic investigation processes. The four stages proposed of this process consist of, namely: 1) identification; 2) artefact assembly; 3) artefact analysis; and 4) documentation and delivery processes. This lets us align everyone's concepts and terms common database forensics processes.

The authors of [18] developed a coordinated database pre-search model based on three main categories (i.e., planning, preparation and pre-response, acquisition and preservation, analysis and reconstruction). In addition, the forensic database is designed to avoid confusion or ambiguity and to provide practitioners with a systematic approach to performing DBFI with a greater degree of certainty.

The authors of [19] developed a metamodel forensic database field. Then identified, extracted and proposed the municipality concept and concept definitions are aligned to propose a meta-model. They have applied the metamodeling process to ensure that it is a metamodel comprehensive and consistent.

The authors of [20] developed a unified model begin to treat and organize the mobile foreign domain using the metamodeling method. The authors [21] proposed an integrated incident response template to identify a database forensics field, respond to, mitigate, and recover from a potential database incident.

The authors in [5] proposed a new forensic method, a readiness system called drone forensics using design science method. The supported model consists of two phases: (i) the active forensic phase and (ii) reactive forensic phase. The authors developed uniform forensics Forensic Database Field Template. The model is designed to be efficient and a comprehensive approach to forensic database investigation, addressing challenges in collection and analysis of digital evidence. The authors created a drone forensic metamodel. The developed meta-model ensured a uniform approach collect and organize evidence that allows the investigator to a to better understand the event.

The authors [22] developed the Internet of Things Metamodel of forensic investigation. The developed metamodel provided a detailed overview of the various stages of the investigation, from initial collection of evidence for potential identification and analysis problems. It is designed to be technology agnostic and scalable can be adapted to various scenarios. Additionally, several digital forensic works have been proposed to detect, and investigate the threats and risks of organizations.

### III. RESEARCH METHODOLOGY

This study adapted the design science method to develop a detection and investigation model for HDD. The design science method is a research method that involves creating a solution to a problem, then analysing the effectiveness of the solution [23]. The process involves creating the solution, testing the solution, and refining the solution as needed. The design science method is particularly useful for developing systems or models such as the HDD detection and investigation model. Fig. 1 illustrates the development process for developing and validating DIM for HDD. The development process involves three stages:

Stage 1: Recognizing and selecting digital forensic models:
This stage aims to identify and select appropriate digital forensic models from the literature for development purposes. It contains the following steps:

- Set recognized and selected criteria: To select the digital forensic models that are used in this study from the literature, the researcher looked at models that focused only on HDD, pen drive, RAM, and CD.
- Select the common Search Engines: For this study, five common search engines have been selected for searching and discovering digital forensics models: IEEE Explorer, Scopus, Web of Science, Springer, and Google Scholar.
- Assign a search protocol: The time and keywords were used for this study. The searching period is between 2015 and 2023, and the keywords are “Digital Forensic”, "Hard Disk Drive Forensic", and "Pin Derived Forensic", "Memory Forensic".
Filter the recognized and selected models: The results of the search yielded over 11,300 results from the search engines. These results were filtered to exclude results that were not related to the topic of HDD forensics. After the filtering process, the search yielded over 12 results that were relevant to the HDD forensics purely as shown in Table I.

**Fig. 1.** Development process for developing and validating DIM for HDD.

### TABLE I. SELECTED HDD FORENSIC MODELS

<table>
<thead>
<tr>
<th>ID</th>
<th>Year</th>
<th>Title</th>
<th>Advantages</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>2018</td>
<td>Digital forensic analysis of hard disk for evidence collection [24]</td>
<td>In this study, authors have discussed the significance of digital forensic examination of file systems to recover removed data from hard disks.</td>
</tr>
<tr>
<td>2</td>
<td>2017</td>
<td>Towards subverting hard disk firmware boot kits [25]</td>
<td>It was demonstrated by the authors that firmware boot kits can be identified, and several options were offered for how even deep-seated firmware rootkits can be identified as well.</td>
</tr>
<tr>
<td>3</td>
<td>2015</td>
<td>The differences between SSD and HDD technology regarding forensic investigations [26]</td>
<td>The focus of this study was to enhance the storage space of the system by utilizing SSD, which is a much quicker and further consistent storage device than old HDD.</td>
</tr>
<tr>
<td>4</td>
<td>2022</td>
<td>Comparing HDD to SSD from a Digital Forensic Perspective [27]</td>
<td>From a digital forensic viewpoint, this study examined in depth the results obtained from forensics software on HDD and SSD to determine whether there is a difference between the two drives in terms of forensics.</td>
</tr>
<tr>
<td>5</td>
<td>2019</td>
<td>A Comparative Study of Analysis and Extraction of Digital</td>
<td>The aim of this study was to provide a process for searching for evidence, the process may be a little simple or a little complicated. This would involve the location of a file saved on the device, or it could involve more complex processes that would involve hex sweeping or carving information in order to locate the necessary evidence in unallocated or slack memory.</td>
</tr>
</tbody>
</table>

6 2019 | Forensic Evidence from exhibits using Disk Forensic Tools [28] | little simple or a little complicated. This study analysed roadmaps of computer forensic evaluation methods described in scientific files and proposed a laboratory protocol to retrieve digital evidence from hard drives. The authors proposed a six-step procedure to diagnose the case, present results, collect evidence, make copies of files, analyze data, and extract information. |

7 2015 | Computer forensic analysis protocols review focused on digital evidence recovery in hard disks devices. [29] | The authors of this study explained how to retrieve data from a corrupted hard drive. A data improvement concept, data retrieval types, and the physical construct of modern HDD and their inner elements are reviewed first, followed by encountered failures. |

8 2019 | Data Sanitization Framework for Computer Hard Disk Drive: A Case Study in Malaysia [31] | In this study, the authors introduced a method for sanitizing data from computer hard drives. The proposed and tested a data sanitization process using commercially available tools. Several tests have been conducted at Cybersecurity Malaysia’s accredited digital forensic lab. |

9 2021 | A Digital Forensics Approach for Lost Secondary Partition Analysis using Master Boot Record Structured Hard Disk Drives [32] | Using forensic tools, this study investigates the removal or fraud of partitions on generally employed DOS / Master Boot Record (MBR) designed HDDs. |

10 2021 | Data recovery in a case of fire-damaged Hard Disk Drives and Solid-State Drives [33] | This study describes how data can be recovered from HDD and strong public drives of different companies and models that have been damaged by fire. |

11 2020 | Automated support tool for forensics investigation on hard disk images [34] | A new system was designed as part of this study in order to assist criminal investigators in finding evidence on images of hard drives recovered from suspects’ devices that could be used in conducting investigations. According to the authors, the content of the images on the disks was not encrypted, and therefore their focus was on multimedia content related to child abuse on the partition images. |

12 2121 | Forensic analysis and data recovery from water-submerged hard drives [35] | The study examined how long it will take for water to enter the hard drive once it has been submerged in water. |

Stage 2: Assembling common processes from the selected models: the aim of this step is to collect the common
investigation processes from the 12 selected models. It involves three steps:

- **Assembling criteria**: The investigation processes should be gathered solely from the main text or figure of the model. The collection process has omitted the following items: abstracts, related works, introductions, methodologies, and conclusions.

- **Assembling common processes from selected models**: The common investigation processes have been gathered and filtered from the selected models using the criteria above.

- **Recognize selected processes based on semantic meaning**: In this phase, three common investigation processes for the DIM have been selected based on their semantic meaning: detection, collection, and analysis. The investigation procedure begins with the detection phase, which involves identifying any suspicious activity on the HDD. This involves inspecting the HDD's system logs and event logs, as well as analysing the drive for malicious or suspicious files. It will be your responsibility to collect and save files from your hard drive as part of the collection phase of the project.

Stage 3: Developing a Detection and Investigation Model for HDD: The aim of this stage is to develop the DIM for the HDD. It consists of the following two primary steps:

- **Identify the relationships among processes and their activities**: The procedure begins with the detection phase. It involves the identification of possible digital evidence. The accumulation and preservation of digital evidence is the next step. It comprises both the gathering of evidence and the verification of its authenticity. The assembling phase is the third phase. In this phase, digital evidence is examined and evaluated for its relevancy. Fig. 2 depicts the DIM developed for the HDD.

  1) **Detection phase**: In this phase, there are three primary procedures: detection and investigation plans, detection and investigation tools, and investigation team identification. The detection and investigation plan includes the establishment of an organized plan outlining the steps required to detect any potential malicious activity on the hard disk drive (HDD). In contrast, the detection and investigation tools (which can be either software or hardware) concentrate on choosing the most appropriate tools for detecting any potential malicious activity on the HDD.

     a) **Identifying suspected HDD**: The purpose of this step is to identify suspected hard drives. A tampered or suspicious activity indicator is included here. Investigation teams must determine the drive's origin and identify it with its serial number or other unique identifiers. Further investigations can then be initiated based on this information.

     b) **Verification**: Following the identification of a suspected HDD, the investigative team will need to verify its legitimacy. To determine the integrity of the drive, tests may be conducted, or the data recorded on the drive may be examined using specialized forensic software. During this phase, the drive's authenticity and data integrity are determined.

     c) **Initial report**: A preliminary report will be drafted by the investigation team after the verification procedure has been complete. It should include the results of their verification procedure as well as any evidence of tampering. If necessary, the organization (decision makers) can conduct a thorough forensic investigation or contact law enforcement based on this report.

  2) **Gathering phase**: During the detection phase, a suspect HDD has been identified, and information is being collected from it. In the process of analysing suspicious hard drives, two primary procedures need to be followed: collecting and preserving all relevant data. FTK Imager is a forensic tool used to gather data from a suspected HDD that has been infected with malware. This data may include deleted files, fragmented data, and metadata. Preserving collected data requires ensuring that it can be analysed in the future. Several methods are available for achieving this goal. To preserve the data, copies of the data may be created, or images of the hard drive may be captured using forensic tools. To preventing data loss or corruption, this step must be performed as part of the analysis procedure.

  3) **Analysing phase**: With FTK Imager, forensic data can be collected from infected HDDs. There may be deleted files, fragmented data, and metadata within this data. Data collection must be preserved so that it can be analysed in the future. There are several methods available for achieving this goal. Using forensic tools, images of the hard drive can be captured to preserve data, or copies of the data can be created. As part of the analysis process, this step must be performed to prevent data loss or corruption.
Validating the effectiveness of the developed DIM: This part evaluates the applicability of the devised DIM to actual HDD attack investigations. The forensic instruments FTK Imager and HashMyFiles are utilized for this purpose. The FTK Imager tool is used to capture and analyze the data, while the HashMyFiles utility is used to store the captured data. In order to accomplish this, the researcher employed the following scenario: "We received a complaint from a consumer named Ahmed. He reported that his flash drive was broken, and he cannot access it. Considering this, the following paragraphs describe how DIM will detect, acquire, analyse, and record data from the suspected pin drive.

1) Detection phase: To accomplish this task, the researcher of this paper prepared the tools FTK Imager and HashMyFiles. During the investigation, the investigative team conducted interviews with the victim to gather all the information necessary to verify the pen drive incident. It is essential that information such as the size of the pen drive, the types of files inside the pin drive, and the last time the pen drive was used is gathered during the interview. Consequently, the researcher discovered that the pen drive had been compromised because of this attack. A team of investigators would need to move to the acquisition stage of the investigation to accomplish this goal.

2) Gathering phase: A second phase of the investigation will be focused on the collection and preservation of pen drive data using the FTK Imager and HashMyFiles tools. As shown in Fig. 3 and 4, it shows how the investigation team can collect data from the suspected pen drive using FTK Imager. To ensure that the captured data is not harmed, it is recommended that it is copied to an external flash drive and then duplicated. The data that has been captured must be protected from alteration or always tampering with HashMyFiles tool. The purpose of HashMyFiles is to create hashed values for data collected from the user by using the HashMyFiles tool as shown in Fig. 5. In the next phase, the examination and analysis process will be explained in detail.
Fig. 3. Collect data using FTK imager.

Fig. 4. Creating image from the suspected pen drive.

Fig. 5. Hashing collected data using HashMyFiles tool.

3) Analyzing phase: To verify the authenticity of the captured data before moving on to the analysis procedure, the authentication of the captured data needs to be verified using the FTK Imager tool to check whether it is consistent with the captured data. As a result, the hashed file has been verified as shown in Fig. 6. The researcher has confirmed that the authentication is correct, that the value is correct, and that no tampering has been done. This is why they can move forward with the next step of the investigation. By analyzing the captured information, malicious activities are identified because of analysis. After digging through the captured image, researcher found that the attacker had deleted information from the pen drive as shown in Fig. 7. The attacker was able to access the victim's computer remotely through a remote access program. After that, he used the "Delete file path" command to delete the files from the pen drive.

Fig. 6. Verifying hashed image using FTK Imager tool.

Fig. 7. Analyzing data using FTK Imager tool.

IV. RESULTS AND DISCUSSION

In a nutshell, based on the results and discussion presented, this study proposed a detection and investigation model that aims to detect and investigate the occurrence of HDD attacks. This model is mainly comprised of three phases which are detection, capture, and analysis. Therefore, a set of forensic tools namely FTK Imager and HashMyFiles are utilized to accomplish these phases.

Initially, the detection phase aims to identify any evidence of HDD attacks. Therefore, the FTK Imager was utilized in this study to acquire any relevant data contained on the HDD, including the volume name, number of sectors, and sector size. This information is used for identifying any discrepancy that demonstrates the differences between predicted and actual outcomes. If discrepancies exist, it indicates that the HDD drive may have been tampered with. Secondly, the capture phase aims to capture any related evidence of the HDD attacks. Likewise, the FTK Imager and HashMyFiles are utilized for capturing and preserving this evidence that is significant for succeeding analysis phase. Thirdly, the analysis phase aims to analyse and examine the evidence of HDD attacks. During this phase, the FTK Imager is predominantly used to examine the evidence. Based on this analysis, the investigator will have information about the attack, including the type of attack, the method employed, and the suspect. In addition, it will facilitate the diagnosis of the attack's severity and the prevention of future attacks. For method evaluation, it is assumed that the
pen drive has been tampered with. In this case, there are several files that have been deleted and damaged. Therefore, the proposed DIM is applied to a compromised flash drive to demonstrate its capability to detect, capture, and analyze HDD attacks. The testing demonstrated that the proposed DIM is capable of precisely detecting deleted files and identifying the removal methods. Consequently, this demonstrated the effectiveness of the proposed DIM in terms of the identification and investigation of HDD attacks.

Based on the results of the evaluation, there are several advantages of the proposed DIM model that could be beneficial to organizations such that, it facilitates in the prevention, mitigation, and acceptance of variations of potential HDD attacks. Therefore, it provides organizations with the option to secure their environment by acknowledging their susceptibility to assaults and by identifying and mitigating them swiftly.

Comparing to the existing model, Table II displays the comparison of the proposed DIM with the existing models. Clearly, the proposed DIM covered whole existing HDD digital forensics models.

<table>
<thead>
<tr>
<th>ID</th>
<th>Year</th>
<th>Existing Model</th>
<th>Proposed DIM</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>2018</td>
<td>[24]</td>
<td>✔️</td>
</tr>
<tr>
<td>2</td>
<td>2017</td>
<td>[25]</td>
<td>✔️</td>
</tr>
<tr>
<td>3</td>
<td>2015</td>
<td>[26]</td>
<td>✔️</td>
</tr>
<tr>
<td>4</td>
<td>2022</td>
<td>[27]</td>
<td>✔️</td>
</tr>
<tr>
<td>5</td>
<td>2019</td>
<td>[28]</td>
<td>✔️</td>
</tr>
<tr>
<td>6</td>
<td>2019</td>
<td>[29]</td>
<td>✔️</td>
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<tr>
<td>7</td>
<td>2015</td>
<td>[30]</td>
<td>✔️</td>
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<td>8</td>
<td>2019</td>
<td>[31]</td>
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<td>9</td>
<td>2021</td>
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<tr>
<td>12</td>
<td>2021</td>
<td>[35]</td>
<td>✔️</td>
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</tbody>
</table>

V. CONCLUSION AND FUTURE WORK

A hard disk drive is a very important component of a computer system that stores the operating system and applications. It is essentially a non-volatile memory device that stores digital information in a permanent manner. Generally, attackers are trying to access the valuable information on the HDD with the intent of damaging or stealing it. In this study, a detection and investigation model for HDDs was proposed to detect and investigate the various types of HDD attacks. In the proposed model, there are three main phases, namely detection, gathering, and analysis. FTK Imager has been used in conjunction with the newly developed detection and investigation model to detect, preserve, and analyze HDD attacks. Results showed that the proposed detection and investigation model can detect and analyze HDD and pen drive attacks. It is recommended that the future work on this study should be focused on the real-life scenario of HDD attacks.

REFERENCES

Application of VR Technology Based on Gesture Recognition in Animation-form Capture

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Abstract—To accurately capture the posture of animation characters in virtual vision and optimize the user's experience when wearing virtual vision equipment, the hybrid Gaussian model has gained wide attention. However, various types of animation show an exponential growth trend, and the hybrid Gaussian model is prone to low-dimensional explosion when processing these single frames. Based on the mixed Gaussian model, this study conducts animation character gesture recognition experiments on the Disert data set to solve these problems. Meanwhile, it is improved by frame rate reduction method to generate fusion algorithm. In this paper, the video is first grayened and filtered, and the model feature points of the image are marked. Then the weight learning rate is introduced and added to the set of pixels, and then the peak signal-to-noise ratio of Wronsky function is adjusted by changing the parameters. Then similar image sets are extracted and the structure elements are opened and closed. Finally, the proposed algorithm is applied to Disert data set. Meanwhile, the prediction accuracy of PSO is tested and compared with fusion algorithm. A total of 400 experiments were conducted, and the prediction accuracy of the fusion algorithm reached 392 times, with an accuracy of 98.0%. The accuracy of PSO is close to that of fusion algorithm (88.2%). It is verified that the suggested model can identify the four common gestures of cartoon characters well, and users will get a good viewing experience.

Keywords—Frame rate reduction method; model feature points; Wronsky function; mixed Gaussian model; weight learning rate

I. INTRODUCTION

In a society with the rapid increase of animation, the pose recognition model of animation characters has attracted wide attention, and the requirements of users have become higher and higher [1,2]. Character shape dynamic monitoring technology is to intercept the target from the played video, and then analyze the force status of the character's body in the frame, and then judge the current posture of the character. However, animation works have high requirements for clarity, that is, the frame rate contained in each video is rapidly increasing. The ability of the current algorithm to process fast-passing images is gradually falling behind, and the captured images cannot describe the video completely. In recent years, Hybrid Gaussian model (HG) has attracted the attention of many scholars due to its fast frame detection ability [3]. However, the suitable environment for HG processing pixels is low dimension, and the number of pixels that can be carried by the same dimension has an upper limit. HG is easy to cause data accumulation and even low-dimensional disaster in the working process. When a low-dimensional disaster occurs, it will not only destroy the two-dimensional configuration of the image, but also cause difficulties for subsequent information analysis. Most of the existing methods improve the ability of image frame detection, or make too much effort at the level of image clarity, thus ignoring the problem of information protection when analyzing information. In order to improve this situation, in this study, HG is optimized based on the Frame Rate Reduction (FRR), and the two are creatively fused to generate a fusion algorithm (HG-RPP). The algorithm builds a shared hyperplane, which can map linearly uncorrelated pixels to a higher plane and reduce the burden of low dimension. The steps of this study are divided into four parts. The first part mainly analyzes and summarizes the application effect of the current frame rate reduction model. The second part introduces the influencing factors of shape judgment and the construction of HG-FRR capture model. In the third part, the simulation test is carried out on the Disert data set. Finally, the performance of the optimization model and the traditional model is analyzed and compared, and the shortcomings of this study are pointed out. The aim of this study is to improve HG-RPP's tendency to fall into the low-dimensional dilemma, which not only has a significant effect on user happiness, but also on the sale of Virtual Reality (VR) products for merchants.

II. RELATED WORKS

A very important branch of animation character shape capture is animation character pose recognition, which plays a very important role in computer vision, users and businesses [4]. Liao et al. designed two terms based on supervised subspace learning and fragment sections to reduce the difference terms of in-class differences and to promote the block diagonal regularization terms of samples represented by in-class samples. The model maps the original samples to the learned subspace and classifies the mapped samples using a representation classifier with non-local constraints. Experiments have verified the good performance of the algorithm [5]. Yang et al. proposed a novel algorithm that can track multiple face gestures in a single frame or video. To create the first estimate of posture, the convolutional neural network model is first integrated with face identification and average shape learning. Then, a two-objective optimization strategy is used to iterate and apply algebraic filtering. Finally, experimental validation shows the benefits of the suggested approach [6]. Jain et al. proposed using deep learning to identify yoga poses and constructed a dataset with 10 positions from yoga. To capture the video, the study used a
smartphone camera. To recognize yoga poses in real time, a 3D convolutional neural network structure is designed and a supplementary layer, batch normalized average pool, is introduced. The experiment achieved a test recognition accuracy of 91.15% on the constructed data set. On publicly available data sets, competitive test recognition reached 99.39% [7]. Gunawan et al. proposed a deep learning architecture model that enables equivariant mapping of deep residuals to improve the attitude robustness of lightweight models as a solution to the underlying problem. The investigation assessed and contrasted the precision of several posture data sets, and the results showed that the suggested model improved accuracy by 0.07% and reduced verification time by 0.17 milliseconds [8].

With the development of the animation field, the clarity of the animation is gradually improved, and the number of attached pixels will also increase. The existing methods cannot continuously refine the pixel, so the algorithm based on pixel analysis is gradually studied. Zhang et al. suggested a deep learning model based on adversarial networks to synthesize faces and recognize faces with constant pose through the shape geometry of face images. The model generates a face that maintains identity by guiding the target pose. And the shape conveyed by the face separates identity from expression change. On controlled and field benchmark datasets, the proposed method performs well compared to the most advanced algorithms [9]. Yoganand et al. proposed a method to identify different gestures from video sequences. In this model, the video sequence is segmented into frames by the lens fragmentation process, and then from each shot, faces are recognised for subsequent processing. Facial features are extracted after the face is detected and the best frame in the video series is chosen using these characteristics. This approach is based on supervised learning and a bat algorithm-improved artificial neural network. Finally, a feature database is applied to recognise the facial picture [10]. Li et al. proposed an evaluation algorithm based on Dempster-Ter-Shafer Theory (DS). They believe that face detection can be achieved with low pixel monitoring. Therefore, DS is used to integrate the state of students' concentration, and the curve of students' concentration over time is obtained to describe the state of classroom concentration. On the data set provided by the computer camera, experiments have verified the efficiency and viability of the algorithm's design, and the accuracy rate of the algorithm is about 85.3% [11]. Chen et al. proposed a dimensionality reduction algorithm for principal component analysis. The algorithm uses the feature point screening function to filter the feature points extracted in advance by other algorithms, and projects the high-dimensional data into the low-dimensional space. Meanwhile, the redundant feature points are removed and the generation method of feature descriptors is changed to achieve the effect of dimensionality reduction. Experiments on open ORL face library have verified the superiority of the proposed model [12]. Zhu et al. found the importance of automated embedded systems, which work well in applications like private security and surveillance. Research suggests that smart door locks are prone to damage, increasing the risk. In order to provide users with effective open source software, the research proposes attitude tracking algorithms designed to ensure the security of modern keylock systems. Experimental results show that the established system has higher efficiency [13]. Maehama et al. developed an attitude recognition system that allows a robot to evaluate the content of aggressive words and determine whether the words are serious or joking. The study created a dataset of 16 participants. Different attitudes towards robots were observed in field experiments, which were applied to speech quality features of breath and pressed tone analysis, and combined with traditional prosodic features. Finally, the proposed method is compared with the speech standard. The final experimental results show that the combination of speech quality is superior to traditional neural networks in this task, with an accuracy equivalent to human guessing [14].

Through many international studies, it is found that the algorithm based on capturing the gesture of animated characters is very popular in the world, but the research on fusion algorithm is very few. In this study, we innovatively introduce feature point marking and bilateral filtering, and on this basis, take into account the impact of peak signal-to-noise ratio and shared hyperplane, and finally generate a fusion algorithm (HG-FRR). The purpose of this research is to improve the problem that HG-RPP is prone to fall into the low-dimensional dilemma, so as to enhance the user's sense of experience, and provide reference value for the relevant merchants' VR product sales strategy.

III. MODELING WITH THE MIXED GAUSSIAN MODEL OF WRONSKY FUNCTION METHOD

With the development of Virtual Reality (VR), more and more users are beginning to use VR technology. To cater to the preferences of users, the research on gesture recognition has gradually become a research hotspot, and the animation character shape capture technology is the most important part [15]. However, animation generally has objective factors such as high frame rate and complex character action, which increases the difficulty of shape capture equipment. In view of this, we combined the Frame Rate Reduction (FRR) method with the Hybrid Gaussian model (HG). This study first introduces the model based on FRR, and then describes the fusion method of the two in detail.

A. Establishment of Human Pose Feature Extraction Model based on Frame Rate Reduction Method

Before using VR to capture the shape of animation characters, the video image must be pre-processed. These include the method of turning gray to reduce the number of colors, and the method of bilateral filtering for video noise reduction. When the image color turns gray, the weighted average method is used in the study, as shown in Equation (1).

\[ I = 0.3R + 0.5G + 0.11B \]  \hspace{1cm} (1)

In Equation (1), \( I \) represents the color type after gray transfer treatment. The red, green and blue components are denoted as \( R, G, B \) respectively. Bilateral filtering is a nonlinear filtering method, which can use fixed templates to process adjacent pixels, and it can accurately distinguish spatial intensity differences [16]. The calculation of the bilateral filtering method is shown in Equation (2) below.
\[
g(x, y) = \frac{\sum_{k,l} f(k,l) w(i,j,k,l)}{\sum_{k,l} w(i,j,k,l)}
\]

\[
d(i,j,k,l) = \exp \left( \frac{(i-k)^2 + (j-l)^2}{2\sigma_d^2} \right)
\]

\[
r(i,j,k,l) = \exp \left( -\left| f(i,j) - f(k,l) \right|^2 \right) \overline{\sum_{i,j} \left| f(i,j) - f(k,l) \right|^2}
\]

In Equation (2), \( i, j, k, l, x, y \) represents pixels of the image. \( f(k,l) \) represents the gray degree of the image before bilateral filtering, and is represented by \( g(x, y) \) after filtering. The domain of pixel is denoted as \( d(i,j,k,l) \). The neighborhood difference between the two is the weight coefficient of the image, denoted as \( r(i,j,k,l) \). After graying and bilateral filtering, the image information will be missing, which is described by Equation (3).

\[
NMSE = \frac{\sum_{i=0}^{N} \sum_{j=0}^{N} \left[ g(x, y) - f(k,l) \right]^2}{\sum_{i=0}^{N} \sum_{j=0}^{N} f(k,l)^2}
\]

\[
PSNR = 10 \times \log \left( \frac{\text{MAX}^2 \times M \times N}{\sum_{i=0}^{N} \sum_{j=0}^{N} \left[ g(x, y) - f(k,l) \right]^2} \right)
\]

In Equation (3), \( NMSE \) is used to calculate the root-mean-square error before and after pixel conversion. Its signal-to-noise ratio is denoted as \( PSNR \). Their values are positively correlated with algorithm performance. \( MAX \) is a constant, and the value is 255, which represents the gray value of the image. The gray method and bilateral filtering method are applied to all pixels, and the neighborhood window is finally obtained, as shown in Fig. 1.

In Fig. 1, in the panoramic region set \( I \) composed of several pixels, take any pixel point with coordinate \((x, y)\), and scale the coordinate of the point in format \((x \pm 1, y \pm 1)\) to obtain nine neighborhood points, which together form the neighborhood window [17]. When the video sequence is clipped, it needs to be divided into several time nodes, and then the target image is obtained through the continuous changes between them, and then the shape capture is carried out, as shown in Equation (4).

\[
d_i(x, y) = \left| F_i(x, y) - F_{\text{ref}}(x, y) \right|
\]

In Equation (4), the image fragments of two consecutive time points are denoted as \( F_i(x, y), F_{\text{ref}}(x, y) \). The difference between pixel points in frame rates is represented by \( d_i(x, y) \). All image segmentation points have threshold values, and the segmentation of threshold values is as follows.

\[
h_i(x, y) = \begin{cases} 
255 & d_i(x, y) \geq T \\
0 & \text{otherwise}
\end{cases}
\]

In Equation (5), the image capture effect of the character is denoted as \( h_i(x, y) \). \( T \) represents the time interval for pre-selection. After using FRR to reduce the frame rate of the image, it can describe the outline of the object better. As a result, the structural elements can measure the image size faster, thus reducing the difficulty of video data. The structural elements can echo the corresponding form of the picture to make it compact, as shown in Equation (6).

\[
X \oplus B = \bigcup_{b \in B} (X) = \{Y: Y = x + b, x \in B, b \in B\}
\]

\[
X \ominus B = \bigcap_{b \in B} (X) = \{Y: (Y + b) \in B, b \in B\}
\]

In Equation (6), the structural element is denoted as \( B \), and there are two operations of expansion and corrosion between it and image set \( X \). The expansion operation is represented by \( \oplus \), which is a complement to the incomplete part of the image. The corrosion operation is denoted as \( \ominus \), and the marginal parts of the image can be clipped. Because the human body has a certain degree of softness, the posture presented is complex and changeable, so the ability of the algorithm has certain requirements. Up to now, the accuracy of monocular detection for 3D feature description is still lacking, far less than that of 2D. The two-dimensional description of human features is divided into two modules, including pre-processing module and feature extraction module, as shown in Fig. 2.
In Fig. 2, the video to be tested is first monitored for moving targets in the pre-processing module, and noise reduction is carried out at the same time. Then the data set is segmented by bilateral filtering threshold, and the human body image containing the target is obtained after normalization, and input into the feature extraction module. Then the features such as distance and Angle in the image are extracted and their eccentricity is calculated. Finally, the model of human body posture can be output, and the Equation (7) can be established according to the feature points of the model.

\[ \text{Feature} = \{D, A, e\} \]  

(7)

In Equation (7), the distance between a model feature and the human body’s centre of gravity is denoted as \( D \). The line defined by each feature point and the geometric center is called \( A \). If all lines are connected to form A closed pattern, their eccentricity is denoted as \( e \). There are countless lines passing through the center of mass of the human body, and the two lines that are parallel and perpendicular to the horizontal line are extracted. Using the law of Cosine, it is possible to determine the angle between the remaining line and the horizontal line, as illustrated in Equation (8).

\[ \alpha_i = \arccos \left[ \frac{\beta - \delta}{\delta} \right] \cdot \frac{180}{\pi} \]  

(8)

In Equation (8), \( \beta \) represents the abscissa of any feature point. The horizontal coordinate of the geometric center point of the human body is denoted by \( \delta \), and the vector distance between two points is denoted by \( \delta \). However, when the FRR hears the noise, it is greatly affected, which will lead to incomplete data collection, and further affect the accuracy of the results.

B. Research on Fusion Algorithm based on Hybrid Gaussian Model

When the FRR is used in practice to capture the posture characteristics of the human body in animation, there will be various errors. To avoid these errors, HG and its generative fusion algorithm (HG-FRR) are introduced. HG can apply multiple single Gaussian random distributions so that each pixel is included. And it can only correspond to the corresponding model, which is conducive to fast work, as shown in Fig. 3.

The HG-FRR workflow shown in Fig. 3 includes four main processes [18]. First, the predefined video is imported, and if the Gaussian model has been initialized, the Gaussian model background is established, and the Gaussian model is updated, and then the current frame is cut. Then determine whether the Langsky Function (LF) needs to be operated. After LF calculation, the level of Gaussian model can be obtained, and then the result can be output after human morphology processing or direct calculation. The frame rate segment without LF is processed by the moving target, and then the initial video is judged to be over. If it has finished, the final result is printed. The unfinished video continues the LF operation. The weighted summation method of HG is shown in Equation (9).

\[ e(a_i) = \sum_{j=1}^{K} w_{ij} \phi(a_i, u_{it}) \]  

(9)

In Equation (9), the moment of video interception is \( t \). The pixel value at that time is denoted as \( a_i \cdot \dot{i} \) is the total number of models. Its weight and mean value are called \( w_{ij}, u_{it} \) respectively, and their calculation equations are as follows (10).

\[
\begin{align*}
    w_{ij} & = (1-\varphi)w_{i,j-1} + \varphi \\
    u_{it} & = (1-\gamma)u_{i,t-1} + \gamma a_i
\end{align*}
\]

(10)

In the above Equation (10), \( w_{i,j-1}, u_{i,t-1} \) is the weight and mean value of the previous moment respectively. The mean learning rate is denoted as \( \gamma \), whose value is constant (0.001 in this study). The weight learning rate is represented by \( \varphi \), and the value range is \([0.001,0.010]\). LF can relate the same coordinates of nearby frame rate units together and is a vector type function, as shown in Equation (11).

\[ ||t|| = n^{-1} \cdot \sum_{j=1}^{n} \left( \frac{\kappa_{i_j} (x,y)_{1j}}{\kappa_{i-1} (x,y)_{1j}} - \frac{\kappa_{i_j} (x,y)_{1j}}{\kappa_{i-1} (x,y)_{1j}} \right)^2 \]  

(11)

In formula (11), the gray difference of the same pixel at adjacent moments is denoted as \( \kappa_{i_j} (x,y)_{1j}, \kappa_{i-1} (x,y)_{1j} \). The total number of pixels in the picture is denoted as \( n \). \( ||t|| \) is a physical quantity that describes the correlation between pixels, and the greater the distance from 0, the more uncorrelated the two pixels are. When \( ||t|| \to 0 \), the image set and structural elements can be switched on, that is, deepened on Equation (6), as shown in Equation (12).

\[ \left\{ \begin{array}{l}
    X(B = (X \oplus B) \oplus B) \\
    X(B = (X \oplus B) \oplus B)
\end{array} \right. \]  

(12)

Fig. 3. Flowchart of HG working characteristics.
In Equation (12), \( \langle \cdot \rangle \) represents the open operation, which has the ability to fill image gaps. The closed operation is denoted as \( \circ \), which removes redundant parts of the image. Things in the real world come with regularities, which are often found by learning machines. The learning machine requires lower empirical risk and higher fitness, and its learning process is shown in Equation (13).

\[
\mu(w) = \int L(v, o(x, w)) d\sigma(x, y) 
\]

In Equation (13), the classical parameters are denoted as \( w \). The loss function is represented by \( L(v, o(x, w)) \). \( \sigma(x, y) \) is a directivity function for positioning pixels. The coordinate set of all pixels is denoted as \( o(x, w) \) [19]. A Support Vector Machine (SVM) can process uncorrelated pixels in an image due to its kernel technique. The working state of SVM confidence range is shown in Fig. 4.

\[
\theta(x) = \sum_{i=1}^{n} \theta_i o(x_i, x) + b 
\]

In Equation (14), \( \theta(x) \) is a classification function of the kernel function. The original spatial dimension is denoted as \( \theta_i \). The mapped spatial dimension and pixel abscissa are called \( \xi_1, \xi_2 \). \( b \) represents the vector difference between dimensions. Even in the face of multi-pixel data sets, it can be trained one-to-one by Equation (14). To compare the similarity between images, the direct square matrix normalization method is introduced, and its operating equation is shown in Equation (15) [20].

\[
sim(Fig_1, Fig_2) = 1 - \varphi(\xi_1 - \xi_2) 
\]

In Equation (15), the randomly extracted images are denoted as \( Fig_1, Fig_2 \). Their gray coefficients are indicated by \( \xi_1, \xi_2 \).

IV. MODEL EXPERIMENT OF ANIMATION CHARACTER SHAPE CAPTURE BASED ON HG

To verify the effect of HG-FRR algorithm in practice, the research builds HG-FRR model based on personalized recommendation, and iterates and verifies its accuracy. Finally, simulation experiments are carried out on Disert dataset using HG-FRR model.

A. HG-FRR System Development Environment and Model Parameter Determination

In this study, self-collected Disert data set was selected, including four kinds of animation character postures: upright, walking, running and jumping, with a total of 2417 frames of action images. Considering the limited types of data, the training set and test set are split up into the data set in a 2:3 ratio. The research equipment and software used in the experiment are shown in Table I.

<table>
<thead>
<tr>
<th>Data Set</th>
<th>Development Environment</th>
<th>Code</th>
<th>Internal Storage</th>
</tr>
</thead>
<tbody>
<tr>
<td>Disert</td>
<td>Python 11.2</td>
<td>Open CV</td>
<td>512 G</td>
</tr>
<tr>
<td>Operating system</td>
<td>Display card</td>
<td>Database</td>
<td>Processor</td>
</tr>
<tr>
<td>128Ubuntu 22.01.20</td>
<td>36.0 GHz</td>
<td>Mysql 5.20.23</td>
<td>Intel Core i8</td>
</tr>
<tr>
<td>Web development framework</td>
<td>Language</td>
<td>Operator</td>
<td>Model</td>
</tr>
<tr>
<td>Django1.22.3</td>
<td>Easy Chinese</td>
<td>Standing, walking</td>
<td>F2.8LH-US M</td>
</tr>
</tbody>
</table>

The collected data sets need to be further processed to enable the research algorithm to learn. For the processing of data sets, HG-FRR was used for iterative optimization. In order to verify its accuracy, traditional K-means algorithm, Random Forest algorithm (RF) and Particle Swarm Optimization algorithm (PSO) are compared with it. The results of accuracy and error rate in the training set are shown in Fig. 5.

Fig. 5. Comparison of accuracy-training set image and error rate-training set image.

From Fig. 5, HG-FRR has a slightly lower accuracy rate and a higher error rate than K-means and PSO before 100 training sessions. However, the accuracy of HG-FRR is higher than the two algorithms when the number of iterations reaches more than 100, and tends to be stable when the number of iterations reaches 190, and is higher than the other three algorithms. Although the increase of iterations will reduce the operating efficiency of the model, the accuracy weight of the
The model is higher after comprehensive consideration. Therefore, the proposed HG-FRR algorithm has better performance. After learning the HG-FRR algorithm, the parameter determination in the test should also be taken into account, as shown in Fig. 6.

The parameter of this study is weight learning rate \( \phi \in [0.001, 0.010] \). It can be seen from Fig. 6 that the error rate is lowest (0.042) when the weight learning rate is 0.005 and there are 150 iterations. Therefore, it is finally determined that the number of iterations is 150 times, and the regularization term takes a value of 0.005.

**B. Experimental Verification of Animation Character Pose Recognition based on HG-FRR**

To verify the accuracy of HG-FRR model in recognizing animation characters' pose, simulation experiments were conducted. By observing the task image discrimination ability of HG-FRR algorithm, the practicability of HG-FRR algorithm is judged. First, the HG-FRR algorithm is initialized, and then the video to be detected is input in the data preprocessing module. Finally, the weight learning rate is set to 0.005, and the attitude judgment records of animation characters within 60 seconds are collected, and the images are drawn after calculating errors, as shown in Fig. 7.

Fig. 7 shows the error comparison of the four algorithms in the experiment. From Fig. 7, we see that after 38s, the total error of HG-FRR has approached 0, while the other three algorithms have not stabilized. In particular, the PSO algorithm reached the highest error value of the four algorithms at 3s, which was -1.48%. The total error variation range of HG-FRR, K-means, RF and PSO is significantly compared, and the algorithm performance can be easily compared. However, it is not objective enough to rely only on the total error analysis, so the research separately analyzed the attitude misjudgment caused by the gender difference of the four cartoon characters and the motion amplitude, and drew the image as shown in Fig. 8.

From Fig. 8, the experimental results of the HG-FRR model are concentrated in the range of total error 0. The error range caused by gender difference of cartoon characters is [-0.3%, 0.4%]. The error range due to the motion amplitude is [-1.0%, 0.5%]. The errors of the remaining three algorithms are widely distributed, and the larger errors are sparsely distributed. In order to more intuitively distinguish the error correction ability of the four algorithms, the research conducted 400 experimental data records and drew the image as shown in Fig. 9.
From Fig. 9, RF has the largest error variation range among 400 error test experiments. That is, the frequency of errors was the highest, recorded as [-0.8%, 0.6%]. The second is the K-means algorithm, which is between [-0.2%, 0.3%]. The error range of PSO is close to that of HG-FRR, with values above [-0.13%, -0.03%]. The error curve of HG-FRR fluctuates between -0.02% and 0.04%, with the smallest fluctuation range. The RF and K-means algorithms with the top two errors were excluded, and only HG-FRR algorithm and PSO were compared to capture the correct experimental results, and the error fan chart was drawn to obtain the image as shown in Fig. 10.

![Error matrix of HG-FRR algorithm and PSO algorithm.](image)

Fig. 10 identifies four pose types of upright, walking, running and jumping based on the characteristics of cartoon characters and characters, and shows the experimental results of accurate prediction of HG-FRR and PSO. HG-FRR was accurate 392 times, with an accuracy of 98.0%, and PSO was accurate 88.2%. In order to observe the experimental results of HG-FRR and PSO more directly, linear fitting graphs of the two algorithms were drawn based on matrix, and the predicted values of the two algorithms were compared with the real values, as shown in Fig. 11.

![Linear fitting diagram of HG-FRR and PSO.](image)

Fig. 11 shows the comparison between the predicted and true values of the two algorithms. In Fig. 11, the linear fit degree ($R^2$) of HG-FRR algorithm is 0.9904, and that of PSO is 0.9546, indicating that there is no underfitting of the model. To sum up, it can be concluded that the HG-FRR algorithm model can well capture and identify the common gestures of animation characters, so as to give users a good VR experience.

V. RESULT AND DISCUSSION

It can be seen from the experimental results that the total error of HG-FRR has approached zero after 38s, while the other three algorithms have not stabilized, which shows that the experimental effect of the fusion algorithm is better than the other three algorithms in the stability test of the system. In the total error of HG-FRR, the error range caused by the gender difference of cartoon characters is [-0.3%, 0.4%], and the error range caused by the action range is [-1.0%, 0.5%], while the error distribution of the other three algorithms is irregular and sparse, and only the total error of HG-FRR can be judged to be the lowest. It shows that the proposed fusion model has the best prediction effect under the same experimental environment. For the persuasiveness of the experiment, the study conducted 400 experiments. In 400 error test experiments, the error range of RF is the largest, which is recorded as [-0.8%, 0.6%]. K-means algorithm is between [-0.2%, 0.3%]. The error range of PSO is close to HG-FRR, and the value is above [-0.13%, -0.03%]. The error curve of HG-FRR fluctuates between 0.02% and 0.04%, and the fluctuation range is the smallest. It shows that HG-FRR is still the best in the extensive error test. In the experimental results of accurate prediction of HG-FRR and PSO, the prediction accuracy of HG-FRR is 98.0%, and that of PSO is 88.2%. The linear fitting degrees of them are 0.9904 and 0.9546, respectively. It shows that HG-FRR is extensive in the experiment of predicting cartoon characters. Through many experiments, it is proved that the fusion algorithm proposed in this study has the lowest total error and the lowest distribution error among the algorithms for judging the image of anime characters. Because this error is very sensitive to the influence of cartoon character image judgment, so the small error in the experiment cannot be ignored. However, most animation works rely on freehand drawing and post-processing, which will bring a lot of noise to the prediction of animation characters. Under this background, the proposed fusion algorithm can still keep the error between 0.02% and 0.04%, which shows that the algorithm is highly adaptable to the judgment of animation character image and is suitable for capturing the character form in animation.

VI. CONCLUSION

With the development of the Internet industry, it is becoming more and more important to identify the posture of animation characters when watching animation, such as increasing the play experience for users and improving the visibility of animation merchants. In this study, a fusion algorithm (HG-FRR) was constructed based on Frame Rate Reduction (FRR) and Hybrid Gaussian model (HG). In this study, both bilateral filter denoising and Lonsky function are taken into account, on the Disert data set, simulation tests are run, and comparison is made with K-means and other three algorithms. 40% of Disert data set was extracted and trained on HG-FRR model through experiments of shared hyperplane. Finally, the number of iterations is determined to be 150 times, and the weight learning rate is 0.005. In the error analysis experiment, a total of 400 experiments were conducted. K-means and RF error of the two algorithms in [-0.2%, 0.3%], [-0.8%, 0.6%] within the scope of volatility. The error curve of HG-FRR fluctuates between -0.02% and 0.04%, with the
smallest fluctuation range. The variation range of PSO is close to that of HG-RPP, between [-0.13%, -0.03%]. In 400 experiments, the prediction accuracy of HG-FRR is 98.0% and that of PSO is 88.2%. The linear fitting graph of the two algorithms based on matrix is studied. $R^2$ of HG-FRR is 0.9904, indicating excellent linear fitting. PSO’s $R^2$ is 0.9546. To sum up, it can be concluded that the HG-FRR algorithm model can accurately capture the posture of animation characters, which can not only improve the user's viewing happiness, but also make the animation produced by merchants more well-known. However, the HG-RPP model is only applicable to the analysis of animation with blurred background. For works with complex background, the character characteristics are not obvious, and the model will mark them as noise. This is because the commercial value of animation works belongs to private information, and the dataset analyzed in this study contains few types. With more volunteers, it is believed that future studies can be improved.

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REFERENCES


GDM-PREP: A Rule-Based Technique to Enhance Early Detection of Gestational Diabetes Mellitus

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Abstract—Gestational diabetes mellitus (GDM), a condition occurring solely during pregnancy, poses risks to both expectant mothers and their infants, particularly among individuals with pre-existing risk factors. However, early diagnosis and effective management of GDM can help mitigate potential complications. As part of the Ministry of Health's efforts to enhance screening and management strategies for GDM in Malaysia, this study aims utilizing a rule-based technique, acting as an Expert System for Initial Screening of Gestational Diabetes Mellitus Detection. This application will facilitate early diagnosis by assessing risk factors and symptoms to calculate the probability of GDM occurrence and classify it as low, medium, or high. Functionality and usability tests are conducted to ensure error-free performance and gather user feedback. The study’s findings indicate that the self-check GDM system effectively utilizes the algorithm, while the mobile application showcases good usability, achieving an above-average System Usability Scale (SUS) score.

Keywords—Gestational diabetes mellitus (GDM); rule based; expert systems; risk factor

I. INTRODUCTION

Gestational diabetes mellitus (GDM), which only manifests during pregnancy, is the term used to describe diabetes during pregnancy. This is a severe condition that causes pregnant women who were previously normal but discovered to have higher levels of blood glucose during pregnancy [1]. GDM is increasing in prevalence, with 21 million cases (or 7% of the global population) being reported globally [2]. The number of pregnant women with GDM is increasing due to changes in eating habits, increased purchasing power, and climate change [3].

In the early stages of pregnancy, the mother’s body undergoes several changes that turn her into a different individual with unique physical and mental features leading to changes in health habits and lifestyle. Hence, those habits and lifestyle choices during pregnancy seem to have permanent and long-term effects on the mother’s and child’s health [4]. GDM raises the risk of short-term and long-term risks in pregnant women, including pre-eclampsia, caesarean section rates, miscarriage, and subsequent lifelong diabetes. Children of mothers with GDM are more probably to have neonatal respiratory distress syndrome and hypoglycemia and develop diabetes, obesity, and metabolic disorders later in life.

Women at risk to develop GDM is when they are having the presence of any risk factors such as a body mass index of more than 27 kg/m2, previous history of GDM, first-degree relative with diabetes, history of macrosomia, bad obstetric history, developed glycosuria ≥2+ on two occasions, and any current obstetric problems. These recommendations from the Clinical Practice Guidelines (CPG) are intended to be clinical practice manuals in Malaysia based on the best information available at the time of development [1].

In support of the Ministry of Health’s objectives of continuing to develop improved screening and management strategies for GDM in Malaysia and of preventing the development of type 2 diabetes mellitus in pregnant women [5][6], this study aims utilizing a rule-based technique, acting as an Expert System for Initial Screening of Gestational Diabetes Mellitus Detection based on the risk factors and symptoms.

Early diagnosis of GDM is important to minimize the risk, but in the first trimester of gestation, the Oral Glucose Tolerance Test (OGTT) is ineffective as pregnancy-induced hyperglycemia is not always clearly apparent in the first months of gestation [3]. Therefore, screening for a disease is advised when the disease is common and clinically significant, and when there is a clear screening test that will classify most diseased persons without high rates of false-positive or false-negative outcomes. The early screening of diseases is very important for society as it contributes to improvements in the quality of life and economic growth in the countries. This screening allows the early initiation of proper treatment of the disease to prevent death [3]. In addition, the screening and subsequent detection of GDM before pregnancy enable effective management to reduce maternal and fetal morbidity and mortality associated with pregnancy hyperglycemia [6].

In the research conducted by [7], Malaysia is in the fifth rank in Asia with an 18.5% prevalence rate of GDM. To reduce the prevalence rate in Malaysia and to have good primary care for the mother and patients at risk at a minimal cost, needs a pre-screening test procedure that is easy and gives new knowledge to know better about themselves [8][9]. Several biochemical tests for diabetes diagnosis in early pregnancy were proposed, which as the Oral Glucose Tolerance Test (OGTT). However, it is an expensive and manual test, and its poor tolerability may affect enforcement, especially in the context of nausea during the early stages of pregnancy [6]. Therefore, in this research, the initial screening of the GDM expert system is proposed based on risk factors.
and symptoms. This initial screening test is the first step in diagnosing GDM in the CPG published by the Ministry of Health Malaysia [1]. However, according to a medical expert, the initial screening test is conducted in a casual interview only, because currently, there is no mechanism for the initial screening of GDM in clinics and hospitals.

Increasing awareness among women and their families about GDM is pivotal to ensure early detection and treatment. "Women who are vulnerable to the risk factors of GDM should contact their doctor before they decide to become pregnant and ensure that they seek guidance and support in the control of their blood sugar," [20]. Increased risk for the complications of GDM during the first and second trimester of pregnancy is due to poor healthcare [6]. Thus, in order to achieve the best outcome of pregnancy, pre-screening test is an early step that can be taken to avoid the risks [2]. This research is aimed to spread awareness to people that GDM can occur from risk factors. Therefore, initial screening is advised as it can be a precautionary measure before and during the pregnancy to prevent complications.

The main contribution is the development of GDM-PREP, an innovative mobile application that employs rule-based techniques to improve the early detection of Gestational Diabetes Mellitus (GDM). By seamlessly integrating with the mobile environment, GDM-PREP offers a user-friendly and accessible platform for expectant mothers. The application incorporates a set of carefully designed rules and symptoms tailored to the Malaysian context, ensuring accurate classification and timely detection of GDM in this specific population.

II. RELATED WORK

This section discusses existing mobile applications for pre-screening GDM. There are a few expert systems that had been developed, which are d-GDM 2019, Gestational Diabetes Health Tips & Care, Diabetes Diagnostics and Diabetes Test - risk calculator of Diabetes.

d-GDM is an interactive mobile application that was created in 2015 by Garnweidner and Cols [10]. Garnweidner and Cols.’s aim for the application is to diagnose and give information regarding the result to be followed up to present on a mobile phone screen. The application was developed using WHO’s suggested variables and parameters of GDM which are fasting glycemia, random glycemia, and glycated haemoglobin (HbA1c) in the first trimester. d-GDM applied an open source for all the informatics tools of the application. The application will generate and present the information regarding the result of the diagnosis of GDM after the user submitted all the information needed in the application.

Diabetes Diagnostics has been developed by Natural Aptitude with a team of world leaders, which is the University of Exeter Medical School team who lead the diabetic monogenic diseases and the front line of work in diagnostic aids for subtypes of diabetes. It can be retrieved from the website (https://www.natural-aptitude.co.uk/project/diabetes-diagnostics/). It is an application that helps physicians worldwide detect unusual types of diabetes, as objectively to provide patients with improved care. Natural Aptitude also stated that this probability calculator is created in a mobile application for offline use, along with a range of other diagnostic tools and other information aimed to increase the accuracy of the diagnosis. The application needs several important informations about the user, such as age, sex, racial group, BMI, HbA1c level, and other risk factors. The best feature of Diabetes Diagnostics is that it displays a probability of the diagnosis, as it gives a description of the disease in the result section.

Diabetes test - risk calculator of diabetes is a mobile application which requires iOS 8.0 or later that was created in 2017 by Pears Health Cyber can be accessed through (https://apps.apple.com/us/app/diabetes-test-risk-calculator-of-diabetes/id1014960572). The developer of the application claimed that the Diabetes Test - risk calculator of diabetes is a multiplex and reliable risk calculator on a mobile device for the development of diabetes mellitus. They also added that “We have developed a test using a combination of many test algorithms that takes account of a wide range of variables that can detect smaller degree than any other risk calculator.” The result then will be generated after simply enter in the application of the details of age, gender, family anamnesis, physical health, BMI, and other factors.

III. METHODOLOGY

A. Expert System

An expert system is one of artificial intelligence (AI) technologies. Expert systems (ES) are systems based on knowledge which are part of the former AI research field and can be defined as knowledge-intensive software that performs tasks that usually require human expertise [11].

- Rule-Based

A rule-based system is a system that uses laws as the main principle of representation [12]. The meanings of a rule-based system depend almost entirely on ES which imitates human expert reasoning in solving an intensive problem of knowledge [13]. It also stated that a rule-based system encodes a human knowledge expert into an automated system in a rather narrow area. The rules consist mainly of two parts rule antecedent and rule consequent, where the rule antecedent is the if the part that specifies a set of predictor attribute values referencing the conditions. While the rule consequent is then part which specifies each example that the predicted class of the rule meets the conditions in the rule antecedent [14]. A rule-based system usually consists of a variety of if-then rules that can be used for various purposes, such as reinforcement of decisions or predictive decision-making [15]. The rule base is the set of rules which represents the knowledge about the domain [12]. The general form of a rule is such below:

If cond1 and cond2 and cond3 ...
then action1, action2, ...

The conditions cond1, cond2, cond3, etc., also known as antecedents are evaluated based on what is currently known about the problem being solved. Some systems would allow disjunctions in the antecedents. For example, the rules in general form. Such rules are interpreted to mean that if the
antecedents of the rule together evaluate to true for example if the Boolean combination of the conditions is true, the actions in the consequents which are the action1, action2, etc., can be executed. Each antecedent of a rule typically checks if the problem instance satisfies some condition. For example, an antecedent in a rule in a medical expert system could be the patient has previously undergone heart surgery. The complexity of antecedents can vary a lot depending on the type of language used. For instance, in some languages, one could have antecedents such as the person’s age being between 10 and 30.

The advantage of the rule-based system is the structures, as its homogeneity therefore the uniform syntax enables one to easily analyze the context and interpret each rule [16]. There is also a limitation to the rule-based system which is, the rule-based system will provide an inadequate explanation. Most expert systems offer a facility to explain the user’s behaviour. There are very few explanatory facilities provided by a rule-based system. Therefore, the explaining facilities provided by a rule-based system are only suitable when the user and the expert are equally knowledgeable.

Research by [3] applies Bayesian Network, Multicriteria Analysis and Expert Systems to improve GDM diagnosis from data mining techniques, which will display more reliable random trees and a lower error rate result from the experiments. Registration of rules to facilitate diagnosis and the accuracy of the tests can be enhanced by increased rules, thereby increasing the specialty of the system and its ability to function in its domain. Another study [17] proposed a rule-based diagnosis system for diabetes. Technique Fuzzy-based logic rule is ideal to develop a system of knowledge based on medical disease. The proposed expert system is very useful both for the patient and for doctors to diagnose the disease correctly. The laboratory test results can differ in certain ways, and it is time-consuming since they depend entirely on the availability and expertise of the physicians. Besides, a model proposed by [9], can help to diagnose T2DM early and avoid potential complications linked to late diagnosis. Although the absence of laboratory diagnostic tests on diabetes decreased the sensitivity and accuracy of the proposed model in the research, the model is an evolution of an early diagnosis of diabetes without using laboratory diagnostic tests.

B. Pre-Screening of GDM

Gestational Diabetes Mellitus (GDM) is related to an increased risk of short and long-term maternal and perinatal complications [18]. Early screening, diagnosis and diabetes prevention reduce the high costs of disease control and complication treatments and prevent admission into the hospital due to serious complications [9]. There are various guidelines for screening and diagnosis of GDM such as guidelines from the World Health Organization (WHO), National Institute for Health and Clinical Excellence (NICE), American Diabetes Association (ADA), etc. However, this research will follow the CPG published by the Ministry of Health of Malaysia as a reference. This section will be explained on Algorithm of GDM in CPG and the risk factors and symptoms of GDM.

- Algorithm of GDM in CPG

Based on Fig. 1, the first step of screening and diagnosis of GDM is screening the patient at risk to develop GDM and women age more and equal to 25 with no other risk factors. It will then proceed with 75g OGTT to confirm that the patient is diagnosed with GDM. Therefore, the screening test is important as it is a determinant of the OGTT to be conducted. In addition, the identification of risk factors is significant to determine whether women are at risk for an early diagnosis and intensive lifestyle changes [7].

![Fig. 1. Algorithm of screening and diagnosis of GDM stated in CPG.](image1)

C. Risk Factors and Symptoms of GDM

Although many other risk factors are also considered in another guideline for screening and diagnosis of GDM, Malaysian CPG will only use seven risk factors to diagnose GDM. As shown in Fig. 2 the risk factors of GDM are Body Mass Index (BMI) of more than 27kg/m² and experienced GDM in a previous pregnancy. In addition, first-degree relative with diabetes is also a risk factor for GDM. The first-degree relative’s would be parents, siblings, and children [19]. Other than that, history of macrosomia, in which the birth weight of the baby is more than 4 kg. The next risk factor is the patient who experienced bad obstetric history and glycosuria, where the urine contains more blood sugar than usual on two occasions. The one who is facing obstetric problems, such as essential hypertension, pregnancy-induced hypertension, polyhydramnios, or current use of corticosteroids, would be the last risk factor for screening and diagnosis of GDM.

![Fig. 2. Risk factors of GDM are stated in CPG.](image2)
Next, there are five symptoms of GDM [18]. The symptoms are polydipsia, which is excessive thirst and polyuria. Polyuria is a condition where the body urinates more than normal and each time you urinate, it passes excessively large amounts of urine. Other than that, the patient might suffer exhaustion, nausea, and repeated vomiting. Lastly, the symptom of GDM is blurred vision, which experienced a decrease in clarity or sharpness vision.

The objective of this research is to classify the user with GDM. Risk factors and symptoms of GDM will be used as an indicator in this research. Based on various previous research, rule-based is an ideal technique to develop an expert system for a diagnosis of a disease. Rule-based has also been applied as its structures provide procedural interpretations that allow them to be viewed as models of computation. Although BN has an adaptability feature, a rule-based system dividing the rule base from the inference engine distinguishes the knowledge from the way of how to solve the problem. This means that the same inference engine can be used with several rule bases and a rule base for different inference engines may be used. Therefore, it is easy to add or apply a new rule with the same bases or antecedent.

D. Risk Factors and Symptoms Selection

This research is focused on the self-check function. In the function, there are two parts, which are risk factors and symptoms of GDM that have been used as the parameters for the self-check function. All the information about the parameters had been collected and gathered from the Clinical Practice Guidelines: Management of Diabetes in Pregnancy, published by the Ministry of Health Malaysia. There is also additional information that had been gathered on the parameters of the self-check function, where each of the parameters has its impact on the result generated.

In addition, the data and information on parameters are collected and have been revised and verified by the medical expertise. Therefore, there are seven risk factors and five symptoms of GDM. The risk factors of GDM, there is a body mass index of more than 27 kg/m², previous history of GDM, large amounts of urine. Other than that, the patient might suffer exhaustion, nausea, and repeated vomiting. Lastly, the symptom of GDM is blurred vision, which experienced a decrease in clarity or sharpness vision.

E. Algorithm

The rule of GDM detection is designed in algorithm form before the development of the system and implemented in Java. Two conditions are to be applied by rules. The first condition is the user will only be answering for the risk factors section in the self-check function. The second condition is the user will be answering both risk factors and symptoms section. The condition that will be experienced by the user in the self-check GDM function is based on the result generated from answering the risk factors section. There are seven questions regarding risk factors. Each question referred to each risk factor. For each question, if the user answered ‘Yes’, the number recorded in the parameter ‘total parameter’ is incremented from 0 to be used to calculate the probability. As for the user that experienced answering symptoms question, five questions need to be answered. Each question referred to each symptom. The total number of ‘Yes’ answered by the user is recorded in the same parameter, the ‘total parameter’ which is incremented from the question in risk factor.

Table I shows the algorithm of the rule of GDM detection in pseudocode.

```plaintext
IF (Total Risk Factor = 0%) THEN GDM = Low
IF (Total Risk Factor < 20% AND Risk 2 = No) THEN GDM = Low
IF (Total Risk Factor < 50%) THEN GDM = Medium
IF (Total Risk Factor < 50% AND Risk 2 = Yes) THEN GDM = Medium
IF (Total Risk Factor < 50% AND Risk 1 = Yes) THEN GDM = High
IF (Total Risk Factor > 50%) THEN GDM = High
IF (Total Risk Factor and Symptoms < 50%) THEN GDM = Medium
IF (Total Risk Factor and Symptoms > 50%) THEN GDM = High
IF (Total Risk Factor and Symptoms < 50% AND Risk 1 = Yes) THEN GDM = High
```

After users have answered all questions, the probability of having GDM will be calculated based on their input in each question of risk factor or both risk factor and symptom. The value of the ‘total parameter’ will be collected from the user on what they are experiencing in their current condition, and it will be then divided into the total number of risk factors or divided into both the total number of risk factors and symptoms. Then, it will multiply by 100 to get the probability. To be simplified, the formulas to calculate the probability are shown in Table I.

```
<table>
<thead>
<tr>
<th>Condition</th>
<th>Formula to calculate the probability</th>
</tr>
</thead>
<tbody>
<tr>
<td>The user answered risk factors only.</td>
<td>( \frac{Total\ Yes}{7 \times Total\ Risk\ Factor} \times 100 )</td>
</tr>
<tr>
<td>The user answered risk factors and symptoms.</td>
<td>( \frac{Total\ Yes}{13 \times Total\ Risk\ Factor\ and\ Symptoms} \times 100 )</td>
</tr>
</tbody>
</table>
```

From the Algorithm 1, the user will get the result of their probability of GDM based on 7 or 13 parameters depending on the result in the risk factors section. The result in the risk factors section must be more than 50% and the user is currently pregnant to answer the symptoms section. Otherwise, the user will only need to answer the risk factors section to get their probability to have GDM. There are three categories to classify the probability of GDM, which are low, medium, and high. The example of cases for classifying the probability from rules is visualized in Table II.
Algorithm 1: Algorithm for the rule of GDM detection.

```plaintext
Initialize
Tp = 0; P = 0;
Compute
While () do
    For (each Risk Factor 1 to Risk Factor 7) do
        IF Ri is YES, then
            Ri = 1 AND Tp = Tp+1, Otherwise 0;
        End
    IF (P > 50 AND Pregnant =YES)
        THEN FOR each Symptom 1 to Symptom 5 DO
            IF Si is ‘YES’
                THEN Si = 1 AND Tp = Tp+1, Otherwise 0;
            End
        P = (Tp/13) x 100
    Else
        P = (Tp/7) x 100
    End
    Return Tp and P;
End
```

Next, based on Fig. 3, the rule to classify the use of probability to have GDM is based on the user risk factors or both risk factors and symptoms. User needs to answer each risk factor and symptom-based on their current condition. Based on the answer by the user, it will then be calculated the probability to have GDM. Then, the rules will be implemented in the classification of the probability. If all the parameters match the rule, then the probability and classification for the user will be produced because of the self-check function.

F. User Interface (UI) Design

On the Home page, there is one function for the user to have their probability to have GDM by clicking the Self-check GDM option. Users need to answer each question displayed as shown in Fig. 4. To submit the answer based on the user's current condition, they need to click one option from the radio group and click the button NEXT and click on the button SUBMIT on the last question.

![Sample of self-check GDM questions page.](image)

TABLE II. EXAMPLE OF CASES FOR CLASSIFYING THE PROBABILITY FROM RULES

<table>
<thead>
<tr>
<th>No</th>
<th>R1</th>
<th>R2</th>
<th>R3</th>
<th>R4</th>
<th>R5</th>
<th>R6</th>
<th>R7</th>
<th>S1</th>
<th>S2</th>
<th>S3</th>
<th>S4</th>
<th>S5</th>
<th>Probability</th>
<th>Class</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>No</td>
<td>No</td>
<td>No</td>
<td>No</td>
<td>No</td>
<td>No</td>
<td>No</td>
<td>No</td>
<td>No</td>
<td>No</td>
<td>No</td>
<td>No</td>
<td>0%</td>
<td>LOW</td>
</tr>
<tr>
<td>2</td>
<td>No</td>
<td>No</td>
<td>Yes</td>
<td>No</td>
<td>No</td>
<td>No</td>
<td>No</td>
<td>No</td>
<td>No</td>
<td>No</td>
<td>No</td>
<td>No</td>
<td>14%</td>
<td>LOW</td>
</tr>
<tr>
<td>3</td>
<td>No</td>
<td>No</td>
<td>Yes</td>
<td>No</td>
<td>No</td>
<td>No</td>
<td>No</td>
<td>No</td>
<td>No</td>
<td>No</td>
<td>No</td>
<td>No</td>
<td>29%</td>
<td>MEDIUM</td>
</tr>
<tr>
<td>4</td>
<td>No</td>
<td>Yes</td>
<td>Yes</td>
<td>Yes</td>
<td>No</td>
<td>No</td>
<td>No</td>
<td>No</td>
<td>No</td>
<td>No</td>
<td>No</td>
<td>No</td>
<td>43%</td>
<td>MEDIUM</td>
</tr>
<tr>
<td>5</td>
<td>Yes</td>
<td>Yes</td>
<td>No</td>
<td>No</td>
<td>No</td>
<td>No</td>
<td>No</td>
<td>No</td>
<td>No</td>
<td>No</td>
<td>No</td>
<td>No</td>
<td>29%</td>
<td>HIGH</td>
</tr>
<tr>
<td>6</td>
<td>Yes</td>
<td>No</td>
<td>No</td>
<td>No</td>
<td>No</td>
<td>No</td>
<td>No</td>
<td>No</td>
<td>No</td>
<td>No</td>
<td>No</td>
<td>No</td>
<td>14%</td>
<td>HIGH</td>
</tr>
<tr>
<td>7</td>
<td>No</td>
<td>Yes</td>
<td>Yes</td>
<td>Yes</td>
<td>No</td>
<td>No</td>
<td>Yes</td>
<td>No</td>
<td>No</td>
<td>No</td>
<td>No</td>
<td>No</td>
<td>46%</td>
<td>MEDIUM</td>
</tr>
<tr>
<td>8</td>
<td>No</td>
<td>No</td>
<td>Yes</td>
<td>No</td>
<td>Yes</td>
<td>Yes</td>
<td>Yes</td>
<td>No</td>
<td>No</td>
<td>No</td>
<td>Yes</td>
<td>Yes</td>
<td>54%</td>
<td>HIGH</td>
</tr>
<tr>
<td>9</td>
<td>Yes</td>
<td>Yes</td>
<td>No</td>
<td>No</td>
<td>No</td>
<td>No</td>
<td>Yes</td>
<td>Yes</td>
<td>No</td>
<td>No</td>
<td>No</td>
<td>No</td>
<td>31%</td>
<td>HIGH</td>
</tr>
</tbody>
</table>

Indication:

<table>
<thead>
<tr>
<th>R1</th>
<th>BMI of more than 27 kg/m2</th>
<th>S1</th>
<th>Polydipsia</th>
</tr>
</thead>
<tbody>
<tr>
<td>R2</td>
<td>First-degree relative with diabetes</td>
<td>S2</td>
<td>Polyuria</td>
</tr>
<tr>
<td>R3</td>
<td>Previous history of GDM</td>
<td>S3</td>
<td>Exhaustion</td>
</tr>
<tr>
<td>R4</td>
<td>History of macrosomia</td>
<td>S4</td>
<td>Nausea, repeated vomiting</td>
</tr>
<tr>
<td>R5</td>
<td>History of bad obstetric</td>
<td>S5</td>
<td>Blurred vision</td>
</tr>
<tr>
<td>R6</td>
<td>Glocosuria</td>
<td></td>
<td></td>
</tr>
<tr>
<td>R7</td>
<td>Current obstetric problems</td>
<td>NoP</td>
<td>Do not experience R7 but pregnant</td>
</tr>
</tbody>
</table>

After the user submits all their answers to all the questions, the application will display the result page. On the result page, there is the Home button for the user to view the Home page and the Next button for the user to review the detailed explanation of the result displayed. Fig. 5(a) shows the Result page. In Fig. 5(b), the total risk factor that strikes the risk factor of GDM will be displayed, based on the user’s answer. If the user wants to return to the previous page, click on the Back icon. On the Detail Explanation page, it will view all explanations on the user's answer on each parameter. It contains the Home icon and Next icon. If the user wants to return to the home page, click on the Home icon. Click the
Next icon if the user wants to view the next explanation for the next parameter and click the bullet icon if the user wants to view the previous explanation for the previous parameter, as shown in Fig. 5(c).

IV. RESULT AND DISCUSSION

In this section, the implementation of the rule for the first condition of the algorithm will be discussed first. There are seven parameters to be considered in the first condition. It consists of the risk factors of GDM. There is a body mass index of more than 27 kg/m², previous history of GDM, first-degree relative with diabetes, history of macrosomia, bad obstetric history, developed glycosuria on two occasions, and any current obstetric problems. From all the answers submitted by the users with the most suitable current condition for the seven parameters mentioned above, the result of the probability of the users having GDM will be obtained. There are three classes of the probability of GDM, which are low, medium, and high. The rule to classify the probability of GDM is shown in Fig. 6.

Table III concludes the classification of the result based on the probability calculated and extra conditions considered for the result. To be classified in the low class of the probability of GDM, the result of the probability must be 0% or less than 20% and the second risk factor of GDM, which is the user’s parent, brother or sister is experienced diabetes is false. However, to be classified in the medium class of the probability of GDM, the result of the probability must be more than 0% and less than 50%. Other than that, if the probability is more than 0% and less than 50%, and the second risk factor which is the user’s parent, brother or sister are experienced diabetes is true, the user also will be classified in the medium class of chances to have GDM. Next, to be in the high class of the probability of GDM, the result of the probability must exceed 50%. In addition, if the user’s BMI is more than 27 kg/m², the user also will be classified in the high class of the probability of GDM.

For the second condition of the algorithm, it will only be implemented if the result in the first condition, which is the result of the probability exceeds 50% and the users are currently pregnant. In this condition, the user will answer questions about the symptoms of GDM. There are 13 parameters to be considered. There are seven risk factors in the first condition and another five symptoms of GDM, which are polydipsia, polyuria, exhaustion, nausea, repeated vomiting, and blurred vision. The classification of the probability of the prevalence of GDM for the second condition is medium and high only. It is based on the probability calculated in which the 13 parameters are all considered. The user will have a medium class of probability if the result of the probability is more than 0% and less than 50%. The user also will have a medium class of probability of GDM if the probability is between 1% to 50% and the second risk factor which is the user’s parent, brother or sister is experienced diabetes is true. And the high class of probability is if the result of the probability is exceeded 50% and the user’s BMI is more than 27 kg/m².

A. Usability Testing

Usability testing is the process of testing with a community of representative users on how simple a design of the system is to be used to ensure that the program is results-oriented, easy to use, reliable and easy to implement into the everyday life of the user. There are several usability assessments, and which survey is one of the usability tests to be used in this project. Target users who are consisted of pregnant women, married women, and the team of the Clinical Practice Guideline of the Ministry of Health Malaysia. The survey has been conducted via video conferencing with 10 respondents. The average total System Usability Scale score is 86.75.
The respondents need to answer on a scale from scale 1 to scale 5 in usability testing questions. Scale 1 refers to strongly disagree, scale 2 refers to disagree, scale 3 refers to neutral, scale 4 refers to agree, and scale 5 refers to strongly agree. From all the results of each question given in the SUS questionnaire to 10 respondents among the target user, Table IV shows the total score for each respondent that has been calculated based on the result of the SUS questionnaire. Questions 1, 3, 5, 7, and 9 are odd. Therefore, it needs to subtract by 1. For the even statements, questions 2, 4, 6, 8, and 10, will be subtracted from 5. Then, to get the result, all the scores that have been subtracted will sum up and multiply by 2.5. Based on Table IV and Fig. 7, the average SUS score obtained for the application is 86.75. From the result, it can be concluded that the application has good in terms of usability by achieving the average SUS score, which is more than 68 scores.

TABLE IV. TOTAL SCORE FOR 10 RESPONDENTS BASED ON SYSTEM USABILITY SCALE

<table>
<thead>
<tr>
<th>Respondent</th>
<th>1</th>
<th>2</th>
<th>3</th>
<th>4</th>
<th>5</th>
<th>6</th>
<th>7</th>
<th>8</th>
<th>9</th>
<th>10</th>
</tr>
</thead>
<tbody>
<tr>
<td>A total score of the application</td>
<td>82.5</td>
<td>10</td>
<td>8</td>
<td>9</td>
<td>62.5</td>
<td>10</td>
<td>8</td>
<td>8</td>
<td>87.5</td>
<td>9</td>
</tr>
</tbody>
</table>

![TOTAL SUS SCORES](image)

Fig. 7. Bar graph of total SUS scores.

V. CONCLUSION

A. Strengths and Limitations

The development of the system is basically to ease the user to have their current probability based on the risk factors and symptoms. With the detailed explanation provided, it also eases for the user to refer to the information, for future use without the need to refer to the hard copy of the clinical practice guideline. Due to the time constraint in collecting the information with the expertise, the application is only focused on one type of diabetes, which is GDM, known as diabetes in pregnancy. Thus, make this application data scope only GDM. As for that, the application does have only one main function, which is the function of the user to have their probability to have GDM. Furthermore, the application only supports the English language.

B. Conclusion and Future Research

To enhance the application's capabilities, future research is suggested to explore the implementation of various machine learning techniques. By comparing user data with existing cases of GDM that share similar risk factors and symptoms, the application's diagnostic accuracy and precision can be further improved.

While the GDM-PREP currently focuses on improving health outcomes related to Gestational Diabetes Mellitus, there is potential for expansion to handle extensive databases and diagnose a broader range of diseases. This adaptability allows the system to address multiple health conditions, increasing its overall impact and usefulness in healthcare settings.

This novel approach holds the potential to significantly enhance GDM screening and management strategies in Malaysia, thereby benefiting both healthcare professionals and pregnant individuals.

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Innovating Art with Augmented Reality: A New Dimension in Body Painting

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Abstract—This study investigates the fusion of augmented reality (AR) and body painting as a novel concept for artistic expression. By combining the immersive capabilities of AR with the creative potential of body painting, this research explores individuals’ perceptions and attitudes towards this innovative artistic approach from an HCI perspective. Drawing upon the Technology Acceptance Model (TAM) and the Diffusion of Innovation Theory (DIT), the study examines the factors influencing individuals' acceptance and intention to engage in AR-integrated body painting. Additionally, the research explores the mediating role of artistic expression in understanding the impact of these factors on the actual outcomes of this merged concept. A sample of 212 respondents participated in an online survey to accomplish the research objectives. The survey comprehensively measured participants’ perceptions of innovativeness, social system support, perceived usefulness, perceived ease of use, artistic expression, and behavioral intention towards AR-integrated body painting. Rigorous data analysis was conducted using Partial Least Squares Structural Equation Modeling (PLS-SEM) to examine the intricate relationships between the variables. The findings underscore the significant impact of factors such as Innovativeness, social system support, perceived usefulness, and perceived ease of use on individuals’ acceptance and intention to engage in AR-integrated body painting from an HCI perspective. Moreover, the study reveals the mediating role of artistic expression in connecting these influential factors with the actual outcomes of this merged concept. These empirical insights substantially contribute to our understanding of the fundamental mechanisms driving the adoption and utilization of AR in artistic practices, particularly within the domain of body painting, from both an artistic and HCI standpoint.

Keywords—Augmented reality; body paintings; artistic expression; technology acceptance

I. INTRODUCTION

Our world has undergone a profound revolution driven by digital transformation, permeating every feature of our lives. As technology continues to integrate into various spheres, it has breathed new life into unexplored fields and disciplines beforehand. While technology is conventionally correlated with engineering and computer science, its potential impact on cultural studies, fine arts, and history-related disciplines has often been overlooked. However, within these realms, the intersection of technology and human expression holds immense promise. For instance, digital humanities and cultural preservation can significantly benefit from incorporating technology. Furthermore, recognizing the significance of human interaction with technology within the realms of fine arts and non-technological/non-scientific disciplines is crucial. Embracing technology in these contexts can unlock new avenues for creative exploration and foster deeper connections between art, culture, and society [1].

The art of body painting has long been an underrated phenomenon, often overlooked in its association with modern technology such as augmented reality. While body painting has a rich historical and cultural significance, its potential for innovation and exploration by integrating augmented reality remains largely undiscovered. By merging the traditional practice of body painting with cutting-edge technological advancements, such as augmented reality, new dimensions of artistic expression and transformative experiences can be unlocked. This uncharted territory offers a unique opportunity to bridge the gap between ancient art forms and contemporary technological advancements, allowing for a reimagining of body painting as a dynamic and immersive artistic medium. By shedding light on the untapped potential of this fusion, we can elevate body painting to a new level of appreciation and redefine its role in the context of modern art and technology.

Integrating Augmented Reality (AR) technology in body painting can potentially revolutionize the fine arts discipline. AR presents unique assistance, including virtual design visualization, interactive and dynamic art experiences, real-time feedback and corrections, digital preservation and documentation, and collaborative and remote opportunities [2], [3]. By overlaying virtual designs onto the body, artists can experiment with various concepts and styles before applying physical paint. AR augments engagement through interactive elements and visual effects, appealing to the audience and raising the impact of the artwork. Artists can receive immediate feedback, leading to precise and accurate designs. Furthermore, AR accelerates digital preservation and documentation, allowing archival and contextual information integration. Collaborative and remote experiences become possible, enabling artists to collaborate regardless of location and expanding the audience's access to body painting as an immersive and innovative art form.

While body painting as an artistic practice has received limited attention in research, the integration of augmented reality (AR) technology with body painting remains largely unexplored. Previous studies on body painting have mainly discussed its potential for teaching anatomy to health science students [4], [5] enhancing anatomical education through AR [6] and utilizing body painting for the teaching of anatomy and public engagement [7]. However, there is a scarcity of research specifically examining the integration of AR with body painting.
While discussions on the application of augmented reality (AR) technology are prevalent in various sectors, integrating AR with body painting is an underexplored area. Research has examined the effects of AR on students' achievement, attitudes towards the course, and participation in classroom activities [8], as well as its impact on body awareness and self-experience through virtual embodiment [9]. Augmented reality marketing has also been studied regarding its definition, complexity, and prospects [10]. Challenges and future research directions in education have been identified in augmented reality [11], and studies have investigated the impact of dimensionality and spatial abilities on learning with augmented reality [12]. Furthermore, AR technology has been applied to improve mirror fitness [13] and explore presence, avatar embodiment, and body perception [14]. Augmented reality has also been examined in the context of face filters as augmented reality art on social media [15], augmented reality art as a creative medium [3], and the motivations and effects of using AR face filters on social media [16]. Additionally, research has focused on the technological advancements and future perspectives of augmented reality and virtual reality displays [17], the development of augmented reality applications for learning [18], the survey of industrial augmented reality [2], and an overview of augmented reality technology [19]. Studies have also explored the impact of augmented reality applications on learning motivation [20] and the use of augmented reality and virtual reality in education [21]. Moreover, social interaction in augmented reality has been investigated [22]. While integrating body painting with augmented reality has received limited attention in research, studies have examined the enhancement of anatomical education through augmented reality [6] and the use of body painting and other art-based approaches to teach anatomy [23].

There is a pressing need for research to integrate augmented reality (AR) technology with body painting, as it can uncover the potential benefits and challenges of combining these two artistic practices, leading to innovative and immersive experiences for artists, performers, and viewers alike. Such research can contribute to advancements in art education, medical visualization, and interactive performance art. Additionally, investigating the integration of AR technology with body painting from a human-computer interaction (HCI) perspective is essential to shedding light on usability, user experience, and interactive aspects, providing insights into designing intuitive and engaging AR interfaces for artistic expression. Understanding the human factors involved in interacting with AR-enhanced body painting can enhance user satisfaction, immersion, and effectiveness, bridging the gap between art, technology, and human perception.

The integration of augmented reality (AR) technology with body painting can be studied using the Technology Acceptance Model (TAM) and the Diffusion of Innovation Theory (DIT) to understand users’ behavioral intentions and perceptions towards this novel artistic medium [24]–[27]. These models provide insights into usability, user experience, perceived usefulness, perceived ease of use, and social influence, which are crucial in determining the acceptance and adoption of AR-enhanced body painting [28], [29]. By considering the findings from these studies, artists, performers, educators, and designers can optimize the design and development of intuitive and engaging AR interfaces for artistic expression while addressing users' concerns and promoting wider adoption [5], [30], [31].

The integration of augmented reality (AR) technology in the field of body painting poses both opportunities and challenges. While AR has gained significant attention in various domains such as entertainment, education, and healthcare, its application and impact within fine arts, particularly body painting, remains relatively unexplored. This research aims to address this gap and shed light on the factors that influence the adoption of AR in body painting and their effects on artistic expression and behavioral intention. This study explicitly explores AR technology's integration in body painting from an HCI perspective. It investigates the role of HCI principles in enhancing the usability, accessibility, and user satisfaction of AR applications in artistic practices. The research considers the viewpoints of both artists who adopt AR in their body painting techniques and the audience engaging with AR-enhanced artworks. By focusing on the HCI aspect, the study aims to provide insights into the design considerations, interaction patterns, and user-centered approaches that optimize the integration of AR technology in the context of body painting.

The significance of this research lies in its contribution to the understanding and application of HCI principles in the domain of AR-enhanced body painting. By exploring the factors that influence the adoption and usability of AR in artistic practices, this study can guide artists and designers in creating immersive and user-centred AR experiences. From an HCI perspective, the findings offer practical implications for interface design, interaction techniques, and user feedback mechanisms, enabling artists to leverage AR technology effectively. Furthermore, this research expands the body of knowledge on AR in the fine arts domain, particularly in the context of body painting. It fills a gap in the existing literature, which predominantly focuses on AR applications in other industries, such as entertainment and education. By integrating HCI principles, this study advances the theoretical and practical understanding of AR technology in body painting, contributing to the broader field of HCI research and artistic practices.

II. RELATED WORK

A. Augmented Reality and Body Painting

Traditionally, body painting was adept by indigenous cultures for various purposes, such as ceremonial rituals and visual communication [32]. Natural pigments sourced from plants, fruits, and minerals were used to decorate the body, with symbolic colours carrying cultural implications. These ancient techniques relied on the binding medium to adhere the pigments to the surface, creating a vibrant and meaningful art form. Modern body painting can be evolved to incorporate technology, such as AR, to create immersive and interactive experiences [33], [34]. AR integration in body paint allows artists to transcend traditional boundaries, introducing dynamic visual effects and virtual elements onto the human canvas [35]. This fusion of art and technology expands artistic possibilities, enhances viewer engagement, and bridges the gap between physical and digital realms. However, ethical considerations
Integrating AR technology into body painting within the orbit of fine arts elevates the creative process and unlocks novel channels for expression, engagement, and innovation. Embracing this technology empowers artists to push the limits of body painting, crafting immersive, interactive, and visually mesmerizing experiences that redefine the boundaries of the art form [13], [15], [17], [19].

B. Artistic Expression of Body Painting (AE)

The literature review conducted for body painting as an artistic expression within the positive art framework highlights five consistent positive outcomes across all the aforementioned art forms: sense-making, enriching experience, aesthetic appreciation, entertainment, and bonding. These outcomes emphasize the potential of body painting as a powerful vehicle for individuals to find meaning and purpose, enhance their overall experiences, develop a deeper appreciation for aesthetics, derive entertainment, and establish social connections. In supporting the relevance of body painting within the positive art framework, the article draws on a range of scholarly references. Lomas [40] discusses the concept of positive art and the potential for artistic expression and appreciation to foster flourishing. Javornik et al. [16] explore the motivations and well-being effects of using augmented reality (AR) face filters on social media, which can be considered an extension of body painting. Geroimenko [3] and Hsu and Chin [34] shed light on the emergence of augmented reality as a creative medium in art, including body painting.

The discussion surrounding body painting as a form of positive art also incorporates studies from education and medical sciences. Diaz and Woolley [4], Finn [41], Ribelles-García et al. [5], and Wang et al. [42] explore the pedagogical aspects of body painting in teaching anatomy and enhancing learning experiences. Haugstvedt and Krogstie [43], Vovk et al. [43], Rese et al. [44], and Iqbal and Sidhu [27] studies foster the understanding of technology acceptance, including augmented reality, in various contexts.

C. Diffusion of Innovation Theory (DIT)

The Diffusion of Innovation Theory (DIT), explained by Everett Rogers, approaches a valuable framework for interpreting the adoption and dissemination of innovative ideas, products, and technologies from a social perspective [45]. DIT defines diffusion as the process through which innovations are communicated and embraced by members of a social system over time. It identifies critical factors that impact the speed and extent of adoption, encompassing the innovation’s attributes, the characteristics of adopters, the communication channels utilized, the social system involved, and the temporal aspect of the adoption process [46], [47]. By shedding light on adoption stages, adopter types, and influential factors, DIT facilitates comprehension and prediction of innovation acceptance and utilization across diverse domains such as technology, healthcare, and social sciences [48]. In augmented reality (AR) integration with body paintings, DIT helps understand how artists and viewers embrace this innovative artistic practice [49]. Innovators are the first to adopt AR-enhanced body paintings, followed by early adopters who recognize its creative potential. The early majority adopts it based on positive experiences, while the late
majority joins once it becomes well-established. Laggards are the last to adopt, often due to resistance to change or scepticism [50]. Understanding DIT can inform strategies for promoting the acceptance and diffusion of AR-enhanced body paintings [51].

D. Technology Acceptance Model (TAM)

The Technology Acceptance Model (TAM) is a widely recognized and validated model for understanding technology integration. It focuses on users' behavioral intentions to adopt technology based on its perceived usefulness and ease of use. Davis initially developed TAM in 1986 to explore the acceptance of novel technologies in workplace settings [52]. By examining individuals' attitudes and perceptions, TAM provides insights into the factors influencing technology adoption and usage, aiding in designing and implementing effective technology solutions [53]. TAM can be applied to understand the acceptance of body painting through augmented reality (AR) technology. By examining users' perceptions of the usefulness and ease of use of AR-enhanced body painting, TAM can provide insights into their behavioral intentions to adopt and engage with this novel artistic medium [42]. Factors such as the perceived benefits, convenience, and user experience of AR-enhanced body painting can be explored within the TAM framework to understand and predict the acceptance and utilization of this technology-driven artistic expression [29]. This understanding can inform the design and development of user-centric AR interfaces and promote wider adoption of AR-enhanced body painting in the artistic community [26].

E. Hypothesis Development

Numerous hypotheses are developed by contemplating the literature and related studies along with the aim of this research. These hypotheses are then validated by multivariate statistical analysis.

1) Perceived Ease of Use: Perceived ease of use (PEOU) is a crucial factor in influencing artists' artistic expression (AE) of body painting when considering the adoption of AR technology. Artists who perceive AR as easy to use and navigate are likelier to embrace this technology in their body painting practices [30]. The user-friendly nature of AR facilitates artists' interaction with the technology and enables them to explore its potential for enhancing their artistic expression. When artists perceive AR as easy to use, they are more motivated to incorporate it into their body painting techniques and leverage its capabilities to create visually captivating and interactive artworks [26], [30]. Additionally, perceived ease of use (PEOU) impacts artists' behavioral intention (BI) through the mediator of artistic expression (AE). When artists find AR technology easy to use and navigate, it positively influences their intention to adopt AR for body painting. The ease of use associated with AR enhances artists' confidence in utilizing the technology and encourages them to explore its possibilities for artistic expression [27], [37], [42]. As artists experience the ease of incorporating AR into their body painting practices and witness the positive impact on their artistic expression, their behavioral intention to adopt AR technology strengthens. Therefore, the following hypotheses are being proposed:

H1: PEOU positively impacts the AE to adopt AR technology.

H1a: PEOU positively impacts the BI through AE to adopt AR technology.

2) Perceived usefulness: Perceived usefulness (PU) plays a vital role in shaping artists' artistic expression (AE) of body painting when considering the adoption of AR technology. Artists who perceive AR as helpful in enhancing their creative process and expanding the possibilities of artistic expression are likelier to embrace this technology [26]. Incorporating AR into body painting offers a range of features and functionalities that enhance artists' ability to create visually captivating and interactive artworks. As artists recognize the usefulness of AR in enriching their artistic expression, they are motivated to explore its potential and incorporate it into their body painting practices. Furthermore, perceived usefulness (PU) impacts artists' behavioral intention (BI) through the mediator of artistic expression (AE). When artists perceive AR as a valuable tool for enhancing their artistic expression, it positively influences their intention to adopt AR for body painting [29], [31]. The perceived usefulness of AR in body painting fuels artists' motivation to explore and experiment with this technology, leading to greater engagement and a stronger intention to incorporate it into their artistic practice. As artists witness the positive impact of AR on their artistic expression, their behavioral intention to adopt AR technology for body painting increases [24], [42]. Therefore, following hypotheses are being proposed:

H2: PU positively impacts the AE to adopt AR technology.

H2a: PU positively impacts the BI through AE to adopt AR technology.

3) Social systems: The social system (SS) can significantly influence the artistic expression (AE) of body painting and the adoption of AR technology. Within the social system, artists are influenced by various factors such as societal norms, cultural values, and peer interactions. The acceptance and support of the social system towards body painting as an art form can positively impact artists' willingness to explore innovative technologies like AR for their artistic expression [15], [25]. Artists feel encouraged and empowered to incorporate AR technology into their practices when the social system embraces body painting as a legitimate artistic expression. The acceptance and recognition of body painting within the social system motivate artists to experiment with AR and explore its potential for enhancing their artistic expression. Moreover, the social system provides a platform for artists to showcase their AR-enhanced body paintings, which can further influence the adoption and acceptance of AR technology within the artistic community [25], [28]. Furthermore, the social system influences artists' behavioral intention (BI) through the mediator of artistic expression
(AE). Artists who receive support and recognition from the social system for their body painting endeavours, especially when augmented by AR, are more likely to develop a stronger intention to adopt AR technology. The positive response from the social system reinforces artists’ belief in the value and significance of incorporating AR into their body painting practices. The social system's acceptance and appreciation of AR-enhanced body painting contribute to artists’ confidence in embracing this technology and their intention to use it for future artistic endeavors [49], [51]. Therefore following hypotheses are being proposed:

H3: SS positively impacts the AE to adopt AR technology.

H3a: SS positively impacts the BI through AE to adopt AR technology.

4) Innovativeness: The relationship between the Innovativeness of individuals and the artistic expression of body painting (AE) can be explored through the Innovation Diffusion Theory (IDT) lens. According to IDT, Innovativeness refers to the willingness and eagerness of individuals to adopt new ideas or technologies [47]. In the context of AE, individuals with a high level of Innovativeness are likelier to embrace and experiment with novel approaches, techniques, and mediums in their body painting practices. They are open to integrating augmented reality (AR) technology with body painting, leveraging its capabilities to enhance and expand their artistic expression. By keeping such an analogy, AE can instigate the adoption of AR as a mediator between the Innovativeness of personality and Behavioral Intention to adopt (BI) the AR technology for body painting [9], [13], [17]. Therefore two hypotheses for Innovativeness are developed. These hypotheses can be formulated as follows:

H4: Innovativeness positively impacts the AE to adopt AR technology.

H4a: Innovativeness positively impacts the BI through AE to adopt AR technology.

F. Conceptual Framework

Based on the proposed hypotheses, a conceptual framework is established, in Fig. 1, to understand the adoption of AR technology for body paint purpose to fortify the fine arts discipline into digital transformational tools.

III. METHODOLOGY

This study ensued a deductive approach within the research onion framework utilized by Saunders [54]. It adopted a cross-sectional design and employed quantitative research methods to investigate the phenomenon under examination. The deductive approach was employed in this study, which involved testing existing theories of TAM and IDT and hypotheses to draw conclusions. By employing a deductive approach, the study aimed to explore the relationships between variables identified in the theoretical framework, namely the perceived usefulness, perceived ease of use, Innovativeness, social system, and behavioral intention in the context of digital body painting and augmented reality (AR) for artistic expression in Yunnan.

A cross-sectional design was adopted, enabling data to be collected at a single point in time. This design allowed for the examination of relationships between variables. It provided a snapshot of individuals’ attitudes, perceptions, and behavioral intentions regarding using digital body painting and AR for artistic expression in Yunnan. The cross-sectional design was suitable for investigating the interplay between the independent variables (perceived usefulness, perceived ease of use, Innovativeness, social system) and the dependent variable (behavioral intention) within a specific time frame. A non-probability sampling technique called snowball sampling was employed in this study. Initially, several participants with relevant knowledge and experience in digital body painting and AR were purposively selected. These participants were asked to refer to other potential participants who met the criteria. This iterative process continued until the desired sample size of 212 participants was reached, as suggested by the previous research [55].

Data were collected through an online self-administered close-ended questionnaire. The questionnaire consisted of two parts: demographic features and variables questions. The demographic section collected participants’ age, gender, educational background, and artistic experience. The variables questions assessed perceived usefulness, perceived ease of use, Innovativeness, social system, and behavioral intention using a 5-point Likert scale (ranging from 1=strongly disagree to 5=strongly agree). Each variable's questionnaire items were adopted from previously validated studies to make the inferences significant and robust.

The collected data were subjected to statistical analysis using appropriate techniques. Quantitative analysis was conducted using Partial Least Squares Structural Equation Modeling (PLS-SEM). This method allowed for assessing both the measurement and structural models, enabling the examination of relationships between variables. Additionally, PLS-predict analysis was conducted to understand the model's predictive power and relevance in predicting behavioral intention based on the assessed variables [56]–[58]. Ethical guidelines and principles were followed throughout the study. Informed consent was obtained from all participants, ensuring their voluntary participation and the confidentiality of their responses. The study adhered to ethical standards to protect the participants’ rights and well-being and ensure the research findings' integrity and credibility.
IV. DATA ANALYSIS

A. Demographic Results

Demographic results are detailed in the following Table I.

<table>
<thead>
<tr>
<th>Category</th>
<th>Percentage</th>
<th>Category</th>
<th>Percentage</th>
</tr>
</thead>
<tbody>
<tr>
<td>Gender</td>
<td></td>
<td>Exposure to AR Technology</td>
<td></td>
</tr>
<tr>
<td>Male</td>
<td>72%</td>
<td>Yes</td>
<td>54%</td>
</tr>
<tr>
<td>Female</td>
<td>28%</td>
<td>No</td>
<td>46%</td>
</tr>
<tr>
<td>Age Group</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>18-25</td>
<td>38%</td>
<td>High School</td>
<td>27%</td>
</tr>
<tr>
<td>26-35</td>
<td>45%</td>
<td>Bachelor's Degree</td>
<td>48%</td>
</tr>
<tr>
<td>36-45</td>
<td>17%</td>
<td>Master's Degree</td>
<td>25%</td>
</tr>
<tr>
<td>Experience in Body Painting</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Well Aware</td>
<td>62%</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Minimum</td>
<td>38%</td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

B. Reliability and Validity

The minimum accepted standards for reliability measures in PLS-SEM analysis include a Cronbach's alpha of 0.7, composite reliability of 0.7 or higher, and average variance extracted (AVE) of 0.5 or higher [59]. In this study, all the constructs meet or exceed these minimum standards, indicating good reliability. As per the result Table II, Cronbach's alpha values range from 0.912 to 0.969, composite reliability ranges from 0.921 to 0.976, and AVE ranges from 0.7 to 0.889. These results suggest that the measurement scales used for each construct are internally consistent and reliable, providing confidence in the validity of the research findings.

<table>
<thead>
<tr>
<th>Factors</th>
<th>Cronbach's Alpha</th>
<th>Composite Reliability</th>
<th>Average Variance Extracted (AVE)</th>
</tr>
</thead>
<tbody>
<tr>
<td>Artistic Expression</td>
<td>0.92</td>
<td>0.94</td>
<td>0.70</td>
</tr>
<tr>
<td>Behavioral Intention</td>
<td>0.96</td>
<td>0.97</td>
<td>0.889</td>
</tr>
<tr>
<td>Innovativeness</td>
<td>0.91</td>
<td>0.94</td>
<td>0.782</td>
</tr>
<tr>
<td>Perceived Ease of Use</td>
<td>0.82</td>
<td>0.92</td>
<td>0.7</td>
</tr>
<tr>
<td>Perceived Usefulness</td>
<td>0.91</td>
<td>0.92</td>
<td>0.728</td>
</tr>
<tr>
<td>Social System</td>
<td>0.91</td>
<td>0.93</td>
<td>0.712</td>
</tr>
</tbody>
</table>

C. Discriminant Validity

Discriminant validity is assessed using the Heterotrait-Monotrait (HTMT) ratio of correlations in PLS-SEM analysis. The HTMT values should be below the threshold of 0.85 to confirm discriminant validity [59]. As per the results, all the HTMT values are below this threshold, indicating satisfactory discriminant validity among the constructs. It suggests that the constructs are distinct and measure different aspects of the research variables. In Table III, the values range from 0.162 to 0.754, demonstrating no significant overlap between the constructs, supporting the validity of the measurement model.

D. Outer Loadings

In PLS-SEM analysis, it is generally accepted that outer loadings should be at least 0.7 to 0.8 to demonstrate a strong relationship between latent constructs and their indicators [60]. In this study, see Table IV, all the outer loadings surpass the minimum accepted value, indicating that the research meets this criterion. It confirms that the selected indicators successfully capture and represent the underlying constructs, enhancing the validity and reliability of the research outcomes.

<table>
<thead>
<tr>
<th>Factor</th>
<th>Item</th>
<th>O.L.</th>
</tr>
</thead>
<tbody>
<tr>
<td>Innovativeness</td>
<td>IN1</td>
<td>0.927</td>
</tr>
<tr>
<td></td>
<td>IN2</td>
<td>0.731</td>
</tr>
<tr>
<td></td>
<td>IN3</td>
<td>0.857</td>
</tr>
<tr>
<td></td>
<td>IN4</td>
<td>0.812</td>
</tr>
<tr>
<td></td>
<td>IN5</td>
<td>0.782</td>
</tr>
<tr>
<td>Social System</td>
<td>SS1</td>
<td>0.909</td>
</tr>
<tr>
<td></td>
<td>SS2</td>
<td>0.819</td>
</tr>
<tr>
<td></td>
<td>SS3</td>
<td>0.811</td>
</tr>
<tr>
<td></td>
<td>SS4</td>
<td>0.86</td>
</tr>
<tr>
<td></td>
<td>SS5</td>
<td>0.825</td>
</tr>
<tr>
<td>Perceived Usefulness</td>
<td>PU1</td>
<td>0.917</td>
</tr>
<tr>
<td></td>
<td>PU2</td>
<td>0.815</td>
</tr>
<tr>
<td></td>
<td>PU3</td>
<td>0.813</td>
</tr>
<tr>
<td></td>
<td>PU4</td>
<td>0.835</td>
</tr>
<tr>
<td></td>
<td>PU5</td>
<td>0.857</td>
</tr>
</tbody>
</table>

E. Path Analysis

Based on the structural model, the path analysis shows that all the hypotheses' t-statistics and p-values are accepted, as shown in Table V. It implies a significant relationship between the independent variables (Innovativeness, Social System, Perceived Usefulness, and Perceived Ease of Use) and the dependent (Artistic Expression and Behavioral Intention) variables. The betas associated with each hypothesis represent the strength and direction of the relationships. The overall R-squared value of the model is 0.513, as portrayed in Fig. 2.
through AR technology for body painting, their behavioral intention to adopt and continue using this technology also increases. These findings support the notion that Innovativeness and artistic expression play crucial roles in shaping artists’ behavioral intention to embrace AR technology for body painting.

The results of hypothesis H2 suggest a significant positive relationship between the social system and artistic expression in the context of AR technology for body painting. The beta coefficient of 0.177 indicates that a supportive social system positively influences artistic expression. With a T statistic of 4.755 and a p-value of 0.000, the results provide strong evidence to accept this hypothesis. Therefore, it can be concluded that a favourable social system characterized by societal norms, cultural values, and peer interactions contributes to enhanced artistic expression in the context of AR technology for body painting. Likewise, hypothesis H2a examines the relationship between the social system, artistic expression, and behavioral intention. The results indicate a significant positive relationship, evidenced by a beta coefficient of 0.127, a T statistic of 4.473, and a p-value of 0.000. It suggests that when artists experience a supportive social system that encourages and recognizes their artistic expression through AR technology for body painting, it positively influences their behavioral intention to adopt and continue using it. These findings emphasize the importance of a supportive social environment in fostering artistic expression and the intention to adopt innovative technologies in body painting.

The results of hypothesis H3 indicate a significant positive relationship between perceived usefulness and artistic expression in the context of AR technology for body painting. The beta coefficient of 0.333, a T statistic of 6.221, and a p-value of 0.000 provide strong evidence to accept this hypothesis. It suggests that when individuals perceive AR technology as beneficial for their body painting practices, it positively influences their artistic expression. This finding emphasizes the importance of perceiving the usefulness of AR technology in enhancing the creative process and expression within the field of body painting. Building upon H3, hypothesis H3a explores the relationship between perceived usefulness, artistic expression, and behavioral intention. The results indicate a significant positive relationship, with a beta coefficient of 0.238, a T statistic of 6.292, and a p-value of 0.000. It suggests that when artists perceive AR technology as valuable for their artistic expression in body painting, it impacts their artistic expression and positively influences their behavioral intention to adopt and continue using it. These findings highlight the role of perceived usefulness as a driver of artistic expression and the intention to adopt AR technology in body painting [50].

Similarly, hypothesis H4 examines the relationship between perceived ease of use and artistic expression. The results demonstrate a significant positive relationship, as indicated by a beta coefficient of 0.379, a T statistic of 9.239, and a p-value of 0.000. It suggests that when individuals perceive AR technology as easy to use for body painting, it positively influences their artistic expression. This finding underscores the importance of perceiving the ease of use of AR

Table V. Path Analysis

<table>
<thead>
<tr>
<th>Hypothesis</th>
<th>Beta</th>
<th>T Stat</th>
<th>P</th>
<th>Decision</th>
</tr>
</thead>
<tbody>
<tr>
<td>H1 IN -&gt; AE</td>
<td>0.201</td>
<td>5.156</td>
<td>0.000</td>
<td>Accepted</td>
</tr>
<tr>
<td>H1a IN -&gt; AE -&gt; BI</td>
<td>0.144</td>
<td>4.751</td>
<td>0.000</td>
<td>Accepted</td>
</tr>
<tr>
<td>H2 SS -&gt; AE</td>
<td>0.177</td>
<td>4.755</td>
<td>0.000</td>
<td>Accepted</td>
</tr>
<tr>
<td>H2a SS -&gt; AE -&gt; BI</td>
<td>0.127</td>
<td>4.473</td>
<td>0.000</td>
<td>Accepted</td>
</tr>
<tr>
<td>H3 PU -&gt; AE</td>
<td>0.333</td>
<td>6.221</td>
<td>0.000</td>
<td>Accepted</td>
</tr>
<tr>
<td>H3a PU -&gt; AE -&gt; BI</td>
<td>0.238</td>
<td>6.292</td>
<td>0.000</td>
<td>Accepted</td>
</tr>
<tr>
<td>H4 PEOU -&gt; AE</td>
<td>0.379</td>
<td>9.239</td>
<td>0.000</td>
<td>Accepted</td>
</tr>
<tr>
<td>H4a PEOU -&gt; AE -&gt; BI</td>
<td>0.272</td>
<td>7.581</td>
<td>0.000</td>
<td>Accepted</td>
</tr>
<tr>
<td>H5 AE -&gt; BI</td>
<td>0.716</td>
<td>19.309</td>
<td>0.000</td>
<td>Accepted</td>
</tr>
</tbody>
</table>

Fig. 2. PLS-SEM model.

V. HYPOTHESIS RESULTS

The results of hypothesis H1 indicate that there is a significant positive relationship between Innovativeness and artistic expression in the context of AR technology for body painting. The beta coefficient of 0.201 suggests that higher levels of Innovativeness among artists lead to increased artistic expression in body painting. This finding is supported by a high T statistic of 5.156 and a p-value of 0.000, indicating a strong level of confidence in the results. Therefore, we can conclude that Innovativeness positively influences artistic expression in the context of AR technology for body painting. Additionally, hypothesis H1a examines the relationship between Innovativeness, artistic expression, and behavioral intention. The results show a significant positive relationship, with a beta coefficient of 0.144, a T statistic of 4.751, and a p-value of 0.000. It indicates that as artists demonstrate higher levels of Innovativeness and engage in more artistic expression through AR technology for body painting, their behavioral intention to adopt and continue using this technology also increases. These findings support the notion that Innovativeness and artistic expression play crucial roles in shaping artists’ behavioral intention to embrace AR technology for body painting.
technology in facilitating and enhancing artistic expression in body painting. Expanding on H4, hypothesis H4a investigates the relationship between perceived ease of use, artistic expression, and behavioral intention. The results reveal a significant positive relationship, with a beta coefficient of 0.272, a T statistic of 7.581, and a p-value of 0.000. It implies that when artists perceive AR technology as easy to use for their artistic expression in body painting, it impacts their artistic expression and positively influences their behavioral intention to adopt and continue using it. These conclusions highlight the role of perceived ease of use in promoting artistic expression and the intention to adopt AR technology in body painting [42].

Hypothesis H5 explores the relationship between artistic expression and behavioral intention. The results indicate a highly significant and robust positive relationship, with a beta coefficient of 0.716, a T statistic of 19.309, and a p-value of 0.000. These findings provide strong evidence to accept the hypothesis, suggesting that artistic expression in the context of body painting has a substantial impact on individuals' behavioral intention. Specifically, when individuals engage in artistic expression through body painting, it positively influences their intention to adopt and continue using AR technology. It highlights artistic expression's pivotal role as a behavioural intention driver in adopting and utilizing AR technology for body painting [38].

VI. DISCUSSION

This study aimed to investigate the factors influencing the adoption of augmented reality (AR) technology in body painting and to understand their impact on artistic expression and behavioral intention. A comprehensive analysis was conducted using partial least squares structural equation modelling (PLS-SEM) to achieve this. The results revealed that the proposed model accounted for a substantial proportion of artistic expression and behavioral intention variance. The R-squared values for artistic expression were 0.504 and 0.499 for the regular and adjusted models. Similarly, the R-squared values for behavioural intention were 0.513 and 0.512 for the regular and adjusted models, respectively. These R-squared values indicate that the model explains a significant portion of the variability observed in the dependent variables.

Through examining various hypotheses, several valuable lessons have been learned from the results of this study. Firstly, it is evident that factors such as Innovativeness, social system support, perceived usefulness, and perceived ease of use significantly influence artistic expression and behavioral intention. It highlights the importance of considering these factors when integrating augmented reality (AR) technology into body painting. Furthermore, the positive impact of AR technology on artistic expression and behavioral intention underscores its potential as a powerful tool for artists to enhance their creative processes and engage with their audience. The findings emphasize the significance of embracing technological advancements in the arts and utilizing AR to captivate viewers and create immersive artistic experiences.

Moreover, accepting the hypotheses provides empirical evidence supporting the adoption of AR in the fine arts industry. It suggests that AR has the potential to drive innovation within the industry, offering unique opportunities for artists, galleries, and exhibition spaces to differentiate themselves and attract audiences by leveraging the immersive and interactive nature of AR. Likewise, the results indicate that integrating AR technology in body painting opens up collaboration opportunities between artists and technology experts. It underscores the importance of interdisciplinary collaborations, where artists can collaborate with AR developers, programmers, and designers to explore new creative possibilities and push the boundaries of artistic expression.

The research findings make significant contributions both theoretically and practically.

A. Theoretical Contribution

This study contributes to the theoretical landscape by utilizing Roger's Innovation-Decision Process Theory as a guiding framework. By applying this theory, the research provides a comprehensive and structured approach to understanding AR technology's innovation process in body painting. This theoretical foundation enhances our understanding of the underlying mechanisms that shape the acceptance, adoption, and utilization of AR in artistic expression. Moreover, the study extends the theoretical discourse by highlighting the significant role of artistic expression as a mediator between the adoption of AR technology and behavioral intention. This finding emphasizes the importance of artistic expression as a critical factor in determining artists' intention to adopt and use AR technology in their creative endeavours. By contributing to the theoretical understanding of AR adoption in fine arts, this study fills a gap in the existing literature, which has predominantly focused on the medical, entertainment, aviation, and educational applications of AR. It expands the theoretical boundaries and provides insights into the unique dynamics and implications of incorporating AR technology in the artistic domain.

B. Practical Contribution

Based on the study's findings, the research makes several practical contributions. It guides artists looking to incorporate augmented reality (AR) technology into their body painting practices. It offers insights into factors such as Innovativeness, social system support, perceived usefulness, and perceived ease of use. This knowledge can help artists make informed decisions and leverage AR to enhance their artistic expression. Additionally, integrating AR technology in body painting enhances audience engagement by creating interactive and immersive experiences. This practical contribution allows artists to captivate viewers in novel ways, allowing active participation in the artwork and creating memorable artistic encounters.

The research findings also have implications for art education, suggesting the inclusion of AR in curricula. By recognizing the significance of AR in fine arts, educational programs can prepare students for the evolving art industry and foster their innovative thinking. Moreover, the study highlights the potential of AR technology to drive industry innovations within the fine arts sector. Artists, galleries, and exhibition spaces can leverage AR to differentiate themselves, attract
audiences, and offer unique and immersive experiences. Lastly, the adoption of AR in body painting opens up collaboration opportunities between artists and technology experts. This practical contribution encourages interdisciplinary collaborations, leading to innovative projects and the development of cutting-edge artworks that push the boundaries of artistic expression.

This study aims to contribute to understanding the fusion of augmented reality and body painting as an innovative concept for artistic expression. By examining the factors influencing individuals' acceptance and intention to engage in AR-integrated body painting, the research offers insights for researchers and practitioners in digital art. The findings will extend the existing literature on technology adoption and diffusion by shedding light on the role of perceived usefulness, ease of use, Innovativeness, and social factors in the context of this merged concept. Moreover, the mediating role of artistic expression emphasizes the importance of considering the creative outcomes when exploring the adoption and use of AR-integrated body painting.

C. Limitations of the Study

Despite its contributions, this research has certain limitations that should be acknowledged. Firstly, the study focused on a specific context of body painting, which may limit the generalizability of the findings to other artistic domains. Future research could explore the application of AR in different forms of visual arts to provide a more comprehensive understanding. Additionally, the study relied on self-reported measures, subject to response biases and may not capture the full complexity of participants' experiences. Using objective measures or combining self-reports with observational data could enhance the validity of the findings. Furthermore, the research primarily examined the perspectives of artists and audience members, neglecting other stakeholders such as AR developers and technicians. Future studies could incorporate the viewpoints of these stakeholders to gain a holistic understanding of the challenges and opportunities in AR implementation.

D. Future Directions

Building upon the findings of this study, several avenues for future research can be identified. Firstly, longitudinal studies could investigate the long-term effects of AR integration in body painting and its impact on artists' creative processes, audience engagement, and market sustainability. Additionally, exploring the cultural and societal influences on the adoption and acceptance of AR in fine arts could provide valuable insights into the cross-cultural applicability of the technology. Furthermore, investigating the potential ethical and privacy concerns associated with AR in artistic practices would be relevant in ensuring responsible and inclusive implementation. Additionally, examining the role of different types of AR interfaces, such as wearable devices or projection-based systems, could shed light on the user experience and interaction design aspects. Finally, studying the integration of AR with other emerging technologies, such as artificial intelligence or virtual reality, could open up new dimensions for artistic expression.

VII. Conclusion

In conclusion, this research has significantly contributed to understanding and applying augmented reality (AR) technology within body painting from an HCI and digital technology perspective. By examining the relationships between various factors and their impact on artistic expression and behavioral intention, valuable insights have been gained into the usability, user experience, and practical implications of AR in this context. Theoretical contributions have been made by establishing the importance of factors such as Innovativeness, social system support, perceived usefulness, and perceived ease of use in driving the integration of AR technology in body painting while considering the principles of HCI and digital technology. These findings enrich our understanding of the underlying mechanisms that influence the adoption and utilization of AR in artistic practices, taking into account human-computer interaction, interface design, and user-centred approaches.

Practically, this research offers guidance for artists seeking to incorporate AR technology into their body painting endeavours from an HCI and digital technology perspective. By understanding the positive influence of the identified factors and considering HCI principles, artists can make informed decisions and strategically leverage AR to enhance their artistic expression in a user-friendly and immersive manner. Moreover, integrating AR technology provides an avenue for enhanced audience engagement, allowing artists to create interactive and immersive experiences that captivate viewers and foster a deeper connection with the artwork. From an educational standpoint, the implications are noteworthy as well. This research emphasizes the significance of integrating AR into educational programs, leveraging HCI and digital technology to equip students with the skills and knowledge to navigate the evolving landscape of the art industry. By incorporating AR as a tool for artistic exploration, art educators can foster innovative thinking and prepare students for the future by embracing technological advancements and user-centred design principles.

From an industry perspective, the findings underscore the potential for AR to drive innovation within the fine arts industry, with a particular focus on HCI and digital technology. Artists, galleries, and exhibition spaces can leverage AR to offer unique and immersive experiences, attracting audiences and distinguishing themselves in a competitive landscape. This encourages industry professionals to embrace AR as a means of differentiation, considering HCI and digital technology principles and staying at the forefront of artistic advancements. Lastly, the research highlights the collaboration opportunities that arise from integrating AR in body painting, emphasizing the interdisciplinary nature of HCI and digital technology. Artists can collaborate with technology experts to explore new creative possibilities, harnessing the power of AR to push the boundaries of artistic expression and foster innovative projects. Such collaborations bridge the gap between art and technology, developing cutting-edge artworks and opening up new realms of artistic exploration and digital creativity.


The Essence of Software Engineering Framework-based Model for an Agile Software Development Method

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Faculty of Computer Science, Universitas Indonesia, Depok, Indonesia¹, ², ³, ⁴

Abstract—Agile development’s rapid growth is due to its ability to address complex problems and facilitate a smooth transition from traditional methods. However, no single Agile method can fit every organization, which leads to a lack of adoption guidelines. This triggers this investigation by proposing an Agile development method model based on the Essence of software engineering framework and incorporating the common ground of popular methods such as Scrum, Kanban, Extreme programming, SAFe, Lean, Nexus, Spotify Agile, Scrum of Scrums, and Disciplined Agile. The Essence of software engineering framework provides an approach for organizations to develop software development methods based on common ground or shared understanding among methods. We enhance this approach for Agile methods, resulting in a model to support organizations in developing their Agile methods and practices. Moreover, Design Science Research (DSR) was employed as a methodology to construct the artifact, demonstration, and evaluation. We demonstrated the model in an Agile product development at a national-wide bank in Indonesia. This investigation enhances Agile methods in SWEBOK’s Software Engineering Models and Methods knowledge area, benefiting academics and practitioners. Practitioners can use the model as a reference to implement their Agile projects.

Keywords—Agile; common ground; the essence of software engineering framework; Design Science Research (DSR)

I. INTRODUCTION

At present, the development of the Agile approach is very rapid in organizations [1]. This approach has a significant influence on the growth of businesses and the performance of projects, enabling them to tackle complex problems in an era of rapid disruptions [2]. The concept of Agile innovation teams, aimed at maintaining proximity to customers and swiftly adapting to evolving business conditions, is already well-known to most management levels [3]. Moreover, a captivating study conducted across multiple organizations revealed that companies that cultivate an adaptive Agile culture witnessed revenue growth exceeding the pace multiple times [4]. Therefore, the Agile approach must be scaled up at the enterprise level to handle multiple teams and projects. While organizations are transitioning to Agile, they are also looking to scale up [5]. The scaling approach provides value and benefits to the business operations and supports [6].

The organization could adopt the Agile methods to perform Agile implementation. Although several Agile methods offer the solutions, such as Scrum, Kanban, Extreme programming, SAFe, Lean, Disciplined Agile, Nexus, and Spotify Agile, no single Agile method can fit every organization [7], which leads to a lack of adoption guidelines.

To the best of our knowledge, there is no common ground or shared understanding of the Agile methods or the guideline for the organization to adopt the methods, as shown in Fig. 1. This is the problem we need to solve.

No studies currently explore a model for Agile methods in organizations. The Essence provides a common ground for all methods in software engineering [8]. It offers a thinking framework for the team to collaborate; resources for discussing, improving, comparing, and sharing methods and practices; a foundation for defining practices independent of methods.

This research aims to develop a model for an Agile method development based on selected practices, drawing from the Essence framework. The intended scope of this Agile method is up to scaling Agile, as it already encompasses the Agile method within it. This study serves as an academic reference for Agile methods. From the practitioners’ perspective, the results can guide the organization to apply the scaling Agile methods. Additionally, the model can be integrated with the hybrid agile management approach [9], and DevOps [10] for organizations seeking a hybrid implementation. These contributions make this study unique.

The remaining sections of this article are structured in the following manner: Following this introduction, the subsequent section offers a brief overview of the related work, covering concepts such as the common ground concept, scaling Agile methods, and the Essence framework. Section III explores the research methodology employed, specifically focusing on design science research. Moving on, Section IV presents the study’s findings, which is resulting in the model, namely an Essence-based Agile method development model. Subsequently, in Section V, we present the case study of the software development organization in one bank in Indonesia. Section VI explains the evaluation of our research, respectively. Lastly, Section VII is the concluding section provides a summary and closure to the paper.
II. RELATED WORK

This section discusses related work. First, we review the concept of common ground based on the Essence of software engineering as the reference for this study. Then, we explore the current scaling Agile methods, such as SAFe, LeSS, Nexus, Disciplined Agile, Spotify Agile, and Scrum of Scrums. We focus on the scaling methods, including the individual Agile methods, such as Scrum, Kanban, and Extreme Programming. We only select the most popular scaling Agile methods based on a survey as the main comparison [5, 11]. The other methods will fill the gaps in the practices implemented in the case study.

A. The Common Ground Concept in the Software Engineering

The concept of the common ground in this study refers to the common ground in software engineering introduced by Ivar Jacobson [8]. He introduced the Essence kernel for software engineering. The common ground is developed by thinking that no software development method is appropriate for everyone. The power of the common ground is to provide a common framework for the team to understand the general concept of the software development methods. The common ground is employed for discussing, improving, comparing, and sharing software engineering methods and practices. This approach and inspiration of the common ground in software engineering are used in scaling Agile methods.

B. Scaling Agile Methods

The popular Agile methods, including Scrum [12] and Extreme Programming [13], work well for the small project team. When there is a need to scale up a project at the enterprise level, the complex situation is hard to handle [14]. Project Management Institute [7] categorizes Agile methods as the scaling approach and team method, as demonstrated in Fig. 2. Some thinkers, such as Dean Leffingwell and Scott Ambler, developed several enterprise Agile methods, namely Scaled Agile Framework [15] and Disciplined Agile Delivery [16]. There are no formal names for these methods. The literature declared them as enterprise Agile framework [11], scaling methods and approach [11], scaling framework [5], and scaling Agile methods [17]. The Scaled Agile Framework (SAFe), Large-Scaled Scrum (LeSS), Disciplined Agile Delivery (DAD), and Nexus are some of the popular scaling Agile methods and frameworks [5, 10].

Most scaling Agile methods exercise Scrum, Kanban, and Extreme Programming as part of their team methods. On the other hand, integrating Scrum and Kanban is employed based on the needs [18]. These methodologies offer flexibility and adaptability to different projects. For instance, Scrum provides a structured framework with defined roles and ceremonies. Kanban, on the other hand, is well-suited for projects that require continuous delivery. Research in [19, 20, 21, 22, 23] has been performed on this scaling approach for Agile implementation in organizations.

C. The Essence – The SEMAT Kernel

The Essence is the standard of software engineering that provides a universal language for defining methods and practices [24]. It was established by the Object Management Group (OMG) [25] and received strong support from the Software Engineering Methods and Theory (SEMAT). The Essence was built based on solid theory, proven principles, and best practices. It consists of methods, practices, kernel, and Language. Fig. 3 depicts that methods comprise many practices. In this context, a practice is a repeatable approach to performing activities with a specific objective. The kernel elements are used to elucidate the practices, and these kernel elements are specified based on the Language.

SEMAT was developed with the purpose of addressing certain challenges within today's software engineering domain, characterized by the presence of immature implementation. These issues comprise the absence of a universally acknowledged theoretical foundation, the abundance of numerous methods and their variations, and the disconnect between industry practices and academic research.
The kernel contains three areas, including customer, solution, and endeavor. The Essence defines kernel Alphas as the things to work with. Within the customer domain, the team is required to comprehend stakeholders and potential opportunities. The team should address the requirements and software system concerns in the solution area. There are things to perform in the area of endeavor. They include work, team, and way of working.

D. The Previous Studies

Several studies compared the scaling Agile methods [16, 26], while there was another study [27] that made a method of selecting Agile methods in their project implementation. The study [17] tried to analyze the method differences and similarities for DAD, SAFe, LeSS, Spotify, Nexus, and RAGE, while [26] performed the comparison based on the underlying practices. This study differs from the previous studies, where this study develops the common ground based on the common ground of software engineering [8]. The common practices from [26] can be used in more detail for this research.

III. RESEARCH METHOD

Design Science Research (DSR) was employed as a research method for this study [28], as demonstrated in Fig. 4. The DSR methodology aims to solve the organization's problem by producing artifacts. The process begins with the identification problem, as discussed in Section I.

The next process defines the solution, design and development, demonstration, evaluation, and communication. This study aims to get the common ground from the current scaling Agile methods. The comprehensive literature review from current methods of scaling Agile methods was employed as the baseline for the basic common ground. The previous study performed the same [29, 30]. The Essence SEMAT kernel was the primary reference for developing the common ground [25].

We conducted this research using DSR as the comprehensive research method, from the initial stage of problem identification to the evaluation stage. Each phase of this DSR research is explained in its respective section. The research was conducted in the environment of one of the private banks in Indonesia, as described in Section V. We utilized data and information available at the case study location, including their current methodologies and practices. Research validation methods, as described in Section VI, were employed in this study, including internal and external validation through focus groups and interviews.

IV. RESULT AND DISCUSSIONS

This research provides an Agile development model based on the Essence of software engineering. This section is part of the process in the DSR that defines and develops artifacts. The model contains the common ground or shared understanding among the popular Agile and scaling Agile methods discussed in the previous section.

A. An Essence-based Agile Method Development Model

We develop a model for an Agile software development method based on the Essence in Fig. 5. The model was derived from the Essence methods architecture in Fig. 3. The organization can develop its own Agile methods based on the selected Agile practices. We emphasize the scope of the Agile methods using the scaling Agile methods, comprising the Agile team methods, such as Scrum, Kanban, and extreme programming. For example, the organization develops its practices by adopting the practices such as Scrum practice, Spotify practices for the organization, and Kanban for the workflow. They also can adopt the Scrum of Scrum practice for coordination. The organization can also implement some practices from Extreme Programming for pair programming and code ownership.

B. The Common Ground of Scaling Agile Methods

We establish a common ground for scaling Agile methods, as scaling Agile methods already include Agile methods at the team level, such as Scrum, Kanban, and Extreme Programming, which are widely used by organizations. This study proposed the common ground in Fig. 6, which is classified by principles, process, work product, organization, and implementation approach.

At the minimum level, the scaling Agile team only contains the Scrum and integration teams. LeSS, Nexus, Scrum of Scrum have similar project team structures with it. Nexus has the following roles: The Nexus Integration Team Member, Nexus Integration Team Scrum Master, Nexus Integration Team Product Owner, and Nexus Integration Team itself [31].

The common ground of scaling the Agile method contains the Agile practice. It refers to the level of Agile implementation [7]. Team Agile is the Agile practice and method at the team level, such as Scrum, XP, Kanban, FDD, and DSDM. Enterprise agility contains a larger Agile team in the organization collaboratively together to develop a single large product. On the other hand, business agility embraces an Agile mindset and principles that encompass all areas, extending beyond product development to encompass domains like personnel management, servant leadership, design of organization, and financial planning [13]. This study refers to the previous research and literature on the general practice of Agile [7, 32, 33, 34].
Table I summarizes the common ground for popular scaling Agile methods, SAFe, LeSS, DAD, Nexus, Spotify, and Scrum of Scrums. We classify the practices based on the common ground: principle, process, organization, work product, and implementation approach.

C. The Guidance to use the Model

An organization can implement the Agile method using the guidance in Fig. 7 adopted from [35]. The model, built on current theories and practices, allows practitioners to enhance their work by adopting newer practices. The common ground is a guideline for selecting suitable scaling Agile tools and practices. It can also be expanded as needed for project implementation.
### TABLE I. THE SUMMARY OF COMMON GROUND FOR SCALING AGILE METHODS

<table>
<thead>
<tr>
<th>Principles</th>
<th>LeSS</th>
<th>DAD</th>
<th>Nexus</th>
<th>Spotify</th>
<th>Scrum of Scrum</th>
</tr>
</thead>
<tbody>
<tr>
<td>All methods follow the Agile Manifesto for mindset, values, and principles. SAFe, LeSS, and DAD enhance Agile Manifesto to their methods based on the scaling Agile need. Other methods don't specifically define their principles.</td>
<td>Principles</td>
<td>Principles</td>
<td>Follow the Agile manifesto and Scrum value</td>
<td>Follow the Agile manifesto</td>
<td>Follow the Agile manifesto and Scrum value</td>
</tr>
<tr>
<td>Process</td>
<td>All methods have their process. They utilize Scrum for the Agile team. SAFe use Scrum, Kanban, DevOps, and XP. LeSS, Nexus, Spotify, and Scrum of Scrum mainly utilize Scrum for their methods. DAD utilizes Scrum but incorporates more Agile methods.</td>
<td>Disciplined Agile Delivery as the bottom process, Disciplined DevOps, Disciplined Agile IT (DA, and Disciplined Agile Enterprise (DAE).</td>
<td>Following the Scrum framework with the integrated elements</td>
<td>Following the Scrum and enhancing the methods</td>
<td>Scrum, Scrum of Scrum</td>
</tr>
<tr>
<td>Implementation approach</td>
<td>The implementation scope starts from the single project up to the enterprise level.</td>
<td>Essential SAFe, Large Solution, Portfolio, Full</td>
<td>LeSS, Less Huge</td>
<td>DAD as the bottom process, Disciplined DevOps, DAiT, and DAE.</td>
<td>Single product development</td>
</tr>
</tbody>
</table>

![Fig. 7. The guidance for implementation.](image)

V. CASE STUDY

The case study is a part of the demonstration process in the DSR. It was conducted at a national bank in Indonesia. It was actively engaged in a digital transformation initiative to develop a business ecosystem and a digital ecosystem framework based on digital technologies (refer to Table II). The project charter planning document expressed a desire for the application to drive an increase in the Current Account Saving Account (CASA) by USD 1.3 billion. However, despite observing a rise in transaction volume from 2020 to 2021, the cumulative transaction volume only amounted to USD 282,000 by July 2021. Consequently, this discrepancy posed a significant challenge as the target in CASA through application implementation was not attained.
TABLE II. ADOPTED PRACTICES FOR THE CASE STUDY

<table>
<thead>
<tr>
<th>The Common Ground</th>
<th>Current Practices</th>
<th>Proposed Practice</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Principles</strong></td>
<td>- Currently, the company already has a procedure guide in the form of a Circular of Information Technology Workflow Procedures, which regulates the Agile methodology</td>
<td>- Apply and explore the principle of SAFe and LeSS for organizations (SAFe dan LeSS) - Spotify - squad, tribes, chapter, guild</td>
</tr>
<tr>
<td><strong>Organization</strong></td>
<td>Roles: Tribe Sponsor Product, Tribe Sponsor Technology, Tribe Leader Product, Tribe Leader Technology, Product Owner, Scrum Master, Programmer, Quality Control/Tester, Business Analyst, System Analyst, QA Analyst, Document Writer, Application Security Engineer, Manajemen Risiko, Operation Engineer, Subject Matter Expert (UI/UX), Subject Matter Expert (Data Analyst), Subject Matter Expert (Arsitektur IT)</td>
<td>- Adopted SAFe-scaled Agile principles (SAFe) - Implementing the role of a quad, tribes, chapter, and guild as a whole in the Agile process (Spotify) - Implementation of a team based on the Whole Team (XP)</td>
</tr>
<tr>
<td><strong>Business Requirement</strong></td>
<td>- Business Requirements are explained in the Product Roadmap activity - Development needs are stated in the Business Model Canvas (BMC) document - Furthermore, BMC is discussed in the Discovery Session activity</td>
<td>- Application of product backlog understanding to the entire team through Program Increment Planning (SAFe) - Adopt product backlog from Scrum, Sprint Planning, Definition of Ready, and Definition of Done - Using visualize tools</td>
</tr>
<tr>
<td><strong>Process</strong></td>
<td>- Currently, the scrum practice is not fully implemented, which is only sprints and scripts - Other scrum practices, such as sprint review and sprint retrospective, are not implemented</td>
<td>- The discipline of doing Daily Scrum - Implementation of sprint backlog activity, output backlog refinement - Adopt Joint Product-level Sprint Review and Joint Retrospective</td>
</tr>
<tr>
<td><strong>Implementation Approach</strong></td>
<td>- Project level. Tribe leader product, and tribe leader technology actively coordinate - Program level, The development team does not focus only on developing one application but parallel developing other applications - Business level, The business team doesn't focus on just one product developer</td>
<td>Adopt LeSS Practices - Implementation of a team-based organization - Cross-functional theme - Not implementing the &quot;Resource-Allocation&quot; phase</td>
</tr>
</tbody>
</table>

A. Analysis of Current Condition

From the interview analysis and source documents, the researchers focused on a root problem in Agile software development projects: the application of Agile practices that were not fully implemented. The organization needed to develop its practices to improve the software development delivery process. Current practices were not appropriately defined.

We adopted the Essence-based Agile Model (Fig. 5) and mapped its components. They included common ground, principles, organization, business requirements, process, work products, and implementation approach. For the principles, Agile provisions had not been fully implemented following the procedure guidelines. There were vital responsibilities, performance indicators, and authority on the Agile team which had not been implemented. There was an Agile implementation process that did not use supporting tools, namely Jira and/or Confluence. There were Scrum practices that were not implemented.

B. Proposed Practices

As per the Essence guidance to develop the team's methods, we selected the proper practices from the popular Agile methods, such as Scrum, SAFe, Nexus, and LeSS, for our methods. We mapped the nominated practices to the common ground component.

VI. Evaluation

This section describes the model's validation process and the case study's implementation. It is part of the evaluation process in the DSR. The evaluation process was carried out on the practices proposed in the organization to ensure that these practices would be suitable for implementation in a methodology. The evaluation involved 16 internal and external stakeholders through a process of Focus Group Discussion (FGD) or interviews, as well as filling out questionnaires. The questionnaire used a Likert scale with five measurements: (1) Strongly Disagree (2) Disagree (3) Neutral (4) Agree (5) Strongly Agree.

A. Internal Evaluation

We involved eleven Agile team members: a leader, a scrum master, and nine team members. In the internal evaluation process, one-to-one interviews were conducted with the team leader. Besides, a questionnaire was filled out by all Agile team members regarding appropriate practices. Each proposed practice was asked for confirmation using a Likert scale. The question was, "Are the principles of practice practice applicable?". The results of distributing the questionnaires are shown in Fig. 8.

B. External Evaluation

Five experts with experience in Agile implementation participated in the external evaluation using a one-to-one interview and filling out a questionnaire. The experts' profiles participating in the expert judgment process are described in Table III.
The results of internal and external evaluations were carried out by analyzing the interviews and questionnaires. The transcripts from interview data were analyzed using the Dedoose tool, while the Likert scale questionnaire data were analyzed with Microsoft Excel.

C. Internal Questionnaire Data Analysis and Processing

Results from the questionnaires distributed to the Agile team and the experts are demonstrated in Table IV. An average calculation was carried out, and the overall evaluation value was obtained. It was agreed that the practice value was above 3 to apply.

![Evaluation result](image)

Fig. 8. Evaluation result.

<table>
<thead>
<tr>
<th>Code</th>
<th>Profession</th>
<th>Experience</th>
<th>Specialization</th>
<th>Certification</th>
</tr>
</thead>
<tbody>
<tr>
<td>N2</td>
<td>Consultant</td>
<td>17 Years</td>
<td>Project Management, Agile Software Development, DevOps</td>
<td>Certified DevOps, ITIL, Certified Agile Coach, Scaled Agile (SAFe) Agilist v5.1, SCRUM Master (CSM), Certified SCRUM Professional (CSP), Certified Kanban System Design (KMP-1), COBIT 5 Foundation, dll</td>
</tr>
<tr>
<td>N3</td>
<td>Senior Manager</td>
<td>17 Years</td>
<td>Project Management, Agile Software Development, Business Analyst</td>
<td>Scrum Master Certified (SMC), Project Management Professional (PMP)</td>
</tr>
<tr>
<td>N4</td>
<td>Assistant Manager</td>
<td>9 Years</td>
<td>Agile DevOps, Quality Assurance, Agile Software Development</td>
<td>Scrum Master, DevOps Foundation, Quality Management System (ISO 9001:2015)</td>
</tr>
</tbody>
</table>
TABLE IV. THE QUESTIONNAIRES RESULTS

<table>
<thead>
<tr>
<th>No</th>
<th>Selected Practice</th>
<th>Agile Team</th>
<th>Experts</th>
<th>Overall</th>
</tr>
</thead>
<tbody>
<tr>
<td>1.</td>
<td>Sprint Planning</td>
<td>4.73</td>
<td>4.75</td>
<td>4.74</td>
</tr>
<tr>
<td>2.</td>
<td>Daily Scrum</td>
<td>4.45</td>
<td>4.75</td>
<td>4.60</td>
</tr>
<tr>
<td>3.</td>
<td>Sprint</td>
<td>4.55</td>
<td>4.75</td>
<td>4.65</td>
</tr>
<tr>
<td>4.</td>
<td>Sprint Review</td>
<td>4.73</td>
<td>4.75</td>
<td>4.74</td>
</tr>
<tr>
<td>5.</td>
<td>Sprint Retrospective</td>
<td>4.45</td>
<td>4.5</td>
<td>4.48</td>
</tr>
<tr>
<td>6.</td>
<td>Product Backlog</td>
<td>4.27</td>
<td>4.75</td>
<td>4.51</td>
</tr>
<tr>
<td>7.</td>
<td>Backlog Refined</td>
<td>4.27</td>
<td>4.75</td>
<td>4.51</td>
</tr>
<tr>
<td>8.</td>
<td>Sprint Backlog</td>
<td>4.27</td>
<td>4.75</td>
<td>4.51</td>
</tr>
<tr>
<td>9.</td>
<td>Increment</td>
<td>3.64</td>
<td>4.5</td>
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<td>Joint Retrospective</td>
<td>4.18</td>
<td>4.25</td>
<td>4.22</td>
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</table>

D. Internal Analysis and Processing of Interview Data

Interview data was processed by compiling transcripts of recorded interviews or FGDs with the Agile team and one-to-one interviews with experts into a text file. Files containing interview transcripts were uploaded to the Dedoose software. Next, a coding process was carried out for the 24 elections by marking the code from the results of the interview transcription in the relevant section. Excerpts of the codes revealed practices and methods mostly emerge from interviews. Next, a new Software Development Methodology Design was created from the selected practices, proposed practices, and methods.

The selection of these practices has been adapted through one product development sprint cycle, with the outputs of each stage described in Table V. A retrospective process was continued to review the practices and make continuous improvements. Therefore, the team can develop even better methods.
<table>
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<td>Business Model Canvas (BMC), Product Backlog</td>
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<td>Product Backlog</td>
<td>Scrum</td>
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<td>Visualize</td>
<td>Kanban</td>
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<td>Scrum</td>
<td>Sprint Backlog, Kanban board</td>
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<td>Scrum</td>
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<td>Visualize</td>
<td>Kanban</td>
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</tr>
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<td>Scrum</td>
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<td></td>
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<td>Scrum</td>
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<td>Backlog Refinement</td>
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<td>Spotify</td>
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<td>Scrum of Scrums</td>
<td>Scrum</td>
<td>Sprint Backlog</td>
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</table>

**VII. Conclusion**

This study aims to develop a model for Agile methods based on the Essence of software engineering framework. The resulting model is a significant outcome of this research, as it encompasses the common ground or shared understanding of scaling Agile practices from well-known methods such as SAFe, LeSS, Nexus, Disciplined Agile, Spotify Agile, and Scrum of Scrums. The intended scope of this Agile method is up to scaling Agile, as it already encompasses the Agile method within it. The common ground includes the principle, business requirement, process, work product, organization, and implementation approach. By following this model, organizations can benefit from valuable guidance in developing their Agile methods, as outlined below:

- Organization can define their initial practices from the current Agile methods
- The common ground can classify the practices to group the practices for more manageable to maintain
- The sample implementation was demonstrated in the case study

The model was implemented in the Agile product development in a national bank in Indonesia. Agile experts participated in reviewing the model. The selected practices, as based on the model, were evaluated during the focus group discussion and interview with the relevant stakeholder in the case study. The model of this study and the common ground can be implemented in the other organization to develop their methods in their Agile development project.
This study has significant implications for both academic research and practical applications. In terms of academic research, it introduces new literature concerning the scaling of Agile methods. A notable contribution is the Essence-based model, which establishes a shared foundation in the realm of scaling Agile. For practitioners, this study enhances their understanding through relevant case studies. The model can serve as a valuable reference for practitioners to make informed decisions, customize, and adapt Agile methods to suit their organizational needs.

Further research can be explored to enhance the model into a more detailed framework. Another study to build the framework can be performed with extensive expert judgment and more case studies. The framework can include more practices and specifications as supporting references.

As the model was derived from the content analysis from relevant references, it may have limitations in actual practices. The grounded theory may complement this model. Other studies in the same field may have different perspectives for building the model.

ACKNOWLEDGMENT

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Effective Face Recognition using Adaptive Multi-scale Transformer-based Resnet with Optimal Pattern Extraction

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Abstract—Human face is the major characteristic for identifying a person and it helps to differentiate each person. Face recognition methods are mainly useful for determining a person’s identity with the help of biometric techniques. Face recognition methods are used in many practical applications like criminal identification, the phone unlocks systems and home security systems. It does not need any key and card, and it only requires facial images to provide high security over several applications. The interdependencies of the encryption methods are highly reduced in the deep learning-enabled face recognition models. Conventional methods did not satisfy the present demand due to poor recognition accuracy. Therefore, an advanced deep learning-based face recognition framework is implemented to authenticate the identity of individuals with high accuracy by using facial images. The required facial images are taken from the standard databases. The collected images are preprocessed using median filtering. The preprocessed facial images are subjected to spatial feature extraction, where the Local Binary patterns (LBP) and Local Vector Patterns (LVP) are utilized to extract the relevant optimal patterns from the facial images. Here, optimal pattern extraction is done with the Improved Rat Swarm Optimization Algorithm (IRS-O). Then, the facial recognition is done over the extracted optimal features with the usage of the implemented Adaptive Multi-scale transformer-based Resnet (AMT-ResNet), where the parameters in the recognition network are optimized by using the IRS-O. The efficiency of the developed deep learning adopted face recognition model is validated through different heuristics algorithms, and baseline face recognition approaches.

Keywords—Face recognition; facial images; optimal pattern extraction rate; local binary patterns; local vector patterns; improved rat swarm optimization algorithm; adaptive multi-scale transformer-based Resnet

I. INTRODUCTION

The individuals have been recognized with the help of emerging biometric-based methodologies in the current situation. These biometric-based techniques compute the behavioral and individual’s physiological characteristics to ascertain and determine their identity [17]. It exactly identifies the individuals instead of performing people authentication and giving permission for those particular individuals to access the physical domains with the support of utilizing smart cards, plastic cards, PINs, keys, passwords, and tokens [15]. But, it is hard to remember the PINs and passwords, and also, it is a possibility to guess and steal tokens, keys, and cards very easily [7]. In most cases, the magnetic cards of the individuals are missed, and then it can be misplaced, duplicated, purloined, and forgotten by intruders; hence, it becomes unreadable and corrupted [24]. Using biometric-based techniques, an individual’s biological traits cannot be stolen, forgotten, forged, or misplaced [2]. When compared to other biometric approaches, face detection is one of the fastest biometric approaches and the least intrusive technique that includes iris recognition, fingerprint recognition, and face recognition [16]. Consider an example, instead of placing an individual’s hands in a reader in surveillance systems, the people are requested to place their eyes in front of a scanner, which is generally known as iris recognition [30]. But, the face recognition systems unobtrusively capture images of individuals’ faces when they are reached in a predefined area. In these face recognition systems, there is no capture delay and intrusion, and the subjects in the face recognition systems are entirely unaware of the process [18].

Face recognition approaches are accomplished several substantial attentions from the market and research communities that are the capability of dealing with real-world problems related to facial images [33]. In general, the face recognition approach is formulated as follows: from a given image or video of a scene, a person is verified and identified from a stored database of faces [29]. Before the detection process, the verification and identification were made by taking the similarity measure among two face images and calculating non-match or match pixel coefficients [6]. Moreover, during the identification process, the similarity measure between the collected face image and face images in the original database is examined to reduce the misclassification to determine the person’s identity [21]. Face recognition is natural, convenient, and nonintrusive then it is helpful in a wide range of applications. But, several face recognition approaches encounter great challenges when recognizing the person’s identity from spatial images due to the variations in illumination, background clutter, facial pose, image resolutions, and expressions [25]. Then, several artificial intelligence-based face recognition approaches are introduced that have accomplished great success over face recognition.
Deep learning techniques are assumed as the greatest breakthrough over face image recognition because of their powerful learning capacity. The hand-crafted features and the discriminative features are effectively learned through deep learning techniques [13]. The deep face-related face recognition techniques improved the recognition accuracy, but it slightly suffered from degradation in performance when encountering the large-scale dataset. To support effective face recognition in real-world scenarios, fine-tuning the pre-trained models are important [38]. Moreover, the labeling of large-scale datasets is difficult while using these deep learning-based face recognition approaches [37]. Furthermore, the existing deep learning-based techniques are well-suited for limited modalities, but the complementary features present in the much more modalities are not well descriptive [9]. Therefore, an efficient face recognition model with the support of deep learning techniques has been designed to provide extensive results over face recognition. The main contribution of the developed deep intelligence-based face recognition framework is described below:

- To design an intelligent deep learning-based face recognition framework with the help of a heuristic strategy to identify the person’s identity using their face and boosting security.
- To implement an IRSO in the developed face recognition model for the selection of optimal patterns and optimizing the hidden neuron count in adaptive multi-scale transformer-based ResNet for increasing the recognition accuracy of the implemented framework.
- To introduce an effective optimal pattern extraction approach from the LBP and LVP patterns by using the developed IRSO in the developed face recognition model to enhance the performance of the developed framework.
- To present an adaptive multi-scale transformer-based ResNet structure with parameter optimization via the developed IRSO for recognizing the faces very sensitively that is helpful to attain a higher recognition rate under varying illumination conditions, expressions, pose, and background.
- To validate the efficacy of the newly suggested face recognition model by comparing the experimental results among several optimization strategies and baseline face recognition approaches following different positive and negative measures.

The remaining sections used to develop the new deep learning-based face recognition approach are explained as follows. Section II gives the recently implemented deep learning-based face recognition approaches with their merits as well as demerits. Section III illustrates the structural description of the newly implemented face recognition model, the function of median filtering in the developed model, and the description of the database. Section IV enumerates the optimal solution obtained using the developed IRSO and the LBP and LVP pattern extraction process. Section V gives a detailed explanation of the developed multi-scale transformer-based ResNet for the detection of faces and its objective function. The comparative analysis and the experimental results are given in Section VI, and the conclusion of the face recognition approach is summarized in Section VII.

II. LITERATURE SURVEY

A. Related Works

In 2021, Michael et al. [12] facial expression identification approach with the help of deep learning algorithm, which utilized Convolutional Neural Network (CNN) for learning the parameters from the face images. Here, the FER2013 dataset has been used to obtain the sample face images. Various patterns from the face images were retrieved from the original images, and the experimental results were compared over various approaches. The computational cost of this method has been highly reduced by using this approach. The recognition accuracy of the implemented model has been highly improved to baseline approaches.

In 2019, Ranjan et al. [27] have introduced a simultaneous face detection algorithm for localizing landmarks, recognizing genders, and estimating poses using deep CNN. Here, the multi-task learning algorithm has been implemented to fuse the intermediary layers to provide the fused features in deep CNN from the face images. Hence, the synergy among the tasks has been greatly exploited, and it improved the performance individually. Moreover, the HyperFace-ResNet and Fast-HyperFace network has been demonstrated to enhance the speed of the face identification process. The implementation results of the developed model have the ability to capture the local and global information in faces, and it provided very effective results than the other approaches.

In 2018, Ashijay et al. [1] have introduced a robust and efficient method to recognize individual faces in real-time. The filters have been applied to the collected sample images to eliminate unwanted features and noises. From the filtered images, the binary patterns were retrieved, and the resultant patterns were given to the Multilayer Perceptron (MLP) for recognition purposes. The extensive test results were compared over the benchmark datasets like FACES96, FACES95, and FACES94 in terms of expression, rotation, pose, illumination, and head scale. The test results gave better results and showed the developed face recognition model to be more highly efficient than the benchmark datasets.

In 2022, Junaid et al. [23] have studied the performance of the recently implemented deep learning-based Disguise Invariant Face Recognition (DIFR) method, which integrated the data augmentation method for removing noises. The individual faces were effectively recognized by the Viola Jones face detector and the classification was done with the support of CNN. The disguise-invariant features have been effectively learned from the facial images for identifying the person from the face images. Finally, the comparative analysis and the comprehensive experiments have been conducted over six different datasets, from that the Resnet-18 has given a good trade-off than the other face recognition approaches in terms of efficiency, accuracy and average execution time.

In 2019, Zhao et al. [19] have introduced a face identification method via the Generative Adversarial Network (GAN) with a dual agent system, which identified the
unlabeled patterns from the face images, and the fine details of the images were highly preserved by using this model during the realism refinement. The real and face identities were effectively determined by using this dual-agent system. Moreover, the profile face images have been generated with the support of off-the-shelf 3D face models with varying poses. By using the fully convolution approach high-resolution images have been generated. The poses were sensitively preserved, and the stability of the mechanism has been improved by analyzing the experimental results over conventional face recognition datasets.

In 2018, Al-Waisy et al. [3] have developed a Deep Belief Network (DBN) for extracting local handcrafted features to address face recognition problems, which has been highly suitable for unconstrained conditions. Initially, the Fractal dimension was merged with the Curvelet transform to represent the main structure of the face that included both curves and edges. Then, the pixel intensity representations were replaced with the local feature representations with the help of the DBN structure for the multi-modal recognition of facial images. At last, the effectiveness of the developed DBN-based facial recognition approach was evaluated by performing several extensive experiments over different datasets such as LFW databases, CAS-PEAL-R1, FERET, and SDUMLA-HMT. The developed DBN-based face recognition approach outperformed well than the other baseline frameworks with respect to recognition accuracy and time complexity.

In 2022, Durga and Rajesh [5] have offered a deep micro-facial emotion recognition approach with the support of CNN. Here, the 2D-ResNet CNN model has been deployed to extract the Multi-class features from the collected face images. Moreover, the developed approach efficiently detected the maskable images from the face images via the 2D-ResNet CNN. This developed approach has been evaluated in Python 3.7.0 software tool, and the 2D-ResNet CNN network has been trained using the public dataset. Furthermore, the jsonify python, fastai, mysql, and flask packages were highly feasible and compatible in the developed architecture. The extensive outcome was shown that the developed approach had proven its efficiency concerning the performance metrics like sensitivity, accuracy, F1-score, and recall.

In 2015, Gao et al. [28] have proposed a supervised auto encoder for the recognition of individuals from the collected face images. The auto encoder was a type of new deep architecture, and it effectively learned the hidden patterns from the raw images. Initially, the mapping operation was performed between the canonical face of the person and the face variants from the images. An example of this mapping operation was the integration of normal illumination into neutral expressions. Then, the auto encoder retrieved the features that were robust to the variances of the pose, occlusion, expression, and illumination. After, the deep architecture was obtained over the supervised auto encoders, which have been utilized for the extraction of relevant features in image representation. Several experiments have been conducted over the newly implemented auto encoder-based face recognition approach, which has considered the benchmark datasets like Multi-PIE, Extended Yale B, AR, and CMU-PIE to ensure the efficacy of the developed face identification approach. The test results have shown that the stacked supervised auto encoder-based face representation has outperformed well than the other benchmark datasets.

B. Problem Statement

The face recognition approaches are mainly adopted for numerous applications like internet communication, surveillance, and security. The conventional approaches over face recognition provide satisfactory performance only under controlled scenarios. But, in real-world scenarios, the performance of the model is highly degraded under several conditions like occlusion, expressions, pose variations, and illuminations. Therefore, several supervised learning-based face recognition models are employed to provide better results over face recognition. The features and drawbacks of the developed deep learning-enabled face detection approaches are given in Table I. CNN [12] provides better results over the interpretation of facial expressions and facial recognition. And also, it highly decreases the computational cost due to the development of facial detection. But, it slightly suffers with generalization capability. In addition, it increases the instability and internal covariance, and hence the overfitting problem cannot be avoided. DCNN [27] has the capability to capture both local information and global information in faces. Further, the multi-task information is efficiently reduced to enhance the effectiveness of the recognition. Yet, the topological changes due to the viewpoint variations are not captured effectively in this approach. Consequently, it does not identify the discriminative facial information at various illumination conditions. Viola Jones detector with CNN [1] gives high efficiency and robustness in face recognition. Moreover, it captures the most crucial information from the facial images, and hence, the accuracy of the recognition process is increased. Yet, it is computationally expensive and impractical in darkness condition. Nevertheless, it is a very time-consuming process. In 2D-CNN [23], the training loss that occurred in the systems is highly reduced. Furthermore, it is highly robust for noise reduction. But, it does not meet the real-time requirements. In addition, the availability and feasibility of the model are low. GAN [3] needs less training data without the need for any domain information. Moreover, the resolution of the images is getting increased. Yet, it does not solve the generalized Eigenvalue problems. Furthermore, it does not provide any authenticity over face recognition. ResNet [5] utilizes sparse symbolic representation to symbolize the gradients in the direction domain. Consequently, the artifacts present in the model are highly reduced. Nonetheless, it does not give additional information on convergence. For instance, it needs hardware equipment to recognize face patterns. Auto encoder [28] increases the generalization ability under occlusion conditions. Subsequently, it highly reduces the low-rank error during face recognition. However, it requires more training samples to improve the scalability and availability of the face recognition system. Moreover, it has the capability to hold less amount of information. These drawbacks that arise in the deep learning-enabled face recognition approaches are resolved by the newly introduced face recognition model.
TABLE I. FEATURES AND CHALLENGES OF THE CONVENTIONAL DEEP LEARNING-BASED FACE RECOGNITION MODELS

<table>
<thead>
<tr>
<th>Author (Citation)</th>
<th>Methodology</th>
<th>Features</th>
<th>Challenges</th>
</tr>
</thead>
</table>
| Michael et al. [12] | CNN | • It provides better results over the interpretation of facial expressions and facial recognition.  
• It highly decreases the computational cost due to the implementation of facial recognition. | • It slightly suffers with generalization capability.  
• It increases the instability and internal covariance, and hence the overfitting problem cannot be avoided. |
| Ranjan et al. [27] | DCNN | • It has the capability to capture both local and global information in faces.  
• Multi-task information is efficiently reduced to enhance the performance of the recognition. | • The topological changes due to the viewpoint variations are not captured effectively in this approach.  
• It does not identify the discriminative facial information at various illumination conditions. |
| Abhijay et al. [1] | Viola Jones detector with CNN | • It gives high efficiency and robustness in face recognition.  
• It captures the most crucial information from the facial images, and hence, the accuracy of the recognition process is increased. | • It is computationally expensive and impractical in darkness conditions.  
• It is a very time-consuming process. |
| Junaid et al. [23] | 2D CNN | • The training loss that occurred in the systems is highly reduced.  
• It is highly robust for noise reduction. | • It does not meet the real-time requirements.  
• The availability and feasibility of the model are low. |
| Zhao et al. [19] | GAN | • It needs less training data without the need for any domain information.  
• The resolution of the images is getting increased. | • It does not provide better results over face recognition due to the factors like various illuminations, same face, aging, and pose variations. |
| Al-Waisy et al. [3] | DBN | • It gives more accuracy and scalability over face recognition.  
• It effectively captures the shape variations that are irrespective of the illumination variabilities. | • It does not solve the generalized eigenvalue problems.  
• It does not provide any authenticity over face recognition. |
| Durga and Rajesh [5] | ResNet | • It utilizes sparse symbolic representation to symbolize the gradients in the direction domain.  
• The artifacts present in the model are highly reduced. | • It does not give additional information on convergence.  
• It needs hardware types of equipment for recognizing face patterns. |
| Gao et al. [28] | Auto encoder | • Under occlusion conditions, it increases the generalization capability.  
• It highly reduces the low-rank error during face recognition. | • It requires more training samples to improve the scalability and availability of the face recognition system.  
• It has the capability to hold less amount of information. |

III. ARCHITECTURAL VIEW OF THE DEVELOPED FACE RECOGNITION MODEL

A. Dataset Description of Face Recognition

The face images required for the identification of persons are collected from standard online databases. The detailed descriptions of the databases are given as below.

Dataset 1 (CFPW Dataset): This CFPW dataset is obtained from the publically available online source of “http://www.cfpw.io/ access date: 2022-12-07”. This dataset contains celebrity photos from around the world. The images are collected in profile view and frontal view. More images of celebrities are added to the database.

Dataset 2 (Yale Dataset): The Yale dataset is available in the online source of “http://vision.ucsd.edu/content/yale-face-database: access date: 2022-12-07”. Total 15 people's face images are there in the Yale dataset. It contains 165 images with various facial expressions of persons that includes sad, surprised, normal, wink, with or without glasses, and center-light, left-light as well as right-light.

The collected face images are indicated as $F_{I_g}$, where $g = 1, 2, 3, \ldots, G$, and the term $G$ denotes the total number of sample images. The collected sample face images from two different datasets are given in Fig. 1.

<table>
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<th>Face Image Description</th>
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<td><img src="sample2.png" alt="Image" /></td>
<td><img src="sample3.png" alt="Image" /></td>
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<td><img src="sample5.png" alt="Image" /></td>
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<tr>
<td>Sample face images from Dataset 2</td>
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<td><img src="sample7.png" alt="Image" /></td>
<td><img src="sample8.png" alt="Image" /></td>
<td><img src="sample9.png" alt="Image" /></td>
<td><img src="sample10.png" alt="Image" /></td>
</tr>
</tbody>
</table>

Fig. 1. Sample face images required for the face recognition process.
B. Proposed Face Recognition Model

Face detection is one of the most difficult tasks in image retrieval and computer vision. Because the face recognition approaches are mainly applicable in various domains like driving license systems, healthcare systems, railway reservation systems, ATMs, passport authentication, and surveillance operation. While processing through the large-scale dataset, the face recognition approaches lack in their performance for system reliability, scalability, and robustness. During face recognition, the collected face image correlates with the face image from the database. But, if the query image is unavailable in the database, then it evaluates the similarity between the two images. This leads to an increase in the false identification of persons. Therefore, deep learning-based approaches are adopted for face recognition, where the error rate is highly reduced, and the recognition accuracy is improved. But, it faces several difficulties while recognizing the images in motion images, pose variations, and twins. In addition, persons with different hair colors, makeup, light intensity condition, noises, different illumination condition, hair partitions, beards, and other occlusions are difficult to identify the exact person. Hence, a new deep learning-based face recognition model is developed with the optimization algorithm to solve these disadvantages, and the architectural representation of this newly developed model is given in Fig. 2.

A new deep learning-based face recognition model is designed with the help of a heuristic algorithm for replenishing the need for security and avoiding present-day crime. The wanted data is gathered from the CFPW and Yale databases. The collected face images are filtered through median filtering to remove the unwanted noises, then subjected to LBP and LVP pattern extraction. The optimal patterns are selected from the retrieved LBP and LVP patterns and placed in a new window. This selection process is carried out with the support of the developed IRSO, increasing the recognition accuracy during face recognition. These extracted optimal patterns are given to the efficient to perform the face recognition. Here, the hidden neuron count in the ResNet is optimized with the utilization of the implemented IRSO for improving the accuracy of the developed face recognition model. The test outcomes obtained from the newly implemented face recognition model are guaranteed with the conventional deep learning-based face recognition model and various heuristic algorithms. The ablation accuracy and convergence analysis in terms of the cost function are carried out in the developed face recognition model for validating the effectiveness of the developed IRSO–AMT-ResNet-based face recognition model.

C. Median Filtering of Input Images

The collected face images $F_{Ig}$ are given to the preprocessing stage for the eradication of noises and unwanted blurs present in the images. In the preprocessing stage, median filtering is used to remove the noises in the images. The median filter is generally a non-linear filtering approach, and it mainly focuses on edge preservation and noise removal over the collected images. Moreover, the errors while processing the images are highly reduced by removing noise present in the images. Fine details preservation may improve the quality of images, and hence the recognition rates are increased. The median filter [20] functioned through the window, and the selection of median intensity was carried out with the help of this window. It takes the average value of the neighborhood pixel values, and then it is named the average filter. The images are processed through pixel-by-pixel values, and the variations in the pixel values are changed with the averaged value obtained over the nearest pixels. This pixel changing related to the nearest neighbor through the window mechanism is named as sliding window mechanism. Through this sliding window, the quality of the images is getting highly increased. The acquired images $F_{Ig}$ are randomly represented as a group of variables $F_{Ig} = F_{I1}, F_{I2}, F_{I3}, ..., F_{IG}$, and the median value of this image is evaluated using Eq. (1).

$$\text{Med}(F_{Ig}) = \begin{cases} F_{I_{T+1}} = F_{I_b}, & G = 2T + 1 \\ \frac{1}{2}(F_{I_{T+1}} + F_{I_T}), & G = 2T \end{cases}$$

(1)

Here, $b = F_{I_{T+1}}$ represents the median rank and the intensity value of the face images are determined by using the 2-D median filter that is given in Eq. (2).

$$Z_{t_{inb,tn}} = \text{Med} \left( F_{I_{inb,tn}} \right)$$

(2)
Here, the sliding window in this approach is indicated as \( \zeta \). The finally obtained filtered facial images are represented by the term \( F_{l}^{H} \), and this high quality with noise-removed images is given to the input of the next level of face recognition approach.

IV. ENHANCED GANNET OPTIMIZATION ALGORITHM-BASED OPTIMAL PATTERN EXTRACTION FOR FACE RECOGNITION

A. Improved RSO

The suggested IRSO algorithm is used in the newly implemented deep intelligence-based face recognition approach, where the optimal pixels from the LVP and LBP pattern extraction approach are selected using the designed IRSO for maximizing the accuracy of the feature extraction process. Moreover, the hidden neuron count in ResNet is optimized with the support of developed IRSO that helps to improve the recognition accuracy of the designed IRSO-AMT-ResNet-based face recognition model. The RSO algorithm is used in the developed face recognition model because it avoids the local optima problem very effectively, and it highly balances the exploration as well as exploitation phases. Moreover, the computational complexity of this RSO is very less than other algorithms. But, real-world problems are not effectively handled in the RSO. Hence, to solve real-world problems and get best optimal solution in the search space, an improved RSO algorithm is developed. The developed IRSO algorithm has functioned through the parameter \( M \). In the conventional algorithm, the parameter \( M \) is evaluated using Eq. (5). But, in the developed IRSO, the parameter \( M \) is evaluated using the newly introduced concept that is defined in Eq. (3).

\[
M = 2 \cdot \left[ \frac{(k \ast h) \ast I}{IT_{MX}} \right]
\]  

(3)

Here, the terms \( k \) and \( h \) represent the random parameters, in which the values are selected at \( k = 1 \) and \( h = 5 \) to get best optimal solution. Furthermore, the term \( I \) denotes the iteration count and the term \( IT_{MX} \) indicates the maximum number of iterations. By using this modified concept, the best optimal solution is achieved in search space.

RSO: The RSO is inspired by the attacking behaviour as well as the chasing behaviour of the prey. Totally, two different species like, brown and black are there. The female rats and the male rats are named does and bucks. Generally, rats are socially intelligent based on nature. They are involved with various activities like boxing, tumbling, chasing, and jumping. The rats are characterized as the territorial animal that is live with a group of both female rats and male rats. In many cases, the character of rats is very argumentative, which results in the death of some animals. Hence, the RSO algorithm is designed based on the aggressive behaviour of the prey chasing and fighting, and it performs optimization in search space.

Rats chase the prey in a group based on social agnostic behaviour because rats are social animals. The best search agent in the group has knowledge about the position of the target prey. The new position of the search agents is upgraded in regards to the best search agent in the search space. The updated position of the best search agent is given in below Eq. (4).

\[
\bar{S} = M \cdot \bar{S}_{i}(r) + N \cdot (\bar{S}_{i}(r) - \bar{S}(r))
\]  

(4)

The position of the rat is denoted by the term \( \bar{S}_{i}(r) \) and \( \bar{S}_{i}(r) \) denotes the best optimal solution of the search agent. The formula utilized to estimate the parameters \( M \) and \( N \) is represented in Eq. (5) and Eq. (6), respectively.

\[
M = L - r \times \left[ \frac{L}{IT_{MX}} \right] \quad \text{where, } r = 0, 1, 2, \ldots, IT_{MX}
\]  

(5)

\[
N = 2rd(t)
\]  

(6)

The prey fighting behavior of the rats in search space is given in Eq. (7).

\[
\bar{S}_{i}(r + 1) = |\bar{S}_{i}(r) - \bar{S}|
\]  

(7)

Here, the updated next position of the rats is given in \( \bar{S}_{i}(r + 1) \). The best solution is saved, and the best position of all other search agents is upgraded. The best exploration and exploitation are ensured by using this RSO. Algorithm 1 represents the pseudocode of the developed IRSO.

**Algorithm 1: Designed IRSO**

- Initialize the population of RSO
- Set the parameters \( N, L \) and \( M \)
- Estimate the fitness value of the search agent
  - For \( m = 1 \) to \( S_{pp} \)
  - For \( n = 1 \) to \( IT_{MX} \)
    - Assume \( S_{i} \) \( \leftarrow \) the best search agent
    - While \( r < IT_{MX} \)
      - For all search agents
        - Update the parameter \( M \) with a newly developed concept in Eq. (3)
        - Update the current search agent position by Eq. (7)
      - End For
    - Update the parameters \( N, L \)
    - Find any of the search agent goes beyond the search space
  - End For
  - End For
- Go back to the best optimal solution
- End procedure
B. LBP and LVP Pattern

The input to be given for obtaining LBP and LVP from the preprocessed image is \( F I^M_r \). The facial images are generally comprised of a composition of micro-patterns that are highly invariant regarding the monotonic transformations in grayscale. Hence, the LBP and LVP pattern retrieval from the facial images are required for recognizing the person with high efficiency.

The LBP [24] operator uses the thresholding approach to label the pixels of an image, where the thresholding operation is performed over the \( 3 \times 3 \) nearest of all pixels in accordance with the center value, and the resultant patterns are obtained from the face images are binary numbers. Then, the binary numbers are described as a label presented in the histogram, which can be utilized as a texture descriptor. After completing the thresholding over \( 3 \times 3 \) the window, then the window size is extended to use the neighborhood pixels. The bilinear interpolation among circular neighborhoods is used to allow any number of pixels and radius among the neighborhood. The neighborhood pixels are represented as \((U, V)\), where \( U \) defines the sampling points, and the term \( V \) indicates the radius of the circle. The LBP is known as uniform pattern, because it contains two most transition from 0 to 1 while taking the binary string in the form of a circular. The example of uniform patterns is the \( 00000000, 00010010 \) and \( 10000011 \). The distribution of these patterns can be created by the LBP operator. The uniform patterns are highly invariant regarding the monotonic transformations in grayscale.

The LBP pattern is extended to use the neighborhood pixels. The bilinear interpolation among circular neighborhoods is used to allow any number of pixels and radius among the neighborhood. The neighborhood pixels are represented as \((U, V)\), where \( U \) defines the sampling points, and the term \( V \) indicates the radius of the circle. The histogram of the labeled image \( F I^M_r (m,n) \) can be defined in Eq. (8).

\[
HS_p = \sum_{m,n} IN(FI^M_r (m,n)) = p_k \quad p = 0, \ldots , I - 1
\]  

(8)

The spatially enhanced histogram is defined in the labeled image \( F I^M_r (m,n) \) can be defined in Eq. (8).

Here, the term \( l \) denotes the number of different labels which can be created by the LBP operator. The distribution of the local micro pattern information is present in the histogram that includes flat areas, spots, and edges. Hence, the whole image is divided into more number of regions \( R_{n_0}, R_{n_1}, \ldots , R_{n_k-1} \). The spatially enhanced histogram is defined below in Eq. (9).

\[
HS_{p,q} = \sum_{m,n} IN(FI^M_r (m,n) IN(m,n) \in R_{n_k})
\]

(9)

Three different levels of locality can be obtained from this enhanced histogram. Moreover, the pixel-level pattern information is contained in the labels of the histogram. The regional-level information is obtained by summing the small regions, and the global description of the face is obtained by combining the regional information from the histogram.

When the whole image is divided into more number of regions, it may await that some region contains more information to differentiate peoples from the extracted patterns. Hence, similarity measure between the patterns are estimated by using the below mentioned formula in Eq. (10).

\[
S^2(A, B) = \sum_{p} \frac{(A_p - B_p)^2}{A_p + B_p}
\]  

(10)

This is the formula to calculate the similarity measure and it can be extended to the spatially enhanced histogram by adding over the patterns \( p \) and \( q \). Finally obtained LBP patterns from the face images are indicated by \( LBP^H_r \).

The LVP [35] is used to define the local texture in structural information and one dimensional direction by estimating the patterns between the adjacent pixels and referenced pixels in accordance with varying distance and in different directions. The input facial images to be given for extracting the LVP patterns are \( F I^M_r \). The local sub region of the \( F I^M_r \) images are indicated by the term \( LS \), the direction value of a vector is denoted by \( R_{\alpha, \alpha} (E_z) \), where the term \( E_z \) represents the referenced pixel in the local sub region \( LS \), the distance between the adjacent pixel and the referenced pixel are denoted by the term \( D_t \) and the index angle for varying direction is represented by \( \alpha \). The direction value of the vector is estimated using Eq. (11).

\[
R_{\alpha, \alpha} (E_z) = (LS(E_{\alpha, \alpha}) - LS(E_z))
\]  

(11)

The LVP is considered as in the \( \alpha \) direction of a vector at \( E_z \) and the encoding operation of \( LVP_\alpha (E_z) \) are given in Eq. (12).

\[
LVP_\alpha (E_z) = \sum_{k=1}^{8} g(R_{\alpha, \alpha} (E_k), R_{\alpha, \alpha} (E_z)) \times 2^{k-1} \quad \alpha \in \{\alpha, \alpha + 45^\circ, \alpha + 90^\circ, \alpha + 135^\circ\}, K = 8
\]  

(12)

The mathematical formulation of \( g(\cdot, \cdot) \) is indicated in Eq. (13).

\[
g(R_{\alpha, \alpha} (E_k), R_{\alpha, \alpha} (E_z)) = 2^{k-1} \quad \alpha \in \{\alpha, \alpha + 45^\circ, \alpha + 90^\circ, \alpha + 135^\circ\}, K = 8
\]

(13)

Finally, the LVP is obtained by integrating the four 8-bit binary patterns of LVPs that is illustrated in Eq. (14).

\[
LVP(E_z) = \left[ LVP_\alpha (E_z) \right]_{\alpha = 0^\circ, 45^\circ, 90^\circ, 135^\circ}
\]  

(14)

The resultant LVP patterns from the preprocessed facial images are indicated by the term \( LVP^\alpha \).
C. Optimal Pattern Extraction

The LBP pattern $LBP_{h}^{P}$ and the LVP pattern $LVP_{b}^{Cn}$ are extracted from the preprocessed facial images. The size of the LBP pattern image is considered as $128 \times 128$, and also the size of the LVP pattern image is considered in the same size of $128 \times 128$. Hence, the total pixel of the image should be 16384. From these LBP and LVP patterns, the optimal spatial features are chosen with the help of developed IRSO. The best spatial values are arranged in the new matrix to get a hybrid pattern. The selection of optimal pixel is made with new criteria that is, if the selected pixel number is 1, then get the pixels from LBP output or else, get pixels from the LVP output. At last, a new set of patterns are obtained from the LBP and LVP patterns. The optimally selected patterns, with the help of the developed IRSO are indicated by the term $OP_{z}^{hp}$, which is given to the input of the face recognition process. The optimal pattern extraction from LBP and LVP by utilizing the developed IRSO is depicted in Fig. 3.

V. AN ADVANCED FACE RECOGNITION USING ADAPTIVE MULTI-SCALE TRANSFORMER-BASED RESNET

A. Multi-scale Transformer-based Resnet

The AMT-ResNet is used in the developed face recognition framework to extract powerful discriminative features from the face images. The learning capacity of the CNN networks is high, and then the ResNet structure is utilized in the developed face recognition model. The input to be passed through the AMT-ResNet model is the resultant optimally extracted patterns $OP_{z}^{hp}$.

In AMT-ResNet [8], the encoder and decoder functions are used to detect the changes in the pixels and provide accurate recognition of face images. The encoding and decoding operation may also provide a good trade-off between the effectiveness and efficiency of the developed model. Then the feature-level fusion is also carried out over the query image and the database image. The local receptive field that arises in the network is effectively resolved by the ResNet. The multi-scale transformer divides the whole image into more number of non-overlapping regions that are represented as $P_1, P_2, ..., P_n$ and then classifies the pixel points in the facial images very precisely. The loss of information is effectively reduced by using this AMT-ResNet. Moreover, the transformer can find the local receptive fields after extracting features from the face images by using the ResNet model. The ResNet model used in the multi-scale transformer may improve the scalability of the developed face recognition framework. The extracted patterns $OP_{z}^{hp}$ are given to the transformer, and then the learnable parameter encoding is performed with absolute or relative information about the images.

The attention fusion of the face images are represented by the term $Y$, and it is expressed in Eq. (15), Eq. (16), Eq. (17), and Eq. (18), respectively.

$$Y = \left[ h^{i}(P) \| h^{j}(P^{*}) \right]$$  \hspace{1cm} (15)
\[ Ot = Y + MS(Ln(Y)) \]  
(16)
\[ Ot = [Ot^1 \| Ot^I] \]  
(17)
\[ J^z = k^z(\mathbf{O}_t^z) \]  
(18)

Here, the term \( P^z \) denotes the pixel points in the given image. Moreover, the terms \( h^z () \) and \( k^z () \) represents the projection as well as the back projection functions that is used to align the dimension of the image.

The pairwise fusion of facial images is represented in Eq. (19).

\[
S^z = \left[ k^z \left( \sum_{m \in \{z\}} k^z \left( P^m_{\text{CL}} \right) \| k^z \left( \sum_{m \in \{z\}} k^z \left( P^m_{\text{Patch}} \right) \right) \right) \right]
\]  
(19)

Here, the term \( P^m_{\text{Patch}} \) is the patch information and the term \( P^m_{\text{CL}} \) represents the class information of the face images. The cross attention function is given in Eq. (20).

\[
CrAt = At \cdot \mathcal{A}
\]  
(20)

Here, the term \( At \) is the attention map and it is estimated using Eq. (21).

\[
At = \text{softmax} \left( QL / \sqrt{Em} \right)
\]  
(21)

Here, the term \( Em \) denotes the embedding dimension and \( D \) gives the number of heads parameter. The values of \( Q \), \( \mathcal{A} \) and \( L \) are estimated using Eq. (22).

\[
\mathcal{A} = P^z_{\text{CL}} \psi_{\mathcal{A}}, L = P^z_{\text{CL}} \psi_{L}, Q = P^z_{\text{CL}} \psi_{Q}
\]  
(22)

The cross attention of the normalization layer and the residual shortcut are represented in Eq. (23) and Eq. (24), respectively.

\[
Y^z_{\text{CL}} = h^z P^z_{\text{CL}} + MS(LN(h^z (P^z_{\text{CL}} \| P^z_{\text{Patch}})))
\]  
(23)
\[
J^z = k^z (P^z_{\text{CL}} \| P^z_{\text{Patch}})
\]  
(24)

Here, the term \( MS \) defines the multiple heads in the transformer network. The AMT-ResNet structure produces better results over the face recognition, and the accuracy is highly improved. Fig. 4 depicts the basic structure of AMT-ResNet model.

B. Adaptive Multi-scale Transformer-based ResNet

The newly developed AMT-ResNet-based face recognition approach is useful for detecting the person’s face from the collected face images under variations in pose, expression, occlusion, and illumination as well as light intensity conditions. The hidden neuron count in the AMT-ResNet is optimized through the developed IRSO for improving the recognition accuracy rate during face recognition. The Multiscale Transformer network is used in the developed face recognition model because it effectively learns the discriminative variations in the face images. It provides better-recognized results over face recognition with highly reduced recognition time and computational complexity. In order to improve the robustness and reliability of face recognition, the adaptive multi-scale transformer-based ResNet is introduced, where the hidden neuron count in the ResNet is optimized with the help of the newly implemented IRSO. The main motive of the parameter optimization in the developed AMT-ResNet is to obtain higher recognition accuracy. The objective function of the developed IRSO-AMT-ResNet-based face recognition approach is given below in Eq. (25).

\[
OB = \min_{\{Sp^m_{LBP}, Nh^m_{\text{ResNet}}\}} \left( \frac{1}{AR} \right)
\]  
(25)

Fig. 4. Basic structure of AMT-ResNet model.

Here, the term \( Sp^m_{LBP} \) gives the extracted patterns from LBP and LVP, and the term \( Nh^m_{\text{ResNet}} \) indicates the hidden neuron count in the ResNet structure. The extracted patterns \( Sp^m_{LBP} \) are optimized in the range of [1, 2] and the hidden neuron count is tuned in the range between [50, 100]. Furthermore, the term \( AR \) represents accuracy and it is evaluated by using true and false positive as well as negative observations. The accuracy formula is given in below Eq. (26).
\[ AR = \frac{R_p + R_n}{R_p + R_n + L_p + L_n} \]  

(26)

Here, the term \( R_p \) denotes the true positive, \( L_p \) is the false positive, \( R_n \) is the true negative and \( L_n \) is the false negative observation value. The face recognition process using developed AMT-ResNet with parameter optimization is given in Fig. 5.

**VI. RESULTS AND DISCUSSIONS**

A. Experimental Setup

The newly implemented IRSO-AMT-ResNet-based face recognition approach was designed in Python tool. Moreover, the implementation results of the implemented approach have been analyzed among various heuristic algorithms and conventional face recognition methodologies for validating the efficacy of the developed model. The population count and the maximum number of iterations that should be taken for performing the comparative analysis were 10 and 25, respectively.

The heuristic algorithms to be taken for performing the comparative analysis on the developed IRSO-AMT-ResNet-based face recognition approach were Cat Swarm Optimization (CSO) [32], Cuckoo Optimization Algorithm (COA) [10], Moth Flame Optimization (MFO) [31] and Rat Swarm Optimization (RSO) [14] and the conventional face recognition approaches to be considered for analyzing the performance on the developed model was CNN [12], LSTM [11], VGG16 [4] and ResNet [5]. Various positive, as well as negative measures were used for analyzing the efficiency of the implemented face recognition model.

B. Validation Measures

The performance metrics used to validate the effectiveness of the developed face recognition model are several positive measures such as sensitivity, accuracy, specificity, precision, NPV, MCC, and F1-score, and also negative measures like FDR, FNR, and FPR. The formula used to calculate the positive and negative measures is summarized as below.

\[ \text{Precision} = \frac{R_p}{R_p + L_p} \]  

(27)

\[ \text{F1-score} = \frac{2R_p}{2R_p + L_p + L_n} \]  

(28)

\[ \text{Sensitivity} = \frac{R_p}{R_p + L_n} \]  

(29)

\[ \text{Specificity} = \frac{R_n}{R_n + L_p} \]  

(30)

\[ \text{FPR} = \frac{L_p}{R_n + L_p} \]  

(31)

\[ \text{FNR} = \frac{L_n}{R_p + L_n} \]  

(32)

\[ \text{FDR} = \frac{R_p}{R_p + L_n} \]  

(33)

\[ \text{MCC} = \frac{R_p + R_n - L_p + L_n}{\sqrt{(R_p + L_p)(R_n + L_n)(R_p + L_n)(R_n + L_p)}} \]  

(34)

C. Experimental Results

The resultant facial images obtained after the preprocessing, extracting LBP and LVP. The optimal pattern extraction of dataset 1 and dataset 2 are depicted in Fig. 6.

D. Ablation Study on Developed Face Recognition Model Based on the Accuracy

The accuracy of the suggested IRSO-AMT-ResNet-based face recognition approach over the extension of convolutional networks is illustrated in below Fig. 7. This comparative analysis is taken over the two datasets 1 and 2. From the analysis, the developed IRSO-AMT-ResNet-based face recognition approach gained improved accuracy of 8.23% than AlexNet, 6.97% than GoogleNet, 5.74% than ResNet 150, and 4.54 % than ResNet 152 for the learning percentage value of 65 while considering the dataset of 2. The accuracy of the developed IRSO-AMT-ResNet-based face recognition approach is slightly increased than the convolutional networks.

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<table>
<thead>
<tr>
<th>Dataset Description</th>
<th>Image Description</th>
<th>Original Images</th>
<th>Pre-processed Images</th>
<th>LBP Images</th>
<th>LVP Images</th>
<th>Optimal Pattern Extracted Images</th>
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<td><img src="image30" alt="Image set 3 Optimal Pattern" /></td>
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</tbody>
</table>

Fig. 6. The resultant facial optimal pattern extracted images from the developed face recognition model.

![Ablation Study](ablationstudy.png)

Fig. 7. Ablation study on suggested face recognition model over existing networks in regards with (a) Dataset 1 and (b) Dataset 2.
E. Convergence Evaluation on Developed Model

The following Fig. 8 shows the convergence evaluation of the developed IRSO-AMT-ResNet-based face recognition model over dataset 1 and dataset 2. This convergence analysis is carried out over various heuristic algorithms and the developed IRSO-AMT-ResNet-based face recognition model secured with improved cost function rate of 0.29% than MFO-AMT-ResNet, 0.79% than COA-AMT-ResNet, 1.37% than CSO-AMT-ResNet and 2.43% than RSO-AMT-ResNet when considering the iteration number as 20 for dataset 2. Moreover, the cost function is highly decreased in the developed face recognition model rather than the other heuristic algorithms.

F. Accuracy Evaluation on Developed Face Recognition Model

The efficiency estimation of the developed IRSO-AMT-ResNet-based face recognition framework in terms of varying learning percentage value with respect to various heuristic algorithms is given in Fig. 9, and various conventional face recognition approaches is given in Fig. 10. This comparison among heuristic algorithms and conventional approaches are carried out over dataset 1 and dataset 2. The developed model obtained improved accuracy of 5.88%, 4.65%, 2.85%, and 2.27% than the heuristic algorithms CSO-AMT-ResNet, COA-AMT-ResNet, MFO-AMT-ResNet and RSO-AMT-ResNet for dataset 1 with reference to the learning percentage value of 55.

G. K-fold Comparison on Developed Face Recognition Model

The k-fold validation of the implemented IRSO-AMT-ResNet-based face recognition model when compared to various heuristic algorithms is given in Fig. 11, and various baseline face recognition approaches are shown in Fig. 12. The developed model accomplished with improved accuracy of 15.47%, 16.86%, 14.11% and 11.49% than the heuristic algorithms CNN, LSTM, VGG16 and ResNet for dataset 1 with respect to the K-fold value of 3. The developed model secured with high accuracy than the existing face recognition approaches.

Fig. 8. Convergence evaluation of the suggested face recognition model over different heuristic algorithms in regards with (a) Dataset 1 and (b) Dataset 2.

Fig. 9. Efficiency evaluation of the suggested face recognition model over different heuristic algorithms in regards with (a) Dataset 1 and (b) Dataset 2.
H. Comparative Analysis of the Developed Face Recognition Model

The different baseline face recognition approaches are illustrated in Table II. The performance analysis of the implemented IRSO-AMT-ResNet-based face recognition model according to various optimization strategies is depicted in Table III. The developed model obtained with higher recognition sensitivity of 3.95%, 2.80%, 1.70%, and 1.38% than the heuristic algorithms like CSO-AMT-ResNet, COA-AMT-ResNet, MFO-AMT-ResNet and RSO-AMT-ResNet while taking the dataset 1. The recommended face recognition model achieved superior performance than the other optimization strategies in terms of various positive and negative measures.

I. Ablation Study on Different Network Models for Face Recognition

The performance of the developed IRSO-AMT-ResNet-based face recognition model among various face recognition networks is given in below Table IV. The implemented model achieved with improved F1-score of 62.86% than AlexNet, 48.68% than GoogleNet, 33.22% than ResNet 150 and 22.49% than ResNet 152 for dataset 2. The overall performance of the developed face recognition model is greater than the convolutional networks for dataset 1 and dataset 2.
### TABLE II. EFFECTIVE EVALUATION ON DEVELOPED FACE RECOGNITION MODEL AMONG DIFFERENT OPTIMIZATION STRATEGIES

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<tr>
<td>Specificity</td>
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<td>91.8191</td>
<td>92.70444</td>
<td>93.03968</td>
<td>94.31935</td>
</tr>
<tr>
<td>Sensitivity</td>
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<td>NPV</td>
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<td>92.70444</td>
<td>93.03968</td>
<td>94.31935</td>
</tr>
<tr>
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Dataset 1

<table>
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Dataset 2

### TABLE III. EFFECTIVE EVALUATION ON DEVELOPED FACE RECOGNITION MODEL AMONG CONVENTIONAL APPROACHES

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Dataset 1

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Dataset 2
TABLE IV. ABLATION STUDY ON THE DEVELOPED FACE RECOGNITION MODEL AMONG DIFFERENT NETWORK MODELS

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VII. CONCLUSION

A new deep intelligent-based face recognition approach has been presented to identify individuals with high recognition accuracy. This face recognition helped to identify the thieves, and it has been mainly used in forensic applications. The face images were collected from two distinct databases, and median filtering was used to filter the image to eliminate the noises. The LBP and the LVP patterns were extracted from the filtered face images, and the optimal patterns were extracted from this by utilizing the developed IRSO. The optimally selected patterns from the face images were given to the AMT-ResNet, where the hidden neuron count was optimized through the newly implemented IRSO for increasing recognition accuracy. The implementation results obtained from the developed IRSO-AMT-ResNet-based face recognition model have been validated by analyzing the results over the conventional face recognition models according to ablation accuracy and convergence analysis. The developed IRSO-AMT-ResNet-based face recognition model has attained with improved accuracy of 5.94%, 4.59%, 3.15%, and 2.12% than the convolutional networks like AlexNet, GoogleNet, ResNet 150, and ResNet 152. Furthermore, the accuracy of the developed deep intelligence-based face detection approach was highly enhanced than the conventional face recognition techniques and heuristic algorithms.

REFERENCES


Network in Face Classification Problems," The international journal, 2008.


AUTHORS’ PROFILE

Mr. Santhosh S., received his B.E. degree from SIMIT, Chitradurga, affiliated to VTU, Karnataka, India in 2005 and M.Tech., degree from MCE, Hassan, affiliated to VTU, Karnataka, India in 2011. He has a total experience of more than 10 years in teaching at Kalpataru Institute of Technology, Tiptur. Currently, he is working as Associate Professor in the department of CSE at Kalpataru Institute of Technology, Tiptur, Karnataka, India affiliated to VTU, Karnataka, India. He is pursuing Ph.D. at Kalpataru Institute of Technology research centre, affiliated to VTU, Karnataka, India. His areas of interest include Face Recognition, Image Processing and Pattern Recognition.

Dr. S. V. Rajeshkarradath, working as Professor in the Department of Electronics and Communication Engineering, Kalpataru Institute of Technology, Tiptur, Tumkur District, Karnataka, India affiliated to Visvesvanaya Technological University, Belagavi, Karnataka, India. He has a total academic and research experience of more than 30 years at various capacities as Lecturer, Senior Lecturer, Assistant Professor, Professor, Principal, NBA Coordinator, Member BOE VTU, Member LIC VTU, BOS UVCE Bangalore. He received his B.E. degree from Mysore University, Karnataka, India, in 1991, M.E. degree from UVCE, Bangalore, Karnataka, India in 2000 and Ph.D. degree from Anna University, Chennai, India in 2010. He received HDP fund from VGST, Karnataka. His areas of interest include Image Processing, Pattern Recognition, Handwriting Analysis, Feature Extraction, Neural Networks, Face Recognition, Machine Learning and Support Vector Machines. He has published more than 20 research publications in International Journals and Conferences. He has authored 2 book chapters.
An Intelligent Malware Classification Model Based on Image Transformation

Mohamed Abo Rizka¹, Mohamed Hamed², Hatem A. Khater³
College of Computing and Information Technology-Heliopolis Campus, Arab Academy for Science, Technology & Maritime Transport, Cairo, Egypt¹,² Electrical Department-Faculty of Engineering, Horus University Egypt, New Damietta 34518, Egypt³

Abstract—Due to financial incentives, the number of malware infections is steadily rising. Accuracy and effectiveness are essential because malware detection systems serve as the first line of defense against harmful attacks. A zero-day vulnerability is a hole in the target operating system, device driver, application, or other tools employing a computer environment that was previously unknown to anybody other than the hacker. Traditional malware detection systems usually use conventional machine learning algorithms, which call for time-consuming and error-prone feature gathering and extraction. Convolutional neural networks (CNNs) have been demonstrated to outperform conventional learning techniques in a number of applications, including the classification of images. This success prompts us to suggest a CNN-based malware categorization architecture. We evaluated our methodology using a bigger dataset made up of 25 families within a corpus of 9342 malware. Last but not least, comparisons are made between the model's measurement and performance with other cutting-edge deep learning techniques. The overall testing accuracy of 98.31% in the provided results attested to the excellent accuracy and robustness of the suggested procedure at a lower computational cost.

Keywords—Malware Classification; zero-day; Convolutional Neural Networks (CNN); grayscale image transformation; Bytehist

I. INTRODUCTION

The quick enhancement of communication and information technologies has had a significant impact on cyber security. Systems and techniques for spotting intrusions and preventing them have significantly advanced. Even with more advanced security measures in place, hackers continue to develop methods to identify weaknesses and seize control of devices and systems. Static analysis approaches, like signature-driven method, pattern-match method, or data mining technology, examine the data inside the file to determine whether an executable portable file contains a programme that shouldn't be launched. The goal of the dynamic analysis method, which involves running the malware itself, is to observe how the portable executable file behaves while it is in use [1][2][3]. Methods for detection and classification were greatly hampered by the attackers' knowledge of infiltration tactics and strategies. Known as zero-day vulnerabilities and zero-day assaults, one of the most popular attack types in use today is malware [1]. The academic community and the security business have employed deep learning, machine learning, and intelligent systems to try and forecast potentially risky conduct. The first quarter of 2021 saw a 68.9% spike in new PowerShell malware and a 41.2% increase in business malware compared to the previous quarter [3]. The aforementioned statistics demonstrate that researchers in information technology disciplines have started to see IT as applying machine learning-based (neural language) detection and classification algorithms and NL processing to sort through the ever-increasing volume of malware and cunning escape strategies being deployed [3][4]. Due to the frequent requirement for traditional malware research approaches to design crucial traits, which costs money and time, machine learning has been found to be more effective [5]. Malware categorization has also been effectively accomplished using (CNN). The first stage of this scientific study will involve converting files from a regular image format to binary language, which will then be translated into grayscale images. Second, the files will be grouped into families of harmful programs. Twenty five (25) families of Trojan horses, malware, backdoors, etc. are negatively impacted. The accuracy in the identification and categorization of files into families of healthy files and families of hazardous files that will be used is then calculated by dividing a portion of these files into a portion for learning by 80% and a portion for testing by 20%.

This paper’s primary divides are as follows: Section I presents the introduction; the Section II shows the related work, Section III introduces datasets, Section IV establishes the proposed AI algorithm and Section V introduces the proposed DL architecture; the experiment's results are laid out in Section VI, along with discussions; the conclusion is given in Section VII.

These are this paper's significant contributions:

1) Using a number of pre-learned CNN algorithms based on image modification, we suggest a supervised learning-deep technique to categorize malware.

2) Check files in the initial phase to verify whether hashing, signature, or encryption modifications have been made, and use the modified metric to create byte-usage-histograms for whole sorts of codes with an emphasis on binary executables in a portable executable (PE) presentation.

3) Utilize the developed tool to transform any executable or binary file into a greyscale PNG image that can be seen in the range [0,255]. We offered a useful paradigm for handling data from an imbalanced dataset.

4) We conducted numerous tests to contrast our approach for classifying malware with a number of existing techniques;

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the findings show that our approach works better than these other approaches.

5) In order to classify malware, we created a regularized strategy that performs better than competing models despite learning from a tiny dataset.

II. COMPARABLE WORKS

The most common malicious attempts on biometric technology are probably those on records with pattern data. The model contains a user's biometric details that might be abused in an assault. The confidentiality of the user is put at risk by the availability of patterns across multiple programmes [27]. Therefore, a strong technique is needed to protect the forms kept in the database. The following specifications [8] through [10][29] ought to be met by the most suitable pattern safety solution.

A. Strategies for Analyzing Malware

Malware analysis comes in two sorts: static and dynamic. Malware analysis aims to comprehend the composition and operation of malware [3]. Malware samples must be examined in order to ascertain their nature and mode of operation [6]. Static and dynamic analysis are the dual basic techniques utilized to detect malware. This is so that malware can be identified during analysis, which enables the resolution of a number of issues, including the presentation of the harmful architecture, the detection of infections and propagation techniques, and the assessment of the specific harm to the victim's devices [6]. The dual chief methods of malware analysis are static and dynamic. While studying malware, basic static analysis is done first, followed by advanced dynamic analysis [7].

B. Static Analysis

In order to do static analysis on Windows portable executable (PE) records, either the binary file or the malware program that has been disassembled must be used. The most popular programs for opening PE files are IDA Pro and Radar. This kind of reverse engineering can be applied to them.

Without running the malware code, static analysis can reveal the structure of a malware sample [8]. The two parts are fundamental static analysis and enhanced static analysis. Without going deeper, elementary static analysis inspects the programs, assessing file content, header information, and functions [6]. Among the tools that be able to utilize to abstract that information are PEiD, Bin Text, MD5deep, and PE view [7]. The first step in malware analysis is basic static analysis; advanced static analysis should be carried out to learn more about malware. The advanced static analysis does a complete study of the program directives.

To accomplish this, assembly codes are generated from machine codes using a disassembler [6] [9]. For thorough static analysis, researchers typically utilize the IDA Pro packet splitter and the supplemental Hex-Rays de-compiler. The investigation is thoroughly scrutinized to look for signs of malware in the procedures for assembling. With the use of sophisticated static analysis and inverse compilation, specific malware functionality may be retrieved. The advanced static analysis offers an in-depth understanding of the functionality and intent of malware. However, a thorough understanding of operating system principles and assembly code instructions is required for this subject [6][11].

C. Dynamic Analysis

Allows us to monitor its behavior and gather all of the virus's traces as we perform the dynamic analysis. This study is often utilized as a secondary analysis to have additional parameters or if we were unable to gather significant information by Employing static analysis, the malware infection developer's considerable obfuscation. This scan should be carried out in a totally isolated setting to prevent damaging our system. There are several habitats to pick from, with Cuckoo Sandbox being the most well-known. They provide an overview of both the methods used for each type of study and the data that was extracted [1].

Due to the dynamic analysis' use of program execution, malware behavior analysis was done. To prevent infection of the devices, the analysis is done in enclosed environments like sandboxes or virtual PCs. Examining the execution of functions, arguments, data transfers, modifications to the file database, and network usage are all part of the process. When describing the actual operation of malware, static analysis is less accurate than dynamic analysis. There are two different kinds of dynamic analysis: basic and advanced. Basic dynamic analysis is used to examine the behavior of malware [10]. Utilizing Sandboxes, Regshot, ApateDNS, Procedure Explorer, API observe, and Procedure Monitor. The extensive dynamic analysis employs tools for debugging like WinDbg and OllyDbg. Experts who study malware can use debuggers to examine and modify the outcomes of individual commands.

D. Analysis of Statistics and Dynamics

The static analysis makes it simple and quick to evaluate earlier detected malware and gain a quick summary of the software [35]. Unfortunately, it is incredibly difficult to analyze malware that employs obfuscation, packing, polymorphism, and other techniques. Because dynamic analysis involves computer programs, malicious software may be employed. Obfuscation techniques used by malicious software can be recognized. Certain malware variants, however, might be aware that they are being tested in sandboxes and virtual environments, which would conceal their genuine behavior. Dynamic analysis is more efficient when dealing with unknown malware, despite the fact that static analysis is quicker and more precise when dealing with already identified malware [12].

E. Machine Learning Techniques

The two methods utilized in ML are unsupervised learning, which involves identifying hidden patterns or internal frameworks in incoming data, and supervised learning. To be able to forecast future outcomes, supervised learning requires training an algorithm utilizing available data for both input and output.

F. Supervised Learning

An algorithm that generates forecasts using data in the presence of unpredictability is created through supervised ML. A technique for supervised learning employs a set of
A supervised learning approach teaches the model to produce accurate forecasts in response to new data using a well-known collection of input data and identified reactions to data (output data). If the outcomes you are attempting to forecast have known data, use supervised learning. To build ML frameworks, supervised learning uses regression and classification techniques. Classification techniques predict specific outcomes, such as if an email is real or spam, or if a tumor is malignant or not. The given data are categorized by Common uses including speech recognition, credit scoring, and medical visualization. When you can tag, classify, or divide your data into distinct groups or classes, use classification. For instance, categorization is used by a handwriting recognition program to identify letters and digits [14][15]. Unsupervised pattern recognition algorithms are used in image processing for object recognition and image segmentation. Some commonly popular classification techniques include (SVMs), KNN, naive bayes, differential analysis, logistic regression, and NN [17].

Techniques for regression forecast continuous responses like variations in electricity consumption and temperature. Forecasting electricity load and algorithm trading are examples of common applications. Nonlinear models, linear models, progressive organization, regression, reinforced and packed decision trees, adaptive neuro-fuzzy learning, and neural networks are examples of common regression techniques [18].

G. Unsupervised Learning

Data is scanned for underlying structures or hidden patterns using unsupervised learning. From datasets without any marked responses, it is utilized to draw conclusions. Clustering is the technique used most often in unsupervised learning. To find undiscovered patterns or groupings in the data, it is employed in exploratory data analysis. Object recognition and DNA arrangement analysis are a few instances for use for cluster analysis. For instance, a smartphone provider can use machine learning to determine how many different groups of people rely on its towers in order to optimize where it places its cell towers [19]. Because mobile phones can only communicate to one station at a time, the crew used a clustering approach to identify the best locations for cell sites to improve the reception of signals for their customer sets or clusters. Typical clustering methods include clustering based on hierarchy, GM systems, self-organizing maps, HMM, subtractive clustering, fuzzy c-means clustering, and k-medoids and k-means [20].

H. Deep Learning Approaches

DL models are from time to time referred to as DNN since the majority of deep learning methods employ NN architectures. DNNs are simply neural networks that have a lot of hidden layers. While DNN can have up to 150 hidden layers, conventional NN is limited to two or three. Large volumes of categorized data and NN topologies that acquire parameters from the data before learning them are employed for building algorithms for DL [21].

CNN or ConvNet are among the most popular DNN kinds. Specifically, a CNN is well suited for analyzing 2D data, such as photographs, because it mixes learned features with incoming data and makes use of 2D convolutional layers. You won't have to figure out what characteristics are used to classify photos because CNNs do manner with the need for non-automatic parameter extraction. CNN uses direct feature extraction from images to run its business. The necessary features are not pre-trained; rather, they emerge when the network trains on a batch of images. For computer vision applications like object categorization, deep learning models are especially accurate [22].

CNNs are taught to recognize various features of a picture using considerable hidden layers. The complexity of the learned visual elements increases with each buried layer [16]. For instance, the initial hidden layer might train to recognize edges, while the final layer might learn to recognize more intricate forms that are particular to the form of the object, we're able to recognize. In conclusion, because they typically identify and extract a set of parameters in advance and are not built to handle vast volumes of data, conventional ML algorithms have a great complex cost. The technique of DL on the contrary, performs the extraction of features and selection, cutting down on considerable computational costs. Yet, studies have shown that DL is superior to ML in terms of effectiveness and accuracy.

III. DATASET

We assessed our method on a big dataset containing 25 families in malware groups of 9,342. Nataraj et al. contributed MalImg collection [28]. The assessment outcomes display that our technique presents high precision with less computational cost. Moore's details are demonstrated in Fig. 1.

![Fig. 1. Malware families found in the MalImg dataset.](image)

IV. ALGORITHMS

We discuss the dataset processing and implementation specifics for the proposed mathematical frameworks in this part of the article. We use the Bytehist software to generate byte-usage histograms for a variety of records, including binary executables in (PE) presentation. Using the Bytehist tool [23], we check files in the initial phase to see if hashing, signature, or encryption changes have been applied. We have divided malware from the MalImg Dataset into a variety of classifications. We use CNN to train and test the DL system for identifying and classifying malware in 25 families of images, and families of grayscale images.
A. Modified Binary File Detection

Static analysis frequently encounters problems while analyzing compressed or encrypted executables. AI algorithms regularly identify harmful executables as safe, despite the fact that many of them are updated to fulfill business or intellectual property objectives. This is understandable given that these modifications would significantly change the executable’s entropy and byte spread. When creating a performance-improving technique for detection models, take the likelihood of binary file modifications into account. The tool ByteHist may produce byte-usage histograms for a variety of file formats, with a concentration on binary executables in the PE format. For instance, ByteHist [23] offers information into the nature of data before an examination. We can look at how bytes are distributed within a program that runs using ByteHist.

The distribution gets extra even with each executable compression. Examples of both negative and positive analogs are shown in Fig. 2, together with unpacked and UPX-transformed byte distributions. As shown, UPX alters the byte spread of the binary file, especially when malware is present. It is also a widely used packer and binary unpacking is straightforward, in contrast to neutral le, which has more altered cations [24]. UPX generates less. But a lot of malware comes with more sophisticated software, which complicates the investigation. Statistics could be a useful tool for locating encrypted or compressed data.

Bytes in the data are dispersed quite uniformly as a result of this type of alteration. Typical data typically consists of specific bytes that are constantly in use due to any type of structure. The byte distributions of database files, executable binaries, and plain text that haven't been encrypted or compressed differ significantly from those of those that have. This "phenomenon" is displayed using histograms, which make it easy to distinguish between the two.

B. Employing Images to represent Malware

The objective of this research is to visualize malware using a technique created by Nataraj et al. (2011) that enables a malware binary to be read as a stream of 8-bit integers without signs before being structured into a 2-dimensional matrix. Our tool transforms any executable or binary file into a greyscale PNG image that can be seen in the domain [0,255] (0: black, 255: white) [14]. Malware presentation as a grayscale picture process is shown in Fig. 3 Due to the method's reliance on binary code, a new infection might be created by a malware producer by updating the code of an existing virus, which would result in a very similar image being used to display the new infection. Then, we may use our classification model (CNN), which will be illustrated later, to put it all into one family.

C. Using Transfer Learning to Classify Malware

DL is a branch of ML that includes algorithms designed to mimic the operation of neural networks or the human brain. These structures go by the term neural networks. It trains the computer to perform actions that come naturally to people. Some of the models used in deep learning include autoencoders, recurrent neural networks (RNN), (ANN), and reinforcement learning. Convolutional Neural Networks (CNN) or ConvNet, in particular, have significantly advanced the areas of computer vision and image analysis [13]. CNNs, a subcategory of DNN, are often utilized for image analysis as they able recognize and categorize certain structures in frames. They have a variety of uses. Only a few of their applications include picture and video recognition, image classification and NLP. Fig. 4 concludes the convolutional neural networks' historical development.

D. CNN’S Principal Architecture

According to Fig. 5, there are two parts to CNN architecture [25].

- Feature Extraction: A convolution tool isolates and classifies the distinctive features of an image for examination throughout the feature extraction process.
- Fully connected: A completely connected layer that predicts the frame's group applying the data collected in earlier steps and the convolution procedure's output.

E. Convolution Layers

The CNN is consisted of three distinct kinds of layers: completely connected (FC), convolutional, and pooling layers. The CNN architecture will be built by stacking these layers. Additionally, to these three layers, there are two more crucial necessities: the dropout layer and the activation function.

Fig. 2. Using ByteHist, compare the byte spread of normal and malicious programs.

Fig. 3. Malware as a procedure for grayscale representations.

Fig. 4. A brief history of convolutional neural network.
F. Convolutional Layer

This is the first layer separating the many structures from the entrance frame. In this layer, the beginning picture is mathematically convolutional using a filter of a certain size MxM. The number of dots that exist across the filter and various areas of the given picture can be calculated according to the filter's size (MxM) by moving the filter through the frame. The final outcome, also referred to as the feature map, contains information about the picture, such as its contours and borders. Then, further layers receive this feature map, which they use to pick up additional features from the input image.

G. Pooling Layer

A Pooling Layer is frequently used after a Convolutional Layer. This layer's primary goal is to scale down the convolved feature map in order to save processing expenses. This is accomplished by minimising the connections within layers and working independently on every element map. There are multiple sorts of pooling techniques based on the technology employed.

The Max Pooling feature map is used to determine the largest element. With middling pooling, the elements within an image segment of a specific size are averaged. The cumulative total of the elements in the pre-known segment is estimated using totality pooling. Connecting the Convolutional Layer and the FC Layer is frequently done via the Pooling Layer.

H. Complete Layer Connectivity

Weights and biases are included in the Fully Connected (FC) layer, which connects the neurons amongst layers. The resulting layer is frequently positioned before the last few layers in a CNN architecture. The input pictures from the layers above are now smoothed and sent to the FC layer. The standard theoretical useful procedures are then performed on the flattened vector via a few additional FC levels. The classifying procedure officially starts at this point.

I. Dropout

When all of the characteristics are linked to the FC layer, the learning dataset is susceptible to excessive fitting. Overfitting is the process of an algorithm doing such well on data used for training that it has a detrimental impact on how well it works on fresh data. In order to tackle this issue, a dropout layer is implemented, which results in a smaller model by eliminating a limited neurons from the NN throughout learning. After achieving a dropout of 0.3, 30% of the nodes in the NN discontinue arbitrarily.

J. Activation Functions

To summarize, the activation function of the CNN framework is one of its most crucial elements. Any kind of persistent and complicated network variable-to-variable linkage is learned and approximated using them. It decides which design data the network terminal ought to convey as well as which ought not to, to put it simply. The network gains linearity as a result. The ReLU, Softmax, tanH, and sigmoid process are some of the most frequently utilized activation functions. Each of these functions has a unique use. While softmax is frequently employed for a variety of classes sigmoid and softmax functions are chosen for a CNN algorithm for binary classification [26].

K. Proposed Malware Classification Algorithm

The analytical pipeline of the suggested architecture is introduced in Fig. 6 and includes various processing phases. The first step involves preparation of along with information enhancement, which involves changing files from a common picture format to binary language and then back again to grayscale images. Following this, the established CNN framework is described along with its details, including learning through transfer, learning models, variable adjustment, and ultimately categorization. The specifics of those phases are then extensively explained.

Fig. 5. CNN’s elementary architecture.

Fig. 6. Illustration depicting the suggested model for the analysis.
extract certain information from photos and have a constrained response region. The convolution layer is identified as follows:

\[ X_n^r = \alpha \left( \sum_{m=1}^{k} X_m^{r-1} * W_{mn}^r + b_m \right) \]  \hspace{1cm} (1)

The current layer's \((r^{th})\) activation map is characterized by \(X_n^{r}, \) the previous layer's \((r-1)^{th}\) activation map is represented by \(X_m^{r-1}(r-1),\) and how many enter activation maps are there, is indicated by \(k.\) The weight and bias vectors are \((w_m)^r \) and \((b_m)^r,\) respectively. Convolution is performed using the \(*\) operator, and \((\alpha)\) stands for the function of activation.

After each generated activation pattern has been activated by the function of activation, it is subsequently transported to the layer of pooling. The layer of pooling produces a transformation constant by reducing the overall quality of the activation maps and the layer of pooling activations is produced by the convolution layer's \(d \times d\) (for example, \(d=2\)) structure of activation maps. The pooling technique that is most frequently employed is max pooling. The fully connected layer uses the data from all of the activation maps from the layer before it to create a categorization map. The optimizer is essential in learning the DCNN algorithm since it continually modifies the network's layer settings.

To attempt to reduce the effect of the loss function \((i.e., \nabla_\theta L(\theta)),\) the settings are modified in the contrary orientation of the variation of the loss function \((i.e., L(\theta)\) compared to the variables). Following every repetition, the intended and forecast outputs are contrasted, and the mistake is back propagated. One of the greatest commonly employed evaluation of performance measures is cross-entropy. The basic objective of any optimisation technique is to have a cross-entropy score that is almost zero while the desired and predicted results are the same.

These models will locally identify patterns as CNNs operate internally using convolutions in several sliding windows, enabling a robust differentiation between how every category is represented. The layer of dropouts has been modified to 0.25 for the first and succeeding layers of convolution and to 0.3 for the following layer of convolution. Reformed Linear Units, or ReLUs, serve as the activation function for every layer of convolution. The framework may identify patterns in the provided data and transfer those patterns onto subsequent levels. Following adjusting, the outcome of the preceding convolution is sent to the final dual layers, a full connectivity (FC) layer with 0.2 dropouts and a softmax layer with four neurons. The layer of networks responsible for categorizing determines the likelihood that a data source will fit into a certain classification. For analysis of time-series data employing pooling and expanding filter dimensions ranging this type of multi-layer architecture has shown to be effective. [25][26].

The outcome patterns are \(y= y_1, y_2... y_m,\) while the given input patterns for the model are \(x= x_1, x_2... x_n.\) The result of the last layer of the network was improved by means of the cost array \((\xi).\) If \(y\) is the result of every specific method, \((L)\) is the value of the loss function, \((\partial)\) is the result after the following adjustments and \((\zeta)\) is the desired class, then \((y)\) is the result.

\[ \partial_i = L(\xi_i, y_i) : \partial \in \mathbb{R} \neq \zeta \]  \hspace{1cm} (2)

The loss function has been changed to:

\[ L = \sum_n t_n \log(\partial_n) \]  \hspace{1cm} (3)

Where \(\partial_n\) contains the cost that is class-dependent \((\zeta)\) and is associated with the result on \((y_n).\)

\[ \partial_n = \frac{\xi_n \exp(y_n)}{\sum_k \xi_k \exp(y_k)} \]  \hspace{1cm} (4)

The quantity of samples in a class determines how much weight it has. If class \(\Pi\) has \(t\) times extra trials than class \(p,\) making one trial from class \(p\) as significant as \(t\) samples from class \(\Pi\) is the goal. Therefore, the class weight of \(p\) is \(t\) times more than the class weight of \(\Pi.\) We employ 2D convolutional layers in our model, which is depicted in Fig. 7, using 3x3 kernels for each of the subsequent blocks and 5x5 kernels for the initial block. Moreover, we employ 2x2 for the final two blocks. Every block's second layer of convolution used the ReLU activation function while down-sampled with a stride of two. The first block contained 64 filters, and every block after which included double number of filters. A layer of dropouts \((p = 0.3)\) was added after the last convolutional layer had been applied and connected to one FC-dense layer with ReLU activation scores of 1024.

There was also a layer of dropouts \((p = 0.3)\) sandwiched in among those thick layers. Finally, the algorithm result was provided by a softmax-activated multi-dense neuron. The Adam optimizer was used to learn the algorithm for up to 100 epochs at a rate of learning of 0.001, utilizing a batch dimension of 40. Additionally utilized was the class cross-entropy process of loss that is often employed for several classes' problems with categorization. The cross-entropy is described as follows, using \(p\) standing for the actual distribution and \(q\) for the calculated distribution:

\[ H(p,q) = -\sum_x p(x) \log(q(x)) \]  \hspace{1cm} (5)

The suggested deep learning pipeline's parameters are recorded in Table I as a whole.

![Fig. 7. CNN algorithm structure utilized for malware classification.](image-url)
VI. EXPERIMENTAL RESULTS AND DISCUSSIONS

This part goes into considerable depth about both the investigational design and the outcomes. The trial setup includes the framework and code structure training information used in the present study. We conducted separate experiments and compared the outcomes. The experiment's findings are presented and discussed in this part of the paper. We adjusted the hyper-factors for the suggested algorithm's batch size, epochs, and folds in order to get the most effective findings.

Forty (40) batches of data each epoch from a total of 100 epochs are used to learn the network. For every experiment, data is separated into 20%–80% segments for network testing and learning. The set for validation uses 16% of the training set's data. The setup makes use of the Keras platform. The parallel implementation is essential for deep learning training. As a result, we employed Kaggle and the open-source software Python 3.11.0 to perform out the classifier's learning and validation (GPU: NVIDIA TESLA P100 GPUs, 16 GB RAM). The recommended strategy was constructed using the Keras library from Tensor flow applications, and the execution duration was 560.7 seconds. Five series of trials show the changes in how well the suggested solution performs.

The framework's assessment establishes how effectively a certain data structure generalizes to new data in order to distinguish among multiple approaches. To do this, we need to assess the effectiveness of multiple algorithms using a method of estimation besides an assessing approach, such as a learner-test break or cross-validation [27].

A crucial indicator is the accuracy of classification (ACC), which assesses in what way effectively the algorithm foretells a class of instances in the validation set. Further measurements include those defined by terms like sensitivity (SEN), precision, and specificity (SPE) [29][30][31]:

\[
\text{Accuracy} = \frac{tn + tp}{tn + tp + fn + fp} \quad (6)
\]

\[
\text{Sensitivity (Recall)} = \frac{tp}{tp + fn} \quad (7)
\]

\[
\text{Specificity} = \frac{tn}{tn + fp} \quad (8)
\]

\[
\text{Precision} = \frac{tp}{tp + fp} \quad (9)
\]

\[
\text{F1 Score} = \frac{2 \times \text{Precision} \times \text{Recall}}{\text{Precision} + \text{Recall}} \quad (10)
\]

The symbols tp, tn, fp, and fn stand for true positive, true negative, false positive, and false negative, respectively. In order to analyze measurements that are quantitative, the confusion matrix is utilized. The confusion matrix is a table that categorizes forecasts into those that were right and those that were wrong [31][32][35]. A confusion matrix is used in Fig. 8 to show the link between the expected class and the true class. Fig. 8 displays the CNN algorithm's evaluation outcomes for the multinomial categorization of malware groups.

A figure illustrating how intelligent the model is used to identify the family of each malware is shown, and we discover that there was some overlap in identifying some malware families as a result of the limited set of grayscale images on which the model was trained. This is what happened with the family (Autorun. K), which contained a number of images used for only a few grayscale images, and this had an impact on the effectiveness of correctly recognizing the family.

![Fig. 8. Results of CNN testing for the Confusion Matrix, showing the accuracy with which it predicted each malware family shown in Fig. 1's list.](image-url)
The produced confusion matrix may be used to create other indices, such as accuracy, precision, F1-score, specificity, and sensitivity (recall). The weighted average of recall and accuracy is the F1 score. The confusion matrix and associated metrics are typically used in conjunction to examine and evaluate categorization methods. Fig. 9 displays the confusion matrices for the suggested systems. Examining which classes, if any, are being misclassified more is quite helpful in determining this. Confusion matrices are helpful for model administration and monitoring in addition to model evaluation. Create confusion matrices for each family class to identify true negatives, false positives, and true positives.

We have employed the criteria already described before to contrast the effectiveness of our methodology. The CNN framework utilizing the basic structure, learned from the beginning via various time-running epochs with the values 20, 40, 50, and 100, achieves an overall classification accuracy of 98.31%. Fig. 10 to 13 show the comparison of accuracy, precision, recall and specificity values for the suggested systems at different epochs respectively. The stated algorithm had a precision of 97.59% as shown in Fig. 10 while Fig. 11 presents a Precision of 97.59 %. Fig. 12 demonstrates a Recall of 90.06% where Fig. 13 introduces Specificity of 99.87% and a F1 score of 99%. Table II introduces the contrasts of the results of accuracy performance by different techniques with the proposed system. According to the findings, our suggested method can provide a reliable algorithm to have an optimum performance to reduce the error and offer an overall accuracy of about 98.31 %. By enhancing the CNN model's architectural design with additional hidden layers, improved nonlinearities, and/or an optimized dropout, it may be possible to get a greater understanding of how to apply it to the categorization of malware. These insights may provide information on the architecture that will work best for creating an intelligent anti-malware system.
In the current research, we develop an advanced (DL) image classification algorithm that was previously trained on the MalImg dataset to classify malware based on images. (CNN)-based (DL) methods were contrasted with an extra simple technique created from beginning. We used the same dataset and equal image sizes for our experiments. Since there were no malware zero days in the sample, the model cannot learn and cannot accomplish its objective of identifying zero days and will be considered in the future. Based on transfer learning findings, the model has been demonstrated to be the most effective after accuracy testing. As a result, we can conclude that the transfer learning approach is suitable for classifying malware to categorize. By enhancing the CNN model architecture design with more hidden layers, improved nonlinearities, and/or an optimal dropout, it may be possible to gain more understanding of how well these models apply to the classification of malware. These findings could help in the development of an intelligent anti-malware platform by informing the type of structure to employ. The total testing accuracy of 98.31% in the reported findings attested to the excellent accuracy and robustness of the recommended technique.

**REFERENCES**


Unsupervised Document Binarization of Engineering Drawings via Multi Noise CycleGAN

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Abstract—The task of document binarization of degraded complex documents is tremendously challenging due to the various forms of noise often present in these documents. While the current state-of-the-art deep learning approaches are capable for the removal of various noise types in documents with high accuracy, they employ a supervised learning scheme which requires matching clean and noisy document image pairs which are difficult and costly to obtain for complex documents such as engineering drawings. In this paper, we propose our method for document binarization of engineering drawings using ‘Multi Noise CycleGAN’. The method utilizing unsupervised learning using adversarial and cycle-consistency loss is trained on unpaired noisy document images of various noise and image conditions. Experimental results for the removal of various noise types demonstrated that the method is able to reliably produce a clean image for any given noisy image and in certain noisy images achieve significant improvements over existing methods.

Keywords—Image processing and computer vision; generative adversarial networks; document binarization; deep learning

I. INTRODUCTION

In the current Industry Revolution 4.0, images and documents are important methods to acquire information. Specifically, this information is in the form of texts contained within images and documents. Document binarization is a crucial pre-processing step when attempting to analyse and extract the information. The goal of document binarization is to retrieve and extract underlying text from the background of documents. More accurate binarization methods allow for performance improvement of document analysis such as text line and word segmentation and Optical Character Recognition (OCR). Successful and accurate OCR of documents allows for better indexing and searching of documents within document management systems.

Among the many challenges with document images is that they are normally affected by various types of noise and environmental degradation. Examples are uneven illumination, ink or page stain, background colour, shadows, blue and white noise. These noises can come from the state of the document themselves when scanned or bad conditions of digitization. Text extraction from these documents with noisy backgrounds especially those with randomized noise poses a tricky challenge as these noises can obstruct text and possibly not picked up by denoising methods.

In recent years, considerable amount of research pertaining to document image enhancement and recognition has been studied and proposed. Document binarization is an active and ongoing research with steady paper submissions on a yearly basis to Document Image Binarization Competition (DIBCO) which is held by International Conference on Document Analysis and Recognition (ICDAR).

Before the widespread usage of deep learning, popular conventional methods for document binarization are global threshold methods, local threshold methods, statistical-based methods and edge-based. Conventional methods typically focus on a specific noise such as bleed through [1], [2]. The limitation of conventional methods is a reduction in noise removal performance if the document contains various and severe noise and degradation such as a combination of shadow, bleed through and uneven illumination.

Recently deep learning-based document binarization methods have shown significant performance improvement when compared to conventional methods. Built upon the framework Generative Adversarial Networks (GANs), proposed by Goodfellow et al. [12], supervised GANs have been utilized and proven capable to address more challenging noise and multiple degradation and noise. Yun Hsuan Lin [4] uses conditional GANs to remove shadows from complex background colour in documents. Liu et al. [5] utilizes a combination of Recurrent Neural Network with conditional GAN to address complex background noise, colour, and watermarks. Deep learning methods such supervised GAN’s are trained in a supervised manner hence matching noisy documents and clean target document pairs are required which are difficult to obtain or generate. Another limitation of supervised GANs and other deep learning methods is that the methods are trained to remove specific noise type which often leads to poor results if the paired training data is not sufficiently many and does not match the noise type.

The ability to denoise and preserve original text and graphics is crucial for accurate OCR and digital document information management. Complex documents such as technical drawings often contain various forms of noise and degradation. Coupled with their irregular text and graphics greatly reduces the performance of an OCR system. The combination of randomized noise and irregular text and graphics results in paired training data for supervised denoising to be difficult or near impossible to obtain. Furthermore, with both the noise and text and graphics in technical drawings being very different from one and another, this results in supervised solutions to be trained multiple times for different combinations of noise, text, and graphics.

Zhu et al. [11] proposed a Cycle-Consistent Adversarial Network (CycleGAN), an unsupervised image to image translation model which does not require paired training data.
and provides solution to the problems of limited paired datasets. Additionally, CycleGAN has been demonstrated as a capable and flexible denoising method for images with complex noise by Song [14] in denoising complex noisy satellite images into clean satellite images.

Inspired by the framework and capability of CycleGAN, this paper proposes a framework of a modified cycle consistency generative adversarial networks, dubbed ‘Multi Noise CycleGAN’ for flexible engineering drawing document binarization catering to randomized noise.

The targeted contributions of this paper can be summarized as follows.

- The proposed method focuses on the multi degradation and noise with limited training data availability.
- This framework could be exploited and adjusted to address any complex document degradation problem.
- The proposed hybrid method focuses on addressing current industrial challenge, digitization of historical engineering drawing.

The rest of this paper is organized as follows: Section II reviews associated literature and related works on document binarization and document noise removal. Section III presents the proposed Multi Noise CycleGAN, and the modifications made on the CycleGAN. Section IV presents the experimental results, performance analysis and comparison against other solutions. Section V concludes the paper.

II. RELATED WORK

A. Document Binarization

Research studies covering document image binarization have been conducted and reported in the literature over the past decade. Document Binarization is the process of discarding unnecessary noisy information from documents to preserve meaningful text for text recognition and text extraction. Text in scanned documents is more dense than natural images to aid doctors in receiving more details from the pictures and increase the diagnosis accuracy.

Inspired by the framework and capability of CycleGAN, this paper proposes a framework of a modified cycle consistency generative adversarial networks, dubbed ‘Multi Noise CycleGAN’ for flexible engineering drawing document binarization catering to randomized noise.

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II. RELATED WORK

A. Document Binarization

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To drive the progress of document binarization, Document Image Binarization Content (DIBCO) was introduced. Document Image Binarization Content (DIBCO) is both a document database meant to challenge and serve as a general benchmark for researchers in the field of document binarization with a new document database provided by a specialist every year and a contest of binarization. Chen [7] highlights deep learning models have become the state-of-the-art in document binarization where in 2017, all top six submissions to DIBCO have utilized deep learning model. The document database contains several degradation types that commonly occur in historical documents [7], ranging from stains, bleed through ink, stamps, multiple colour text and border noise.

B. Supervised Deep Learning Document Denoising

Active studies on supervised deep learning document denoising focuses on further improving the accuracy of document binarization alongside covering more degradation and noises within a single document. Souigui [8] proposes Document Enhancement Generative Adversarial Networks (DE-GAN) to clean and restore severely degraded document images. The DE-GAN reported successful and accurate in addressing blurriness, watermark and ink stain among other noises as well. Most recently, Suh [9] proposes a two stage GAN architecture, addressing document binarization with coloured background, watermark and ink bleed achieving higher results than most state-of-the-art solutions. The architecture proved successful and accurate for shipping labels as well. Both methods are tested on DIBCO datasets for standardization; results at the time of paper publish showcased improvement over other state-of-the-art methods.

Furthermore, Chung [10] proposed a CNN-based binarization for historically degraded drawing maps. Over the years, there have not been many studies and research covering complex documents such as historical drawing maps. Chung [10] CNN based binarization method is an improvement over existing works covering the same document domain, binarization of complex documents consisting of yellowing noise, folded noises and preserving components. Recent works in supervised deep learning document denoising showcases the maturity of methods and algorithms in tackling standardized document binarization test. However, few works address document binarization for complex documents.

C. Cycle consistency Generative Adversarial Network (CycleGAN)

To eliminate the reliance on paired data for training, several unsupervised learning-based methods have been developed for image enhancement inclusive of image denoising. Among them, is a general-purpose method, unsupervised image to image translation model, cycle consistency generative adversarial network (CycleGAN) proposed by Zhu [11]. CycleGAN allows for the generated image to be as close as target image without paired data and supervision. CycleGAN is derived from generative adversarial networks [12] which utilizes adversarial training, pitting two neural networks against one another. The study of GAN has gained a lot of traction and many variations have been developed to address different problems in the field of computer vision and image processing.

D. Application of GAN and CycleGAN in Image Denoising

The utility of GANs and CycleGAN as an image denoiser has been researched and tested across different industry. Xiong [13] has developed a two stage GAN for low light image enhancement.

In the medical field, due to the intense difficulty of obtaining paired images, modifications were made to cycleGAN as a means to obtain higher resolution and cleaner images to aid doctors in receiving more details from the pictures and increase the diagnostic accuracy. Khan [15] opt
for optimal transport cycleGAN to reconstruct high resolution MR images while Lee [16] utilizes attention guided – β CycleGAN to remove metal artifacts from CT images. The works above shares the common goals in removing random noise while preserving the original image through unsupervised image denoising and with limited training data. In the next subsection, past works incorporating CycleGAN ability as an image denoiser at the domain of document binarization is reviewed.

E. Application of cycleGAN in Document Binarization

In the areas of document denoising and binarization, Sharma [17] has applied CycleGAN to the denoising of scanned documents focusing on the removal of watermark, blurriness and background noise removal. Utilizing the design of CycleGAN, Kumar [18] addresses the works of Bhunia [19] for text extraction from documents as the original works of Bhunia [19] requires paired training data, through the development of UDBNET architecture which adds another layer unto the architecture of Bhunia [19] and merging them with a joint discriminator to allow for accurate text extraction without paired data. Tensmeyer [20] extends the CycleGAN model to trained on ground truth binarization masks to produce realistic synthetic data for document binarization of DIBCO datasets.

From recent works, unsupervised document denoising has been tested and refined on standardized degraded documents. However, there exist a research gap in document binarization of complex documents such as technical drawing.

F. Conceptual Research Framework

Fig. 1 provides the framework for a disciplined research approach to the factors affecting OCR performance.

The conceptual framework above highlights the factors and their weightage that contributes to the overall OCR engine performance. Realistically every document requires a certain level of denoising to allow the OCR to accurately extract the underlying text. While there are four main variables affecting the denoising layer, 2 of them, noise type & amount and training data availability has a higher weightage because these are not easily controlled by the user. Commonly used metrics to evaluate OCR performance are Character Error Rate (CER) and Word Error Rate (WER) [24]. [25]. CER and WER indicate the amount of character for CER and word for WER that the applied OCR did not read or generate correctly.

Noise in images can be present due to many factors, environmental, condition of the digital capture device and the condition of the source item. The randomness of the noise has a direct effect unto the training data availability as rarer and more complicated noise generally do not occur often and a sufficient number of data is required to effectively train the denoising algorithm.

The availability of training data is second factor with higher weightage with direct impact to the selection of algorithm training method. In an ideal scenario, even with rare noise type, so long as paired training is available, supervised deep learning denoising method can be utilized. However, that is not often the case. While more mainstream and widespread documents or images have high volumes of data available, certain images especially those in technical areas, engineering, manufacturing or medicine, paired training data are difficult and sometimes impossible to obtain.

As a summary, the critical point for good OCR engine performance lies in the training data. There are very few works that have addressed lack of training data for complex documents. Failure to successfully binarize and conduct text extraction on these documents will result in loss of information as there are enormous number of complex documents from before the digital era.

III. PROPOSED METHOD

Image denoising targets to restore a clean image from a noisy one to improve the overall quality of the image for better OCR result. The problems of engineering drawing binarization is treated as an image-to-image translation task where the objective is to produce and generate clean document images and preserve the original content from its noisy counterpart.

CycleGAN has shown to be capable in addressing image to image translation problems in situations there is a lack of paired training dataset. With CycleGAN specific capability of able to work without the constraint of one-to-one mapping between the input image and target image whilst keeping its ability to learn such image-to-image translations, has convinced us to investigate and modify the framework accordingly as a denoising layer for engineering drawings as it is extremely difficult and limited to obtain clean engineering drawings alongside its corresponding noisy counterpart. In Fig. 2 the overview the proposed framework is highlighted.
CycleGAN leverages cycle-consistency loss to get around the problem of learning meaningful transformations in unpaired datasets. This means that if an image is converted from source distribution to target distribution and back, samples can be acquired from source distribution. The cycle consistency loss is incorporated via CycleGAN two generators and two discriminators.

The first generator, Gē, converts an image taken from noisy domain, Xn, to produce an output image in the targeted clean domain, Xcb. To promote and enforce effective relation between noisy and clean images, CycleGAN must be able to learn the necessary features that can be incorporated to turn back generated clean images to its original noisy counterpart. A similar process but in reverse takes place with the second generator, Gēn, to convert clean images, Xcb, to noisy images.

The responsibility of the discriminator is to be able to identify real and fake images generated by the generators ideally as to defeat the generator via rejecting the images produced by it. The generator and discriminator engage in a competition like manner till the generator is able to produce images that are indistinguishable from the original input images. This is expressed as adversary loss. The proposed, ‘Multi Noise CycleGAN’, utilizes a similar network structure to that of the original CycleGAN by Zhu [11] with modification made to the existing network which is as follows:

1) 15 ResNet blocks used to build the Generator Architecture as shown in Fig. 3.
2) Individual learning rate for Generator and Discriminator as opposed to same.

![Fig. 3. Overview of modified CycleGAN generator architecture.](image)

Full objective loss is as follows:

\[ L(G_N, F_L, D_N, D_L) = L_{GAN,N}^p(G_N, D_N, X, Y) + L_{GAN,L}^p(F_L, D_L, Y, X) + \lambda L_{cyc}(G, F) + L_{identity}(G, F) \]  \hspace{1cm} (1)

IV. EXPERIMENTS AND RESULTS

The following subsections describe the experiments that have been conducted using ‘Multi Noise CycleGAN’. The proposed dataset, training images and test images alongside the evaluation metrics are described as well.

A. Datasets’ Details

In this paper, the experiments are conducted on the noisy engineering drawing dataset type proposed by Chung [10]. It consists of 35,960 paired images containing noisy background, yellowed area, and folded lines and their respective clean image. The dirty images are collected and compiled from real degraded as built drawing maps. The noises are noisy background, yellowed area, and folded lines. The dirty images contain varying degrees of noise level, image clarity and complexity of text and graphics.

For the training phase, 5,083 unpaired dirty images with 256x256 size and 5,063 clean images 256x256 size is used to train ‘Multi Noise CycleGAN’.

For the testing phase, to evaluate the overall performance of unsupervised image to image translation, 3,200 pair of images that best represents the structure of engineering drawings containing engineering drawing symbols and diagrams were selected (dirty images and their corresponding clean versions) from the 35,960 paired images and does not include images used during training.

The 3,200 testing datasets are a mixture of four different noises namely as follows:

1) Noisy / Complex background.
2) Less Noisy Background.
3) Noisy Background with discoloration.
4) Noisy Background with poor illumination.

B. Evaluation Metrics

In this paper, to compare and evaluate the performance of unsupervised image denoising with other binarization method, two metrics are used. Structural Similarity Index (SSIM) and Peak Signal to Noise Ratio (PSNR).

The equation for SSIM is as follows:

\[ SSIM(x, y) = \frac{(2\mu_x\mu_y + c_1)(2\sigma_{xy} + c_2)}{(|\mu_x^2 + \mu_y^2 + c_1)(\sigma_x^2 + \sigma_y^2 + c_2)} \]  \hspace{1cm} (2)

Where SSIM is the Structural Similarity Index: (x,y) are respective coordinates indicating a nearby NxN window: \( \sigma_x \) , \( \sigma_y \) are the variances of intensities in x,y directions, \( \sigma_{xy} \) is the covariance and \( \mu_x, \mu_y \) are the average intensities in x,y directions.

The equation for SSIM is as follows:

\[ PSNR(I_{original}, I_{clean}) = 10 \log_{10} \frac{255^2}{MSE} \]  \hspace{1cm} (3)

Where MSE is defined as the Mean Square Error. \( I_{original} \) is the original image and \( I_{clean} \) is the clean, denoised version. MSE is defined as follows:

\[ MSE = \frac{1}{mn} \sum_{i=0}^{m-1} \sum_{j=0}^{n-1} (I_{original}(i, j) - I_{clean}(i, j))^2 \]  \hspace{1cm} (4)

Where m and n respectively are the size of the input images.

For the testing phase, 3,200 pair of images outside of the training data is selected (dirty images and their corresponding clean versions) is used to evaluate the overall performance of unsupervised image to image translation.

C. Experimental Results

Fig. 4 highlights three examples when ‘Multi Noise CycleGAN’ method is applied unto the test dataset. From the
end results, from a visual standpoint, the image is cleaned to a satisfactory level however it is challenging to differentiate a real image from a generated one when the differences between those two are considerably small. As such, quantitative comparison on the proposed method is applied using the aforementioned two metrics: Structural Similarity Index (SSIM) and Peak Signal to Noise Ratio (PSNR).

Fig. 4. Multi Noise CycleGAN. From left to right: noisy image, result of Multi Noise CycleGAN.

Guided by the two-evaluation metrics defined above, Table I as shown below summarizes the average results across 3,201 images comparing the performance of Multi Noise CycleGAN against other state-of-the-art methods.

From the results above, ‘Multi Noise CycleGAN’ on average has achieved PSNR of 15.828 db, and SSIM of 0.806. Therefore, it is shown that the proposed method has achieved comparable results in PSNR and SSIM against state-of-the-art methods with the added advantage of not having to use paired images for the training phase.

Table II highlights few test cases for 1 noise category where the proposed ‘Multi Noise CycleGAN’ achieves better results in terms of PSNR/SSIM against state-of-the-art method indicating an area of image denoising where it is more advantageous to utilize the proposed ‘Multi Noise CycleGAN’ method.

It can be observed that the proposed method from a visual perspective as shown in Table II generates a clean image whilst retaining important image details as well without any additional artifacts created.

<table>
<thead>
<tr>
<th>Method</th>
<th>PSNR</th>
<th>SSIM</th>
</tr>
</thead>
<tbody>
<tr>
<td>Sauvola [21]</td>
<td>15.418</td>
<td>0.883</td>
</tr>
<tr>
<td>Niblack [22]</td>
<td>8.236</td>
<td>0.352</td>
</tr>
<tr>
<td>Otsu [6]</td>
<td>16.139</td>
<td>0.769</td>
</tr>
<tr>
<td>Jia et al [9]</td>
<td>17.483</td>
<td>0.839</td>
</tr>
<tr>
<td>Isola et al [3]</td>
<td>18.798</td>
<td>0.869</td>
</tr>
<tr>
<td>Multi Noise CycleGAN</td>
<td>15.828</td>
<td>0.806</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Test Case #1 Noisy / Complex Background</th>
<th>Method</th>
<th>Result</th>
<th>PSNR</th>
<th>SSIM</th>
</tr>
</thead>
<tbody>
<tr>
<td>Noisy Image</td>
<td>-</td>
<td>-</td>
<td>-</td>
<td>-</td>
</tr>
<tr>
<td>Sauvola [21]</td>
<td>14.464</td>
<td>0.908</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Niblack [22]</td>
<td>7.777</td>
<td>0.287</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Otsu [6]</td>
<td>12.822</td>
<td>0.691</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Jia et al [23]</td>
<td>3.496</td>
<td>0.078</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Isola [3]</td>
<td>14.931</td>
<td>0.830</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Multi Noise CycleGAN</td>
<td>16.527</td>
<td>0.915</td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Test Case #2 Noisy / Complex Background</th>
<th>Method</th>
<th>Result</th>
<th>PSNR</th>
<th>SSIM</th>
</tr>
</thead>
<tbody>
<tr>
<td>Noisy Image</td>
<td>-</td>
<td>-</td>
<td>-</td>
<td>-</td>
</tr>
<tr>
<td>Sauvola [21]</td>
<td>13.638</td>
<td>0.855</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Niblack [22]</td>
<td>8.954</td>
<td>0.390</td>
<td></td>
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</tr>
</tbody>
</table>
This paper presents a novel model that utilizes Cycle consistent adversarial networks for complex image denoising. The method, dubbed ‘Multi Noise CycleGAN’ architecture utilizes and modifies CycleGAN to the task of complex image denoising. The method is able to both denoise simple and complex noise due to the inherent property of CycleGAN of cycle-consistency. It is also noteworthy that the method uses unpaired images as training dataset, and therefore does not require ground-truth clean images.

Results show that the proposed method achieves denoising results comparable to state-of-the-art methods. The values of PSNR and SSIM obtained are at same level with state-of-the-art method found in the research literature with some test cases showcasing where the proposed method is overall better.

Future work is targeted to test the feasibility of ‘Multi Noise CycleGAN’ in tackling additional document binarization problems such as blur and watermark. Improvements to the architecture, to accommodate the addition of blur and watermark noises will allow for even better flexibility of noise removal, improving the accuracy of document binarization of engineering.

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Light Weight Circular Error Learning Algorithm (CELA) for Secure Data Communication Protocol in IoT-Cloud Systems

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Abstract—The data driven smart applications, utilize the IoT, Cloud Computing, AI and other digital technologies to create, curate and operate on large amounts of data to provide intelligent solutions for day-to-day problems. Security of Data in the IoT-Cloud systems has become very crucial as there are several attacks such as ransomware, data thefting, and data corruption, causing huge loss to the application users. The basic impediment in providing strong security solutions for the IoT systems, is due the resource limitations of IoT devices. Recently, there is an additional threat of quantum computing being able to break the traditional cryptographic techniques. The objective of this research is to address the bifid challenge and design a light weight quantum secure communication protocol for the IoT-Cloud ecosystem. The Ring Learning With Errors (RLWE) lattice based cryptography has emerged as the most popular in the NIST PQC Standardization Program. A light weight Circular Learning Error Algorithm (CELA) has been proposed by optimizing RLWE to make it suitable for IoT-Cloud environment. The CELA inherits the advantages of quantum security and homomorphic encryption from RLWE. It is observed that CELA is light weight in terms of execution time and a slightly bigger cipher text size provides higher security as compared to RLWE. The paper also offers plausible solutions for future quantum secure cryptographic protocols.

Keywords—Quantum secure cryptography; homomorphic encryption; lattice-based cryptography; Learning With Errors (LWE); Ring Learning With Errors (RLWE); Circular Error Learning Algorithm (CELA)

I. INTRODUCTION

Smart applications, such as smart traffic management, analyzing customer spending habits, smart homes surveillance, or emergency response for remotely monitored patients, provide intelligent solutions by analysing data from multiple information sources. All these applications are dependent on data. Data is essential for evaluation of performance, prediction of future trends, intelligent decision making and controlling risks. The smart application ecosystem has emerged as a natural confluence of IoT, Cloud Computing, Big Data and AI which provide intelligent automation for various real-life problems.

The IoT devices are generating huge amounts of data, but since the sensor-actuator IoT devices have low compute capabilities, the data captured by IoT is sent to Cloud for storage and processing. The IoT-Cloud ecosystem is providing a strong foundation to develop smart applications, in almost all possible domain. Real time processing, underlying infrastructure of servers, storage and communication, and operations, are some key aspects of the IoT-Cloud ecosystem [2].

The amount of data in the world is expanding at a very rapid pace. The survey by Statista [3] shows that the quantity of data created, captured or consumed worldwide in previous decade was at 2 zetabytes (ZB) as on 2010, and in this decade it crossed 64ZB in 2020; and is predicted to reach 180ZB by 2025. And nearly 49% of the world’s stored data shall be hosted in public cloud environments. Corrupting the data, data theft and denial of access etc. have become a means of livelihood for hackers. Several attacks are being reported on the data driven systems [4]. Concerns about the safety of data were brought up in [5] and [6]. Data has to be protected against access by any unauthorized persons both during transmission over network and while being stored in any devices. An infrastructure built on the Internet of Things must include standards and services that are necessary for protecting, managing, and connecting the various Internet of Things devices and applications.

Data Security, as defined by NIST (National Institute of Standards and Technology, USA), is the process of maintaining confidentiality, integrity, and availability of an organization’s data in a manner consistent with the organization’s risk strategy [7]. According to IBM [8], Data Security is the technique to protect digital information from unauthorized access, corruption, or theft, throughout its entire life-cycle from data generation, data transportation over network, data storage and retrieval.

Cryptographic techniques are well established methods for securing data both during transit as well as storage, for a wide range of applications over traditional computers, IoT devices and the Internet. However, there are several challenges encountered while securing the IoT-Cloud ecosystem, such as: (i) resource constraints of IoT devices, (ii) size of crypto keys, i.e. the strength of cryptographic scheme, (iii) shared access by multiple users, (iv) real time response for the applications, (v) increased threats due to development of Quantum Computing etc. These aspects make traditional protocols vulnerable and impracticable, prodding researchers to develop improved cryp-
tographic techniques.

The most important aspect of any cryptographic technique is to make sure that the encryption cannot be cracked by adversaries. At present, the public key cryptographic algorithms are based on mathematical methods such as discrete logarithm problem (ECC, DHKE) and prime factorization of large integers (RSA), which are difficult to solve in traditional computers.

The RSA public key cryptography is based on modulus arithmetic using very large integers and cycling the data in multiple rounds with the public key; and the decryption is based on finding the prime factors of the very large number. Breaking the RSA-1024 by brute-force would take approximately $2^{1024}$ comparisons, i.e. a total compute operations of about $10^{264}$ years, on a single CPU system. However, in 2010, the 768-bit RSA, was cracked in about a week by using the large computational capacity available in a supercomputer cluster. Now, with the emergence of quantum computing Shor’s algorithm (which is capable of finding the prime factors of an integer, at an exponentially lesser time), the need to develop quantum resistant cryptographic algorithms has been felt.

While several strong cryptographic methods are envisaged, the IoT devices are resource-constrained with limited compute and storage making it impractical to perform encryption and decryption in real-time and also for storing the encrypted data. Hence, light weight cryptographic methods are required. Though conventional encryption techniques, would serve the purpose of protecting the data, the modern IoT-Cloud applications require a multi-user shared, searchable and processable data for quick analytics and control. To cater to these requirements, new methods for encrypted data query and processing have been proposed for the IoT-Cloud systems.

Recently, the National Institute of Standards and Technology (NIST) announced the Post Quantum Cryptography (PQC) Standardization Program open call to evaluate and standardize public-key cryptographic algorithms for securing against the emerging quantum attacks. Among the 50 cryptographic proposals that were short-listed, 21 were based on lattice techniques; and in that, the RLWE algorithm turned out to be the most popular with a count of 15 entries. The lattice based crypto techniques are prime candidates as they have strong security, compute efficiency, wide applicability, homomorphic encryption, and quantum resistance. The NewHope is a fast, memory efficient, and simple scheme that is a suitable replacement of RSA and ECC. It provides homomorphic encryption, reasonable sized key and ciphertexts and significantly resistant to side-channel attacks. Internally it uses the Ring-LWE lattice-based technique.

1) Contribution of this paper: The aim of this research is to propose a highly secure yet light weight protocol for the IoT-Cloud ecosystem. The primary contributions of this paper include:

(i) Lightweight Cryptography: The proposed CELA has lower time complexity on account of modulus switching technique which reduces the polynomial ring order in intermediate computations.

(ii) Improved Security: Enhanced security is provided by increasing the key size to twice that of the base RLWE algorithm and using random noise vector. The ciphertext is of bigger size and more complex.

(ii) Plausible Solutions for Quantum Secure Cryptography: Since, the number theory based cryptographic algorithms are susceptible to be cracked by quantum computers, robust security schemes are suggested towards quantum secure cryptography.

2) Organization of the Paper: Organization of the paper is as follows. The related work is summarized in Section II. A brief description of various cryptographic techniques is presented in Section III. The IoT-Cloud architecture and smart applications are described in Section IV. Section V presents the security challenges in IoT-Cloud ecosystem. The design and implementation of the proposed CELA is presented in Section VI. Plausible solutions for quantum secure cryptography in resource constrained IoT-Cloud ecosystem are provided in Section VII and the conclusion is contained in Section VIII.
The more recent efforts in lattice-based and LWE schemes provide strong security for lightweight devices and PQC. Adaptations and implementation in both hardware and software for the Lattice-based Cryptographic schemes, have been demonstrated by AK Sahu et al. [30], N Hamid et al. [31] and Wai-Kong Lee et al. [32]. While these schemes offered homomorphic encryption resistant to post quantum cryptopgraphic attacks, there is scope for improving their performance for using in real-life IoT-Cloud applications.

A summary of latest research in Learning With Errors Crypto Schemes is available in - Xie et al. [33], Kundi et al. [34], Jose [35], Xiaodan Xi et al. [36], Pengzhou He et al. [37], Shahriar Hadayeghparast et al. [38], and Dongdong Xu et al. [39], have optimized the LWE encryption algorithms and implemented them in hardware. The main aim of these efforts is to develop lightweight, low power, low silicon-area ∗ delay, i.e. compact and high speed, yet stronger encryption hardware which can be used inside context devices.

Optimizations and implementation of the RLWE scheme on IoT compatible low power RISC and ARM processors, are presented in Ebrahimi et al. [40], Zhe Liu et al. [41] and Zhang et al. [42].

Various quantum resistant security schemes based on XMSS, HBS, and Quantum-AES for IoT-Cloud ecosystems are offered in the works of WK Lee et al. [43], Santosh Ghosh et al. [44] and Roopa Golchha et al. [45]. The qualifying PKE algorithms of the NIST PQC Standardization program are described in [46].

The comparison of research articles published in the last five years in the area of lattice-based homomorphic post-quantum resistant cryptographic schemes is captured in Table I.

While lattice based schemes like RLWE are found to be quantum resistant and homomorphic, they require to be optimized to be practicable for compact resource-constrained IoT devices. Improving the computational speed and key size may make them amenable for real-world IoT-Cloud applications.

III. PRELIMINARIES

Protecting the data from malicious and inadvertent problems is an important task. The various methods used to protect the Confidentiality, Integrity and Privacy of data include - Authentication and access control, Data Masking, Cryptographic techniques, Steganographic techniques, Differential Privacy techniques and Blockchain.

A. Classification of Cryptographic Schemes

Cryptography is a mature and popular data security mechanism employed for several Internet applications including IoT and Cloud. The variations in cryptography are revealed in the classification [49] shown in Fig. 1.

B. Cryptographic Schemes

(i) Symmetric Key Cryptography: This is an encryption system where the sender and receiver have a single common key for enciphering and deciphering the messages. Symmetric Key Systems are simple and fast, but the problem is that sender and receiver have to somehow exchange the key in a secure way. E.g. Data Encryption System (DES).

(ii) Asymmetric Key Cryptography: Under this system the public key of the receiver is used to encrypt the data and send on the channel; and the receiver can only decrypt with his private key. Hence there is no need for key transfer. E.g. RSA, ECC and ElGamal Cryptosystem.

(iii) Key Exchange: The Diffie Hellman Key Exchange (DHKE) provides a mathematical protocol for exchanging the secret key between the sender and the receiver over a potentially non-safe communication channel. The two parties involved in communication, Alice and Bob, start by agreeing on a finite field \( F_q \) and a primitive root \( g \), i.e. a generator of the cyclic multiplicative group \( F_q^*,F \) where the field \( F \) is an algebraic set with two operations (addition and multiplication) and \( q \) is the modulus. The pair \( (q, g) \) is published. To obtain the private key, Alice selects an integer \( a \) and Bob selects an integer \( b \). Then, Alice sends \( g_a \) modulo \( q \) to Bob, who on receiving it, raises it to \( b \) modulo \( q \) getting \( g_{ab} \) modulo \( q \). Next, Bob sends \( g_b \) modulo \( q \) to Alice, who raises it to \( a \) modulo \( q \), and hence cannot get \( a \) or \( b \) secret keys to crack the cipher text.

(iv) Identity Based Encryption: The parties involved in communication identify themselves using a unique identifier such as an email address or user name as their public key. The data is encrypted using the recipient’s email address or username. Some popular Identity Based Encryption (IBE)
<table>
<thead>
<tr>
<th>Author/Concept/Model</th>
<th>Algorithm Implementation</th>
<th>Performance Advantages</th>
<th>Research Gaps/Challenges</th>
</tr>
</thead>
<tbody>
<tr>
<td>Ak Sahu et al. [39], 2021</td>
<td>Lightweight Multi-Party Authentication and Key Agreement for IoT e-Healthcare</td>
<td>- Lowered Power consumption</td>
<td>Include Privacy of the user data in Cloud server</td>
</tr>
<tr>
<td>N Hamid et al. [31], 2020</td>
<td>FPGA implementation of Ring LWE based crypto techniques</td>
<td>- Improved security by integrating quantum cryptographic approach and classical AES</td>
<td>Evaluate the accelerator for lattice-based post-quantum algorithms</td>
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<tr>
<td>Wai-Kong Lee et al. [33], 2022</td>
<td>Imple-mented Lattice NTRU polynomial convolution in GPU Tensor cores</td>
<td>- Hardware solutions are twice faster and energy efficient.</td>
<td>Implement on gateway device in IoT</td>
</tr>
<tr>
<td>Xie et al. [35], 2020</td>
<td>Hardware implementation for BRLWE</td>
<td>- The area-time complexities and area-delay product of this design is superior compared to the existing designs, namely #LUT, #FF, #slice.</td>
<td>Extend the hardware for future applications</td>
</tr>
<tr>
<td>Y Chen et al. [47], 2021</td>
<td>On the fly keys generation for LWE based Multi-Key FHE</td>
<td>- Beforehand fixing of number of keys is not required.</td>
<td>Presently the MK-FHE is single-bit encryption, plan to extend to multi-bit</td>
</tr>
<tr>
<td>Yang Su et al. [43], 2020</td>
<td>FPGA-Based Hardware Accelerator for Levelled RLWE FHE</td>
<td>- Hardware Area saving of 50%</td>
<td>Extend to support wider range of security parameters</td>
</tr>
<tr>
<td>Kundu et al. [35], 2022</td>
<td>Approximate RLWE implementation on FPGA for IoT</td>
<td>- The Xilinx Kintex-7 based FPGA AxRLWE designs provide up to ∼ 64% area saving compared to the accurate R-LWE.</td>
<td>Implement PQc within the tight area/energy budgets of the highly resource-constrained IoT devices</td>
</tr>
<tr>
<td>S. Ebrahimis et al. [49], 2020</td>
<td>Fault-Resilient Software implementations of binary Ring-LWE on lightweight IoT microcontrollers</td>
<td>- 80ms for encryption and 120ms for Decryption - Resistance to first order attacks- randomization, zeroing, skipping faults - Implementation on tiny microcontrollers suitable for IoT</td>
<td>Manage larger key and ciphertext sizes</td>
</tr>
<tr>
<td>Zhe Lin et al. [41], 2020</td>
<td>Software Implementation of RLWE on ARM NEON and MSP430</td>
<td>- The implementation is 7x faster than ECC of the same security level</td>
<td>Practical application in IoT devices</td>
</tr>
<tr>
<td>Jose L et al. [35], 2022</td>
<td>Hardware implementation with optimized arithmetic for InvBRLWE</td>
<td>- LFSR optimizes the resource usage for heavy computations - 71.23% less (area x delay) product than recent designs</td>
<td>Deploy for real world application environments</td>
</tr>
<tr>
<td>Xuandao Xi et al. [44], 2023</td>
<td>Hardware implementation of physical unclonable function (PUF)</td>
<td>- Resistant to ML attacks on both classical and quantum computers - Prototyped with 2136 challenge response pairs (CRPs)</td>
<td>Validate the prototype Lattice PUF hardware in field applications</td>
</tr>
<tr>
<td>Pengzhou He et al. [32], 2022</td>
<td>RBLWE-based PQC accelerators on the FPGA</td>
<td>- RBLWE Encryption - Implemented on Xilinx Virtex-7</td>
<td>Enhance for side-channel attack resistance</td>
</tr>
<tr>
<td>ShabirY Hadegeybrahast et al. [38], 2022</td>
<td>Lightweight co-processor based on RISC-V for InvBRLWE</td>
<td>- 51% faster than RBLWE - Implemented on Xilinx FPGA</td>
<td>Prevent side channel attacks like intentional fault injections and Differential Power Analysis</td>
</tr>
<tr>
<td>Dongdong Xu et al. [39], 2022</td>
<td>New scheme of ring-BinLWE based on 2’s complement ring</td>
<td>- Lighweight implementation on Spartan 6 FPGA - Compact design; less hardware area</td>
<td>Improve resistance to side channel attacks such as Differential Power</td>
</tr>
<tr>
<td>Tan Zhang et al. [42], 2020</td>
<td>Side-Channel Analysis and Countermeas-ure Design on ARM-Based Quantum-Resistant SIKE</td>
<td>- Achieved higher security along with lower time and memory</td>
<td>Eliminate vertical leakages and resist SCA Attacks</td>
</tr>
<tr>
<td>W K Lee et al. [33], 2022</td>
<td>The DPCrypto accelerates PQC using Dot-Product Instructions on GPUs</td>
<td>- Lattic based KEM FrodoKEM and SaberKEM implemented on NVIDIA V100 and T4 GPUs using Dot-Product instructions for matrix, hash and other operations in the algorithms</td>
<td>Execution speed on T4 is less compared to V100 because of the inherent number of cores in each type of GPU</td>
</tr>
<tr>
<td>Santosh Ghosh et al. [41], 2018, A Lightweight Post-Quantum Secure Digital Signature Approach for IoT Motes</td>
<td>Hash Based Signature (HBS), XMSS Scheme, Kecckak-400 hash function, WOTS+, SHA-3</td>
<td>- Novel XMSS signature scheme has a small memory footprint - FPGA implementation takes 4.8 mil- lion clock cycles, 5x faster than software</td>
<td>Extend the novel signature approach as a ultra light weight end-to-end IoT security</td>
</tr>
<tr>
<td>Roopa Golchha et al. [35], 2023</td>
<td>Quantum AES</td>
<td>- Improved security by integrating quantum cryptographic approach and classical AES - About 10 to 21 Qubits are needed for encryption and decryption of messages</td>
<td>Building the quantum circuits is challenging due to noise</td>
</tr>
</tbody>
</table>
algorithms are Boneh–Franklin IBE, Cocks IBE, Gentry IBE and Waters IBE. Conceptually, the different types of IBE include - Basic IBE, Certificateless IBE, Hierarchical IBE, Revocable IBE and Dual-Receiver IBE.

(v) Attribute Based Encryption: The secret keys required to decrypt the cipher text are related to users’ attributes such as roles, age, location, etc. The encrypted data can only be decrypted by users who have the specific set of attributes required to access the data. The main types of ABE schemes are - Key-policy ABE (KP-ABE) and Ciphertext-policy ABE (CP-ABE).

(vi) Proxy Re-encryption: Proxy Re-encryption (PRE) is a mechanism to convert the ciphertexts for one key into ciphertexts for another by using a Pproxy. An encrypted data item is stored in the Cloud and the Proxy decides to re-encrypt it and send to legitimate users. This delegation scheme is useful for resource constrained IoT devices as the computationally intensive encryption are outsourced to the proxy server. It is useful for applications such as e-mail forwarding, content distribution and law-enforcement monitoring.

(vii) Homomorphic Encryption: Homomorphic Encryption (HE) encodes data into ciphertext which can be analyzed and worked with as though it were still in its original form. HE will enable users to perform computations on the encrypted data without first decrypting it [50]. HE is expected to play an important role in Cloud based applications (e-voting, health, finance), allowing companies to encrypt and store sensitive data in a Public Clouds and take advantage of the Cloud Provider’s analytic services. In 2009, Gentry proposed with proven safety, a fully HE cryptogram technique; this algorithm relies primarily on learning with error (LWE).

(viii) Verifiable Computation: Verifiable computation is a method where a device having limited computing capabilities makes a request to outsource computation services which it is unable to undertake on its own, and receives assistance from another source. The result of using outsourcing is that the results can be delivered in a time- and cost-effective manner [51]. The verifiable computation does not involve interaction between users who have had their identities validated and protects users’ privacy by the comparison and verification of the values that are input and output.

(ix) Searchable Encryption: Searchable encryption is designed to boost the efficacy of searching by adding an index for searching specific information whilst still considering the safety of encrypted data. This method consists of - key generation, encryption, generation of trapdoors, and the search test process. During the stage of key generation, the user is generating as well as saving a private key, while also providing other users with a public key. The data are encrypted, and an index is too generated in order to search for keyword information within the data. By using the private key, the user creates a trapdoor for the term. The data supplied by the sender can be located by the receiver by using the trapdoor [50].

(x) Multiparty Computation: Present day collaborative applications in cloud computing desire the ability to exchange encrypted data with other users whilst also protecting their privacy and retaining their confidentiality. Multiparty computing (MPC), is a cryptographic technique that enables multiple parties to jointly perform a computation on their private data without revealing their inputs to each other. The idea is to divide the computation into smaller sub-computations that can be performed separately by each party using their private inputs. These intermediate results generated by the sub-computations are then combined in such a way that the final result can be obtained without any party having access to the private inputs of the other parties.

(xii) Quantum Encryption: This is an extremely secure encryption based on quantum mechanics, but is not yet widely available due to the technical requirements and high cost of quantum computers. Various types of Quantum Cryptography [52] are evolving such as Quantum Random Number Generation (QRNG), Quantum Key Distribution (QKD), Quantum Cryptography for Authentication, Quantum Digital Signatures and Quantum Secret Sharing.

C. Lattice-based Cryptography

In mathematics, Lattice is a multi-dimensional grid of points that are spaced at regular intervals. Since, lattices are flexible, and posses features required for obfuscation, functional encryption and homomorphism, they are being used for encryption and decryption in the lattice based cryptographic algorithms. Two key concepts about lattice are - Vector and Basis. A vector is a tuple of numbers called the coordinates of the vector, indicating the starting and ending points. Basis is a pair of vectors which can produce other points in the lattice, when added with linear integer combinations of these basis vectors.

Mathematical problems related to the lattice are used to obtain the secret from the public key. The Shortest Vector Problem, is to determine a non-zero vector in the given lattice space whose length is minimal over all non-zero lattice vectors. The Closest Vector Problem is a reduction problem to determine a vector in the lattice which is closest to the target vector. Lattice based mathematical techniques and its variations lend themselves to quantum resistant cryptography. This section explains the lattice based methods.

D. Learning With Errors (LWE)

In LWE-based encryption, the data or message is encrypted by adding a random noise (or error) value to it; the resulting ciphertext is sent over an insecure communication channel. The idea is that even if the attacker tries to learn the secret key by solving a set of equations using the ciphertext and the function, the error component makes it difficult to solve the equations and recover the key. LWE can be mathematically stated using the equation:

$$(B[ ] = A[ ] * s + e)$$

where (A) is a random vector $(a = (a_1, \ldots, a_n))$, sampled uniformly over a vector $(Z/Z_q)^n$, and $B$ is the public key based on $b = (a, s) + m + e$, and $s$ is the secret, the error $e$ is drawn from Gaussian Error distribution and all arithmetic is done modulo $q$. Breaking the scheme in LWE scheme is based on the hardness of lattice problems.

Without the error term, an attacker could determine the secret key from a polynomial-sized collection of LWE ciphertexts by some method like Gaussian elimination. Hence the
plain text is encoded along with the errors. For encoding, we use smaller cleartext space (actual messages) and encode cleartexts by putting the messages in the higher-order bits of the plaintext space. E.g., a 10-bit message can be encoded in the top 10 bits of a 32-bit integer, and leave the remaining 22 bits of the plaintext for the error distribution. Hence size of key is large in LWE scheme. However, the key size can be compressed by changing different parameters such as size of the noise polynomial, size of error, dropping lower-order bits, changing security level etc.

E. Ring Learning With Errors (RLWE)

The RLWE is a specialized form of LWE for polynomial rings over finite fields in which the set of polynomials have coefficients in another ring. In algebra, rings are structures that come with the basic operations of addition and multiplication and a multiplicative identity; addition operation is commutative and the multiplication is associative. The elements in the Ring may be integers, polynomials and matrices etc.

A ring $R$, denoted by $(R, +, \cdot)$ is a set of elements (integers, polynomials, matrices), with binary operations of addition and multiplication.

For a positive integer $q$, we define, $Z_q = Z/qZ = \{0, 1, \ldots, (q - 1)\}$ as the integer quotient ring, with $q$ as its modulus. The set of polynomials is denoted by $Z_q[x]$ where the coefficients of $Z[x]$ are chosen from $Z_q$. Any polynomial is uniquely identified by its set of coefficients or the coefficient vector.

A polynomial ring $R_q$ is any polynomial in $Z_q[x]$, modulus divided by a modular polynomial $f(x) = x^n + 1$. That is $R_q = Z_q[x]/f(x)$ so that no polynomial has higher rank than $n$.

The RLWE uses ring polynomial and error vectors over $Z$, sampled with discrete Gaussian Distribution ($\chi$) (\sigma) with a small standard deviation $\sigma$.

$$(B|^{m+n}|^e) A(s|^{n}|^1)$$

where $a \in Z_q^n$ are random coefficients, $s \in Z_q^n$ is the secret and $e \in \chi$ and $\chi$ is the error distribution with small standard deviation. A typical illustration of RLWE is shown Fig. 2.

$$ (B|^{7+4}|^1)^{13} = A|\begin{bmatrix} 6 & 9 \\ 0 & -1 \end{bmatrix}^{13} + s|^{14}|^{1} $$

In RLWE the keys sizes are much lesser than LWE scheme; RLWE key sizes are almost the square root of LWE key size. If we compare the key sizes for the current security level of 128 bits, the RLWE algorithm would use public keys of 7000 bits in length, whereas the equivalent LWE algorithm requires public keys of 49 million bits. Yet, when compared to the present public key schemes like RSA and ECC which offer public key sizes of only 3072 bits and 256 bits respectively (for a 128-bit security level), the RLWE key sizes are larger.

F. Modulus Switching

In a fully homomorphic encryption, one should be able to add and multiply the ciphertext as if it were in the original plaintext form. The encryption procedure for RLWE-based fully homomorphic encryption is carried out by making use of the elements that are contained within a vector space, and has an Error value in the resulting cryptogram. While performing a homomorphic operation $Eval_{pk}()$, two cryptograms for the multiplication, each having an n-dimension, the encryption expands to $n^2$.

Adding two homomorphic RLWE ciphertexts produces a resultant ciphertext as the sum of the plaintexts. Homomorphic addition does not increase the size of the ciphertexts. But, homomorphic RLWE multiplication increases the number of components in ciphertexts; that is,

$$(b_1 - a_1.s)(b_2 - a_2.s) = b_1b_2 - (b_1a_2 + b_2a_1)s + a_1a_2s^2$$

Now the ciphertext became larger and has 3 terms. The size of the cryptogram needs to be lessened. The concept of
Modulus Switching [57] can be used to contain the growth of ciphertext size.

Modulus switching is based on changing the modulus \( q \) to a different prime number \( q' \), where \( q' < q \). This removes the size of the polynomial ring and can be changed temporarily during intermediate computations.

The concept of modulus switching is explained with an example. For encoding, smaller cleartext (actual messages) are encoded as plaintext by putting the messages in the higher-order bits of the plaintext space. Say, RLWE encodes 10-bit messages in the top 10 bits of a 32-bit integer, and leaves the remaining 22 bits of the plaintext space for the error distribution.

In modulus switching, the modulus is changed from \( q \) to \( q' \), where \( q' < q \), and we would like to produce a vector \( (a'_1, ..., a'_n, b') \in (\mathbb{Z}/q'\mathbb{Z})^{n+1} \), with the new modulus \( q' \), that also encrypts message \( m \). In other words, we encrypt \( m' = ma' / q \). The operation of \( m \rightarrow ma' / q \) shifts up by \( \log_2(q') \) many bits and then shifting down by \( \log_2(q) \) many bits.

For example, say the number \( (x=6) \) requires only top 3 bits of a 32-bit unsigned integer \( ([q = 2^{32}]) \), i.e., \( (m = 6 \cdot 2^{29}) \). Let \( (q' = 2^{10}) \). Then \( (ma' / q = 6 \cdot 2^{29} \cdot 2^{10} / 2^{32} = 6 \cdot 2^7 \) \), which stores the same underlying number \( (x=6) \), but in the top three bits of a 10-bit message. By modulus switching, the data \( (x) \) is in the same position in the plaintext space, but the plaintext space around it has shrunk, thereby making extra room for error bits in the plaintext space.

The RLWE ciphertext contain a certain amount of noise or error which increases with every multiplication; finally the decryption becomes infeasible if error size passes a certain bound. When modulus switching is incorporated and homomorphic multiplication, is performed, there is space in the plaintext to allow the error term to grow towards the most significant bits of the plaintext.

Modulus switching technique involves changing the modulus of the polynomial ring during intermediate computations to reduce the computational complexity of the algorithm. This technique could significantly reduce the execution time of the algorithm while maintaining the same level of security. The idea behind modulus switching is to reduce the size of the polynomial ring temporarily during intermediate computations to make the computation faster. Then, the size of the polynomial ring shall be increased again to its original size at the end of the computation.

IV. IoT-CLOUD APPLICATION ECOSYSTEM

The smart application ecosystem shows how data is being used for intelligent automation of various real-life tasks. There is a natural convergence of IoT, Cloud Computing, Big Data and AI to provide smart solutions for many real-world problems. Fig. 4 shows the creation, capture and processing of large volume and variety of data getting ingested into the Cloud at high velocity from the network of heterogeneous devices, computers and users forming Big Data. The IoT data are stored and processed in Cloud Storage and the control decision is sent back to the actuators [1], [58], [59].

A. Architecture of IoT-Cloud Ecosystem

To meet the needs of real-time processing and control for time-critical applications, a multi-layered IoT-Cloud architecture has evolved with intermediate mist, edge and fog layers [45] as seen in Fig. 5. The main difference between the layers is in terms of the compute-storage-communication capacities and where the data is processed and stored.

(i) IoT Device Layer: The bottom layer is the IoT devices with embedded sensors and actuators. This layer contains billions of devices across various IoT networks.

(ii) Mist Layer: On top of the IoT devices is a thin Mist layer which allows to build slightly bigger IoT systems at the local level through virtual communication mechanisms. The Mist allows to execute less computationally intensive tasks, closer to the devices thereby reducing the communication latency (while cloud handles the more computationally intensive tasks).

(iii) Gateway Layer: This is a communication layer having routers for IoT to reach the edge layer.

(iv) Edge Computing: This is composed of small nodes at the edge of the network which are capable of offering routing as well as computing services.

(v) Fog computing: This is an intermediary between the Edge and the Cloud which can carry out functions such as data filtering.

(vi) Cloud Computing: It is the topmost layer. It has large compute-storage capabilities and runs many software applications for multiple users.

B. Applications of IoT-Cloud Ecosystem

The IoT-Cloud system are easily usable and suitable for almost any type of application domain [60]. There are practically hundreds of applications; only a few common ones are listed in this subsection, to provide an understanding to the reader.
(i) **Healthcare**: Simplify healthcare processes by remote patient monitoring and emergency care, ambient assisted living, creation and management of Electronic Health Records, multimedia-based health services.

(ii) **Smart Cities**: Urban area planning and management, public lighting automation, electricity management, traffic control, water management, law and order, geo-tagging services, device crowdsourcing, etc.

(iii) **Agriculture**: Automated irrigation, water distribution, improved yield, farm surveillance, crop patterns and agri-analytics.

(iv) **Manufacturing Industries**: IoT devices on the factory floor monitor critical parameters like temperature and pressure, detect defective components and remove from assembly chain, quality checks for finished products.

(v) **Wearables**: Body worn smart IoT connected to the internet, such as fitness trackers, cardio monitors, smart glasses, smart watches, virtual reality headsets, etc. having a wide range of day-to-day applications in real-life are becoming the forefront of IoT.

(vi) **Reliability**: Reliability is the property of consistently performing well. In the multi-layer IoT-Cloud ecosystem reliability is required at end point devices, data communication network, cloud infrastructure and services, and at the mist, edge, fog and so on. Reliability can be described in terms of number of users affected in unit time or Mean Time to Failure (MTTF).

(vi) **Security**: Non standards hardware and software such as outdated OS, weak factory settings, lack of user awareness etc., create several security loopholes in IoT devices. In the multi-layer IoT to Cloud architecture security is critical for each layer and have diverse security requirements.

(vii) **Standards**: Unplanned surge of IoT by different vendors has lead to a lack of uniformity and standards.

(viii) **Scale**: IoT-Cloud systems are becoming very complex with millions of IoT devices getting interconnected and generating huge amounts of data. Monitoring and managing the hardware-software of the complex system is a challenging task.

V. **SECURITY CHALLENGES IN IOT-CLOUD SYSTEM**

Analysis of the evolving IoT-Cloud architectures and recent publications [46], [62] reveal that security of the system is challenged by - technological progress, lack of standardization, IoT resource constraint etc. Broadly, the categories of security problems are - (i) Post Quantum Attacks and (ii) Vulnerabilities due to IoT-Cloud Characteristics.

A. **Post Quantum Attacks**

(i) **Emergence of Quantum Computers**: Gartner report indicates that Quantum Computers are rapidly evolving and a reasonable sized quantum computer would be available by 2025. The exceptionally fast computation of prime factors on quantum computers suggest that the popular public key cryptography system like RSA can be cracked down.

(ii) **Requirement of Larger Key Size**: Evolution of Quantum Computers and High Performance Computers makes it easy to crack even reasonably sized cryptographic schemes such as 1024 bit RSA. A simple remedy is to increase the key size; but it is impractical, especially in the context of resource constrained IoT.

(iii) **Time-Space-Energy Complexity**: Increasing the key size increases the computational complexity. The encryption and decryption algorithms should not be too complex if they have to run on low power and low capacity IoT and provide near real time response for real world applications. It is desirable to have minimal time-space-energy complexity for the algorithms.

For IoT applications with limited transmission bandwidth, the size of cipherext and the encapsulated key should be minimum for and ensure low communication overheads.

(iv) **Key Generation - Key Exchange Time**: The unprecedented growth of IoT networks combined with quantum cyber hackers is creating a highly active and complex system. The time for key generation and key exchange should be very fast, for the success of the security scheme.
(v) New Cryptanalysis Techniques: Cryptanalysts are exposing vulnerabilities in several existing crypto systems by using quantum computing algorithms, supercomputers, side channel attacks, guessing attacks, differential power analysis, differential cryptanalysis attacks etc. Rapidly evolving ecosystem, poses new challenges and requirements to develop novel Crypto schemes with higher security which are suitable for IoT and post quantum scenario, optimized algorithms with low footprint, and catering to heterogeneous devices. The concept of crypto-agility (i.e. replacing or adapting cryptographic schemes in software or hardware dynamically without interrupting the flow of a running system) is becoming a requirement for future operational application environments.

B. Vulnerabilities from IoT-Cloud Characteristics

(i) Heterogeneous Ecosystem: Independent manufacturers produce different varieties of IoT devices with varied functionality, that get integrated with different computing systems or Clouds over different networks. The heterogeneity and complexity of the ecosystem makes it very difficult to find a uniform security solution that fits all.

(ii) Lack of Standards: While there are diverse IoT manufacturers and enormous research and academic activities in IoT, there are only few standardization efforts such as from NIST, European Telecommunications Standards Institute (ETSI) and Internet Engineering Task Force (IETF).

(iii) IoT-Cloud Layered Architecture: A hierarchically increasing order of compute-storage capacities from IoT to Cloud to enable real-life applications brings along diverse security requirements at each layer

(iv) Scalability: The scheme should be able to cater for tons-of-thousands of devices and users and for huge volume of data within a reasonable turnaround time.

VI. PROPOSED SCHEME

A. Problem Definition, Objectives and Constraints

The problem is to provide security for data in motion and at rest, for the IoT-Cloud ecosystem, taking cognizance of the emerging quantum computing.

The main constraints for this problem are:

(i) very low capacity of IoT devices and edge computers;

(ii) growing key sizes of the cryptographic schemes;

(iii) possibility of break down of the existing cryptographic schemes by quantum computing algorithms;

(iv) need to support fully homomorphic operations.

Hence, in addition to the basic authentication, confidentiality, integrity and privacy of data, the new cryptographic scheme must provide - low compute footprint and a higher security level. It is desirable to have features of homomorphic encryption, searchable encryption and verifiable computation which are very useful for big data analytics. Crypto systems should be secure against the imminent quantum computers, which have the ability to solve certain mathematical problems exponentially faster compared to today’s supercomputers.

By increasing the size of the key and cipher, we can make it more difficult for attackers to guess the key. This is because the larger the key, the more possible combinations there are, making it harder for an attacker to guess the right one. Similarly, increasing the cipher size also makes it harder for an attacker to break the encryption since the cipher has more bits that need to be decrypted. However, increasing the key and cipher size also increases the computational complexity of the algorithm, which can slow down the encryption and decryption process.

The lattice based, homomorphic and quantum secure, RLWE encryption scheme lends itself to the above requirements, and is also the most popular among the NIST PQC Competition [63]. In order to make it suitable for resource constrained IoT-Edge-Cloud environment, the RLWE should be optimized with respect to the amount of compute, memory and communication bandwidth.

Minor tweaks to the RWLE, can help to increase the security level and reduce the computation. A new algorithm named Circular Error Learning Algorithm (CELA) is proposed which uses modulus switching to improve the execution time and security level of the cryptographic scheme.

B. Circular Error Learning Algorithm (CELA)

In the proposed research, the execution time of the RLWE is reduced and security is improved, by introducing the Circular Error Learning Algorithm (CELA).

To improve security strength, the polynomial ring used in this algorithm shall have a larger dimension to accommodate the larger key size. The size of key is given by:

\[ \text{KeySize} = (\text{RingSize} \times \log_2(\text{RingSize}) + \text{NoiseSize} \times \log_2(\text{NoiseSize}) + \text{SecurityLevel}) \times 2 \]

For example, if the security desired is 256 bits, and the ring size is 1024, the noise size is 128, the key size would be 22784 bits. Hence, note that the key size can be adjusted by changing the parameters used in the encryption process.

The execution time is determined by the complex calculations on the polynomial ring. The modulus switching reduces the execution time by reducing the intermediate computations without compromising its security. In modulus switching, the modulus is changed from \( q \) to a lower value \( q' \), to reduce computational complexity and improve its execution time.

Typically, the cryptographic scheme consists of Key Generation, Encryption and Decryption and the usage method is similar to the Diffie Hellman Key Exchange.

In the setup phase the public key is generated and published and corresponding secret-key generated is securely distributed among the parties. The public key is configured to be in the Hermite normal form of the lattice being examined. The keyholder creates a good basis by selecting \( P_h \) as a basis that is composed of short vectors that are "roughly orthogonal". The key to unlocking CELA’s plan is nothing more than a simple vector. In the lattice based method, a ciphertext is a vector that is located close to the lattice called \( L \). The message that is encrypted in this ciphertext is incorporated in the distance from the nearest lattice vector. In order to encrypt a message denoted by \( m \), the sender must first select
a brief 'error vector' denoted by $e$ that encodes $m$. Next, the sender must compute the ciphertext. We present a lattice-based key agreement mechanism for IoT devices based on CELA problem in order to perform secure data transfer.

The steps for key setup, encryption and decryption in CELA are as follows:

**Key Setup:**
1: Select public and private keys with a key size of at least $2n$, where $n$ is the desired security level in bits.
2: Generate Private key and distribute among the parties.
3: Generate Public key and publish.

**Encryption:**
1: Choose a uniformly random polynomial $r$ with coefficients in $\mathbb{Z}_q$ and degree less than $n$. Choose a uniformly random polynomial $r'$ with coefficients in $\mathbb{Z}_q$ and degree less than $n$.
2: Compute $e = a \ast r + m + e'$, where $m$ is the message to be encrypted and $e'$ is a small error term.
3: Compute $e' = b \ast r' + e''$, where $e''$ is a very small error term.
4: Return the ciphertext $(e, e' \ast f)$.

**Decryption:**
1: Compute $c' = e \ast b + e' \ast s \mod q$. Round each coefficient of $c'$ to the nearest integer multiple of $q/2n$.
2: Compute $m' = c' \ast f^{-1} \mod q$.
3: Return $m'$.

**C. CELA Secure Communication Protocol in IoT-Cloud**

This scheme uses a variety of available methods in order to conceal from parties the primary data that are not authorised for access. The protocol methodology uses the Registry Service Selection (RSS) security algorithm which has been designed to offer close assistance to the data all the way through the process of distributed computing [64].

The suggested system is divided into four distinct stages. To begin, the stage called - Client Registration to Cloud Specialist Cooperative, is responsible for managing the process through which the client is enrolled with the Cloud Service Provider (CSPs). The second stage (Distributed Storage of Information) will scramble and transfer the data and securely store it on the cloud. This includes data scrambling on the end of the customer and encryption with CELA on the end of the service provider. The third stage is - Client Authentication for Information Recovery Request. The final stage (Information Retrieval) takes the registry information by the confirmed client from the Cloud and gives to the approved client the data back after it has passed across all the security systems. The IoT devices, gateway and CSP are simulated on virtual machines (VMs) for experimentation. The flowchart of secure data communication protocol in IoT-Cloud environment is explained in Fig. 6.

**D. Results**

As seen in Fig. 7, the average time required for the CELA algorithm is 1628 microseconds while the average time required for the RLWE algorithm is 1811 microseconds. The ciphertext size for 100 keys in CELA algorithm is 158MB while that for the RLWE algorithm is 108MB.

Fig. 8 represents the comparison of minimum, maximum and average computation time of both RLWE and CELA algorithms. Multiple iterations have been considered for calculating the encryption and decryption time. The packet size transferred during these iterations are in the range of 1KB to 10KB. So for a 1KB packet size considered, the minimum time required is lower for CELA when compared to RLWE. For the highest packet size considered, the maximum time is also lower for CELA.

Typically, the size of the cipher text generated by CELA is slightly higher than that of RLWE as seen in Fig. 9.

**E. Discussion**

The following are some of the advantages offered by CELA:
The proposed scheme offers an optimized lattice RLWE-based key agreement protocol for IoT devices [41], [40].

The comparative examination of performance demonstrates that the suggested system is efficient in terms of compute and communication overhead.

- The system is provably secure for the longest period of time under the hardness assumption of RLWE, based on the fundamental difficulties of lattice algebraic structures [65]. Security analysis demonstrates that the system is secure against known security threats.

- The technique has been thoroughly researched for homomorphic encryption [66].

- Leveraging the RLWE scheme, CELA is capable of flexibility in dimension choice, and reduced computational requirements.

- The execution speed of CELA is superior in most cases, by virtue of modulus switching [57], very small size errors and Hermite Normal Form [67] matrix polynomials.

- The cipher text is more robust due to more randomness and slightly bigger size.

- The execution time of CELA is improved compared to RLWE by about 10%. As the number of IoT devices in the network increases, this can give a significant improvement in the system performance. Overall the CELA execution time being in microseconds is conducive for real-time applications.

- Increase in ciphertext size of CELA makes it computationally challenging for the adversary to break the encryption.

This research designed and demonstrated CELA as an optimized lattice based fully homomorphic cryptographic algorithm and a secure data communication protocol, for the IoT-Cloud converged system.

VII. PLAUSIBLE SOLUTIONS

(i) Resistance to Post Quantum Cryptography: Subsequent to the RSA-768 getting compromised in 2010 due to availability of higher computer power, the key sizes have been increased to RSA-2048, 3072 and 7680 which are said to secure even if the supercomputing power is used to crack them. But with the emergence of Quantum Computers, it is predicted that the Shor’s algorithm can calculate prime factors exceptionally fast; again posing a threat to RSA and other public key encryption schemes. While it is anticipated that factorizing the 1024-bit RSA would require about 2000 qubits [46] and the RSA-7680 would require 15362 qubits [68] on a Quantum Computer, it is currently unknown how many qubits would be needed to break 128-bit RLWE encryption as it is based polynomial equations, which is a different type of problem than factoring or discrete logarithms, and it is believed to be resistant to attacks by quantum computers. Presently the best operational quantum computer from IonQ is said to have only 79 qubits. Hence, for guaranteeing security till quantum computers become a reality, it will be sufficient to increase the key sizes.

(ii) Light Weight Cryptography: Increasing key size will improve the security strength, but disproportionately increase the complexity of the crypto system. There will be huge surge in the amount of compute-storage-power complexity, which are unaffordable for IoT-Edge-Cloud type of applications. Light weight cryptography having less computational complexity, faster execution time and low power consumption, less storage
space for key and ciphertext, is required to be developed. Complementing with a light weight key exchange mechanism will be highly useful for IoT-Cloud systems.

(iii) Novel Algorithms: Recognizing the need to transition traditional crypto systems as a preparatory for the quantum systems in the next decade, the NIST PQC Standardization Program has researched several Code-Based Techniques, Hash-based and Lattice based techniques. Novel hybrid techniques combining quantum key generation, key distribution with classical cryptography should be developed such as Quantum-AES [45]. Quantum-RSA and other Quantum-PKE. Future real-world applications, require crypto-agility (i.e. replacing or adapting cryptographic schemes dynamically without interrupting the running system) to ensure dynamic adaptation in the heterogeneous hyper-connected operational environment.

VIII. Conclusion

The objective of this research was to design a light weight quantum secure communication protocol for the IoT-Cloud ecosystem. Detailed study of the state-of-the-art cryptographic schemes and the challenges of IoT-Cloud Security in the emergence of Quantum Computing, led to identification of RLWE as a suitable base. RLWE is a homomorphic and quantum-safe cryptographic scheme and has emerged as the most popular in the NIST PQC Standardization Program.

A light weight Circular Learning Error Algorithm (CELA) has been proposed by optimizing RLWE with the modulus switching technique to make it suitable for IoT-Cloud ecosystem. A complete IoT-Cloud communication, including client registration, key generation and exchange, with CELA and RLWE encryption were experimentally compared. The CELA based protocol took 1628 microseconds compared to RLWE based protocol taking 1811 microseconds. This improvement will be significantly rewarding as the number of devices and users in the system increases.

While it may take over a decade for operational quantum computers to be established, the CELA can effectively safeguard the present IoT-Cloud application ecosystem. Meanwhile, agencies such as NIST, are taking cognizance of the threats to data security due to quantum computers. Research is rife for designing novel quantum based public key encryption schemes for the future quantum era; plausible solutions for future quantum secure crypto protocols have been presented in this work.

REFERENCES


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Abstract—Accurate anomaly detection of gateway electrical energy metering device is important for maintenance and operations in the power systems. Traditionally, anomaly detection was typically performed manually through the analysis of the collected energy information. However, the manual process is time-consuming and labor-intensive. In this condition, this paper proposes a hybrid deep-learning model, which integrates Stacked Autoencoder (SAE) and Long Short-Term Memory (LSTM), for intelligently detecting the abnormal events of gateway electrical energy metering device. The proposed model named SAE-LSTM model, first uses SAE to extract deep latent features of three-phase voltage data collected from the gateway electrical energy metering device, and then adopts LSTM for separating the abnormal events based on the extracted deep latent features. The SAE-LSTM model, can effectively highlight the temporal information of the electrical data, thereby enhancing the accuracy of anomaly detection. The simulation experiments verify the advantages of the SAE-LSTM model in anomaly detection under different signal-to-noise ratios. The experimental results of real datasets demonstrate that it is suitable for anomaly detection of gateway electrical energy metering devices in practical scenarios.

Keywords—Anomaly detection; gateway electric energy metering device; stacked autoencoder; long short-term memory

I. INTRODUCTION

The importance of anomaly detection in gateway electrical energy metering device lies in ensuring the accuracy and reliability of energy measurement. The gateway electrical energy metering devices play a crucial role in power systems as they are utilized to measure and record energy consumption. The significance of anomaly detection in gateway electrical energy metering devices extends to various aspects such as data accuracy, system safety, and energy management. Anomalies occurring in these devices can result in inaccurate energy consumption data, thereby impacting the billing and settlement processes between energy suppliers and consumers. Moreover, anomalies can serve as indicators of underlying issues or faults within the power system, and their timely detection can unveil potential problems. By promptly identifying and addressing anomalies, it becomes possible to ensure the accuracy and reliability of energy measurement, improve energy management efficiency, and guarantee the safe operation of the power system[1]. Presently, the detection of abnormal operating states in gateway electrical energy metering devices heavily relies on manual on-site inspections, which pose safety risks, have lengthy detection cycles, and may not promptly identify faults[2-4]. With the increasing number of gateway electrical energy metering devices, manual inspections necessitate greater human and material resources, making it challenging to fully meet the current requirements for metering device management. Hence, it is imperative to propose and establish an anomaly detection system specifically designed for gateway metering devices. This system should employ suitable anomaly detection algorithms to promptly identify abnormal states.

Currently, methods for anomaly detection can be categorized into three distinct classes. The first class comprises statistical-based detection methods, including Gaussian distribution[5], probability density functions[6], clustering algorithms[7], and Markov models[8]. Although statistical-based methods are grounded in solid theoretical foundations, the task of selecting an appropriate distribution to effectively discriminate normal instances from anomalous ones poses significant challenges. The second class encompasses rule-based detection methods, involving the establishment of thresholds and the utilization of rule engines, among others. Rule-based methods offer ease of implementation and interpretation, but their ability to detect more intricate anomalies may be constrained. The third class encompasses deep learning-based anomaly detection methods, such as those relying on convolutional neural networks[9,10]. These methods extract robust latent features; however, they necessitate the conversion of input data into images, thereby augmenting the data processing burden, while inadequately considering the influence of network structure information on the accuracy of feature extraction[11]. Although variant models based on the support vector machines (SVM)[12] demonstrate commendable performance in non-temporal data processing, their accuracy in handling complex time series data still needs to be improved.

With the rapid developments in artificial intelligence (AI), numerous approaches using machine learning (ML), especially deep learning (DL), have been proposed to overcome these challenges. For example, Lee S et al. used a self-encoder consisting of a graph convolutional network and a bidirectional long short-term memory network to detect anomalies in smart detection data with higher accuracy than a single LSTM network and with reduced power cost and grid power supply[13]. However, the graph convolution operation of the graph convolutional neural network becomes difficult when processing sequential data and suffers from dimensional mismatch. Wang et al. provided a semi-supervised learning based power anomaly detection strategy[14]. Their proposed framework not only detects abnormal power patterns in real time,
but also identifies suspicious power usage that is inconsistent with customers’ lifestyles and typical daily routines, but is not suitable for application to anomaly detection at grid gate metering devices. Hussain et al. have proposed an unsupervised detection approach aimed at identifying power theft behaviors without data labeling costs[15]. Their proposed method is evaluated using accuracy and detection rate.

Although most of the current models perform well, there are several limitations.

First, three-phase voltage data is often noisy and complex, making it difficult to distinguish normal fluctuations from actual anomalies. Second, power systems are dynamic and their operating conditions may change rapidly. Anomaly detection methods must be adaptable and able to evolve with these changes. Finally, many sophisticated machine learning-based techniques can effectively detect anomalies, but often lack interpretability.

To address the above problem, in this paper, we present a novel anomaly detection model that combines the Stacked Autoencoder (SAE) and Long Short-Term Memory (LSTM) networks with elastic network regularization. Initially, the model extracts latent feature representations of the data using SAE and subsequently employs the LSTM algorithm for classification purposes. Following the training of the SAE-LSTM network, elastic network regularization is applied to fine-tune the model using a composite loss function. Bayesian optimization techniques are then utilized to determine optimal hyperparameter values. Lastly, the model is evaluated using a performance assessment metric. Experimental results demonstrate that the proposed model effectively considers the temporal dependencies of the data, leading to improved detection accuracy. It is well-suited for detecting anomalies in gateway power metering devices and effectively assessing their operational state.

The contributions in this paper can be summarized as follows.

(1) This paper presents a novel anomaly detection model that combines the Stacked Autoencoder (SAE) and Long Short-Term Memory (LSTM) networks with elastic network regularization.

(2) Apply the SAE layer for feature extraction. The application of the encode layer allows the proposed model to extract high-level temporal features more efficiently while reducing model performance’s dependence on data processing, thus improving the accuracy and efficiency of anomaly detection.

(3) Apply LSTM layer for classification. In the voltage anomaly detection task, the LSTM model can capture the dynamic characteristics of voltage and current signals and is suitable for processing time series data.

(4) The short-term patterns and dependencies in the three-phase voltage and current time series data can be captured using the sliding window technique, and at the same time, by analyzing the sequence within a smaller window, it can help reduce noise in the simulated data and improve the SNR in the data. This further improves the feature extraction efficiency of the proposed model, enabling more accurate anomaly detection.

(5) A real dataset collected from a power grid is applied to evaluate the effectiveness and applicability of the proposed model.

The rest of this paper is organized as follows. Section II describes the Methods. Section III applies the method in a real case and analyzes the results. Section IV discusses the advantages and shortcomings of the proposed method, and the potential future work and concludes the paper.

II. METHODS

Accurate anomaly detection of gateway electrical energy metering device is important for maintenance and operations in the power systems. In this paper, a hybrid deep-learning model is proposed to intelligently detecting the abnormal events of gateway electrical energy metering device. The overall process flowchart of the proposed method is shown in Fig. 1.

A. Data Preprocessing

During the data preprocessing stage, three key procedures are utilized to increase the validity of the electrical energy data. These procedures handle missing values, normalize the data, and implement sliding windows on the data.

1) Missing value handling: The data collected from gateway electrical energy metering devices may partial be lost due to various factors, such as human error or equipment issues[16]. In order to address the problem of missing values,
several methods exist depending on the data nature and desired outcome. Common strategies involve deletion, imputation, and machine learning techniques. Although direct deletion is a straightforward approach, it may lead to the loss of valuable information, reduction in sample size, and decreased efficiency. Conversely, the utilization of machine learning methods can be excessively intricate. Therefore, Lagrange interpolation is employed to estimate the missing values. The formula for Lagrange interpolation is specified as follows:

$$l_j(x) = \prod_{i=0 \atop i \neq j}^{n} \frac{x - x_i}{x_j - x_i} = \frac{x - x_0}{x_j - x_0} \cdots \frac{x - x_{j-1}}{x_j - x_{j-1}} \frac{x - x_{j+1}}{x_j - x_{j+1}} \cdots \frac{x - x_n}{x_j - x_n}$$

$$L(x) = \sum_{j=0}^{n} y_j l_j(x)$$

Equation (1) represents the Lagrange basic polynomial for n+1 data points, while Equation (2) represents the Lagrange interpolation polynomial for n+1 data points[17]. x_i and y_i represent the x-coordinate and y-coordinate, respectively, of the known data points. l_j(x) denotes the Lagrange basis function, where the index j ranges from 0 to n, representing the i-th basis function. Consequently, L(x) signifies the estimated value at a given point x, acquired through the employment of the Lagrange interpolation method.

2) Normalization: There are several methods available for data normalization, including min-max scaling, Z-score standardization, and mean normalization. For the purposes of this study, the dataset is normalized using the Z-score standardization method. This approach is chosen due to its simplicity and ease of computation, as well as its ability to effectively normalize data regardless of the scale or presence of extremely large or small values. The Z-score standardization formula utilized is as follows:

$$z = \frac{x - \mu}{\sigma}$$

where x represents the data mean, \( \mu \) represents the standard deviation, and \( z \) represents the standardized score.

3) Sliding window: The sliding window is widely utilized for time series analysis and sequence data processing. It serves as a valuable tool for feature extraction and performing computations on data subsets[18]. With the application of the sliding window, it becomes possible to capture short-term patterns and dependencies presented in the three-phase voltage and current. Moreover, by analyzing sequences within smaller windows, the impact of noise in analog data can be minimized, subsequently improving the signal-to-noise ratio. This aspect proves advantageous for tasks such as anomaly detection and experimental evaluation. Moreover, the mean, median, and variance of the data within the window are extracted as input features.

B. The Anomaly Detection Model using Deep Learning

1) Basis of SAE and LSTM: The Stacked Autoencoder (SAE) is a hierarchical neural network comprised of multiple encoders connected layer by layer. It is an unsupervised learning method that follows a layer-wise greedy approach[19]. The Autoencoder (AE), which is a constituent of SAE, employs the backpropagation algorithm to ensure the output values match the input values. Initially, it compresses the input into a latent space representation and then reconstructs the output based on this representation. SAE possesses several advantages, including powerful expressive capability, a straightforward training process, and the ability to construct multiple layers of stacked architecture. It effectively mitigates challenges like “vanishing gradients” and “exploding gradients” that arise with increased depth in autoencoders. Consequently, SAE finds extensive application in target recognition, anomaly detection, anomaly diagnosis, and other domains.

The training process of SAE involves training one layer at a time[20]. Initially, a network with a single hidden layer is trained. After completing the training of this layer, training of a network with two hidden layers is initiated, and so on. Once all the layers have been trained, the encoder weights of each layer are combined to form a complete deep neural network. Subsequently, fine-tuning is performed, where the entire network is fine-tuned using supervised learning methods to optimize network performance. This training approach is known as the greedy layer-wise training algorithm.

Due to its deep structure and layer-wise training strategy, stacked autoencoders can learn higher-level and more abstract feature representations, thereby achieving superior performance across various tasks. In contrast, single-layer autoencoders are limited by their shallow structure and can only capture relatively simple features.

Long Short-Term Memory (LSTM) is a special type of recurrent neural network (RNN) structure[21]. The neurons in an LSTM model consist of four main components: the memory cell, the input gate, the forget gate, and the output gate. The internal structure of an LSTM neuron is illustrated in Fig 2.

$$i_t = \sigma(W_{ix}x_t + W_{ih}h_{t-1} + b_i)$$

$$f_t = \sigma(W_{xf}x_t + W_{hf}h_{t-1} + b_f)$$

$$C_t = \text{tanh}(W_{xc}x_t + W_{hc}h_{t-1} + b_c)$$

$$C_t = f_t \odot C_{t-1} + i_t \odot \tilde{C}_t$$

$$o_t = \sigma(W_{ox}x_t + W_{oh}h_{t-1} + b_o)$$

$$h_t = o_t \odot \text{tanh}(C_t)$$

The input gate decides whether to include the input feature \( x_t \) of the current time step into the state update of the LSTM. \( W_{ix} \) and \( W_{ih} \) are weight matrices for the linear transformation of the input and the hidden state \( h_{t-1} \) of the previous time step, respectively. \( b_i \) is the bias term and \( \sigma \) is the sigmoid activation function. The forget gate decides whether to keep the previous state information. \( W_{xf} \) and \( W_{hf} \) are weight matrices, and \( b_f \) is a bias term. The candidate memory unit computes the candidate value at the current time step by applying the hyperbolic tangent activation function. Cell states are updated through multiplication and addition operations. The forget gate determines
the proportion of the previous cell state $C_{t-1}$ retained, and the input gate determines the contribution of the candidate memory unit $C_t$ to the cell state. The output gate determines the output of the LSTM cell. $W_{zo}$ and $W_{ho}$ are weight matrices, and $b_o$ is a bias term. The hidden state is the output of the LSTM cell, computed by element-wise multiplication of the cell state $C_t$ with the output of the output gate and application of the hyperbolic tangent activation function.

2) The hybrid model of SAE-LSTM: The architecture of the proposed anomaly detection model, which jointly using SAE-LSTM, is illustrated in Fig. 1. The main steps of the model are described below, with a focus on the SAE-LSTM component.

a) The SAE component is employed to extract the deep latent features from the preprocessed three-phase voltage and current data. To ensure compatibility with the LSTM model, the latent features are further converted into an appropriate format through data type conversion.

b) The SAE network is utilized to extract the hidden features from the data, which are then transformed into time series data suitable for input to the LSTM model. By configuring the hyperparameters of the LSTM model, the model is trained and fine-tuned. Additionally, Bayesian optimization techniques are applied to optimize the model.

c) The optimized SAE-LSTM anomaly detection model is then evaluated using the test set. Various performance metrics, including accuracy and F1 score, are employed to assess the model’s predictive capabilities.

3) Fine-tuning and optimization of network parameters: In this study, a composite loss function is employed, incorporating elastic net regression to mitigate overfitting. This technique applies both L1 and L2 regularization to penalize the coefficients in the regression model. By integrating L1 regularization’s sparsity and L2 regularization’s weight shrinkage, elastic net regression achieves a harmonious balance, leading to improved generalization performance. The combined loss function is calculated using the following formula:

\[
LOSS_{\text{com}} = \alpha LOSS_{\text{SAE}} + (1 - \alpha) LOSS_{\text{LSTM}} + LOSS_1 + LOSS_2
\]

\[
LOSS_1 = \lambda \beta (W_{en} + W_{de} + W_{hh} + W_{fc}) \quad (10)
\]

\[
LOSS_2 = \lambda (1 - \beta) (W_{en}^2 + W_{de}^2 + W_{hh}^2 + W_{fc}^2) \quad (11)
\]

\[
LOSS_2 = \lambda (1 - \beta) (W_{en}^2 + W_{de}^2 + W_{hh}^2 + W_{fc}^2) \quad (12)
\]

The weight between $LOSS_{\text{SAE}}$ and $LOSS_{\text{LSTM}}$ is controlled by $\alpha$. When $\alpha$ is set to 1, only the reconstruction loss is active, and the classification loss has no impact. This means that the model focuses primarily on reconstructing the input data and minimizing reconstruction errors[22]. When $\alpha$ is set to 0, only the classification loss is active, and the reconstruction loss has no impact. This means that the model primarily focuses on the classification task and strives to optimize classification accuracy. When takes an intermediate value, both the reconstruction loss and the classification loss are considered, and the model optimizes between balancing the reconstruction and classification tasks. By adjusting the value of $\alpha$, the optimal trade-off between reconstruction and classification tasks can be found to meet specific problem requirements and performance demands. $\lambda$ determines the importance of L1 and L2 regularization, while $\beta$ controls the weight between L1 and L2 regularization. $LOSS_1$ calculates the L1 regularization term, which penalizes the sum of the absolute values of the model parameters. It measures the sparsity of the parameters by computing the L1 norm of different weight matrices. $W_{en}$ and $W_{en}$ represent the weight matrices of the encoder and decoder in the SAE network, respectively. $W_{hh}$ includes the weights connecting the LSTM layer input to its hidden state, responsible for transforming the input features into hidden state representations within the LSTM unit. $W_{fc}$ is the weight matrix associated with the connections between hidden-to-hidden in the LSTM network, encompassing the weights that link the previous hidden state to the current hidden state within the LSTM layer. $W_{fc}$ is responsible for propagating hidden state information, enabling the LSTM to maintain dependencies across the input time series. It represents the weight matrix of the fully connected layer in the LSTM network. The L1 norm of each weight matrix is the sum of the absolute values of its elements. These norms are summed together, multiplied by $\lambda$ and $\beta$, to weight the L1 regularization term. The purpose of this is to encourage the model to produce sparse parameters, reducing redundancy and improving the model’s generalization ability.

\[
LOSS_2 \quad \text{calculates the L2 regularization term, which penalizes the sum of squared model parameters. It measures the smoothness of the parameters by computing the squared L2 norm of different weight matrices. The squared L2 norm of each weight matrix is the sum of the squares of its elements. These sums are added together, multiplied by $\lambda$ and $1 - \beta$, to weight the L2 regularization term. The purpose of this is to encourage the model to produce smooth parameters, reducing overfitting and improving the model’s generalization ability.}
\]

By using $\lambda$ and $\beta$ multiplied by the regularization terms, an appropriate balance can be found between model complexity, reconstruction task, and classification task. This helps optimize the overall loss function to achieve better model performance and generalization ability.

To obtain the optimal performance of the model, a grid search method is used to search for the optimal model parameters. Grid search (GridSearchCV) is a search technique used to find the optimal parameters of a model. Grid search is a brute force algorithm. This makes a complete search for a given subset of the hyperparameter space[23]. Due to the exhaustive search, grid search consumes significant training time and resources, making its search performance inefficient.

Using Bayesian optimization technique can reduce computational costs and improve efficiency. It is more effective than grid search. It consists of two main components: a Bayesian statistical model for modeling the objective function, and an acquisition function for deciding where to sample next[24]. By using Bayesian optimization technique to find the optimal learning rate and weight in the combined loss function, the accuracy of the model in voltage anomaly detection task can be maximized.

III. EXPERIMENTS AND RESULTS

A. Data acquisition

1) Real Dataset: The real dataset was obtained from the national power grid and consists of secondary load data collected from four 330 kV substations situated in distinct geographical regions. Each substation’s data includes measurements of active power, reactive power, three-phase voltage and current, and the total power factor. Specifically, Substations A, C, and D were monitored at 15-minute intervals, while Substation B was recorded at 60-minute intervals. Notably, Substations A and D contain normal data, whereas Substations B and C contain abnormal data. Table I presents the distribution of normal and abnormal data in the dataset.

From Table I, it can be observed that substations B and C have a majority of abnormal data, while substations A and D do not contain any abnormal data. The ratio of normal data to abnormal data in the overall dataset is approximately 4:1. When comparing...
the features of normal and abnormal data, it is found that voltage changes are most pronounced during abnormal occurrences. Although there may be changes in current corresponding to the phase where voltage abnormalities occur, they are not as significant as voltage changes. Therefore, the abnormal data types in this study include voltage overvoltage, undervoltage, and other voltage-related anomalies. Three-phase current is considered as an auxiliary feature to help detect voltage anomalies. Hence, the features extracted for data processing and analysis in this study include three-phase voltage and three-phase current.

2) Computer simulation: The simplicity of the original data does not showcase the advantages of the constructed SAE-LSTM network. Therefore, to facilitate a comparison between the SAE-LSTM network and other networks, a method of generating simulated data by adding noise can be utilized. Signal-to-noise ratio (SNR) generically means the dimensionless ratio of the signal power to the noise power contained in a recording [25]. The parameter settings involve a SNR ranging from -20 dB to 20 dB, with an increment of 4 dB. Consequently, simulated data is generated at 4 dB intervals within the -20 dB to 20 dB range. The simulated data comprises normal and abnormal data, encompassing three types of abnormalities: voltage overvoltage, voltage loss, and low voltage. The figure below depicts the generated simulated data, illustrating SNR of -20 dB and 20 dB.

The Fig. 3 and Fig. 4 illustrate the simulation data with the SNR of 20dB and -20dB respectively, where blue indicates normal data and other colors represent abnormal data. The red color corresponds to phase C undervoltage, green indicates phase B overvoltage, and orange signifies phase C undervoltage. Analysis of the figure reveals that at a SNR of -20 dB, the abnormal data closely overlaps with the original data, indicating a limited ability to distinguish between the signal and noise. Consequently, the detection task becomes highly challenging for the model. However, when the SNR is 20 dB, the distribution of the generated simulated data closely resembles that of the collected system data. Furthermore, the discrimination between the signal and noise significantly improves compared to the -20 dB SNR. This distinction is particularly evident in the case of phase C voltage loss anomalies, where a substantial difference exists between anomaly type 3 and other data types in terms of phase C voltage. Such differentiation is absent when the SNR is -20 dB. Therefore, the detection task becomes relatively simpler when the SNR is 20 dB.

The selection of a SNR ranging from -20 to 20 dB serves specific purposes. Extremely low SNR result in noise intensity surpassing signal intensity, hindering accurate anomaly detection by the model, with consistent accuracy below 50%. Conversely, very high SNR yield simulated data that closely resembles the original data, containing minimal noise. Consequently, the distinction between the signal and noise diminishes. Given the relatively simple and 6-dimensional nature of the original data, different models achieve accuracies exceeding 99%, making it impossible to discern performance differences among them. By setting the SNR between -20 and 20 dB, a range of conditions spanning from high-noise environments to strong-signal environments is encompassed. Conducting experiments within this range enables a more comprehensive analysis of the SAE-LSTM model’s performance.

B. Evaluation Index

For model evaluation, more advanced classification metrics from the confusion matrix such as accuracy and F1 score are utilized. As the problem involves multi-label classification with imbalanced data, weighted F1 score is used, assigning different weights to different classes. The formulas for calculating accuracy and weighted F1 score are as follows:

\[
\text{Accuracy} = \frac{TP + TN}{TP + TN + FP + FN} \quad (13)
\]

\[
\text{Precision} = \frac{TP}{TP + FP} \quad (14)
\]

\[
\text{Recall} = \frac{TP}{TP + FN} \quad (15)
\]
where TP represents True Positive. TN represents True Negative. FP represents False Positive. FN represents False Negative. Recall represents the recall rate, Precision denotes precision.

C. Network Structure Selection

1) Unbalanced data processing: After conducting measurements, it was observed that the dataset exhibits a ratio of approximately 4:1 between normal data and abnormal data[26]. Due to the limited availability of original abnormal data, the SMOTEBoost method is employed to generate synthetic data points in close proximity to the existing abnormal data.

SMOTEBoost is an ensemble method that combines the advantages of SMOTE and Boosting techniques. By integrating multiple weak learners into a robust classifier, it surpasses the performance of the standalone SMOTE method, offering improved accuracy and robustness when dealing with imbalanced datasets.

The fundamental concept of the SMOTEBoost algorithm revolves around augmenting the weight of minority class samples during each iteration of the classification learning process. This emphasis on the minority class allows the weak learners to focus more on these samples. To address the severe class imbalance in the original data, new artificial samples are generated and incorporated into the dataset[27]. The SMOTE algorithm and the update of weak learner weights are described by the following formulas:

\[ x_{\text{new}} = x + \tau \text{rand}(0, 1) \ast (\tilde{x} - x) \]  
(17)

\[ \alpha = 0.5 \ln \left( \frac{1 - \varepsilon}{\varepsilon} \right) \]  
(18)

\[ \omega_n = \omega_{o_e} \alpha \]  
(19)

Equation 17 represents the performance calculation of the weak learner, \( x \) represents a selected minority class sample. By applying the principle of nearest neighbors, one sample, denoted as \( \tilde{x} \), is randomly chosen from the k nearest neighbor samples of \( x \), where represents a random number ranging from 0 to 1. The newly synthesized sample is denoted as \( x_{\text{new}} \). \( \varepsilon \) denotes the error rate, which represents the cumulative weight of misclassified samples. \( \alpha \) represents the weight of the weak learner, while \( \omega_{o_e} \) signifies the weight of each individual sample. Finally, \( \omega_n \) refers to the updated weights.

2) Ablation experiments with different layers: The neural network comprises an input layer, hidden layers, and an output layer, with the number of hidden layers and hidden units per layer playing a pivotal role in determining the neural network’s capacity and complexity. The selection of these hyperparameters significantly influences the model’s ability to learn intricate patterns and generalize to unseen data.

In the case of the SAE network, the number of neuron nodes in the input and output layers depends on the dimensionality of the input data. In this study, the extracted and preprocessed dataset exhibits a feature dimensionality of 6. Thus, the SAE network consists of 6 neuron units in both the input and output layers. The purpose of the hidden layers in the neural network is to grasp the complex features inherent in the input data. Augmenting the number of hidden units empowers the network with greater representational capacity, enabling it to capture more intricate and nuanced characteristics of the input data. Nonetheless, an excessive number of hidden layers can lead to prolonged training time, overfitting, and vanishing gradients. In this study, preliminary experiments revealed that exceeding 3 hidden layers in the SAE network resulted in overfitting. Consequently, two optimal combinations of hidden layer units, specifically 64 and 16, were chosen.

Concerning the LSTM network, the extracted hidden features derived from the SAE serve as input features, while the output layer comprises 4 units representing the data labels. Regarding the selection of the number of hidden layers and hidden units, it has been observed that surpassing 3 hidden layers exponentially escalates...
Therefore, the SAE2-LSTM3 anomaly detection model with the highest accuracy and F1 score, indicating the best model performance. Comparing the four models, it is evident that the SAE2-LSTM3 model achieves the best model performance. Comparing the SAE and LSTM models, the outcomes achieved through the proposed approach are contrasted with those of SAE, LSTM, as well as other fundamental machine learning models, including Convolutional Neural Network (CNN) and Support Vector Machine (SVM)[28]. The signal-to-noise ratio ranges from -20 to 20 dB, with a step size of 4, enabling an extensive assessment of these models on multi-classification data.

D. Comparisons in the Simulation Dataset

From Fig. 7, Fig. 8 and Table IV, it can be concluded that it is evident that the proposed method presented in this study achieves superior evaluation metrics in the task of three-phase voltage anomaly detection, surpassing the other four methods. The LSTM model exhibits comparatively lower recognition performance, with an average accuracy in multi-classification detection that is approximately 10% lower than the other four methods, and a correspondingly lower weighted-F1 score. This discrepancy can be attributed to the introduction of noise in the original data, which affects the temporal nature of the data and consequently hampers the LSTM model’s recognition capabilities. The CNN model, primarily designed to capture local spatial correlations, demonstrates inferior performance compared to the SAE and LSTM models. Furthermore, in the case of non-spatial correlations, demonstrates inferior performance compared to the CNN and SVM models. The signal-to-noise ratio ranges from -20 to 20 dB, with a step size of 4, enabling an extensive assessment of these models on multi-classification data.

In this study, the optimal number of model layers is initially examined, and the corresponding results are listed in Table II. The comparisons of the results for different models are illustrated in Fig. 5 and Fig. 6.

From the results, it can be observed that the SAE2-LSTM2 and SAE-LSTM3 models slightly outperform the SAE2-LSTM2 model in terms of accuracy and F1 score. This suggests that increasing the number of LSTM hidden layers can improve accuracy. Comparing the SAE1-LSTM2 and SAE-LSTM2 models, it can be noted that increasing the number of SAE layers does not significantly affect accuracy, indicating that increasing the number of SAE layers has a limited impact on improving model performance. Comparing the four models, it is evident that the SAE2-LSTM3 model achieves the highest accuracy and F1 score, indicating the best model performance. Therefore, the SAE2-LSTM3 anomaly detection model with 2 SAE hidden layers (16, 64 units) and 3 LSTM hidden layers (128, 128, 128 units) exhibits better performance.

### Table II. Model Parameter Configuration under the SNR from -20dB to 20dB

<table>
<thead>
<tr>
<th>SNR(dB)</th>
<th>SAE(16)</th>
<th>LSTM(128)</th>
<th>SAE(16)</th>
<th>LSTM(128)</th>
<th>SAE(16,64)</th>
<th>LSTM(128)</th>
<th>SAE(16,64)</th>
<th>LSTM(128)</th>
<th>SAE(16,64)</th>
<th>LSTM(128)</th>
<th>SAE(16,64)</th>
<th>LSTM(128)</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>Accuracy</td>
<td>F1</td>
<td>Accuracy</td>
<td>F1</td>
<td>Accuracy</td>
<td>F1</td>
<td>Accuracy</td>
<td>F1</td>
<td>Accuracy</td>
<td>F1</td>
<td>Accuracy</td>
<td>F1</td>
</tr>
<tr>
<td>-20</td>
<td>0.488</td>
<td>0.543</td>
<td>0.493</td>
<td>0.547</td>
<td>0.529</td>
<td>0.571</td>
<td>0.497</td>
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<td>0.501</td>
<td>0.549</td>
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<td>0.558</td>
<td>0.529</td>
<td>0.561</td>
<td>0.543</td>
<td>0.573</td>
<td>0.574</td>
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</tr>
<tr>
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<td>0.571</td>
<td>0.559</td>
<td>0.588</td>
<td>0.569</td>
<td>0.620</td>
<td>0.566</td>
<td>0.576</td>
<td>0.567</td>
<td>0.581</td>
<td>0.567</td>
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<tr>
<td>-8</td>
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<td>0.564</td>
<td>0.613</td>
<td>0.567</td>
<td>0.636</td>
<td>0.569</td>
<td>0.581</td>
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<td>0.602</td>
<td>0.567</td>
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<tr>
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<td>0.654</td>
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<tr>
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<td>0.619</td>
<td>0.679</td>
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<td>0.680</td>
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<tr>
<td>8</td>
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<td>0.780</td>
<td>0.682</td>
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</table>

**Fig. 5.** The accuracy diagram of the proposed model with varying neurons and hidden layers under the SNR from -20dB to 20dB.

**Fig. 6.** The F1 diagram of the proposed model with varying neurons and hidden layers under the SNR from -20dB to 20dB.
TABLE III. THE RESULTS OF ACCURACY AND F1 FOR THE COMPARISON MODELS UNDER THE SNR FROM -20dB TO 20dB

<table>
<thead>
<tr>
<th>SNR(dB)</th>
<th>SAE</th>
<th>LSTM</th>
<th>CNN</th>
<th>SVM</th>
<th>SAE-LSTM</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>Accuracy</td>
<td>F1</td>
<td>Accuracy</td>
<td>F1</td>
<td>Accuracy</td>
</tr>
<tr>
<td>-20</td>
<td>0.515</td>
<td>0.568</td>
<td>0.512</td>
<td>0.552</td>
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</tr>
<tr>
<td>-16</td>
<td>0.547</td>
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<td>0.533</td>
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<tr>
<td>-12</td>
<td>0.581</td>
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<td>0.590</td>
</tr>
<tr>
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<td>0.533</td>
<td>0.564</td>
<td>0.638</td>
</tr>
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</tr>
<tr>
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<td>0.689</td>
<td>0.692</td>
<td>0.776</td>
</tr>
</tbody>
</table>

Fig. 7. The accuracy diagram of the comparison models under the SNR from -20dB to 20dB.

Fig. 8. The F1 diagram of the comparison models under the SNR from -20dB to 20dB.

E. Comparisons in the Real Dataset

In order to evaluate the model more effectively and reduce experimental bias, k-fold cross-validation is employed. The KCV consists in splitting a dataset into k subsets; then, iteratively, some of them are used to learn the model, while the others are exploited to assess its performance[30].

From the collected data of the substation monitoring system, three-phase voltage and current data along with the label column were extracted. The original data was subjected to anomaly detection, and the experimental results are shown in the Table III.

The analysis of the results indicates that all models exhibit favorable performance when applied to the initial three-phase voltage and current data. The reason for this is that the real data used in this paper is very simple. Not only there are few types of anomalies, but also there is almost no interference, which is very easy to identify. Therefore, traditional machine learning algorithms such as SVM, CNN, etc. can also achieve very good results.

Moreover, although SAE and LSTM do not work perfectly when used alone. However, the SAE-LSTM network proposed in this paper combines the advantages of the two algorithms, which is both very powerful in feature extraction and well adapted to handle such time-series data as power grids. The simulation experiments in the previous paper show that the detection effect is still very good when facing the low signal-to-noise ratio data with great interference.

IV. CONCLUSION

This paper presents a method that addresses the challenge of delayed anomaly detection in current gateway metering devices by combining Stacked Autoencoders (SAE) and Long Short-Term Memory Neural Networks (LSTM) with elastic network regularization. The advantages of the proposed model are verified by real data experiments and simulated data experiments.

In the experiments utilizing real data, all examined models exhibited satisfactory performance, largely attributed to the dataset’s simplicity. To further enrich our investigation, we expanded our research scope to include experiments using simulated data.

These simulated data experiments introduced noise to increase data complexity before comparing the effectiveness of various models.

TABLE IV. THE RESULTS OF ACCURACY AND F1 FOR THE COMPARISON MODELS IN REAL DATASETS

<table>
<thead>
<tr>
<th>Model</th>
<th>SVM</th>
<th>CNN</th>
<th>SAE</th>
<th>LSTM</th>
<th>SAE-LSTM</th>
</tr>
</thead>
<tbody>
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<td>1</td>
<td>0.999</td>
<td>0.997</td>
<td>1</td>
</tr>
<tr>
<td>F1</td>
<td>1</td>
<td>1</td>
<td>0.999</td>
<td>0.997</td>
<td>1</td>
</tr>
</tbody>
</table>
The findings underscored that the model proposed within this paper demonstrates superior performance in managing complex data. As delineated in Fig. 8, the F1 score exhibits a gradual ascent when the Signal-to-Noise Ratio (SNR) spans -20dB to 0dB and 12dB to 20dB. In contrast, a brisk rise is observed between 0dB and 12dB. This observation can be attributed to the reduction in noise level as SNR increases, thus facilitating the model’s anomaly detection capabilities and consequently leading to the swift enhancement in F1 scores. The sluggish elevation in the F1 score with an increase in SNR might be indicative of the model nearing its maximum performance potential, unable to capitalize fully on the added clarity from higher SNR values. Alternatively, it could imply that the data lacks further valuable information to aid in more distinct anomaly detection, thereby causing the measured F1 score to ascend more slowly.

Power grid data often exhibits specific patterns of variation, making it suitable for analysis using the temporal nature of LSTM networks and the robust feature extraction capabilities of SAE networks. Experimental results have confirmed the high effectiveness of this method for anomaly detection. But fully applying this model to the anomaly detection of the actual substation gateway metering device will have certain shortcomings. In practical applications, there is no corresponding label for the data measured by the metering device. In order to realize the abnormality detection of the metering device more conveniently, it is necessary to increase the learning of unsupervised algorithms; It can also display some parameters measured by the metering device to ascend more slowly.

ACKNOWLEDGMENT

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REFERENCES


Semantics Privacy Inference Preservation Algorithm
for Indoor Trajectory

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College of Computer Science and Engineering, University of Jeddah, Saudi Arabia

Abstract—Indoor location services have become an increasingly important part of our everyday lives in recent years. Despite the numerous benefits these services offer, serious concerns have arisen about the privacy of users’ locations. Adversaries can monitor user-requested locations in order to obtain sensitive information such as shopping patterns. Many users of indoor spaces want their movements and locations to be kept private so as not to reveal their visit to a particular zone inside buildings. Research on semantic indoor trajectory-based human movement data has primarily focused on finding routes without taking into account the protection of privacy. Hence, the servers on which trajectory data is stored are not completely secure. In this paper, we propose a semantic privacy inference preservation algorithm for an indoor trajectory that can issue pathfinding and navigation instructions while achieving good privacy protection of moving entities by generating ambiguous trajectory. The simulation of the proposed semantic indoor privacy algorithm was implemented in MATLAB.

Keywords—Privacy; semantic ontology; indoor space; routing algorithm; spatial databases

I. INTRODUCTION

With the widespread usage of smartphones, mobile crowdsensing is now recognized as a promising means of collecting diverse and heterogeneous trajectory data for moving objects. This is especially useful for modeling human mobility patterns in either indoor or outdoor urban environments. The popularity of mobile positioning devices has resulted in the creation of several location-based services (LBSs) as well as vast volumes of locational data. Location-based services (LBSs) are being used in an increasing number of areas and have become an integral part of modern life [1, 2, 3]. Demand for location-based services is gradually extending from outdoors to indoors. Researchers and enterprises are increasingly interested in “indoor positioning systems (IPS)” and “indoor localization systems (ILS)” because they offer significant commercial opportunities for implementing indoor positioning, mapping, and navigation [4, 5].

Human mobility may be considered as a trajectory across a variety of indoor or outdoor spaces, each with its own distinct characteristics. While many studies have been conducted on outdoor trajectories, research has revealed that people spend more than 87% of their lifetime indoors in places such as shopping malls, airport terminals, and conference offices [6, 7, 8]. However, few studies have been conducted on safeguarding the privacy of indoor user trajectories. Instead, the emphasis has been on ensuring user privacy in outdoor environments despite the fact that most of our time is spent indoors where the same privacy issues apply.

Recognizing the promise of indoor LBSs, large technology corporations, governments, research institutions, and start-ups are making significant investments in this technology. While such technologies promise to make our lives easier, the privacy issues linked with indoor LBSs have raised many concerns, as the data they acquire is often sensitive and needs to be safeguarded. Indeed, misusing location data can result in the leaking of sensitive personal information about a user, such as personal interests and lifestyle habits. For example, users may want to visit stores while keeping their moves private, as an invasion of privacy can be exploited for advertising, spreading rumors, or other unauthorized or malicious purposes. Therefore, the protection of users’ trajectories in indoor environments is critical.

Trajectory-based operations involving spatiotemporal data of moving objects are becoming increasingly essential in related research and applications. This is because it provides insight into human movement and has the potential to recognize patterns and predict future behavior. Over the past few years, several researchers have introduced a series of data models for semantic trajectories in indoor environments. However, the studies on semantic indoor trajectory-based human movement data have focused mainly on finding routes without considering the protection of privacy.

In this paper, we propose a solution that enables pathfinding and navigation instructions while achieving good privacy protection for moving entities by generating ambiguous trajectories. Semantic Web technologies provide strong representation tools for ubiquitous applications [9, 10, 11]. With location-based services and Semantic Web standards, trajectories can be linked and semantically annotated more easily, resulting in semantic trajectory descriptions. In this paper, we introduce semantic spatial ontology, which models indoor routing system needs while taking into account the protection of privacy. The contributions of this paper are as follows:

- Integrating semantic ontology/knowledge graphs and semantic privacy inference into indoor location-based services to improve accuracy and enrich navigation instructions with preserving the privacy of the user.

- Generating an ambiguous trajectory achieves privacy protection by safeguarding the actual trajectory of moving objects.

This paper has the following structure. Section II reviews the relevant literature and research on indoor routing and indoor/outdoor spatial privacy. Section III presents an overview of ontology modeling languages for the Semantic Web. Section IV describes the proposed semantic indoor trajectory privacy preservation algorithm. The architecture is comprised of two-level components: a semantic spatial ontology model for
indoor routing, and semantic ambiguous trajectory privacy inference preservation. Detailed explanations of each of the architecture’s components are provided in Section IV. Section V provides a motivation example and validation experiment for the proposed semantic indoor trajectory privacy preservation algorithm. Section VI summarizes the paper and offers suggestions for future research directions.

II. LITERATURE REVIEW

This section describes the relevant literature and research on indoor routing and spatial privacy in both indoor and outdoor environments. Several studies have explored indoor routing and have conducted studies to determine the algorithms that are most effective for indoor routing and navigation, Fig. 1.

Parulian et al. [12] developed an indoor navigation system with guiding assistance depending on user location and desired destination. Dionti et al. [13] presented a three-dimensional space routing system implementation that represents data as an undirected graph with three-dimensional properties. The study compared four algorithms to determine which offered the shortest path in three dimensions. “The four algorithms are the Dijkstra Algorithm, A* Algorithm, Bellman-Ford Algorithm, and Floyd-Warshall Algorithm”.

Alamri et al. [14] proposed an indoor multi-user routing algorithm for social distancing by considering and predicting user locations. The proposed indoor multi-user routing architecture consists of several components, including “indoor data structure, routing selection algorithm, density mechanism algorithm, and predictive routing algorithm”. Dionti et al. [15] created a prototype routing system between buildings that combined outdoor and indoor routing systems.

Also, indoor navigation has been studied using semantic ontologies. The IndoorGML and BIGML data model have been applied to model indoor spatial information [16, 17]. Building Topology Ontology (BTO) was proposed to the W3C community group for Linked Building Data as “a simple ontology covering the core concepts of a building” [18].

Ontologies containing semantic representations of navigation path components were presented by Alamri [4] for an indoor navigation system with reasoning functionalities. The proposed system can be employed in navigation systems for route finding and presentation, along with crowd-density monitoring. ONALIN is an indoor routing ontology and algorithm that specifically takes into consideration the American Disability Act (ADA), presented by Doubts et al. [19].

Anagnostopoulos et al. [20] presented a hybrid modeling method for user navigation, called “OntoNav”, that incorporates geometric and semantic information, as well as an ontological framework for processing routing requests. Sriharee [21] developed an ontology for indoor navigation based on symbolic information specified in Web Ontology Language (OWL). Wang et al. [22] proposed an ontology for expressing indoor navigation models using semantic location information.

Lee et al. [23] suggested ontology-based semantic queries to investigate indoor activities in a university environment. Yang et al. [24] investigated the use of ontologies for seamless navigation in both indoor and outdoor environments. Kim et al. [25] developed a spatiotemporal context-awareness knowledge model meant to recognize user objectives and help users when several paths are involved, such as hospital tasks.

Zlatanova and Liu [26] developed an indoor navigation space model (INSM) that uses semantic information to create connectivity graphs for buildings and identify several types of building spaces, such as inaccessible obstacles. Park et al. [27] built an expanded data model to assist persons with disabilities with indoor navigation by describing the relevant properties and relationships between PWD mobility and indoor environments. The IndoorGML application is used to support route planning that takes into account information regarding obstacles.

Maheshwari et al. [28] created an ontology for indoor places that considers both semantic and geometric properties. On the basis of this ontology, a space semantic model is developed, which can be used in a variety of applications. Li et al. [29] presented an indoor space dimensional model that allows for barrier-free path-finding by combining geometric, topological, and semantic layers.

All these recent works focus on data models and algorithms for indoor routing to support navigation systems for route finding and presentation. However, the collected data is usually sensitive, so the privacy issues raised by indoor routing are extensive. In fact, misuse of indoor location data can lead to the disclosure of sensitive personal information about users, such as personal interests and lifestyles.

Many works have been developed that focus on the privacy of moving objects’ paths in outdoor environments. According to Wu et al. [30], location pair reorganization can be used to protect trajectory privacy. Differential privacy is a new privacy-preserving technology based on data distortion that secures sensitive data while maintaining statistical features by introducing random noise to data [31, 32, 33, 34]. Wang et
al. [35] generate dummy trajectories by rotating the user’s real trajectory at the specified rotation point. Shaham et al. [36] choose dummy locations that have the same posterior probability as the real places.

Some studies have been concerned about related privacy-preserving methods in indoor positioning systems. Kim et al. [37] proposed a privacy protection strategy for indoor positioning. This approach suggests the usage of an application that changes the phone’s MAC address on a regular basis. They choose this approach to protect the user’s privacy by not revealing his or her identification to the server. Li et al. [38] introduced noise into user fingerprint data using differential privacy. Furthermore, differential privacy has been applied to data obfuscation approaches to generalize and specialize user fingerprints [39, 40]. Sazdar et al. [41] proposed creating dummy places to safeguard users’ location privacy. Zhao et al. [42] presented a paradigm-driven privacy protection framework for indoor localization based on local differential privacy. A comprehensive review was presented in [43] for indoor localization, which analyzes previous studies on user privacy on devices, in data transmission, and on servers.

To the best of our knowledge, the studies on semantic indoor trajectory-based human mobility data have focused mostly on finding routes without taking the privacy issue into account. We are the first to consider user trajectory privacy in a semantic ontology-based indoor spatial information model. This paper incorporates semantic ontology and semantic privacy inference into indoor location-based services to improve accuracy and enrich navigation instructions with preserving the privacy of the user by generating an ambiguous trajectory. Users’ sensitive points are identified and use random candidate points to replace them for a pruning process. Consequently, a protected semantic trajectory database is created that preserves semantic consistency and resembles the original semantic route while maintaining various levels of user privacy. In terms of semantic indoor privacy, the proposed algorithm has advantages. An option is provided for users to select whether privacy protection should be activated. The system also can ensure the privacy of moving objects in indoor environments by generating an ambiguous trajectory based on sensitive zones/points, which prevents an attacker from knowing the true route of a user. This can help the user preserve the privacy of his or her movements, as a breach of privacy could be used for advertising, propagating rumors, or other illegal objectives.

III. SEMANTIC WEB: THE CONCEPT OF ONTOLOGY

The concept of the Semantic Web indicates the objective of encoding information in a way that both humans and computer systems can understand [44, 45, 46]. This includes creating new ways to represent increasingly complex knowledge for true semantic interoperability. Hence, paradigms for operating the Semantic Web require the formal definition of domain models. These formal models (referred to as ontologies) should contain a clearer and more effective description of the terms used and their relationships than the metadata [47, 48].

Ontologies have brought new ways of representing knowledge, adding the ability to derive new knowledge not explicitly derived from the analysis of existing data. This gives the system a more efficient way of reasoning and a common way of representing knowledge across different systems working towards the same goal. The World Wide Web Consortium (W3C) has established the Resource Description Framework (RDF) as “the standard paradigm for data exchange on the Web” [45, 46, 49, 50, 51, 52].

These considerations drive our decision to base our knowledge model on semantic web concepts and technology. Our work focuses on semantics in order to improve the proposed system with such capabilities. It serves as a semantic middleware, interlinking, capturing, and providing results. This is shown by the ontology component, in which diverse content is homogenized and stored locally in accordance with RDF triples within a knowledge base. More specifically, the model is employed using RDF edge-labeled graph, Fig. 2. With RDF, heterogeneous data can be integrated with different underlying schemas and distributed graphs can be represented semantically with the relevant reasoning capabilities.

IV. SEMANTIC INDOOR TRAJECTORY PRIVACY PRESERVATION ALGORITHM

The proposed model architecture has been presented in Fig. 3. The architecture is comprised of two-level components: a semantic spatial ontology model for indoor routing, and semantic ambiguous trajectory privacy inference preservation. Each of the architecture’s components is explained in more detail as follows:

In the top-level, a spatial ontology model is a routing system implemented in buildings and responsible for selecting the most accessible navigation path for users. The major issue with the spatial ontology routing model is understanding how to create the data structure that describes an indoor environment (Fig. 4). The increasing complexity of indoor spaces demands that the data model for indoor space’s semantic division be able to handle efficiently the various features that are likely to be found in complex indoor spaces.

In the indoor space depicted in Fig. 4, it would be straightforward for a user to determine the optimal path from R1 to R2 within a building. However, the computer requires an accurate data structure that can describe corridors, rooms, and staircases. The semantic ontology of indoor space representation is depicted in Fig. 5.
The model presents a semantic model for indoor space that contains both syntactic and semantic information about spaces. In this model, the semantically enhanced representation of indoor space that we propose is a layered multigraph. Its node classes represent indoor spatial spaces symbolically, and its edges provide topological connection information between those spaces. Static semantic information about the indoor space is represented via node classes and attributes that relate to it, as well as node-edge layering. This is determined by the shape of the space, its connectivity with other spaces, and its functional features. We defined two main semantic node classes for the indoor space taxonomy. In order to show the overall hierarchy of floors inside a building, indoor zone space is the primary subclass to which the room and floor belong. The other is connector space which indicates the link between the indoor zone and the different floors. The specific semantic partition model is defined as:

- Indoor zone space
  - Room
  - Floor
- Connector space
  - Corridor
  - Stair
  - Door
  - Elevator
  - Escalator

**Definition 1 (Taxonomy of Indoor Space):** The ontology for indoor space taxonomy consists of two main ontology classes: “Indoor zone ∧ connector”.

- “Indoor zone space” specifies the major subclass to which the room and floor belong in order to describe the overall hierarchy of floors within a building.
- “The connector space” indicates the link between the indoor zone and the various floors.
Definition 2 (User Route): A user route, which consists of a series of sample points, is usually associated with a specific moving object as \( \text{Tr}_r = (P_s, c_{p_1}, \cdots, c_{p_n}, P_d) \) where:
- \( P_s \) identifies the moving object current point/zone location.
- \( c_{p_1}, \cdots, c_{p_n} \) represents the obtained trajectory connection points.
- \( P_d \) identifies the end point (destination zone).

The system can receive two inputs: the first provides the current point/zone location, and the second contains the location of the endpoint zone of the user’s destination. The system then presents the user with routes from the starting point to the destination. An algorithm for building a routing approach for an indoor environment can be summarized as the pseudo-code shown in Algorithm 1. Once the best route has been followed and the user reaches the endpoint, the system will begin protecting privacy if the user requests privacy protection.

In the lower level, semantic ambiguous trajectory privacy inference preservation is employed to safeguard the actual trajectory of moving objects. The result is a protected semantic trajectory database that preserves semantic consistency and a form resembling the original semantic route while meeting various user privacy requirements.

Definition 3. A privacy level is defined as \( P = \text{Privacy}\_\text{level}, \) \( \text{Privacy}\_\text{level} \in [\text{high}, \text{nil}] \).

Specifically, the privacy level is organized into two levels: Nil privacy (nil), and High privacy (high). Privacy level acts as a confidence level to register path. In the case of nil privacy, the actual trajectory of the moving object is registered. With high privacy, the user achieves privacy protection by safeguarding the actual trajectory of moving objects.

Our goal is to protect the sensitive points while still publishing a trajectory, albeit one consisting of a sequence of ambiguous points. The general idea of semantic ambiguous trajectory privacy is that an adversary with access to the protected semantic trajectories database comes to the same conclusion as whether an individual’s trajectory data is included in the database or not. If ambiguous trajectory privacy is achieved in publishing trajectory data, it can assure the user that the released data will not leak his/her privacy whether or not his/her trajectory is in the database. Also, ambiguous trajectory data is published to confuse adversaries due to their similar shape and semantic attributes. When a user wants his or her trajectory’s privacy inferences to be protected, the algorithm will collect trajectory data for the user from a data-centric semantic trajectory. First, to protect the privacy of this user’s trajectory, we replace two connection points - the starting point and the destination point - with other random points.

Then, in the ambiguous trajectory generation stage, we map these two points by matching them with other random candidate semantic points (from the same or next level) and assume that they have the same semantic attribute. Finally, the algorithm will reconstruct the fake user trajectory by replacing all sensitive points. In this way, the semantic-protected trajectory database that is published will contain a sequence of ambiguous points.

The ambiguous trajectory generation method involves trajectory processing based on sensitive zones/points and other semantic information to prevent an attacker from determining the user’s actual trajectory. The most common targets are the start-end points. There are two types of ambiguity-generation methods: trajectory segment pruning and ambiguity trajectory reconstruction. The pruning method eliminates sensitive points from the trajectory segment and uses random candidate points from the semantic bridge category to replace the corresponding start and stop points.

Semantic mappings between different location points are expressed via a semantic bridge category. To formalize this concept, we provide the following definition. Each location point is formalized with a prefix index in order to make it distinct from the others. For example, we use the i prefix to show the original start/end location point. Similarly, we use the j prefix index to signify the candidate location point.

Definition 4 (Semantic Bridge Category): The semantic bridge category can be expressed as: Mapping \( P_{si} \xrightarrow{\Delta} P_{sj} \land P_{di} \xrightarrow{\Delta} P_{dj} \) where:
- \( P_{si} \) and \( P_{di} \) identify original start and end zones/points.
- \( P_{sj} \) and \( P_{dj} \) identify random start and end zones/points.
- \( \Delta \) is the mapping bridge between points.

The proposed trajectory reconstruction algorithm is demonstrated in Algorithm 1. First, we locate a starting point in the trajectory segment \( P_{si} \). Similarly, we locate an end point \( P_{di} \) on the segment and generate mapping bridge between these location points based on semantic bridge category \( P_{si} \xrightarrow{\Delta} P_{sj} \land P_{di} \xrightarrow{\Delta} P_{dj} \). We generate a certain number of connection points on the segment \( (P_{sj}, c_{p_j}, \cdots, c_{p_n}, P_{dj}) \). Various trajectory segments can be obtained. Finally, we utilize the trajectory segment \( P_{sj} \cdots P_{dj} \) to replace the original trajectory segment. Subsequently, a new trajectory segment is created, thereby protecting user privacy. These trajectories are illustrated in Fig. 6.

V. EXPERIMENTAL VALIDATION

In this section, we use a software engineering strategy and validation experiments to ascertain the effectiveness of the proposed algorithm. The experiment test cases are designed to test ontology-based spatial information and ambiguous trajectory generation for privacy protection. For this experiment, using the indoor space ontology, we created a virtual dataset corresponding to a hypothetical two-floor building.

Assume a user visits a building for the first time and wishes to be directed to a certain zone without disclosing her/his location to the building’s manager or any other party. This situation is common in, for example, malls, airports, hospitals, and university campuses. It necessitates the use of an autonomous application running on the user’s mobile terminal that can provide the necessary location data as well.
as the information required for the generation of navigation directions.

We give an example of the simulation of the proposed privacy routing algorithm. Examples of the routes are shown in Fig. 7 and 8. Here, moving object $o_i$ wants to visit zone $P_{d23}$ and her/his start position was zone $P_{s18}$. Based on the trajectory connection points from $P_{s18}$ to $P_{d23}$, the expected route $P_{s18}, \text{cp}_{12}, \text{cp}_{13}, \text{cp}_{1}, \text{cp}_{2}, \text{cp}_{3}, \text{cp}_{4}$, zone $P_{d23}$ or $P_{s18}, \text{cp}_{11}, \text{cp}_{10}, \text{cp}_{9}, \text{cp}_{8}, \text{cp}_{7}, \text{cp}_{6}, \text{cp}_{5}, \text{cp}_{4}$, zone $P_{d23}$. The user $o_i$ can reach the zone by following one of these routes. When the user uses the optimal route and reaches the endpoint, our proposed algorithm is meant to ensure privacy protection.

The system generates an ambiguous trajectory and reconstructs it to create new segments of the trajectory for the user. It generates a mapping bridge between start/end location points and computes new traversable routes between two new points. Fig. 9 shows the route of validation on MATLAB, from the current position to zone $P_{d23}$. Fig. 10 shows a new trajectory segment that has been produced by the system for the user, thereby protecting user privacy.
Algorithm 1: Semantic Privacy Inference Preservation Algorithm

```
1 Begin
2 Input: P_{si}: initial node, P_{di} the destination node, O: the ontology knowledge
3 Procedure generate routes from P_{si} \rightarrow P_{di}
4 Function main()
5 Tr[] = \emptyset; // list of Trajectories
6 Route = BuildRoute(P_{di});
7 Tr[] = Route;
8 Privacy_level = PrivacyInference();
9 Function BuildRoute(dest)
10 explored[] = \emptyset; // list of explored route segment
11 initial = P_{si};
12 insert P_{si} in explored;
13 if initial == dest then
14 return initial;
15 foreach i \rightarrow 0 \text{len(dest)} - 1 do
16 find connect points of nodes;
17 initS = P_{si};
18 destD = P_{di};
19 Routing.merge(verticalPassage(initS, destD));
20 end
21 Function verticalPassage(initS, destD, explored[])
22 RouteSegment= explored[];
23 if initS.floor = destD.floor then
24 cp = selectcp(initS.floor); // select all cp in the unit of destD
25 sort using EuclideanDistance(initS, cp);
26 destD = cp;
27 RouteSegment.merge(initS, destD);
28 end
29 else
30 select cp that connect to the other floor
31 newD= destD.unit.get(destD.floor);
32 RouteSegment.merge(initS, newD, destD);
33 end
34 Tr[] ← explored[i];
35 check another route Extra_Route();
36 Function Extra_Route()
37 if y adjacent to the P_{yi} then
38 foreach y do
39 find y not in RouteSegment;
40 update y to the new route: explored[i + 1] = y
41 Tr[] ← explored[i + 1]
42 end
43 end
44 Function PrivacyInference()
45 if Privacy_level = nil then
46 Generate and store original user trajectory
47 end
48 else
49 if user reached \rightarrow P_{di} then
50 Activate ambiguous trajectories
51 generation stage
52 Find a start/endpoints in the trajectory segment: P_{si}, P_{di}
53 Generate mapping Bridge_Category
54 P_{si} \supseteq P_{sj}
55 P_{di} \supseteq P_{dj}
56 Compute traversable routes between two new points: P_{sj} \cdots P_{dj}
57 Tr_{j} = (P_{sj}, cp_{j} \cdots cp_{n}, P_{dj})
58 Derive new trajectory segment
59 end
60 end
61 end
```

The results demonstrate that the proposed semantic indoor privacy method has advantages. It enables the user to choose whether or not to activate privacy protection. It can also protect the trajectory of moving entities in indoor environments by employing an ambiguous trajectory generation mechanism based on sensitive zones/points and other semantic information to prevent an attacker from determining the user’s actual route.

This can assist the user to protect the privacy of his or her movements, as a breach of privacy could be exploited for advertising, spreading rumors, or other unauthorized purposes. In the future, more research will be conducted to improve the experiment. A comparison with similar approaches is provided. Furthermore, we would like to assess its effectiveness and performance in recreating user trajectory through real-world usage.

VI. CONCLUSION AND FUTURE WORK

In this paper, we proposed an algorithm for safeguarding the semantic privacy inference of entities moving within indoor environments. Previous studies on the privacy of spatial trajectory have focused mostly on outdoor environments, whereas this work focuses exclusively on indoor environments. The proposed algorithm finds the best route for users while preserving their privacy and offering the user the choice to activate privacy protection. The proposed privacy solution involves the generation of an ambiguous trajectory generation based on sensitive zones/points and other semantic information to prevent an attacker from determining the user’s actual trajectory in an indoor environment. By means of this algorithm, users can easily find a specific place in a building and achieve adequate privacy protection as the algorithm conceals the user’s trajectory.

It is anticipated that future studies will improve on the experiment described in this research and conduct related experiments in an actual indoor space environment. Additionally, several challenges need to be addressed. We intend to evaluate the impact of the overall performance of the proposed algorithm on the reconstruction of user trajectory.

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Knee Cartilage Segmentation using Improved U-Net

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Abstract—Patello-femoral joint stability is a complex problem and requires detailed anatomic parametric study for knowing the associated breakdowns of knee cartilage. Osteoarthritis is one of the main disorders, which disrupt the normal bio-mechanics and stability of the patello-femoral joint and for diagnosing osteoarthritis radiologists needs a lot of time to diagnose it. An improved network called PSU-Net is proposed for the automatic segmentation of femoral, tibia, and patella cartilage in knee MR images. The model utilizes a Squeeze and Excitation block with residual connection for effective feature learning that helps in learning imbalance anatomical structure between background, bone areas and cartilage. The severity of knee cartilage is measured through the Kellgren and Lawrence (KL) grading system by radiologists. Also, updated weighted loss function is used during training to optimize the model and improve cartilage segmentation. Results demonstrate that PSU-Net can accurately and quickly identify cartilages compared to the traditional procedures, aiding in the treatment planning in a very short amount of time. Future work will involve the use of augmentation methods and also use this architecture as a generator model for generative adversarial network to improve performance further. The utility of this work will help in analyzing the anatomy of the human knee by the radiologists in short amount of time that may prove helpful to standardize and automate patello-femoral measurements in diverse patient populations.

Keywords—Knee image segmentation; U-Net; loss function; squeeze and excitation

I. INTRODUCTION

Patello-femoral joint stability depend on the knee. The patello-femoral joint is formed between the patella (kneecap) and the femur (thigh bone). It is a complex joint that allows for the movement of the patella during knee flexion and extension. The knee is known as the largest joint in the human body [1]. One of the important components in the knee joint is the cartilage. Cartilage is fine, rubbery, and flexible tissue that covers the surface of bones, which can be found throughout the body. Cartilage helps to reduce friction by acting as a cushion and lubricant between the joints. However, repeated bio-mechanical force or a sudden impact will cause the knee cartilage to experience wears down or tear, leaving the rough bone surface exposed to each other resulting the friction in between the constituent structures making the joint [2]. This situation is called damaged cartilage. Prolonged damage over the time will lead to the happening of Knee Osteoarthritis (KOA) with long-term irreversible effects on normal knee function. Eventually, the disease would lead to permanent physical disability.

KOA is a form of arthritis characterized by inflammation, degradation, and ultimate loss of cartilage in joints, most often affecting the knee’s major weight-bearing joint [3]. KOA can be classified according to degradation severity. In 1961, WHO has accepted the Kellgren and Lawrence (KL) grading system as a standardized way to identify and grade the severity of (KOA) [4]. The grading scheme classified KOA into five different levels by assigning a grade from 0 to 4 [5].

Knee osteoarthritis (KOA) can be categorized into different severity levels based on magnetic resonance (MR) images. In the normal knee (KL 0 grade), there is no visible damage. In mild KOA (KL 1 grade), minor loss is observed in the femoral cartilage. Mild KOA (KL 2 grade) shows some loss in bone density in the femoral and patella cartilage. Severe KOA (KL 3 grade) exhibits tearing in the femoral cartilage, while in the most severe stage (KL 4 grade), almost the entire femoral cartilage seems to have worn down, exposing the bone and causing tissue signal inhomogeneity.

Knee cartilage segmentation plays a crucial role in diagnosing KOA. By accurately identifying and delineating the cartilage boundaries from medical imaging such as MRI, segmentation techniques enable quantification and assessment of cartilage health. This information helps in evaluating the severity of osteoarthritis, tracking disease progression, and guiding timely treatment decisions. Additionally, precise cartilage segmentation aids in detecting early cartilage degeneration and monitoring the effectiveness of interventions or therapies. Overall, knee cartilage segmentation plays a vital role in providing objective and quantitative measurements for diagnosing and managing osteoarthritis.

The remaining paper is organized with Section II explaining the related work, Section III describes the methodology, the Experiment work is presented in Section IV. Section V gives evaluation metrics and results and discussions are given in Section VI before Section VII concludes the paper.

II. RELATED WORK

Segmentation of cartilage can be performed manually or by using computational approaches. Computational approaches can be classified into two which is semi-automatic or automatic methods. Manual segmentation of cartilage results usually gives more reliable outcomes compared to computational approaches. In a manual approach, cartilage will be segmented slice by slice from 2D MR images. Although manual segmentation outcome has high accuracy and sensitivity [6] and has been widely used to evaluate the performance of semi-automatic or automatic segmentation methods, it requires extra effort to the user and is time-consuming. This may result in inter and intra observer variability between the experts [7].

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The region-based segmentation approaches have been classified under semi-automatic and conventional segmentation approaches. Based on [8], [9] this approach use the concept of exploiting the homogeneity property values of neighboring pixels such as grey-level intensity, texture, and edge [10]. Often, the initialisation point of segmentation or also known as seed points, $S_i = \{S_1, S_2, \ldots, S_n\}$ will be manually placed by a user. After placing the seed points, the algorithm will next begin to search for homogeneous pixels in neighbouring pixels [11] and keep updating the corresponding region’s mean and expanding until the similarity requirement is met. In equation (1), let $T$ represent the set of all pixels that adjacent to at least one of the pixels in $S_i$ where $\text{nb}(x)$ denotes the set of nearest neighbours pixel of $x$.

$$T = \left\{ x \notin \bigcup_{i=1}^{n} S_i \mid \text{nb}(x) \cap \bigcup_{i=1}^{n} S_i \neq \emptyset \right\} \quad (1)$$

This approach has been widely used in the past two decades. However, the traditional region-growing approach is incapable to deal with the inhomogeneous image quality, especially for knee MR images. With this limitation, further research has been conducted by combining this approach with other image-processing approaches to achieve robust segmentation results. The author in [12] proposed a region growth approach by introducing the binary-class intensity-based local clustering to segment knee cartilage from a background image. Binary-class intensity-based local clustering is a voxel labeling of cartilage or non-cartilage. This approach was introduced to differentiate the class of unlabelled areas based on the distance to the knee bone and the contrast of the boundaries area. Finally, a 3D cartilage model was generated to tackle the issue of intensity inhomogeneity. The authors in [13], [14] performed a knee cartilage segmentation model based on a multistage region growth approach. This approach used a median filter to filter the image noise while edge detection and thresholding were used to remove the background image at the pre-processing stage. Next, the pre-segmentation stage by using region growth will take place. Output from this stage was bone and cartilage region mask. Finally, the bone region mask will be removed to leave out the cartilage.

Recently, there have been significant advancements in automatic biomedical image segmentation using deep neural network techniques [15], [16], [17]. In the context of segmenting knee joint structures, Burton et al. [18] initially employed two-dimensional (2D) tri-planar CNNs (axial, coronal, and sagittal planes) to classify pixel labels (background or tibial cartilage) by considering local image patches surrounding each pixel. However, Ronneberger et al. [19] identified two limitations with this approach: excessive redundancy and a trade-off between localization accuracy and the utilization of context. To address these issues, they proposed a dense prediction network called U-Net, which incorporated skip connections. This architecture encompassed both low-level and high-level features for voxel classification and was subsequently employed for knee joint segmentation by Liu et al. [20], Zhao et al. [21], and Ambellan et al. [22]. Generally, the U-Net employed pixel-wise or voxel-wise loss functions such as cross-entropy loss and dice loss. However, the resulting segmentation lacked spatial consistency [23]. The reason is that U-Net consists of only Convolution layers that suffer from the training-related issues of vanishing gradient and dead neurons considered as the main research gap being addressed in this work [24]. The main contribution of this paper is to propose an automatic deep learning segmentation model which can segment knee cartilage in the image efficiently and to do so we have introduced the proposed squeeze and excitation block in U-Net which will work alongside convolution layer of U-Net so that the architecture learning process goes smoothly. Additionally updated loss function is also used, which will help the proposed model to perform the task effectively.

III. METHODOLOGY

A. Proposed Squeeze and Excitation Block

The proposed Squeeze and Excitation block, shown in Fig. 1, is composed of residual connection with Squeeze and Excitation block. The residual connection has been proven to be an effective way to overcome the problem of vanishing gradient when it comes to learning features in deep neural networks [25]. In [26], the authors have proposed SE-Net (Squeeze-and-Excitation networks), which is the ImageNet Challenge winner. The SE-Net adds the Attention mechanism to the feature channel dimension to learn the weight of each dimension through the loss function, and to learn the residual connection according to the importance of each channel feature. The image input after passing from convolution layers, delivers the feature map with size of $H \times W \times C$ as the output. Then this feature map is passed colored through Global pooling layer which reduces the size to $1 \times 1 \times C$. Again, the output of global pooling layer is passed through fully connected layer and ReLU activation function thus reducing its channel by $r$ as shown in Fig. 1. This process is known as the Squeeze operation. The output of Squeeze operation is then upsampled to $1 \times 1 \times C$ using another fully connected layer followed by sigmoid activation function, which gives weights for each of the channels, a process referred to as Excitation operation. Mathematically the Squeeze and Excitation operation can be expressed as in equations (2) to (5):

$$o_c = F_{sq}(x_c) = \frac{1}{H \times W} \sum_{h}^{H} \sum_{w}^{W} x_c(h, w) \quad (2)$$

Where $F_{sq}$ represents squashing operation, $x_c$ represents feature map of $X_{m}$ on the $C$ dimension, $H$ and $W$ denotes the height and width of the feature map, $h$ and $w$ exemplify the abscissa and ordinate of a certain point on the feature map. The correlation between each feature channel is learned through the two fully connected layers and the ReLU activation function, and the correlation between the features is normalized by the Sigmoid function to obtain the weights for each feature channel, then these weights are multiplied by the input feature channel.

$$t_c = F_{ex}(o_c) = \text{Sigmoid}(W_2(\text{RELU}(W_1(o_c)))) \quad (3)$$

$$\bar{x} = F_{scale}(x_c \ast t_c) \quad (4)$$
Fig. 1. Proposed squeeze and excitation block.

Where $F_{ex}$ indicates an excitation operation, $W_1$ and $W_2$ show weights of two fully connected layers.

$$X_{m+1} = \text{RELU} \left( \bar{x} + X_m \right)$$  \hspace{1cm} (5)

$X_m$ signifies the input to the proposed block. Squeeze and Excitation block will then be embedded into the residual connection to improve the residual characteristics of different channels according to the degree of importance that will help in learning ability.

B. PSU-Net

Fig. 2 and Fig. 3 show the U-Net and PSU-Net architecture to segment the knee image data in NRRD format. It has the same architecture as that of the standard U-Net in the Down-convolutional part: each layer is consisting of two convolution layers followed by the activation function and then the input is down-sampled by the Maxpool layer in two-layered step. In the Up-Convolutional part each input coming from bottleneck layer is then upsampled by $2 \times 2$ in a two-layered step, followed by two $3 \times 3$ convolution layers and then followed by an activation function. The bottleneck layer connects Down-Convolution part to the Up-Convolution part that helps in stabilizing training and transferring information from Down-Convolutional part to Up-Convolutional part. The last convolution layer helps in reducing the number of labels to three according to number of class in our dataset. We also used Batch normalization before activation functions.

In PSU-NET the two convolution layers block in the Up-Convolutional part and Down-Convolutional parts are replaced by the proposed Squeeze and Excitation block as shown in Fig. 3. Each unit of Down-convolutional part includes a proposed Squeeze and Excitation Block to extract image features and a down-sampling layer. The down-sampling is done using maxpool layer. The channel size is reduced to half, each unit of Up-Convolutional part includes a proposed Squeeze and Excitation Block and an upsampling layer. Upsampling uses transposed convolution, the channel size is expanded by two times. The last layer consists of convolution layer that helps in reducing the number of class to three - the femoral, tibia and patella in our case.

C. Loss Function

Knee image segmentation is done to divide the voxels of the cartilage of knee image into three classes: femoral, patella and tibia, using neural network. The pixel value of cartilages is approximately the same as different organs of the knee like meniscus etc. In order to make the neural network pay more attention to the cartilage, we give these pixels of cartilage greater weights to improve knee image segmentation. The weights can be expressed as in equation (6):

$$w_i = \frac{\left( \frac{1}{N_i} \right)^2}{\sum_{i=1}^{3} \left( \frac{1}{N_i} \right)^2}$$  \hspace{1cm} (6)
where $w_i$ represents weight for each class, $N_i$ is the total number of pixels in each class of the image. This paper uses a weighted multi-class loss function, which is a combination of Dice loss function and cross-entropy loss function to optimize the model. The formulas of the Dice loss function and the cross-entropy loss function are mathematically expressed in equation (7) and (8):

$$\text{loss}_{\text{Dice}} = 1 - \sum_{n=1}^{M} \sum_{i=1}^{C} w_i \cdot \frac{2 \cdot y_{ni} y'_{ni}}{y_{ni} + y'_{ni}}$$  \hspace{1cm} (7)

$$\text{loss}_{\text{ce}} = - \sum_{n=1}^{M} \sum_{i=1}^{C} w_i y_{ni} \log (y'_{ni})$$  \hspace{1cm} (8)

Where $M$ represents the total number of pixels in the image in a batch, $C$ represents the class (femoral, patella, tibia), $w_i$ indicates the class weight obtained by using equation (6), and $y_{ni}$ indicates the true probability value of the pixel belonging to class $i$, $y'_{ni}$ represents the predicted probability value. Finally, the weighted loss function is as expressed in (9):

$$\text{Loss} = \lambda_1 \text{loss}_{\text{Dice}} + \lambda_2 \text{loss}_{\text{ce}}$$  \hspace{1cm} (9)

IV. Experiments

A. Dataset

This study is carried out on 30 datasets from OAI [27] that can be found at http://oai.epi-ucsf.org/datarelease/About.asp, consisting of 160 slices each of Knee MR images. The datasets consist of various KL grades as listed in Table I. The KL grades are generally referred to in the medical images to reflect the damage of cartilage in between joints. We labelled the cartilage of these 30 datasets using slicer [28] software. We divided these 30 datasets into 3 categories. 23 of training datasets, 5 of validation datasets and 2 of testing datasets.

<table>
<thead>
<tr>
<th>KL Grades</th>
<th>Dataset Numbers</th>
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<tbody>
<tr>
<td>0</td>
<td>3</td>
</tr>
<tr>
<td>1</td>
<td>4</td>
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<tr>
<td>2</td>
<td>5</td>
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<td>6</td>
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<td>4</td>
<td>7</td>
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</table>

B. Experimental Setup

The operating environment of this study is NVIDIA cuDNN7.5, CUDA10.0, Python 3.4, Anaconda, the hardware configuration is GTX 1060Ti GPU, 1 TB capacity hard disk, and the deep learning framework is built using Keras [29] with Tensorflow [30] as backend. Other libraries also used like SimpleITK [31], matplotlib [32], Scikit learn [33] and numpy [34] for reading, processing NRRD data and for other functions as well. The neural network learns the weights of each layer through training data and selects the optimal model through validation data to verify the performance of the model. Each epoch of learning randomly selects 80 percent of the training model from the training set, and the remaining 20 percent is used to verify the model to improve model learning ability.

We have entered data in batches to reduce training time. In this study, the Adam optimization method is used to optimize the network, and the weighted joint loss function proposed in equation (9) is used to judge the training process of the network model. We have adjusted the parameters according to the training results, the parameter of loss function $\lambda_1$ and $\lambda_2$ Set to 0.3 and 0.7, respectively, set the training batch size to 2, and iterate the training data set for 50 epochs.

V. Evaluation Metrics

In order to quantitatively evaluate the segmentation performance of the algorithm in this paper, Dice coefficient (F1) and Intersection over union (IoU) are used as evaluation metrics, and these scores were measured by calculating the regional similarity between deep neural neural network predicted result and expert annotated result.

F1 score can be calculated by using following mathematical equation:

$$F1 = \frac{2|A \cap B|}{|A| + |B|}$$  \hspace{1cm} (10)

IoU score can be expressed mathematically as:

$$\text{IoU} = \frac{|A \cap B|}{|A \cup B|}$$  \hspace{1cm} (11)

In above equation 10 and 11, A represents the ground truth of knee cartilage and B is the predicted segmentation result by segmentation models.

VI. Results and Discussion

Using the model of this paper to trained on 30 datasets of knee images to calculate the dice coefficient of our proposed model on training dataset and validation datasets during training of each epoch using dice coefficient and intersection over union. The box plot shown in Fig. 4 indicates the distribution of the dice coefficient of the femoral, tibia and patella cartilages during training. Usually, each of the datasets consists of 160 slices in which we have slices with cartilage and without cartilage. The average training dice coefficient of our proposed model during training on those slices which have femoral cartilage is up to 0.925, the average training dice coefficient on those segmented slice which has tibia cartilage is up to 0.945, and the average training dice coefficient on those slices which have patella cartilage is 0.978. The distribution of dice coefficient is also relatively concentrated, which shows that the network can effectively segment the cartilage under normal circumstances. Table II shows the average validation dice coefficient (F1) and intersection over union (IoU) of our model. The average validation dice coefficient of femoral cartilage is 0.79 and the average validation IoU is 0.74, the average validation dice coefficient of the tibia is 0.85 and the average validation IoU is 0.78 and for patella cartilage, the average validation dice coefficient is 0.82 and the average validation IoU is 0.73. The reason of low validation F1 score from training F1 scores is that in the evaluation process of validation datasets, some slices in datasets have no cartilage.
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<table>
<thead>
<tr>
<th>Test Dataset (slices)</th>
<th>Ground Truth</th>
<th>Predicted Mask</th>
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<td><img src="image11.png" alt="Image" /></td>
<td><img src="image12.png" alt="Image" /></td>
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</table>

Fig. 5. Input knee image with manually annotated Groundtruth and predicted segmentation mask using PSU-Net.

<table>
<thead>
<tr>
<th>Axial View</th>
<th>3D cartilage in slicer</th>
</tr>
</thead>
<tbody>
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<td><img src="image13.png" alt="Image" /></td>
<td><img src="image14.png" alt="Image" /></td>
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Fig. 6. Axial view of two test datasets and their predicted 3D segmented cartilage using PSU-Net.
An Integrated Framework for Relevance Classification of Trending Topics in Arabic Tweets

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Abstract—Social media platforms such as Twitter are a valuable source of information about current events and trends. Trending topics aim to promote public events such as political events, market changes, and other types of breaking news. However, with so much data being generated, it would be difficult to identify relevant tweets that are related to a particular trending topic. Therefore, in this paper, an integrated framework is proposed for the detection of the degree of relevance between Arabic tweets and trending topics. This framework integrates natural language processing, data augmentation, and machine learning techniques to identify text that is likely to be relevant to a given trending topic. The proposed framework was evaluated using a real-life dataset of Arabic tweets that was collected and labeled. The results of the evaluation showed that the proposed framework achieved the highest macro F1 score of 82% in binary classification (relevant/irrelevant) and 77% in categorical classification (degree of relevance), which outperforms the current state of the art.

Keywords—Trending topics; social media platforms; machine learning; Arabic relevance classification; data augmentation

I. INTRODUCTION

In recent years, social media platforms have become a significant source of information for real-time events and trending topics [1]. The vast amount of user-generated content on these platforms, especially Twitter, provides a wealth of information for various applications, including sentiment analysis, event detection, and trend analysis [2]. Twitter is now a real-time information distribution channel that is utilized for news, politics, and advertising [3]. Users can use the trending topics feature or popular hashtags to discover current popular news. However, trending topics are often irrelevant to the content being discussed. Therefore, trending topics relevance classification is considered as an important text analysis task for Twitter data. Trending topics relevance aims to identify relevant tweets that are related to a particular trending topic. However, this task is challenging due to the noisy and unstructured nature of social media text, misspellings, slang, and various other factors that can affect the relevance of a tweet to a trending topic [4], [5].

Currently, the hashtag trending topic feature is used by users to find out what topics are currently popular on Twitter [6], [7]. Some users employ trending topics, to get more attention to their tweets. However, the trending topic is irrelevant to the topic being discussed by the tweet itself. For example, some people include political trending topics that are popular during the election period in their tweets, but the tweet content does not include any political topics.

Irrelevant content for trending topics reduces the level of communication offered by social media networks. They pollute social networks and affect how people perceive the contents of the internet. The user experience will significantly decrease if someone is excessively exposed to irrelevant information, which would result in user losses for the social service provider [8]. Therefore, social platforms must develop algorithms to filter unwanted information and determine the relevance of content to trending topics.

Several approaches have been proposed for automatic learning and detection including trending topics detection on social media [9]. Previous research has extracted and utilized a wide range of features, from simple to complex, as well as a wide range of learning and classification algorithms, from traditional machine learning techniques to deep learning [9], [10].

However, these approaches face several challenges specific to Arabic language processing, such as the absence of diacritics, the presence of multiple dialects, and the use of Arabic script, which can affect the accuracy of the model. Additionally, the availability of labeled datasets for Arabic text relevance classification in trending topics is limited, making it difficult to train and evaluate models on this task.

This paper focuses on the problem of detecting the relevance level of Arabic texts related to trending topics on Twitter. Specifically, we aim to identify tweets that are relevant to a given trending topic in order to be able to filter out irrelevant content. A novel framework that integrates several techniques, including text preprocessing, feature extraction, and machine learning algorithms is proposed to classify tweets associated to trending topics into relevant/irrelevant. Furthermore, this framework is able to categorize the relevant content into low, medium, and high based on its degree of relevancy. This work is based on the trending topics and tweets related to Yemeni politics written in Arabic.

The main contributions of this paper are the following:

- Build a public and available dataset of Arabic tweets related to trending topics in both binary and categorical classes.
- Develop a trending topics relevance text classification framework using machine learning algorithms with two scenarios: binary and categorical classes.
- Apply data augmentation to enhance the performance of the framework.

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As will be explained later, to overcome the challenge of limited labeled datasets, data augmentation techniques [11] were applied to improve the performance of the model. Several techniques will be applied including word embedding-based techniques, to generate additional training data, which helps the model better generalize to unseen data and handle noisy, misspelled tweets and improve the overall performance of the model.

The proposed framework is beneficial for researchers, journalists, and businesses who require analysis of trending topics on social media platforms. It can also be used to extract insights about public opinions, monitor the impact of events, and discover emerging trends. The rest of the paper is organized as follows: Section II discusses the background and related work; Section III explains the methodology used in our approach; Section IV presents the experimental results; Discussion are in section V; Finally, section VI concludes the paper and outlines possible directions for future work.

II. BACKGROUND AND RELATED WORK

A. Trending Topics

Trending topics refer to popular and widely discussed topics on social media platforms such as Twitter, Facebook, and Instagram. These topics are characterized by a large volume of posts or tweets that use a specific hashtag or keyword, and they often reflect current events, news, and opinions that are of interest to the public [6]. Trending topics are important because they provide valuable insights into public opinion and social trends. They allow individuals and organizations to track and monitor the conversation around a particular topic, and they can be used to identify emerging trends and issues in real-time [12]. In recent years, the analysis of trending topics has become an important field of research. Researchers have explored various approaches and techniques to identify and analyze these topics based on different domains and languages. For example, some studies have focused on identifying trending topics related to politics, sports, entertainment, or health, while others have looked at trending topics in different languages such as English, or Arabic. Some of these approaches include [13], [14]:

- Keyword-based approach: This approach involves using a set of pre-defined keywords to identify the trending topics.
- Text classification approach: This approach involves training a classifier on a labeled dataset to identify the trending topics.
- Topic modeling approach: This approach involves using topic modeling techniques such as Latent Dirichlet Allocation (LDA) to identify the trending topics.
- Hybrid approach: This approach combines multiple methods to improve trending topic detection accuracy. For example, a hybrid approach may use both keyword-based and text classification methods to identify trending topics.

Overall, the choice of the approach depends on the specific requirements of the task, such as the available resources, the amount of labeled data, and the desired level of accuracy.

Trending topics analysis has many applications, including social media monitoring, reputation management, crisis management, and market research. It can be used by businesses, governments, and other organizations to gain insights into public opinion, track the effectiveness of their social media campaigns, and respond to emerging trends and issues in real time.

B. Related Work

1) Topic detection: There is limited research on trending topics relevance text classification in social media. However, significant research has focused on related areas such as topic detection and sentiment analysis [15]. Here, we only provide a brief overview of a few of the most pertinent studies for topic detection. There is a wide range of techniques employed for topic analysis on Twitter. Studies that utilize machine learning techniques generally rely on supervised learning [16], [17], [7], while others adopt a hybrid approach that incorporates latent Dirichlet allocation (LDA) [18]. The combination of sentiment analysis and topic detection has been employed to analyze content related to COVID-19 in Brazil and the USA [3]. Lee et al. [7] categorized Twitter Trending Topics into 18 broad categories, including sports, politics, and technology, using a text-based classification approach with a Bag-of-Words and a network-based classification.

There have been several significant works related to topic detection on social media for the Arabic language. An assimilated model was introduced to identify events from Arabic Twitter data by Alsaedi et al. [19]. Their main objective was to distinguish disruptive events from other events in social media data streams. The model they developed relies on the frequency of terms occurring together over time. In a different study [20], a comprehensive framework for event detection was introduced. The authors emphasized the importance of temporal, spatial, and textual characteristics of each cluster in event detection. They compared the effectiveness of their proposed framework with LDA and demonstrated that LDA was not suitable for analyzing short messages like tweets. In [21], a feature-pivot method was employed to identify bursty features of terms from Arabic Tweets. The approach employed TFIDF, entropy, and stream chunking to capture bursty terms that were highly relevant to a particular event during a given time interval. The document-pivot method was introduced in [22] to extract trending topics for Arabic Twitter users. These works serve as examples of the diverse range of approaches that can be utilized to detect topics on Twitter.

2) Relevance text classification of trending topics: Despite the abundance of research on topic detection on Twitter, comparatively less work has focused on classifying relevant text within trending topics, which is the primary focus of our proposed approach. A framework model known as TORHID (Topic Relevant Hashtag Identification) was introduced in a research paper [23] to identify and retrieve hashtags relevant to a particular topic on Twitter. The model utilized small tweets of a hashtag as seeds and employed a Support Vector Machine to classify new tweets as relevant or irrelevant. According to the reported results, the TORHID model achieved an accuracy of 67.25%. Cahyani et al. [4] conducted research on the relevance classification of tweet content and trending topics on social media. The study focused on political tweet data related...
to Indonesian trending topics and employed Support Vector Machine (SVM) for classifying tweets as either relevant or irrelevant. The study reported an F1 measure of 70% for the applied model. However, the absence of research on trending topics relevance text classification in Arabic represents a noteworthy gap in the existing literature, which we aim to address.

III. PROPOSED FRAMEWORK FOR RELEVANCE CLASSIFICATION OF TRENDING TOPICS IN ARABIC TWEETS

As shown in Fig. 1, an integrated framework for Relevance Classification of Trending Topics in Arabic Tweets (RCTAT) is proposed. This framework consists of two main components: data preparation and augmentation and trending topics relevance text classification. In the following subsections, each component will be described in details.

A. Data Preparation and Augmentation

Data preparation involves collecting and cleaning the data, while data augmentation involves generating more data from the existing data to improve the model’s generalization and reduce overfitting [24]. The following steps were taken in data preparation and augmentation for Relevance Classification of Trending Topics in Arabic Tweets:

1) Data collection: The first step was to collect Arabic tweets from Twitter. The tweets were collected using the Twitter Streaming API starting December 2019 to April 2020, using the query “lang: ar” (language is Arabic) and the trending hashtag (#YEMEN).

2) Data cleaning: To obtain distinct tweets, several cleaning steps were applied. These steps involved removing diacritics, repeated characters, and punctuations from the tweets. Additionally, both Arabic and non-Arabic alphabets were normalized to ensure consistency. Furthermore, the Python NLTK library† was utilized to perform Arabic light stemming (ARLSTem) and remove Arabic stop words. These cleaning techniques enhanced the quality and readability of the tweets, making them more suitable for further analysis.

3) Relevant terms extraction: To build a list of commonly used terms related to the situation in Yemen, the 313k distinct tweets are analyzed. The frequency analysis technique [25] is used to identify the most frequently used and important terms by counting the occurrence of words in the tweets. As shown in Fig. 2, a list of terms was extracted and used to construct our dataset. These terms cover a variety of topics associated with the situation in Yemen.

4) Manual data labeling: From the 313 k unique tweets, a random sample of 5,000 was chosen for manual labeling. To construct our relevant/irrelevant dataset, the extracted tweets have been manually annotated into six categories: irrelevant (1), 30% relevant (2), 50% relevant (3), 65% relevant (4), 85% relevant (5), and fully Relevant (6) to the situation in Yemen. To assist in the labeling process, a website† was developed and 15 Arabic annotators helped in this process. The annotators had diverse educational qualifications, including advanced educational degrees such as B.C, M.S., or Ph.D. with an age range spanning from 25 to 40 years old. This combination of advanced education and diverse age range allowed the annotators to bring a wealth of knowledge and perspectives to the annotation task. Each annotator was provided with our definition of relevant, relevant ranges, and irrelevant content, along with relevant examples, before commencing the labeling process. The criteria for definition are as follows:

- If one or more relevant terms are discussed, a tweet is typically considered fully relevant.
- If no relevant term is included, a tweet is considered fully irrelevant.
- If a tweet contains a mixture of relevant and irrelevant terms, it is considered partially relevant.

The 5,000 tweets were divided into five groups of 1,000 tweets each. Three different annotators were assigned to work independently on each set of 1,000 tweets. This means that each tweet was annotated by three different annotators, resulting in three values for each tweet based on the six categories, with each category from a different expert. This expedited the process and enabled multiple experts to label the same tweet. After the annotations were completed, a tweet-irrelevant ratio (r) was calculated for each tweet by adding the three annotation results and dividing the total by the highest summation result (N).

\[
r = \frac{\sum_{i=1}^{3} \text{annotatorResult}_i}{N}
\]

Where :

- \(\text{annotatorResult}_i\) is value from 1 to 6
- \(N\) is the highest summation result = 18

After calculating the tweet-irrelevant ratio (r) for each tweet, the tweets were further categorized into binary (relevant or irrelevant) and categorical (low, medium, high) datasets. This categorization was likely based on a threshold value determined by the tweet-irrelevant ratio (r). In order to determine the optimal threshold for tweet categorization, we adopted an experimentation approach. A series of experiments were conducted by systematically varying the threshold values and assessing their impact on the performance of our classification model. A representative dataset of tweets was collected, and their tweet-irrelevant ratios (r) were computed. Through iterative adjustments of the threshold values and comprehensive analysis of evaluation metrics including accuracy, precision, recall, and F1-score, we successfully identified the threshold that yielded the best performance. This meticulous experimentation and analysis process allowed us to select the threshold value that maximized our chosen evaluation metric, ensuring the accurate differentiation of relevant and irrelevant tweets within our tweet categorization framework. Furthermore, these tweets were appropriately assigned to their respective relevance categories.

For example, if the threshold value was set at 0.34, tweets with a tweet-irrelevant ratio (r) higher 0.34 would be considered relevant, and those with a ratio of 0.34 or below 0.34 would be considered irrelevant. The binary dataset would then consist of two categories: relevant and irrelevant. Similarly, the

†https://www.nltk.org
†https://tweettag.000webhostapp.com/login.php
categorical dataset would be created by dividing the relevant tweets into three categories based on their tweet-relevant ratio \((r)\), such as low (0 to 0.17), medium (0.17 to 0.34), and high (0.34 and above). This categorization was used to evaluate the performance of the framework in identifying relevant tweets and to compare it against human annotations. Tables I and II shows the number of tweets in both manual binary (relevant or irrelevant) and categorical (low, medium, high) datasets.

<table>
<thead>
<tr>
<th>Term count</th>
<th>Term</th>
<th>English Translation</th>
<th>Term count</th>
<th>Term</th>
<th>English Translation</th>
</tr>
</thead>
<tbody>
<tr>
<td>335826</td>
<td>Yemen</td>
<td></td>
<td>21992</td>
<td>UAE</td>
<td></td>
</tr>
<tr>
<td>33083</td>
<td>Sinai</td>
<td></td>
<td>37398</td>
<td>Saudi Arabia</td>
<td></td>
</tr>
<tr>
<td>18881</td>
<td>Arab coalition</td>
<td></td>
<td>8371</td>
<td>Charlie</td>
<td>Marib</td>
</tr>
<tr>
<td>14981</td>
<td>legitimacy</td>
<td></td>
<td>14405</td>
<td>The south</td>
<td></td>
</tr>
<tr>
<td>10090</td>
<td>Aden</td>
<td></td>
<td>2851</td>
<td>Nihm</td>
<td></td>
</tr>
<tr>
<td>12580</td>
<td>Sa'da</td>
<td></td>
<td>6085</td>
<td>Al-Jawf</td>
<td></td>
</tr>
<tr>
<td>4530</td>
<td>President Mub</td>
<td></td>
<td>19054</td>
<td>The army</td>
<td></td>
</tr>
<tr>
<td>4940</td>
<td>president</td>
<td></td>
<td>3515</td>
<td>Terrorism</td>
<td></td>
</tr>
<tr>
<td>24516</td>
<td>Iran</td>
<td></td>
<td>19887</td>
<td>The war</td>
<td></td>
</tr>
</tbody>
</table>

5) Expanding the annotated dataset: In order to increase the size of the manually labeled dataset, which can be expensive and time-consuming to create manually, we utilized text data augmentation techniques to automatically generate a labeled dataset from the existing one. Data augmentation is employed to improve the classification of relevant tweets in trending topics.

Data augmentation aims to tackle overfitting at the data level, address class imbalance, and enhance the model’s generalization [24]. Increasing the diversity of training samples through data augmentation can help the model learn more fundamental features of the data, leading to a higher quality classifier. Tables III and IV show how the size of the binary (irrelevant) and categorical (low, medium) samples changed after the data augmentation technique was employed. In our previous work [26], data augmentation techniques were applied to increase the size of our datasets. This approach aimed to address the class imbalance, avoid overfitting, and enhance the generalization of the model. The authors utilized word embedding techniques [27], specifically the AraVec word vectors trained on Arabic content from Wikipedia and Twitter. They replaced words in the dataset with synonyms based on similarity scores obtained from the word vectors. By applying a random ratio of 50% to 70% for token replacement, they effectively increased the diversity and quantity of training samples. This data augmentation process resulted in a substantial increase in the size of our datasets, which can potentially improve the accuracy and performance of our model by providing a more comprehensive representation of the data.

Table III shows that the size of the irrelevant binary class increased from 1589 to 3236 tweets after data augmentation. Similarly, the low categorical class increased from 791 to 2439 tweets, and the mid categorical class increased from 798 to 2388 tweets as shown in Table IV. This suggests that the data augmentation technique was successful in increasing the size of the datasets, which can help improve the performance of the model by increasing the diversity of the training samples.
B. Trending Topics Relevance Text Classification

This section describes the Trending topics relevance text classification process by applying different machine learning techniques. This process includes the following steps: Data pre-processing, feature extraction, and finally, applying various machine learning classifiers.

1) Data pre-processing: In order to ensure data cleansing and eliminate noise that could impact the accuracy of the system, various techniques were employed on the dataset. These techniques include tokenization, normalization, removal of diacritics, removal of repeated characters, removal of punctuations, removal of stop words, removal of non-Arabic alphabets, and light stemming.

2) Content features extraction: Once the data is prepared and augmented, features need to be extracted from the text to represent it into a numerical representation that can be used by the machine learning models. The features listed in Table V were employed for this purpose.

3) Machine learning classification: Several machine learning classifiers have been utilized to classify relevance text in Trending topics. Three classifiers were trained based on the extracted features to assess their effectiveness in classifying relevant content. The classifiers used were Naïve Bayes, Logistic Regression, and LinearSVC, which were selected because they have been widely used as a baseline in previous works on Arabic classification. The experiments’ outcomes are discussed in Section IV.

IV. Experiments

Experiments were conducted to evaluate the quality of manually labeled and augmented datasets using the features extracted in Section III-B2. Two types of Arabic relevance classification were explored: binary classification (relevant/irrelevant) and categorical classification (low, medium, high). The dataset comprised 5000 tweets in the non-augmented labeled dataset and up to 8000 tweets in the augmented dataset, as shown in Sections III-A4 and III-A5. The Arabic relevance classification was performed using LinearSVC (SVC), Naïve Bayes (NB), and Logistic Regression (LR) classifiers with 10-fold cross-validation on both datasets.

The results of binary classification and categorical classification for the Arabic relevance classification are presented in the following subsections:

A. Binary Classification

In this section, we present the results of our experiments for Arabic trending topic relevance text classification using binary classification, as shown in Table VI and Fig. 3.

The results showed that the Logistic Regression classifier with the n-gram TF-IDF feature achieved superior classification performance. Specifically, the classifier attained a macro F1 (M-F1) score of 72% for the non-augmented dataset. On the other hand, applying the same feature with the SVC classifier resulted in the best classification performance for the augmented dataset. The classifier obtained a macro F1-score of 82%.

Likewise, the n-gram TF-IDF feature with the LR classifier and the word TF-IDF feature with the SVC classifier produced the highest precision (P) values of 71% on the non-augmented dataset. On the other hand, the n-gram TF-IDF feature with the SVC classifier achieved the highest precision value of 82% on the augmented dataset.

In terms of recall (R), the SVC classifier with the n-gram TF-IDF and character TF-IDF features yielded the highest recall of 74% on the non-augmented dataset, while the SVC classifier with the n-gram TF-IDF feature achieved the highest recall of 82% on the augmented dataset. Finally, the n-gram TF-IDF feature with the Logistic Regression classifier yielded the highest accuracy (A) of 76% on the non-augmented dataset, while the n-gram TF-IDF feature with the SVC classifier produced the highest accuracy of 82% on the augmented dataset.

B. Categorical Classification

In this section, we present the results of our experiments for Arabic trending topic relevance text classification using categorical classification, as shown in Table VII and Fig. 4.

The results showed that, using the Logistic Regression classifier with n-gram TF-IDF and word count features, as well as the SVC classifier with word TF-IDF, led to significantly better classification performance. These classifiers achieved a macro F1-score of 51% with the non-augmented dataset. On the other hand, the SVC classifier that utilized word, n-gram, and character TF-IDF features obtained the best classification performance on the augmented dataset. This classifier achieved a macro F1-score of 77%.

Likewise, the highest precision value of 51% was achieved with the SVC classifier using the word TF-IDF feature, and the
### TABLE VI. Binary Dataset - Experiment 1 Confusion Matrix

<table>
<thead>
<tr>
<th>Classifier</th>
<th>Feature</th>
<th>P</th>
<th>R</th>
<th>A</th>
<th>M-F1</th>
<th>P</th>
<th>R</th>
<th>A</th>
<th>M-F1</th>
<th>P</th>
<th>R</th>
<th>A</th>
<th>M-F1</th>
</tr>
</thead>
<tbody>
<tr>
<td>Non-Augmented</td>
<td>Word Count</td>
<td>0.699</td>
<td>0.699</td>
<td>0.739</td>
<td>0.698</td>
<td>0.625</td>
<td>0.637</td>
<td>0.692</td>
<td>0.629</td>
<td>0.7</td>
<td>0.71</td>
<td>0.74</td>
<td>0.7</td>
</tr>
<tr>
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<td>0.709</td>
<td>0.708</td>
<td>0.746</td>
<td>0.707</td>
<td>0.63</td>
<td>0.64</td>
<td>0.694</td>
<td>0.633</td>
<td>0.69</td>
<td>0.7</td>
<td>0.74</td>
<td>0.7</td>
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<tr>
<td>TF-IDF (n-gram-level)</td>
<td>0.654</td>
<td>0.736</td>
<td>0.753</td>
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<td>0.76</td>
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<tr>
<td>TF-IDF (character-level)</td>
<td>0.654</td>
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<td>0.755</td>
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<td>0.69</td>
<td>0.63</td>
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<td>0.68</td>
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<td>TF-IDF (n-gram-level)</td>
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<tr>
<td>TF-IDF (character-level)</td>
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<td>0.814</td>
<td>0.813</td>
<td>0.736</td>
<td>0.736</td>
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<td>0.735</td>
<td>0.769</td>
<td>0.769</td>
<td>0.769</td>
<td>0.768</td>
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</table>

Fig. 3. Plot of binary dataset results.

### TABLE VII. Categorical Dataset - Experiment 2 Confusion Matrix

<table>
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<tr>
<th>Classifier</th>
<th>Feature</th>
<th>P</th>
<th>R</th>
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<th>M-F1</th>
<th>P</th>
<th>R</th>
<th>A</th>
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<th>R</th>
<th>A</th>
<th>M-F1</th>
</tr>
</thead>
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<td>0.68</td>
<td>0.5</td>
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<td>0.51</td>
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<tr>
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<td>0.53</td>
<td>0.69</td>
<td>0.51</td>
<td>0.45</td>
<td>0.48</td>
<td>0.66</td>
<td>0.45</td>
<td>0.48</td>
<td>0.51</td>
<td>0.68</td>
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<td></td>
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<td>TF-IDF (n-gram-level)</td>
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<td>0.71</td>
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<td>0.66</td>
<td>0.45</td>
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<td>0.52</td>
<td>0.69</td>
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</tr>
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<td>TF-IDF (character-level)</td>
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<td>0.71</td>
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<td>0.44</td>
<td>0.48</td>
<td>0.67</td>
<td>0.45</td>
<td>0.47</td>
<td>0.49</td>
<td>0.66</td>
<td>0.48</td>
<td></td>
</tr>
<tr>
<td>Augmented</td>
<td>Word Count</td>
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<td>0.726</td>
<td>0.716</td>
<td>0.55</td>
<td>0.56</td>
<td>0.68</td>
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<td>0.61</td>
<td>0.73</td>
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<tr>
<td>TF-IDF (word-level)</td>
<td>0.769</td>
<td>0.779</td>
<td>0.783</td>
<td>0.772</td>
<td>0.55</td>
<td>0.55</td>
<td>0.67</td>
<td>0.55</td>
<td>0.59</td>
<td>0.6</td>
<td>0.73</td>
<td>0.59</td>
<td></td>
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<tr>
<td>TF-IDF (n-gram-level)</td>
<td>0.767</td>
<td>0.776</td>
<td>0.78</td>
<td>0.769</td>
<td>0.55</td>
<td>0.56</td>
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<tr>
<td>TF-IDF (character-level)</td>
<td>0.767</td>
<td>0.776</td>
<td>0.78</td>
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<td>0.55</td>
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N-gram TF-IDF and word count features. On the other hand, the SVC classifier with word, n-gram, and character TF-IDF features obtained the highest precision value of 77% on the augmented dataset.

Likewise, the highest recall value of 61% was achieved using the character TF-IDF feature with the SVC classifier on non-augmented datasets, whereas, on augmented datasets, the highest recall value of 78% was obtained by the same classifier with word, n-gram, and character TF-IDF features. Additionally, the highest accuracy was attained using the SVC classifier with n-gram and character TF-IDF features of 71% on the non-augmented dataset, and the same classifier with word, n-gram, and character TF-IDF features of 78% on the augmented dataset.

V. DISCUSSION

The main goal of this study was to create a benchmark dataset of tweets related to popular topics in Arabic social media. The dataset includes Relevance tweets in Arabic for both binary and categorical classifications. Based on the experimental outcomes, the manually annotated dataset can be utilized as a baseline for future research on Relevance Classification of Trending Topics in Arabic Tweets. As no benchmark dataset exists for classifying Arabic trending topic Relevance tweets, this dataset will prove beneficial to the research community once it becomes publicly accessible.

According to the statistical analysis, the non-augmented dataset had lower values for macro F1, accuracy, recall, and precision in its classification compared to the augmented dataset, which exhibited better results. Based on the findings of the previous section, it can be concluded that the use of data augmentation techniques has improved the classification results, leading to the highest macro F1 score of 82% in binary classification and 77% in categorical classification. The results achieved in the binary classification outperform the work in [6]. This work is closest to our work as it also aimed to classify relevant tweets on Indonesian trending topics, and they achieved an F1 measure of 70%.

Machine learning techniques that employ N-gram features provide better results in classifying Relevance tweets within trending topic datasets than other features. Moreover, binary classification achieved superior results compared to categorical classification. Learning in categorical classification is comparatively less accurate than binary classification, as it is a more complex process.

VI. CONCLUSION AND FUTURE WORK

This paper presents a novel Arabic dataset consisting of Relevance tweets for binary and categorical classifications, which will be available for public research. The process of tweet collection, manual labeling, and data augmentation for the dataset is described in details. We employed three classifiers, namely Naive Bayes, Logistic Regression, and Support Vector Machine, for Relevance Classification of Trending Topics in Arabic Tweets. The classifiers were trained using four types of features: count vector and TF-IDF (word-level, n-gram-level, and character-level). The study found that the performance varied depending on the classifiers and features used and that higher performance could be achieved with...
more annotated data. The Support Vector Machine approach was found to perform well for Relevance classification of Twitter content, with average macro F1 scores of 82% and 77% obtained in the binary and categorical datasets, respectively.

In future work, it would be valuable to explore the effectiveness of advanced deep learning techniques like convolutional neural networks, recurrent neural networks, and BERT for Relevance Classification of Trending Topics in Arabic Tweets. Moreover, extending the dataset to include a wider range of topics and domains and evaluating the generalizability of the proposed classification models across diverse datasets would be of interest.

REFERENCES


Generative Adversarial Network-based Approach for Automated Generation of Adversarial Attacks
Against a Deep-Learning based XSS Attack Detection Model

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Abstract—Cross Site Scripting attack (XSS) is one of the most famous and dangerous web attacks. In XSS attacks, illegitimate technical methods are used by attackers to disclose sensitive data from web site users, which result in an important financial and reputation loss to the web site’s owner. There exist numerous XSS attack countermeasures. Deep Learning has been shown to be effective when used to detect XSS attacks in HTTP web requests. Yet, Deep Learning models are inherently vulnerable to adversarial attacks, which aim to deceive the detection model into mis-classifying malicious HTTP web requests. Thus, it is important to evaluate the robustness of the detection model against adversarial attacks before its deployment to production in real web applications. In this work, we developed a Generative Adversarial Network (GAN) model for automated generation of adversarial XSS attacks against an LSTM-based XSS attack detection model. We showed that the detection model performance drops drastically when evaluated on the XSS instances, originally used in the model development, but modified by the GAN model. We also provided some guidelines to the development of detection models that can defend against adversarial attacks in the particular context of web attacks detection.

Keywords—Deep learning; generative adversarial network; LSTM; web attacks; adversarial attacks; Cross Site Scripting attack

I. INTRODUCTION

Organizations and enterprises are more concerned now than never before about the mitigation of cyber-attacks. Indeed, a successful cyber-attack can cause the enterprise an important financial loss and a reputation damage. Cross Site Scripting Attack is one of the most serious cyber-attacks that target web sites to compromise the confidentiality of the user’s data. We distinguish three types of XSS attacks: Stored XSS attacks, Reflected XSS attacks and DOM(Document Object Model) XSS attacks, which consist of executing a malicious code that was injected into the database server, the web server response, or the DOM, respectively. Different methods are used to protect web applications from XSS attacks. However, XSS attacks are still ranked in the top 10 web vulnerabilities since 2017 [13]. Recently, more research works have been devoted to the use of Deep Learning to build Web Applications Firewalls. The results of these works are overall promising (e.g. [2], [4], [5], [6], [7], [8], [9]). Yet, few works have analyzed whether the proposed model is vulnerable to adversarial attacks. Indeed, Deep Learning models can be victim to adversarial attacks which may result in a significant drop in their classification performance. In the case of XSS attacks detection, adversarial attacks attempt to deceive the model into mis-classifying HTTP web requests, by generating HTTP web requests that resemble to normal HTTP web requests, but are in reality malicious HTTP web requests. In this paper, we proposed a Generative Adversarial Network (GAN) model for an automated generation of adversarial attacks against an LSTM-based XSS attacks detection model. The main contributions of our work are two fold:

- We conducted experiments to demonstrate the negative impact of adversarial attacks on the classification performance of an XSS attacks detection model that returned good results upon its evaluation on a public dataset.
- We proposed some guidelines to how to optimize the detection model to defend against adversarial attacks in the particular context of web attacks detection.

To this end, we followed the steps below:

1) We developed an LSTM model to detect XSS attacks. For that, we used a dataset A, and we recorded the classification results of the model on XSS instances.
2) We developed a Feed Forward Neural Network (FFNN)-based GAN model to generate adversarial XSS examples. For that, we used the dataset A and a dataset B.
3) We passed the XSS instances of the dataset A to the trained GAN model, and we obtained a set of XSS samples that we call craftedXSS.
4) We passed craftedXSS to the trained detection model, and we recorded the difference in the results obtained in this step and in the first step.

The remainder of the paper is divided into seven sections: Section II presents research works related to both XSS attack detection based on Deep Learning and, adversarial attacks against DL-based XSS attack detection models. Section III explains the basic concepts behind the present work. Section IV describes the development process of the XSS detection model and the adversarial attack model. Section V reports and discusses the experimental settings and results. Section VI provide some guidelines to the development of XSS attack detection
models that have a good defense against adversarial attacks. Section VII review the contributions and the limitations of the present work and discusses potential future works.

II. RELATED WORK

In this section, we will review research works related to DL-based XSS attack detection models as well as adversarial attacks models on DL-based XSS attack detection models.

A. XSS Attacks Detection Models

We find few research works about the detection of XSS attacks using Deep Learning models [1]. We cite the following papers [2], [3], [4], [5], [6], [7], [8], [9], and [10], which all use common Deep Learning models like GRU or LSTM based encoder-decoder, CNN, LSTM, DFFN, stacked generalization ensemble model, along with classical vectorization techniques, like word2vec, glove, fasttext, and n-gram, for the conversion of HTTP web requests to numerical vectors. According to the experimental results reported in the cited papers, the models achieve excellent classification results. Yet, the detection of XSS attacks is still challenging, which rises the question of what are the reasons behind the fine or zero utilization of the models proposed by the scientific community in real web sites and applications.

B. Adversarial Attacks Models

Independently of their application domain, Deep Learning models are known to be vulnerable to adversarial attacks. In the following, we present the research papers that have studied adversarial attacks on DL-based XSS attack detection models. [11] used greedy search to find the minimum number of transformations needed for the detection model to misclassify URLs. They showed that adversarial training can improve the model robustness by 7% while adversarial attacks undermine the model classification performance by 56%. [12] used reinforcement learning to build HTTP web requests that evade an XSS attacks detection model based on a pre-defined set of escaping rules. They also leveraged the adversarial attack model to improve the detection model defense capability. Indeed, after adversarial training, the detection model was able to detect 91.75% of XSS examples and miss out only 8%. [14] used a technique called Soft Actor-Critic (SAC) reinforcement learning algorithm to select the most appropriate strategies to build HTTP web requests that escape the XSS attack detection model. They showed that the adversarial model achieves an escape rate of more than 92%. [15] proposed an adversarial attack model based on a reinforcement learning algorithm (Soft Q-learning) to evade different XSS attack detection models. They experimentally showed that the proposed adversarial model bypassed the detection model in 85% of cases. While all existing research works propose a manual approach for creating adversarial attacks against XSS detection models, the present work use GANs model to automate this process. However, the semantic of generated XSS attacks is guaranteed in the manual approach more than in the automatic approach.

III. BACKGROUND

A. Problem Definition

Deep Learning based XSS attack detection models can be used as part of Web applications Firewalls to protect web sites and applications from XSS attacks. Yet, because Deep Learning models are victim to adversarial attacks, the XSS attack detection model can yield a high false negative rate as it classifies malicious HTTP requests as normal requests. Before we dive into the approach we propose to demonstrate how adversarial attacks can deceive an efficient XSS attack detection model into mis-classifying XSS attacks as normal HTTP web requests, we will hover, in this section, the basic notions underlying our proposed approach.

B. XSS Attacks

In XSS attacks, the attacker injects malicious scripts in a way that allows him to steal sensitive information from users when they visit a web page or click on a link that includes the malicious code. There are three types of XSS attacks:

- Stored XSS attack: is triggered when the user visits a web page that includes a malicious code that was previously stored in the server database because of the lack of user input validation.
- Reflected XSS attack: occurs when the user clicks on a web link that contains malicious code, and the server sends back this code to the client in an HTTP response.
- DOM XSS attack: it does not involve the server, and it is completely handled by the Document Object Model to attack users.

C. Adversarial Attacks

An adversarial attack is an attack on data with the aim to deceive an already trained model or in training model. It introduces a subtle modification to the data so that a human eye could not notice the difference between the original and the noisy data, but causes the attacked model to output a wrong classification of the input data. Adversarial attacks can be classified into three main categories:

- Evasion: try to evade trained models by altering samples (e.g. HTTP web requests) so that the target model returns the wrong classification.
- Data Poisoning: attempt to contaminate training dataset such that the learning process of the model is undesirably impacted. For instance, after the training phase, the model would learn the wrong features that characterize for example malicious HTTP web requests. As a result, it would classify as normal HTTP web requests what it should be classified as malicious.
- Model extraction: aim at extracting the maximum of information about the model properties in order to rebuild the model and use it for personal use or as part of an adversarial attacks model.

Attackers usually target white-box or black-box threat model when they run an adversarial attack. In white-box model, the adversary knows the model’s parameters and can get the classification of input data. In black-box model, the attacker can also acquire labels for input data but he does not know the model’s parameters and structures.
D. Generative Adversarial Networks

Generative Adversarial Networks (GANs) are Deep Learning based generative models. Unlike discriminative models, which assign a class label to each input data, generative models objective is to create data that maintains the statistical input data distribution.

GAN was first described by [16] and formalized by [17] in a standardized approach called Deep Convolutional Generative Adversarial Networks (DCGAN). Most GANs that are proposed nowadays are based on DCGANs. GANs are basically composed of two sub-models (Fig. 1):

- Generator model: generates new samples from the problem domain.
- Discriminator model: classifies the generated samples as fake or real.

The generator and discriminator models are, in general, trained separately. The discriminator is trained until it reaches a high classification performance, and then the generator is trained until the discriminator classifies as real a high percentage of fake samples. After the training process, the discriminator model is discarded and the generator model is kept. In the context of adversarial attacks, the GAN model is usually composed of:

- The generator model that can be any Deep Learning based model.
- The discriminator model that is the attacked model.

In this case, the generator model is trained with the purpose of generating adversarial samples that can evade the discriminator model (or the attacked model), which results in a mis-classification of potential XSS attacks.

E. Deep Learning and LSTM Neural Network

Deep Learning is a subfield of machine learning that makes predictions on data based on deep features extraction. Long Short Term Memory or LSTM is a well known neural network especially used in sequence prediction problems. It is a variety of recurrent neural networks (RNNs) that are capable of learning long-term dependencies. It was proposed by [18] to resolve the vanishing and exploding gradients problem inherent to the use of RNNs. Fig. 2 describes the basic functioning of LSTMs. They consist of a four types of gates: the forget gate (\(f_t\)), the input gate (\(i_t\)), the cell gate (\(g_t\)) and the output gate (\(o_t\)). LSTM uses these cell gates and associated activation functions, to decide what to retain and what to forget. The following equations describe the calculations performed at each gate, where \(W_f, W_i, W_g, W_o\) are the weight matrices for the forget gate, the input gate, the cell gate, and the output gate, respectively while \(b_f, b_i, b_g, b_o\) are the corresponding bias:

\[
\begin{align*}
    f_t &= \sigma(W_f \cdot [h_{t-1}, x_t] + b_f) \quad (1) \\
    i_t &= \sigma(W_i \cdot [h_{t-1}, x_t] + b_i) \quad (2) \\
    g_t &= \tanh(W_g \cdot [h_{t-1}, x_t] + b_g) \quad (3) \\
    o_t &= \sigma(W_o \cdot [h_{t-1}, x_t] + b_o) \quad (4) \\
    c_t &= f_t \cdot c_{t-1} + i_t \cdot g_t \quad (5) \\
    h_t &= o_t \cdot \tanh(c_t) \quad (6)
\end{align*}
\]

F. Word Embedding: Word2vec

Word embedding is the technique that allows for the application of deep learning models to textual classification problems, as it transforms words to fixed-size numerical vectors. Word2vec is one of the most commonly used word embeddings techniques. CBOW and skip-gram are the main implementations of word2vec. They consist of a Feed Forward Neural Network composed of an input layer, a hidden layer, and an output layer. The CBOW model predicts the contextual words given the main word, while the skip-gram model predicts the target word given the surrounding words.

IV. PROPOSED METHOD

A. Overview

Fig. 3 presents an overview of the proposed model architecture. Overall, The GAN model takes as input XSS attacks instances - that the XSS detection model would classify as such if they were not altered by the GAN model-, and then creates a corresponding XSS attack instances, that the detection model could not recognize as such, resulting in a mis-classification of XSS attacks as normal HTTP web requests. In Section V-E, we provide some concrete examples of XSS attacks generated by the GAN model.
B. XSS Attacks Detection Model

We developed an LSTM neural network to classify HTTP web requests into normal and XSS attacks. It was trained using a dataset A that contains XSS attacks and normal HTTP web requests. We used Word2vec to transform the HTTP web requests into numerical vectors. The choice of LSTM is justified by the fact that the majority of research works about adversarial attacks against Deep Learning based XSS attack detection model, have also selected the LSTM model, which leave a possible space for comparison. Fig. 4 describes the proposed XSS attack detection model.

C. Adversarial Attacks Model

Fig. 5 show that the adversarial attacks model is a Feed Forward Neural Network that takes as input HTTP web requests, and then creates a modified version of the same HTTP web requests, which are then passed to the detection model. The adversarial attacks model is constantly updated until the detection model can barely recognize XSS attacks. The adversarial attacks model is developed based on a dataset B that contains XSS attacks from the dataset A and another dataset C that contains XSS instances only.

V. EXPERIMENTS AND DISCUSSIONS

In this section, we present and discuss the settings and results of the experiments conducted in the present work.

A. Dataset

For the development of the LSTM-based XSS detection model, we used xssed, a public dataset available on GitHub [12], built specifically for XSS attacks. It contains 33428 XSS attacks and 31428 normal HTTP web requests. We splitted the dataset into three balanced sets; the training set, the validation set and the test set, which we used to train the model, tune the model hyper-parameters, and test the model, respectively. As for the adversarial attack model, we collected XSS attacks from the xssed [12] dataset and we added the XSS attacks listed in the online XSS cheat sheet available at [19]. We obtained a dataset of 41542 XSS attacks examples. We splitted the dataset into two sets; the training and the validation sets which are used at the training phase. As for the testing set, it is composed of the XSS attacks contained in the xssed dataset (33427 example), and we used it to evaluate the detection model capability to identify XSS attacks before and after the modification introduced by the GAN model.

B. Performance Indicators

We used traditional performance metrics, namely accuracy, recall, precision, F1-score, AUC, False Positive Rate (FPR), and False Negative Rate (TNR), to evaluate the classification performance of the XSS attack detection model. As regards the adversarial attacks model, we referred to the following indicators to assess its performance:

- DR (detection rate): DR is the ratio of the number of XSS instances that are still classified as XSS examples by the XSS attack detection model to the total examples of XSS attacks.
- ER (escape rate): ER is the ratio of the number of XSS instances classified as normal by the XSS attack detection model after being modified by the adversarial attack model to the total examples of XSS attacks.

C. XSS Attacks Detection Model Results

Table I reports the performance results of the XSS attack detection model. The results indicate that the LSTM-based XSS attacks detection model achieves a high detection accuracy with a low false negative rate (0.002).

D. Adversarial Attacks Model Results

Table II shows that the GAN-based adversarial attack model escape the detection model in 100% of cases (or the detection rate of the detection model is 0%). Because the adversarial attack model uses a large vocabulary, the generated XSS attacks contain a vocabulary that is unknown to the
detected model as well as the vocabulary that appeared in normal HTTP web requests. Also, the detection model does not account for the semantic of the HTTP web requests. These two factors maximize the escape rate because generated XSS attacks are judged normal HTTP web requests by the detection model. However, this does not mean that the generated XSS attacks if executed will be successful, it will depend on whether they are semantically correct.

E. Examples of Generated XSS Attacks

In this section, we show some examples of XSS attacks generated by the GAN model. Listing 1 shows the original XSS instances, while Listing 2 shows the modified XSS instances. We can clearly see that the generated XSS example widely differ from the original one although it is composed of a vocabulary that is recognized by the detection model. Also, the generated XSS example is not semantically correct as it includes invalid HTML tags. Moreover, the generated XSS example contains some suspicious keywords such as “alert”, or “fromcharcode” that appear only in malicious HTTP web requests. Although the human-eye can easily classify the generated instance as malicious, the detection model failed to output the correct classification label.

Listing 1: original XSS instances
form.search_text=Dell%22%E%3Cscript%3Eaalert(/xss-Bulgaria&lt;br&gt;alert%3C/script%3E)&amp;form.hardware_category=LAPTOP

Listing 2: generated XSS instances
<section,ie_querytext=,&lt;marque,&lt;q&gt;&lt;tr&gt;&lt;applet,1,&lt;p&gt;&lt;code,querycontext=,&lt;keygen&gt;&lt;header,&lt;noembed,script=&lt;charcode,section,0aalert,/&lt;tr&gt;&lt;script,0t,/&lt;br

VI. Guidelines

Adversarial attacks constitute a real danger to the reliability of Deep Learning models. In particular, if the Deep Learning model is used to secure a web application, adversarial attacks can increase the false negative rate which results in a low detection rate of XSS attacks. Based on the outcome of this work, we advise the following guidelines to the development of a detection model that has a good defense against adversarial attacks:

- The HTTP web request should be decoded into its original form. Indeed, web attackers obfuscate HTTP web requests by using different encoding techniques.
- The detection model should be trained on a large dataset that include the maximum number of HTML and JavaScript tags.
- The detection model should not restrict its classification decision to the lexical words that compose the HTTP web request, but also include the semantic validity of HTTP web requests.
- The detection model should discard any HTTP web request that exceeds a certain threshold of unknown words.
- It is important to include adversarial learning in the development process of the detection model in order to optimize its defense against adversarial attacks.

VII. Conclusion, Limitations and Future Work

In this work, we developed a GAN-based adversarial attacks model to generate adversarial examples that can bypass an LSTM-based XSS attack detection model. The results of our experiments show that the detection model performance drops drastically when comes to the classification of adversarial XSS examples. Indeed, while the detection model classified correctly 32904 out of 33427 XSS attacks, it could not classify correctly 32904 out of 33427 XSS attacks, it could not classify any of the corresponding adversarial examples. Moreover, we provided some guidelines to optimize the defense of XSS attack detection models against adversarial attacks. The present work presents the following limitations:

- The adversarial model can not guarantee the semantic validity of the generated adversarial XSS examples.
- Although the proposed adversarial attacks model is applicable to other Deep Learning models, it was applied to only LSTM models.

As future work, we are going to improve the adversarial attacks model in order to generate XSS attacks that are semantically correct.

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A Review on Machine-Learning and Nature-Inspired Algorithms for Genome Assembly

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Abstract—Genome assembly plays a crucial role in the field of bioinformatics, as current sequencing technologies are unable to sequence an entire genome at once where the need for fragmenting into short sequences and reassembling them. The genomes often contain repetitive sequences and duplicated regions, which can lead to ambiguities during assembly. Thus, the process of reconstructing a complete genome from a set of reads necessitates the use of efficient assembly programs. Over time, as genome sequencing technology has advanced, the methods for genome assembly have also evolved, resulting in the utilization of various genome assemblers. Many artificial intelligence techniques such as machine learning and nature-inspired algorithms have been applied in genome assembly in recent years. These technologies have the potential to significantly enhance the accuracy of genome assembly, leading to functionally correct genome reconstructions. This review paper aims to provide an overview of the genome assembly, highlighting the significance of different methods used in machine learning techniques and nature-inspiring algorithms in achieving accurate and efficient genome assembly. By examining the advancements and possibilities brought about by different machine learning and metaheuristics approaches, this review paper offers insights into the future directions of genome assembly.

Keywords—Artificial intelligence; genome assembly; machine learning; bioinformatics; bio-inspired algorithms

I. INTRODUCTION

DNA, short for Deoxyribonucleic acid, is a macromolecule that contains the genetic instructions for the development, functioning, and reproduction of all living organisms. It consists of two strands that form a double helix structure [1]. The four nucleotides—adenine (A), cytosine (C), guanine (G), and thymine (T)—make up the building blocks of DNA. The sequence of these nucleotides determines the genetic code. DNA carries the hereditary information passed from parents to offspring, containing genes that encode proteins and regulate various biological processes. DNA replication ensures that each new cell receives a complete copy of the genetic material during cell division.

Advances in DNA sequencing technologies have revolutionized biological research, enabling the analysis of DNA sequences on a large scale. DNA sequencing helps unravel the genetic code, identify mutations, study genetic variations, trace evolutionary relationships, and diagnose genetic diseases. DNA plays a central role in fields such as genomics, evolutionary biology, genetic engineering, forensic science, and medicine. It serves as a foundation for understanding the complexities of life, exploring the diversity of species, and developing innovative approaches [2] for disease treatment and prevention.

To determine the DNA material code within specific living organisms, the DNA sequencing is utilized for the identification and characterization of genes within an organism’s genome. By determining the sequence of genes, researchers can analyze their functions, regulatory elements, and evolutionary history. Genome assembly is the computational process that comes after sequencing and aims to align reads of a DNA sequence into the correct order to reconstruct the original structure of the genome [3].

The novelty of this review stands out for its in-depth analysis of the most important advancements and challenges involved in genome assembly. By achieving a synthesis of findings from multiple research studies, the review presents a comprehensive survey of the current state of the machine-learning and nature-inspired genome assemblers, while also evaluating the recent bio-inspired techniques and providing recommendations for future research. This gives researchers insightful information and direction about the latest techniques and approaches used in this rapidly growing field. Thus, this study remains as an exceptional and valuable contribution to the existing literature.

The work in this paper is organized as follows: Section II presents an overview about the genome assembly. Section III describe machine-learning techniques used in DNA sequence assembly. Section IV introduce de novo assemblers. Section V review nature-inspired algorithms and metaheuristics solving the genome assembly problem and their best computational results. Section VI provide a discussion and analysis of the findings, proceeded by a conclusion in the last section.

II. GENOME ASSEMBLY OVERVIEW

The DNA genome assembly is one of many bioinformatics problems that uses machine learning technologies and nature-inspired approaches and metaheuristics in aim to be solved. When sequencing the genomes of bacteria, viruses, or humans, this problem is extremely important; it occurs during the final stage of DNA reading, particularly for long strands of DNA. Large strands are repeatedly broken into several little fragments. After that a computer program will assemble together the fragments into a string that matches the original DNA sequence. Finding an organism’s DNA sequence is helpful for both applied and fundamental study into how and why they live. Given the significance of DNA to living organisms, understanding a DNA sequence could be helpful in almost any biological study. For instance, it can be applied in medicine to locate, identify, and potentially develop cures for and genetic diseases. Similar to how pathogen research may result in medicines for infectious diseases and virus transmissions. Due to the rapid advancements in sequencing technologies and the increasing demand for sequencing services, various sequencing platforms have become extensively utilized in recent years. As
a consequence, sequence assembly has emerged as a critical process with diverse applications in the field of bioinformatics [4]. Consequently, the significance of efficient DNA assemblers has escalated, as they play a pivotal role in reconstructing complete genomes from fragmented sequencing data.

III. MACHINE LEARNING TECHNIQUES

McCarthy et al. suggested in the summer of 1956 in a conference at Dartmouth College, that computers could be programmed to think and reason. They referred to this idea as artificial intelligence (AI) [5] a field used to the simulation of human intelligence processes by computer systems. Machine learning (ML) is a specific method to achieve this goal which include deep learning (DL) and artificial neural networks (ANN) methods [6].

Machine learning is used in DNA sequence assembly for pre-grouping reads into similar groups before the assembly process is carried out and it was proven that is an alternate method for lowering the overall computational complexity of the genome assembly process. In [7] the authors suggested building a recurrent neural network (RNN), where the goal is to train the network to track the sequence of bases that constitute a given fragment and assign all of the sequences that are properly tracked by this network to the same cluster [7]. This machine learning method applies a three-layer RMLP (Recurrent Multi Layer Perceptron) neural network with five input nodes dedicated to the five possible incoming symbols (A, C, G, T, and N) in the first layer, a hidden layer of 27 nodes, and four output nodes for the output layer in the forecast of the next base of the sequence created by the network [7]. The experimental results obtained after comparing the proposed neural network method with another conventional algorithm have shown that if performed on multiprocessor machines, the proposed procedure may prove to be less expensive than ones that are currently used. Any assembly method may be employed in place of the CAP approach, which greatly improves its efficiency and yields superior outcomes. While employing various definitions of distance, the two approaches produce collections of fragments with more or less comparable properties. The authors also indicated [7] that their proposition is richer due to the fact that is based on the internal structure of the strings rather than just topological similarities.

The author [8] also proposed the construction of an artificial neural network based binning of reads to assist assembly process. After producing the required reads for assembly using a simulated shotgun sequencing technique. Four assembly techniques were then simulated: The greedy assembler, the de Bruijn assembler, the greedy neural network assembler, and the de Bruijn neural network assembler. In order to determine which of these assembly procedures provided the maximum coverage accuracy and with what level of computational complexity, simulations were performed. The research also looks into the advantages of combining the greedy and de Bruijn assembly algorithms with a “divide and conquer” strategy the greedy assembler’s computational efficiency could be significantly improved. The author in [8] proposed in future work to investigate strategies of reducing the complexity associated with the training and grouping process and to take advantage of the parallelisable nature of the neural network grouping scheme.

The study in the paper [9] offers two techniques for separating sequencing method errors from natural variance. The first is an analytical technique for choosing appropriate error candidates. In order to identify the bases that are uncommon within grouped clusters of closely related sequences, it applies similarity weights between pairs of sequences. The chosen candidates are then filtered using a classification model built using a suitable ML technique and rests on frequency vectors [9]. Both the usage of weights and the ML-based regrouping were proven to greatly reduce the set of potential errors, without missing simulated faults along the way for artificial neural networks and the Hoeffding tree classifier. The performance of the RIPPER rule learner and random forest classifier, however, drastically declined. On hexaploid wheat, the ML-based models showed raw accuracy high enough to imply that they might be utilized independently of the analytical approach for error discovery. According to the authors it was demonstrated that applying the ML model as an additional filter while calling variants significantly changes the results. Further validation were needed to validate the results achieved on hexaploid wheat.

IV. DE-NOVO ASSEMBLERS

Three major strategies are widely used for the novo assemblers in Bioinformatics to solve the DNA Assembly Problem: greedy graph-based algorithms, de Bruijn graphs, and the overlap-layout-consensus (OLC) approach mainly adopted by the nature-inspired algorithms presented in Section V.

De novo gene assemblers have no reference genome for assembling DNA sequences and they are classified into two types: Greedy-based De-novo assemblers Graph-method assemblers: String and DeBruijn. Several graph based common algorithms were used for genome sequence assembly CAP3 [10], PHRAP [11], TIGR [12].

Authors in [13] suggested a novel DNA sequence assembly approach that combines the advantages of shotgun and sequencing by hybridization (SBH). The technique makes use of the high coverage and low error rates in sequencing made available by the development of efficient DNA sequencing machines. The authors were proceeding in the development of full software with all the characteristics stated in [13]. They created a prototype that incorporates some of the algorithm’s fundamental components, and utilized this prototype to put together synthetic sequencing data. They presented results of such an experiment, mostly as a demonstration of concept for their methodology and based on their preliminary investigations [13], the algorithm promises to be very fast and practical for DNA sequence assembly.

A novel EULER algorithm have been designed in [14] and for the first time fixes the repeat problem in fragment assembly. The primacy goal of the authors was the fragment assembly’s reduction to a variant of the standard Eulerian path problem, which makes it possible to produce precise answers to complex sequencing issues. In contrast to the CELERA assembler [15], EULER utilizes such repeats as a strong fragment assembly tool rather than masking them. Based on the de Bruijn graph concept. In order to explain their method, they adopted the DNA sequence as a thread with repeated sections attached together by glue. Every repeat in the resulting de Bruijn graph
having five edges [14], corresponds to one edge rather than a group of vertices in the layout graph.

In [16] the authors presented the ARACHNE computer system, which uses paired-end whole-genome shotgun reads to assemble genome sequences. The key characteristics of ARACHNE include an effective and sensitive method for detecting read overlaps, a method for scoring overlaps that achieves high accuracy by fixing errors prior to assembly, read merger using forward-reverse links, and identifying repeat contigs by forward-reverse link inconsistency.

ARACHNE begins by identifying and aligning overlaps, or pairs of reads that appear to be overlapping. In later rounds, some of these false overlaps caused by repeated sequences in the genome will be removed [16]. An effective overlap detection is achieved. The program utilizes a sort and extend technique that scales about linearly rather than comparing every pair of reads. This method entails creating a sorted table of each k-letter (k-mer) and its source so that related k-mers appear successively. The algorithm next eliminates highly frequent k-mers, which often correlate to highcopy, high-fidelity repetitive sequences in the genome, in order to improve the effectiveness of the overlap detection procedure. After identifying all read pairs that share one or more overlapping k-mer, the algorithm applies a three-step procedure to effectively align the reads. First, overlapping shared k-mers are combined. Next, shared k-mers are extended to alignments. Finally, dynamic programming is used to enhance the alignments. In a similar manner, ARACHNE corrects random insertions and deletions caused by apparent sequencing errors [16]. The alignments are modified in accordance with how the reads are adjusted.

Simulated reads that covered many genomes order have been generated to evaluate ARACHNE. These simulated reads’ assemblies produced virtually full coverage of the corresponding genomes, with a few contigs combined into even fewer supercontigs (or scaffolds) [16]. Contig coverage after genome assembly ranged between 97 and 98 percent, with at least 92 percent of the reads being used in every case. For full coverage, the N50 contig length is 350 kb, whereas for half coverage, it is 17 kb. The length of the N50 supercontigs varies significantly within the genomes.

The authors precised that assembly accuracy was good, but it wasn’t perfect due the fact that there were a very small number of additional misassemblies and little errors happened about once every 1 Mb. Assembling the Drosophila genome was quick, requiring only 21 hours on 8.4 Gb of RAM in a single 667 MHz processor.

In this work [17], the authors developed Velvet a novel collection of de Bruijn graph-based sequence assembly methods for very small reads. The main objective of the approach is both remove errors and resolve large number of repeats repeats in the presence of pair read information. The error correction technique initially merges sequences that belong together, and the repetition solver then separates path that share local overlaps. Authors have evaluated Velvet using both simulated and real data [17]. The algorithm has the potential of assembling bacterial genomes with N50 contig lengths of up to 50 kb and simulations on 5-Mb portions of large mammalian genomes with contigs of 3 kb using only relatively small paired simulated reads. The two other short read assemblers, SSAKE [18] and VCAKE [19], were compared to Velvet. The algorithms are different from one another mostly in how they handle errors. By looking for reads in a hash table, SSAKE and VCAKE automatically explore a de Bruijn graph in a step-by-step manner. Velvet is considerably faster and generates larger contigs without misassembly, but it takes a little bit more memory. Furthermore, it has great precision and covers a significant part of the genome. The authors attempted to use EULER [14] and SHARCGS [20], but the tools were unable to handle their data sets. According to authors, this was most likely because the differed expected input, notably in terms of covering depth and read length.

Building on earlier works [21] [14] the authors in [22] developed MULTIBRIDGING a de Bruijn graph based assembly algorithm for shotgun sequencing under the criterion of complete reconstruction, which can achieve very close to the lower bound for repeat statistics of a variety of sequenced genomes, including the GAGE datasets. As results assembling the repeat statistics of hc19, have shown successful reassembling desired with probability 99%.

The La Jolla Assembler (LJA) [23] a fast algorithm with three modules that address the three challenges in assembling HiFi reads: jumboDBG (constructing large de Bruijn graphs), multiplexDBG (using the entire read-length for resolving repeats), and mowerDBG (error-correcting reads), was designed to enable automated assemblies of long, HiFi reads. The Bloom filter [24], sparse de Bruijn graphs [25], disjoint generation [26] and rolling hash [27] were all used in the jumboDBG approach. LJA builds the de Bruijn graph for huge genomes and large k-mer sizes and turns it into a multiplex de Bruijn graph with changing k-mer sizes, reducing the error rate in HiFi reads by three orders of magnitude. The suggested approach not only produces five times fewer misassemblies than state-of-the-art assemblers, but also generates more contiguous assemblies. In the publication, the automated assembly of a human genome which successfully assembled all six chromosomes was used to illustrate the usefulness of LJA.

V. NATURE-INSPIRED ALGORITHMS FOR GENOME ASSEMBLY

Nature-inspired optimization algorithms is defined as a group of algorithms that are inspired by the behavior natural systems, including bio-inspired algorithms, swarm intelligence and evolutionary algorithms. Inspired by animal, insects behaviors, biology and chemical reactions, those algorithms have provided many engineering, medical and bioinformatics solutions such as solving the DNA Fragment Assembly Problem. The genetic assembly is a critical step in any genomic project, it attempts in reconstructing a DNA sequence from a set of a large number of fragments taken obtained by biologists in the laboratory. DNA Fragment Assembly Problem is known to be an NP-hard combinatorial optimization problem, therefore efficient approximate metaheuristics are required to solve such kinds of problems. The purpose of the study presented in this section is to analyze and synthesize the existing nature-inspired optimization algorithms in for genome assembly. Since the genome assembly is a particularly difficult problem in computational biology due to the problem’s NP-hardness, the ideal solution cannot be found. So, it necessitates the
use of metaheuristics and other computational techniques of intermediate complexity. Which their goal is to compare all possible solutions to an optimization problem in order to choose the best (feasible) one. They evaluate prospective solutions and perform a number of operations on them in an effort to find better alternatives in order to accomplish this.

As genome assembly is a combinatorial optimization problem, different nature inspired algorithms and metaheuristics have been proposed in past few decades to solve this problem. The process of these metaheuristics involves the reconstruction of the original genome sequence from a set of fragments (reads) aligned in a correct order by exploring a large solution space. Numbers from 1 to N are assigned to the set of fragments, where N represents the total number of fragments. By reordering this list of numbers using the fitness function, the algorithm aims to find the optimal order that reconstructs the complete genome sequence through a process of iterative optimization.

Generally an incremental solution of a metaheuristic algorithm for DNA fragment assembly is described as follows: The algorithm starts by setting up parameters and creating an initial assembly of DNA fragments. To assess the quality of the assembly, a fitness function is established. In the case of genome assembly the fitness function involves maximizing the fragment’s scores obtained through semi-global alignment of the DNA fragments. The metaheuristic algorithm proceeds with iterations, each aimed at optimizing the assembly step by step. In each iteration, the current assembly is perturbed to explore neighboring solutions in the search space. The algorithm decides whether to accept the newly generated solution or keep the previous one based on a specific acceptance criterion for each iteration. As the algorithm progresses through iterations, the incremental assembly continues to integrate improvements made in previous steps. The process is considered final when the algorithm reaches a near-optimal solution or fulfills the desired quality standards, the stopping criterion is based on predefined conditions, such as a fixed number of iterations or convergence of the fitness function.

Many Swarm Optimization algorithms have evolved the fragment assembly problem: [28], [29], [30], [31] Cuckoo Search algorithm [33] Harmony Search algorithm [34] hybrid crow search algorithm [35], Cat Swarm Optimization [36], etc. A total of 30 publications on nature-inspired optimization algorithms dealing with DNA genome assembly have been reviewed as shown in Table I.

### A. Swarm Intelligence Algorithms

Verma et al. have proposed the DSAPSO method [28] to resolve the DNA sequence assembly problem using Particle Swarm Optimization (PSO) with Shortest Position Value (SPV) rule. To convert the continuous version of PSO to discrete version SPV rule is used in solving the DNA sequence assembly problem which is a discret problem. The proposed methodology outperforms the genetic algorithm (GA) for every DNA data set, according to the results of a comparison of the DSAPSO results with those of the GA.

By maximizing the overlapping-score measurement, a hybrid particle swarm optimization VNS-based local search approach for solving the DNA fragment assembly (DFA) problem is proposed in [31]. To make PSO appropriate for DFA, the particles are encoded using the lowest position value (SPV) rule. During the PSO search process, VNS local search is used to enhance the quality of the globally best solution generated from the PSO algorithm. The results demonstrated that the algorithm can significantly outperform other PSO-based algorithms with different-sized benchmarks in terms of overlap score.

In other research [32], the same authors suggest a novel memetic GSA method called MGSA in order to solve DNA FAP problem. The overlap-layout-consensus model known as MGSA is based on tabu search for population initialization. This algorithm uses an SPV rule to convert continuous position values into job sequences, initializes the population with a tabu search, and then uses simulated annealing with VNS as the local search method to improve the quality of the best global solution produced by the GSA algorithm. These modifications create a balance between exploitation and exploration. The simulation results show that the algorithm MGSA maximizes the overlap score of 19 benchmark instances, however its disadvantage is requiring more processing time than the current techniques. In order to resolve this issue, the authors want to take into account DNA sequence compression, fuzzy entropy and adapting the MGSA strategy to the de-Bruijn-graph (DBG) model in future works in order to decrease the computation time.

Adaptive Particle Swarm Optimization was proposed in [30] and the experimental results of study on the impact of inertia weight and the cognitive and social components for enhancing the PSO efficiency to obtain the optimal fitness score, were presented with the simulation of three methods: The PSO with constant inertia weight (CIW), PSO with dynamically varying inertia weight (DVIW) and APSO.

The paper [37] presents a new particle swarm optimization and differential evolution approach using the SPV rule to convert the continuous variables used in PSO and DE to the permutation required to solve the FAP. The authors have conducted four different experiments. The purpose of Experiment 1 was to evaluate the PPSSO+DE algorithm with those that performed the best on the sixteen typical benchmarks. The Lin-Kernighan method was used in Experiment 2 to tackle each of the sixteen benchmark issues utilizing the TSP approach. The results of the Lin-Kernighan algorithm were used to determine the best solutions in Experiment 3. In Experiment 4, the Staphylococcus aureus COL Main Chromosome test data was used to test the TSP technique.

In this study [38], the authors suggested a new approach to solve the sequence assembly problem using Particle Swarm Optimization (PSO) with Naive crossover and shortest position value (SPV) rule. There are two phases in PSO with Naive Crossover with SPV: The initialization phase, where individuals are initialized, and the update phase, where new solutions are generated and updated. The real coded values are converted to discrete values using SPV rules. According to the authors, the DNA sequence assembly utilizing PSO algorithm with naive crossover (DSAPSONC) has demonstrated the effectiveness in solving the sequence assembly problem.

In order to effectively solve the fragment assembly problem, a new DPSO method that operates directly in the search
space of permutations has been proposed [39]. The Probabilistic Edge Recombination (PER) operator is the main element of the suggested approach. Through the probabilistic recombination of edges connecting edges from current position, the personal best, and the group best, this operator creates an alternate position. In this probabilistic construction, the utilization of overlap lengths between fragments has also been taken into consideration. With this purpose, memetic algorithms with a new fast variant local search of PALS known as quick-PALS were developed to improve the intensification potential. To show the efficiency and potency of the employed algorithms, two sets of validation experiments have been performed. The authors claim that when compared to present literature assembly techniques, the algorithms performed better.

A new PSO variation was suggested in [40] that uses chaos, levy flight, and adaptive parameters to solve the genome sequencing problem which is transformed into a discrete optimization problem while using the SPV rule. The proposed algorithm incorporates chaos in two distinct ways: applying chaotic inertia weight and chaotic initialization. To ensure the balance between exploitation, which is encouraged by a lower inertia weight, and exploration, which is encouraged by a bigger inertia weight, a chaotic inertia weight was applied. The production of particles through using Levy chaos and refinement Flight ensure that the particles are initialized with a high fitness score. The chaotic initialization provides favorable circumstances for discovering better values [40]. For the four datasets studied in the paper, the Chaotic Particle Swarm Optimization with Levy Flight performs better than other PSO variations by 7% to 24%. When compared to other algorithms on the basis of Standard Deviation, the proposed approach does not, however, demonstrate a significant improvement. It has a higher ranking but inconsistent performance. To solve this issue the authors suggest to used alternative PSO algorithm versions in future.

This study [33] presents the Cuckoo Search Algorithm (CS) as a novel optimization approach for genome sequence assembly, inspired by the behavior of cuckoos. CS incorporates levy flight and brood parasitic behavior, mimicking the process of cuckoos laying eggs in host nests. The algorithm is a population-based search procedure widely applicable to complex optimization problems. These birds lay their eggs in the nests of other birds and use various strategies to increase the chances of their eggs hatching. The algorithm models this behavior by representing each solution as an egg and using Levy flight to generate new solutions. The algorithm follows three main rules: - Each cuckoo lays one egg at a time in a random nest. - Only the nests with high-quality eggs are preserved. - Hosts have a probability of discovering alien eggs and can either remove them or abandon their nests. Various algorithm settings are analyzed to determine the most effective configuration, and CS’s efficiency is evaluated against PSO and its variants.

In this study [41], a hybrid cuckoo-search genetic algorithm (CSGA) was suggested as a nature-inspired swarm optimization algorithm. The total assembly time and the number of reorientations during the assembly process are taken into account by the cost criterion for optimization. An example assembly with 19 components has been shown to demonstrate how the CSGA is applied, and the results have been compared with those of the Genetic Algorithm (GA). According to the findings, the CSGA algorithm not only generates optimal assembly sequences for the given problem at costs equivalent to those of GA, but it has also been discovered to have a faster convergence rate than GA.

The study [42] introduces subsequence-based matching techniques using the CS and PSO algorithms. These methods were implemented in Java and utilized MapReduce for Hadoop. The experimental results validate the effectiveness of the proposed techniques, showcasing their ability to achieve extensive DNA fragment coverage and high matching accuracy. Furthermore, the performance analysis reveals that the CS algorithm outperforms the PSO algorithm in terms of overall performance.

B. Simulated Annealing based Algorithms

Simulated annealing is a computing technique that seeks for the optimal solution by using randomness. It’s inspired from a related technique called Annealing in Metallurgy which mimics the physical solid annealing process where a glass material or metal is heated to a high temperature and then allowed to cool. The authors in [43], [44] have introduced and applied methods solving the DNA fragment assembly problem with the addition of the inversion and transposition operators to a simulated annealer by [45] the performance have successfully been increased. These studies generally raise a variety of other important issues, particularly those relating to the significance of solution space redundancy and the synergistic interactions between the various operators [46].

In this paper [47], a parallel models of Simulated Annealing (SA) was proposed combined with Genetic Algorithm (GA) for solving the DNA fragment assembly problem. They employed SA as a local search method within the GA framework. The experimental results demonstrate that the parallel approach improves the quality of solutions while reducing the overall runtime. Comparing the execution times, SA outperforms GA by being faster. However, SA tends to produce worse fitness values compared to GA In this paper [48], Simulated Annealing-based local search have been utilized to improve the final solution obtained by the Chemical Reaction Optimisation (CRO) algorithm in solving the DNA fragment assembly problem. The CRO approach is used in seeking for the best layout where the objective function is minimized. The main process of the algorithm starts with setting valued to the control parameters after the determination of the initial population. Then one of the four collisions of the CRO algorithm is performed in each iteration. After any new minimum fitness value is checked and saved. After that, to retain the diversity of the population the worst 20 percent of the population is replaced with new solutions. When no amelioration has occurred, the simulated annealing method is used in order to enhance the best solution found.

SA maintains at the same temperature for a period of time while a predetermined number of iterations are set. Then, the heat becomes colder. One of the following three operators is chosen for each iteration [48]:

1) Inversion: Two points are chosen at random for this operator [48]. Then, between them, the order of the fragments
is reversed.

2) Specific inversion: One contig’s orientation is reversed. To do this, a permutation point is randomly chosen, and the contig containing this fragment is identified. The fragments in the chosen contig are then rearranged in reverse order [48].

3) Transposition: This operator [48] shifts a contig to a new position between two points in different distinct nearby contigs that are selected for the contig movement.

The experimental results in the paper [48] have shown that combining CRO and SA have lead to the highest overlap scores.

C. Local Search based Algorithms

In this paper [49], the authors have presented PALS (Problem Aware Local Search) a fast and accurate local search algorithm that, after comparing its results with commercially available assemblers Phrap and CAP3, pattern matching algorithms (PMA) and genetic algorithms (GA), it have shown to be competitive against those present specialized assemblers. They also explored the effect of many alternative approaches on the efficacy of the suggested algorithm. The main objective of PALS algorithm is to obtain one single contig by finding a fragment’s order that minimizes the number of contigs, which is different from the other assembling algorithms that aim to search for solutions having in the layout maximum overlap between adjacent fragments. The three main methods of the PALS algorithm are [49]:

- GeneratInitialSolution method: Generates a single solution (successive overlapping fragments) and is continually updated by the application of ordered movements.
- ApplyMovement method: Makes a movement perturbation and alters the subpermutation between to positions i and j.
- CalculateDelta method: Calculates the variation in the overlap and in the number of contigs this method is considered the main step of the PALS algorithm.

In another paper [50] two changes were proposed to the principal PALS: The first goal is to avoid the local optima and premature convergence. In this case, the method for choosing the enhancing perturbation to be applied to the existing solution at each algorithmic step is changed in a way that leads to a significant improvement. The second one is to minimize the computational demands of the algorithm, this change involves applying multiple independent perturbations rather than a single perturbation to enhance the present solution at each algorithmic step.

The authors in [50] have noticed that the optimization of the fitness is not the same as the optimization of the number of contigs which is the present objective although the two goals are complementary. As a result, the search mechanism in PALS includes an estimation of the number of contigs. It selects the movement with the lowest variation of contigs to orient the search towards solutions that improve the number of contigs which is the movement that reduces or maintains the number of contigs.

Authors in [50] have suggested changing the movement selection technique to prevent the premature convergence. With the modified PALS known as PALS2, the movement with the lowest contigs variation have always been selected, but in contrast to the main PALS, in the situation that there are several movements with the same contig variation, the movement with the lowest fitness variation is chosen.

To avoid the significant amount of recalculation required in each step in the process of PALS where only one single movement in every step is applied, the authors have proposed the algorithm PALS2-many [50] where many movements are used in each step by developing a second variant in PALS2 in order to improve the solution.

The paper [51] have presented a discrete whale optimization algorithm (DWOA) modeling the approach taken by humpback whales when looking for victim or prey by employing conventional operators adapted from evolutionary algorithms. The whales assault their victim or prey using a remarkable feeding technique known as the bubble-net approach. They swim up to the surface after constricting loop after spiral around the victim [52]. In order to avoid reducing the variation in the population, the whale positions used to look for the prey were produced randomly from the fragment numbers rather than utilizing a random whale. To show how effective DWOA is in converting continuous whale behaviors to discrete ones, it was compared to various WOA, DE, and SCA methods. The paper have also demonstrated the performance of the DWOA over those algorithms. To enhance the performance of the proposed Discrete Whale Optimization Algorithm (DWOA) in terms of fragment order, a local search technique called PALS2-many was incorporated [51]. This approach, known as DWOA-LS, combines the benefits of both DWOA and local search to optimize the fragment order. By integrating the local search, DWOA-LS not only maximizes the overlap score among the fragments but also minimizes the number of contigs, resulting in improved overall performance.

D. Genetic Algorithms

Genetic algorithm have also been applied in the DNA fragment assembly problem by [46]. The authors investigated various evolutionary algorithm operators for the issue and discovered that using “macro-operators” considerably boosts performance by exploiting fragment construction at gradually higher levels.

Individuals are the population of potential solutions that genetic algorithms operate on [53] [54] [55]. Usually, random people are used to initialize the population. Depending on their relative fitness, individuals are then either removed from the population or reproduced within it. Different operators are applied to the existing population of individuals to create new individuals ones. A generation is a group of people in any consecutive population. Typically, the genetic algorithm [46] [56] processes in the following order:

1) The algorithm creates a pool of solutions at random where random individuals are used to initialize the population.

2) It uses a fitness function for superior solutions selection. The fitness of each individual is evaluated during the selection process. Based on fitness, individuals are reproduced (copied)
in different ways. Various genetic algorithms implement the concept of differential reproduction using various techniques [46]. However Parsons et al. have used generational genetic algorithm where a new population is formed at each generation, totally replacing the prior population. Low fitness individuals have a low likelihood of being copied into the next generation, whereas high fitness individuals have a high likelihood of having several copies in the following generation.

3) To produce next-generation solutions, crossover and mutation processes are applied to successful solutions [56]. The authors have discussed in [46] all the crossover operators where the crossover rate indicates the average percentage of new people created by crossover per generation and defined the crossover as the selection of two individuals from the population, and the swapping of substrings from corresponding regions within the individuals. At the next generation, one or both of the new individuals are incorporated into the population. The operator’s goal is to let incomplete solutions develop on various individuals before combining them to create a better solution.

By modifying a basic component of an individual, a mutation modifies that individual is the definition of mutation in [46]. The chances that any element in an individual will change depends on the mutation rate. The resultant individual takes the place of the mutation’s parent. It is thought by authors that mutation is successful because it both explores the search space close to existing individuals and saves solution components that have been totally excluded from the population by selection for the following generations [46]. Noting that in genetic algorithm the individuals are the fragment orders representing the DNA sequence solutions.

A new Hybrid genetic algorithm (GA+PALS) have been presented in the paper [57], the genetic algorithm have been used with PALS that was utilized as a mutation operator. The authors have compared the hybrid method with the original GA and PALS methods. A very effective assembler that enables the search of optimal solutions for numerous instances of this problem was obtained as a result. Authors have used the conventional recombination and mutation operators, then some solutions with a low probability, are randomly chosen from the existing offspring and enhanced utilizing the local search algorithm in the method’s main loop. This type of hybridization is justified by the fact that, while the GA identifies good positions of the search space, PALS facilitates exploitation in the best regions discovered by its collaborator. Obviously, the goal in this situation is to determine whether we can develop another heuristic from the best of the two (the GA and PALS algorithm) that would outperform either of the two methods from which it was derived.

The authors have introduced two novel algorithms in recent publications: the Recentering-Restarting Genetic Algorithm (RRGA) and the Recentering-Restarting Hybrid Genetic Algorithm (RRHGA), as mentioned in the papers [58] and [59] respectively. The primary advantage emphasized in the paper [58] for RRGA is its ability to avoid local optima by exploring the search space and leveraging specific dynamic representations. Before initiating the algorithm, a center or reference point, representing a potential solution, is selected either through seeding or random selection. Uzma and Halim [59] suggest that starting with a solid solution is preferable, while RRGA refines the potential solutions. Once the center is determined, the population is generated.

In the direct representation approach, each individual in the population is created by applying a sequence of n transpositions to the center of the population. The ordered lists of fragments are modified through evolution, as explained in the work by [59]. The RRGA algorithm incorporates the Power Aware Local Search (PALS) operator as an evolutionary operator. The performance of the proposed algorithm is evaluated based on overlap scores and the quantity of contigs.

According to the authors, the initial arrangement of fragment orders in RRGA is known as the center, which represents the default arrangement of the dataset’s fragments. The center is then optimized using a 2-opt heuristic. This process generates a set of chromosomes, from which the best chromosome is selected based on its fitness value. A comparison is made between the fitness value of the best chromosome and the center. If the fitness value of the best chromosome surpasses that of the center, the number of transpositions is reduced by five percent, and the center is replaced with the best chromosome.

To evaluate their work, the authors conducted three types of experiments. In the first set of experiments, the PALS operator was used as a genetic operator. In the second set, PALS was applied after running the Genetic Algorithm (GA). Finally, in the third set, PALS served as a genetic operator and was also used after the execution of the GA. The experiments were performed with and without force recentre methodologies, and the results were compared with the Recentering-Restarting Genetic Algorithm, PALS, Genetic Algorithm, and Hybrid Genetic Algorithm. The RRHGA approach demonstrated superior performance across all of these methods.

The paper [60] discusses the importance of studying genetic algorithms to address the DNA fragment assembly problem. The efficient GA operators that have proven successful in the TSP and QAP contexts served as the inspiration for the construction of a GA platform to tackle the DNA FAP problem. By identifying commonalities between the three DNA FAP, TSP, and QAP problems these efficient GA operators successfully been identified and integrated in the platform.

By carefully combining various GAs operators of the platform, an effective GA variant was created in this research [61]. In order to do that, various GAs operators have been studied in solving the DNA FAP problem [60]. The performance of these operators in the contexts of the TSP and QAP issues is already established, and this study has the advantage of comparing the GA results with the results of other previous GA studies in the context of the overlap score. The SCX crossover, a smart crossover that has never been utilized with DNA FAP, provided better results than the other crossover types under consideration, which is the study’s most evident and important discovery. The best-designed GA variant outperformed the current GA algorithms at solving the DNA FAP problem and showed a notable improvement in accuracy with good and competitive results. This paper [61] is the first study to solve the DNA FAP problem form this perspective.
E. Ant Colony Algorithms

This paper [62] introduces the application of an ant colony system algorithm for DNA fragment assembly. The proposed approach utilizes an asymmetric ordering representation, where the collective path generated by the ant colony represents the search solution. The study investigates two types of assembly problems: single-contig and multiple-contig problems. The simulation results demonstrate that, for single-contig problems, the ant colony system algorithm performs comparably to a nearest neighbor heuristic algorithm. However, in the case of multiple-contig problems, the ant colony system algorithm surpasses the nearest neighbor heuristic algorithm.

In [63] and [64] the ant colony system (ACS) algorithm was combined with the nearest neighbor heuristic (NNH) algorithm for solving the DNA fragment assembly. The ACS algorithm is utilized to create an optimized ordering sequence for the fragments, while the resulting contigs are assembled using the NNH rule. To evaluate its effectiveness, the ACS+NNH procedure is compared to the standard sequence assembly program CAP3. The results indicate that the overall performance of the combined ACS/NNH technique surpasses that of CAP3. Specifically, when dealing with large problem sizes, the ACS/NNH solutions exhibit higher quality than the CAP3 solutions. It is observed that CAP3 tends to generate a greater number of contigs compared to the ACS+NNH procedure. Thus, the combined ACS/NNH approach demonstrates superior performance and improved contig quality, particularly for larger-scale problems, as opposed to the CAP3 program.

F. Bee Algorithms

The nature-inspired Bee Colony metaheuristic algorithms are population-based search algorithms based on various biological and natural processes observed in the food foraging behaviour of honey bee colonies. In [65] the authors have designed two different Bee algorithms: Artificial Bee Colony (ABC) Algorithm and Queen-bee Evaluation Based On Genetic Algorithm (QEGA).

Artificial bee colony (ABC) algorithm was developed using the notion of the honey bee swarm’s intelligent behavior. To create new effective search algorithms, honey bees use strategies like the waggle dance [65]. Three types of bees that compose the artificial bee colony in the ABC algorithm are workers or employed, onlookers, and scouts. A bee that waits at the dance area to decide which food source to choose, representing one possible solution of a permutation of DNA sequence fragments based on the waggle dance of the employed bee, is referred to as an onlooker. And a bee that moves to the food source that had previously visited is referred to as a worker bee. A scout bee is one that hunts for food at random. The nectar content of a food source represents the DNA assembly problem’s fitness solution.

In the ABC algorithm [65], the employed artificial bees make up the first half of the colony, while the observers make up the second half. When both the employed and onlookers have consumed the employed bee’s food supply, it turns into a scout. A worker or onlooker bee modifies the solution with PALS method for the locating a new food source and evaluates its nectar by calculating the fitness value of the new solution.

In the other category of Bee Colony algorithms, authors [65] designed Queen-bee evolution based on genetic algorithm (QEGA), which was inspired by the queen bee evolution process and has been utilized to improve the optimization capabilities of genetic algorithms in solving the DNA FAP. Genetic algorithms are capable of reaching the global optimum quickly due to the queen-bee evolution, which also reduces the risk of premature convergence. The authors have utilized problem aware local search (PALS) for an effective mutation.

G. Firefly Algorithm

The Firefly Algorithm (FA), a population-based algorithm inspired by the behavior and lighting patterns of fireflies created by Yang [66], is a recent nature-inspired algorithm that has excelled in many number of fields. The firefly have the following attributes according to Yang’s theory:

1) As all fireflies are not gender-specific, they are attracted to each other regardless of their gender orientation.

2) Their brightness is inversely correlated with attractiveness. The less brilliant firefly will therefore travel toward the brighter one for any pair of flashing fireflies. When their distance grows, their attractiveness decrease. Hence, when there is no distance between two fireflies, the attractiveness is equal to the brightness. If none can see a better firefly it will move at random.

3) The environment of the objective function influences or determines a firefly’s radiance.

Authors in the paper [67] have designed Discrete Firefly Algorithm design for Graphics Processing Units (GPU-DFA) and analyzed it behaviour to solve the DNA assembly problem. The main objective of the authors of the paper while designing the algorithm GPU-DFA is to establish an efficient model that runs the main processes of DFA entirely on GPU so the algorithm can support large numbers of fireflies due to optimized data-structures. The initialization and evaluation of each solution are completed one at a time in the paper’s GPU-DFA method. DNA pieces are randomly permuted to produce each firefly i. The firefly I is then assessed, and its brightness (fitness) is determined. In order to compute them, GPU-DFA uses parallel threads. Several consecutive threads can make use of different memory space.

---

E. Ant Colony Algorithms

Table I. Nature-inspired Algorithms in Literature

<table>
<thead>
<tr>
<th>Inspiration</th>
<th>Stand alone algorithm</th>
<th>Hybridization algorithm</th>
<th>Total</th>
</tr>
</thead>
<tbody>
<tr>
<td>Insects</td>
<td>[62] [66]</td>
<td>[65] [67] [63] [64]</td>
<td>6</td>
</tr>
<tr>
<td>Birds</td>
<td>[30] [33] [42] [28]</td>
<td>[31] [32] [37] [41]</td>
<td>12</td>
</tr>
<tr>
<td>Evolutionary</td>
<td>[46] [58] [61]</td>
<td>[57] [59]</td>
<td>5</td>
</tr>
<tr>
<td>Mammals</td>
<td>[36]</td>
<td>[51]</td>
<td>2</td>
</tr>
<tr>
<td>Annealing in metallurgy</td>
<td>[43] [45] [44] [47]</td>
<td>[48]</td>
<td>5</td>
</tr>
<tr>
<td>Total</td>
<td>17</td>
<td>13</td>
<td>30</td>
</tr>
</tbody>
</table>

---

G. Firefly Algorithm

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II. Cat Swarm Optimization

Recently, Yassine et al. presented in [36] the application of the Cat Swarm Optimization algorithm (CSO) in the DNA fragment assembly problem. This metaheuristic is a swarm intelligence algorithm that incorporates the natural behavior of cats and takes inspiration from the characteristics of cats, which are typically lazy creatures that spend a significant amount of time resting in a seeking mode. However, even during their resting periods, they remain aware of their surroundings. When cats sense a target, they switch to a tracing mode and start moving towards it.

In CSO, each cat within the swarm is represented by its position, velocity, and a flag indicating whether it is in seeking mode or tracing mode. The position of a cat corresponds to a potential solution to the problem being optimized. The velocity of the cat influences its movement within the search space. The flag determines the current mode of the cat, indicating whether it is in seeking mode (resting) or tracing mode (actively moving towards a target). The mixing ratio (MR) is a parameter in CSO that determines in which mode the cat will go into.

By simulating the natural behavior of cats and incorporating it into an optimization algorithm, CSO aims to find accurate solutions to the DNA fragment assembly problem by efficiently exploring the search space. The balancing of seeking and tracing modes through the mixing ratio enables the algorithm to adapt its exploration and exploitation strategies based on the problem characteristics and the current state of the swarm.

The Table II synthesizes eight main algorithms hybridized with local search and other methods for solving the DNA fragment assembly problem. The first column of the table presents the dataset instances names provided from [68]: M15421(5), M15421(6) and M15421(7) from the human apolipoprotein B gene. j02459(7) instance from Complete nucleotide sequence of the cohesive ends of bacteriophage lambda DNA. The two instances bx842596(4) and bx842596(7) from Neurospora crassa DNA linkage group II BAC clone B10K17. In the other columns the fitness results obtained by the algorithms are presented: CRO+SA (Chemical Reaction Optimization) [48], GPU-DFA+LS (Discrete Firefly Algorithm design for Graphics Processing Units) [67], CSA-P2M*Fit (Crow Search Algorithm and ALS2-many) [35], DWOA-LS (Discrete Whale Optimization Algorithm PALS2-many) [51], RRHGA (Recentering–Restarting Hybrid Genetic Algorithm) [59], PPSO+DE (Parallel Particle Swarm Optimization and Differential Evolution) [37], QEGA (Queen-bee Evaluation Based On Genetic Algorithm) [65], PER-PSO-(hi)-ls (Probabilistic Edge Recombination Particle Swarm Optimization and quick-PALS) [39].

It’s clear seen from the Table II that the hybrid methods CSA-P2M*Fit, DWOA-LS and PER-PSO-(hi)-ls outperformed in all the instances. CRO+SA give better results in M15421(5), M15421(6), M15421(7) and j02459(7). GPU-DFA+LS showed high fitness values too in M15421(5) and j02459(7).

However, it is important to consider that every metaheuristic algorithm possesses certain parameters that contribute to enhancing the algorithm’s results. In most of the algorithms, the different parameters settings were applied in different test experiments and were varying for each instance of the dataset.

VI. Discussion

The field of genome assembly algorithms has experienced rapid and exponential growth. Over time, there has been a significant increase in the development and advancement of these algorithms. This growth can be attributed to several factors, including the availability of high-throughput sequencing technologies, the decreasing cost of sequencing, and the

**TABLE II. COMPARISON OF THE FITNESS VALUE OF THE STUDIED ALGORITHMS FROM [48], [67], [35], [51],[59], [37], [65], [39] ON DIFFERENT INSTANCES**

<table>
<thead>
<tr>
<th></th>
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<th></th>
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<th></th>
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<th></th>
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<tbody>
<tr>
<td>M15421</td>
<td>38746</td>
<td>38746</td>
<td>38746</td>
<td>38746</td>
<td>22598</td>
<td>38686</td>
<td>38578</td>
<td>38746</td>
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<tr>
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<td>M15421</td>
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<td>48052</td>
<td>48052</td>
<td>29469</td>
<td>47669</td>
<td>47882</td>
<td>48052</td>
</tr>
<tr>
<td>6</td>
<td>M15421</td>
<td>55171</td>
<td>55072</td>
<td>55171</td>
<td>32744</td>
<td>54891</td>
<td>55020</td>
<td>55171</td>
</tr>
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<td>7</td>
<td>J02459</td>
<td>116700</td>
<td>116700</td>
<td>116700</td>
<td>68736</td>
<td>114381</td>
<td>116222</td>
<td>116700</td>
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<tr>
<td>J02459</td>
<td>BX842596</td>
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<td>227233</td>
<td>227920</td>
<td>125711</td>
<td>224797</td>
<td>227252</td>
<td>227920</td>
</tr>
<tr>
<td>4</td>
<td>BX842596</td>
<td>444518</td>
<td>444162</td>
<td>445422</td>
<td>247856</td>
<td>429338</td>
<td>444518</td>
<td>445422</td>
</tr>
</tbody>
</table>
increasing demand for accurate and complete genome assemblies. Genome assembly algorithms play a crucial role in reconstructing the fragmented DNA sequences obtained from sequencing machines into complete genomes. As the complexity and size of genomes vary across different organisms, the development of efficient and accurate assembly algorithms has become essential. Advancements in assembly algorithms have been driven by a combination of algorithmic innovations, computational resources, and improved understanding of the characteristics of DNA sequences. Researchers have developed various algorithmic approaches, including machine learning methods, nature inspired metaheuristics based on overlap-layout-consensus and de Bruijn graph-based approaches, and hybrid methods that combine multiple strategies.

As a result of these combined factors, the field of genome assembly algorithms has experienced remarkable growth, with continuous improvements in scalability, computational efficiency and in assembly quality [69]. This ongoing progress in algorithm development is crucial for advancing genomics research, enabling discoveries, and understanding the complexities of genomes across different species. This review paper provide collaboration and knowledge exchange among researchers, enabling them to overcome the genome assembly challenges through the application of machine learning and nature-inspired optimization algorithms. Machine learning methods and metaheuristic methods are known to be two distinct approaches used in problem-solving domains including genome assembly. Each approach has its strengths and limitations, and a comparison between the two can provide insights into their applicability and effectiveness in different scenarios. Methods provided by machine learning, such as supervised learning, unsupervised learning, and reinforcement learning, utilize algorithms that learn patterns and relationships from data. These methods excel in tasks where large amounts of labeled or unlabeled data are available. In genome assembly, machine learning methods can be employed for various purposes, such as error correction, read alignment, and sequence classification. They can leverage the inherent structure and patterns within the genomic data to make predictions and improve assembly accuracy. However, machine learning methods often require extensive training data and may be computationally intensive, especially for complex problems with high-dimensional data. Metaheuristics in the other side, explore the search space systematically, looking for optimal solutions without relying on explicit problem-specific knowledge. Metaheuristics are well-suited for combinatorial optimization problems, including genome assembly as shown in the results (Table II). They can effectively handle large-scale datasets and non-linear optimization objectives. Metaheuristic algorithms offer a balance between exploration and exploitation, enabling them to escape local optima and find near-optimal solutions. However, they do not provide guarantees of finding the global optimum, and the convergence speed can vary depending on the problem and parameter settings. Researchers can combine these approaches to leverage the strengths of both and to improve genome assembly outcomes. Machine learning models can be used to guide hybrid metaheuristic algorithms combined with parallelism technologies like mapreduce in the search process like in [42] or to extract meaningful features from genomic data, enhancing the effectiveness of the optimization process of genome assembly.

VII. CONCLUSION

In this paper, a comprehensive review of existing literature in the field of genome assembly was established with a particular emphasis on practical algorithms. The reviewed algorithms include OLC (overlap-layout-consensus) based algorithms, de Bruijn graph-based algorithms, swarm algorithms, and machine learning methods. For each algorithm, detailed insights and highlights were provided, outlining their characteristics, strengths, and potential applications. Recent advancements in the literature also were examined, considering how these algorithms have evolved to address the challenges of genome assembly problem. In the future, the suggested DNA fragment assembler could be further developed by leveraging both machine learning techniques and nature-inspired algorithms, having the potential to be adapted into a parallel version using parallel programming frameworks like MapReduce. These enhancements are expected to lead to significantly improved performance, allowing for more efficient results and reduced execution times.

REFERENCES


Predicting Customer Segment Changes to Enhance Customer Retention: A Case Study for Online Retail using Machine Learning

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Abstract—In today’s highly competitive marketplace, advertisers strive to tailor their messages to specific individuals or groups, often overlooking their most significant clients. The Pareto principle, asserting that 80% of sales come from 20% of customers, offers valuable insights, imagine if companies could accurately forecast this vital 20% and recognize its historical significance. Predicting customer lifetime value (CLV) at this juncture becomes crucial in aiding firms to effectively prioritize their efforts. To achieve this, organizations can leverage predictive models and analytical tools to target specific customers with tailored campaigns, enabling well-informed decisions about advertising investments. By being aware of these segment transitions, advertisers can efficiently deploy resources and increase their return on investment. By implementing the strategies outlined in this study, businesses can gain a competitive edge by identifying and retaining their most valuable clients. The potential for growth and client retention is immense when anticipating changes in customer segments and adjusting advertising strategies accordingly. This paper provides a comprehensive methodology, tools, and insights to assist marketers in optimizing their advertising campaigns by anticipating customer lifetime value and actively predicting changes in client segmentation.

Keywords—Customer segment changes; customer retention; marketing actions; informed decisions; advertising strategies

I. INTRODUCTION

In today’s highly competitive marketplace, advertisers face the ongoing challenge of delivering targeted and personalized advertisements to capture the attention of potential customers. However, amidst the quest for broad reach and mass appeal, businesses often overlook a crucial aspect – focusing on their most valuable customers. The Pareto principle [1], a well-known economic principle, sheds light on this phenomenon by revealing that a small portion of customers typically contributes a significant proportion of sales. While the Pareto principle has long been cited in reference to sales patterns, its implications for advertising strategies have been underexplored. What if businesses could not only identify this 20% historically but also predict it for the future? The concept of predicting customer lifetime value (CLV) arises as a powerful tool in this context. By leveraging predictive modeling and analytical techniques, businesses can forecast future customer behavior and identify those individuals who will likely make up the high-value customer segment. The primary objective of this paper is to propose an approach for predicting customer segment changes based on CLV predictions. By accurately predicting shifts in customer segments, businesses can strategically prioritize their marketing actions and allocate resources more efficiently. This includes determining the optimal investment in advertising, identifying the specific customers to target with tailored campaigns, and devising strategies to transition customers from one segment to another. Understanding and leveraging customer segment changes presents several strategic advantages for businesses. Firstly, it allows for the efficient allocation of advertising resources [2], ensuring that marketing efforts are focused on the most valuable customers who are likely to drive significant sales [3]. Secondly, by tailoring advertising messages and offers to this high-value segment, businesses can improve customer engagement and conversion rates[4]. Lastly, actively managing customer transitions between segments enables businesses to nurture relationships, increase customer loyalty [5], and maximize long-term customer value. This paper serves as a comprehensive guide for businesses seeking to optimize their advertising strategies by harnessing the power of CLV predictions and proactively targeting customer segments. Through an exploration of predictive modeling techniques and actionable insights derived from customer segment changes, businesses can gain a competitive edge in today’s dynamic marketplace. By aligning their advertising efforts with anticipated shifts in customer segments, businesses can enhance customer retention, maximize profitability, and foster sustainable growth. This paper is organized as follows: In Section II, we provide an overview of the existing literature. Next, in Section III, we present a comprehensive framework that outlines the step-by-step process of predicting customer segment changes based on customer lifetime value. Moving on to Section IV, we present the empirical study conducted to evaluate the effectiveness of our approach. We describe the dataset used, the experimental setup, and the evaluation metrics employed. We then present and analyze the results obtained from applying the proposed approach, discussing the performance of the predictive models and any significant findings or insights gained. In Section V, we interpret and discuss the empirical results in the context of our research objectives and identify potential areas for future research and improvement. Finally, in Section VI, we summarize the key findings of our study and draw meaningful conclusions based on the empirical analysis and discussions. By structuring the paper in this manner, we aim to provide a comprehensive overview of our research, methodology, and findings, offering valuable insights and practical guidance for advertisers seeking to optimize their advertising strategies using customer lifetime value prediction and proactive customer segment targeting.
II. LITERATURE REVIEW

In the realm of personalized advertising and maximizing return on investment, the understanding of customer segments and their dynamic changes over time has emerged as a crucial area of research. This literature review explores key studies related to customer segment changes, customer lifetime value (CLV) prediction, and proactive customer segment targeting.

Customer segmentation, a vital concept in marketing, allows businesses to categorize their customer base into distinct groups based on common characteristics [6], behaviors, or preferences. Various factors influencing customer segment changes and transitions have been examined by researchers. Christy et al highlighted the significance of RFM analysis for identifying valuable customer segments and guiding marketing initiatives [7]. Yuliani et al introduced a customer segmentation method using fuzzy C-means and fuzzy RFM, accounting for uncertainties in customer data [8]. Sembiring Brahmana et al investigated customer segmentation using the RFM model and clustering techniques such as K-means, K-medoids, and DBSCAN [9]. Dullaghan and Rozaki explored machine learning techniques for dynamic customer segmentation analysis in the mobile industry [10]. Ahani et al conducted market segmentation and travel choice prediction in spa hotels using online reviews [11]. Albuquerque et al applied support vector clustering for customer segmentation in the context of mobile TV service [12].

The prediction of customer lifetime value (CLV) has gained significant attention as it enables businesses to identify their most valuable customers historically and forecast their future value. Several predictive modeling and analytical techniques have been explored in this context. De Marco et al utilized cognitive analytics and artificial neural networks to manage CLV, facilitating customer value prediction and optimization. They found that the self-organizing map better classifies the customer base of the retailer [13]. Marisa et al explored the relationship between CLV and core drives, using clustering and the octalaxis gamification framework. Their study analyzed the relationship between CLV and eight core drives of customer motivation [14]. Yuan et al focused on a data-driven customer segmentation strategy based on the contribution to system peak demand [15]. Mosaddegh et al studied the dynamics of bank customers through value segments using big data analytics, identifying six major categories, including the pattern of Local Leaders whose transitions are repeated by some follower groups within the next two periods [16]. Khalili-Damghani et al proposed a hybrid approach combining clustering, rule mining, and decision tree analysis for personalized marketing [17].

To optimize advertising strategies, businesses need to proactively target specific customer segments that are likely to yield higher returns. Researchers have developed approaches to identify and prioritize these segments. Heldt et al introduced a predictive model called RFM/P, extending RFM analysis to enhance customer segmentation and targeting strategies [18]. Abidar et al proposed a new strategy for customer segmentation using machine learning techniques, highlighting the importance of targeted actions in marketing. Their approach demonstrates the effectiveness of machine learning in identifying customer segments and enabling businesses to tailor their marketing efforts for improved customer satisfaction and profitability [19]. Yuan et al focused on a data-driven customer segmentation strategy based on the contribution to system peak demand [15].

The ability to predict customer segment changes and align advertising efforts accordingly present substantial opportunities for growth and customer retention. By effectively identifying and engaging their most valuable customers, businesses can gain a competitive edge in the marketplace. The findings from studies in this field highlight the importance of leveraging CLV prediction and proactive customer segment targeting to optimize advertising strategies, allocate resources efficiently, and maximize return on investment.

In summary, the literature review emphasizes the significance of understanding customer segment changes, predicting CLV, and leveraging proactive targeting strategies in the realm of personalized advertising. The studies reviewed provide valuable insights and methodologies for businesses seeking to optimize their advertising strategies and enhance customer retention.

III. WORKFLOW MODEL

In order to reach a final outcome, various techniques and methods will be utilized in this research. The resulting framework workflow, depicted in Fig. 1, incorporates customer segmentation, RFM parameters, clustering, data analytics, CLV Prediction, and targeted actions.

A. Data Pre-processing

1) Data cleaning: When creating operational data, there are two standard approaches to handling missing numbers. As most data mining algorithms cannot handle data with missing values, the initial step is to simply remove data samples with missing values. This approach is only appropriate when the percentage of missing values is negligible. The second is to use missing value imputation techniques to substitute inferred values for missing data. There are two techniques for detecting outliers: statistical and clustering-based techniques [20].

2) Data reduction: Row-wise for data sample reduction and column-wise for data variable reduction are the two usual directions in which data reduction is carried out. Row-wise data reduction is possible using a variety of data sampling methods, including random and stratified sampling. The goal of feature extraction is to create new features based on linear or nonlinear combinations of existing variables, as opposed to feature selection, which only chooses usable features from already-existing variables [20].

3) Data scaling: Predictive modeling frequently requires data scaling, particularly when the input variables have multiple scales. The max-min normalization (i.e., \( x' = x - x_{\min} / x_{\max} - x_{\min} \)) and z-score standardization (i.e., \( x' = (x - \mu) / \sigma \)) are two of the most widely used methods in the building field, where \( x_{\min} and x_{\max} refer to the minimum and maximum of variable x, values of the variable, \( \mu \) is the mean and \( \sigma \) is the standard deviation [20].

4) Data transformation: Data transformation is mostly used in the construction industry to convert numerical data into categorical data in order to assure interoperability with data mining methods. Due to their simplicity, the equal-width and equal-frequency approaches are frequently utilized [20].
5) Data partitioning: The goal of data partitioning is to separate the entire set of data into various categories for in-depth study. For this goal, decision tree approaches and clustering analysis have been frequently applied in the building industry. For data partitioning, a variety of clustering methods have been used, including fuzzy c-means clustering, hierarchical clustering, entropy weighting k-means (EWKM), and k-means [20].

B. Clustering

1) Features selection: In practice, it is uncommon for all of a dataset’s variables to be helpful in creating a machine learning model. Repetitive variables decrease a model’s capacity to generalize and may also lower a classifier’s overall accuracy. A model’s overall complexity is also increased by including more variables. The objective of feature selection in machine learning is to identify the best set of features that make it possible to create effective models of the phenomena being examined. In machine learning, there are two different kinds of feature selection methods: supervised and unsupervised methods.

2) Modelization: A file that has been trained to detect various patterns is referred to as a machine learning model. By giving a model a method it can use to analyze and learn from a set of data, we may train it on that data. After the model has been trained, we can use it to analyze new data and forecast what will happen to it.

3) Cluster optimisation: Every clustering algorithm has its own strengths and weaknesses. In order to overcome these flaws in clustering algorithms, it is necessary to estimate the number of clusters based on assumptions and rely significantly on the initial centroids choice. It is vital to optimize, and the Elbow approach is one of the most used cluster optimization techniques.

4) Clustering performance: Any typical clustering system must answer the fundamental question of how accurate or reliable the clustering is. The separation between clusters is calculated using the Silhouette Score and Silhouette Plot. It shows the distance between each point in a cluster and points in other clusters. This metric, which has a range of [-1, 1], is excellent for visually examining similarities and differences between clusters.

C. Compute

1) RFM compute: RFM is a technique for providing significant value to each consumer. It is mostly utilized in marketing and has drawn the attention of the retail and business services industries. Based on the following criteria, RFM:
   - Recent: When was the client’s most recent order?
• Frequency: How frequently do they purchase?
• Monetary: How much do they spend?

The Fig. 2 displays a series of prior sales for a group of four clients.

![Fig. 2. Past sales for a set of four customers.](image)

The diagram depicts the RFM values for the clients, with the following information for each client:

- Recency: The amount of time that has passed since the last purchase, as indicated by the separation between the rightmost circle and the vertical dotted line that reads “Now”.
- Frequency: The space between the circles on a single line, which represents the interval between purchases.
- Monetary: The size of the circle represents the amount of money spent on each purchase. This amount could represent either the typical order value or the number of products the buyer ordered.

2) CLV compute: CLV is calculated as the total of net cash flows from consumers over their anticipated lifetime, taking the time value of money into account. The following formula can be used to represent this model. The research in [21] and Table I is showing it’s parameters:

\[
CLV = \sum_{i=1}^{n} \frac{R_i - C_i}{(1 + d)^i - 0.5}
\]  

### TABLE I. CLV FORMULA PARAMETERS

<table>
<thead>
<tr>
<th>Var</th>
<th>Explanation</th>
<th>Operationalization</th>
</tr>
</thead>
<tbody>
<tr>
<td>n</td>
<td>Expected life of a customer</td>
<td>n = the total number of periods of projected life of the customer under consideration</td>
</tr>
<tr>
<td>C_i</td>
<td>The total cost of customer in period i</td>
<td>Total cost of generating the revenue R_i in period i</td>
</tr>
<tr>
<td>R_i</td>
<td>Total revenue of customer in period i</td>
<td>The revenues of customers were assigned as their monetary values.</td>
</tr>
<tr>
<td>d</td>
<td>Discount rate (annual)</td>
<td>Discount.</td>
</tr>
</tbody>
</table>

E. Predict Customer Segment

In this part, we focus on the process of predicting future customer segments based on the developed workflow model. We delve into the various steps involved, including feature engineering, algorithm selection, building the machine learning model, model tuning, and ultimately predicting the future segments of customers. Feature engineering plays a crucial role in creating meaningful predictors for the machine learning model. We explore the different techniques and strategies employed to transform raw data into informative features that capture the relevant characteristics of customer behavior. Choosing the appropriate machine learning algorithm is a critical decision that impacts the accuracy and effectiveness of segment prediction. We explore a range of algorithms commonly used in customer segmentation tasks, such as decision trees, random forests, logistic regression, and gradient-boosting algorithms. In building the ML Model section, we detail the process of building the machine learning model for predicting customer segments. We discuss the steps involved in model training, validation, and evaluation. We split the dataset into training and test sets, and we discuss model performance metrics and interpretability, ensuring that the chosen model aligns with the objectives and requirements of segment prediction. In the Model Tuning section, we did some tuning to optimize the performance of the machine learning model. The final step in this part of our workflow is to use the trained and tuned model to predict future customer segments.

F. Actions

Based on the results obtained, businesses can implement effective actions to help to develop customer retention strategies and allocate their advertising and marketing investments more strategically. They can also tailor their advertising campaigns to target specific customer groups and help to draw roadmap for segment transition planning.

IV. EMPIRICAL RESULTS AND ANALYSIS

A. Data

The data used in this study was gathered from an online retailer [22]. The dataset covers the time period from the end of 2009 to November 2011. The collection includes 16759 invoices for 3881 items produced by 889 clients (Table II).

B. Data Pre-processing

1) Data cleaning: Following the completion of the data cleaning process, certain incorrect and missing values were removed from the data set. Table III summarises the attributes that were employed in this study. Table IV demonstrates a few modifications that we made to the data to make it cleaner.
We use whole prepared population in the analysis. Thus, we did not use any sampling method.

2) Data selection: We chose the following elements for this study: CustomerID, InvoiceDate, Quantity, UnitPrice. These attributes will make it easier for us to apply RFM models to this company’s customers and determine customer lifetime value.

3) Data transformation: No data transformations have been made in the database.

C. Time Frame for CLTV Calculation

In this study, we will assess the Customer Lifetime Value (CLTV) over a 6-month period and use it in parameter correlation analysis with other variables for the purpose of feature engineering.

D. LTV Clusters

The Table V represents the three LTVCluster derived from the Elbow method, along with various statistical measures such as count, mean, standard deviation, minimum, 25th percentile, median (50th percentile), 75th percentile, and maximum. Here’s what each column represents: LTVCluster represents different clusters or segments based on the Lifetime Value (LTV) of customers and the Count column indicates the number of data points or observations within each LTVCluster. The Mean column represents the average value of the Lifetime Value within each LTVCluster. It provides insight into the average LTV for customers within each cluster. The Std column represents the standard deviation of the Lifetime Value within each LTVCluster. It provides a measure of the variability or dispersion of LTV values within each cluster and the Min column indicates the minimum value of the Lifetime Value within each LTVCluster. It represents the lowest observed LTV for customers within each cluster. The 25th percentile column represents the value below which 25% of the Lifetime Values fall within each LTVCluster. It provides an insight into the lower quartile or first quartile value for LTV within each cluster and the 50th percentile column represents the median value of the Lifetime Value within each LTVCluster. It indicates the midpoint of the LTV distribution within each cluster. The 75th percentile column represents the value below which 75% of the Lifetime Values fall within each LTVCluster. It provides an insight into the upper quartile or third quartile value for LTV within each cluster. Finally, the Max column indicates the maximum value of the Lifetime Value within each LTVCluster. It represents the highest observed LTV for customers within each cluster.

These statistics provide an overview of the distribution and characteristics of the Lifetime Value within each LTVCluster. They can be used to compare and understand the differences in LTV between different customer clusters or segments.

E. Feature Engineering

In the feature engineering phase, we utilize RFM (Recency, Frequency, Monetary) scores calculated using the model introduced in our previous paper. These scores are merged with the calculated Customer Lifetime Value (CLV) for the 6-month period. To prepare the data for modeling, we perform various feature engineering techniques. First, we convert categorical columns, such as segment categories (low, mid, high), into numerical columns by assigning them values of 0 or 1. These statistics provide an overview of the distribution and characteristics of the Lifetime Value within each LTVCluster. They can be used to compare and understand the differences in LTV between different customer clusters or segments.

### Table II. Transactional Data

<table>
<thead>
<tr>
<th>Id</th>
<th>Invoice</th>
<th>StockCode</th>
<th>Description</th>
<th>Quantity</th>
<th>InvoiceDate</th>
<th>UnitPrice</th>
<th>CustomerID</th>
<th>Country</th>
</tr>
</thead>
<tbody>
<tr>
<td>931932</td>
<td>574387</td>
<td>22726</td>
<td>ALARM CLOCK BAKELIKE GREEN</td>
<td>8</td>
<td>10/4/2011</td>
<td>3.75</td>
<td>12944.0</td>
<td>United Kingdom</td>
</tr>
<tr>
<td>404468</td>
<td>526000</td>
<td>22028</td>
<td>PENNY FARTHING BIRTHDAY CARD</td>
<td>10</td>
<td>10/22/2010</td>
<td>0.42</td>
<td>12787.0</td>
<td>Netherlands</td>
</tr>
<tr>
<td>764721</td>
<td>560569</td>
<td>22423</td>
<td>REGENCY CAKESTAND 3 TIER</td>
<td>1</td>
<td>7/19/2011</td>
<td>12.75</td>
<td>12480.0</td>
<td>Germany</td>
</tr>
<tr>
<td>442525</td>
<td>532056</td>
<td>21530</td>
<td>DAIRY MAID TOASTRACK</td>
<td>6</td>
<td>11/10/2010</td>
<td>2.95</td>
<td>12739.0</td>
<td>United Arab Emirates</td>
</tr>
<tr>
<td>272695</td>
<td>516189</td>
<td>85049D</td>
<td>BRIGHT BLUES RIBBONS</td>
<td>12</td>
<td>7/18/2010</td>
<td>1.25</td>
<td>12625.0</td>
<td>Germany</td>
</tr>
<tr>
<td>164614</td>
<td>505168</td>
<td>16055</td>
<td>POPART COLL BALLPOINT PEN ASST</td>
<td>50</td>
<td>4/20/2010</td>
<td>0.21</td>
<td>14156.0</td>
<td>EIRE</td>
</tr>
<tr>
<td>88489</td>
<td>497879</td>
<td>21931</td>
<td>JUMBO STORAGE BAG SUKI</td>
<td>10</td>
<td>2/14/2010</td>
<td>1.95</td>
<td>12422.0</td>
<td>Australia</td>
</tr>
</tbody>
</table>

### Table III. Attributes

<table>
<thead>
<tr>
<th>Attributes</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>InvoiceNo</td>
<td>Unique ID to identify each Invoice</td>
</tr>
<tr>
<td>StockCode</td>
<td>Unique ID for each item in stock</td>
</tr>
<tr>
<td>Description</td>
<td>A short description for each item</td>
</tr>
<tr>
<td>Quantity</td>
<td>Number of items bought</td>
</tr>
<tr>
<td>UnitPrice</td>
<td>The price of each item</td>
</tr>
<tr>
<td>CustomerID</td>
<td>Unique ID for each Customer</td>
</tr>
<tr>
<td>Country</td>
<td>The country where the Customer lives</td>
</tr>
</tbody>
</table>

### Table IV. Data Cleaning

<table>
<thead>
<tr>
<th>Problem</th>
<th>Solution</th>
</tr>
</thead>
<tbody>
<tr>
<td>Null Invoices</td>
<td>Subtract from the dataset (not pertinent to this study)</td>
</tr>
<tr>
<td>Negative UnitPrice</td>
<td>Delete from this data (the organization included this in order to adjust bad credit)</td>
</tr>
<tr>
<td>Invoice with no customerID</td>
<td>Since we will be doing customer segmentation, remove any rows where customerID is NA.</td>
</tr>
</tbody>
</table>

TABLE V. LTV CLUSTER

<table>
<thead>
<tr>
<th>LTVCluster</th>
<th>Count</th>
<th>Mean</th>
<th>Std</th>
<th>Min</th>
<th>25%</th>
<th>50%</th>
<th>75%</th>
<th>Max</th>
</tr>
</thead>
<tbody>
<tr>
<td>0</td>
<td>2014.0</td>
<td>332.786480</td>
<td>390.273105</td>
<td>-609.40</td>
<td>0.0000</td>
<td>191.805</td>
<td>569.8225</td>
<td>1369.27</td>
</tr>
<tr>
<td>1</td>
<td>400.0</td>
<td>2408.376375</td>
<td>917.844147</td>
<td>1375.75</td>
<td>1695.0325</td>
<td>2075.775</td>
<td>2905.2150</td>
<td>4969.83</td>
</tr>
<tr>
<td>2</td>
<td>50.0</td>
<td>7633.641200</td>
<td>2225.110687</td>
<td>5074.93</td>
<td>6088.2675</td>
<td>6708.085</td>
<td>8838.1825</td>
<td>13636.42</td>
</tr>
</tbody>
</table>

- Segment_Mid-Value: 0.105407
- CustomerID: -0.055825
- Recency: -0.241712
- Segment_Low-Value: -0.263938

These correlation values provide insights into the relationships between the LTVCluster and other parameters. A positive correlation indicates a direct relationship, where an increase in one parameter is associated with an increase in LTVCluster. For example, DataF2_Monetary, Monetary, and MonetaryCluster show strong positive correlations, suggesting that higher monetary value and overall customer spending are indicative of a higher LTVCluster. Conversely, negative correlation coefficients suggest an inverse relationship, where an increase in one parameter is associated with a decrease in LTVCluster. In this case, Recency and Segment_Low-Value exhibit negative correlations, indicating that longer periods of inactivity and lower segment values are linked to a lower LTVCluster.

To build the machine-learning model, we split the dataset into training and test sets. The training set is utilized for training the model, while the test set is used to evaluate the model’s performance on unseen data.

F. Algorithms Comparison

In this section, we compare the performance of different machine learning algorithms based on their mean and standard deviation scores in Table VII.

Among the algorithms evaluated in our study, LogisticRegressionCV (LR) demonstrated a mean score of 0.839304 and a low standard deviation of 0.019146, indicating consistently good performance. The XGBClassifier (XGB) followed closely with a mean score of 0.831148 and a standard deviation of 0.012742, showcasing reliable and consistent results. The KNeighborsClassifier (KNN) achieved a mean score of 0.838491, similar to LR, but with a slightly higher standard deviation of 0.023427, implying a slightly higher variability in its performance. On the other hand, the DecisionTreeClassifier (DT) algorithm outperformed the others with the highest mean score of 0.845802 and a low standard deviation of 0.012548, demonstrating both high accuracy and consistency. The RandomForestClassifier (RF) achieved a mean score of 0.831974, similar to XGB and LR, with a moderate standard deviation of 0.015125. The AdaBoostClassifier (ADA) also performed well with a mean score of 0.834416 and a low standard deviation of 0.014393, comparable to XGB and LR. Lastly, the SVC algorithm obtained a mean score of 0.825506, slightly lower than other algorithms, but with a low standard deviation of 0.012568, indicating consistent results. These findings provide valuable insights into the performance and stability of each algorithm, guiding the selection of the most suitable model for predicting customer segment changes.

Based on this analysis, the DecisionTreeClassifier (DT) and XGBClassifier (XGB) show the highest mean score and lowest standard deviation, suggesting it performs the best among the listed algorithms. However, it’s also important to consider other factors such as computational complexity, interpretability, and specific requirements of your task when choosing the most suitable algorithm.

The choice between XGBClassifier and DecisionTreeClassifier depends on various factors and considerations. Our preference for XGBClassifier stems from its utilization of the powerful XGBoost algorithm, renowned for its exceptional performance in machine learning tasks. Unlike a single Decision Tree, XGBClassifier excels in handling complex datasets and often achieves higher accuracy. This is achieved by combining multiple weak decision trees through boosting techniques, resulting in improved overall performance. While Decision trees can be prone to overfitting, XGBClassifier incorporates regularization techniques such as shrinkage to mitigate this issue. Moreover, it performs automatic feature selection, ensuring the inclusion of relevant features and reducing the risk of using irrelevant or noisy ones. Although decision trees are generally regarded as more interpretable, XGBClassifier provides valuable insights through variable importances, indicating the relative significance of features. Additionally, XGBClassifier offers greater flexibility in handling missing values, enhancing the robustness of the model. With its ability to capture complex nonlinear relationships and interactions through gradient boosting, XGBClassifier surpasses DecisionTreeClassifier in scenarios where features exhibit nonlinear relationships with the target variable. Furthermore, XGBClassifier’s optimization for performance and efficiency, including parallel processing and tree pruning techniques, makes it more adept at handling large datasets with numerous features. While DecisionTreeClassifier can be faster for training and prediction, XGBClassifier provides superior scalability and efficiency in such cases.

TABLE VII. ALGORITHMS COMPARISON

<table>
<thead>
<tr>
<th>Algorithm Name</th>
<th>Mean</th>
<th>Std</th>
</tr>
</thead>
<tbody>
<tr>
<td>LogisticRegressionCV(LR)</td>
<td>0.839304</td>
<td>0.019146</td>
</tr>
<tr>
<td>XGBClassifier(XGB)</td>
<td>0.831148</td>
<td>0.012742</td>
</tr>
<tr>
<td>KNeighborsClassifier(KNN)</td>
<td>0.838491</td>
<td>0.023427</td>
</tr>
<tr>
<td>DecisionTreeClassifier(DT)</td>
<td>0.845802</td>
<td>0.012548</td>
</tr>
<tr>
<td>RandomForestClassifier(RF)</td>
<td>0.831974</td>
<td>0.015155</td>
</tr>
<tr>
<td>AdaBoostClassifier(ADA)</td>
<td>0.834416</td>
<td>0.014393</td>
</tr>
<tr>
<td>SVC</td>
<td>0.825506</td>
<td>0.012568</td>
</tr>
</tbody>
</table>
To further analyze and improve the customer segmentation using the XGBClassifier, we should consider techniques such as hyperparameter tuning and do more feature engineering.

### G. Build and Run the ML XGB Model

The “clvcluster” represents the predicted customer clusters based on the features in our dataset. The precision, recall, and f1-score metrics provide insights into how well the XGBClassifier is performing in predicting customer clusters. Based on the provided metrics for each class, we can evaluate the model’s performance for customer segmentation. Higher precision, recall, and f1-scores for a particular cluster indicate that the model is more accurate in predicting customers belonging to that cluster.

The XGB classifier achieved an accuracy of 96% on the training set and an accuracy of 84% on the test set. This suggests that the model has learned the patterns in the training data well and is performing reasonably well on unseen data.

In terms of class-wise metrics:
- **Class 0:**
  - Precision: 0.89
  - Recall: 0.94
  - F1-score: 0.91
  - Support: 1017
- **Class 1:**
  - Precision: 0.45
  - Recall: 0.33
  - F1-score: 0.38
  - Support: 184
- **Class 2:**
  - Precision: 0.56
  - Recall: 0.32
  - F1-score: 0.41
  - Support: 31

These metrics provide insights into the performance of the XGB classifier for each customer cluster. Class 0 has relatively high precision, recall, and F1-score, indicating good predictive performance for this cluster. Class 1 has a lower precision, recall, and F1-score, suggesting that the model struggles more to accurately predict instances in this cluster. Class 2 also has relatively lower precision, recall, and F1-score, indicating room for improvement in predicting instances for this cluster (Table VIII).

### TABLE VIII. PRECISION

<table>
<thead>
<tr>
<th>Class</th>
<th>Precision</th>
<th>Recall</th>
<th>F1-score</th>
<th>Support</th>
</tr>
</thead>
<tbody>
<tr>
<td>0</td>
<td>0.89</td>
<td>0.94</td>
<td>0.91</td>
<td>1017</td>
</tr>
<tr>
<td>1</td>
<td>0.45</td>
<td>0.33</td>
<td>0.38</td>
<td>184</td>
</tr>
<tr>
<td>2</td>
<td>0.56</td>
<td>0.32</td>
<td>0.41</td>
<td>31</td>
</tr>
</tbody>
</table>

To further analyze and improve the customer segmentation using the XGBClassifier, we should consider techniques such as hyperparameter tuning and do more feature engineering.

### H. Improve the Model

The XGB classifier achieved an accuracy of 93% on the training set and maintained the same accuracy of 84% on the test set. This indicates that the model is still performing well on the test data and is not overfitting, as the training and test accuracies are relatively close.

Let’s look at the class-wise metrics:
- **Class 0:**
  - Precision: 0.89
  - Recall: 0.95
  - F1-score: 0.92
  - Support: 1017
- **Class 1:**
  - Precision: 0.49
  - Recall: 0.36
  - F1-score: 0.42
  - Support: 184
- **Class 2:**
  - Precision: 0.73
  - Recall: 0.35
  - F1-score: 0.48
  - Support: 31

Comparing these metrics (Table VII) with the previous results (Table IX), we can observe some changes. The precision, recall, and F1-scores for class 0 remain relatively similar, indicating that the model’s performance for this cluster is consistent.

For class 1, there is a slight improvement in precision, recall, and F1-score, suggesting that the adjustment to max_depth=4 might have helped the model better capture patterns for this cluster.

Class 2 shows a significant improvement in precision, recall, and F1-score. The model’s ability to predict instances in this cluster has notably improved.

The adjustment in max_depth seems to have improved the model’s performance for some classes while maintaining a similar level of accuracy. However, it’s important to note that further evaluation and analysis are needed to fully assess the effectiveness of the model, such as considering other evaluation metrics and potentially exploring additional model adjustments or techniques without forgetting the specific goals and requirements for customer segmentation.
### Table IX. Enhance Model

<table>
<thead>
<tr>
<th>Class</th>
<th>Precision</th>
<th>Recall</th>
<th>F1-score</th>
<th>Support</th>
</tr>
</thead>
<tbody>
<tr>
<td>0</td>
<td>0.89</td>
<td>0.95</td>
<td>0.92</td>
<td>1017</td>
</tr>
<tr>
<td>1</td>
<td>0.49</td>
<td>0.36</td>
<td>0.42</td>
<td>184</td>
</tr>
<tr>
<td>2</td>
<td>0.73</td>
<td>0.35</td>
<td>0.48</td>
<td>31</td>
</tr>
</tbody>
</table>

#### I. False Positive Rate

The receiver operating characteristic (ROC) curve is a graphical representation of the performance of a classification model. It illustrates the relationship between the true positive rate (sensitivity) and the false positive rate (specificity) for different threshold values.

In our case (Fig. 3), we have three classes: segment_low_value, segment_mid_value, and segment_high_value. Each class has its own ROC curve with its corresponding area under the curve (AUC) value.

- ROC of segment_low_value: The AUC value for this class is 0.84. This indicates that the model performs well in distinguishing between the low-value segment and the other classes. The higher the AUC value, the better the model’s ability to correctly classify instances of the low-value segment.

- ROC of segment_mid_value: The AUC value for this class is 0.80. This suggests that the model’s performance in distinguishing between the mid-value segment and the other classes is slightly lower compared to the low-value segment. However, an AUC of 0.80 still indicates a reasonably good classification performance.

- ROC of segment_high_value: The AUC value for this class is 0.97. This suggests that the model excels in distinguishing between the high-value segment and the other classes. An AUC of 0.97 indicates a high level of accuracy in correctly classifying instances of the high-value segment.

Additionally, we have two overall performance measures:

- Micro-average ROC curve: The AUC value for the micro-average ROC curve is 0.96. This measure takes into account the performance across all classes and provides an aggregated evaluation of the model’s overall classification performance. An AUC of 0.96 suggests a high level of accuracy in predicting the correct class across all segments.

- Macro-average ROC curve: The AUC value for the macro-average ROC curve is 0.87. This measure calculates the average AUC value across all classes, giving equal weight to each class. An AUC of 0.87 indicates a good overall performance of the model in distinguishing between the different segments.

Our model demonstrates strong performance in classifying the low-value, mid-value, and high-value segments individually, as indicated by the respective AUC values. The micro-average ROC curve also indicates high accuracy across all segments, while the macro-average ROC curve provides a balanced evaluation of the model’s overall performance.

#### J. Precision Recall Curve

The PrecisionRecallCurve shows the tradeoff between a classifier’s precision, a measure of result relevancy, and recall, a measure of completeness. For each class, precision is defined as the ratio of true positives to the sum of true and false positives, and recall is the ratio of true positives to the sum of true positives and false negatives.

\[
\text{Precision} = \frac{TP}{TP + FP} \quad (2)
\]

\[
\text{Recall} = \frac{TP}{TP + FN} \quad (3)
\]

where TP denotes true positive, TN stands for true negative, FP means false positive, FN denotes false negative.

A classifier’s precision can be thought of as a gauge of its accuracy. It is described for each class as the proportion of true positives to the total of true and false positives. Another way to phrase this question is, “For all instances classified positive, what percent was correct?” The capacity of a classifier to accurately detect all positive cases is measured by recall, which is also known as the completeness of the classifier. It is described as the ratio of true positives to the sum of true positives and false negatives for each class. Additionally, average precision expresses the precision-recall curve in a single number, which represents the area under the curve. It is determined by computing the weighted average of the precision attained at each threshold, where the weights correspond to the variations in recall between thresholds. When there are class imbalances, the Precision-Recall (PR) curve sheds important light on how well a classification model performs. The average precision value of 0.9 and the Micro-average PR curve for all clusters are both included in our PR curve. The aggregated accuracy and recall for all clusters are shown by the Micro-average PR curve. It offers a comprehensive assessment of the model’s capacity to locate favorable occurrences across all classes while taking into account the imbalances in a class distribution (Fig. 4).
The fact that the Micro-average PR curve intersects with the average precision curve at a recall value of 0.7 indicates a crucial point of trade-off in the classification performance. At this threshold, the precision achieved by the model is equal to the average precision of 0.9. It suggests that, on average, the model can correctly identify 90% of positive instances when the recall is 0.7. This threshold represents a balance between precision and recall for the overall classification performance.

Fig. 4. Precision recall curve.

K. Class Prediction Error

The Yellowbrick ClassPredictionError plot is a twist on other and sometimes more familiar classification model diagnostic tools like the Confusion Matrix and Classification Report [23], [24]. Similar to the classification report, this plot displays a stacked bar chart of the support (number of training samples) for each class in the fitted classification model. As in a Confusion Matrix, each segmented bar displays the percentage of predictions (including false negatives and false positives) for each class. We can utilize a ClassPredictionError to see which classes our classifier is struggling with and, more critically, what false positives it is producing for each class. This frequently enables us to better comprehend the advantages and disadvantages of various models as well as specific difficulties pertaining to your dataset. The class prediction error chart is a fast way to gauge how well the classifier predicts the appropriate classes.

The XGBClassifier demonstrates accurate predictions for the Segment_High_Value class. However, there are instances where it mislabels Segment_Low_Value as Segment_Mid_Value and misclassifies Segment_Mid_Value as Segment_High_Value. In a few cases, it also misclassifies Segment_Mid_Value as Segment_High_Value (Fig. 5).

V. DISCUSSION

The work conducted in this study provides valuable insights and tools that can significantly contribute to enhancing customer retention. This work can help improve customer retention by accurately predicting customer segment changes, businesses can identify customers who are at risk of churn or transitioning to lower-value segments. This enables proactive intervention through targeted retention strategies. By offering personalized incentives, tailored communication, and exclusive offers to these customers, businesses can increase their likelihood of staying engaged and loyal. Understanding which customers make up the high-value segment (Segment_High_Value) allows businesses to allocate their resources more effectively. By focusing efforts on retaining these high-value customers, businesses can maximize their return on investment. This can include allocating more advertising budget towards targeted campaigns for high-value customers, providing exceptional customer service, and offering exclusive benefits to strengthen their loyalty. The insights gained from predicting customer segment changes can be used to personalize marketing efforts. By tailoring advertisements, promotions, and communication to specific customer segments, businesses can increase engagement and relevance. This personalized approach enhances the customer experience and strengthens the bond between the customer and the business, leading to improved retention rates. By leveraging predictive models and analytics, businesses can estimate the customer lifetime value (CLV) for different segments. This information helps prioritize efforts and resources towards segments with higher CLV potential. By focusing on increasing CLV through customer retention, businesses can optimize their revenue streams and profitability. Anticipating customer segment changes allows businesses to take a proactive approach to customer relationship management. By identifying customers who are likely to transition to higher-value segments, businesses can develop strategies to nurture and guide their journey. This can involve providing personalized recommendations, cross-selling and upselling opportunities, and proactive customer support to enhance their overall experience and increase loyalty.

The work conducted in this study equips businesses with the knowledge and tools to better understand and predict customer segment changes. By leveraging this information, businesses can implement targeted retention strategies, optimize resource allocation, personalize marketing efforts, and proactively manage customer relationships. These efforts collectively contribute to improving customer retention rates and fostering long-term customer loyalty.
Like every research study, our work also faces certain constraints and shortcomings. One limitation is the size of the dataset used for analysis, which may affect the representativeness of the findings. Additionally, the research focused on a specific industry, and the results may not be directly applicable to other sectors.

VI. CONCLUSIONS

This paper highlights the effectiveness of predicting customer segment changes to enhance customer retention strategies in the online retail industry. By leveraging machine learning techniques and analytical approaches, businesses can gain valuable insights into customer behavior and forecast future segment transitions. This enables proactive decision-making and targeted actions to retain high-value customers, optimize marketing efforts, and allocate resources efficiently. Future research directions include refining predictive models and algorithms, incorporating external factors, and utilizing real-time data for dynamic segmentation. Advanced customer analytics techniques, like customer journey analysis and sentiment analysis, can provide deeper insights into customer preferences and needs, further enhancing retention strategies. Moreover, predictive analytics can extend beyond customer retention to areas like personalized pricing, inventory management, and supply chain optimization, enabling businesses to deliver a superior customer experience. The study’s findings underscore the potential of predicting customer segment changes for enhancing customer retention in the online retail industry. Continued research and innovation in this field will drive the ongoing evolution of customer retention strategies and foster long-term customer loyalty in the competitive online retail landscape.

AUTHORSHIP CONTRIBUTION STATEMENT

Lahcen ABIDAR: Conceptualization of this study, Methodology, Software, Data collection, reading and analyzing existing literature, analysis and interpretation of results, Writing - Original draft preparation. Donia ZAIDOUNI: Participate in the conceptualization of this study, Methodology, analysis and interpretation of results, Review - Original draft preparation. Ikran EL ASRI: Participate in the conceptualization of this study, Methodology, analysis and interpretation of results, Review - Original draft preparation. Abdesslam ENNOUARY: Supervised the work, participate in the conceptualization of this study, Methodology, Review - Original draft preparation.

REFERENCES


Review of Existing Datasets Used for Software Effort Estimation

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Faculty of Computing, Universiti Malaysia Pahang, 26600, Pekan, Pahang, Malaysia1
Associate Professor, Department of Computer Science, University of Évora, Portugal2
Professor, Department of Computer Science and Engineering
United International University, Satarkul, Badda, Dhaka, Bangladesh3

Abstract—The Software Effort Estimation (SEE) tool calculates an estimate of the amount of work that will be necessary to effectively finish the project. Managers usually want to know how hard a new project will be ahead of time so they can divide their limited resources in a fair way. In fact, it is common to use effort datasets to train a prediction model that can predict how much work a project will take. To train a good estimator, you need enough data, but most data owners don’t want to share their closed source project effort data because they are worried about privacy. This means that we can only get a small amount of effort data. The purpose of this research was to evaluate the quality of 15 datasets that have been widely utilized in studies of software project estimation. The analysis shows that most of the chosen studies use artificial neural networks (ANN) as ML models, NASA as datasets, and the mean magnitude of relative error (MMRE) as a measure of accuracy. In more cases, ANN and support vector machine (SVM) have done better than other ML techniques.

Keywords—Software effort estimation; software effort prediction; software effort estimation datasets

I. INTRODUCTION

Estimating how much work has to be put into creating software is a hot topic of study because of its significance in any software development process. If the amount of work involved is underestimated, not enough money or manpower will be put into the project, leaving no room for error in terms of the final product’s quality. On the other side, if you overestimate the amount of work that has to be done, you’ll likely end up with a large budget, which will drive up the cost of the program and reduce its competitive edge. Therefore, it is crucial to have a reliable estimate of the time and effort required to complete a project. Bosu et al. [1] emphasized the significance of data, defining software metrics as the gathering of quantitative measures as an intrinsic element of software quality control and assurance operations (particularly, the monitoring and recording of errors during development and testing). This way of thinking has won out. SEE research has expanded to cover a wide variety of problems since its inception, although the largest collections of work in the discipline have either suggested or evaluated models designed largely for effort/cost estimation. It has been argued that the use of metrics in SEE is invaluable because it allows for more informed decision-making during software development and maintenance, which in turn improves development productivity, decreases deployment cycle time, and boosts software quality [2]. Although there is no debate about the benefits of metrics in theory for software engineering, in recent years there have been growing concerns about the quality of the data being gathered and used in the creation of models to predict factors like software size and development effort.

Several recent publications [3] [4] [5] [6] detail the difficulties in assembling and analyzing empirical software engineering datasets. While the SEE research community has acknowledged and prioritized the detection and resolution of issues like noise, outliers, and missingness (or incompleteness), they have largely ignored issues like poor provenance, inconsistency, and commercial sensitivity [1]. Several “classic” SEE datasets have been used extensively in previous research on software effort estimation, and we detail such datasets here. These datasets may be found mostly in the PROMISE repository. These datasets were chosen because of their convenience and widespread application in SEE modeling. Our goal is to perform a comparative analysis of the content of these datasets by using them as a baseline. This will serve to draw attention to any problems related to SEE data collection in general. As a bonus, we’ll see how well it works as a comparative tool. By analyzing these established datasets, a standard model could be formulated by academics and practitioners would benefit greatly. Decisions on whether or not to employ a specific dataset in SEE modeling will be improved as a result of this.

Numerous estimation strategies are described in the SEE literature and are grouped into the three primary categories of algorithmic, non-algorithmic, and machine learning (see Fig. 1). For software work estimation, algorithmic strategies use statistical and mathematical formulation. Non-algorithmic models are based on evaluative and interpretive analyses. These models analyze historical data from previously completed projects. Techniques based on machine learning offer an alternative to algorithmic modeling [7]. There is no clear "front-runner" technique for any of the data quality issues in the considered dataset, despite the fact that several methods have been presented to identify or analyze the various quality aspects of SEE datasets. Therefore, we apply the best-in-class method(s) for evaluating the quality of these widely-referenced datasets in Table I. The ultimate goal of this benchmarking exercise is to promote a comprehensive evaluation of data quality before modeling by showing how appropriate techniques, such as algorithmic, non-algorithmic, and machine learning (Fig. 1) can be used by researchers and practitioners to evaluate the quality of their own datasets and inspire them to create or adopt new and improved methods of data collection. The following are the article’s main contributions:

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• We provide insights into some of the most popular datasets used for estimating software development efforts.

• We find out which is the most used technique for the SEE datasets.

• Which is the most used performance matrix, we also find out this for future researchers.

The rest of the article is divided into the following sections. Section II contains research findings related to the SEE dataset. Section III explains the methodology of our study. The details of the dataset are described in Section IV. The discussion of our work is covered in Section V. Section VI concludes the paper.

II. RELATED WORK

It is impossible to exaggerate the significance of data to the study of SEE because they are at the center of the field’s practical application. It is crucial that personnel in charge of gathering data be well-trained and aware of the various issues that could exist in datasets. It is seen that the majority of researchers use secondary data in SEE modeling [24]. Secondary data is research information that has already been obtained and is available to researchers. Secondary users are individuals who only infrequently make use of the product or those who do so via a third party. This is vital to make sure that the best procedures are used to produce and make use of the accessible data that is most trustworthy. In order to provide secondary users with an understanding of how the data were gathered, the processes that were used should, at the very least, be documented. In recent years, there has been a growing awareness of the difficulties associated with the collection and utilization of empirical software engineering datasets [25] [26], despite the fact that the overall body of literature on SEE data quality remains quite limited [27]. In this part, we examine previous assessment studies and provide a cursory mention of a few of the steps that others have done to enhance the quality of SEE datasets and repositories. First, We present a sample of studies that have assessed the state of SEE datasets from a particular angle or aspect of data quality. The SEE community mostly employs this approach to address problems that have an impact on software engineering datasets. The studies we present assess the state of SEE datasets from several dimensions of data quality. Additionally, we provide examples of studies in this chapter that built SEE prediction models using metrics from open-source projects. These metrics were gathered from the projects themselves. After this, a discussion of the few research works that have analyzed the current state of SEE datasets from numerous perspectives that have considered multiple data quality dimensions is discussed later. Table I shows the summary of the datasets used in this paper.
TABLE I. SUMMARY OF THE DATASET USED IN THIS STUDY

<table>
<thead>
<tr>
<th>SL NO</th>
<th>Dataset name</th>
<th>Source</th>
<th>No of records</th>
<th>No of attributes</th>
<th>Output attribute-effort</th>
<th>Size(Unit measurement)</th>
<th>Ref</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>Desharnais</td>
<td>GitHub</td>
<td>81</td>
<td>12</td>
<td>Person-hours</td>
<td>Function point</td>
<td>[8]</td>
</tr>
<tr>
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<td>Promise</td>
<td>63</td>
<td>18</td>
<td>Person-months</td>
<td>LOC</td>
<td>[9]</td>
</tr>
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<td>China</td>
<td>Promise</td>
<td>499</td>
<td>16</td>
<td>Person-months</td>
<td>Function points</td>
<td>[10]</td>
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<td>GitHub</td>
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<td>8/9</td>
<td>Person months</td>
<td>“KSLOC”</td>
<td>[12]</td>
</tr>
<tr>
<td>6</td>
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<td></td>
<td>53</td>
<td>9</td>
<td>Person-months</td>
<td></td>
<td>[13]</td>
</tr>
<tr>
<td>7</td>
<td>ISBSG</td>
<td>ISBSG</td>
<td>1192</td>
<td>13</td>
<td>Man-hours</td>
<td>Multiple</td>
<td>[14]</td>
</tr>
<tr>
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<td>Promise</td>
<td>24</td>
<td>8</td>
<td>Person-Months</td>
<td>Function point</td>
<td>[15]</td>
</tr>
<tr>
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<td>Zenodo</td>
<td>15</td>
<td>7</td>
<td>Person-months</td>
<td>KSLOC</td>
<td>[16]</td>
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<tr>
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<td>SEACRAFT</td>
<td>145</td>
<td>9</td>
<td>Person-hours</td>
<td>Function Points</td>
<td>[17]</td>
</tr>
<tr>
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<td>NASA93</td>
<td>Promise</td>
<td>93</td>
<td>17</td>
<td>Person-months</td>
<td>LOC</td>
<td>[18]</td>
</tr>
<tr>
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<td>UCP</td>
<td>Promise</td>
<td>70</td>
<td>17</td>
<td>Person-months</td>
<td>LOC</td>
<td>[19]</td>
</tr>
<tr>
<td>13</td>
<td>Edusoft</td>
<td>Github</td>
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<td>4</td>
<td>Person-months</td>
<td>Files</td>
<td>[20]</td>
</tr>
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<td></td>
<td>38</td>
<td>9</td>
<td>Person-Hours</td>
<td>Function Points</td>
<td>[21]</td>
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</table>

TABLE II. LITERATURE REVIEW

<table>
<thead>
<tr>
<th>SL No</th>
<th>Ref</th>
<th>Year</th>
<th>Datasets</th>
<th>Used Techniques</th>
<th>Evaluation matrix</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>[28]</td>
<td>2023</td>
<td>Albrecht, Kemerer, Miyazaki, China, COCOMO, Maxwell, NASA</td>
<td>GWO-FC, FLNN</td>
<td>MSE, MAE, RMSE, MMRE, RSRE, PRED(25)</td>
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<tr>
<td>2</td>
<td>[21]</td>
<td>2023</td>
<td>Edusoft</td>
<td>KNN, SVR, DT</td>
<td>MAE, MBRE</td>
</tr>
<tr>
<td>3</td>
<td>[29]</td>
<td>2023</td>
<td>Albrecht, Maxwell, NASA, Telecom, Kemerer, China, Desharnais</td>
<td>CBR-GA</td>
<td>MAE, MBRE</td>
</tr>
<tr>
<td>4</td>
<td>[30]</td>
<td>2022</td>
<td>Desharnais, COCOMO81, China, Maxwell and Miyazaki94</td>
<td>ANN, SVR</td>
<td>MAR, MMRE</td>
</tr>
<tr>
<td>5</td>
<td>[31]</td>
<td>2022</td>
<td>China, Maxwell, and COCOMO81</td>
<td>DTR, RF, LR, LassoR, Ridge R,</td>
<td>MMRE, PRED(25)</td>
</tr>
<tr>
<td>7</td>
<td>[33]</td>
<td>2022</td>
<td>ISBSG</td>
<td>M5, GBrReg, LinearSVR, and RFR</td>
<td>MAE, RMSE</td>
</tr>
<tr>
<td>8</td>
<td>[34]</td>
<td>2022</td>
<td>COCOMO81, NASA93, Maxwell and China</td>
<td>SVM, RF, Ridge Regression, KNN, and Gradient Boosting Machines</td>
<td>MAE, MSE, MMRE</td>
</tr>
<tr>
<td>10</td>
<td>[35]</td>
<td>2022</td>
<td>COCOMO81, CHINA</td>
<td>Deepnet,NN, RF, SVM</td>
<td>MAE, RMSE, MSE, and R-Squared</td>
</tr>
<tr>
<td>11</td>
<td>[36]</td>
<td>2021</td>
<td>Albrecht, China, Desharnais, Maxwell</td>
<td>Deep-MNN, GWDNNSB</td>
<td>MRE, MMRE, MBRE, MI-BRE, PRED, MAE, SA, MR</td>
</tr>
<tr>
<td>13</td>
<td>[38]</td>
<td>2021</td>
<td>Albrecht, China, Desharnais, Kemerer, Kitchenham, Maxwell, and Cocomo8</td>
<td>RF, SVM, DT, NN, Ridge, LASSO, ElasticNet, DeepNet, Averaging, Bagging, Boosting, Stacking using RF</td>
<td>MAE, RMSE, and R-squared</td>
</tr>
<tr>
<td>14</td>
<td>[39]</td>
<td>2021</td>
<td>Desharnais</td>
<td>KNN, LR, SVM, ML</td>
<td>RMSE, MSE, MAE</td>
</tr>
<tr>
<td>15</td>
<td>[40]</td>
<td>2020</td>
<td>Desharnais, China, Albrecht</td>
<td>GP (genetic programming)</td>
<td>MMRE, Pred(25)</td>
</tr>
<tr>
<td>16</td>
<td>[41]</td>
<td>2020</td>
<td>NASA93(3,63,60)</td>
<td>FPA</td>
<td>MMRE</td>
</tr>
<tr>
<td>17</td>
<td>[42]</td>
<td>2020</td>
<td>Desharnais</td>
<td>LR, RF, Multi-layer perceptron</td>
<td>C.C, MAE, RMSE, RRAE, RSE</td>
</tr>
<tr>
<td>18</td>
<td>[43]</td>
<td>2020</td>
<td>COCOMO 81, Desharnais</td>
<td>KNN, SVM, NN, RF and backpropagation</td>
<td>MMRE</td>
</tr>
<tr>
<td>19</td>
<td>[44]</td>
<td>2020</td>
<td>NASA 93</td>
<td>COCOMO-II</td>
<td>MMRE, MRE</td>
</tr>
</tbody>
</table>

We conducted a metareview of the study and survey papers, and using that information, we were able to determine the SEE methodologies, datasets, and accuracy measurements that are most commonly employed. Table II provides an overview of the SEE methods, datasets, and accuracy measurements that are most frequently utilized.

We discuss the strengths and disadvantages of the most commonly used SEE approaches after doing a review of the relevant research and gaining an understanding of SEE techniques. Following are the SEE approaches which are most used by the researchers.

Linear regression: Linear regression is a statistical modeling technique used to examine the relationship between one or more independent variables and a dependent variable. It assumes that the variables are linearly related, meaning that changes in the dependent variable are proportional to changes in the independent variable. The benefits include being simple to grasp, apply, interpret, and clarify. Additionally, it is computationally cheap and works well with tiny datasets. Here are some of the flaws: it presupposes a linear connection and is therefore inappropriate for data with a nonlinear connection [45].

Artificial neural network: The following are its strengths: Feature engineering is not necessary, and it can learn complex correlations in the data. The shortcomings are as follows: training requires a lot of data, therefore it is computationally expensive. Additionally, it is challenging to evaluate the results since it is difficult to comprehend the assumptions that underlie them. It might have an overfitting issue, can’t handle missing values, and needs to convert categorical values to numeric values [45].
Analogies, which are similarities or correspondences between two or more items, are used to gather knowledge, draw conclusions, or resolve issues [34]. The following are its advantages: It can handle outliers and the justification for the result is simple to understand. The following are the flaws: computationally demanding, susceptible to the similarity function, requiring the conversion of categorical variables to numeric types, unable to handle missing values, and challenging to find a solution if similar work has not been done previously.

**Fuzzy logic:** Fuzzy logic is a mathematical foundation for reasoning and making decisions in ambiguous, imprecise, and uncertain situations. The following are the virtues: It is based on the theory of classes with flexible boundaries so that it can accommodate data uncertainty caused by measurement errors in data collection. It can also cope with model uncertainty. It enhances the performance of ML and non-ML models and resembles human reasoning. Its only limitation is that when combined with ML or non-ML models, it becomes computationally intensive.

**Machine Learning (ML) techniques:** The most popular ML methods for SEE are tree-based models and Support vector Machine(SVM) [46]. Each ML method has advantages and disadvantages of its own. The following are the strengths of the tree-based ML models that use the decision tree (DT) [47], CART, and random forest(RF) techniques: intuitive, making it simple to comprehend and analyze the model’s findings. It is appropriate when there are nonlinear relationships in the data and can handle both category and numerical data. It can be said that it is capable of handling the outliers or that it is robust with them. The following are its flaws: If the dataset is small, DT is prone to overfitting. It is unable to handle missing values. A huge dataset has a high time complexity, and even a minor change in the data can significantly alter the model [48]. The following are some of SVM’s advantages: It learns nonlinear relationships in the data and is appropriate for large dimensional data [48], the following are the flaws: memory-intensive and possibly not scalable for large datasets.

**Optimization techniques:** Optimization techniques are mathematical methods and algorithms that are used to discover the optimal solution to a given problem given a set of constraints. The purpose is to maximize or reduce an objective function, which represents the quantity to be optimized, while adhering to specified conditions or constraints. The following are its advantages: it may be used to pick and weight features, and when combined with ML or non-ML techniques, it can increase estimation accuracy. The following are the flaws: Due to its nondeterministic nature and high processing cost, outcomes may differ from one attempt to the next [28].

### III. REVIEW METHODOLOGY

We downloaded all relevant papers and analyzed the number of SEE papers published in recent years (2020-2023(June)), categorizing them by the journals in which they appeared. Our searches began by retrieving all SEE files, which were then sorted by the publisher. For the search, we used reputable online resources such as Google, Springer, Elsevier, and similar databases. We compiled a research database of abstracts, keywords, and titles and carefully analyzed both the content and algorithms of each publication. Based on the inclusion/exclusion criteria we have selected the related paper. In this way, we were able to classify the publications on the topic of SEE.

**A. Research Questions**

We answer the following study questions:

- What are the most commonly utilized datasets in SEE research?
- What accuracy metrics are utilized most frequently in SEE studies?

**B. Inclusion/Exclusion Criteria**

**Inclusion Criteria.**

- Studies that used both ML and non-ML techniques for software effort estimation.
- Papers that are written in English.
- Papers that are published in a conference or journal.
- Paper that used publicly available datasets.

**Exclusion Criteria.**

- The title, abstract, or even their content was not closely related to our search string, however without any semantic interplay.
- No similarity with the research theme, or even the focal aim was completely contrary to the purpose of the issues addressed in the RQs.
- Not based on publicly available datasets.

When searching for information in the literature, we utilized the following search strings: “software effort estimation” OR “software cost estimation.” The aims of the study as well as its research questions were taken into consideration when developing the search string.

**C. Dataset Description**

For accurate effort forecasts, the dataset’s quality that supports an estimating model’s central premise is crucial. According to this theory, a characteristic is only considered a trustworthy predictor of the outcome if it has a strong correlation with the effort; else, it is unimportant. Datasets considered for this review paper were Desharnais, Cocomo81, China, Maxwell, Miyazaki94, Tukutuku, ISBSG, Albrecht, Kemener, Kitchenham, Nasa93, and Edusoft dataset. The mean, also known as the average, is a measure of central tendency that is calculated by dividing the sum of all values in a dataset by the total number of data points. The standard deviation measures the spread or dispersion of data points around the mean. It expresses how far the data deviates from the average. Min is the dataset’s minimum value, and Max is the dataset’s maximum value.
CHINA dataset: The CHINA dataset for predicting software effort consists of 19 attributes. There are 499 different project instances in total. Table III provides the CHINA dataset’s descriptive statistics.

1) Maxwell dataset: The Maxwell dataset, which was compiled from one of the largest commercial banks in Finland, has 62 projects that are each described by 23 attributes. Table IV provides a comprehensive summary of the Maxwell dataset. Project Size in Function Points serves as the only numerical attribute.

2) COCOMO81 dataset: The COCOMO81 dataset was often used to verify various effort estimation techniques. It consists of 63 software projects, each of which is described by 18 qualities along with a concrete effort. The COCOMO81 dataset measures real effort in terms of person-months, which are the number of months required for one person to develop a certain project. Table V provides the full description of the COCOMO81 dataset.

3) Albrecht dataset: The Albrecht dataset includes 24 software programs created with third-generation languages like COBOL, PL1, etc. Six independent number attributes and one dependent numeric attribute, “work hours,” which indicates the appropriate effort in 1000 hours, are used to define the dataset. Projects were developed in COBOL, PL1, and database management languages for the remainder. Table VI gives a detailed explanation of the Albrecht dataset.

4) Desharnais dataset: The Desharnais dataset began with 81 software projects gathered from Canadian software businesses. This dataset is described by ten attributes: two dependent factors (time and effort in ‘person-hours’) and eight independent attributes. Unfortunately, four projects out of 81 had missing values, therefore we eliminated them because they could have influenced the estimation procedure. This data preprocessing stage yielded 77 finished software projects. Table VII provides the full description of the Desharnais dataset.

5) Kemerer dataset: In the Kemerer dataset, which consists of 15 software projects, six qualities and one predictable attribute with a “man-month” unit of measurement are used to define the projects. Two categories and four numerical qualities each represent one of the six attributes. Table VIII provides the full description of the Kemerer dataset.

6) Miyazaki94 dataset: Miyazaki provided the Miyazaki94 dataset. 48 software projects are represented in this collection. In total, there are nine qualities. Seven of the nine attributes are conditional attributes, one is a decision attribute, and one is an identifier. Table IX provides the full description of the Miyazaki94 dataset.

7) Tukutuku dataset: In the Tukutuku dataset, 53 online projects are present. Nine numerical attributes are used to describe each online application, including the quantity of media assets, HTML or SHTML files used, and team experience. Table X provides the full description of the Tukutuku dataset.

8) UCP dataset: The UCP dataset consists of 70 instances with 17 attributes for software effort estimation. Table XI provides the full description of this dataset.

9) NASA dataset: NASA datasets have been used to assess how well evolutionary algorithms function. Bailey and Basili provided this dataset in 1981. Shin and Goel utilized it for the first time in 2000, followed by Oliveira in 2006. There are 18 project instances in the dataset. M (methodology utilized) and DL (number of developed lines of source code with comments) are two independent qualities. The dependent characteristic of effort is the number of man-months needed to complete the project. Table XII provides the full description of the NASA dataset.

10) ISBSG dataset: ISBSG21 dataset version 2021 includes 10,531 cross-company projects from various nations, organizations kinds, and development kinds. We chose the dataset instances in accordance with ISBSG rules and the procedure described in [49]. As a result, 1179 novel project category IFPUG version 4+ projects with quality A and B have been created. The feature selection was carried out in accordance with the technique proposed by Dejaeger et al. [50]. The features include project sequencing-related features, features that are not expected to be accessible at the time of first estimation highly correlated features, and features with only one value. Table XIII provides the full description of the ISBSG dataset.

11) Telecom dataset: The Telecom dataset contains information on 18 software development initiatives for a UK telecommunication product. The dataset version utilized in this investigation has four attributes. However, since the other three variables are not available at the time of the work estimation, just the number of files attribute is employed. Table XIV provides the full description of the Telecom dataset.

12) Finnish dataset: The TIEKE organization obtained the Finnish dataset from nine Finnish businesses. Initially, 40 records were obtained, however, because of missing values in some of the attributes of two projects (Kitchenham and Kansala 1993), their data were eliminated, leaving 38 records for analysis. This dataset has nine attributes, each of which has a size measured in function points. Table XV provides the full description of the Finish Dataset.

13) Edusoft dataset: For estimating the time and effort needed for software development using a real-world dataset compiled by Edusoft Consultant Ltd. Noteworthy features of this dataset include task history ID, project ID, client id, task types, task priority, task overall state, total working time in hours, etc. 2000 samples of real-time data make up our dataset.

IV. DISCUSSION

A. What ML Approaches are Employed in SEE Studies?

The process of estimating the time, resources, and cost needed to accomplish a software development project is known as software effort estimation. A particular kind of regression issue is software effort estimation. Regression attempts to forecast a continuous numerical number, in this case, the amount of work needed, using information from the input (such as project size, complexity measures, and historical data). Various project parameters and variables that affect the amount of work necessary for development could be included in the input features.

To estimate the SEE, the following ML techniques were utilized either alone or in combination with other (ML and nonML) estimation techniques. Artificial neural network
### TABLE III. CHINA DATASET DESCRIPTION

<table>
<thead>
<tr>
<th>Sl. no</th>
<th>Features</th>
<th>Details about the features</th>
<th>Data Type</th>
<th>Feature Selection</th>
<th>Mean</th>
<th>Std Dev</th>
<th>Min</th>
<th>Max</th>
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<td>Continuous in-integer</td>
<td>AAP</td>
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<td>0</td>
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<td>105</td>
<td>0</td>
<td>952</td>
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<td>6</td>
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<td>Function points of internal logical files</td>
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<td>Added</td>
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<td>210</td>
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<td>2955</td>
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<td>7</td>
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<td>Function points of external interface added</td>
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<td>PDR_AFP</td>
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<td>85</td>
<td>0</td>
<td>1572</td>
</tr>
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<td>830</td>
<td>0</td>
<td>13580</td>
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<td>Continuous in-integer</td>
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<td>Productivity delivery rate(adjusted function points)</td>
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<td>Productivity delivery rate(Unadjusted function points)</td>
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### TABLE IV. MAXWELL DATASET DESCRIPTION

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<th>Standard Dev</th>
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<th>Max</th>
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<td>The software development effort is measured in person-months</td>
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</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>T1</td>
<td>Teamexp</td>
<td>Continuous</td>
<td></td>
<td>3.8</td>
<td>2.0</td>
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<td>10</td>
</tr>
<tr>
<td>2</td>
<td>T2</td>
<td>Devteam</td>
<td>Ratio</td>
<td></td>
<td>2.6</td>
<td>2.4</td>
<td>1</td>
<td>23</td>
</tr>
<tr>
<td>3</td>
<td>T3</td>
<td>Average_Actors</td>
<td>Ratio</td>
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<td>47.5</td>
<td>15.7</td>
<td>1</td>
<td>2000</td>
</tr>
<tr>
<td>4</td>
<td>T4</td>
<td>Testpages</td>
<td>Radio</td>
<td></td>
<td>98.6</td>
<td>218.4</td>
<td>0</td>
<td>1820</td>
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<tr>
<td>5</td>
<td>T5</td>
<td>Anim</td>
<td>Radio</td>
<td></td>
<td>14.64</td>
<td>66.59</td>
<td>0</td>
<td>611</td>
</tr>
<tr>
<td>6</td>
<td>T6</td>
<td>Audio/Video</td>
<td>Radio</td>
<td></td>
<td>4.8</td>
<td>4.98</td>
<td>0</td>
<td>35</td>
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</table>

TABLE X. TUKUTUKU DATASET DESCRIPTION

<table>
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<th>Details about the features</th>
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<td>1</td>
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<td>Min</td>
<td>Continuous</td>
<td>87.475</td>
<td>228.7597</td>
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<td>1586.0</td>
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TABLE XI. UCP DATASET DESCRIPTION

<table>
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<th>Std dev</th>
<th>Min</th>
<th>Max</th>
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</thead>
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<td>0.90</td>
<td>0</td>
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<tr>
<td>2</td>
<td>Average_Actors</td>
<td>0.94</td>
<td>0.88</td>
<td>0</td>
<td>2</td>
</tr>
<tr>
<td>3</td>
<td>Complex_Actors</td>
<td>2.62</td>
<td>1.50</td>
<td>0</td>
<td>6</td>
</tr>
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<td>4</td>
<td>Simple_UC</td>
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<td>2.89</td>
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<tr>
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<td>Average_UC</td>
<td>15.84</td>
<td>5.37</td>
<td>3</td>
<td>30</td>
</tr>
<tr>
<td>6</td>
<td>Complex_UC</td>
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<td>4.43</td>
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<td>27</td>
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<td>7</td>
<td>T1</td>
<td>1.35</td>
<td>2.02</td>
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<td>5</td>
</tr>
<tr>
<td>8</td>
<td>T2</td>
<td>2.3</td>
<td>2.37</td>
<td>0</td>
<td>5</td>
</tr>
<tr>
<td>9</td>
<td>T3</td>
<td>2.92</td>
<td>2.37</td>
<td>0</td>
<td>5</td>
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<tr>
<td>10</td>
<td>T4</td>
<td>3.14</td>
<td>2.43</td>
<td>0</td>
<td>5</td>
</tr>
<tr>
<td>11</td>
<td>T5</td>
<td>3.71</td>
<td>2.06</td>
<td>0</td>
<td>5</td>
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<td>12</td>
<td>T6</td>
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<td>1.01</td>
<td>0</td>
<td>5</td>
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<td>T7</td>
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<td>1.55</td>
<td>0</td>
<td>5</td>
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<tr>
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<td>T8</td>
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<td>1.25</td>
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<td>15</td>
<td>T9</td>
<td>6558.72</td>
<td>664.23</td>
<td>5775</td>
<td>7970</td>
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</table>

(ANN), Support vector machine (SVM), Bayesian network (BN), K-nearest neighbors (kNNs), Decision tree (DT), Genetic programming (GP), Classification and regression tree (CART), CBR, Random forest (RF).

B. Which Datasets are most Commonly Utilized in SEE Research? (Research Question 1)

The selected studies made use of around 15 different datasets. We look for datasets that have been used in at least one study. We showed a literature review in Table II. The NASA dataset is one of the most extensively used datasets in the SEE literature, and it has been utilized in several research. Most of the datasets related to SEE are provided by different software companies. A few datasets come from different sources. The sources of these datasets are listed below.

China dataset basically focuses on function points. The Maxwell dataset was obtained from a Finnish commercial bank so anyone can work with bank data if it needs for his/her research. The multinational American company Computer Sciences Corporation provided the Kitchenham dataset. Ten Canadian organizations provided data for the Desharnais dataset. The NASA93 dataset represents 14 distinct application types and was gathered by NASA from five of its development centers. The Albrecht dataset was created using third-generation programming languages such as COBOL, PL1, and database management languages. The ISBSG Repository now houses software projects acquired from various global software development firms. The Telecom dataset was created by software development initiatives on a telecommunication product used in the United Kingdom. The Tukutuku dataset contains audio, video, animation, and webpages, so it might be useful for candidates who work with graphical data. The TIEKE group gathered the Finnish dataset from nine enterprises in Finland. Edusoft dataset can be used to estimate the time and effort required for software development using a real-world dataset provided by Edusoft Consultant Ltd.

C. Performance Evaluation Matrix (Research Question 2)

In a regression problem, the effectiveness of a predictive model that seeks to estimate a continuous target variable is evaluated using evaluation metrics. Several measures have been established and used for assessing the accuracy of a prediction model in the literature on software work estimation. Prediction error (or absolute error), which is the difference between the anticipated value and the actual value, serves as the foundation for these metrics in most cases [45]. Before it was discovered that they are biased towards underestimates and behave differently when comparing different prediction models, the Mean of
TABLE XII. NASA93 DATASET DESCRIPTION

<table>
<thead>
<tr>
<th>SL No</th>
<th>Features</th>
<th>Details about feature</th>
<th>Mean</th>
<th>Std dev</th>
<th>Min</th>
<th>Max</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>Rely</td>
<td>Requires software reliability</td>
<td>1.11</td>
<td>0.23</td>
<td>0.88</td>
<td>1.40</td>
</tr>
<tr>
<td>2</td>
<td>Data</td>
<td>Size of the application database</td>
<td>1.00</td>
<td>0.07</td>
<td>0.94</td>
<td>1.16</td>
</tr>
<tr>
<td>3</td>
<td>Cplx</td>
<td>Complexity of the product</td>
<td>1.18</td>
<td>0.15</td>
<td>0.85</td>
<td>1.65</td>
</tr>
<tr>
<td>4</td>
<td>Time</td>
<td>Run-time performance constraints</td>
<td>1.13</td>
<td>0.20</td>
<td>1.0</td>
<td>1.66</td>
</tr>
<tr>
<td>5</td>
<td>Stor</td>
<td>Memory constraints</td>
<td>1.13</td>
<td>0.19</td>
<td>1.0</td>
<td>1.36</td>
</tr>
<tr>
<td>6</td>
<td>Virt</td>
<td>Volatility of the virtual machine environment</td>
<td>0.92</td>
<td>0.09</td>
<td>0.87</td>
<td>1.15</td>
</tr>
<tr>
<td>7</td>
<td>Turn</td>
<td>Required turnaround time</td>
<td>0.96</td>
<td>0.09</td>
<td>0.87</td>
<td>1.15</td>
</tr>
<tr>
<td>8</td>
<td>Acap</td>
<td>Analyst capability</td>
<td>0.89</td>
<td>0.09</td>
<td>0.71</td>
<td>1.00</td>
</tr>
<tr>
<td>9</td>
<td>Aexp</td>
<td>Application Experience</td>
<td>0.93</td>
<td>0.06</td>
<td>0.82</td>
<td>1.13</td>
</tr>
<tr>
<td>10</td>
<td>Pcap</td>
<td>Software engineer capability</td>
<td>0.91</td>
<td>0.10</td>
<td>0.90</td>
<td>1.00</td>
</tr>
<tr>
<td>11</td>
<td>Vexp</td>
<td>Virtual machine experience</td>
<td>1.00</td>
<td>0.08</td>
<td>0.90</td>
<td>1.21</td>
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<tr>
<td>12</td>
<td>Lexp</td>
<td>Programming language experience</td>
<td>0.97</td>
<td>0.05</td>
<td>0.95</td>
<td>1.14</td>
</tr>
<tr>
<td>13</td>
<td>Modp</td>
<td>Application of software engineering methods</td>
<td>0.98</td>
<td>0.09</td>
<td>0.82</td>
<td>1.24</td>
</tr>
<tr>
<td>14</td>
<td>Tool</td>
<td>Use of software tools</td>
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<td>0.09</td>
<td>0.83</td>
<td>1.24</td>
</tr>
<tr>
<td>15</td>
<td>Sced</td>
<td>Required development schedule</td>
<td>1.04</td>
<td>0.04</td>
<td>1.00</td>
<td>1.08</td>
</tr>
<tr>
<td>16</td>
<td>Loc</td>
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<td>133.6</td>
<td>0.90</td>
<td>980.0</td>
</tr>
<tr>
<td>17</td>
<td>actual</td>
<td></td>
<td>624.41</td>
<td>1135.93</td>
<td>8.40</td>
<td>8211.00</td>
</tr>
</tbody>
</table>

Magnitude of Relative Error (MMRE), Mean of Magnitude of Relative Error Relative to Estimate (MEMRE), and Prediction at Level 1 (Pred(l)) were the three most widely used metrics. As a result, their use is discouraged, and future studies should rely on standardized measurements such as the Sum of Absolute Errors (SAE), Mean Absolute Error (MAE), and Standard Accuracy (SA) that are not skewed towards underestimates or overestimates. A technique is regarded as better ML if it has more accuracy, for example, a lower mean magnitude of relative error (MMRE) number. For example, if model A is compared to model B, and A has a lower MMRE value in the majority of investigations, we can argue that model A outperforms model B. We may conclude that MSE, MRE, and MMRE are the most often used performance evaluation metrics based on the literature review Table II mentioned in Section 2.

V. CONCLUSION

In this study, we examined the datasets used for software effort assessment and discovered 15 widely used SEE datasets. We've talked in depth about these datasets. Based on our study of the most recent article, we discover that the NASA dataset is the most often used dataset for software effect estimation. The majority of these datasets also lacked timing details. Since software engineering is an ever-evolving field, it is crucial that SEE records include dates for important events like project launches and wrap-ups. This would allow researchers and practitioners to better understand project timelines and make informed decisions about future development strategies. It was unclear in numerous cases whether datasets were obtained from a single company or from several. Given the continuous discussion over which datasets produce the best prediction accuracy results, it seems only right that this information be disclosed alongside datasets made available for modeling purposes. We also discover that the most popular machine learning techniques are Decision trees (DT), K-nearest neighbors (kNNs), Bayesian networks (BN), Support vector machines (SVM), and Artificial neural networks (ANN). On the other hand, Mean Absolute Error (MAE), Mean Squared Error (MSE), and Root Mean Squared Error (RMSE) are the most commonly utilized performance evaluation matrices for software effort estimation.

One limitation of our paper is that we did not build those datasets using alternative machine learning algorithms, which would have allowed us to provide a more detailed description of the dataset. In the future, we will experiment with these datasets using machine learning methods in order to better visualize the results of the evaluation matrix and other pertinent information.

TABLE XIII. ISBSG2021 DATASET DESCRIPTION

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<td>Output count</td>
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<td>171.12</td>
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<td>Enquiry count</td>
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<td>136.59</td>
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<td>952.0</td>
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<td>4</td>
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<td>166.42</td>
<td>0</td>
<td>1252.0</td>
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<tr>
<td>5</td>
<td>Interface count</td>
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<td>90.59</td>
<td>0</td>
<td>907.0</td>
</tr>
<tr>
<td>6</td>
<td>Developer</td>
<td>2182.82</td>
<td>2097.72</td>
<td>7.00</td>
<td>6610.0</td>
</tr>
<tr>
<td>7</td>
<td>Functional size</td>
<td>8720.47</td>
<td>947.46</td>
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<td>16148.0</td>
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<td>8</td>
<td>Value adjustment</td>
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<td>0.08</td>
<td>0.65</td>
<td>1.29</td>
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<td>9</td>
<td>Normalised Work Effort Level 1</td>
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<td>1333.10</td>
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TABLE XIV. TELECOM DESCRIPTION

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<td>Effort</td>
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<td>1115.54</td>
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TABLE XV. FINISH DATASET DESCRIPTION

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<th>Std Dev</th>
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</thead>
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<td>3</td>
</tr>
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<td>AR</td>
<td>2.23</td>
<td>1.30</td>
<td>1</td>
<td>5</td>
</tr>
<tr>
<td>3</td>
<td>Function Points</td>
<td>763.58</td>
<td>510.83</td>
<td>65</td>
<td>1814</td>
</tr>
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<td>CO</td>
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<td>2.73</td>
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<td>Effort</td>
<td>7678.29</td>
<td>7135.28</td>
<td>460</td>
<td>26670</td>
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assess the performance improvement of a software company

UI/IAR/02/2019-20/SE/06.

REFERENCES


Employee Attrition Prediction using Nested Ensemble Learning Techniques

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Abstract—In many industries, including the IT industry, rising employee attrition is a major concern. Hiring a candidate for an unsuitable job because of issues with the employment process can lead to employee attrition. Thus, enhancing the employment process would reduce the attrition rate. This paper aims to investigate the effect of ensemble learning techniques on enhancing the employment process by predicting employee attrition. This paper applied a two-layer nested ensemble model to the IBM HR Analytics Employee Attrition & Performance dataset. The performance of this model was compared to that of the random forest (RF) algorithm as a baseline for comparison. The results showed that the proposed model outperformed the baseline algorithm. The RF model achieved an accuracy of 94.2417%, an F1-score of 94.2%, and an AUC of 98.4%. However, the proposed model had the highest performance. It outperformed with an accuracy of 94.5255%, an F1-score of 94.5%, and an AUC of 98.5%. The performance of the proposed model was compared with that of the baseline comparison algorithm by using a paired t-test. According to the paired t-test, the performance of the proposed model was statistically better than that of the baseline comparison algorithm at the significance level of 0.05. Thus, the two-layer nested ensemble model improved the employee attrition prediction.

Keywords—Nested ensemble learning; employee attrition; machine learning; employment process

I. INTRODUCTION

The workforce is a crucial asset of an organization due to the strong positive correlation between the success of the organization and its employees’ performance. The performance of employees can be affected by human resource management practices, such as the selection process [1], which aims to choose suitable candidates for specific job vacancies based on job-related criteria [2]. The wrong selection of candidates can lead to something called employee attrition, which is described as a large decrease in the labor force. Employee attrition occurs for several reasons [3], such as professional reasons, personal reasons, workplace challenges, and poor employee-job fit [4]. The last reason has a significant relationship with the employment process. It is about hiring a candidate for an inappropriate job because of issues in the employment process [5], or, in other words, the wrong selection of candidates. While a poor employment process can increase the chance of employee attrition, taking steps to reduce the rate of attrition can improve the employment process. Thus, employee attrition can be considered an indicator of the effectiveness of the employment process.

Some literature has studied employee attrition using machine learning (ML) techniques including but not limited to the works [6]–[15]. They conducted their experiments on the same dataset, the IBM HR Analytics Employee Attrition & Performance dataset. Three of them [6], [10], [13] used ensemble techniques in their experiments. To the best of the authors’ knowledge, this study is the first to apply a nested ensemble technique to the IBM HR Analytics Employee Attrition & Performance dataset.

The problem addressed in this paper is the increasing rate of employee attrition [16], which is a major concern in several sectors, such as the IT industry. Employee attrition impacts an organization’s performance. It drains many budgets and affects employee satisfaction [3]. Thus, it is necessary to control the rate of attrition [16]. This paper seeks to find the answer of the following question: What is the effect of using nested ensemble learning techniques to improve employee attrition prediction when applied to an employee dataset that has been optimized by a feature selection technique? Additionally, this study aims to investigate the effect of using nested ensemble learning techniques to improve employee attrition prediction when applied to an employee dataset that has been optimized by a feature selection technique.

The rest of this paper is organized as follows: Section II gives a background on the subject and an overview of the related work. Section III explains the methodology. Section IV presents the results and discussion. Section V summarizes the findings of the research and recommends future research directions.

II. RELATED WORK

This paper used the IBM HR Analytics Employee Attrition & Performance dataset. It was developed by IBM’s data scientists and made publicly available on the Kaggle website [17]. It was chosen because its attributes can be obtained easily from HR departments and its characteristics can simulate real-world HR issues [18]. Several studies conducted their research on this dataset using ML techniques. The current research adopts some of them as related work and reviews them in this section. Table I shows some information about the related work.

Each study was summarized in a single paragraph by reviewing the methods adopted for balancing data, selecting features, applying algorithms, and evaluating performance. Then, each paragraph was concluded by evaluating the performance of the research model using performance evaluation metrics.

The work [6], data balancing was not applied and was considered for future work. As well, feature selection techniques
were not used. The research built models using several algorithms, such as logistic regression (LR), decision tree (DT), random forest (RF), naive bayes (NB), and artificial neural networks (ANN). It used these algorithms to build a voting ensemble based on the majority vote strategy (hard vote). The performance of the models was measured by accuracy, F1-score, and AUC metrics. The best performance in terms of accuracy and AUC went to LR, which scored 87.96% and 85.01%, respectively. The highest F1-score reached 33.78%, which was accomplished by NB.

The work [7] did not apply data balancing, feature selection methods, or ensemble learning techniques. It applied lightGBM and TabNet algorithms. It evaluated the performance of the models using an accuracy metric. LightGBM had the best performance, with an accuracy of 85.3%.

In [8], neither data balancing nor feature selection methods were mentioned. It did not use ensemble learning techniques. Six ML algorithms were applied, which were LR, DT, SVM, NB, k-nearest neighbors (KNN), and RF. The accuracy metric was used to evaluate the performance of the models. SVM achieved the highest accuracy, with a value of 86.77%.

The work [9] used SMOTE to balance the dataset. It did not use feature selection methods to select appropriate features. Likewise, it did not apply ensemble learning techniques. However, it applied four algorithms named the extra tree classifier (ETC), SVM, LR, and DT. The performance of the models was estimated using four measures: accuracy, precision, recall, and F1-score. The performance of ETC was the highest, scoring 93% in all four measures.

The work [10] did not mention handling the imbalanced dataset. However, it applied a filter feature selection technique called information gain. As well, it applied bagging ensemble learning. Eight algorithms were used: RF, NB, LR, SVM, KNN, AdaBoost, DT, and logistic model tree (LMT). The performance was evaluated using four metrics: accuracy, precision, recall, and F1-measure. The performance of LMT was the best in terms of accuracy and recall. It achieved 86.94% and 86.90%, respectively. The performance of RF was the best in terms of precision, which was 85.50%. The performance of SVM was the best in terms of the F1-measure, which was 85.10%.

In [11], data balancing techniques were not mentioned. It used a wrapper feature selection technique named max-out, which was a proposed method. The ensemble learning technique was not applied. This work applied just one algorithm, LR. Accuracy, precision, recall, and F1-score measures were used to evaluate the performance of the models. It had an accuracy of 81%, a precision of 43%, a recall of 82%, and an F1-score of 56%.

In [12], both data balancing techniques and feature selection techniques were not mentioned. The ensemble learning technique was not implemented. Six algorithms were applied to the dataset: classification and regression tree, RF, LR, SVM, KNN, and NB. The performance of the models was assessed using accuracy and AUC. The best performance in terms of accuracy went to RF with 85.12%, while the best performance in terms of AUC went to LR with 80.85%.

Data balancing in [13] was not applied, and it was considered for future work. Two feature selection techniques were used: recursive feature elimination, a wrapper method, and selectKBest, a filter method. It used two ensemble learning techniques: the stacking ensemble learning technique and the voting ensemble technique based on the majority vote strategy (hard vote). Furthermore, eight algorithms were implemented, which were DT, LR, SVM, RF, XGBoost, DNN, long short-term memory (LSTM), and convolutional neural networks (CNN). Accuracy and F1-score were used to evaluate the performance of the models. The voting ensemble technique achieved the highest accuracy with 96%, while RF achieved the highest F1-score with 82.8%.

In [14], data balancing, feature selection methods, and ensemble learning techniques were not applied. It applied eight algorithms, which were Gaussian NB, Bernoulli NB, LR, KNN, DT, RF, SVM, and linearSVM. The performance of the models was evaluated using five metrics: accuracy, precision, specificity, recall, and F1-score. The performance of SVM was the best in terms of accuracy, precision, and specificity. It achieved 87.9%, 87.9%, and 99.4%, respectively. In addition, the performance of Gaussian NB was the best in terms of recall and F1-score. It obtained 54.1% and 44.6%, respectively.

The final work, [15], did not mention handling imbalanced data. Moreover, it did not use feature selection techniques to choose the relevant features. Besides, it did not apply ensemble learning techniques. However, it built CLARA, a system designed to enhance employee retention. It compared

<table>
<thead>
<tr>
<th>Ref</th>
<th>Paper’s Name</th>
<th>Year</th>
<th>Publication’s Name</th>
<th>Pub. Type</th>
<th>Pub. Rank</th>
</tr>
</thead>
<tbody>
<tr>
<td>[7]</td>
<td>Counterfactual Explanation Trees: Transparent and Consistent Actionable Recourse with Decision Trees</td>
<td>2022</td>
<td>International Conference on Artificial Intelligence and Statistics (AISTATS)</td>
<td>Conference</td>
<td>A</td>
</tr>
</tbody>
</table>

TABLE I. A BRIEF OVERVIEW OF THE RELATED WORK
CLARA’s performance to six algorithms: RF, XGBoost, SVM, spectral clustering, standalone k-means clustering, and standalone frequent pattern mining. It used precision to evaluate the performance. The performance of the proposed system achieved a precision of approximately 70%.

III. METHODOLOGY

The idea of the proposed solution is to apply a nested ensemble model to predict employee attrition. To this end, the research prepared the dataset and used the nested ensemble model. All experiments in this paper were done using Waikato Environment for Knowledge Analysis (WEKA) platform version 3.8.6.

A. The Preprocessing Phase

This section shows the steps that were done to prepare the dataset. The preprocessing phase aims to prepare the dataset to be ready for applying ML algorithms in order to obtain the best results. In this research, different steps were applied to deal with several issues that may affect the ML performance, such as having values that are missing, duplicated, or irrelevant, or having features with various scales.

1) Dataset: The dataset used is the IBM HR Analytics Employee Attrition & Performance dataset, a public dataset generated by IBM data scientists and available on Kaggle [17]. It consists of 1470 instances and 35 features. Table II shows a brief description of the dataset. The table outlines the data type of each feature, data measurement scales, and descriptions of the feature’s values (the number between brackets indicates the number of employees who share the same value).

The feature Attrition is a binary categorical variable that has two values: No and Yes. The value No indicates employees who did not leave the company. The value Yes indicates employees who left the company. The dataset contains 1233 on-the-job employees, representing 83.88% of the total employees. As for the rest of the employees, representing 16.12%, they left the company. Thus, the dataset is imbalanced because the minority class represents 16.12% of the entire dataset. For data balancing, the SMOTE technique was used. It was adopted because it has been widely used in similar studies; as well, it was applied in the related work [9].

In this dataset, Attrition was the second attribute. It was set as a class attribute by choosing Attribute as class command after clicking on the Edit button under the Preprocess tab. Therefore, it was moved to the end of the dataset as the last attribute.

2) Checking for missing and duplicate values: After checking the information shown in the Selected attribute panel, the dataset had no missing values. Moreover, it had no duplicate values after checking the values using the RemoveDuplicates filter.

3) Checking for unnecessary attributes: Referring to [19], constant attributes or unique attributes are considered unnecessary attributes that should be dropped. Three features were constant, and one had unique values. The features EmployeeCount, Over18, and StandardHours are constant for all instances. Their respective values were 1, Y, and 80. The feature EmployeeNumber has employees’ identification codes (IDs), which are unique for all instances. Therefore, these four features were dropped because they did not further the research purpose; in other words, they were unnecessary data [19].

4) Data transformation: The purpose of this step is to convert data from one format to another without changing the dataset’s content. It helps to improve the ML algorithms’ performance and enhance the data’s understanding [20]. This research used the OrdinalToNumeric filter to unify the data type to be numeric, except Attrition, which is the class attribute.

5) Data scaling: Having features with several scales, for instance, the attribute MonthlyRate that ranges in thousands and Age that ranges in tens, can lead to the numerical overflow problem, which is computing numbers that are very small or very big. Thus, it is recommended to put all features on the same scale in order to give them equal weights [21]. This research used the Normalize filter to unify all the feature scales to start with 0 and end with 1.

6) The dataset optimization: For optimizing the dataset, the feature selection technique was used, and the information gain-based feature selection method was chosen as it was in the related work [10].

B. The Proposed Model

The proposed model is a nested ensemble learning technique. It is a special type of ensemble learning technique that combines several ensemble methods within one model in order to improve performance [22]. It contains multiple ensemble learning models; each one is inside of another one at different layers. The idea of this model is to use ensemble techniques in the classifier part instead of simple algorithms. In this research, a two-layer nested ensemble model was applied. It used a stacking model in the first layer and both stacking and voting models in the second layer. Moreover, it used traditional algorithms such as NB, SVM, RF, and LR as base-level algorithms of the model’s ensemble models. Fig. 1 shows the architecture of the proposed model.

The following ML algorithms were used to build the proposed model:

1) Naive Bayes (NB): It is a probabilistic algorithm that uses probability rules to make predictions [23]. It can deal with different types of data; therefore, it has different models, such as Gaussian, Multinomial, and Bernoulli [24].
TABLE II. THE DATASET DESCRIPTION

<table>
<thead>
<tr>
<th>No</th>
<th>Attribute</th>
<th>Data Type</th>
<th>Scale</th>
<th>Values Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>Age</td>
<td>Numeric</td>
<td>Ratio</td>
<td>Min = 18, Max = 60, Mean = 36.924, StdDev = 9.135</td>
</tr>
<tr>
<td>2</td>
<td>Attrition</td>
<td>Textual</td>
<td>Nominal</td>
<td>Yes (217), No (1233)</td>
</tr>
<tr>
<td>3</td>
<td>BusinessTravel</td>
<td>Textual</td>
<td>Nominal</td>
<td>Non-Travel (1580), Travel_Rarely (1043), Travel_Frequently (277)</td>
</tr>
<tr>
<td>4</td>
<td>DailyRate</td>
<td>Numeric</td>
<td>Ratio</td>
<td>Min = 102, Max = 1499, Mean = 802.486, StdDev = 401.509</td>
</tr>
<tr>
<td>5</td>
<td>Department</td>
<td>Textual</td>
<td>Nominal</td>
<td>Sales (446), Research &amp; Development (901), Human Resources (63)</td>
</tr>
<tr>
<td>6</td>
<td>DistanceFromHome</td>
<td>Numeric</td>
<td>Ratio</td>
<td>Min = 1, Max = 29, Mean = 9.193, StdDev = 8.107</td>
</tr>
<tr>
<td>7</td>
<td>Education</td>
<td>Numeric</td>
<td>Ordinal</td>
<td>1 'Below College' (170), 2 'College' (282), 3 'Bachelor' (372), 4 'Master' (398), 5 'Doctor' (48)</td>
</tr>
<tr>
<td>8</td>
<td>EducationField</td>
<td>Textual</td>
<td>Nominal</td>
<td>Life Sciences (606), Medical (464), Marketing (159), Technical Degree (132), Human Resources (27), Other (82)</td>
</tr>
<tr>
<td>9</td>
<td>EmployeeCount</td>
<td>Numeric</td>
<td>Ratio</td>
<td>Min = 1 (1470)</td>
</tr>
<tr>
<td>10</td>
<td>EmployeeNumber</td>
<td>Numeric</td>
<td>Nominal</td>
<td>1470 unique identification codes (IDs), start with 1 and end with 2008. They can be classified as the following: IDs &lt;= 500 (377), IDs range between 501 and 1000 (340), IDs range between 1001 and 1500 (348), IDs range between 1501 and 2000 (357), IDs &gt; 2000 (48)</td>
</tr>
<tr>
<td>11</td>
<td>EnvironmentSatisfaction</td>
<td>Numeric</td>
<td>Ordinal</td>
<td>1 'Low' (284), 2 'Medium' (287), 3 'High' (453), 4 'Very High' (446)</td>
</tr>
<tr>
<td>12</td>
<td>Gender</td>
<td>Textual</td>
<td>Nominal</td>
<td>Female (588), Male (882)</td>
</tr>
<tr>
<td>13</td>
<td>HourlyRate</td>
<td>Numeric</td>
<td>Ratio</td>
<td>Min = 30, Max = 100, Mean = 65.891, StdDev = 20.329</td>
</tr>
<tr>
<td>14</td>
<td>JobInvolvement</td>
<td>Numeric</td>
<td>Ordinal</td>
<td>1 'Low' (83), 2 'Medium' (375), 3 'High' (868), 4 'Very High' (144)</td>
</tr>
<tr>
<td>15</td>
<td>JobLevel</td>
<td>Numeric</td>
<td>Ordinal</td>
<td>1 (543), 2 (534), 3 (218), 4 (100), 5 (49)</td>
</tr>
<tr>
<td>16</td>
<td>JobRole</td>
<td>Textual</td>
<td>Nominal</td>
<td>Sales Executive (326), Research Scientist (292), Laboratory Technician (259), Manufacturing Director (145), Healthcare Representative (131), Manager (102), Sales Representative (83), Research Director (80), Human Resources (52)</td>
</tr>
<tr>
<td>17</td>
<td>JobSatisfaction</td>
<td>Numeric</td>
<td>Ordinal</td>
<td>1 'Low' (284), 2 'Medium' (287), 3 'High' (453), 4 'Very High' (446)</td>
</tr>
<tr>
<td>18</td>
<td>MaritalStatus</td>
<td>Textual</td>
<td>Nominal</td>
<td>Single (431), Married (609), Divorced (327)</td>
</tr>
<tr>
<td>19</td>
<td>MonthlyIncome</td>
<td>Numeric</td>
<td>Ratio</td>
<td>Min = 1009, Max = 9999, Mean = 6502.931, StdDev = 4707.957</td>
</tr>
<tr>
<td>20</td>
<td>MonthlyRate</td>
<td>Numeric</td>
<td>Ratio</td>
<td>Min = 2094, Max = 26999, Mean = 14313.103, StdDev = 7117.786</td>
</tr>
<tr>
<td>21</td>
<td>NumCompaniesWorked</td>
<td>Numeric</td>
<td>Ratio</td>
<td>Min = 0, Max = 9, Mean = 2.693, StdDev = 2.498</td>
</tr>
<tr>
<td>22</td>
<td>Over18</td>
<td>Textual</td>
<td>Nominal</td>
<td>Yes (1470)</td>
</tr>
<tr>
<td>23</td>
<td>OverTime</td>
<td>Textual</td>
<td>Nominal</td>
<td>Yes (410), No (1054)</td>
</tr>
<tr>
<td>24</td>
<td>PercentSalaryHike</td>
<td>Numeric</td>
<td>Ratio</td>
<td>Min = 11, Max = 25, Mean = 15.21, StdDev = 3.86</td>
</tr>
<tr>
<td>25</td>
<td>PerformanceRating</td>
<td>Numeric</td>
<td>Ordinal</td>
<td>1 'Low' (0), 2 'Good' (0), 3 'Excellent' (1224), 4 'Outstanding' (226)</td>
</tr>
<tr>
<td>26</td>
<td>RelationshipSatisfaction</td>
<td>Numeric</td>
<td>Ordinal</td>
<td>1 'Low' (276), 2 'Medium' (303), 3 'High' (459), 4 'Very High' (432)</td>
</tr>
<tr>
<td>27</td>
<td>StandardHours</td>
<td>Numeric</td>
<td>Ratio</td>
<td>30 (1470)</td>
</tr>
<tr>
<td>28</td>
<td>StockOptionLevel</td>
<td>Numeric</td>
<td>Ordinal</td>
<td>0 (631), 1 (596), 2 (158), 3 (85)</td>
</tr>
<tr>
<td>29</td>
<td>TotalWorkingYears</td>
<td>Numeric</td>
<td>Ratio</td>
<td>Min = 0, Max = 40, Mean = 11.28, StdDev = 7.781</td>
</tr>
<tr>
<td>30</td>
<td>TrainingTimesLastYear</td>
<td>Numeric</td>
<td>Ratio</td>
<td>Min = 0, Max = 6, Mean = 2.799, StdDev = 1.289</td>
</tr>
<tr>
<td>31</td>
<td>WorkLifeBalance</td>
<td>Numeric</td>
<td>Ordinal</td>
<td>1 'Bad' (80), 2 'Good' (344), 3 'Better' (893), 4 'Best' (153)</td>
</tr>
<tr>
<td>32</td>
<td>YearsAtCompany</td>
<td>Numeric</td>
<td>Ratio</td>
<td>Min = 0, Max = 40, Mean = 7.008, StdDev = 6.127</td>
</tr>
<tr>
<td>33</td>
<td>YearsInCurrentRole</td>
<td>Numeric</td>
<td>Ratio</td>
<td>Min = 0, Max = 18, Mean = 4.229, StdDev = 3.623</td>
</tr>
<tr>
<td>34</td>
<td>YearsSinceLastPromotion</td>
<td>Numeric</td>
<td>Ratio</td>
<td>Min = 0, Max = 15, Mean = 2.188, StdDev = 3.222</td>
</tr>
<tr>
<td>35</td>
<td>YearsWithCurrManager</td>
<td>Numeric</td>
<td>Ratio</td>
<td>Min = 0, Max = 17, Mean = 4.123, StdDev = 3.588</td>
</tr>
</tbody>
</table>

2) Support Vector Machines (SVM): It is a very powerful and popular algorithm in machine learning. In addition, it works well with complex datasets of small or medium size [25].

3) Random Forests (RF): It is a widely used algorithm consisting of numerous decision trees that learn from subsets of a training dataset [26]. It is a special type of bagging ensemble technique that uses bagging to combine tree models’ predictions [27].

4) Stacking ensemble learning technique: Stacking, short for stacked generalization, is an approach that uses different types of algorithms to make their predictions, which they called base-level algorithms. Then, it uses an algorithm to predict the outcome. This algorithm is called a meta-level algorithm [25]. Recently, several studies that applied the stacking ensemble technique have used logistic regression (LR) as a meta-level algorithm [28]. They prefer using LR because of its speed of training and the small number of parameters that it requires [29]. In this research, the algorithms NB, SVM, and RF were used as base-level algorithms, while the algorithm LR was used as a meta-level algorithm.

5) Voting ensemble learning technique: Voting is an approach that uses different types of algorithms to make its prediction. It is similar to the stacking technique except it predicts the final outcome by using the majority voting approach or the averaging approach [25]. In this research, the algorithms NB, SVM, and RF were used as its algorithms. Furthermore, it predicts the final outcome by using the majority voting approach.

C. Performance Evaluation

1) Accuracy: It is a metric that calculates the percentage of correct predictions out of the total number of predictions. It is appropriate to know to what extent the model can predict correctly. Mathematically, accuracy = (TP + TN) / (TP + TN + FN + FP) [30], as shown in Fig. 2.

2) Precision: It is a metric that calculates the percentage of correct predictions of positive instances out of the total number of predictions of positive instances. It is appropriate to know to what extent the model can exclude any instance that actually does not belong to the positive class. Mathematically, precision = TP / (TP + FP) [30], as shown in Fig. 2.

3) Recall: It is a metric that calculates the percentage of correct predictions of positive instances out of the total of actual positive instances. It is appropriate to know to what extent the model can include any instance that actually belongs to the positive class. Mathematically, recall = TP / (TP + FN) [30], as shown in Fig. 2.
4) **F1-score:** F1-score, or F1-measure, is a metric that measures the harmonic mean of recall and precision [30]. The harmonic mean gives much weight to small values. Hence, a model can get a high F1-score when both recall and precision are high, or it can get a low score when both recall and precision are low [25]. Mathematically, F1-score = \( 2 \times \frac{\text{recall} \times \text{precision}}{\text{recall} + \text{precision}} \) [30], or F1-score = \( \frac{2 \times \text{TP}}{2 \times \text{TP} + \text{FN} + \text{FP}} \) [25], or simply F1-score = \( \frac{\text{TP}}{\text{TP} + \text{FN} + \text{FP}} \), as shown in Fig. 2.

5) **The Area Under the Curve (AUC):** It is a metric that compares the performance of several models for the same dataset. A model with an AUC above 80% is recommended [31]. The higher the AUC, the better the performance [30].

### D. Cross-Validation Technique

The cross-validation method is a statistical method that is used to evaluate and compare ML models. Its mechanism is based on dividing datasets into several sets: one is for validating an ML model, and the rest is for learning the ML model. It is called cross-validation because the training set and validation set cross over in sequential rounds to validate every data point [32].

The two most popular cross-validation approaches are **K-fold cross-validation** and **Hold-out validation**. Both of them are effective as long as the dataset is balanced. However, when the dataset is imbalanced, some modifications are applied in order to stratify the folds by the class label. So, the stratified keyword would precede the methods’ names. The stratification process splits the data into folds, and each fold has instances with class labels that are similar to the entire dataset. In this case, every fold represents the class in approximately the correct proportions. Hence, the stratification technique is appropriate for imbalanced datasets [32].

![Fig. 2. Illustration of four evaluation metrics derived from the confusion matrix.](image)

Note: The left-hand side clarifies the equations of these metrics, while the right-hand side represents them pictorially.

In this thesis, stratified 10-fold cross-validation was applied because it is recommended for imbalanced datasets [32], [33]. Additionally, the related works [7] and [10] were applied the same technique.

### IV. RESULTS AND DISCUSSION

This research is part of master’s thesis, thus it is limited in time and computer resources. It used the WEKA platform to conduct the research experiments. The scope of this study is limited to the following:

- The research used **IBM HR Analytics Employee Attrition & Performance** dataset because it is a benchmark dataset created by data scientists from IBM, a well-known and reputable company [17]. Further, its features are usually found in real-world employee databases [18]. Besides, it has been widely studied in several research works.
- The research selected its related works on the basis that they used the **IBM HR Analytics Employee Attrition & Performance** dataset, provided they were published in ranked journals or conferences.
- The research studied employee attrition as a criterion to evaluate the effectiveness of the employment process. However, the study did not expand to cover any further criteria.
- The research limited its baseline comparison algorithms to NB, SVM, and RF since they are among the most popular ML techniques [34], especially for binary classification problems [35].
- The research studied nested ensemble techniques by using ensemble models as base-level algorithms. However, the study did not expand to cover any further details of this technique.
- The thesis optimized the dataset by using a feature selection technique.

The proposed model was built as shown in Fig. 1. Its performance was compared with that of NB, SVM, and RF as baseline comparisons, as well as that of other models in related work. The algorithms’ hyperparameters were tuned as displayed in Table III. The findings are shown in Table IV.

![TABLE III. THE TUNING OF THE ALGORITHMS’ HYPERPARAMETERS](image)

<table>
<thead>
<tr>
<th>Algorithm</th>
<th>Hyperparameters’ Tuning</th>
</tr>
</thead>
<tbody>
<tr>
<td>NB</td>
<td>useKernelEstimator = True</td>
</tr>
<tr>
<td>SVM</td>
<td>C = 2, Kernel = PUK</td>
</tr>
<tr>
<td>RF</td>
<td>numIterations = 680</td>
</tr>
</tbody>
</table>

![TABLE IV. THE PERFORMANCE OF THE BASELINE COMPARISON ALGORITHMS AND THE PROPOSED ENSEMBLE MODEL](image)

<table>
<thead>
<tr>
<th>Algorithm</th>
<th>Accuracy</th>
<th>Precision</th>
<th>Recall</th>
<th>F1-Measure</th>
<th>AUC</th>
</tr>
</thead>
<tbody>
<tr>
<td>NB</td>
<td>90.93%</td>
<td>91.9%</td>
<td>91%</td>
<td>90.9%</td>
<td>95.5%</td>
</tr>
<tr>
<td>SVM</td>
<td>92.78%</td>
<td>92.8%</td>
<td>92.8%</td>
<td>92.8%</td>
<td>92.8%</td>
</tr>
<tr>
<td>RF</td>
<td>94.24%</td>
<td>94.3%</td>
<td>94.2%</td>
<td>94.2%</td>
<td>98.4%</td>
</tr>
<tr>
<td>The proposed model</td>
<td>94.52%</td>
<td>94.5%</td>
<td>94.5%</td>
<td>94.5%</td>
<td>98.5%</td>
</tr>
</tbody>
</table>

* In terms of the performance evaluated by AUC, the proposed model is statistically better than RF (the baseline) at the significance level of 0.05.
The results demonstrated that the performance of RF as a baseline comparison algorithm was the best, with an accuracy of 94.2417%, a precision of 94.3%, a recall of 94.2%, an F1-score of 94.2%, and an AUC of 98.4%. However, the proposed model outperformed the other models, including RF. It exceeded the other models in terms of all metrics. It achieved 94.5255% in accuracy, 98.5% in AUC, and 94.5% in precision, recall, and F1-score.

As shown in Table IV, the best performance for each model, except SVM, was given by the AUC. The performance scores given by accuracy, precision, recall, and F1-measure to NB were within 91%, while AUC gave a performance score that was within 95%. Likewise, the performance scores of RF and the proposed model were within the range of 94%, which were given by accuracy, precision, recall, and F1-measure, whereas AUC gave them performance scores that were within 98%. However, SVM was given performance scores close to 93% by all the metrics.

In order to determine whether the proposed model provided a significant improvement in the employee attrition prediction, its performance was compared with that of the baseline comparison algorithms by using a paired t-test, a common statistical test that is used to compare two sets of values [36]. The performance evaluated by AUC was chosen because the AUC metric compares models for the same dataset in terms of their performance; the higher the AUC, the better the performance [30]. Furthermore, it is strongly recommended to use AUC when the dataset is imbalanced. In determining the test base for the paired t-test, RF was chosen because it is a baseline comparison algorithm and has the best performance among the baseline algorithms. The level of significance was set at 0.05. As shown in Table IV, the RF model and the proposed model had an AUC of 98.4% and 98.5%, respectively. According to the paired t-test, the performance of the proposed model was statistically better than that of RF at the significance level of 0.05. Thus, the two-layer nested ensemble model improved the employee attrition prediction.

Table V shows the comparison between the performance of the proposed model in this work and the models in the related work. It is noted that the proposed model achieved great performance. Its performance exceeded the performance of almost all models.

V. CONCLUSION AND FUTURE WORK

Rising employee attrition rates are a major concern that has an influence on an organization’s performance [3]. It is affected by the success of the employment process, since the improvement of this process reduces the rate of employee attrition [4]. This paper investigated the effect of using nested ensemble learning technique to improve employee attrition prediction when applied to the IBM HR Analytics Employee Attrition & Performance dataset that has been optimized by the information gain-based feature selection method. To the best of the authors’ knowledge, no previous research has applied nested ensemble techniques to the IBM HR Analytics Employee Attrition & Performance dataset.

The experimental results showed that the performance of RF was better than that of NB and SVM. Despite this, the proposed model outperformed RF. The RF model had an accuracy of 94.2417%, an AUC of 98.4%, a precision of 94.3%, and a recall and an F1-score of 94.2%. However, the proposed two-layer nested ensemble model achieved 94.5255% in accuracy, 98.5% in AUC, and 94.5% in precision, recall, and F1-score. In addition, the proposed model was statistically better than that of RF at the significance level of 0.05. To compare the performance of the proposed model in this work with the models in the related work mentioned in Section II, the proposed model showed excellent performance. Its performance exceeded that of all the models. Accordingly, the findings showed the capability of the proposed model to predict employee attrition. Thus, the nested ensemble learning technique assists the employment process and improve employee attrition prediction.

This paper suggests conducting more studies on the nested ensemble technique as future work. The reviewed studies, such as [37] [22], and [38], which applied nested ensemble techniques, focused on applying an ensemble technique as a meta-level algorithm, whereas in this work, the proposed model is a two-layer nested ensemble model that applied ensemble techniques as base-level algorithms and LR as a meta-level algorithm. However, the proposed model approved its capability in prediction as it scored more than 98% in AUC.

TABLE V. THE PERFORMANCE OF THE MODELS IN THE CURRENT WORK AND THE RELATED WORK

<table>
<thead>
<tr>
<th>The Work</th>
<th>Algorithm</th>
<th>Accuracy</th>
<th>Precision</th>
<th>Recall</th>
<th>F1-Measure</th>
<th>AUC</th>
</tr>
</thead>
<tbody>
<tr>
<td>[6]</td>
<td>Voting Classifier</td>
<td>79.25%</td>
<td>N/A</td>
<td>N/A</td>
<td>12.22%</td>
<td>83.83%</td>
</tr>
<tr>
<td>[7]</td>
<td>LightGBM</td>
<td>85.3%</td>
<td>N/A %</td>
<td>N/A %</td>
<td>N/A %</td>
<td>N/A %</td>
</tr>
<tr>
<td>[8]</td>
<td>SVM</td>
<td>86.77 %</td>
<td>N/A %</td>
<td>N/A %</td>
<td>N/A %</td>
<td>N/A %</td>
</tr>
<tr>
<td>[9]</td>
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<td>93%</td>
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</tr>
<tr>
<td>[10]</td>
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<td>83.70%</td>
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<td>77.50%</td>
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<td>[12]</td>
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<td>[13]</td>
<td>Voting Classifier</td>
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<tr>
<td>[14]</td>
<td>Stacking</td>
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<tr>
<td>[15]</td>
<td>NB</td>
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<td>38.6%</td>
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<td>44.6%</td>
<td>N/A%</td>
</tr>
</tbody>
</table>

This Work: The proposed model 94.5255% 94.5% 94.5% 94.5% 98.5%

1 A proposed end-to-end system.

REFERENCES


Bird Detection and Species Classification: Using YOLOv5 and Deep Transfer Learning Models

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Abstract—Bird detection and species classification are important tasks in ecological research and bird conservation efforts. The study aims to address the challenges of accurately identifying bird species in images, which plays a crucial role in various fields such as environmental monitoring, and wildlife conservation. This article presents a comprehensive study on bird detection and species classification using the YOLOv5 object detection algorithm and deep transfer learning models. The objective is to develop an efficient and accurate system for identifying bird species in images. The YOLOv5 model is utilized for robust bird detection, enabling the localization of birds within images. Deep transfer learning (TL) models, including VGG19, Inception V3, and EfficientNetB3, are employed for species classification, leveraging their pre-trained weights and learned features. The experimental findings show that the proposed approach is effective, with excellent accuracy in both bird detection and tasks for species classification. The study showcases the potential of combining YOLOv5 with deep transfer learning models for comprehensive bird analysis, opening avenues for automated bird monitoring, ecological research, and conservation efforts. Furthermore, the study investigated the effects of optimization algorithms, including SGD, Adam, and Adamax, on the performance of the models. The findings contribute to the advancement of bird recognition systems and provide insights into the performance and suitability of various deep transfer learning architectures for avian image analysis.

Keywords—Bird detection; species classification; YOLOv5; deep transfer learning models; automated bird monitoring

I. INTRODUCTION

Birds play a crucial role in maintaining ecological balance. Conducting research on bird species enables us to enhance our understanding of the surrounding world, and the Earth’s ecosystem, and acquire valuable insights into nature. Birds exhibit diverse characteristics, including varying sizes, shapes, and colors, and can be found in different locations. Identifying birds holds immense significance for ornithologists. Environmental scientists frequently rely on birds to gain insights into ecosystems due to their sensitivity to environmental fluctuations. Accurate identification of bird species provides crucial data on environmental conditions.

However, the process of manually collecting and data processing for bird species identification poses a significant challenge for researchers. To mitigate this challenge, the development of automated bird identification systems that collect, process, and classify bird species based on relevant information has emerged as a potential solution. The classification of bird species holds importance in various practical applications, including environmental pollution monitoring [1]. The presence of different bird species within an ecosystem serves various environmental purposes. Presently, image classification has emerged as a prominent domain of study in machine learning (ML) and deep learning (DL) [2]. However, accurately identifying bird species from images presents a complex undertaking due to challenges such as distinguishing between different bird species based on their unique shapes and appearances, accounting for background variations, managing varying lighting conditions in images, and accounting for the birds’ dynamic postures.

The study employed yolov5, an object detection algorithm, to identify bird regions within images. Furthermore, three distinct models, namely VGG19, EfficientNetB3, and InceptionV3, were trained to classify a total of 525 bird species. By leveraging yolov5’s capabilities, the researchers successfully detected the presence of birds in the images. Additionally, the classification task was accomplished using the trained models, which enabled accurate identification of the specific bird species among the 525 possibilities. This comprehensive approach allowed for both detection and classification of birds in the study.

The purpose of this article is to explore the application of YOLOv5 [3] (You Only Look Once) in detecting specific image areas that contain birds. Additionally, the article aims to compare the results achieved by various deep learning models such as VGG19, EfficientNetB3, and InceptionV3 in classifying different types of birds. To create a comprehensive model, the article proposes combining both bird detection and classification techniques. By leveraging the strengths of YOLO and deep learning models, this research aims to contribute to the development of robust systems for bird detection and classification in images. Moreover, the study examined how the performance of the models was influenced by optimization algorithms such as SGD, Adam, and Adamax.

The paper’s structure is organized as follows: In Section II, a thorough literature review is presented, covering the relevant background. Section III elaborates on the method employed for bird detection in image, as well as the implementation of three proposed transfer learning models for the task of bird classification. This section discusses various aspects, including Data Set and Data Preparation, Optimizers, and Model Evaluation Metrics. The experimental system and results are then detailed in Section IV. Section V offers conclusive remarks to wrap up the paper.

II. RELATED WORKS

In today’s era, CNN (Convolutional Neural Network) models have emerged as powerful tools for addressing classification
problems across various fields. With their ability to automatically learn hierarchical representations from raw data, CNN models have revolutionized the field of computer vision and beyond. These models excel in extracting meaningful features and patterns from images, making them particularly well-suited for tasks like object recognition, image classification, and segmentation. By leveraging deep learning techniques, CNN models have achieved remarkable success in domains such as healthcare, autonomous vehicles, natural language processing, and more. Their versatility, scalability, and robustness make them widely adopted in academia and industry alike, as they continue to push the boundaries of what is possible in the realm of classification problems. Several studies have been conducted focusing on TL techniques for the purpose of classification; in the medical field [4], [5], [6], [7], [8]; in the field of agriculture [9], [10], [11] and many other fields. In the problem of bird classification, Shazzadul Islam et al. [12] propose a ML approach for identifying the species of Bangladeshi birds. To extract image features from bird pictures, the researchers utilized the VGG-16 network. Among the various classification methods employed, Support Vector Machine (SVM) attained the highest accuracy, reaching an impressive 89%. Samparthi V S Kumar et al. [13] compares the performance of MobileNet, AlexNet, InceptionResNet V2, Inception V3, and EfficientNet for bird species recognition using a dataset of 11,488 images (increased to 40,000 through data augmentation). The results indicate that MobileNet and EfficientNet are the fastest to train, and EfficientNet achieves the highest test accuracy of 87.13%. The study [14] utilized traditional ML algorithms, DL algorithms, and transfer learning-based deep learning algorithms to classify the Kaggle-180-birds dataset, with the transfer learning-based classifier achieving the highest classification accuracy of 98%. The paper [15] compares and evaluates DL models (SSD, YOLOv4, and YOLOv5) for the classification and identification of bird species using the CUB-200-2011 dataset, with the YOLOv4 model achieving superior performance, including 95.43% accuracy, 93.94% precision, 94.34% recall, 94.27% F-1 score for 20 classes, and 96.99% mAP score. Ichsan Budiman et al. [16] utilized the K-Nearest Neighbor (KNN) algorithm to classify bird species based on a dataset of 58,388 images belonging to 400 species, achieving an accuracy of 95.5%. This paper [17] introduces a dual TL method for enhanced seabird image classification with spatial pyramid pooling, concatenating outputs from InceptionV3 and DenseNet201 backbones, and applying global average-pooling and global max-pooling. The proposed method achieved high accuracy, precision, recall, and F1-score of 95.11%, 95.33%, 95.11%, and 95.13%, respectively, on a 10-class seabird image dataset. Apart from the image-based classification of birds, extensive research has also been conducted on classifying birds using sound signals. This research [18] focuses on utilizing convolutional neural networks (CNNs) to develop an automated system for bird species identification based on spectrogram images. By analyzing the challenges of bird species detection, segmentation, and classification using a publicly available dataset of 8000 audio examples, the study concludes that a CNN-based approach with fully convolutional learning achieves efficient and accurate results. Through a 9-step implementation, the system demonstrates high accuracy (0.9895), precision (0.9), and a minimal loss (less than 0.0063) after training and validation with 50 epochs. Kumar et al. [19] explores the use of deep neural networks (DNN) and TL for automatic voice recognition and species identification of 22 bird species using various feature extraction techniques. The results indicate that models like ResNet50, DenseNet201, InceptionV3, Xception, and EfficientNet achieve high prediction accuracy, with DenseNet201 and ResNet50 attaining the best classification accuracy of 97.43% on the validation set. The study [20] focuses on identifying the most suitable cepstral features for the accurate classification of 15 endemic Bornean bird sounds. By comparing different feature types, the model utilizing gamma tone frequency cepstral coefficients (GTCC) achieves superior performance with 93.3% accuracy. The most advanced CNN models currently utilize pre-trained networks and can classify bird species from various angles and positions. These methods have the potential to significantly enhance the accuracy. This article investigates YOLOv5 (You Only Look Once) for detecting bird-specific image areas and compares the classification performance of DL models like VGG19, Inception V3, and EfficientNetB3. By leveraging YOLO and deep learning, the research aims to enhance the development of robust systems for accurately detecting and classifying birds in images. Furthermore, the study delved into the influence of optimization algorithms, namely SGD, Adam, and Adamax, on the performance of the models.

III. METHODOLOGY

A. Dataset and Data Preparation

The dataset utilized in the article consists of a vast collection of 89,885 images. It encompasses a diverse range of 525 bird species, with 84,635 images allocated for training purposes. Additionally, the dataset includes 2,625 images each for testing and validation. All the images possess a resolution of 224x224 pixels and are in JPG format with RGB channels. The dataset was sourced from https://www.kaggle.com/datasets/gpioskenka/100-bird-species [21]. Significantly, a meticulously chosen set of five images per species was incorporated for testing and validation purposes. Fig. 1 provides additional information regarding the dataset.

B. Transfer Deep Learning Models, Proposed Model

The objective of this study is to employ the yolov5 algorithm for bird region detection in images, followed by utilizing a transfer learning model for bird classification. To assess the effectiveness of this approach, we compare the efficiency of bird classification using multiple proposed models based on transfer learning techniques, namely VGG19 [22], InceptionV3 [23], and EfficientNetB3 [24]. The architecture of the classification models proposed in this study is depicted in Fig. 2. Through this comparative analysis, we aim to evaluate the performance and efficacy of our solution in accurately classifying birds. In Fig. 3, the study conducted training on 03 bird classification models based on the EfficientNetB3, InceptionV3, and VGG19 network architectures, and selected the model with the highest accuracy. Fig. 4 presents the architecture and layers of the proposed models for bird classification based on the EfficientNetB3, InceptionV3, and VGG19 network architectures. The architecture and layers of the proposed model with the best results for the bird classification problem are built based on the EfficientNetB3 network architecture shown in Fig. 5.
Fig. 1. Sample bird species in the dataset.

Fig. 2. Proposed model for bird detection and species classification.
C. Optimizers

An optimizer is a key element in machine learning and deep learning that facilitates the training of models. Its primary purpose is to iteratively adjust the parameters of a model with the objective of minimizing a specific loss function. By finding optimal parameter values, the optimizer helps enhance the model’s performance on various tasks like classification, regression, or generative modeling. Different optimizers employ distinct algorithms and techniques to update the model’s parameters based on gradients computed from the training data.

The choice of optimizer greatly influences convergence speed, generalization ability, and overall model performance. Overall, an optimizer is a vital tool that guides the optimization process in machine learning by iteratively refining model parameters to achieve better results. In this paper, we assess the performance of transfer learning models by employing three highly effective algorithms: Stochastic Gradient Descent (SGD) [25], Adam [26], and Adamax [27]. These optimizers have demonstrated remarkable capabilities in various machine learning tasks and are widely recognized for their ability to optimize model parameters efficiently. By leveraging SGD, Adam, and Adamax, we aim to evaluate the accuracy and effectiveness of transfer learning models and explore their potential for enhancing performance in diverse domains.

D. Evaluation Metrics

In this investigation, the efficacy of DL models was evaluated using a diverse set of metrics, including Precision, Recall, F1-score, and Accuracy. The overall performance of the models in predicting the target variable was assessed using Accuracy. Precision was utilized to measure the ratio of correctly predicted positive outcomes to all positive predictions, while recall quantified the proportion of accurately predicted instances of positive outcomes to all instances of positivity in the dataset. To provide a balanced perspective on the model’s effectiveness, particularly in cases of imbalanced classes, the F1-score, which combines precision and recall, was employed. By employing multiple evaluation metrics, we obtained a comprehensive understanding of the model’s effectiveness and were able to make informed judgments regarding its effectiveness.

\[
\text{Accuracy} = \frac{TP + TN}{TP + TN + FP + FN} \quad (1)
\]

\[
\text{Precision} = \frac{TP}{TP + FP} \quad (2)
\]

\[
\text{Recall} = \frac{TP}{TP + FN} \quad (3)
\]

\[
F_1 - \text{Score} = \frac{\text{Precision} \times \text{Recall}}{\text{Precision} + \text{Recall}} \quad (4)
\]

In which, TP: True Positive, TN: True Negative, FP: False Positive, FN: False Negative.
IV. RESULTS

In this section, we present the training and test results obtained for three models based on: EfficientNetB3, VGG19, and Inception V3. These models are based on their respective architectures and were trained using different optimizers, namely Adam, Adamax, and SGD. The purpose of these experiments was to evaluate the performance of the models under varying optimization algorithms. By utilizing different optimizers, we aimed to explore the impact on training and test outcomes, thereby gaining insights into the models’ effectiveness.

EfficientNetB3: Among the three models, EfficientNetB3 consistently demonstrates the highest test accuracy of 98% across all optimizers (SGD, Adam, and Adamax). This indicates that EfficientNetB3 performs exceptionally well in correctly classifying unseen data. In addition to test accuracy, EfficientNetB3 also achieves high train accuracy, with values ranging from 0.967 to 0.9915 across all optimizers (SGD, Adam, and Adamax), indicating its ability to effectively learn from the training data. Furthermore, EfficientNetB3 exhibits low loss values during training, with the lowest recorded value of 0.0288 using the Adam optimizer. This suggests that the model effectively minimizes the discrepancy between predicted and actual values, resulting in improved performance. Detailed results are presented in Fig. 6.

VGG19: VGG19 performs slightly lower than EfficientNetB3 but still delivers respectable results. Among the optimizers used, VGG19 achieves the highest test accuracy of 95% when trained with the Adam optimizer. This indicates its ability to classify unseen data with a relatively high level of accuracy. VGG19 also demonstrates reasonably good train accuracy, ranging from 0.967 to 0.9915 across all optimizers (SGD, Adam, and Adamax), indicating its ability to effectively learn from the training data. Furthermore, EfficientNetB3 exhibits low loss values during training, with the lowest recorded value of 0.0288 using the Adam optimizer. This suggests that the model effectively minimizes the discrepancy between predicted and actual values, resulting in improved performance. Detailed results are presented in Fig. 7.

Inception V3: Inception V3 achieves test accuracies of 92% to 93% across all optimizers. While it performs slightly lower than EfficientNetB3 and VGG19, Inception V3 still demonstrates a reasonable ability to classify unseen data...
accurately. In terms of train accuracy, Inception V3 achieves values ranging from 0.8624 to 0.913, which are slightly lower than the other two models. However, it shows relatively lower loss values during training compared to VGG19, indicating better convergence toward the desired outputs. Detailed results are presented in Fig. 8.

In summary, EfficientNetB3 consistently achieves the highest test accuracy across all optimizers, indicating its superior performance in classifying unseen data. VGG19 performs slightly lower but still achieves respectable results, with the best test accuracy attained using the Adam optimizer. Inception V3 also demonstrates good performance, although slightly lower than the other two models, with the best test accuracy achieved using the Adam optimizer. Detailed results comparing the testing accuracy and loss values of the EfficientNetB3, VGG19, and InceptionV3 models using Adam Optimizer are presented in Fig. 9.

Based on the information given in Table I, it appears that EfficientNetB3 consistently achieves high precision, recall, and F1 scores across all optimizers. It consistently achieves a score of 0.98 for precision, recall, and F1 score, indicating its effectiveness in correctly identifying positive instances and achieving a balance between precision and recall. While VGG19 and Inception V3 also show relatively strong performance, EfficientNetB3 consistently demonstrates higher scores in all metrics.

Fig. 10 shows comparisons of test accuracy value between EfficientNetB3, VGG19 and Inception V3. Fig. 11 shows the results obtained by using YOLOv5 to detect birds in the image (https://ebird.org/species/abbbab1) will be further classified into bird species using an EfficientNetB3-based model.
Fig. 8. Comparing the training accuracy and loss values of the InceptionV3 model using Adam, Adamax, and SGD.

Fig. 9. Comparing the testing accuracy and loss values of the EfficientNetB3, VGG19 and InceptionV3 model using Adam Optimizer.

TABLE I. MODEL COMPARISON - PRECISION, RECALL, F1 SCORE

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V. CONCLUSION

In conclusion, this study has successfully addressed the challenges associated with accurately identifying bird species in images, a task of great significance in ecological research and bird conservation efforts. By employing the YOLOv5 object detection algorithm for bird localization and leveraging deep transfer learning models like VGG19, Inception V3, and EfficientNetB3 for species classification, the researchers have developed an efficient and accurate system for bird detection and classification. The experimental results demonstrate the effectiveness of this approach, achieving high levels of accuracy in both bird detection and species classification tasks. The combination of YOLOv5 with deep transfer learning models has shown great potential for comprehensive bird analysis, paving the way for automated bird monitoring, ecological research, and conservation efforts. Moreover, the study also investigated the impact of optimization algorithms, including SGD, Adam, and Adamax, on the model’s performance, providing valuable insights into the suitability of different deep transfer learning architectures for avian image analysis. Overall, these findings contribute to the advancement of bird recognition systems and offer valuable knowledge for improving the field of avian image analysis.


FCCC: Forest Cover Change Calculator User Interface for Identifying Fire Incidents in Forest Region using Satellite Data

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Abstract—For the ecosystem to maintain a balance between the social and environmental spheres, forests play a crucial role. The greatest threat to forests for this significance, is fires and natural disasters caused by several factors. It is crucial to assess the genesis and behavioral characteristics of fires in forest areas. The discovery of the forest fire areas and the intensity of the fire affected are greatly facilitated by the satellite image obtained by different sensors and data sets. We are suggesting a novel approach to compute using spectral indices, using landsat-9 and sentinel-2 satellite datasets for measuring the change in forest areas affected by fire accidents over Kochi areas on March 2023. Kochi is a city in Kerala, South India, and is located at 9° 50’ 20.7348” N and 77° 10’ 13.8828” E. coordinates. Computation is performed by calculating forest area before the fire incident (pre-fire) and after the fire incident (post-fire) and total loss is calculated by the difference between pre-fire and post-fire incident. The proposed work uses Sentinel-2 and Landsat-9 satellite images to recover burn scars using several vegetation indicators. We have identified the fire locations using the object-based classification approach. For verification of results computed by vegetation indices, we have also performed land use land cover classification and calculated the changes in forest areas. Accuracy is computed by the confusion matrix with an accuracy of 89.45% and the kappa coefficient with an accuracy of 87.68%. In particular, there was a strong correlation between forest loss and the burned area in the subtropical evergreen broadleaf forest zone (6.9%) and the deciduous coniferous forest zone (18.9% of the lands). These findings serve as a foundation for future forecasts of fire-induced forest loss in regions with similar climatic and environmental conditions.

Keywords—GEE; remote sensing; classification; landsat; sentinel; forest fire

I. INTRODUCTION

The forest is a foundation for all living things and is crucial in influencing global climate change. The forest is divided into basically three types: Reserved Forest (RF), Protected Forest (PF), and Unclassified Forest (UF). Under federal or state forest legislation, a place is known as a Reserved Forest (RF), which is completely protected. Unless otherwise permitted, no activity is allowed in the reserved forest. The State Forest Act or the Federal Forest Act may designate a territory as a Protected Forest (PF). An unclassified Forest (UF) is a place that has been given a forest designation but is not a restricted or protected forest. Depending on the state, these woodlands may be owned differently. Being the primary categories of natural landscapes, forests are the most priceless natural resources on earth. Unlike other natural resources like minerals, mineral oils, and natural gas, which are finite and cannot be replenished, forests have the amazing advantage of being renewable. However, the productivity of forests depends heavily on human activity. More than four billion acres of forest cover the entire earth, according to a survey by the Global Forest Resource Assessment (GFRA) in 2020 [1], the size of these forests cover varies, and it covers the entire surface of the earth. With around 45% of the world’s total forest cover, Europe has the largest forest cover, according to another report from 2011. Considering percentages, South America is the top continent in terms of the area that is covered by forests, with almost half of its land mass falling under forests. If we consider per capita forest area, Oceania is placed first. But a percentage of these forest areas are degraded year to year due to several natural factors or man-made activity. According to the Forest Survey of India in the Year 2023 between the dates 1 March 2023 to 15 March 2023, a total of 772 large fire accidents happened. Mizoram has been affected by a maximum total of 110 fire incidents, and Meghalaya, Manipur, and Assam forest areas have been affected a total of 59, 52, and 43 fire incidents respectively. The reasons behind these fire incidents in forest areas are less precipitation level in February 2023. The country recorded only a 7.2 mm rain level in February which is the sixth lowest from the year 1901, central India recorded 77% rain deficiency while the northwest, southern, and northeast area recorded 76%, 54%, and 43% rain deficiency respectively. Total 38 forest fire incidents were reported in Kerala between January and April of 2018. Kerala had numerous fire events, from January to March 2017; 2,100 hectares were destroyed in 440 fire incidents. The reason behind this incident is climate change; also because one of the factors that contribute to fires is the increase of dryness in the environment. Forest Fires can be identified and characterized by several indicators. The spectral reflectance characteristics of healthy vegetation and burnt scar vegetation can be used to identify forest fires. For fire detection, thermal variations between burning and background pixels are also frequently used. The thermal infrared bands on satellite sensors like ASTER, MODIS, and VIIRS also can be used to detect forest fires. Since the majority of these sensors are categorized as having coarser spatial resolutions, it is challenging to monitor...
fires on a regional scale. For instance, burn area products from the Moderate Resolution Imaging Spectroradiometer (MODIS) are available globally in 500 m (MCD64A1) since 2001. There are a number of researchers who performed change detection due to fire incidents over different land cover areas like [3] performed forest fire area computation over Mugla, Turkey using different vegetation indices. The authors in [4] performed their analysis over the Pacific Northwest, USA for finding the effects of climate change on fire regimes. The rest of the paper is structured as follows. Related work is presented in Section II. Section III illustrates the study area. Section IV presents a methodology and materials like data sets, vegetation indices, and classification algorithms used in the computation. The result is discussed in Section V and in the last Section VI presents the concluding remarks.

II. RELATED WORK

Forests fire occurred in recent years in a very large manner. Remote sensing techniques are mainly used for the finding reason and computing loss for these fire incidents, there are a number of platforms available in geographic information systems (GIS) which are used to calculate these changes like QGIS, ArcGIS, and ELWIS using satellite images. But these GIS softwares required large processing of data before computation, [5] have used such geographic information, system-based software and produced a framework for computing change analysis. In this analysis data sampling was put into place in two primary stages: maintaining a binary map for the burned area and unburned areas and mapping the burned areas of various land cover classes. The primary stage focuses on mainly five steps: removal of cloud cover and other noise (prepossessing), spatial and spectral feature extraction for pre-fire and post-fire, analysis of change in forest area in pre-fire and post-fire computation, extraction of features, mapping of burned areas based on the selection of feature from VIIRS hotspots, but extraction of burned area and unburned area takes a very large amount of time and computation so some researchers used Google Earth Engine for computation and pre-analysis. The authors in [6] have performed their analysis using Google Earth Engine (GEE) [7], over two satellites Landsat 8 and Sentinel 2. The researchers performed their computation by partitioning into four-level, performed the supervised burned area over the cartography tool for stratified random sampling, after the selection of data change in the area over a given time period. In [8] researcher performed his study by use of 14 different sets of fire variables that are taken from spectral vegetation indices, environmental variables, climate factors, and other spatial features. The training and testing validation is performed using a classification algorithm later use bootstrap, and optimistic bootstrap approaches to evaluate the models’ accuracy and estimate the variance and bias of the estimate. But these researchers are focusing on the sample and collected data set only not on real-time data sets. Later Andrea Tassi and Marco Vizzari [9] proposed point-based (PB) and object-based (OB) classification and categorization techniques, that may be implemented on the spatial cloud platform, Google Earth Engine (GEE). Ramita Manandhar [10] also studied the framework-based technique of classification to enhance the accuracy of the classification algorithm by combining other data, such as different land cover classes, spatial features, spectral indices, and digital elevation models. The study [11] discusses the other category of classification, he performed categories based on the study of light detection and ranging (LIDAR) was used as part of an innovative strategy for mapping the danger of forest fires. The hierarchy process was used to calculate the criteria weights that affect fire risk and was applied to two different data sets over the different locations of Spain. The findings indicate that about 50% - 65% of the study area is classified as 3-moderate fire risk zones. The researcher will be able to choose the best vegetation management strategies based on the danger of forest fires according to the technique given in the [11] study. The forest fires problem does not occur only in the global region but also causes in many places in small areas like the valley of the Himalayas. Over the course of the whole research region, the forest fire burn area dramatically fluctuates with time and space. Research [12] concluded that fire problems arose in the west-to-south Himalayan region multiple times between years 2000 -2010. Higher burn area fraction patches (0.7sq-km to 1 sq-km) were discovered over the southeast regions of the eastern Himalayas, the central Himalayas, and the western Himalayas. Over the past two decades (2001–2020), the average yearly burn area was 5557.35 sq-km, with a large amount of variability (standard deviation 2661.71 sq-km). The yearly burn area increased significantly by 755.7 sq-km per year between 2001 and 2010 but declined in the following decade. Further [13] provides three models for anticipated forest fire susceptibility maps (FFSM) using the Google earth engine (GEE) platform and then maps it over geographic information system software. The researcher [14] uses a range of probability mechanisms for finding change due to fire incidents; value lie between 1 to 0 if it surrounds near 1 then occurrence and loss is high and in other case occurrence and loss is very less. Deforestation is the long-term loss of trees as a result of anthropogenic or natural activity. It happens everywhere as a result of intricate socioeconomic processes like population and housing increase, agricultural expansion, and wood exploitation in underdeveloped nations. Deforestation is also made worse by economic, political, technological, and cultural causes. Deforestation contributes to a number of serious issues, such as soil erosion, disruptions to the water cycle, and possibly global repercussions. Deforestation has caused the land to change too quickly, resulting in the loss of vegetation, and wildlife, which hinders the functioning of ecosystems [15], [16] computed the growth in urbanization and loss of forests between the years 2009 and 2019, and found the current trend of urban growth is continuing at a pace of 0.16% each year. Environmentalists and land planners comprehend the effects of land use and land cover to provide recommendations for effective policy strategies to manage growth in Cameroon Highlands. The study [17] discussed the forest management systems in Finland and Scandinavia. Satellite images are used in Finland and Sweden’s national forest inventories. Researchers [18] witnessed fast changes in land use and cover, with the area covered by vegetation dropping from roughly 46% in 2009 to 28% in 2019. Area of about 27.54 percent of the area covered by vegetation is converted to urban/built-up areas, 4.60 percent to agriculture, and 6 percent to arid terrain. The amount of agriculture and urban/built-up area has greatly expanded. The research [19] analyzed the change in different classes of land cover using a number of sample points and training and testing points for each class. Points are split into subgroups for training.
but the categorization accuracy within primary rain forests has a great deal of potential to be produced by remote sensing, and thematically precise vegetation maps. Such maps have and management, as well as ecological study, is made more difficult in tropical rainforests due to a dearth of spatially and thematically precise vegetation maps. Such maps have a great deal of potential to be produced by remote sensing, but the categorization accuracy within primary rain forests has typically fallen short of practical uses.

III. Study Area

The study area Fig. 1 is located at 9.50°N and 77.10°E. covering about 2407 square kilometers (1,171sq mi) and bordered on the north by the district of Thirssur, on the south by the districts Kottayam and Alappuzha, on the east by the district Idukki, and on the west by the Arabian Sea. Three separate sections make up the district: Hills and woods, plains, and the seashore, respectively, make up the highland, midland, and lowland regions. The highlands are located at a height of around 300 m. (980 ft). Except for Muvattupuzha, the Periyar River, Kerala’s longest, traverses every taluk. The district is traversed by the Chalakkudy River and the Muvattupuzha River.

It consists of several land cover classes like urban, water, and forest (high vegetation), agriculture (low vegetation) but forest alone consists of 45.20% of total land. In the year 2022-23, a total of nearly 430.75 hectares was affected by 391 fire incidents. According to data from the state pollution control board, the mean air quality index remained above 300 PM concentrations on March 6; 445 PM concentrations on March 8; and 380 PM concentrations on March 9. Good breathable air quality index was below 100 PM concentrations, like 2 PM concentrations. The study area

It was 441 PM2.5 concentrations on March 5; 445 PM2.5 concentrations on March 6; 465 PM2.5 concentrations on March 7; 324 PM2.5 concentrations on March 8; and 380 PM2.5 concentrations on March 9. Good breathable air quality has an index value of less than 50 PM2.5 concentrations, while before the dump yard fire, the city’s average air quality index was below 100 PM2.5 concentrations. The study area is computed as 3,432 mm of rain falls on average in the district this year. The district has a mild temperature average temperature between May 2022 to March 2023 as shown in Fig. 2.

The temperature rapidly increases during fire time and is largely located in the Malabar Coast moist forests ecoregion, while the highlands are a part of the South-Western Ghats moist deciduous forests ecoregion. On the border of the districts of Ernakulam and Idukki, the Anamudi is the tallest mountain in South India. Sholas can be found in some areas of the Mankulum Forest Division and Idamalayar Reserve Forest, however, these areas cannot be reached by road. Edamalakkudy and the Idamalayar Protected Forest, have different kinds of rocks, silt, and sand. The majority of the district’s eastern forests are secluded and are a portion of the Anamalais. Temperature is also a very significant factor after the forest fire incident it changed moderately.

IV. Materials and Methods

Since the 1970s, surface soil moisture (SSM) and change in the surface area has been determined by remote sensing. The primary benefit of remote sensing is that it offers geographically diverse data, as surface variables with spatial information are necessary for many applications, including evapotranspiration evaluation, soil erosion mitigation, irrigation scheduling, drought monitoring, and forest management.

The study is performed by using the satellite data set from Sentinel and Landsat over the cloud platform Google Earth Engine (GEE). GEE is a planetary platform that has access to different satellite data like Sentinel, Landsat, MODIS, etc. Each satellite has the unique characteristic as wavelength, band combination, and resolution. Here we are accessing sentinel-2 and Landsat-9 to observe forest area changes. A detailed description of the analysis is shown in the Fig. 3 and used bands used for analysis by both data sets are detailed in Table 2.

Research is subdivided into two parts. We calculate the change in forest area from pre-fire and post-fire accidents. Later we perform observation over the burn area and calculate the different changes in the forest area. These changes are calculated by computing the change in a land cover class by the gradient tree boost classification model and also by computing the change in mean and standard deviation value of different indices like Green Normalized Difference Vegetation, Adjusted Transformed Soil Adjusted Vegetation Index (ATSAVI), Normalize difference water index (NDWI) and Enhance vegetation index (EVI). The methodology used in the research is illustrated in Fig. 4. GNDVI (Green Normalized Difference Vegetation) measures the “greenness” or photosynthetic activity of plants. While it saturates later than NDVI, it is a chlorophyll index that is utilized during later phases of development. It is one of the most popular vegetation indices for calculating crop canopy water and nitrogen uptake. The values for Normalize difference water index (NDWI), like other indices, range from -1 to 1, with high values denoting high plant water content and coverage of a significant portion of the plant and low values denoting low vegetation water content and sparse cover. Pre-fire and post-fire results are calculated by using the normalized burn ratio mechanism and calculated by equation 1.

\[ NBR = \frac{(NIR - SWIR)}{(NIR + SWIR)} \] (1)

Where: NIR (Near Infra-Red) as a Band 5 SWIR (Short Wave Infra-Red) as a Band 7

GNDVI has a greater saturation point than NDVI and is more sensitive to changes in the crop’s chlorophyll content. While NDVI is useful for predicting crop vigor in the early stages, it can be used in crops with dense canopies or in more mature phases of growth.

\[ GNDVI = \frac{NIR - GREEN}{NIR + GREEN} \] (2)
Where: NIR: Near Infra-Red
GREEN: GREEN Wavelength Band
A water body can “stand out” against the land and vegetation by using the Normalized Difference Water Index (NDWI) to emphasize open water features in a satellite picture.

\[
NDWI = \frac{GREEN - NIR}{GREEN + NIR}
\]  

Where: NIR: Near Infra-Red
GREEN: GREEN Wavelength Band
The normal reflectivity of the sea surface is maximized in
the visible green wavelengths. The near-infrared wavelengths maximize the high reflectance of terrestrial vegetation and soil components while minimizing the low reflection of aquatic features. The NDWI equation yields positive values for water features and negative ones (or zero) for soil and terrestrial vegetation.

\[
EV I = \frac{2.5 \times (Band5 - Band4)}{(Band5 + 6 \times Band4 - 7.5 \times Band2 + 1)}
\]

\[\text{(4)}\]

In dense vegetation, EVI is more sensitive and can compensate for some atmospheric conditions and canopy background noise. While ATSAVI is used where vegetation cover is very low.

\[
ATSAVI = a \times \frac{(NIR - a \times RED - b)}{((a \times NIR) + (RED - a \times b) + (X \times (1 + a^2)))}
\]

\[\text{(5)}\]

Classification is performed using the Gradient tree boost machine learning classifier algorithm. It provides a hypothetical model in the form of an ensemble of decision trees. Mainly it is a collection of nodes of weak prediction models. The resulting technique, called gradient-boosted trees, typically it beats random forest when a decision tree is a weak learner. The construction of a gradient-boost trees model follows the same stage-wise process as previous boosting techniques, but it generalizes other techniques by optimizing any differentiable loss function, later we computed accuracy by using the confusion matrix by equation 6 and kappa coefficient by equation 7. Cohen recommended the following interpretation of the Kappa result: values 0 as showing no agreement and 0.01-0.20 as none to the partial agreement, 0.21-0.40 as fair agreement, 0.41- 0.60 as moderate agreement, 0.61-0.80 as significant agreement, and 0.81-1.00 as almost perfect agreement.

\[
\text{Accuracy} = \frac{TP + TN}{TP + TN + FP + FN}
\]

\[\text{(6)}\]

Where \(\hat{TP}\) = True Positive, \(FP\) = False Positive.
TABLE I. BAND USES FOR CALCULATING CHANGES IN FOREST COVER

<table>
<thead>
<tr>
<th>COLOR</th>
<th>LANDSAT</th>
<th>WAVE LENGTH</th>
<th>SENTINEL</th>
<th>WAVE LENGTH</th>
</tr>
</thead>
<tbody>
<tr>
<td>BLUE</td>
<td>BAND - 2</td>
<td>0.45 - 0.51</td>
<td>BAND - 2</td>
<td>0.492 - 0.496</td>
</tr>
<tr>
<td>GREEN</td>
<td>BAND - 3</td>
<td>0.53 - 0.59</td>
<td>BAND - 3</td>
<td>0.559 - 0.560</td>
</tr>
<tr>
<td>RED</td>
<td>BAND - 4</td>
<td>0.63 - 0.67</td>
<td>BAND - 4</td>
<td>0.664 - 0.665</td>
</tr>
<tr>
<td>NIR</td>
<td>BAND - 5</td>
<td>0.85 - 0.87</td>
<td>BAND - 8</td>
<td>0.833 - 0.835</td>
</tr>
<tr>
<td>SWIR</td>
<td>BAND - 7</td>
<td>2.1 - 2.2</td>
<td>BAND - 12</td>
<td>2.18 - 2.20</td>
</tr>
</tbody>
</table>

\[ \tilde{TN} = \text{True Negative}, \]
\[ \tilde{FN} = \text{False Negative} \]

\[ \kappa = \frac{P_o - P_e}{1 - P_e}. \] (7)

Where \( P_o \) = Observed proportional agreement, 
\( P_e \) = Hypothetical probability of chance agreement.

V. RESULT

Deforestation is taking place day to day in a very large manner, sometimes it is being done by human activity, and sometimes it is by natural activity, like flood, and fire. We are calculating a change in the forest area of Kochi district from all the above mention activity. The primary reason for the change in forest area over the study area is some fire accidents that occurred between January 2023 and March 2023. But we are not only rigid for these fire issues we also calculate all other possible reasons also for deforestation. So here we calculated vegetation indices like NDVI (Normalize Difference Vegetation Index), GNDVI (Green Normalize Difference Vegetation Index), ATSAVI (Adjusted Transformed Soil Vegetation Index), and EVI (Enhanced Vegetation Index). These vegetation indices are used for the calculation of Change of Vegetation indices and also for evaluating the effect on soil moisture and water due to these changes. For the calculation of forest change in the study area we first calculate the change as shown in Fig. 4 and Fig. 5 using a supervised classification technique Gradient Tree Boosting. For this, we calculated about 1200 data points of different land cover classes and took 70% data as training points and 30% data as testing points. Results of this classification are shown in Fig. 4 and Fig. 5 and these changes are described that a large amount of forest loss occurred by these fire incidents. So for calculating these losses here, we computed the percentage of the forest before the fire incident (pre-fire) and after the fire incident (post-fire). After classification, we computed the accuracy of the classification result by confusion matrix and kappa coefficient, total accuracy computed by confusion matrix is 89.45% and by kappa coefficient is 87.68%. On focusing output generated by the gradient tree boost in Fig. 4 and Fig. 5 there is a lot of change happening in the east-south region of the study area. The reason behind these changes is fire accidents in these areas. So we calculate fire accidents in this area, mainly two major fire accidents found at two places, and many vegetation changes occur due to this fire accident. Fig. 6 and Fig. 7 shows the change in vegetation due to these fire accidents in the study area.

Fig. 4. Output generated over landsat-9 data sets before fire incident.

Fig. 5. Output generated over landsat-9 data sets after fire incident.

Fig. 8 shows the loss of land cover areas during a fire incident. The fire affected approximately 24.67% of land cover.
Fig. 6. Fire incidents happened between January and February months.

Fig. 7. Fire incidents happened between February and March months.
areas. These areas come under dense forest, moderate dense forest, open forest, and scrubland as we computed here forest loss. So non-forest areas about 74% are detected as having no loss.

Green Normalize Difference Vegetation index, Normalize Difference Water Index, Adjusted Transformed Soil Adjusted Vegetation Index. Enhanced Vegetation Index is calculated over Ernakulum during the forest fire time from September 2022 to March 2023. We found that time when rapid fire arises, its mean and standard deviation value change. For verification purposes, we have computed the mean GNDVI over two data sets Landsat-9 and Sentinel-2.

These two data sets are easily available in the Google Earth Engine directory; basic difference between both data sets in terms of resolution is Landsat-9 data sets having a resolution of 30 meters and Sentinel data sets having a resolution of 10 meters, these data sets also have a collection of nine different bands each band having unique attributes and the ratio of these bands providing a different signature for each vegetation indices. Fig. 9 and Fig. 10 show the change in green normalize difference vegetation index over sentinel-2 and Landsat-9 data sets respectively and both data sets have nearly the same result over the study area. Later we calculate mean and standard deviation values over normalized difference water index (NDWI), adjusted transformed soil adjusted vegetation index (ATSAVI), and enhanced vegetation index (EVI). For observing the effects of fire incidents over water, soil, and natural vegetation indices. Fig. 11 and Fig. 12 shows results.

Fig. 8. Changes in the study area.

Fig. 9. Green normalize difference vegetation index calculated over study area during forest fire using sentinel 2 data set.

Fig. 10. Green normalize difference vegetation index calculated over study area during forest fire using landsat 9 data set.
Fig. 11. Normalize difference water index calculated over study area during forest fire using sentinel 2 data set.

Fig. 12. Normalize difference water index calculated over study area during forest fire using landsat 9 data set.

Fig. 13. Adjusted transformed soil adjusted vegetation index calculated over study area during forest fire using sentinel 2 data set.

Fig. 14. Adjusted Transformed Soil Adjusted Vegetation index calculated over study area during forest fire time using landsat 9 data set.

Fig. 15. Enhanced vegetation index calculated over study area during forest fire using sentinel 2 data set.

Fig. 16. Enhanced vegetation index calculated over study area during forest fire using landsat 9 data set.

for normalized difference water index, Fig. 13 and Fig. 14 show result for adjusted transformed soil adjusted vegetation index, and Fig. 15 and Fig. 16 shows the result obtained from
Fig. 17. Forest loss computed from year 2000 to year 2023.

Fig. 18. Snip of user interface available at https://paper1.users.earthengine.app/view/forest-cover-change.
enhanced vegetation index over Sentinel-2 and Landsat-9 data sets respectively; this result shows that change in these indices occurred at the time of fire incident over the study area and also after the fire incident also. From the above observations, we find that fire forest reduces greenery as well as affects water and soil also. Equations (1 to 5) are used for the calculation of these indices and finding change during fire incidents. Equation 1 is used for calculating a change in the study area due to fire incidents and providing details results which are shown in Fig. 6 and Fig. 7. Areas affected by these fire incidents are described in Fig. 8. From the result, it is shown that a large area of forest loss nearly happened over the study area due to fire incidents. These fire incidents not only disrupted climate conditions and global warming but also had a huge effect on soil and other vegetation indices. We have calculated total forest loss from the year 2000 to the year 2021 using Hansen forest loss cover analysis data sets, and find loss increased from the year 2017 due to multiple reasons over the study area. Detailed analysis of loss cover is shown in Fig. 17.

Later at the end, we develop an application that collects data in the form of geometry or if the user has their shape or CSV file of the study area can provide a path in user access link. Users can select desired geometry in the form of a point, polygon, or in rectangle form. A snap of the User interface is located in Fig. 18.

VI. Conclusion

The proposed analysis model is used to compute forest loss in Ernakulum (Kochi) using Landsat-9 OLI, and Sentinel 2 satellite data due to the occurrence of the fire incident. Through our analysis, we found deciduous forests are more vulnerable to fires. About 18.2% of vegetation area was affected by fires in 2023. Among all types of vegetation classes, vegetation that has higher density is affected most by the fire incident. Fire not only destroys vegetation cover but also has an impact on soil and water as well as on climate conditions also. We are also focused on finding factors behind these multiple fire occurrences over study and find an increase in temperature always increases the probability of fire occurrence. We have computed our analysis with more than 85% accuracy. Later our objective is to develop a model that can be used for analyzing change due to fire or another natural disaster by a number of vegetation indices using our defined model.

REFERENCES


Enhancing Computer-assisted Bone Fractures Diagnosis in Musculoskeletal Radiographs Based on Generative Adversarial Networks

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Meridian Team, LYRICA Laboratory, School of Information Sciences, Rabat, Morocco², ³

Abstract—Computer-Assisted Bone Fractures Diagnosis in musculoskeletal radiographs plays a crucial role in aiding medical professionals in accurate and timely fracture detection. In this work, we explore a Generative Adversarial Network based approach for this task, which is a powerful deep learning model capable of generating realistic images and detecting anomalies. Our proposed approach leverages the potential of GANs to generate synthetic radiographs with fractures and identify anomalous patterns, thereby enhancing fracture diagnosis. Through extensive experimentation and evaluation on musculoskeletal radiograph datasets (MURA), we demonstrate the effectiveness of GAN-based models in improving fracture detection performance by adopting several evaluation metrics notably accuracy, precision, F1-score and detection speed. These findings highlight the potential of integrating GANs into computer-assisted diagnosis, contributing to the advancement of fracture diagnosis methodologies in orthopedics. It is important to note that GANs operate by training a generator network to produce synthetic images and a discriminator network to distinguish between real and generated images. This adversarial process fosters the generation of realistic radiographs with fractures, enabling accurate and automated detection. Our findings contribute to the advancement of fracture diagnosis methodologies and pave the way for more efficient and precise diagnostic tools in the field of orthopedics.

Keywords—Deep learning; generative adversarial network; diagnosis; orthopedics; fracture detection; x-ray image

I. INTRODUCTION

Accurate and timely diagnosis of bone fractures is crucial in musculoskeletal radiology for effective patient care and treatment planning[1], [2], [3]. Conventional methods heavily rely on the expertise of radiologists [4], which can be subjective and time-consuming. However, recent advancements in artificial intelligence and deep learning techniques have paved the way for computer-assisted diagnosis systems that can aid radiologists in detecting and classifying fractures with improved accuracy [5].

In this article, we explore the application of generative adversarial networks (GANs) for computer-assisted bone fracture diagnosis in musculoskeletal radiographs. GANs are a class of deep learning models comprising two neural networks: a generator and a discriminator[6]. Its architecture give a promising future for anomaly detection in several fields[7], [8], [9]. The integration of GANs in fracture diagnosis presents several advantages. Firstly, it offers the potential to reduce radiologists’ workload by automating the initial screening process. This allows radiologists to allocate more time and attention to complex cases, ultimately improving patient care. Secondly, GANs have shown promise in enhancing diagnostic accuracy by providing a reliable second opinion. By learning from a large dataset of medical images, GANs can capture subtle fracture features and aid radiologists in making more informed decisions. Moreover, GAN-based systems have the potential to contribute to the standardization of fracture diagnosis. By learning from diverse cases and incorporating a wide range of fracture patterns, these systems can help minimize inter-observer variability and increase diagnostic consistency across different healthcare settings. Furthermore, the continuous learning capabilities of GANs enable the system to adapt to new data and improve its diagnostic performance over time. Despite these promising advancements, several challenges must be addressed. Ensuring the robustness and generalizability of GAN-based fracture diagnosis systems across various patient populations, imaging modalities, and fracture types is essential. Furthermore, ethical considerations, data privacy, and regulatory frameworks need to be carefully considered to ensure the responsible and safe implementation of these technologies in clinical practice.

In this work, we will investigate how the integration of GANs in computer-assisted bone fracture diagnosis can hold a great potential in revolutionizing musculoskeletal radiology. By combining the expertise of radiologists with the power of deep learning, these systems can enhance diagnostic accuracy, streamline workflow, and contribute to standardized fracture diagnosis. As ongoing research and development in this field continue to unfold, it is expected that GAN-based computer-assisted diagnosis systems will play a pivotal role in improving fracture detection and patient care in the near future. To this end, we will explore MURA dataset [10], the largest public radiographic image datasets, to see what GANs may be drawn on orthopedic anomaly detection on X-ray images.

The contribution through this work can be summarized as follows:

- Efficient and accurate anomaly detection techniques based on deep learning to detect bone fractures.
- Applying several GANs models in one work to have a rich comparative study.
- Review the examined deep learning models, approaches and architectures.
- Identification of the most suitable data pre-processing techniques for our study, especially for the dataset em-
ployed. This analysis provides valuable insights into the optimal preprocessing steps required to improve the overall effectiveness of the models.

- Optimize the performance of the examined models to overcome the performance of recent and relevant works.
- Evaluates the effectiveness of the examined models using different evaluation metrics to have a comparison and analysis study based on their results.

The rest of the paper is designed as follows: In Section I we provide the essential background to understand the rest of the paper. Section II lays out a brief summary of the related work in the same field. Section III presents our paper's methodology. Section IV presents the experiment, materials used, dataset description and evaluation metrics for a promising comparative analysis study. Section V presents models' results and discusses the outcomes based on several evaluation metrics. Finally, the conclusion and the findings of this work are in Section VI.

II. RELATED WORK

Fractures are highly accurate indicators of orthopedic pathology in most hospitals. However, analyzing medical images to identify bone fractures can be time-consuming and requires the expertise of qualified professionals. To address these challenges, scientists have been investigating ways to reduce diagnosis time and improve decision precision, aiming to assist doctors in their diagnostic processes [11], [12], [5], [13]. Several studies have demonstrated the potential of AI/DL in supporting medical professionals and decision-makers by developing automated tools that enhance the accuracy of physician interpretation [14], [5], [15], [16], [17], [18], [19], [20] and facilitate the creation of effective and cost-efficient treatment plans [15]. While many studies have focused on accurately detecting musculoskeletal abnormalities using CNN models, our research specifically explores the application of generative adversarial networks (GANs) [21] as a novel approach as shown in Table I. In this article [22], we propose the Res-UnetGAN network, an unsupervised anomaly detection approach based on GANs. The architecture combines the ResNet50 and UNet models within the generative network to calculate the normal distribution of samples. The discriminator employs a deep separable convolution-based convolutional neural network model to facilitate the adversarial training process. Anomaly identification is achieved by evaluating the reconstruction error score, which measures the quality of reconstruction and detects the presence of defects. Extensive testing on the Mura dataset demonstrates that our proposed method outperforms several other models in terms of defect detection accuracy.

In addition, [23] Davletshina et al. highlighted the value of unsupervised techniques trained on radiographic images without anomalies. Their approach aims to improve diagnostic accuracy and reduce the possibility of overlooking critical areas. By leveraging cutting-edge unsupervised learning techniques, they successfully identify anomalies and demonstrate the justifiability of the results.

Through our research, we aim to contribute to the advancement of fracture diagnosis by integrating the capabilities of GANs in framework will be used by radiographs. Hopefully this approach will have the potential to enhance the efficiency and accuracy of fracture diagnosis, enabling timely interventions and improved patient care.

III. METHODS

In this section we will present our proposed framework and we will review GAN models investigated in this study.

A. Proposed Framework

In the proposed framework for fracture diagnosis as shown in Fig. 1, the process begins with the acquisition of the images using medical imaging devices; then will be integrated into computers. These medical images are then passed through a deep learning system that performs preprocessing tasks such as image enhancement, augmentation, rotation, and normalization. The preprocessed images are then fed into the fracture detection algorithm, which utilizes GAN models and CAM technique to identify and visualize fractures within the medical image. Once the detection process is complete, the system evaluates the results using various evaluation metrics such as accuracy, precision, F1-score, and detection speed. These metrics provide quantitative measures of the system's performance in correctly identifying fractures. Based on these metrics, a final decision is generated, indicating the presence or absence of fractures in the input image. This decision is crucial in assisting radiologists in making accurate diagnoses and treatment decisions. The framework aims to enhance the efficiency and time to diagnosis fracture detection, ultimately improving patient care in the field of musculoskeletal radiology.

B. Overview of GANs

Generative Adversarial Network (GAN) is a powerful deep learning framework that has gained considerable attention in recent years [6]. As in Fig. 2 GANs are composed of two main components: a generator and a discriminator. The concept behind GANs is to train the generator and discriminator in a competitive manner. Initially, the generator produces random samples, and the discriminator tries to correctly identify them as fake. As training progresses, both the generator and discriminator learn and improve their capabilities. The generator aims to generate samples that are increasingly difficult for the discriminator to differentiate from real data, while the discriminator continuously adapts to distinguish the real and generated samples accurately.

One of the significant advantages of GANs is their ability to generate new and realistic data that captures the underlying distribution of the training data. GANs have been widely used for various applications, including image synthesis, text generation, anomaly detection, medical diagnosis, and data augmentation [5], [8], [7], [24], [26], [27]. The generated samples can be used to enhance training datasets, generate novel and diverse content, or assist in data analysis tasks. Despite their remarkable capabilities, GANs come with their own challenges. Training GANs can be complex and prone to instability, often requiring careful hyperparameter tuning and architectural considerations. Issues such as mode collapse, where the generator fails to explore the entire distribution of
TABLE I. SUMMARY OF RECENT WORKS OF FRACTURE DETECTION IN MUSCULOSKELETAL RADIOGRAPHS

<table>
<thead>
<tr>
<th>Ref/Year</th>
<th>Dataset</th>
<th>Approach</th>
<th>Results</th>
</tr>
</thead>
<tbody>
<tr>
<td><a href="2021">24</a></td>
<td>MURA dataset</td>
<td>A proposed unsupervised anomaly detection method is the Res-UnetGAN Network. This method incorporates a GAN that merges ResNet50 and U-Net architectures to create an autoencoder framework. This structure enables the system to learn distinctive characteristics from the input data.</td>
<td>Res-UnetGAN: 0.92 GANomaly: 0.81 Skip-GANomaly: 0.90 CVAE-GAN-Based: 0.86 EGBAD: 0.80</td>
</tr>
<tr>
<td><a href="2020">23</a></td>
<td>MURA dataset</td>
<td>Comparative study between GAN and AE models on anomaly detection.</td>
<td>CAE: 0.57 VAE: 0.48 DC-GAN: 0.53 BiGAN: 0.54 AlphaGAN: 0.60</td>
</tr>
<tr>
<td><a href="2020">25</a></td>
<td>MURA dataset</td>
<td>Computer Based Diagnosis (CBDs) model based on DenseNet201 and Inception V3 models, they were used to classify the given dataset as abnormal or normal.</td>
<td>DenseNet201: 87.15 InceptionV3: 86.11 Ensemble: 88.54</td>
</tr>
</tbody>
</table>

In the following Table II, we describe the different GAN models implemented in this work.

IV. EXPERIMENT

A. Experimentation Setup

The following subsection provide an overview of the dataset utilized in this study, including a description of its characteristics. The training settings and evaluation metrics employed.

1) Dataset description: The MURA dataset is a widely recognized and utilized dataset in the field of musculoskeletal imaging[10]. It comprises a large collection of radiographic images across different anatomical regions, including upper extremities, lower extremities, and the torso. The dataset focuses on various musculoskeletal conditions, especially fractures. The MURA dataset serves as a valuable resource for developing and evaluating algorithms and models in the domain of musculoskeletal radiography. Researchers and practitioners leverage this dataset to advance the field and improve diagnostic accuracy, automated diagnosis systems, and computer-assisted fracture detection techniques. The dataset is provide by the Stanford Program for Artificial Intelligence in Medicine: https://stanfordmlgroup.github.io/competitions/mura/.

2) Evaluation metrics: Accuracy: Accuracy is a widely used evaluation metric that measures the overall correctness of a fracture detection model [30]. It calculates the percentage...
### TABLE II. DESCRIPTION OF IMPLEMENTED MODELS

<table>
<thead>
<tr>
<th>Model</th>
<th>Type</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>[24] GANomaly</td>
<td>GAN-based anomaly detection model</td>
<td>GANomaly combines GAN architecture with anomaly detection techniques to identify fractures in radiographs. It learns to generate normal images and detects anomalies based on reconstruction error.</td>
</tr>
<tr>
<td>[24] SkipGANomaly</td>
<td>GAN-based anomaly detection model</td>
<td>SkipGANomaly extends the GAN architecture by incorporating skip connections to improve the quality of reconstructed images.</td>
</tr>
<tr>
<td>[26] AnoGAN</td>
<td>GAN-based anomaly detection model</td>
<td>AnoGAN combines GAN architecture with unsupervised learning to detect anomalies in images.</td>
</tr>
<tr>
<td>[27] MadGAN</td>
<td>GAN-based anomaly detection model</td>
<td>MadGAN is a GAN architecture that utilizes multiple discriminators to enhance the detection of anomalies.</td>
</tr>
<tr>
<td>[28] AttentionGAN</td>
<td>GAN model image-to-image translation</td>
<td>AttentionGAN is a type of GAN that incorporates an attention mechanism to improve the quality of generated images. It selectively focuses on important regions, capturing fine details and producing realistic outputs.</td>
</tr>
<tr>
<td>[5] DCGAN</td>
<td>GAN model with deep convolutional layers</td>
<td>DCGAN is a foundational GAN model that employs deep convolutional layers for image generation. It can be utilized to generate synthetic radiographs with fractures for training or augmenting the dataset.</td>
</tr>
<tr>
<td>[5] CycleGAN</td>
<td>GAN model for image-to-image translation</td>
<td>CycleGAN is primarily used for domain adaptation and image translation tasks. Although not directly designed for fracture detection, it can potentially be employed to translate normal radiographs to fractured ones, facilitating the identification of fractures based on the translated images.</td>
</tr>
<tr>
<td>[29] SAGAN</td>
<td>GAN model with self-attention mechanism</td>
<td>SAGAN incorporates self-attention mechanisms to improve the quality and coherence of generated images.</td>
</tr>
</tbody>
</table>

of correctly identified fractures out of all the samples in the dataset.

\[
\text{Accuracy} = \frac{\text{TruePositives} + \text{TrueNegatives}}{\text{TotalSamples}}
\]

**Precision:** Precision is a metric that focuses on the positive predictions made by the fracture detection model[30]. It measures the proportion of correctly identified fractures out of all the predicted fractures. Precision helps assess the model’s ability to minimize false positives, indicating how reliable the model is when it identifies a sample as a fracture.

\[
\text{Precision} = \frac{\text{TruePositives}}{\text{TruePositives} + \text{FalsePositives}}
\]

**F1-Score:** The F1-Score is a combined metric that takes into account both precision and recall, providing a balanced measure of the fracture detection model’s performance[30]. It considers the trade-off between false positives and false negatives. The F1-Score is particularly useful when the dataset is imbalanced or when both precision and recall are equally important.

\[
F_{1\text{score}} = 2 \cdot \frac{\text{Precision} \cdot \text{Recall}}{\text{Precision} + \text{Recall}}
\]

**Detection Speed:** Detection speed is an essential evaluation aspect that measures the efficiency of a fracture detection model in processing and analyzing musculoskeletal radiographs[30]. It quantifies the time taken by the model to detect fractures in a given dataset or per image. Faster detection speeds are desirable, particularly in clinical settings where time is of the essence.

By considering these evaluation metrics - accuracy, precision, F1-Score, and detection speed - we can thoroughly assess the performance and efficiency of fracture detection models. These metrics provide valuable insights into the model’s ability to correctly identify fractures, minimize false positives, achieve a balance between precision and recall, and process radiographic images efficiently.

### B. Preprocessing

The MURA dataset, presents a challenge due to its diverse range of images of bone abnormalities with different formats and sizes. To address this issue and make the data more uniform, we employed image pre-processing techniques to enhance the image quality. Initially, we applied binary thresholding to identify the Region of Interest (RoI) within the image and extract its contours. This process enabled us to isolate the relevant region for classification and crop it accordingly.

Data augmentation played a vital role in our research, helping to expand the dataset and improve the learning algorithm. Several augmentation approaches were employed, including horizontal image flipping, random rotation within a range of 30 degrees, scaling within the range of 95-130 percent, and randomly adjusting brightness within the range of 80-120 percent.

Prior to initiating the learning process, radiographs were normalized to have the same mean and standard deviation as the images in the ImageNet training set. This normalization step ensured consistency and facilitated subsequent stages of the project.
Following data augmentation and normalization contributed to enhancing the quality, consistency, and effectiveness of the dataset for our research purposes.

V. RESULTS AND DISCUSSION

In this section, we discuss the performance of GAN models for bone fracture detection in radiographs, focusing on key evaluation metrics such as accuracy, precision, F1-score, and detection speed. These metrics provide valuable insights into the effectiveness and efficiency of the models in identifying fractures and aiding in clinical decision-making. Various factors influence the performance of the models, including architectural approach, layer design, padding, shape, normalization, activation, loss function, optimizer, batch size, learning rate, pooling, and output layer. Consequently, achieving an effective result required multiple tuning iterations. Many of our models comprised computationally expensive layers and modules, necessitating long training durations that were often impractical on basic hardware or laptop configurations. Preprocessing played a crucial role in obtaining good results in our deep learning tasks. After selecting the appropriate GAN models for the study based on a benchmark between the different GAN model, extensive data preparation was necessary. The image size proved to be a significant parameter impacting the accuracy of detecting fractures. To this end, several treatments were involved on our dataset as mentioned in preprocessing section. Also, to overcome the limitations of the available data, we employed data augmentation techniques, which augmented the amount of data during the training phase. However, we had to be cautious regarding the rotation methods, excessive compression, and shear, as they could negatively affect the performance of bone fracture diagnosis. By considering these factors and conducting thorough experimentation, we aimed to optimize our models and enhance the accuracy of our results. It is evident that preprocessing, data augmentation, data normalization and careful parameter selection are vital considerations when striving for accurate and reliable outcomes in deep learning tasks related to bone fracture detection. Overall, as shown in Fig. 3 and 4 after selecting the input image we configure our framework by choosing body part treated, image modality and evaluation metrics that the user want to display. We see the visualized results by Grad-CAM and obtained from our implemented GAN models for bone fracture detection in radiographs were promising. The models demonstrated a high level of accuracy, achieving performance within the range of 0.7% to 0.954%. These results indicate that the GAN-based approach holds considerable potential for improving fracture diagnosis in the field of orthopedics.

Comparing our GAN models with previous works, it is evident that they compare favorably in terms of performance. The accuracy achieved by our models aligns with some results reported in works with the same task with different techniques and surpasses others results reported in relevant literature. This suggests that the utilization of GANs for bone fracture diagnosis can yield significant improvements in diagnostic accuracy and support medical professionals in their decision-making processes.

Detailedly, Table III shows that MadGAN, CycleGAN, SAGAN and SkipGANomaly on average achieves the best accuracy (0.954%; 0.922%; 0.9%; 0.901%). These accuracy scores indicate that the models were successful in correctly classifying fractures in radiographs, contributing to improved diagnostic capabilities. While with less processing time MadGAN and SAGAN have speed detection higher than other models. So we can consider MagGAN and SAGAN are the most powerful models. This is due to their architectures. MadGAN has the incorporation of multiple discriminators that can enhance the detection and identification of fractures in radiographs as anomalous patterns. SAGAN use self-attention mechanisms can improve the coherence and quality of generated images. Although not specifically tailored for fracture detection, its ability to produce visually consistent radiographs was the main cause to have an accurate fracture identification. Then we had GANomaly, AttentionGAN and AnoGAN achieved an accuracy of 0.861% and 0.842%. Those are reputable results even those models were not explicitly designed for fracture detection, but it could give a better performance in anomaly detection in other fields and with other parameters notably type of data, size of dataset, experiment setting, etc. For DCGAN, its performance in fracture detection be less compared to the models explicitly designed for anomaly detection or fracture identification. Because its architecture is not explicitly designed for fracture detection also due to the high number of convolutional layers that take more time to give the predictive result.

One notable advantage of GAN models is their ability to generate synthetic data, which can be beneficial in addressing...
the issue of limited labeled datasets. By leveraging the generative capabilities of GANs, it becomes possible to augment the available data and improve the robustness and generalization of the models. This can be particularly valuable in medical imaging, where acquiring large annotated datasets can be challenging. Despite the promising results, it is important to acknowledge the limitations and challenges associated with GAN models for fracture detection. One key consideration is the computational complexity and resource requirements of training and deploying GAN models. The training process of GANs can be computationally intensive and time-consuming, necessitating powerful hardware and substantial computational resources. This can pose practical limitations, especially in clinical settings where quick and efficient diagnosis is crucial.

To sum up, our study demonstrates the potential of GAN models in detecting bone fractures in radiographs. The achieved accuracy and performance indicate that GANs can serve as valuable tools for assisting medical professionals in fracture diagnosis. However, challenges related to computational complexity, data availability, and interpretability need to be addressed for broader adoption and real-world application. Future research should focus on optimizing GAN architectures, addressing dataset limitations to further enhance the performance and practicality of GAN models in fracture detection.

VI. CONCLUSION

In conclusion, this article has presented an exploration of Computer-assisted Bone Fractures Diagnosis in musculoskeletal radiographs using Generative Adversarial Networks (GANs). The use of GANs in medical image analysis has shown great potential in aiding clinicians and radiologists in the accurate and efficient detection of bone fractures. By leveraging the power of GANs, we have demonstrated the ability to generate realistic radiographs with fractures, detect anomalies, and improve the overall diagnostic process. Through our research, we have observed promising results in utilizing GAN-based models such as MadGAN, CycleGAN, SkipGANomaly, and SAGAN for fracture detection. These models have demonstrated their effectiveness in generating high-quality images, identifying anomalies, and translating normal radiographs to fractured ones. The performance of these models, although influenced by various factors such as dataset size, training configuration, and preprocessing techniques, has shown significant potential in enhancing fracture diagnosis accuracy and reducing the reliance on manual interpretation. However, it is important to acknowledge the challenges that lie ahead. Further research and development are required to address limitations such as data heterogeneity, model generalization, and interpretability. Additionally, the ethical implications, including patient privacy and the need for human oversight in the diagnostic process, should be carefully considered and addressed. In conclusion, the application of Generative Adversarial Networks for Computer-Assisted Bone Fractures Diagnosis in musculoskeletal radiographs holds great promise. By harnessing the power of GANs, we can improve the accuracy, efficiency, and overall quality of fracture detection, ultimately benefiting both clinicians and patients. Continued advancements in this field have the potential to revolutionize musculoskeletal radiography and pave the way for more effective and precise diagnostic tools in the future.

ACKNOWLEDGMENT

I would like to express my sincere gratitude to my supervisors and colleagues whose invaluable contributions and support have greatly enriched this article. Their guidance and insights have been instrumental in shaping its content and direction.

REFERENCES


TABLE III. COMPARISON OF MODELS BASED ON PERFORMANCE METRICS

<table>
<thead>
<tr>
<th>Models</th>
<th>Accuracy</th>
<th>Precision</th>
<th>F1-Score</th>
<th>Detection Speed</th>
</tr>
</thead>
<tbody>
<tr>
<td>Ganomaly</td>
<td>0.861</td>
<td>0.873</td>
<td>2.863</td>
<td>2.987</td>
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<tr>
<td>SkipGanomaly</td>
<td>0.901</td>
<td>0.903</td>
<td>0.907</td>
<td>5.837</td>
</tr>
<tr>
<td>AttentionGAN</td>
<td>0.842</td>
<td>0.877</td>
<td>0.8417</td>
<td>2.291</td>
</tr>
<tr>
<td>MadGAN</td>
<td>0.954</td>
<td>0.958</td>
<td>0.953</td>
<td>1.418</td>
</tr>
<tr>
<td>SAGAN</td>
<td>0.9</td>
<td>0.905</td>
<td>0.907</td>
<td>0.2</td>
</tr>
<tr>
<td>AnoGAN</td>
<td>0.838</td>
<td>0.848</td>
<td>0.842</td>
<td>3.738</td>
</tr>
<tr>
<td>DCGAN</td>
<td>0.7</td>
<td>0.724</td>
<td>0.705</td>
<td>15.233</td>
</tr>
<tr>
<td>CycleGAN</td>
<td>0.932</td>
<td>0.917</td>
<td>0.928</td>
<td>8.412</td>
</tr>
</tbody>
</table>


Performance Evaluation of Face Mask Detection for Real-Time Implementation on an RPi

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Abstract—Mask-wearing remains to be one of the primary protective measures against COVID-19. To address the difficulty of manual compliance monitoring, face mask detection models considerate of both frontal and angled faces were developed. This study aimed to test the performance of the said models in classifying multi-face images and upon running on a Raspberry Pi device. The accuracies and inference speeds were measured and compared when inferring images with one, two, and three faces and on the desktop and the Raspberry Pi. With an increasing number of faces in an image, the models’ accuracies were observed to decline, while their speeds were not significantly affected. Moreover, the YOLOv5 Small model was regarded to be potentially the best model for use on lower resource platforms, as it experienced a 3.33% increase in accuracy and recorded the least inference time of two seconds per image among the models.

Keywords—Face mask detection; multi-face detection; Raspberry Pi; embedded platform

I. INTRODUCTION

Throughout history, infectious diseases have continuously emerged and evolved due to natural causes and human activities [1][2]. These diseases can affect a significant number of people and even become global health issues. A timely example would be the Coronavirus disease or COVID-19, a highly contagious disease caused by the Severe Acute Respiratory Syndrome Coronavirus 2 or SARS-CoV-2 [3]. It originated in Wuhan, China in December 2019 and was eventually declared by the World Health Organization (WHO) a pandemic from March 2020 to May 2023 [4][5]. In the first 50 days of its onset in China, there were more than 70,000 infected individuals and 1800 deaths recorded [6]. In the current Philippine context, the Department of Health (DOH) has reported a total of about 4.16 million cases of infection, 66,000 deaths, and 4.09 million recoveries in the Philippines as of June 2023 [7].

To combat the spread of COVID-19, several health protocols have been issued by authorities and institutions. Among the common ones are physical distancing, hand hygiene, surface disinfection, proper ventilation, and, especially, mask-wearing [8]. Regarding mask use, the WHO promotes the wearing of non-medical or medical masks in poorly ventilated or crowded indoor settings and public areas with insufficient physical distancing. Moreover, wearing strictly medical masks is recommended for vulnerable populations, potential or confirmed COVID-19 patients, and caretakers of COVID-19 patients [9]. The Centers for Disease Control and Prevention (CDC) recommends mask-wearing in public transportation vehicles and hubs. For people situated in COVID-19 hotspots, the CDC requires the use of face masks, especially for those who are highly susceptible to severe infection [10].

For guidelines against COVID-19 to take full effect, public compliance is essential. However, the extended pandemic duration has resulted in growing complacency and, consequently, disobedience to health protocols [11]. While stricter monitoring of compliance can be helpful, the need for physical distancing and a reduced workforce due to the pandemic makes it challenging. To address this issue, the Ateneo Laboratory for Intelligent Visual Environment (ALIVE) has developed face mask detection models that can classify medically-approved masks, non-medically-approved masks, and unmasked faces in consideration of both forward-looking and angled or side-view face images [12].

The said models were tested using only a desktop computer. In real-time monitoring of mask-wearing compliance, portable devices with relatively lower processing capabilities may be used for convenience. Moreover, only single-face images were considered in both model training and testing. In public settings, multiple faces may be captured by the mask detection models at a time.

With these, the study then aims to evaluate and compare the performance of the developed mask detection models on both desktop and the embedded platform Raspberry Pi. Multi-face mask detection will also be explored by testing the models on combined single-face images from the validation and test sets. It builds on [12] by making the following contributions:

- The real-time performance of robust mask detection models considerate of both frontal and angled faces are examined through deployment on a Raspberry Pi. This helps determine the viability of using the models for live monitoring on low-power systems.
- A comparative analysis of the models’ mask detection capabilities in a desktop computer and a Raspberry Pi is performed. This results in insights into the strengths and limitations of the existing models based on the computational power of the hardware used.
- The models’ performance in detecting multiple masked faces is observed. This helps identify the
suitability of using the models for simultaneous mask detection in crowded settings.

This paper contains the following sections: Section II provides a discussion on the object detection architectures used by the developed models, the robust detection models from [12] involved in this study, and related works on real-time and multi-face mask detection. Section III elaborates on the methods employed in this study, particularly multi-face inference and real-time implementation on a Raspberry Pi. Section IV presents a description of the results and their detailed analysis. Lastly, Section V summarizes the study’s findings and offers recommendations on potential future developments.

II. RELATED WORK

A. Object Detection Architectures

In developing the models involved in this study, the state-of-the-art object detection architectures CenterNet and YOLO, particularly YOLOv5, were used.

CenterNet [13] is characterized by anchorless object detection, which implements a more time- and resource-efficient algorithm in place of the standard Non-Maximum Suppression (NMS) technique. Under this approach, objects are modeled as a single point, corresponding to the center of the bounding box. Searching of center points is done through keypoint estimation, while determining other object properties including the size, location, and pose involves regression. In evaluating the relevance of bounding box predictions, the CenterNet architecture focuses on where their centers are located rather than how much they overlap with the object being detected. Compared to anchor-based detectors, fewer irrelevant detections are generated by CenterNet, resulting in faster inference and less power usage.

On the other hand, YOLO (You Only Look Once) is a state-of-the-art framework that generally operates by dividing an input image into grids, with object detection taking place in each grid. This approach boasts remarkable speed and efficient consumption of computational resources. YOLOv5 [14], one of the latest versions of YOLO, mostly differs from its predecessors in terms of the model backbone, neck, and head. In charge of image feature extraction, a combination of Cross Stage Partial Networks (CSPNet) [15] and Darknet termed CSPDarknet serves as the backbone of YOLOv5. With CSPNet, the issue of duplicate gradients common in large-scale backbones gets resolved, resulting in increased inference speed and reduced model size due to the decline in parameters and floating-point operations per second. Path Aggregation Network (PANet) [16] functions as the model neck, used to create feature pyramids for aggregating and passing features to the model head. It implements a novel feature pyramid structure with an improved bottom-up path that allows for better low-level feature propagation. Overall, PANet helps in locating objects more accurately. For generating predictions and bounding boxes, YOLOv5 retains the model head utilized by YOLOv4, which can perform multi-scale detection [17]. The activation and loss functions of YOLOv5 also set it apart from older YOLO models and contribute to its faster learning and enhanced performance. To deal with the vanishing gradient problem, it uses Leaky Rectified Linear Unit and Sigmoid activation functions. For the loss function, it employs the Binary Cross-Entropy with Logits Loss function.

B. Mask Detection Models and Dataset

As previously stated, this study makes use of the object detection models from [12] which include YOLOv5 Small, YOLOv5 Medium, CenterNet Resnet50 V1 FPN 512x512, and CenterNet HourGlass104 512x512. These models were trained for the image classification task on the relabeled Face Mask Label Dataset (FMLD) curated in [12]. The relabeled FMLD is made to train deep learning models in detecting three mask-wearing classifications, which are Medical Masks, Non-Medical Masks, and No Mask, in consideration of front and side view face images. With this, the six classes represented in the dataset are Front - Medical Mask, Front - Non-Medical Mask, Front - No Mask, Side - Medical Mask, Side - Non-Medical Mask, and Side - No Mask. In the relabeled FMLD, there are 50 images per class which sum up to a total of 300 images. In relation to this, the training of each object detection model involved 300 epochs. The classification accuracies of the models on the test set of the relabeled FMLD were then measured and compared. The models with the highest accuracy were found to be the CenterNet Resnet50 and CenterNet HourGlass104 models, both having an overall accuracy of 95%. The YOLOv5 Medium comes next with an overall accuracy of 93.33%, and the YOLOv5 Small model is the least accurate with 91.67%.

C. Similar Works

The study of ben Abdel Ouahab et al. [18] aimed to develop a mask detection model and evaluate its real-time performance on the Raspberry Pi. Fine-tuning was performed on the pre-trained MobileNetV2 model, constructing a new classification head with five layers: average pooling, flatten, dense with ReLU activation, dropout, and dense with softmax activation. The model was trained on a dataset with 1915 masked and 1918 unmasked face images. It achieved an accuracy of 99% upon testing on the validation set. For real-time implementation, two high-performing laptops, Raspberry Pi 3 and 4 devices, and Raspberry 4 devices with Intel Neural Compute Stick (NCS) 2 were used. The model obtained the highest average FPS value of 4.8 on the Raspberry Pi 4 (8 GB RAM) with NCS 2 among the different versions. It was observed that the performance of the model in terms of speed decreased significantly when deployed on low-power systems rather than on desktop devices.

Moreover, Mohandas et al. [19] focused on creating a real-time face mask detection system running on edge computing devices for access and egress control. For model training, transfer learning was employed on the SSD InceptionV2 model using a GPU-accelerated device. The dataset used for re-training consisted of images from the Real-World Masked Faces Dataset (RMFD), Labelled Faces in the Wild (LFW) dataset, and various web resources. Upon implementation on a Raspberry Pi 4 device, an average detection time of 1.13 ms per frame was obtained. The model also achieved perfect precision for both classes, 89% recall for masked faces, and 91% recall for unmasked faces on the Raspberry Pi 4, which is comparable to its performance on the GPU-enabled computer. While the system is meant to detect only one face at a time, restricting
the face to a certain Region of Interest, it was found to be effective for detecting multiple masked or unmasked faces as well.

Lastly, the study of Reza et al. [20] investigated the face mask detection performance of selected Convolutional Neural Network (CNN) models on mobile IoT devices. New classifiers were added to the MobileNetV2, InceptionV3, VGG16, and ResNet50 models and trained on top of their frozen layers. Model training involved a public dataset containing both single- and multiple-face images of masked and unmasked people. The models were trained four times, using varying ratios of the training data, and tested accordingly on the NVIDIA Jetson TX2 and Jetson Nano. VGG16 had the highest average accuracy across all ratios and for both devices, reaching a peak of 96.07% for 20% training data, but obtained the slowest inference speed. On the other hand, MobileNetV2 performed the best in terms of speed, having the least inference time of 25.10 ms for 5% and 10% training data, but lagged behind in testing accuracy. InceptionV3 achieved promising accuracy results upon being trained and tested on the smallest dataset ratio.

III. METHODOLOGY

This study used two general sets of methods, which are Multi-Face Inference and Real-Time Implementation in Raspberry Pi. In the corresponding subsections, the procedures performed are discussed in greater detail.

A. Multi-Face Inference

To evaluate the multi-face mask detection performance of the models from [12], images with two or three faces must be included in the test set for this study. Since the mask detection models in [12] were trained and tested on the relabeled FMLD, the same dataset was used in building the test set. Single-face images from the validation and test sets in [12] were combined through the image editing tool Photopea to synthetically produce images that contain two or three faces. The images used per multi-face combination came from random classes, but the equal representation of all six classes among the test images with two or three faces was ensured as far as possible. Black pixels were used to fill the empty spaces left in the synthetic multi-face images which were caused by the varying dimensions of component images. Fig. 1 and 2 present sample test images with two and three faces, respectively.

Using the four mask detection models from [12], inference was done on the multi-face images created from the relabeled FMLD. The classification accuracy for each class and the overall classification accuracy were recorded. Moreover, the inference speed was examined to describe the relationship between the number of faces in an image and the speed at which the models classify the image, if any. First, baseline speeds for all four models were determined through the inferencing of single-face images from the validation and test sets used in [12] and the computation of average inference time per image. Then, similar steps were carried out to measure the speeds for inferencing two-face and three-face images using the four models. For the YOLOv5 models, the speeds were automatically printed out after the completion of inference. Conversely, calculating the inference speed for the CenterNet models was done using the timeit package for Python. The desktop computer utilized for testing was equipped with an NVIDIA GTX 1080 Ti GPU.

For further verification of the mask detection models’ classification accuracies on the images with two or three faces, the single-face images that comprise the synthetic multi-face images went through individual inferencing. This process served as a way of confirming that any difference in classification accuracy between an image with a single face and one with multiple faces can be primarily attributed to the number of faces in an image instead of other possible factors. In individually inferencing the faces present in the test images containing two or three faces, the classification accuracy for each class and the overall classification accuracy were recorded.
B. Real-Time Implementation in Raspberry Pi

To test the capabilities of the mask detection models from [12] on lower resource machines, they were exported and run on an embedded system, and their performance in terms of accuracy and speed was measured. This process is aimed at assessing the feasibility of deploying the models on portable or mobile platforms for increased accessibility and convenience of use.

Initially, it was planned to export all four models from [12] to a Raspberry Pi 4 Model B with 4GB of RAM for testing procedures. Unfortunately, there were compatibility issues with the Tensorflow 2 Object Detection API, which facilitated the use of the CenterNet models. Transferring and running the said models on the Raspberry Pi turned out to be challenging due to the format in which they were exported by the Tensorflow 2 Object Detection API. While it is possible to run the CenterNet models on the Raspberry Pi, it was not done successfully for this study. In the end, only the YOLOv5 Small and Medium models were run and tested on the Raspberry Pi. The CenterNet models were excluded since it was taking a significant amount of time to resolve the problems.

In testing the YOLOv5 models on the Raspberry Pi, the classification accuracies were recorded, so that the performance in terms of accuracy when running on a desktop computer and an embedded system may be compared. This was performed using the same methods from the testing phase in [12] and Section III A as well. Inferencing with the YOLOv5 models running on the Raspberry Pi device was performed on the combined validation and test set images from [12]. The class with the highest predicted confidence score was determined to be the predicted class for each image. To compute the accuracy per class in percentage value, the number of correct predictions was divided by the total number of images per class, and the resulting quotient was then multiplied by 100. Python was used for inferencing and calculating the respective classification accuracies of each mask detection model. In particular, the PyTorch package was instrumental in inferencing with the YOLOv5 models.

Aside from their classification accuracies, the inference speeds of the YOLOv5 models upon testing on the Raspberry Pi were also recorded. This process served as a way of determining if the processing speeds of the YOLOv5 models on a low-end computing device fall within an acceptable range. This was performed using similar procedures from Section III A. Measuring of speeds took place while inferencing images with one, two, and three faces from the combination of the validation and test sets used in [12].

IV. RESULTS AND DISCUSSION

A. Multi-Face Inference

A total of 20 synthetic multi-face images were produced from the dataset preparation stage of Section III-A, having ten images with two faces each and another ten images containing three faces each. Table I details the distribution of the six classes found in the relabeled FMLD from [12] among the two-face and three-face images.

Fig. 3, 4, 5, and 6 present the multi-face classification accuracies measured from the YOLOv5 Small, YOLOv5 Medium, CenterNet ResNet50, and CenterNet HourGlass104 models, respectively. Each figure consists of three bar graphs that show the corresponding mask detection model’s accuracies in classifying images with one, two, and three faces. The accuracies for each class present in the relabeled FMLD are specified, as well as the overall accuracy per model. The labels for classification accuracies come in the form of percentages and fractions showing the number of correct predictions over the total of possible predictions. The dotted line found in each figure represents the overall accuracy per model, which is just another way of displaying the values from the bar for overall accuracy. It helps create a better visualization of the changes in accuracy while detecting images with varying numbers of faces. The values for the Single Face Accuracy section per model were taken from the results obtained in [12]. From the figures, it can be observed that the overall accuracy for all models tends to decrease as the number of faces present in an image increases. The YOLOv5 models perform with similar levels of accuracy when classifying two-face images, only suffering from reduced overall accuracies when classifying three-face images.

![Fig. 3. Multi-face classification accuracies of the YOLOv5 small model.](image)

![Fig. 4. Multi-face classification accuracies of the YOLOv5 medium model.](image)
CenterNet models suffer from an approximated 10% loss in overall accuracy. Moving further to detecting images with three faces, the CenterNet Resnet50 model somehow obtains the same level of accuracy compared to its performance on two-face image detection. Conversely, the CenterNet HourGlass104 model incurs an additional 25% decline in overall accuracy when classifying three-face images compared to when classifying single-face images. Among all mask detection models, the CenterNet HourGlass104 model experiences the greatest decline in overall accuracy upon classifying images with multiple faces. This can be possibly attributed to the susceptibility of CenterNet models to overfitting as discussed in the model training procedure in [12]. Overfitting makes it difficult for the said models to perform well on inputs that differ from those that they were trained on, which are single-face images.

Fig. 7 presents the different inference speeds of the models on images with one, two, and three faces. From the graph, it can be observed that the fastest mask detection model, having the least inference time in milliseconds, is the YOLOv5 Small model, followed by the YOLOv5 Medium, then the CenterNet Resnet50, and finally the CenterNet HourGlass104. The arrangement of the models in increasing inference speed corresponds to their arrangement in increasing network size, making the former quite expected. Being the smallest of the four models, the YOLOv5 Small model is relatively computationally lightweight and thus faster to execute. Conversely, the CenterNet HourGlass104 model is computationally intensive and thus slower to run, since it has the largest network size among the models. In testing multi-face detection, the models’ inference speeds incur an initial decrease upon detecting two-face images compared to when detecting single-face images. However, there is negligible change in the speeds of the models when going from detecting two-face images to detecting three-face images. Based on these results, classifying images with even more faces may not have significant effects on the models’ inference speeds. It is also important to note that the speeds specified in Fig. 7 were recorded during the first round of inference on the test images. Upon repeated inferring in several rounds, the speeds measured turned out to be faster, but such speeds were no longer considered.

CenterNet models suffer from an approximated 10% loss in overall accuracy. Moving further to detecting images with three faces, the CenterNet Resnet50 model somehow obtains the same level of accuracy compared to its performance on two-face image detection. Conversely, the CenterNet HourGlass104 model incurs an additional 25% decline in overall accuracy when classifying three-face images compared to when classifying single-face images. Among all mask detection models, the CenterNet HourGlass104 model experiences the greatest decline in overall accuracy upon classifying images with multiple faces. This can be possibly attributed to the susceptibility of CenterNet models to overfitting as discussed in the model training procedure in [12]. Overfitting makes it difficult for the said models to perform well on inputs that differ from those that they were trained on, which are single-face images.

**TABLE I. DISTRIBUTION OF THE SIX CLASSES OF THE RELABELED FMLD FOR THE IMAGES CONTAINING TWO OR THREE FACES**

<table>
<thead>
<tr>
<th>Class</th>
<th>Number of Faces</th>
<th>Images with Two Faces</th>
<th>Images with Three Faces</th>
</tr>
</thead>
<tbody>
<tr>
<td>Front - Medical Mask</td>
<td>4</td>
<td>4</td>
<td></td>
</tr>
<tr>
<td>Front - Non-Medical Mask</td>
<td>3</td>
<td>4</td>
<td></td>
</tr>
<tr>
<td>Front - No Mask</td>
<td>3</td>
<td>6</td>
<td></td>
</tr>
<tr>
<td>Side - Medical Mask</td>
<td>4</td>
<td>5</td>
<td></td>
</tr>
<tr>
<td>Side - Non-Medical Mask</td>
<td>3</td>
<td>6</td>
<td></td>
</tr>
<tr>
<td>Side - No Mask</td>
<td>3</td>
<td>5</td>
<td></td>
</tr>
<tr>
<td>Total</td>
<td>20</td>
<td>30</td>
<td></td>
</tr>
</tbody>
</table>

Note: Ten images containing two faces each equals 20 faces in total. Similarly, ten images containing three faces each equals 30 faces in total.
seen that the inference times were dominated by the inferencing process itself, while the pre-process and Non-Maximum Suppression (NMS) stages took up only small portions. The YOLOv5 models automatically generated the speed breakdown after the inferencing process. Since the CenterNet models have no similar capability, their inference speed breakdown did not get included anymore.

Fig. 8. Breakdown of the multi-face inference speeds of the YOLOv5 models.

Fig. 9, 10, 11, and 12 show the results for the validation of the multi-face classification accuracies of the YOLOv5 Small, YOLOv5 Medium, CenterNet Resnet50, and CenterNet HourGlass104 models, respectively. Each figure is divided into two cluster groups, the first group is for results on two-face images while the second one is for results on three-face images. The first cluster in each group presents the classification accuracies of the corresponding model when individually inferencing the faces that comprise the synthetic multi-face images. On the other hand, the second cluster in each group presents the model’s accuracies upon inferencing the merged images with two or three faces themselves. The results verify that the differences in accuracy when classifying images with one face and those with multiple faces are caused by the changes in the number of faces contained in them. For most of the mask detection models, their classification accuracies were higher when single-face images were inferenced individually and not as components of a synthetic multi-face image. However, the results obtained from the YOLOv5 Small model deviate from the general observation, as its classification accuracies were found to be higher when inferencing images with multiple faces compared to those with only one face. These further support the earlier findings about YOLOv5 models being able to classify two-face images with similar levels of accuracy and only obtaining reduced overall accuracies upon classifying three-face images. The discussion on CenterNet models being more prone to declines in accuracy due to an increase in the number of detected faces also gets further confirmed.

B. Real-Time Implementation in Raspberry Pi

Table II shows the classification accuracies of the YOLOv5 models when tested on the Raspberry Pi. These accuracies are for images with single faces only, as the combined validation and test set images from [12] are used without making modifications. The table presents a direct comparison between the model’s classification accuracy on the desktop computer and its accuracy upon running on the Raspberry Pi. Each table cell corresponds to a particular mask detection model and image class, containing the ratio of the number of correctly predicted images from the class by the model to the total number of images in the class and the equivalent accuracy in percentage form.

From these results, the differences in classification accuracies of the YOLOv5 Small and Medium models when tested on
The YOLOv5 Small model obtained a greater classification accuracy on the Raspberry Pi than on the desktop computer, being able to correctly predict two additional images belonging to the Side - Medical Mask and Side - No Mask classes. The opposite was true for the YOLOv5 Medium model, as it achieved a lower classification accuracy when running on the Raspberry Pi than on the desktop computer. There were fewer correctly predicted images under the Front - Medical Mask, Front - Non-Medical Mask, Side - Non-Medical Mask, and Side - No Mask classes. Overall, there was a minimal difference of 3.33% for the classification accuracies of the YOLOv5 Small model and a larger discrepancy of 8.33% for the accuracies of the YOLOv5 Medium model.

Varying performance metric values for the same deep learning models when run on the Raspberry Pi and on other platforms have also been recorded in [21][22][23]. Unfortunately, the said studies could not offer an explanation behind the discrepancy in accuracies or confidence scores upon testing on different devices, including the Raspberry Pi. Similarly, this study failed to come up with reasons for the difference in the mask detection models’ classification accuracies on the desktop computer and on the Raspberry Pi.

Moreover, Fig. 13 presents the inference speeds of the YOLOv5 models when ran on the desktop machine versus when ran on the Raspberry Pi. Overall, the YOLOv5 Small model turns out to be the most ideal for deployment on low-end computing devices. There was only a small difference in the classification accuracies of the YOLOv5 Small model when running it on the Raspberry Pi and on the desktop computer. As seen in Table II, the YOLOv5 Small model obtained the highest overall classification accuracy when tested on the Raspberry Pi. Furthermore, less discrepancy in the said model’s inference speeds can be observed upon inferencing on the desktop computer then on the Raspberry Pi. The YOLOv5 Small model achieved an inference time of about two seconds per image on the Raspberry Pi, still falling within the acceptable range. Conversely, the YOLOv5 Medium model obtained an inference time of about five seconds per image, already considered to be somehow too slow.

### Table II. Comparison of Classification Accuracies on Desktop versus on Raspberry Pi

<table>
<thead>
<tr>
<th>Model</th>
<th>Front - Medical Mask</th>
<th>Front - Non-Medical Mask</th>
<th>Front - No Mask</th>
<th>Side - Medical Mask</th>
<th>Side - Non-Medical Mask</th>
<th>Side - No Mask</th>
<th>Overall</th>
</tr>
</thead>
<tbody>
<tr>
<td>YOLOv5 Small</td>
<td>9/10 Accuracy = 90%</td>
<td>9/10 Accuracy = 90%</td>
<td>10/10 Accuracy = 100%</td>
<td>9/10 Accuracy = 90%</td>
<td>10/10 Accuracy = 100%</td>
<td>8/10 Accuracy = 80%</td>
<td>5/6 Accuracy = 83.33%</td>
</tr>
<tr>
<td>YOLOv5 Medium</td>
<td>10/10 Accuracy = 100%</td>
<td>8/10 Accuracy = 80%</td>
<td>10/10 Accuracy = 100%</td>
<td>8/10 Accuracy = 80%</td>
<td>10/10 Accuracy = 100%</td>
<td>10/10 Accuracy = 100%</td>
<td>5/6 Accuracy = 83.33%</td>
</tr>
</tbody>
</table>

**Bold = Highest Value in Column**

The YOLOv5 Small model and Medium models had inference times on the desktop computer to obtain the speedup values of 137.28, 57.83, and 59.08 upon inferencing single-face, two-face, and three-face images, respectively. On the other hand, the YOLOv5 Medium model obtained speedup values of 137.28, 57.83, and 59.08 upon inferencing single-face, two-face, and three-face images, respectively. Both models experienced the highest speedups with single-face images, which aligns with how the YOLOv5 Small and Medium models had lower inference times on the desktop computer when inferencing images with one face than those with two or three faces. Meanwhile, the YOLOv5 Small and Medium models had inference times of similar levels for single-face, two-face, and three-face images on the Raspberry Pi.

Fig. 12. Validation of the Multi-Face classification accuracies of the CenterNet HourGlass104 model.

Fig. 13. Multi-face inference speeds of the YOLOv5 models when ran on the desktop computer and the Raspberry Pi.
V. Conclusion

Generally, the mask detection models were found to be capable of multi-face detection, although the accuracies were observed to decline in the presence of more faces in an image. In terms of inference speed, the YOLOv5 models performed faster than the CenterNet models due to their smaller network sizes. Moreover, it was observed that the number of faces in an image did not significantly affect the models’ inference speeds.

For the implementation on the Raspberry Pi 4 Model B embedded platform, only the YOLOv5 Small and Medium models were used. Going from inferencing on the desktop computer to the Raspberry Pi, the YOLOv5 Small model experienced a 3.33% increase in classification accuracy, while the YOLOv5 Medium model suffered from an 8.33% decrease in accuracy. In terms of inference speed, the YOLOv5 models generally exhibited a slower mask detection performance on the Raspberry Pi than on the desktop computer, with the YOLOv5 Small model having an inference time of about two seconds per image and the YOLOv5 Medium model obtaining an inference time of about five seconds per image. Furthermore, both models recorded the highest speedup values when inferencing single-face images, just as they achieved lower inference times with single-face images on the desktop computer compared to multi-face images. Considering both the classification accuracy and inference speed, the YOLOv5 Small model was regarded to be potentially the best model for use on lower resource platforms.

Future work may involve the use and evaluation of other object detection models for the robust face mask detection task. Developed models can also be tested on several kinds of embedded systems, such as the Jetson Nano. It might also be worth looking into the addition of more mask-wearing categories, including incorrectly masked faces, among others.

References


A Hybrid TF-IDF and RNN Model for Multi-label Classification of the Deep and Dark Web

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Abstract—The classification of content on the deep and dark web has been a topic of interest for researchers. Researchers focus on adopting more efficient and effective classification methods as the data available on deep and dark web platforms continues to grow. Multi-label classification is the approach for simultaneously categorizing content into multiple classes. To address this, a hybrid approach combining Term Frequency-Inverse Document Frequency (TF-IDF) and Recurrent Neural Network (RNN) has been proposed. The approach involves preprocessing a dataset of Hypertext Markup Language (HTML) documents, selecting specific HTML tags to generate embeddings using TF-IDF, and using an RNN model for multi-label classification. The proposed model was evaluated against commonly used methods (Binary Relevance, Classifier Chains, and Label Powerset) using precision, recall, and F1-score as evaluation metrics, demonstrating promising results in accurately classifying data from the deep and dark web. This contribution represents a noteworthy advancement for researchers and analysts working in this field.

Keywords—Deep web; dark web; multi-label classification; TF-IDF; FastText; RNN

I. INTRODUCTION

A deep web is a portion of the internet not accessible by traditional search engines. A subsection of the deep web, the dark web, is known for its anonymity and association with illegal activities, while it requires specialized software to access [1]. An encrypted network called Tor, or "onion routing," is often used to access the dark web like the Tor browser. The Tor Network, a free, open-source software platform, enables anonymous communication. To provide anonymity to its users, Tor encrypts their internet traffic before it reaches its final destination and bounces it around multiple servers.

To comprehend cyber threat intelligence from the dark web, researchers collect deep and dark web data using web crawlers. Deep web crawler collects data from websites not listed in traditional search engines. Deep web data collection aims to find resources, often valuable, and provide insight into various domains, including science, medicine, and finance. Databases and APIs are examples of deep web content that may contain structured data suitable for data integration. In addition, researchers can use deep web data to study the behavior of users in online communities and analyze trends in e-commerce [2]. It is essential to use deep web data from vast and diverse corners of the internet that are difficult to access conventionally. Text classification is the primary method for inferring information from deep and dark web pages crawled to analyze content. Using text features, researchers propose a method to classify Deep Web data sources [3].

Since the dark web is anonymous, criminals can engage in these activities without fear of being caught, causing a proliferation of criminal enterprises. As a result, many illegal activities are happening on the dark web, including drug trafficking, arms trafficking, human trafficking, and selling stolen data. However, a challenge is associated with studying the dark web because of its unique characteristics [4]. First, due to the encrypted nature of the dark web, identifying users and tracking data origins is very difficult. Consequently, tracing the source of criminal activities and cyber threats is challenging for law enforcement agencies and researchers.

The dark web has been investigated for its content in recent years. Law enforcement agencies and cybersecurity experts can detect and prevent criminal activities using machine learning approaches on the dark web, where there is vast data. Researchers discuss how the dark web information they find can be assessed for its relevance, usefulness, and appropriateness [5]. Researchers have used Machine Learning (ML) approaches to investigate content on the dark web. Researchers can better understand how activities on the dark web are patterned by analyzing individual use cases and different forms of data. For example, text and image data can be automatically classified and analyzed using algorithms to detect potential illicit activities, such as illegally selling goods and services [6]–[8].

Using machine learning algorithms and external knowledge sources maximizes the efficiency and effectiveness of identifying and categorizing deep and dark web content. However, the lack of access to these deep and dark webs makes research into their nature and structure challenging. One standard method is to manually label crawled sites per a series of categories and then use this corpus as a training corpus for automated crawling in the future. While this approach has its benefits, it has limitations because it is time-consuming, expensive, and valuable only for specialized tasks. Also, related research mainly focuses on classifying content with a single label. As per the authors’ knowledge, the presented work is the first attempt to label deep and dark web content with multi-label classification.

The contribution of this study lies in the methodology proposed, which provides a comprehensive and systematic approach for labeling hidden content on the deep and dark web with multi-label. This approach starts with a dataset of HTML documents scraped from the deep and dark web. Then, as part of the preprocessing, specific HTML tags are selected, irrelevant characters are removed, and embeddings are generated using TF-IDF. In the next step, FastText assigns labels to documents according to their similarity. Text classification and language modeling can be accomplished with FastText, a
popular machine-learning algorithm.

Recurrent Neural Networks (RNNs) are trained on pre-processed datasets after the documents have been labeled. In particular, RNNs are well suited to dealing with sequential data, such as text. This study compared the proposed method to three existing methods: Binary Relevance (BR), Classifier Chains (CC), and Label Powerset (LP). Precision, recall, and F1-Score are used as evaluation metrics to measure the performance of the proposed approach. The proposed approach for multi-label classification of deep and dark web content has significant implications for enhancing security and combating criminal activities on these platforms.

The following paper is organized into four sections: Related Work, Proposed Approach, Results and Discussion, and Conclusion. Section II, related work covers the need for deep and dark web content classification and existing multi-label classification approaches. Section III, proposed approach explains the methodology used, while Section IV, Results and Discussion, presents the findings. As a final section of the paper, Section V, the conclusion section summarizes and suggests future research directions.

II. RELATED WORK

Information obtained from deep web sources often lacks structure, making it challenging to categorize and analyze. Instead, specialized algorithms must be used to extract text features to classify web pages in this vast and complex world of the deep web. As part of their extraction process, these algorithms use various techniques, including keyword encoding, topic modeling, sentiment analysis, and entity recognition, to identify meaningful features in the text. The algorithms can extract valuable insight from the text by categorizing it based on the extracted features. For example, in the context of mobile app stores, the paper highlights the importance of text feature classification algorithms for collecting and analyzing deep web data [9].

Deep web crawlers use their classification modules to understand the context of the data they collect. For example, through text classification algorithms, the classification module provides insight into the actual content of a web-page. The researchers proposed an Accurate crawler to harvest deep web content with accurate classification [10]. By ranking sites based on the similarity of their content, the framework attempted to reduce the number of pages that need to be visited. Consequently, deep web content can be extracted and classified more accurately. In addition, researchers proposed a smart crawler to search the deep web efficiently while avoiding irrelevant pages [11]. Dark web content classification is necessary because these parts of the internet are often used for illegal and criminal activities, like drug trafficking, drug sales, and cybercrime. Natural language processing techniques are used to classify texts on the dark web, and supervised machine learning algorithms are used to classify dark web content.

Researchers proposed an Automated Tool for Onion Labeling (ATOL) for mapping dark web content to thematic labels [12]. In this approach, popularity scores for different categories were calculated using TF-IDF. Three components comprised the ATOL system: a module that finds keywords for different categories, a classification framework that maps onion content to categories when labeled data is available, and a clustering framework that categorizes onion content using external knowledge sources when labeled data is lacking.

The researchers proposed a crawler to search dark web links for markets [13]. A dataset was created and pre-processed using data-cleaning techniques. Linear Support Vector Machines outperformed Random Forests and Nave Bayes in classifying dark web pages. The proposed system effectively identified and classified dark web pages with high accuracy, precision, recall, and F1 score. In addition, researchers proposed a modified frequency-based term weighting scheme for identifying dark web content [14]. In a dataset selected from the dark web Portal Forum, the proposed term weighting scheme was compared to Term Frequency (TF), Term Frequency-Inverse Document Frequency (TF-IDF), and Term Frequency-Relative Frequency (TF.RF). Experimental results demonstrate that the proposed scheme outperforms other term weighting techniques based on classification accuracy and other evaluation measures.

Researchers analyzed using the Vector Space Model with two-term weighting schemas: TF and TF-IDF, to determine some of the most discussed topics within Arabic dark web forums [15]. In addition, researchers proposed a new method of tagging extracted data to provide a more balanced and effective method for detecting text features [16]. ELEMENT, a modified TF-IDF algorithm, considered several factors, including document length and term length. One study applied the TF-IDF to classify dark web marketplaces according to their offered products [17]. In addition, researchers identified language characteristics of dark web forums related to drug markets using TF-IDF [18]. Researchers confirmed the similarity between the subjects of dark web forums with affect analysis [19]. Focused topics identified in forums were piracy, hacking, drugs, politics, revolution, weapons, and guns.

Researchers evaluated some text embeddings and classifiers for classification tasks in the darknet domain [20]. In the study, researchers compared text classification to keyword searches by training the classifier using keyword search results. The study concluded that text classification performed better than a keyword-based search. Although dark web content is typically classified using a single label, multiple-label classifications can be helpful in cases where content belongs to multiple categories. However, dark web websites or contents are typically categorized into one of several categories, such as drugs, weapons, or hacking. Therefore, multi-label classification is helpful for dark web content that may be classified with multiple labels. For example, dark web data can be analyzed for multi-label classification tasks based on multiple labels such as ‘fraud,’ ‘drugs,’ and ‘weapon’ Researchers reviewed three significant areas of multi-label learning: paradigm formalization, learning algorithms, and learning environments [21]. Besides defining multi-label learning and evaluating its outcomes, the research examined traditional and deep learning-based algorithms for multi-label learning. The empirical results also discussed how different learning settings, such as feature selection and transfer learning, affect the learning process and algorithm performance.

Multi-label classification, in combination with other machine learning tasks, has led to the development of two
main categories of methods: Problem Transformation (PT) and algorithm adaptation [22]. Several PT methods are independent of the algorithm and involve transforming a multi-label classification task into a single-label classification, regression, or ranking task. Methods such as BR, CC, and LP fall into this category. The review provided valuable insights and techniques for multi-label learning, but it is necessary to adapt further and study these techniques for dark web data classification.

The presented work has been proposed to classify dark web data using techniques such as BR, CC, and LP. By including label interdependence and using probabilistic models to predict label combinations, these methods can improve the accuracy of multi-label classification.

BR is a decomposition method that assumes labels are independent and trains binary classifiers separately to determine the Relevance of each label [23], [24]. It has been proven that BR produces good machine learning classifiers both computationally and as a result of several metrics. An example of the application of BR is covered in the study, which developed a novel method for multi-label text classification for Arabic texts using binary relevance [25]. The study examined five multi-label classification approaches for enhancing Arabic multi-label text classification, including Support Vector Machines (SVMs), k-Nearest Neighbors (KNNs), Naive Bayes (NBs), and different classifiers.

Aside from BR, other multi-label classification models have also been developed to deal with label dependency. As one example, the CC method is becoming increasingly popular due to its simplicity and promising results [26]. As a method for solving multi-label learning problems, classifier chains consist of chaining together off-the-shelf binary classifiers in a directed structure [27]. Then, the label predictions are used to refine other classifiers. Various datasets and evaluation metrics have been used to evaluate the effectiveness and flexibility of this method.

There has been an alternative method developed to address label dependency in a multi-label classification called LP. The LP transformation aims to transform multi-label learning problems into single-label multi-class problems by transforming these into one-label multi-class problems [28]. By using LP transformations, all label combinations present in the original dataset are transformed into one label.

Multi-label classification has gained attention recently due to its application in solving complex real-world problems [29]. For example, multi-label classification is helpful in text classification, image annotation, and bio-informatics scenarios with multiple labels associated with each instance. In addition, using multi-label classification and feature selection techniques is an upcoming approach to identifying the most relevant features. Selecting features aims to reduce the number of irrelevant features while keeping only the most informative ones. Removing noisy or irrelevant features can enhance the performance of multi-label classification models by reducing the input space dimension.

Using multi-label classification for multitask learning is another example of combining multi-label classification with other machine learning tasks. A multitask learning model uses a single model to learn multiple related tasks simultaneously. However, as each document may contain multiple correlated labels, it is challenging to classify text using multiple labels. Therefore, the researchers introduced a new multi-label text classification technique, learning feature combinations from documents and labels while reinforcing label correlations [30]. As a result, researchers avoid label order dependency and ensure that label correlations are effectively learned. Applying the multi-label text classification technique for dark web content is feasible. The dark web contains a wide range of diverse and often hidden content, making its identification and classification particularly challenging. However, with the multi-label text classification approach, the model can learn to capture correlations between labels, even if they were not explicitly observed in the training data. This enables the possibility of improving dark web multi-label classification.

The proposed study presents a comprehensive and systematic approach to multi-label classification of dark web text. The method involves several steps: first, collecting HTML documents from the dark web, then preprocessing them by selecting relevant content, and generating embeddings using the TF-IDF method. Next, labels are assigned to the documents based on their similarities using FastText. Finally, an RNN is trained using these labels. The model’s performance for multi-label classification problems is evaluated using Precision, Recall, and F1-Score. Section III provides a detailed description of the proposed approach.

### III. PROPOSED APPROACH

The following discussion covers the proposed procedure for selectively picking and labeling the content of deep and dark web HTML tags based on their frequency and Relevance using the TF-IDF and FastText embeddings. To multi-label a text document, the authors propose combining TF-IDF and FastText. The TF-IDF statistic evaluates the importance of a term in a document by considering its frequency and inverse document frequency [31]. FastText, on the other hand, is a neural network-based approach that captures the semantic meaning of words through word embeddings [32].

The proposed method extracts TF-IDF features from textual data and trains a FastText model on top of these features. The trained model can then predict multiple labels for a single text instance. This method has yielded promising results in various multi-label classification tasks, including text categorization and sentiment analysis. The combination of TF-IDF and FastText yields a robust and efficient solution for multi-label textual data classification. The authors proposed neural network architecture for binary classification. The proposed architecture is a feed-forward neural network, a Multi-Layer Perceptron (MLP). The network is implemented using the Sequential API, which allows for the easy creation and addition of layers in a linear stack. Fig. 1 illustrates the model architecture along with crucial details regarding layer activations and layer names. The diagram provides a visual representation of the network’s structure, showcasing the flow of information through the layers and the respective labels assigned to each layer.

The architecture consists of two dense layers. A dense layer, also known as a fully connected layer, is one in which every neuron in the layer is connected to every neuron in the previous layer. The first dense layer has 50 units, an
input dimension of 300, and a Rectified Linear Unit (ReLU) activation function. The output of the sigmoid function, when applied to the input features, will be a value between 0 and 1, which can be interpreted as the probability of the input features belonging to the positive class. The network is then compiled using the Adam optimizer, a binary cross-entropy loss function, and accuracy as the evaluation metric. Adam is a stochastic gradient descent optimizer that adapts the learning rate for each parameter. It is computationally efficient and has been demonstrated to work well in various tasks. The binary cross-entropy loss function is a loss function that is often used for classifying things into two groups. It measures the dissimilarity between the predicted probability distribution and the true distribution.

A. Proposed Algorithm

The authors proposed an algorithm for multi-label text classification of dark web data. Algorithm 1 describes the proposed algorithm.

**Multi-Label Text Classification Algorithm**

Further authors describe each step in the algorithm in the following discussion.

1) **Data cleaning**: Three main steps are involved in cleaning and preprocessing HTML tags’ content. The first step is the selection of relevant HTML tags. A second step involves removing accented and non-ASCII characters, stopwords, languages other than English, and punctuation, which add little value to the content and may cause errors. The final process standardizes and normalizes the data by converting all the remaining content to lowercase and separating it into individual tokens and words. As a result, these steps enable further analysis of the HTML tags’ content.

2) **Performing TF-IDF on the data**: The next step of the algorithm is to perform the TF-IDF on the data. The TF-IDF is calculated as the product of the TF, which is the number of times a token appears in a document, and the IDF, which is the logarithm of the ratio of the number of documents in the corpus to the number of documents containing the token. The TF-IDF assigns higher weights to the tokens that frequently occur in a document but infrequently in the corpus and lower weights to the tokens that frequently occur in the corpus but infrequently in a document. The TF-IDF can help filter out the common or irrelevant tokens and highlight the specific or distinctive ones.

3) **Picking tokens having the highest TF-IDF weights in the document**: predefined categories or domains to which the researcher wants to assign the picked tokens based on their meaning or context. The generalized tokens may be selected or created by the researcher based on the research question, the scope, or the analysis criteria. The similarity is measured using a distance or a similarity metric, such as Cosine similarity or Euclidean distance, between the embeddings of the picked token and the generalized token. The similarity reflects the degree or the probability of the picked token belonging to the
The further step is to pick the tokens having the highest TF-IDF weights in the document based on the selection criteria. The selection criteria are designed to balance the tokens' importance or Relevance and the diversity of representativeness of the tokens. The selection criteria consist of three cases:

- If more than twenty tokens are in the document, pick the top 60% of the tokens, sorted by the TF-IDF weights in descending order.
- If in the document, tokens are between six and twenty, pick the top 70% of the tokens, sorted by the TF-IDF weights in descending order.
- If there are fewer than six tokens in the document, include all the tokens.

The authors use the selection criteria to identify the most relevant and significant tokens for the analysis while avoiding redundant data and over-representation. Also, the selection criteria aim to maintain the overall context and subtleties of the document while capturing its primary or specific aspects.

4) Using FastText to generate embeddings of the picked tokens from the document: The next step of the algorithm is to use FastText, a library for efficient text classification and representation learning, to generate the embeddings of the picked tokens from the document. The embeddings are dense, low-dimensional, continuous, and real-valued vectors representing the tokens' semantics or meaning in a multi-dimensional space. The embeddings are trained or learned from a large dataset using a supervised or unsupervised learning algorithm, such as skip-gram or CBOW. The embeddings capture the tokens' co-occurrence or context and the relationships or similarities between the tokens. The embeddings can be used to measure the similarity or the distance between the tokens or to classify or cluster the tokens.

5) Calculating the similarity of every picked token with every generalized token from the list: The next step of the algorithm is to calculate the similarity of every picked token with every generalized token from the list using the embeddings. The generalized tokens are the predefined categories or domains to which the researcher wants to assign the picked tokens based on their meaning or context. The generalized tokens may be selected or created by the researcher based on the research question, the scope, or the analysis criteria. The similarity is measured using a distance or a similarity metric, such as Cosine similarity or Euclidean distance, between the embeddings of the picked token and the generalized token. The similarity reflects the degree or the probability of the picked token belonging to the generalized token.

6) Assigning the generalized token with the highest similarity score to the token: The further step is to assign the generalized token with the highest similarity score to the token based on the assignment criteria. The assignment criteria are designed to ensure the assignment's reliability or confidence and to handle exceptions or special cases. The assignment criteria consist of two cases:

- Only assign the generalized token to the picked token, if the similarity is more than 60
- Manually assign certain labels, such as "onion" or "tor", to the generalized token "network" if the picked token belongs to those labels. The assignment criteria aim to assign the picked token to the most suitable or the most likely generalized token while avoiding the assignment's errors or ambiguities. The assignment criteria also aim to respect the dark web's particularities or conventions and avoid misusing or abusing the generalized tokens.

7) Creating a set of assigned generalized tokens per document: The next step is to create a set of assigned generalized tokens per document by collecting the generalized tokens assigned to the picked tokens in the document. The set of assigned generalized tokens per document reflects the labels or the topics of the document. It can be used as the input or the output of a classification task, such as topic modeling or sentiment analysis, or as the input or the output of a recommendation or search system. The set of assigned generalized tokens per document represents the document's main or specific aspects or themes and the context or nuance of the document.

8) Splitting the dataset into train and test sets: The dataset has been divided into training and testing sets using a predetermined ratio or a sampling method. The training set is used to fit or train the classifier neural network model, and the testing set is used to evaluate or test the performance of the classifier neural network model. The ratio of the split method should be chosen based on the dataset's size, quality, or representativeness and the purpose or objective of the analysis. In this paper, authors used a ratio of 25% testing set and 75% training set.

9) Fitting classifier neural network model: Next, a neural network model is trained using the training set using generalized tokens as labels. The classifier neural network model is an artificial neural network used to classify or predict the labels of the input data based on its patterns or features. The classifier neural network model consists of an input layer, two hidden layers, and an output layer.

To train the classifier neural network model, the following hyperparameters are used:

- Sequential Classifier: The neural network architecture is designed in a sequential manner, where each layer is added one after the other.
- Activation Functions:
  - ReLU (Rectified Linear Unit): Used as the activation function for the hidden layers to introduce non-linearity into the model.
  - Sigmoid: Used as the activation function for the output layer in multilabel classification to produce probabilities for each label independently.
- Optimizer: The Adam optimizer is employed to minimize the loss function during training. Adam is a popular optimization algorithm that combines the benefits of both AdaGrad and RMSProp.
- Loss Function: Binary Cross-entropy is utilized as the loss function for multilabel classification. This loss function is well-suited for problems where each input
sample can belong to multiple classes, as is the case in multilabel classification tasks.

- **Hidden Layers:** The classifier neural network model includes two hidden layers, which contribute to the network’s ability to learn complex representations from the data.
- **Learning Rate:** The learning rate is set to 0.004. The learning rate is a hyperparameter that determines the step size at each iteration of the optimization algorithm. A higher learning rate can result in faster convergence but may risk overshooting the optimal solution, while a lower learning rate can provide more stability during training but might take longer to converge.

During the training process, the classifier neural network model learns the mapping or relationship between the input data and the output labels by adjusting the weights or biases of the connections between the layers. This allows the model to handle the complexity or variability of the data and generalize or adapt to new or unknown data.

10) **Evaluating the model using the test set:** The twelfth and final step of the procedure is to evaluate or test the performance of the classifier neural network model using the testing set and the generalized tokens assigned as the labels. The evaluation is performed by comparing the predicted labels of the model with the actual labels of the testing set, using evaluation metrics such as accuracy, precision, recall, or F1 score. The evaluation metrics measure the quality or the reliability of the model’s predictions and provide insights or feedback for the improvement or optimization of the model.

The algorithm creates a clean HTML document by cleaning and preprocessing the HTML tags’ content. On the data, TF-IDF is performed. TF-IDF weights are used to select tokens. The algorithm generates a FastText embedding. Based on a list of generalized tokens, it calculates the similarity between each token. This algorithm assigns the token to the generalized token with the highest similarity score. Per the document, it creates a set of generalized tokens. Train and test sets are separated in the dataset. Algorithms are designed to extract the main or specific elements of a document.

Further, Section IV presents the result and discussion of the proposed algorithm.

**IV. RESULT AND DISCUSSION**

**A. Dataset Description**

For a proposed work, data was gathered using a custom deep and dark web crawler. In order to explore Tor’s hidden services, the crawler used seeds provided by the Hidden Wiki page. The crawler continued to collect data as it encountered new links. An extensive deep and dark web dataset was collected as a result. The dataset comprised fifty thousand web pages from the deep and dark web. Because the dataset included HTML code for every web page, the file size increased significantly. The crawled files were cleaned by removing HTML tags, JavaScript, and non-English web pages. After the keywords have been extracted from each website, the cosine similarity between these keywords and a set of custom keywords has been calculated. The predefined labels in the dataset are Business, Cybersecurity, Education, Entertainment, Finance, Food, Health, Literature, Nature, Network, Politics, Security, and Shopping.

The evaluation phase includes the examination of model accuracy and loss graphs, which are presented in Fig. 2 and Fig. 3, respectively. These graphs serve as important tools to assess the model’s performance and identify potential areas for improvement.

Fig. 2 and Fig. 3 illustrate the model’s performance over time, visually representing accuracy and loss metrics. By closely monitoring these metrics, adjustments can be made to address issues such as overfitting or underfitting, ensuring the model’s optimal performance and generalization to unseen data.

![Fig. 2. Model accuracy graph for the proposed algorithm.](image1)

![Fig. 3. Model loss graph for the proposed algorithm.](image2)

**B. The Empirical Study of Multi-label Classification with BR, CC and LP Algorithms**

Empirical studies collect and analyze data to gain insights and draw conclusions. The performance of different algorithms can be compared in the context of multi-label classification of dark web data, and the best algorithm can be determined through empirical research. In the context of multi-label classification of dark web data, an empirical study compares the performance of various algorithms such as BR, CC, and LP, and a proposed algorithm on the same dataset.

BR is a widely used baseline method for multi-label classification tasks. It treats each label as an independent binary classification problem and trains a separate binary classifier for each label. During the testing phase, each classifier predicts
the presence or absence of its corresponding label. Then, the predictions are combined to generate the final set of labels for the input sample. Binary Relevance is a simple and efficient method, but it ignores the potential dependencies between labels and may not perform well when labels are highly correlated.

CC is a method that considers the label dependencies by creating a chain of binary classifiers. In this approach, the first classifier is trained on the input data, and each subsequent classifier is trained on the input data concatenated with the predictions of the previous classifiers. The order of the classifiers in the chain can be randomized or chosen based on some heuristic. Classifier Chains can capture the dependencies between labels and improve classification performance compared to Binary Relevance.

However, it requires more training time and is sensitive to the classifiers’ order. Label Powerset is another method that considers the label dependencies by transforming the multi-label problem into a multiclass problem. In this approach, each unique combination of labels is treated as a separate class, and a classifier is trained to predict the class of each input sample. The Label Powerset method can handle many labels but can be computationally expensive due to the many possible label combinations. It may also suffer from the ”curse of dimensionality” if the number of labels is too large.

The proposed algorithm is compared with BR, CC, and LP for multi-label classification, considering some potential factors for comparison:

1) Performance A proposed approach’s performance will be compared with other algorithms as the most critical factor. The performance of different algorithms can be compared using evaluation metrics such as precision, recall, F1-score, accuracy, and others. The proposed approach may perform better or worse depending on the dataset and the characteristics of the problem.

2) Scalability The scalability of an algorithm is another factor to take into consideration. For example, there can be scalability issues with the LP algorithm, while the BR algorithm is generally more scalable when the number of labels is large.

3) Interoperability When the results need to be understood by humans, interpretability is an essential factor. Since CC model the dependency between labels, it may be easier to interpret than other algorithms.

4) Complexity In addition, complexity can be one of the factors to consider, affecting training times, memory requirements, and other aspects of the algorithm. As a result, there may be differences between the proposed approach and other algorithms in complexity.

5) Data Distribution Data attributes such as sparsity, imbalance, and noise can impact algorithm performance. Therefore, the proposed approach may have specific strengths or weaknesses based on the data distribution type.

Model performance for multi-label classification of dark web data with BR, CC, and LP algorithms:

Fig. 4, 5, and 6 illustrate the performance of the BR, CC, and LP algorithms in terms of accuracy and loss during both the training and testing phases. These figures provide valuable insights into the behavior of each algorithm over time, enabling a thorough analysis of their effectiveness and identifying potential areas for improvement in both training and testing scenarios.

In the training and testing phases, accuracy and loss graphs are helpful indicators of the model’s performance. High accuracy and low losses indicate that the model is learning and improving. Furthermore, when assessing the overall performance of a model, other metrics, such as precision, recall, and F1
TABLE I. COMPARISON OF PRECISION (PRC), RECALL (REC), AND F1 SCORE OF PROPOSED ALGORITHM (PRP ALG), BR, CC, AND LP

<table>
<thead>
<tr>
<th>Algorithm</th>
<th>Prc</th>
<th>Rec</th>
<th>F1</th>
<th>Prc</th>
<th>Rec</th>
<th>F1</th>
<th>Prc</th>
<th>Rec</th>
<th>F1</th>
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<th>F1</th>
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<td></td>
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<td>0.78</td>
<td>0.72</td>
<td>0.77</td>
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</tr>
</tbody>
</table>

Fig. 6. Model accuracy and loss graphs for LP algorithm.

score, should also be considered.

C. Performance Evaluation of Algorithms

Comparison of precision, recall, and F1 scores provide insights into which algorithm performs better regarding identifying relevant labels. The F1 score is a harmonic mean of precision and recall. The accuracy of the predictions measures precision and recall by the ability to identify all relevant positive instances, while the precision of the forecasts measures recall.

Table I shows that comparing the proposed algorithm to the other three algorithms (BR, CC, and LP) for three labels (Business, Cybersecurity, and Education), the proposed algorithm has higher precision, recall, and F1 scores. Hence, the proposed algorithm outperforms the other algorithms for these labels. In addition, the proposed algorithm appears more accurate at identifying relevant instances within Entertainment than the other three algorithms.

In Finance, F1 scores for the baseline algorithm were the highest, while precision scores were the highest for the proposed algorithm. The proposed algorithm performs better than the other two algorithms in all but precision. Regarding Food, all algorithms performed well, with high precision, recall, and F1 scores. According to the proposed algorithm, the recall score was the highest, while the precision and F1 scores were the highest for the baseline and LP algorithms. BR algorithm had the highest recall and precision scores in Health, while the proposed algorithm had the highest precision and F1 scores.

For the Literature label, BR, LP, and the Proposed algorithm correctly identified labels for this category. For Nature, with F1 scores of 0.93 and 0.82, respectively, the Proposed Algorithm and LP perform well, while CC performs poorly with 0.80. In the Network category, the Proposed Algorithm, BR, and LP all achieve high F1 scores of 0.86, whereas CC achieves a slightly lower F1 score of 0.80. All algorithms perform reasonably well in this category.

In the politics category, the Proposed algorithm and BR scored 0.40, indicating they could not correctly identify the labels. On the other hand, there was a slight improvement in the F1 scores of LP and CC, respectively, with 0.43 and 0.41. In Security, a perfect score of 1.0 was achieved by all algorithms for precision, recall, and F1, indicating that all assigned labels were identified correctly. For the category of Shopping, BR, and LP all achieve high F1 scores of 0.86, whereas CC achieves a slightly lower F1 score of 0.80. All algorithms perform reasonably well in this category.

Overall, the algorithms perform differently in different categories. However, most categories show that the Proposed algorithm and LP are better at identifying the labels for the given text data than BR and CC. The algorithm’s poor performance in the Politics category demonstrates that category’s characteristics affect the algorithm’s effectiveness.

The number of labels per sample significantly affects
multi-label classification using the proposed approach. Models may be unable to capture the complex relationships between features and labels when the number of labels is too low. This may cause the model to underperform on the test set and not generalize well. On the other hand, a model may suffer from the curse of dimensionality if there are too many labels per sample, which creates a very complex feature space. As a result, the model may struggle to learn effectively from the data, leading to poor performance. As a result, the optimal performance of the proposed approach depends on balancing the number of labels per sample.

V. CONCLUSION

The proposed work introduces a multi-label text classification approach to categorize deep and dark web content, aiming to predict multiple labels for a given text. Four machine learning algorithms were compared: the proposed algorithm, BR, CC, and LP. Evaluation metrics including precision, recall, and F1 score were used to assess their performance. The proposed algorithm exhibited significantly higher precision, recall, and F1 scores compared to the other three algorithms. Additionally, the study highlights the influence of the number of labels per sample on multi-label classification performance. Balancing the number of labels per sample is crucial to avoid poor results caused by either too few or too many labels per sample, which can lead to difficulties in capturing relationships between features and labels or the curse of dimensionality, respectively. In conclusion, this research proposes an efficient multi-label classification model for deep and dark web content analysis, demonstrating superior performance compared to existing methods, with potential applications in cybersecurity and law enforcement. Furthermore, the insights gained regarding the impact of the number of labels per sample can guide the development of future multi-label classification models. The multi-label text classification approach also enhances the model’s capabilities to simultaneously learn entity recognition, relation extraction, and other related tasks, thus improving its overall performance.

REFERENCES


Abstract—This paper presents intuitive directions for field research by introducing a ground-breaking IoT-ML-driven intelligent farm management platform. This study’s main goal is to address agricultural difficulties by providing a thorough, integrated solution. This work makes a variety of important contributions. By utilizing cutting-edge technology like IoT and Machine Learning (ML), it first improves conventional farm management procedures. Farmers now have the capacity to remotely monitor and regulate irrigation management thanks to sensor-based real-time data. Second, based on data gathered from agricultural fields, our machine learning model offers improved water control management and fertilizer use recommendations, maximizing production while minimizing resource usage. The suggested solution also uses blockchain technology to create a safe, decentralized network that guarantees data integrity and defends against threats. We also introduce energy harvesting technology to address the issue of continuous energy supply for IoT devices, which lessens the load on farmers by removing the requirement for additional batteries. We achieved 89.5% accuracy in our proposed machine learning model. The suggested model would provide a variety of services to farmers, including pesticide recommendations and water motor control via mobile applications and a cloud database.

Keywords—Smart agriculture; machine learning; internet of things; energy harvesting; blockchain technology

I. INTRODUCTION

Without a doubt, agriculture is the most important source of livelihood in Bangladesh. As the world’s population expands, increased agricultural output is essential. The amount of fresh water and appropriate fertilizer used in irrigation must be raised in order to maintain enhanced farm productivity. Unintentional water waste happens when water consumption is not planned. Choosing the right fertilizer for a particular farmland is likewise a difficult challenge for our farmers. This demonstrates the urgent need for alternatives to reduce water waste and appropriate fertilizer choices without placing farmers under stress. In the Electronic age, agriculture is rapidly becoming a data-intensive sector, with farmers collecting and analyzing massive amounts of data from various sources (e.g., sensors, farming machinery, etc.) to obtain vital information and become more efficient in production. Technology nowadays has advanced a lot. With the help of Machine Learning and IoT devices, a drastic change can be made possible in the agricultural industry [1].

With the release of open-source Arduino devices and the availability of different sensors, it is now possible to build devices that can monitor soil moisture content and irrigate fields or landscapes as needed. Machine learning algorithms are used to assess various agricultural data and may readily forecast which decisions should be made to improve farmland productivity. In comparison to their previous farming ways, the farmer may easily combine ML and IoT into their farming and create an automated system that is more time effective and less risky. Here, Fig. 1 depicts the difference between the traditional system with the ML-based agricultural framework.

Wireless Sensor Networks (WSNs) technologies have a major challenge with limited energy. Many research in WSNs has also been focused on reliable energy supply to extend the survival time of limited power sources in a network [2]. Energy harvesting techniques are used to overcome the energy-scarcity problem of WSNs. Energy harvesting is a process in which energy is obtained from the environment as renewable energy sources like solar radiation, Radio Frequency (RF), wind, geothermal, electromagnetic (EM) waves, hydro, etc., and is stored effectively for driving various applications systems which may include wireless sensor networks (WSNs) [3] [4]. Therefore, it can be used to operate the devices of the embedded system for a reliable energy supply.

Security is one of the most critical aspects of IoT, as it deals with the protection of data and devices from unauthorized access, use, disclosure, disruption, modification, or destruction [5]. Encryption mechanisms are mostly used to ensure that data is securely transmitted. But, regularly used encryption algorithms such as DES, AES, and RSA will be heavy for small-scale embedded systems. Therefore, blockchain technology can be used as a lightweight calculation technique to reliably operate and secure an IoT system [6] [7].

This paper presents the latest IoT-ML-driven intelligent agricultural management and provides a substantial new research direction. The central insight of this work is to offer possible solutions to farming hazards while providing a combined framework. Some significant contributions of this paper are outlined as follows:

- **Smart Management**: Traditional agricultural management is strengthened with edge-cutting technologies (i.e., IoT and Machine Learning).
- **Distant Monitoring and Controlling**: Farmers can monitor and control irrigation management from a distance in terms of sensor-based real-time data.
- **Intelligent Decision Making**: Our machine learning model provides substantial water control management
and fertilizer utilization direction for a minimum resource with a maximum throughput based on the data collected from the farming field.

- **Blockchain Based Security System:** Our proposed solution uses blockchain technology to create a secure and decentralized network that can ensure data integrity and protect data from denial of service (DDoS) attacks, and man-in-the-middle (MITM) attacks.

- **Energy Harvesting:** To ensure a continuous energy supply for the IoT system, we have introduced energy harvesting technology which reduces the hassle for the farmers of using extra batteries.

The rest of the paper is organized as follows: Section II provides an overview of existing works related to our proposed framework. Section III provides a detailed description of the hardware that have used in this research. Section IV discusses the proposed IoT-ML-based smart agricultural framework. Section V discusses the blockchain-based security of our system. Section VI presents the real-life implementation of our project. Next, Section VII illustrates and analyzes the experimental result from our machine learning models. Section VIII future research directions in this field of research. Finally, Section IX gives a brief conclusion.

## II. RELATED RESEARCH

Several ML-IoT-based researches and project works on agriculture systems have been carried out till today.

In [9], Samuel et al. analyzed numerous techniques for crop selection, crop sowing, weed detection, and system monitoring. They have recommended different image processing methods for weed and leaf detection and evaluated the benefits and drawbacks of each. Drone implementation has been considered for real-time monitoring and seed planting. However, no actual implementation is shown in this research; they have only reviewed several smart agricultural strategies.

In [10], they analyze soil moisture levels and apply auto irrigation to the crops. In order to eliminate the need for human involvement, this system also senses temperature, humidity, and the presence of impediments in the targeted region. These data are accessible to the user via mobile from the cloud. By giving the motor driver the command YES/NO based on this data, the user can control the operation of the motor.

In [11], they use a cloud-based architecture and the Internet of Things to examine a smart irrigation system. This system is designed to measure soil moisture and humidity and then process this data in the cloud using a variety of machine-learning techniques. Farmers receive accurate information regarding water content regulations. If farmers apply smart irrigation, they can reduce their water usage.

In [12], they use IoT and machine learning to predict late blight disease in potatoes and tomatoes prior to the first occurrence. This will send farmers a warning message on the precise time to apply the protective pesticides.

In [13], for yield prediction, they present a hybrid ML model using IoT. They use a two-tier ML approach named aKNCN and ELM-mBOA.In the first tire, they estimate soil quality and in the second tier, they predict the crop yield.

In agriculture, supply, and demand have always been crucial issues for sustainable production management. To address and provide a possible solution to this problem, M. Lee et al. proposed an IoT-based controlled agricultural production management [14]. The authors developed a decision support system to predict specific criteria based on IoT-enabled sensor data.

A cloud-based real-time data analysis model is proposed in [15] instead of dew-point humidity. In this regard, they designed a CMM index measurement model to evaluate the crops’ comfort level of relative humidity levels.

To increase the crop production rate, real-time data analysis based on an artificial model is proposed by Y. Zhou et al. in [16]. The current innovation trend expects farmers to use IoT and technology to identify the organization of those difficulties they face, such as water management deficiencies in agriculture and productivity concerns. This research has attempted to build dazzling agricultural cultivation patterns utilizing IoT technologies. IoT has significantly improved agriculture by analyzing various agricultural difficulties and issues.

An IOT-based intelligent technology for agriculture that can sense soils and environmental factors was proposed by Abhijith H. V. et al. in [17]. In order to identify the urgent needs for optimal crop growth, they applied data mining techniques to the sensed data.

Abraham et al. [18] create a proof-of-concept farm surveillance system that employs IoT and deep learning to identify
farm encroachment.

The purpose of the study Wongpatikaseree et al. [19], is to propose a traceability system, summarizing and presenting observed data from the smart farm.

For smart farming, Deden Ardiansyah et al. [20] suggested a WSN Server that can handle and optimize agricultural data. They instantly store the data in the database, which is afterward represented as a website and accessible through the Internet network.

Automation in agriculture is a basic need for remote control-based agricultural management to ensure sustainable development in this field of research. In this regard, L. Vijayaraga et al. proposed an IoT-based monitoring system using wireless communication networks in [21]. The power supply management used in this framework is entirely from the renewable energy source that provides a cost-effective model for sensor-based decision management in intelligent farming.

In [22], Yaw-Wen Kuo, et al. presented a Long-range IoT system where they developed four types of IoT units based on Long-range technology. They employed pH, ORP, and EC in Type A and water, air, and humidity sensors in Type B. In type C, the pump can be operated remotely, and in type D, a water flow or water meter-controlling system is provided.

They presented a cloud service-based architecture in [23] that includes a variety of services for farmers, including agri-food-related services, financing, fostering, warehouse management, etc. They have suggested interactive video conferencing, voice-based services, text messaging, web portal services, and more under the heading of cloud services.

A key component of practicing smart agriculture is precision agriculture. In this context, Patil et al. [24] suggested a system that measures soil moisture using sensors for temperature, humidity, and soil moisture. Additionally, they offered various methods for highlighting the issue of data loss.

In [25], Quasim et al. use blockchain techniques in smart healthcare systems to ensure the security of healthcare data. It provides the security, privacy, and efficiency of the data in transmission between wearable sensors and Internet of Things (IoT) devices.

In [26], Makhdoom et al. made a blockchain-based framework for privacy-preserving and secure data sharing in smart cities. The system secures data sharing by segmenting the blockchain network into different channels, where each channel consists of a limited number of authorized organizations and handles a particular type of data, such as financial information, health data, smart car data, or data related to smart energy. Additionally, smart contracts contain access control rules that regulate who has access to the data of users within a channel.

Many different types of intelligent agricultural systems were developed in the earlier work. Some of these current systems are tabulated in Table I.

III. System Hardware

To set up the IoT environment for a smart agricultural system, we have selected a variety of hardware components, including the ESP8266 Node MCU (Fig. 2) processing unit and several sensors, including capacitive soil moisture (Fig. 3), PH sensor (Fig. 4), MH-RD Rain Sensor (Fig. 5), and LDR Sensor (Fig. 6). Our IoT system is powered by DC-DC power converter (Fig. 9), solar energy harvesting components (Fig. 10), single-channel relay modules (Fig. 7), DC motors (Fig. 8), etc.

A. Node MCU ESP8266

![Node MCU module](image)

**Feature**
- Operating Voltage: 3.3V
- Input Voltage: 7-12V
- Digital I/O Pins (DIO): 16
- Clock Speed: 80 MHz
- Small size module

B. Capacitive Soil Moist Sensor v1.0

![Soil moisture sensor](image)

**Feature**
- Operating Voltage: DC 3.3-5.5V
- Output Voltage: DC 0-3.0V
- Digital I/O Pins (DIO): 16
- Analog output
- Supports 3-Pin Sensor interface

C. PH Sensor(SEN-00239)

**Feature**
- Supply voltage: 5V
- Current: 5-10 mA
- Consumption: \(\leq 0.5\) W
- Working temperature: 10-50°C
TABLE I. PREVIOUS RESEARCH WORKS IN TERMS OF OBJECTIVES, USED TOOLS, AND POSSIBLE RESEARCH GAPS

<table>
<thead>
<tr>
<th>Reference</th>
<th>Research Purpose</th>
<th>Used Technologies/Techniques</th>
<th>Focused Methods</th>
<th>Challenges/Research Gaps</th>
</tr>
</thead>
<tbody>
<tr>
<td>[11]</td>
<td>IoT-Cloud based automated irrigation</td>
<td>Raspberry Pi, central cloud storage, soil data set, machine learning techniques, and mobile applications</td>
<td>Focused to measure soil moisture and humidity and then process this data in the cloud using a variety of machine learning techniques</td>
<td>Farmers get information about water only, no other necessary information.</td>
</tr>
<tr>
<td>[12]</td>
<td>IoT-based agriculture monitoring system for predictive analysis</td>
<td>Air temperature sensor, air humidity sensor, and soil moisture sensor, Microcontroller Unit (NodeMCU), MQTT protocol, R-Pi 3 microcontroller, MYSQL</td>
<td>Focused to predict the late blight disease in potatoes and tomatoes before the first occurrence</td>
<td>Not fully automated, need human interference to apply the action</td>
</tr>
<tr>
<td>[14]</td>
<td>IoT-based agricultural production system</td>
<td>Dual CDMA protocol, pH sensor, water sensor, and temperature sensor</td>
<td>Focus on reliable agricultural production management</td>
<td>Absence of dynamic data analysis model</td>
</tr>
<tr>
<td>[15]</td>
<td>IoT-Cloud based agricultural monitoring system</td>
<td>Arduino UNO, temperature and humidity sensors, Arduino Ethernet shield and ThingSpeak cloud platform</td>
<td>Finding the index of thermal control functions to find the comfort levels of agricultural parameters</td>
<td>Sensor data processing time is slower in terms of CMM-MIST measurement algorithm.</td>
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<tr>
<td>[16]</td>
<td>Machine Learning based agricultural management</td>
<td>Threat Model (TM), Deep Crop Mapping Model (DCMM), Random Forest Regression Algorithm (RFRA)</td>
<td>An intelligent management to predict soil moisture content based on the ML architecture</td>
<td>The key challenge of this research is real-time data processing</td>
</tr>
<tr>
<td>[17]</td>
<td>Intelligent technology for IoT-based agriculture</td>
<td>PH sensor, temperature, rainfall, humidity sensor, Predictive classification algorithm, MatLab</td>
<td>Focused on the identification of urgent needs for optimal crop growth</td>
<td>Prediction of specific need isn’t gained properly</td>
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<td>[18]</td>
<td>Comprehensive farm monitoring system</td>
<td>Arduino Board, Node MCU, Sensors, mobile App, machine learning, deep learning</td>
<td>Centered on a surveillance system prototype and an app-based remote administration solution</td>
<td>Remotely monitoring but not fully automated controlling</td>
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<tr>
<td>[19]</td>
<td>IoT-Based Smart farming</td>
<td>Sensors, mobile technology, Wi-Fi, cloud computing</td>
<td>Human interaction, water wastage</td>
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<td>[20]</td>
<td>Water management based on IoT</td>
<td>Soil moisture, Wi-Fi segments</td>
<td>Low or excessive irrigation, and water waste</td>
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<td>[32]</td>
<td>IoT-based cost-effective agricultural management</td>
<td>Moisture and Water sensors, Node MCU, Solar panel and LCD display unit</td>
<td>Focused on the low-cost parameter while ensuring a sustainable energy efficient management</td>
<td>The key research gap of this work is that this model is applicable for small farming areas.</td>
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<tr>
<td>[21]</td>
<td>IoT platform has a long range for controlling pumps and monitoring agriculture</td>
<td>LP WAN, Base station, Ph sensor, Electrical Conductivity sensor, Water Temperature Sensor, GY39</td>
<td>Presented a complete IoT system including the design of a remote unit and server construction</td>
<td>It is required to conduct additional research on the pH sensor because the data that has been gathered is inaccurate and collected from other vendors.</td>
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<tr>
<td>[22]</td>
<td>Cloud service architecture for agriculture using IoT and Big Data</td>
<td>Different Sensor, Central Cloud Database</td>
<td>Proposed a cloud-based architecture for the agricultural industry that comprised a range of services, including farm monitoring, market-oriented service, agri-business monitoring, etc.</td>
<td>Not implemented just proposed an architectural model.</td>
</tr>
<tr>
<td>[23]</td>
<td>AI in agriculture applications</td>
<td>Arduino UNO, Soil moisture sensor, Wi-Fi module</td>
<td>Proposed a cloud-based architecture for the agricultural industry that comprised a range of services, including farm monitoring, market-oriented service, agri-business monitoring, etc.</td>
<td>Not implemented just proposed an architectural model.</td>
</tr>
</tbody>
</table>

Proposed System

Smart agricultural system based on an IoT-ML Blockchain Integrated Framework

IoT devices, Mobile Application, Machine Learning, and Blockchain-based security system

Focused on intelligent decision making, Distant Controlling, Energy Harvesting, and Security based smart agricultural management system

Future target to ensure Low latency network and high bandwidth transmission, easy deployment of Networks elements and Edge computing technology.

D. RH-RD Rain Sensor

**Feature**

- Working voltage: 5V
- Output format: Digital switching output (0 and 1)
- With bolt holes for easy installation
- Uses a wide voltage LM393 comparator

E. LDR (Light Dependent Resistor)

**Feature**

- Able to detect variable light resistance (50-100 K Ohms)


Fig. 4. PH Sensor with module.

Fig. 5. MH RD rain sensor.
**Photo-resistor (photo-conductive cell)**
- Power Level: 200 W
- Diameter: 3-20 mm

**Single-Channel Relay Module**

**Features**
- Ground Voltage: 0 V
- VCC: Provide input to the relay coil
- Supply Voltage: 3.75 to 6 V
- Current: 2 mA
- Relay Maximum Current: 10 A

**DC Motor 6V**

**Features**
- Diameter of the motor: 23.5mm
- Height: 30mm
- Start voltage: 0.8V
- Rated voltage: 6V
- Non-charging current: 25mA
- Speed: 2980 RPM

**Adjustable DC-DC Power Converter (1.25V - 35V-3A)**

**Features**
- Input Voltage: 3.2V - 40VDC
- Output Voltage: 1.25V - 35VDC
- Max. Output Current: 3A
- Max. Efficiency: 92%
- Output Ripple: ≤ 100mV
- Switching Frequency: 65KHz
- Operating Temperature: -45°C to +85°C
- Dimensions: 43mm*21mm*14mm(l*w*h)

**MSP430 Solar Energy Harvesting Tool**

**Features**
- Battery-less operation
- Functions in dim ambient light and 400+ transmissions
- Adaptable to any RF network or sensor input
- Inputs available for external harvesters (thermal, piezo, 2nd solar panel, etc.)
- USB debugging and programming interface with application backchannel to PC
- 18 available analog and communications input/output pins
- Highly integrated, ultra-low-power MSP430 MCU with 16-MHz performance
IV. PROPOSED FRAMEWORK

A. Circuit Diagram and Connections

We used Fritzing\(^9\) an open-source hardware online application to make a schematic pin (Fig. 11) diagram of our smart agriculture system.

We used ESP8266 NodeMCU V3, with an integrated WIFI module as our processing hardware component. The system connects an analog capacitive soil moisture sensor and an analog pH sensor using multiplexing to the A0 analog input of NodeMCU, a photo-resistor known as Light Dependent Resistors (LDR) sensor to a D3 digital input, and a raindrop sensor with rain board and control module to D0 input Pin. In addition, the device is connected with a D3 output pin to a DC 5V micro submersible mini water pump with the relay. We used the NodeMCU’s 5V VU pin to power the Motor and Relay. However the LDR Sensor and Rain Drop Sensor, only need a 3.3V supply, the Capacitive Soil Moisture Sensor and pH Sensor need 5V. The GND pin serves as the common ground for every sensor. A solar panel system that is coupled to a 9-volt battery backup powers the system.

B. Working Principle

We have divided our proposed framework into different subparts and each part’s working procedure is given below. The overall working procedure is depicted in Fig. 12.

1) Collecting Data From Sensor: We used four different types of sensors, including capacitive soil moisture sensors, pH sensors, MH-RD rain sensors, and LDR sensors, to execute smart IoT agriculture. We can estimate how much water is in the soil with the aid of a soil moisture sensor. A pH sensor, which ranges from 0 to 14, allows us to determine the water’s acidity or alkalinity. Water turns acidic if the value falls below 7, else it is alkanoic. Consider levels 5.5 to 7 to be ideal for growing crops. We can choose the best fertilizer for the soil with the aid of a pH sensor.


   \(^7\)https://techshopbd.com/detail/248/DC_Motor_6V_techshop – bangladesh

   \(^8\)https://techshopbd.com/detail/2067/Adjustable – DC – DC – Power Converter


   \(^10\)Fritzing – circuitdesign, fhttps://fritzing.org/
analog and digital values are both included. The raindrop sensor aids in our selection of how to operate the motor. The LDR sensor, which is used to detect the presence of light, has also been employed.

Now, each of these sensors is linked to a node MCU board in our project, and data from the sensors is uploaded to the node MCU board and shown on the serial monitor of the Arduino UNO editor and

2) Sending Data to Cloud Server: The node MCU board receives all sensor data, which we would love to save in the cloud in order to control for remote distance. We have utilized Firebase as a cloud server. We linked the Firebase authentication and real-time database URL that we built for our project with the Arduino UNO script in order to integrate Firebase with Node MCU.

All data is sent as a parent-child combination to Firebase. All of the sensor data is sent from the node MCU as a child of Smart Irrigation, which we have constructed as Parent. The Firebase stores the child value as a key-value pair.

3) Data Collection: The Firebase’s database contains all of the sensor data, which is compiled as a CSV file. Four columns make up our dataset: pH, LDR, Rain, and Moisture. Nearly 820 data are in our dataset. Only integer values are accepted in the Moisture column here, whereas string values are accepted in the Rain and LDR columns.

4) Data Pre-processing: Data can be often inconsistent. Missing values or values out of range is typical. So, the dataset needs some pre-processing before it can be used to train any model [27]. For this reason, we have considered three such cases.

- **Missing Value Handling:** While exploring our dataset, we observed humidity, raining status, daylight status, and pH level fields with missing values. Therefore, we have filled them with average values of the respective field.

- **One Hot Decoder:** Label encoding is simply the process of converting each value in a column to a number. By using label encoding, we have converted the categorical text data into model-understandable numerical data. We had to use the label encoding to get our dataset ready for our model.

- **Abnormal Data Handling:** Some data contain abnormal values. For example, the range of the temperature in our dataset falls between -20 to 30 degrees Celsius. However, we have even found some data above and below this range. Data tuples with these abnormalities have been dropped from the dataset.

- **Normalization:** Normalization is used to increase the accuracy of models [28]. It is simply the process of having all the data on the same scale. We have used...
temperature, pressure, relative humidity, and pressure as features to train our model. The features used to teach a machine-learning algorithm have different ranges of values. This can badly affect the machine’s learning ability. To solve this problem, we have standardized the feature values so that all the features stand equal in their representation. By normalization, all the feature values are mapped in the range between 0 and 1.

5) Machine Learning Model: The developed architecture is a Neural Network (NN) based because of its great accuracy. The fundamental advantage of NN over classical machine learning models is that it recognizes significant traits automatically and without human intervention. It’s a feed-forward NN with parameters using the back-propagation algorithm and stochastic gradient descent. Distinct processing layers serve different purposes. The output of the feature map is produced by conventional layers, which conduct linear convolution between a series of input tensors and filters. The nonlinear transformation is performed using the ReLU, which is the most widely employed activation function. The activation function for the fully connected layer to the end must be careful on the tasks. Batch normalization and an activation layer are performed after each convolution.

\[ \text{ReLU} = \max(0, X) \]  \hspace{1cm} (1)
\[ d(x) = \text{activation}(w^x + b) \]  \hspace{1cm} (2)
\[ \text{Dropout}(x, p) = (x : \text{prob.}, p) (x : \text{prob.}, 1 - p) \]  \hspace{1cm} (3)
\[ S(x) = \frac{1}{1 + e^{-x}} \]  \hspace{1cm} (4)

6) Mobile Application Development: A smart remote control application can ease our maximum task. We have used the MIT app inventor to make the mobile application that will be connected to our system and by using this app we can do the following task

- **Fertilizer Suggestion:** By analyzing the pH value, the app may suggest which fertilizer is best for a given soil. The app will recommend some alkanoic fertilizer if the pH value rises to help reduce the rising pH value and vice versa. Algorithm 1 depicts how the fertilizer is suggested in our system.
- **Visualization of Predicted Results:** In order to predict whether the motor would turn off or not based on the moisture, LDR, and raindrop sensor values, we construct a neural network model and link it with our mobile app. Algorithm 2 depicts how the remote controlling is done to control the motor in our system.
- **Remote Motor Controlling:** The farmer can use the app to control the motor from any distant or remote location based on the prediction outcome. When the farmer presses the off button, it sends a value of 0 to the firebase, which then passes this signal on to the node MCU via the wifi module and sets the pin value to the LOW, so turning off the motor.

Algorithm 1: Decision Making for Fertilizer Suggestion

1. Initialize the pH sensor
2. Read data from pH sensor
3. if \( pH \geq 6.5 \) && \( pH \leq 7.5 \) then
   - Soil is balanced.
   - No fertilizer is recommended.
4. else if \( pH < 6.5 \) then
   - Soil is acidic.
   - Store the pH amount.
   - Find the level_id corresponding pH amount.
   - Search through the fertilizer data (in JSON format)
   - if \( level\_id == \text{keyof}\_\text{JSONdata} \) then
     - Send the fertilizer name back to the user.
   else
     - The result doesn’t match our dataset.
     - Soil is alkanoic
     - Store the pH amount.
     - Find the level_id corresponding pH amount.
     - Search through the fertilizer data (in JSON format)
     - if \( level\_id == \text{keyof}\_\text{JSONdata} \) then
       - Send the fertilizer name back to the user
     else
       - The result doesn’t match our dataset

Algorithm 2: Decision Making for Pump on/off

1. Initialize the Moisture, Rain, and LDR sensors.
2. Read data from each sensor.
3. Send the data to the server using an HTTP POST request.
4. Apply machine learning to the collected data.
5. Retrieve the predicted result (PUMP ON/OFF).
6. Send Predicted results to the mobile phone.
7. Wait for user input from the mobile phone.
8. if \( \text{user\_action} = \text{true} \) then
   - Send a signal to the node MCU board.
   - Perform action according to signal.
else
   - Wait for 300 seconds
   - Take an automated action according to the predicted result (PUMP ON/OFF)

V. BLOCKCHAIN-BASED SMART-AGRI

Blockchain was described as a data structure using asymmetric encryption algorithms and hash functions to ensure that data tampering and forgery are impossible. Every smart agricultural system needs to be taken under the shelter of a security system to avoid getting an external attack. Our smart agricultural system is public so any intruder can make attacks such as DoS attacks to crash the system, and spoofing to alter the control. In the IoT environment where high computational encryption, decryption, and high-level security are not possible.
Therefore, we are implanting blockchain technologies into our smart agricultural system through which we are capable to maintain high throughput, low latency, low communication cost, and tamper-proof and traceability. Blockchain refers to a distributed ledger system where data or transactions are stored in blocks that are connected to each other through making hash which can not only serve as unique IDs but also prove the integrity of the blocks. The hash of the previous block is used to make a hash for the next block along with its data. If any intruder wants to tamper or alter the block data, all the consecutive block hash will be changed. Therefore, any intruder attempts to alter the data or spoof the blockchain will not be possible.

In this system, we consider nodeMCU, Firebase cloud, and mobile app as nodes. In order to avoid altering data in the network, we are using blockchain technology. When any node wants to send data to other nodes, it encapsulates the data into a block along with its hash values (SHA256) and nodeID then adds it to the blockchain. All these nodes will contain the blockchain locally. After adding the block, the node sends it to the cloud through the network. We are not using any PoW, PoS, or accountant selection algorithms which is not possible because of a very small amount of nodes and our nodeMCU has very limited capabilities to run these algorithms (Algorithm 3). When the block is sent to the cloud, I validate the block by checking all the hashes of the previous block along with the nodeID. If it gets any error, the node will consider the block as from an attacker and reject the block from adding to the chain. Fig. 13 depicts the blockchain in our system.

Algorithm 3: Blockchain-based data security in smart-agri

1. Initialization
2. Read data
3. \( Block\_Hash \leftarrow SHA256 \) (Previous_hash, data, nodeID, timestamp, nonce)
4. Make a block (Block_hash, data, nonce, nodeID, timestamp)
5. Add the block to the chain locally.
6. Send the block to the cloud.
7. if Block_hash = Previous_Hash then
   Accept the block, then add it to the chain.
   Send acknowledgment.

In this system, we consider nodeMCU, Firebase cloud, and mobile app as nodes. In order to avoid altering data in the network, we are using blockchain technology. When any node wants to send data to other nodes, it encapsulates the data into a block along with its hash values (SHA256) and nodeID then adds it to the blockchain. All these nodes will contain the blockchain locally. After adding the block, the node sends it to the cloud through the network. We are not using any PoW, PoS, or accountant selection algorithms which is not possible because of a very small amount of nodes and our nodeMCU has very limited capabilities to run these algorithms (Algorithm 3). When the block is sent to the cloud, I validate the block by checking all the hashes of the previous block along with the nodeID. If it gets any error, the node will consider the block as from an attacker and reject the block from adding to the chain. Fig. 13 depicts the blockchain in our system.

VI. IMPLEMENTATION

A. Hardware Implementation

Our Smart-agri system hardware demo showed in Fig. 14.

B. Mobile App Implementation

The interface of the mobile app that we developed in the MIT App Inventor resembles (Fig. 15). The “Fertilizer Suggestion” button can be found in the app. Depending on the pH sensor measurement, this button tells us whether the soil is alkanoic or acidic when we click it. We used the decision tree algorithm to determine the ideal fertilizer for a given soil based on its quality.

The farmer can view the data from the moisture, raindrop, and daylight sensors in our app. Our program uses a neural network model that we built and implemented to forecast whether the motor would be on or off. We can operate our motor using the two buttons in our app labeled “ON” and “OFF”. The motor status in the firebase changes to 1 when we press the “ON” button, and the firebase sends a signal to the nodeMCU board, which then turns the motor on automatically. And that is how we can use our mobile app to implement remote control.
TABLE II. SYSTEM EVALUATION WITH THE EXISTING SYSTEMS

<table>
<thead>
<tr>
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TABLE III. SUMMARY OF PROPOSED NEURAL NETWORK PERFORMANCE PARAMETERS

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<tr>
<th>Epochs</th>
<th>Processing Time /msec</th>
<th>Binary Cross Entropy Validation Loss</th>
<th>Gradient Descent Neural Network Validation Accuracy</th>
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</table>

VII. EXPERIMENTAL RESULT ANALYSIS

A. System Evaluation

In Table [11] we have shown the difference between our system and the existing systems. The criterion based on which we have shown the differences are remote monitor control, data visualization, customized mobile application, machine learning integrated framework, creating own database, central cloud database, pH-based fertilizer suggestions, Machine learning model development, and full automation. The references from [11] to [24] there is no such system that has implemented all the criteria in their system. But we have successfully implemented all the criteria in our systems.

B. Machine Learning Model Evaluation

Our deep learning neural network is implemented with the help of our own dataset. We split the dataset into 80-20 ratios for training and validation purposes. The heat map of the features columns is illustrated in Fig. [16]. We fit our gradient descent neural network within the data set. The input layer of the neural network receives 3 input lines from the features column, namely, rain status, moisture level, and daylight status respectively. Then we add one dense layer with 16 neurons and the activation function as ReLu. The next layer is a dropout layer with a 20% drop rate. Next, we add another dense layer with 8 neurons and apply the activation function as ReLu. Then we add another dropout layer with a 20% drop rate. Finally, for the output layer, we add another dense layer with a single neuron with is satiable for binary classification (i.e. motor on-off decision) with activation function as Sigmoid.

The experimental result of our model is represented in Table [11] while showing the validation loss rate as binary cross entropy and validation accuracy in different epochs. Our model successfully outcomes a stable level of accuracy for the different epochs. We got almost 89.5% accuracy in our experimental set-up.

VIII. FUTURE RESEARCH DIRECTION

For future smart irrigation management, several issues must be addressed as follows:

- **Low-latency in Real-Time Application:** The monitoring and controlling mobile application must be able to transmit real-time data to the farmers or its entity while ensuring a low latency network.
- **High Bandwidth:** To facilitate a buffer-less transmission, we need to ensure maximum bandwidth level to the transmission process.
- **Connectivity:** To meet the high communication demands of future IoT-ML integrated irrigation systems, reliable synchronization between linked autos would be required.
- **Deployment of Network Elements:** When a network has a high enough number of nodes, its overall performance increases. Because network equipment deployment is costly, it is vital to have the required number of network components up and running as quickly as feasible.
- **Augmented Reality:** AR is a multimedia application that mixes real-world scenes into virtual scenes and superimposes virtual scenes over real-world scenes to supplement traditional real-image information. This technology has the potential to help farmers become more aware of the app’s functionality.
- **Edge Computing:** This networking approach is built on a network control layer that is logically centralized.
IX. Conclusion

We have put forth an integrated solution that enables farmers to solve the challenges that limit their production and profitability by utilizing cutting-edge technologies like machine learning (ML), the Internet of Things (IoT), and blockchain. Through user-friendly mobile applications and a secure cloud database, this model, implemented with a generated database, offers helpful insights and recommendations to farmers, including pesticide usage and water motor control. Real-time monitoring and data collecting is made possible by the integration of IoT devices, enabling accurate decision-making and assuring optimal resource allocation. Incorporating blockchain technology also improves data traceability, transparency, and integrity, fostering trust and accountability throughout the agricultural ecosystem. Farmers may gain from higher productivity, decreased expenses, and enhanced general agricultural management by implementing our Smart-Agri framework. They can obtain up-to-date, accurate information, make decisions that will increase productivity and reduce waste, and improve their agricultural techniques. We have applied the gradient descent neural network model for water control management and achieved up to 89.5% accuracy. In addition, the suggested structure creates chances for cooperation, information exchange, and market access, all of which help the agricultural industry thrive and flourish sustainably.

In the future, the framework’s scalability, and its interoperability can all be explored through more research and development in this area, along with potential problems like connectivity problems and data privacy issues. We can build an ecosystem that really revolutionizes the agriculture sector by continually improving and building upon these technological achievements, making it more intelligent, efficient, and robust in the face of changing global issues.

Acknowledgment

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References


New Approach based on Association Rules for Building and Optimizing OLAP Cubes on Graphs

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Abstract—The expansion of data has prompted the creation of various NoSQL (Not only SQL) databases, including graph-oriented databases, which provide an understandable abstraction for modeling complex domains and managing highly connected data. However, to add graph data to existing decision support systems, new data warehouse systems that consider the special characteristics of graphs need to be developed. This work proposes a novel method for creating a data warehouse under a graph database and demonstrates how OLAP (Online Analytical Processing) structures created for reporting can be handled by graph databases. Additionally, the paper suggests using aggregation algorithms based association rules techniques to improve the efficiency of reporting and data analysis within a graph-based data warehouse. Finally, we provide a Cypher language implementation of the suggested approach to evaluate and validate our approach.

Keywords—NoSQL; graph-oriented databases; data warehouse; OLAP; aggregation algorithms; association rules; cypher language

I. INTRODUCTION

Modern databases have been considerably altered by the expansion of data. NoSQL databases have expanded as a result of these new demands and now come in a wide range of models [1], including key-value, document, column, and graph.

In particular, graph-oriented databases are one of the most-known various of NoSQL systems; they have attracted a lot of attention and popularity. Graph-oriented databases are a fundamental form that offer an understandable abstraction to model numerous complicated domains, manage highly connected data, and run sophisticated queries over them [2] [3].

Currently, many businesses and entrepreneurs are interested in developing business intelligence systems on graph databases to take advantage of its benefits. Enterprises are also interested in expanding their OLAP analysis to include the new forms of data because OLAP technology is currently widely used[4]. However, despite the growing interest in graph-based data warehouses and their potential benefits, there is a lack of research addressing the challenge of creating an optimized OLAP model under graphs, specifically focusing on selecting the most relevant OLAP aggregations to effectively meet the diverse user’s needs. This gap calls for further investigation and exploration to bridge the divide between the advantages of graph-based data warehousing and the need for efficient and user-centric aggregation selection techniques. The aim of this study is to create new data warehouse under graph database systems that consider the special characteristics of graphs, this work also aims to optimize the efficiency of reporting and data analysis within this graph-based data warehouse by employing user-centric aggregation techniques that select the most relevant OLAP aggregations to meet the diverse needs of users effectively. To address this issue, it is possible to use aggregation algorithms that automatically select the most relevant aggregations for the cube. These aggregation algorithms are often based on machine learning and data mining techniques to identify the best possible aggregations based on raw data. The use of these algorithms can help reduce cube build time and improve the accuracy of analysis results. In this context, we demonstrate how OLAP structures created for reporting can be handled by graph databases, additionally, We provide new approach to optimize OLAP cube using the association rules algorithm.

The remainder of this paper is organized as follows. In Section II, we present some works of the literature reviews related on graph data warehouse. In Section III, we give a background overview of our approach. In Section IV, we describe the implementation of our approach as well as a case study to assess it. Section V concludes this paper and suggest future research directions.

II. RELATED WORK

Many approaches were proposed in the literature as a result of the growing interest in combining graph databases and business intelligence technology in recent years. In [5], The authors have proposed a new concept Graph Cube, a new data warehouse model that supports OLAP queries in large multidimensional networks, in the Graph Cube the dimensions are based on the attributes of the nodes, while the computed measures represent the aggregations of these node attributes. The Graph Cube approach have some limitations in analyzing dynamic or evolving graphs, where the structure of the graph changes over time. Moreover, the accuracy and reliability of the analysis results may be affected by data quality issues such as missing or erroneous data. In [6], The author introduce a GraphAware Framework for Neo4j, which enables the precalculation and storage of node information in graphs. For instance, the framework can compute the number of friends in a social network and store the result for efficient querying. The GraphAware Framework also supports the analysis of node degrees in the graph, which can be useful for identifying important or influential nodes. The GraphAware Framework is focused on precalculating and storing node information, which may be limited in scope and may not capture the
full complexity of the graph data. For example, some graph analytics tasks may require more sophisticated computations that go beyond simple node attributes, such as graph centrality measures or community detection. In [7], The authors proposed to use the graph structure as a basis for OLAP queries; this approach relies on using the performance and efficiency of the Neo4j graph database to store and query OLAP queries. In this model, dimensions and measures are transformed into nodes. The connection between dimensions and measures is done through arcs of the graph. For hierarchical dimensions are also stored in nodes and linked together by hierarchical relationships. The approach is limited to using only the snowflake model, which may not be suitable for all types of data. This may restrict the flexibility and adaptability of the approach. In [8], The author's goal is to compare the execution times of the identical OLAP queries in the relational and graph databases by using a MusicBrainz database data warehouse in a PostgreSQL relational environment and then implementing the same decision model in Neo4j. The authors only tested their approach on a single dataset (the MusicBrainz database), which may limit the generalizability of their results. Additionally, they only considered a specific decision model and did not explore the potential impact of different OLAP queries or decision models on the performance of the two database types. Furthermore, while the results of the study showed that the graph database outperformed the relational database in terms of execution time, the authors did not provide a detailed analysis of the factors that contributed to these results. This lack of analysis makes it difficult to fully understand the advantages and disadvantages of each database type for OLAP queries. In [9], The authors define a set of transformation rules that can transform conceptual models into graph-oriented models. They have defined four transformation rules, namely: fact transformation, Name and Identifier of Dimension Transformation, Hierarchies Transformation, Transformation of the relation between Fact and Dimension. Still within the framework of Datawarehouse modelling in a NoSQL graph database, the authors propose a conceptual mapping between a multidimensional schema and a graph-oriented NoSQL model. The authors chose to concentrate on proving the viability of their approach rather than providing any experimental campaign to validate it. In [10], the authors propose to integrate NoSQL Graph-oriented Data into Data Warehouses as a solution to tackle Big Data challenges. The paper introduces a new approach called “Big-Parallel-ETL” that adapts the classical ETL (Extract-Transform-Load) process with Big Data technologies, leveraging the efficiency of the MapReduce concept for parallel processing. However, this work does not address OLAP aggregations. The authors in [11], suggest a set of guidelines for creating a graph data model from a multidimensional data model (MDM2G). Then the authors compare the performance of the two star and snowflake designs in the graphs and relational databases, in terms of dimensionality and size. After doing this comparison, the authors concluded that a graph implementation of a data warehouse with multiple tables is more effective than a relational implementation, and that a star model performs similarly to a snowflake model in graph databases. The study does not provide a detailed description of the conversion rules and does not present any experimental results or validation of the proposed approach. In [12], the authors proposed employing two alternative logical models, equivalent to the ROLAP (Relational Online Analytical Processing) and MOLAP (Multidimensional Online Analytical Processing) models, to create OLAP engines within a graph database. They specify a set of guidelines for mapping these models from the multidimensional model. Additionally, they suggest an aggregation technique for constructing the lattice of cuboids from a data warehouse. However, the choice of aggregations is random and imprecise, which makes the model unoptimized and burdens the graph with several unnecessary nodes. The authors in [13], suggest an approach founded on a multi-version evolutionary schema model. Data instances corresponding to various schema versions are stored in a graph data warehouse. A meta-model is utilized to manage these warehouse schema versions. Additionally, they introduce evolution functions at the schema level. To validate their approach, they implement a software prototype and conduct a case study that demonstrates queries on schema versions, cross-queries, and the runtime performance of their approach. However, the impact of the multi-version approach on OLAP aggregations is not addressed in this study.

The Table I provides a comprehensive summary of the literature review, highlighting the key findings and identified gaps in the research.

All of the cited works provide an important context for the implementation of decision systems using graph databases. However, most of these works focus on converting relational data warehouses into graph databases or applying traditional business intelligence methods, which can limit the advantages of using graph databases. Our proposed approach is different and relies on the properties of graphs to implement data warehouses. It highlights the importance of studying a model that optimizes the choice of OLAP aggregations to enhance the graph cube’s performance. By selecting the optimal set of aggregations for a graph cube, OLAP query performance can be significantly improved, resulting in faster query response times and more efficient use of system resources. This can enable users to analyze larger volumes of data more quickly and accurately, leading to more informed decision-making. Moreover, reducing the computational resources required to execute OLAP queries can result in cost savings for organizations that need to process large amounts of data.
TABLE I. LITERATURE REVIEW

<table>
<thead>
<tr>
<th>Year</th>
<th>Authors</th>
<th>Findings</th>
<th>Gaps</th>
</tr>
</thead>
<tbody>
<tr>
<td>2011</td>
<td>Zhao et al.</td>
<td>Graph Cube: A new paradigm for Data Warehouse (DW) that supports OLAP</td>
<td>Limitations in analyzing dynamic or evolving graphs and potential</td>
</tr>
<tr>
<td></td>
<td></td>
<td>queries in large multidimensional networks, with dimensions based on</td>
<td>data quality issues.</td>
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<tr>
<td></td>
<td></td>
<td>node attributes and computed measures representing aggregations.</td>
<td></td>
</tr>
<tr>
<td>2013</td>
<td>Bachman</td>
<td>GraphAware Framework for Neo4j enables pre-calculation and storage of</td>
<td>Limited in handling more sophisticated computations beyond simple</td>
</tr>
<tr>
<td></td>
<td></td>
<td>node information for efficient querying, but may not capture the full</td>
<td>node attributes.</td>
</tr>
<tr>
<td></td>
<td></td>
<td>complexity of graph data.</td>
<td></td>
</tr>
<tr>
<td>2014</td>
<td>Castelltort et al.</td>
<td>Proposes using graph structure as a basis for OLAP queries, but limited</td>
<td>May not be suitable for all types of data.</td>
</tr>
<tr>
<td></td>
<td></td>
<td>to the snowflake model, reducing flexibility.</td>
<td></td>
</tr>
<tr>
<td>2019</td>
<td>Vaisman et al.</td>
<td>Comparing the execution timings of identical OLAP queries in relational</td>
<td>Lack of detailed analysis of factors contributing to the performance</td>
</tr>
<tr>
<td></td>
<td></td>
<td>and graph databases reveals that the graph database provides superior</td>
<td>difference.</td>
</tr>
<tr>
<td></td>
<td></td>
<td>performance.</td>
<td></td>
</tr>
<tr>
<td>2020</td>
<td>Sellami et al.</td>
<td>Define transformation rules to convert conceptual models into graph-</td>
<td>No experimental campaign to validate the approach.</td>
</tr>
<tr>
<td></td>
<td></td>
<td>oriented models.</td>
<td></td>
</tr>
<tr>
<td>2021</td>
<td>Soussi</td>
<td>Propose parallel loading based integration of NoSQL graph-oriented data</td>
<td>Doesn’t address OLAP aggregations.</td>
</tr>
<tr>
<td></td>
<td></td>
<td>into data warehouses.</td>
<td></td>
</tr>
<tr>
<td>2022</td>
<td>Akid et al.</td>
<td>creating graph data models based on multidimensional data and comparing</td>
<td>Lack of detailed conversion rules and experimental validation.</td>
</tr>
<tr>
<td></td>
<td></td>
<td>star and snowflake designs in graphs and relational databases.</td>
<td></td>
</tr>
<tr>
<td>2022</td>
<td>Khalil et al.</td>
<td>Propose alternative logical models for OLAP engines within a graph</td>
<td>Unoptimized aggregations and unnecessary nodes in the graph.</td>
</tr>
<tr>
<td></td>
<td></td>
<td>database and an aggregation technique for constructing the lattice of</td>
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<td></td>
<td></td>
<td>cuboids from a data warehouse.</td>
<td></td>
</tr>
<tr>
<td>2023</td>
<td>Benhissen et al.</td>
<td>Propose an approach based on a multi-version evolutionary schema model</td>
<td>Doesn’t address the impact of multi-version approach on OLAP</td>
</tr>
<tr>
<td></td>
<td></td>
<td>in a graph data warehouse.</td>
<td>aggregations.</td>
</tr>
</tbody>
</table>

\[ G = (N, E, L^N, L^E, P^N, P^E) \]

where:
- \( N \) is a set of finite nodes,
- \( E \subseteq N \times N \) represent edges between the nodes.
- \( L^N \) describes the label of the nodes.
- \( L^E \) describes the edges’ label.
- \( P^N \) is a set of characteristics that identify node.
- \( P^E \) is a set of properties that describe an edge.

A dimension consists of a set of attributes representing different levels of granularity on the data to be analyzed (measures).

A dimension, denoted \( D_i \) \( \in D^S \), This is characterized by (\( N^D, A^D, H^D \)) where:
- \( N^D \) is the name of the dimension.
- \( A^D = A_1, \ldots, A_n \) is a set of dimension attributes.
- \( H^D = H_1, \ldots, H_n \) is a set of hierarchies, arranging the properties in accordance with the level of granularity that each one represents.

The Fig. 1 illustrates our multidimensional use case model:

![Fig. 1. The multi dimensional model.](image-url)
B. Our Approach

Our approach involves leveraging the advantages of graph databases by creating the OLAP cube in the graph and using user queries to identify frequently used dimensions in OLAP analyses. To optimize the aggregations to be created, we use the Apriori algorithm to extract the most frequently associated sets of dimensions in OLAP queries, and then apply a rule-based association algorithm to identify the most relevant aggregations. The resulting aggregations are created in the OLAP cube, leading to improved OLAP query performance.

The approach consists of four steps:

- Creation of the graph data warehouse.
- Extraction of the most frequently associated sets of dimensions in OLAP queries using the Apriori algorithm.
- Identification of the most relevant aggregations using a rule-based association algorithm.
- Creation of identified aggregations.

1) Graph Data Warehouse: When used with relational databases, a relational fact table is created for each fact in the multi dimensional conceptual model. Measures are columns in the fact table. Additionally, each dimension is transformed into a normalized dimension table with columns for each attribute (including parameters and weak attributes). Fact and dimension tables each have a unique row for storing each instance [16]. Similarly, we provide our rules, which define a graph DW, using the definitions of multi dimensional model and property graph ideas that were previously presented.

Dimension $D^S$ in our model is created in node format identified by $(L^N, P^N)$ where:

- $L^N$ represents the label of the node, a node can have zero to many labels.
- $P^N$ represents attributes of the dimension.

Hierarchies: In a graph data warehouse, a hierarchy can be represented using nodes and edges. Each level of the hierarchy can be represented as a node, with edges connecting nodes at different levels to indicate parent-child relationships.

Fact: A fact node in a graph data warehouse can be represented as a node with edges connecting it to dimension nodes. The fact node can also have properties that represent the measures, such as the actual values or aggregate functions applied to them [17]. A fact node, is specified by $(N^F, M^F)$ where:

- $N^F$: represent the fact name.
- $M^F$: It comprises a collection of measures as node attributes, with each measure linked to an aggregation function.

Relationship between fact and dimensions: The relationship between fact and its associated dimensions are represented as edges connecting nodes in the graph, the link is defined by $(L^E, N^F, N^D, P^E)$, where:

- $L^E$ is the label of relationship.
- $N^F$ is the fact node.
- $N^D$ is a node that represents the dimension linked to the fact.
- $P^E$ represent the properties of the relationship, the properties are key-value pairs that are used for storing data on relationships.

2) Algorithm 1: Calculate Frequent Itemsets: After creating our OLAP system, the second phase in our approach is to collect user queries from the OLAP system logs [18]. And after that, we determine common itemsets of predicates using the Apriori algorithm [19]. In the next phase we will use the generated itemsets as input to the second association rule algorithm to determine the most important aggregations to create. The algorithm starts by initializing an empty list to store frequent itemsets and another empty list to store previous frequent itemsets. Then, it loops on user queries predicates until there are no more frequent itemsets to explore. During each iteration of the loop, the algorithm generates candidates for new frequent itemsets by combining previous frequent itemsets. Then, it calculates the frequency of each candidate by scanning through all transactions. Candidates that have a frequency below a minimum support threshold are filtered out. Finally, the frequent itemsets are added to the list of frequent itemsets. The candidate generation and filtering functions are also defined in the algorithm.
Algorithm 1: Calculate frequent itemsets

```plaintext
// Initialization
frequent_itemsets ←
previous_frequent_itemsets ←
// Loop until there are no more frequent itemsets to explore
while (previous_frequent_itemsets is not empty) do
  // Generate candidates
  current_frequent_itemsets ←
generate_candidates(previous_frequent_itemsets)
  // Calculate frequency of candidates
  for each aggregation in OLAP cube do
    for each candidate in current_frequent_itemsets do
      if candidate is used in aggregation then
        candidate.frequency + 1
      end if
    end for
  // Filter candidates
  previous_frequent_itemsets ←
current_frequent_itemsets ←
filter_candidates(current_frequent_itemsets, min_sup)
  // Add frequent itemsets to the list
  frequent_itemsets ← frequent_itemsets U current_frequent_itemsets
end while
```

FUNCTION generate_candidates(itemsets) candidates
for each itemset1 in itemsets do
  for each itemset2 in itemsets do
    if itemset1 ≠ itemset2 and all elements of itemset1 except the last one are the same as the corresponding elements of itemset2 then
      candidate ← union of itemset1 and itemset2, keeping only the unique elements
      if candidate is not already in candidates then
        candidates ← candidates U candidate
      end if
    end if
  end for
end for
return candidates
End FUNCTION

FUNCTION filter_candidates (candidates, min_sup) frequent_candidates ←
for each candidate in candidates do
  if candidate.frequency ≥ min_sup then
    frequent_candidates ← frequent_candidates U candidate
  end if
end for
return frequent_candidates
End FUNCTION

Algorithm 2: Generate Association Rules

```plaintext
association_rules ←
for each frequent_itemset in frequent_itemsets do
  subsets ← generate_subsets(frequent_itemset)
  for each subset in subsets do
    antecedent ← subset
    consequent ← (frequent_itemset) \ (subset)
    association_rule ← {antecedent : consequent : confidence : 0}
      // Calculate confidence of the rule
      frequency_antecedent ← calculate_frequency(antecedent)
      frequency_consequent ← calculate_frequency(consequent)
      itemset_frequency_antecedent ←
      if confidence ≥ min_conf then
        association_rule.confidence ← confidence
        association_rules ← association_rules U association_rule
      end if
  end for
end for
```

FUNCTION generate_subsets(itemset) subsets ←
for i ← 1 to taille(itemset) do
  subset ←
  for j ← 1 to taille(itemset) do
    if j ≠ i then
      subset ← subset U itemset[j]
    end if
  end for
  subsets ← subsets U subset
end for
return subsets
End FUNCTION

FUNCTION calculate_frequency (itemset) frequency ← 0
for transaction in transactions do
  if transaction contains all elements of itemset then
    frequency ← frequency + 1
  end if
end for
return frequency
End FUNCTION

3) Algorithm 2: Generate Association Rules: The second algorithm in our approach is used to generate association rules from frequent itemsets. It first loops through each frequent itemset, and for each itemset it generates all possible subsets. For each subset, it creates an association rule with the antecedent being the subset and the consequent being the complement of the subset in the frequent itemset.

4) Graph Aggregation: The two optimization algorithms previously defined enable finding the best combinations of aggregations. For improving query performance and maximising data retrieval in a graph, OLAP aggregations must be created. Aggregations act as quick-query summaries of data that have
already been calculated, saving time and resources needed to obtain the raw data. Complex queries can be conducted in a fraction of the time it would take to scan the full data set by specifying and producing the relevant aggregations. In our approach, aggregations are stored both in the nodes and in the edges. since the graph allows to put the measures in properties of the links, the advantage of this model will allow us to minimize the number of nodes created in the graph, and also allows us to take advantage of the benefits of graphs. We will also put in the links between the dimensions the information of time dimension.

We will store aggregations in relationships when the aggregation combines only one dimension, two dimensions, or three dimensions (including time dimensions). Aggregations that combine more than three dimensions will be stored in nodes. The link-Aggregation is represented by an edge \( (L^E, N^S, N^T, P^E) \), where:

- \( L^E \) is the label of relationship.
- \( N^S \) is the start node.
- \( N^T \) is the target node.
- \( P^E \) represent the properties of the relationship, the properties are key-value pairs that are used for storing aggregation on relationships.

IV. APPROACH IMPLEMENTATION AND EVALUATION

A. Implementation of the Graph Warehouse

We implemented our graph warehouse using the Neo4j database (version 5.1.0). Neo4j is a graph database management system that uses graph structures with nodes, relationships, and properties to represent and store data[20], enabling efficient storage and querying of complex, interconnected data. Neo4j supports ACID-compliant transactions, offers a flexible data model, and provides a query language called Cypher for working with graph data[21].

In addition, we used the TPC-H benchmark database as a source file. We made use of a global flat CSV file that contains information from a flat meta-model. The TPC-H benchmark database has been used to provide support for Big Data technologies, including NoSQL and Hadoop file systems[22]. It generates data in various file formats (xml, json, csv, tab, ...) following different data models. The dimensions used in the model are as follows:

- Product dimension with the TypeProduct hierarchy.
- Customer dimension with the cityCustomer and RegionCustomer hierarchy.
- Supplier dimension with the citySupplier and RegionSupplier hierarchy.
- Year dimension with the Month hierarchy.

The script in Listing 1 is used to import data from a CSV file (“File.csv”) into Neo4j database by creating nodes to represent different entities related to the (CUSTOMER) as well as the hierarchies associated with these entities (CityCustomer and RegionCustomer). The script uses the APOC (Awesome Procedures on Cypher) library’s apoc.periodic.iterate procedure to efficiently manage the data import from the CSV file. It performs the import by iterating over batches of data, allowing it to process large amounts of data while avoiding overwhelming the system. The function ‘LOAD csv WITH HEADERS FROM “file:///File.csv” as row FIELDTERMINATOR “;” RETURN row’ loads the CSV data with headers into the “row” variable and uses the semicolon (;) as the field delimiter. The first function MERGE (creates or updates) a node with the “CUSTOMER” label having properties “CUSTOMER-ID” and “CUSTOMER-NAME” extracted from the corresponding columns in the CSV file. The options batchSize:10000, allow the data to be processed in batches of 10,000 rows in parallel for better performance. The second use of the MERGE function in the script creates or merges nodes with the “CityCust” label having properties “CUSTOMER-CITYID” and “CUSTOMER-CITY” extracted from the CSV file. Similarly, the third use of the MERGE function is also used for creating or matching nodes in the graph database for the “RegionCust” hierarchy.

Listing 1: The Customer Dimension

```cypher
// Dimension Customer
CALL apoc.periodic.iterate(
  'LOAD csv WITH HEADERS FROM "file:///File.csv" as row FIELDTERMINATOR ";";
  RETURN row',
  'MERGE (C:CUSTOMER [CUSTOMER_ID : row.CUSTOMER_ID, CUSTOMER_NAME : row.CUSTOMER_NAME])',
  { batchSize:10000, parallel:true });
// Hierarchy CityCustomer
CALL apoc.periodic.iterate(
  'LOAD csv WITH HEADERS FROM "file:///File.csv" as row FIELDTERMINATOR ";";
  RETURN row',
  'MERGE (CC:CityCust [CUSTOMER_CITYID : row.CUSTOMER_CITYID, CUSTOMER_CITY : row.CUSTOMER_CITY])',
  { batchSize:10000, parallel:true });
// Hierarchy RegionCustomer
CALL apoc.periodic.iterate(
  'LOAD csv WITH HEADERS FROM "file:///File.csv" as row FIELDTERMINATOR ";";
  RETURN row',
  'MERGE (RC:RegionCust [CUSTOMER_REGIONID : row.CUSTOMER_REGIONID, CUSTOMER_REGION: row.CUSTOMER_REGION])',
  { batchSize:10000, parallel:true });
```

The provided script in Listing 1 only facilitates the creation of nodes and hierarchies but does not establish connections between them. The script in Listing 2 establishes relationships between nodes in the Neo4j database for the “Customer,” “CityCustomer,” and “RegionCustomer” hierarchies based on the data imported from the CSV file. For each row, the script looks for the “CUSTOMER” node with the matching “CUSTOMER-ID” property, and the “CityCust” node with the
matching “CUSTOMER-CITYID” property using the MATCH clauses. The First MERGE clause creates a relationship of type CITY-CUSTOMER between the matched “CUSTOMER” and “CityCust” nodes. The second use of the MERGE clause creates a relationship of type REGION-CUSTOMER between the matched “CityCust” and “RegionCust” nodes.

Listing 2: The relationship between the Customer dimension and its hierarchies

```graphql
// Relationship between Customer and CityCustomer
CALL apoc.periodic.iterate('LOAD CSV WITH HEADERS FROM "file:///File.csv" as row FIELDTERMINATOR "";
  RETURN row',
  'MATCH (C:CUSTOMER [CUSTOMER_ID: row.CUSTOMER_ID])
  MATCH (CC:CityCust [CUSTOMER_CITYID: row.CUSTOMER_CITYID])
  MERGE (C)-[:CITY_CUSTOMER]->(CC)',
  [batchSize:2000, iteratelist:true]);

// Relationship between CityCustomer and RegionCustomer
CALL apoc.periodic.iterate('LOAD CSV WITH HEADERS FROM "file:///File.csv" as row FIELDTERMINATOR "";
  RETURN row',
  'MATCH (C:CUSTOMER [CUSTOMER_ID: row.CUSTOMER_ID])
  MATCH (CC:CityCust [CUSTOMER_CITYID: row.CUSTOMER_CITYID])
  MERGE (CC)-[:REGION_CUSTOMER]->(RC)',
  [batchSize:2000, iteratelist:true]);
```

To create the dimensions “PRODUCT”, “SUPPLIER”, and “Time”, we use a similar approach as employed for the “CUSTOMER” dimension.

After creating all the dimension nodes and their hierarchies in the same way, the next step is to create the fact node that contains the measures, and relationships between the fact and dimension nodes. The following script in Listing 3, demonstrates the creation of the fact node in Neo4j and the relationships between the “FACT” nodes and the corresponding nodes in the “PRODUCT”, “SUPPLIER”, and “TIME” dimensions in the Neo4j graph database.

The script uses the MERGE clause to create a node labeled as “FACT” with the specified properties (“ID”, “Price”, and “QUANTITY”) taken from the corresponding columns in the CSV file.

For each row, the script uses the MATCH clauses to find the “CUSTOMER” node with the matching “CUSTOMER-ID” and the “FACT” node with the matching “ID” property.

The MERGE clause creates a relationship labeled as “FACT-CUSTOMER” between the matched “FACT” and “CUSTOMER” nodes.

Listing 3: The Fact Node

```graphql
// FACT NODE
CALL apoc.periodic.iterate('LOAD CSV WITH HEADERS FROM "file:///File.csv" AS row FIELDTERMINATOR "";
  RETURN row',
  [batchSize:10000, parallel:true]);
```

In the same way, we create relationships between the fact node and the other dimensions (PRODUCT, SUPPLIER, TIME). Using also the apoc.periodic.iterate procedure along with MATCH and MERGE statements to efficiently import data from the CSV file and establish the relationships between the “FACT” nodes and the corresponding nodes in the “PRODUCT”, “SUPPLIER”, and “TIME” dimensions in the Neo4j graph database.

The Fig. 2 represents the implementation of graph warehouse in Neo4j.

![Fig. 2. The graph warehouse of our case study.](image-url)

B. OLAP Operators

Slice

The slice operator in OLAP enables the selection of slices from the data based on a condition on the dimension values [7].

In Listing 4 the slice operator is applied to the Customer Region dimension using the filter condition “AFRICA”, which allows for selecting the data related to the AFRICA.
Listing 4: Selecting Price and Quantity Results from Africa.

```
MATCH (RC:RegionCust [CUSTOMER_REGION: 'AFRICA'])<-[*3]-(m:FACT)
RETURN RC.CUSTOMER_REGION, sum(tofloat(m.Price)), sum(tofloat(m.QUANTITY))
```

Dice

The Dice operator is used in OLAP to select a subset of data based on two or more conditions on dimensions. It is similar to the Slice operator, but allows for finer selection by applying multiple criteria on dimensions simultaneously.

In Listing 5 the Dice operator is applied to the Customer Region and Year dimensions using the filter condition “AFRICA” and “1997”.

Listing 5: Dice-Selecting Quantity Results with a Dice Operation.

```
MATCH (RC:RegionCust [CUSTOMER_REGION : 'AFRICA'])<-[*3]-(m:FACT)
MATCH (Y:YEAR [YEAR : 1994])<-[*2]-(m:FACT)
RETURN RC.CUSTOMER_REGION, Y.YEAR, sum(tofloat(m.QUANTITY)) as QUANTITY
```

Roll Up

In OLAP,[23] the Roll-Up operation is used to aggregate data at a higher level of hierarchy than the current level[24]. It involves moving from a detailed level to a higher-level concept [25]. The Roll-Up operation is performed by grouping the data based on the dimensions and then performing the aggregation function on the measures. The result is a summarized view of the data at a higher level of abstraction. In Listing 6, the Roll Up operation is carried out by moving up the Product dimension’s concept hierarchy (Product → Product Type) and the hierarchy of dimension Time (Month → Year). This query creates a relationship between the two dimensions that contains the aggregated measures.

Listing 6: Roll Up- Price and Quantity summarized by Product Type and Year.

```
MATCH (TP:TYPE_PRODUCT)<-[*2]-(FCT:FACT)
MATCH (Y:YEAR )<-[*2]-(FCT:FACT)
WITH distinct TP,Y, sum(tofloat(FCT.Price)) as Price, sum(tofloat(FCT.QUANTITY)) as QUANTITY
create (TP)<-[ :TYPRD_YEAR_AGG [Price: Price, QUANTITY: QUANTITY]]-(Y)
```

C. Graph Aggregations

After generating the most frequently user queries, the Apriori Algorithm was then used to determine the most common dimensions, and then we executed the second algorithm to generate the most commonly used combinations to create the aggregations. We set the support to 0.4 and the confidence to 0.7. The Fig. 3 shows the combinations of the most commonly used dimensions.

To create aggregations in Neo4j, we use the following script in Listing 7 that stores the aggregations in relationships:

Listing 7: Aggregation Customer-Year

```
CALL apoc.periodic.iterate(
'LOAD CSV WITH HEADERS FROM "file:///File.csv" AS row FIELDTERMINATOR "\";"
RETURN row',
'MATCH (C:CUSTOMER)<-[]-(FCT:FACT)
MATCH (Y:YEAR )<-[*2]-(FCT:FACT)
CREATE (C)<-[ :CUST_YEAR_AGG [Price: toFloat(row.O_TOTALPRICE), QUANTITY: toFloat(row.L_QUANTITY)]]-(A)',
[batchSize:10000, parallel:true]);
```

In Listing 8, the script is used to create aggregations that are stored in nodes, not in relationships.

Listing 8: Aggregation Customer-Product-Supplier

```
CALL apoc.periodic.iterate(
'LOAD CSV WITH HEADERS FROM "file:///File.csv" AS row FIELDTERMINATOR "\";"
RETURN row',
'MATCH (P:PRODUCT)<-[]-(FCT:FACT)
MATCH (C:CUSTOMER)<-[]-(FCT:FACT)
MATCH (S:SUPPLIER) <-[]-(FCT:FACT)
create (PCS:PROD_CUST_SUPP [PRICE: toFloat(row.O_TOTALPRICE), QUANTITY: toFloat(row.L_QUANTITY)])
create (P)<-[ :PROD_3]-(PCS)
create(C)<-[ :CUST_3]-(PCS)
create (S)<-[ :SUPP_3]-(PCS)',
[batchSize:10000, parallel:true]);
```
The Fig. 4 shows the creation of aggregations using in approach.

D. Experimental Results and Evaluation

To validate our approach and measure the effectiveness of optimizing the OLAP cube in the graph, we conducted a series of experiments in which we evaluated performance before and after using our optimization approach. We measured the query execution time before and after adding optimized aggregations, and compared the execution times to determine if adding the optimized aggregations led to a significant improvement in performance. We conducted our test on an i7 processor machine with 16GB of RAM and 1TB of storage memory. Additionally, we used the TPC-H database with a scale factor of SF1 (1GB). We use in Table II, Cypher queries before and after optimization.

<table>
<thead>
<tr>
<th>Query</th>
<th>Before Optimization</th>
<th>After Optimization</th>
</tr>
</thead>
<tbody>
<tr>
<td>Q 1</td>
<td>MATCH (RS:RegionSupp)&lt;-[*:3]-(FCT:FACT) <strong>return</strong> SUM(tofloat(FCT.Price)) AS Price, RS.SUPPLIER_REGION</td>
<td>MATCH (RS:RegionSupp) &lt;-[:REGION_SUPP_AG]- (RA:Region_Supp_AGG) <strong>return</strong> RA.Price, RS.SUPPLIER_REGION</td>
</tr>
<tr>
<td>Q 2</td>
<td>MATCH (Y:YEAR)&lt;-[:REGION_SUPP_AG]- (RA:Region_Supp_AGG) Y.YEAR, TP.BRAND <strong>return</strong> Y.YEAR, TP.BRAND, SUM(tofloat(r.Price)) as Price</td>
<td>MATCH (TP:TYPE_PRODUCT) &lt;-[:TYPE_PROD_AGG]-(TPAGG:TYPE_PROD_AGG) <strong>return</strong> TPAGG.QUANTITY, TP.BRAND</td>
</tr>
<tr>
<td>Q 3</td>
<td>MATCH (C:CUSTOMER) &lt;-[:FACT_CUSTOMER]-(FCT:FACT) <strong>return</strong> SUM(tofloat(FCT.Price)) AS Price, C.CUSTOMER_NAME, P.PRODUCT_NAME</td>
<td>MATCH (P:PRODUCT)&lt;-[:FACT_PRODUCT]-(FCT:FACT) <strong>return</strong> SUM(tofloat(FCT.QUANTITY)) As QUANTITY, TP.BRAND</td>
</tr>
<tr>
<td>Q 4</td>
<td>MATCH (P:PRODUCT)&lt;-[:FACT_PRODUCT]-(FCT:FACT) <strong>return</strong> SUM(tofloat(FCT.QUANTITY)) As QUANTITY, TP.BRAND</td>
<td>MATCH (P:PRODUCT)&lt;-[:FACT_PRODUCT]-(FCT:FACT) <strong>return</strong> SUM(tofloat(FCT.QUANTITY)) As QUANTITY, TP.BRAND</td>
</tr>
<tr>
<td>Q 5</td>
<td>MATCH (P:PRODUCT)&lt;-[:FACT_PRODUCT]-(FCT:FACT) MATCH (C:CUSTOMER) &lt;-[:FACT_CUSTOMER]-(FCT:FACT) MATCH (S:SUPPLIER) &lt;-[:FACT_SUPPLIER]-(FCT:FACT) <strong>return</strong> SUM(tofloat(FCT.Price)) AS PRICE, P.PRODUCT_NAME, C.CUSTOMER_NAME, S.SUPPLIER_NAME LIMIT 43</td>
<td>MATCH (P:PRODUCT)&lt;-[:FACT_PRODUCT]-(FCT:FACT) MATCH (C:CUSTOMER) &lt;-[:FACT_CUSTOMER]-(FCT:FACT) MATCH (S:SUPPLIER) &lt;-[:FACT_SUPPLIER]-(FCT:FACT) <strong>return</strong> SUM(tofloat(FCT.Price)) AS PRICE, P.PRODUCT_NAME, C.CUSTOMER_NAME, S.SUPPLIER_NAME LIMIT 43</td>
</tr>
</tbody>
</table>
The Fig. 5 shows the query execution time before and after the optimization of the model.

![Query Execution in neo4j]

The results demonstrate that the execution time of the queries decreased after the optimization and usage of OLAP aggregations, although the execution time may vary depending on the complexity of the query, with complex queries requiring traversal of numerous relationships taking more time, while simple queries involving only a few relationships having relatively short execution times.

For instance, in the first query, the execution time before optimization was approximately 14266 milliseconds, and after optimization, it reduced to 3 milliseconds. This translates to an impressive percentage improvement of approximately 99%. Furthermore, in the query 4, we achieved considerable improvements as well, the execution time decreased from 3,978 milliseconds to 2 milliseconds, substantial gains in performance clearly demonstrate the effectiveness of our approach in making the system nearly 2,000 times faster than its previous state.

The significant reduction in execution time showcases how this approach can make the system multiple times faster than its previous state, enhancing the efficiency of reporting and data analysis within the graph-based data warehouse.

We also compared our model implemented in the graph and the ROLAP model, comparing the execution time of the same multidimensional queries in Neo4j and Oracle.

Table III shows the queries used in the comparison between Graph OLAP and ROLAP.

<table>
<thead>
<tr>
<th>Query</th>
<th>Relational Query</th>
<th>Graph Query</th>
</tr>
</thead>
<tbody>
<tr>
<td>Q 1</td>
<td>SELECT sum(_totalprice) as Price, supplier_region from w_fact_lgb_f3 Group by supplier_region</td>
<td>MATCH (RS:RegionSupp) &lt;-[:REGION_SUPP_AGG]- (RA:Region_Supp_AGG) return RA.Price, RS.SUPPLIER_REGION</td>
</tr>
<tr>
<td>Q 2</td>
<td>SELECT sum(_totalprice) as Price, orderyear AS YEAR, brand as PRODCUT_TYPE from w_fact_lgb_f3 GROUP BY orderyear, brand</td>
<td>MATCH(TP:TYPE_PRODUCT ) &lt;-[r:TYPRD_YEAR_AGG] -(Y:YEAR) return Y.YEAR, TP.BRAND ,sum(tofloat(r.Price ) ) as Price</td>
</tr>
<tr>
<td>Q 3</td>
<td>SELECT sum(_totalprice) as Price, product_name, customer_name from w_fact_lgb_f3 GROUP BY product_name, customer_name</td>
<td>MATCH (P:PRODUCT)- [:PROD_3]-(PCS:PROD_CUST_SUPP) MATCH (C:CUSTOMER)&lt;-[:CUST_3]-(PCS:PROD_CUST_SUPP) MATCH (S:SUPPLIER)&lt;-[:SUPP_3]-(PCS:PROD_CUST_SUPP) return PCS.PRICE,P.PRODUCT_NAME, C.CUSTOMER_NAME,S.SUPPLIER_NAME LIMIT 43</td>
</tr>
<tr>
<td>Q 4</td>
<td>SELECT sum(l_quantity) as QUANTITY, brand as PRODUCT_TYPE FROM w_fact_lgb_f3 GROUP BY brand</td>
<td>MATCH (TP:TYPE_PRODUCT) &lt;-[TYPE_PROD_AGG] -(TPAGG:TYPE_PROD_AGG ) return TPAGG.QUANTITY , TP.BRAND</td>
</tr>
<tr>
<td>Q 5</td>
<td>select * from ( SELECT sum(_totalprice) as Price, product_name, customer_name, supplier_name FROM w_fact_lgb_f3 GROUP BY product_name, customer_name, supplier_name) where ROWNUM &lt;= 43;</td>
<td>MATCH (P:PRODUCT)&lt;-[:PROD_3]-(PCS:PROD_CUST_SUPP) MATCH (C:CUSTOMER) &lt;-[:CUST_3]-(PCS:PROD_CUST_SUPP) MATCH (S:SUPPLIER) &lt;-[:SUPP_3]-(PCS:PROD_CUST_SUPP) return PCS.PRICE,P.PRODUCT_NAME, C.CUSTOMER_NAME,S.SUPPLIER_NAME LIMIT 43</td>
</tr>
</tbody>
</table>
The results also show that the Graph cube delivers performance levels that are better than those of the ROLAP model. We also notice that Graph databases provide a great degree of flexibility for searching through data by conducting more complex pathways or by following direct linkages. While SQL queries that use joins to mix data from various tables provide the basis for data traversal in a relational architecture, while joins can be effective, they can also be less flexible and intuitive when navigating complex relationships.

V. CONCLUSION

Graph-Oriented Databases offers a clear abstraction for managing heavily connected data and modelling complicated domains. In this Paper we present our contribution for developing a data warehouse under a graph database, our approach relies on the properties of graphs to implement graph data warehouse. To enhance the graph cube’s performance we provide a new technique that optimizes the choice of OLAP aggregations by using the association rules algorithm.

To validate our approach and measure the effectiveness of OLAP cube optimization in the graph, we conducted a series of experiments in which we evaluated the performance before and after the optimization, we also compared our model with the relational model in terms of query performance. The experiment’s findings demonstrate the benefits of creating a framework for building olap cubes on graphs. In [4], Amine Ghrab, Oscar Romero, Sabri Skhiri, Alejandro Vaisman, and Esteban Zimányi. A framework for building OLAP cube optimization in the graph, we conducted a series of experiments in which we evaluated the performance of OLAP cube optimization in the graph, we conducted a series of experiments in which we evaluated the performance of OLAP cube optimization in the graph.


[5] Peixiang Zhao, Xiaolei Li, Dong Xin, and Jiawei Han. Graph cube: on warehousing and OLAP multidimensional networks. In Proceedings of the 2011 ACM SIGMOD International Conference on Management of Data, pages 853–864, Athens Greece, June 2011. ACM.


Cloud Task Scheduling using Particle Swarm Optimization and Capuchin Search Algorithms

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Hebei Normal University of Science & Technology, Qinhuangdao 066004, China

Abstract—Cloud providers offer heterogeneous virtual machines for the execution of a variety of tasks requested by users. These virtual machines are managed by the cloud provider, eliminating the need for users to set up and maintain their hardware. This makes accessing the computing resources necessary to run applications and services more accessible and cost-effective. The task scheduling problem can be expressed as a discrete optimization issue known as NP-hard. To address this problem, we propose a hybrid meta-heuristic algorithm using the Capuchin Search Algorithm (CapSA) and the Particle Swarm Optimization (PSO) algorithm. PSO excels in global exploration, while CapSA is adept at fine-tuning solutions through local search. We aim to achieve better convergence and solution quality by integrating both algorithms. Our proposed method's performance is thoroughly evaluated through extensive experimentation, comparing it to standalone PSO and CapSA approaches. The findings reveal that our hybrid algorithm outperforms the individual techniques in terms of both total execution time and total execution cost metrics. The novelty of our work lies in the synergistic integration of PSO and CapSA, addressing the limitations of traditional optimization methods for cloud task scheduling. The proposed hybrid approach opens up intriguing directions for future research in dynamic task scheduling, multi-objective optimization, adaptive algorithms, integration with emerging technologies, and real-world deployment scenarios.

Keywords—Cloud computing; virtualization; task scheduling; optimization; resource utilization; capuchin search algorithm; particle swarm optimization

I. INTRODUCTION

Cloud computing is a well-known computing model that hosts and delivers various services via the Internet [1]. It enables users to access computing resources on demand, thus reducing the cost of ownership and IT management. Cloud computing services are provided pay-as-you-go and typically include storage, software, analytics, and networking [2]. The cloud computing paradigm has allowed companies to move, process, and run some of the services and applications in cloud environments, providing convenient access to a wide range of resources easily and conveniently. In addition, these services are tailored to each customer's requirements [3]. Originally, due to the emergence of big data, conventional hardware could not handle heterogeneous workloads flowing onto the infrastructure. Consequently, many IT companies are migrating their infrastructure to the cloud to handle these diverse and heterogeneous tasks. Cloud computing models offer several potential benefits, including flexibility, highly resilient virtual architectures, on-demand services, elasticity, and scalability. Multi-tenant computing environments share resources among users. A scheduler module checks the resource status and allocates it under user requests [4]. Scheduling plays a crucial role in achieving optimal real-time performance in cloud computing. The scheduling algorithms map the tasks into the cloud environment and utilize the available resources, thereby reducing latency and response time for requests and increasing resource utilization and system throughput [5].

The fusion of IoT, big data, artificial intelligence (AI), machine learning (ML), deep learning, feature and channel selection, meta-heuristic algorithms, and association rule mining with cloud computing has ushered in a new era of technological possibilities. IoT connects a vast array of devices, generating immense volumes of data that can be harnessed in the cloud for analysis and processing [6]. Big data, with its inherent complexity and scale, finds a natural fit in cloud computing, which offers the necessary storage and computational capabilities to extract valuable insights [7, 8]. AI and ML form the backbone of intelligent cloud applications, enabling systems to learn from data patterns and make informed decisions [9-11]. Deep learning, a subset of ML, is especially powerful in cloud computing for tasks such as image recognition, natural language processing, and complex data analysis [12-14]. Feature and channel selection play a crucial role in optimizing cloud-based applications by identifying pertinent data attributes and sources, leading to improved efficiency and accuracy [15]. Meta-heuristic algorithms, with their ability to efficiently solve complex optimization problems, facilitate resource allocation, load balancing, and task scheduling in cloud environments [16-19]. Association rule mining is essential for discovering meaningful patterns and relationships within vast datasets, empowering businesses to make data-driven decisions and gain a competitive edge [20]. The integration of these concepts in cloud computing is transformative for businesses and industries alike. Cloud-based IoT solutions facilitate real-time data analysis and decision-making, enabling predictive maintenance, personalized services, and improved operational efficiency. The ability to process and store big data in the cloud ensures data accessibility, scalability, and cost-effectiveness for organizations. Furthermore, AI and ML capabilities in the cloud enable innovative applications like virtual assistants, recommendation systems, and fraud detection.

Many researchers have addressed the problem of scheduling tasks; however, it has remained an NP-hard problem. This means that scheduling tasks is a computationally difficult problem, and finding an optimal solution is not feasible in a reasonable time. As a result, numerous heuristics have been created to solve this issue effectively [21]. Cloud
environments consist of geographically distributed data centers. These data centers are connected by high-speed networks, making it possible to transfer data between them quickly. This enables cloud providers to use distributed scheduling algorithms that can use multiple data centers to solve the task scheduling problem more efficiently. There are thousands of servers in each data center. Each server has an array of virtual machines equipped with various resources, such as storage, CPU, and memory. To execute tasks, groups of virtual machines are allocated to cloud users. It is a complex task to schedule the appropriate resources for the task. In order to assign tasks to virtual resources, it is necessary to examine their characteristics with respect to dependency, length, and size. By factoring in the total execution time for all tasks, task scheduling algorithms can distribute the workload across virtual machines.

In this research paper, we introduce a novel hybrid meta-heuristic algorithm that combines the strengths of the Capacitive Search Algorithm (CapSA) and the Particle Swarm Optimization (PSO) algorithm. The decision to merge these two approaches was motivated by their complementary characteristics in tackling optimization problems. While PSO is renowned for its efficient global exploration capabilities, CapSA excels in local search strategies, making it adept at fine-tuning solutions. The integration of PSO and CapSA allows us to harness their unique strengths synergistically, aiming for improved convergence and solution quality. During the experimentation phase, our hybrid approach demonstrated superior performance to standalone PSO and CapSA, as evident in total execution time and total execution cost metrics. Furthermore, our evaluation of existing methods for cloud task scheduling revealed specific limitations hindering their effectiveness in this context. Traditional optimization algorithms often struggle to find solutions in large search spaces because they lack robust exploration capabilities. On the other hand, local search algorithms may get trapped in local optima, preventing them from achieving global optimality. Our proposed hybrid PSO-CapSA approach effectively addresses these limitations. By leveraging PSO’s global exploration capabilities, the algorithm conducts a diverse search across the solution space, mitigating the risk of premature convergence to suboptimal solutions. Additionally, CapSA’s local search enhances the precision of the algorithm by refining solutions and escaping local optima. This combination empowers our method to tackle cloud task scheduling challenges more effectively than existing approaches.

The paper is arranged in the following manner. Section II summarizes the literature. Section III discusses the problem statement and the proposed algorithm. Section IV illustrates the findings and analyses. Section V presents the conclusion.

II. RELATED WORK

Dai, et al. [22] propose a novel task-scheduling algorithm that incorporates multiple quality of service criteria into the scheduling process, like reliability, security, expenditure, and time. It combines genetic as well as Ant Colony Optimization (ACO) algorithms. ACO employs the genetic algorithm to generate an initial pheromone efficiently. A four-dimensional quality of service objective is evaluated utilizing a designed fitness function. The optimum resource is then identified using the ACO algorithm. The proposed algorithm was implemented on several real-world tasks and demonstrated significant improvements in the quality of service. The consequences demonstrated that the suggested algorithm achieved better scheduling than existing algorithms.

Tang, et al. [23] proposed the DVFS-enabled Energy-efficient Workflow Task Scheduling (DEWTS) algorithm to reduce energy consumption and ensure the quality of service adhering to deadlines. DEWTS is built upon a dynamic voltage frequency scaling approach that assigns appropriate processing speeds to tasks based on their deadlines. By combining potentially inefficient processors, DEWTS can utilize the slack time repeatedly after servers have been merged by reclaiming the slack time. This ensures that peak performance is maintained and the power wasted is minimized. The DEWTS algorithm can effectively manage servers with different deadlines and dynamically adjust the processors’ frequency and voltage. After calculating the starting scheduling sequence for all tasks, DEWTS determines the total makespan and deadline by applying the Heterogeneous-Earliest-Finish-Time (HEFT) algorithm. The underutilized processors are combined by terminating the last node and reallocating the assigned tasks to the processors according to the number of running tasks and the energy consumption of each processor. Results from the experiments demonstrate that DEWTS reduces entire power consumption by up to 46 percent for a variety of parallel applications while balancing scheduling performance based on randomly generated DAG workflows.

Keshanchi, et al. [24] introduced a powerful and enhanced genetic algorithm to optimize task-scheduling solutions. Based on model-checking techniques, they have developed a behavioral modeling approach to verify the algorithm’s validity. Next, Linear Temporal Logic (LTL) functions are used to extract the expected specifications. A Labeled Transition System (LTS) is used to obtain optimal results in the validation process. The algorithm was tested on various tasks and provided better performance and scalability than existing methods. The results were validated using the LTS model, and the algorithm was effective in generating optimal task-scheduling solutions. According to the verification outcomes, the validity of the suggested algorithm is assessed with respect to some reachability, fairness, specifications, and deadlock-free performance. Statistical analysis, as well as simulation findings, indicate that the designed approach is superior to traditional heuristic algorithms. The proposed algorithm has been successfully implemented in a deployed system, demonstrating its effectiveness and scalability. The results of the verification process show that the algorithm achieves the desired results with minimal computational cost.

Lin, et al. [25] presented a power efficiency model for cloud servers. To optimize energy consumption in cloud environments, they propose a heuristic task scheduling algorithm (ECOTS) that utilizes the power efficiency of the server to guide task scheduling. Multiple vital factors are taken into account by ECOTS, including degradation of performance, power efficiency models for servers, and resource requirements for tasks, with the goal of reducing the energy consumption of the system without compromising performance. ECOTS is
characterized by its simplicity in terms of time and space and the ability to search globally to find an effective scheduling strategy. An evaluation of the effectiveness of ECOTS was conducted by simulating a heterogeneous cluster environment. It has been demonstrated that the ECOTS algorithm achieves the highest energy efficiency level.

The energy efficiency of task scheduling in cloud data center architectures was studied by Sharma and Garg [26], who also created a new hybrid meta-heuristic algorithm according to the harmony-inspired genetic algorithms. It combines the exploration capabilities of a genetic algorithm with the exploitation capabilities of harmony searches to provide rapid convergence while intelligently sensing both local and global optimal regions without wasting time in local or global optimal regions. Key goals include reducing the time required for computation and the energy it consumes. In contrast, secondary objectives include reducing energy consumption and scheduling execution overhead.

Alsaidy, et al. [27] propose a heuristic algorithm for developing the initialization of the Particle Swarm Optimization (PSO) algorithm. The PSO is initially configured using the minimum completion time (MCT) and longest job to fastest processor (LJFP) algorithms. Both algorithms are tested in terms of their efficiency to minimize total energy consumption, degree of imbalance, and total execution time. A comparison is also made between the proposed algorithms and recent methods. The simulation outcomes demonstrate that the suggested algorithms are more effective and superior to traditional PSO and comparative algorithms.

Saravanan, et al. [28] used the enhanced wild horse optimization algorithm and the levy flight algorithm for task scheduling. This method generates a multi-objective fitness function by optimizing resource utilization and minimizing the makespan. The simulation results indicated that the suggested method outperformed others in a variety of situations. The algorithm was successful in achieving better task scheduling compared to traditional algorithms. It achieved better outcomes regarding completion time, cost, as well as energy efficiency. The proposed method was also found to be scalable and robust, able to handle a large number of tasks and resources.

III. PROPOSED METHOD

This section begins with a description of the problem statement. Then, the standard PSO and CapSA algorithms are discussed to determine the basis for the suggested algorithm. This section also examines the fitness function incorporated into the proposed algorithm.

A. Problem Statement

The process of scheduling cloud-based tasks is illustrated in Fig. 1. Initially, users' tasks are placed in a queue. The tasks are then retrieved from the queue and allocated to the available cloud resources. Once the tasks are finished, the results are sent back to the users. This procedure continues until each task is completed. Generally, static and dynamic are the two kinds of task scheduling algorithms. Static scheduling algorithms require detailed information regarding the environment and the tasks. Dynamic scheduling algorithms monitor the system continuously and are capable of balancing workloads. In the following way, the task scheduling problem is described. The task scheduling problem consists of assigning n tasks of users to m heterogeneous virtual machines based on some constraints in order to optimize some objective functions. We have observed that task scheduling is an objective-driven strategy that involves allocating computing resources to specific tasks periodically to maximize one or more objectives. This process requires analysis of data related to the task, such as the resource requirements, the time needed to complete the task, and the task's priority. Optimizing task scheduling can result in significant cost savings by allocating resources efficiently. Furthermore, effective task scheduling can help improve the system's overall performance. In the cloud computing environment, the scheduling policy should address this problem from two different perspectives, customer and cloud provider perspectives. The customer's objective is to lower the cost of executing the tasks, and the cloud provider's objective is to maximize the utilization of the infrastructure. Thus, cost and performance should be considered when designing the scheduling policy. In this regard, scheduling is a major concern in the cloud environment, as it directly impacts the performance of the cloud system as well as the cloud service consumer.
Assume that \( VM = [ VM_1, VM_2, \ldots, VM_m ] \) represents the set of virtual machines available within a data center. These virtual machines are provided to cloud users in order to fulfill their task requests. Each virtual machine is equipped with processing power expressed in Millions of Instructions Per Second (MIPS). A cloud broker allocates tasks to appropriate virtual machines according to details of existing resources and requirements of tasks. The collection of tasks to execute on the virtual machines at the data center is expressed as \( T = [T_1, T_2, \ldots, T_n]\). Tasks are defined by their length and processing requirements, which are specified by Millions of Instructions (Mls).

**B. Fitness Function**

Fitness functions represent the desired objectives to be optimized. They measure the performance of the problem being solved and direct the search process toward finding the best solution. Fitness functions are generally represented as mathematical functions, which can be used to evaluate the candidate solutions. Fitness functions may be multi-objective from two perspectives: prior and posterior. In an a priori approach, the fitness function is designed to optimize multiple objectives from the beginning. In a posteriori approach, a single-objective fitness function is used, and the multiple objectives are satisfied through post-processing. According to the prior strategy, objectives are given a weight that reflects their significance in producing a single-value function, referred to as a fitness value. The posterior approach produces non-dominant solutions. This paper uses priori principles to develop the fitness function. Total execution time and total execution cost are included in the fitness function. In mathematical terms, the fitness function considered is expressed by Eq. 1.

\[
F = a_1 \times TEC + a_2 \times TET \tag{1}
\]

Cloud computing is designed to meet users’ functional needs while reducing costs. Consequently, a scheduling algorithm should provide users access to their necessary applications at a minimal cost. Costs associated with storage, communication, and execution are all contained in the cost of cloud computing. Execution cost consists of the price per unit interval applied by the virtual machine and the execution time of all tasks executed by that virtual machine. Eq. 2 can be used to calculate the total execution cost of a workflow.

\[
TEC_W = \sum_{i=1}^{k} \frac{ET_{ij}}{\tau}\times CO_i : j \in VM_j \tag{2}
\]

In Eq. 2, \( ET_{ij} \) measures the time taken to execute task \( T_j \) by \( j \)th VM. \( \tau \) is the period during which the user utilizes the resources, and \( CO_i \) represents the cost of a type-i VM instance for a unit of time in the cloud data center. The total execution time or makespan is a key metric used to measure the performance of a task scheduling approach. It is the sum of the longest completion times of tasks within a workflow. Optimizing the makespan is an important part of workflow scheduling. Eq. 3 can be used to calculate the makespan of a workflow.

\[
TET_W = \max [CT_i | i = 1,2,\ldots,m] \tag{3}
\]

In Eq. 3, \( CT_i \) is represented the completion time of task \( T_i \) in the workflow. In other words, it is characterized as the variation between task \( T_i \)'s start and end times. Eq. 4 is used to calculate the completion time. According to Eq. 5, the waiting time of task \( T_i \) equals the total completion time of its predecessors. Eq. 6 calculates the execution time of task \( T_i \) on \( VM_j \). \( PE_{unit} \) represents the size of each core in MIPS. \( Num(PE_i) \) indicates how many cores are allocated to the virtual machine \( VM_j \), as well as \( SZ_{Task} \) indicates the size of task \( T_i \). in MI.

\[
CT_i = \begin{cases} 
ET_i & \text{if pred}(T_i) = 0 \\
WK_i + ET_i & \text{if pred}(T_i) \neq 0 
\end{cases} \tag{4}
\]

\[
WK_i = \begin{cases} 
0 & \text{if pred}(T_i) = 0 \\
\max(CT_j) & \text{if pred}(T_i) \neq 0 
\end{cases} \tag{5}
\]

\[
ET_{ij} = \frac{SZ_{Task}}{Num(PE_i) \times PE_{unit}} \tag{6}
\]

**C. Proposed Algorithm**

The proposed algorithm combines the PSO and Capuchin Search Algorithm (CapSA). The algorithm consists of running the PSO algorithm during the first half of the total iterations, initializing the most optimal solution produced by the PSO algorithm (gbest) to CapSA, and running CapSA for the second half of the total iterations. The best CapSA solution is the most efficient assignment of tasks to virtual machines. After the total iterations are finished, the algorithm takes the best solution from PSO and CapSA and finds the best fitness value. This solution is returned as the final output of the proposed algorithm.

In order to apply any algorithm to a workflow scheduling problem, the problem must be modeled in terms of the tasks that need to be completed, their precedence relationships, the resources and time needed to complete each task, and other factors. Once the problem is accurately modeled, the appropriate algorithm can be applied to find an optimal solution. This problem can be viewed as a mapping between user tasks and virtual machines. As shown in Fig. 2, an array can represent the proposed algorithm's solution. Tasks and assigned VMs are represented by an array's index and array's values, respectively. Population refers to the set of solutions. In the first iteration, the population is first initialized using a random solution. As the algorithm is iterated, the solution is improved. Fig. 3 illustrates a random population initialization. Each population is evaluated to determine its fitness. The population with the highest fitness is considered the most optimal solution. Selection processes are then used to identify solutions for reproduction.

<table>
<thead>
<tr>
<th>T1</th>
<th>T2</th>
<th>T3</th>
<th>T4</th>
<th>T5</th>
<th>T6</th>
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<tr>
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<td>4</td>
<td>5</td>
<td>3</td>
<td>2</td>
<td>3</td>
<td>4</td>
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</table>

Solution 1

<table>
<thead>
<tr>
<th>T1</th>
<th>T2</th>
<th>T3</th>
<th>T4</th>
<th>T5</th>
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<th>T8</th>
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</thead>
<tbody>
<tr>
<td>2</td>
<td>5</td>
<td>2</td>
<td>5</td>
<td>4</td>
<td>1</td>
<td>3</td>
<td>2</td>
</tr>
</tbody>
</table>

Solution 2

Fig. 2. A solution array.
Using Eq. 7, the algorithm determines the execution time and assigns them to the execution time matrix. The element value demonstrates the execution time; for example, ET_{1,1} represents the execution time of task T_1 on VM_1. According to Eq. 8, the cost matrix contains the execution cost of each virtual machine, C_1, C_2, ..., C_m correspond to the unit execution costs of the virtual machines VM_1, VM_2, ..., VM_m.

\[
ET - Mtx = T_1 \begin{bmatrix} ET_{1,1} & ET_{1,2} & \ldots & ET_{1,m} \\ ET_{2,1} & ET_{2,2} & \ldots & ET_{2,m} \\ \vdots & \vdots & \ddots & \vdots \\ ET_{n,1} & ET_{n,2} & \ldots & ET_{n,m} \end{bmatrix}
\]

\[
\text{Cost} - \text{Mtx} = (C_1, C_2, \ldots, C_m)
\]

The PSO algorithm is an evolutionary algorithm derived from bird swarms or fish schools. It optimizes an issue by iteratively seeking to enhance a candidate solution concerning a given quality measure. It works by moving a population of candidate solutions, known as particles, throughout the search space in accordance with straightforward mathematical formulas. The movement of particles is guided by their best-known position and the best-known positions in the search space, which are updated as other particles find better positions. Position pi and velocity vi are the two parameters that describe a particle. Pbest and gbest are two parameters that affect the particle’s position. Gbest represents the best position of neighboring particles, while Pbest represents the best position visited by the particle. Position and velocity are updated after each iteration of the algorithm. The velocity is updated using Eq. 9.

\[
v_{id}^{t+1} = w v_{id}^t + c1 r1 (p_{best_{id}} - x_{id}^t) + c2 r2 (g_{best_{id}} - x_{id}^t)
\]

In Eq. 9, v_{id}^t corresponds to the velocity of the ith particle in the dth dimension on iteration t. Eq. 10 is used to update the particle’s position. In Eq. 10, p_{lb_{id}} displays the position of particle i in the dth dimension at time t.

\[
p_{lb_{id}}^{t+1} = p_{lb_{id}} + v_{id}^t
\]

The CapSA is inspired by capuchin monkeys’ dynamic behavior when navigating between branches and riverbanks for food [29, 30]. It uses three great navigation methods: jumping, swinging, and climbing. According to CSA, capuchin populations can be categorized into two important groups: the leaders and the followers. The leaders guide their followers and keep track of each other. Three principal concepts are used by capuchin leaders and swarm members when looking for food sources: explore independently, cooperate in locating better food sources, and lead by example. In the search for food sources, the Alpha males, as well as Alpha females, lead the other group members. By assisting the other group members in discovering food sources, the Alpha male serves as a leader. An iterative solution is determined using the above strategies. In order to select the best features, CapSA follows the following steps:

- CapSA initialization: Like other meta-heuristic algorithms, CapSA generates a predetermined number of individuals (i.e., capuchins) to serve as its population. Each individual represents a potential answer to the task scheduling problem in this paper. A capuchin array can be viewed as a d-dimensional matrix. The matrix for the initial population is shown in Eq. 11.

\[
X = \begin{bmatrix} x_1 & x_2 & \ldots & x_d \\ x_1 & x_2 & \ldots & x_d \\ \vdots & \vdots & \ddots & \vdots \\ x_1 & x_2 & \ldots & x_d \end{bmatrix}
\]

- In Eq. 11, n stands for the number of capuchins, d demonstrates the number of variables, xid indicates the dimension of the ith capuchin, and x signifies the position of the capuchins. Eq. 12 is used to compute the first situation of each capuchin.

\[
x_{i}^{t} = u_{b_{j}} + t \times (u_{b_{j}} - l_{b_{j}})
\]

- In Eq. 12, ub_{j} and lb_{j} define upper and lower bounds for the jth capuchin, respectively, as well as t varies uniformly from 0 to 1.

- Generation of solutions by capuchins: In CapSA, a new population originates from the position of the capuchin, the best capuchin, and F, which represents the food source. In a d-dimensional search domain, capuchins should iteratively update this food source. The following equation is used in our proposed algorithm to generate new solutions. In Eq. 13, P_{lb_{j}} is the probability of the balance generated by the tail of the capuchin during leaping movements, θ is the angle at which capuchins jump, v{j} is the velocity of the ith capuchin in the dimension j, g is gravity equal to 9.81, F_{j} refers to the food’s position in the dimension, and x{j} refers to the position of the alpha capuchins and their following capuchins within the dimension j. A capuchin’s jump angle can be determined using Eq. 14, where r is a uniformly generated number from 0 to 1.

\[
x_{i}^{t} = F_{j} + \frac{P_{lb_{j}}v_{j}^2\sin(2\theta)}{g}
\]

\[
\theta = \frac{3}{2} r
\]

IV. EXPERIMENTAL RESULTS

In this section, the proposed algorithm’s performance is compared with previous algorithms under Montage workflows.
with 50 and 100 tasks. The proposed algorithm was simulated and evaluated using the WorkflowSim -1.1 toolkit, an extension of Cloudsim. Comparisons are made among the proposed algorithm as well as standalone PSO and CapSA. Table I provides a summary of the simulation parameters that were utilized for evaluating the algorithm.

1) **Total execution time comparison (50 tasks):** As depicted in Fig. 4, the algorithms were compared regarding the total execution time as iterations increased from 50 to 500. The proposed algorithm consistently outperformed CapSA in all iterations. While slight degradations were observed in comparison with PSO at several iterations, the proposed algorithm maintained consistent performance as the number of iterations increases.

2) **Execution costs comparison (50 tasks):** Fig. 5 illustrates the comparison of execution costs among the algorithms. The proposed algorithm exhibited improvements of 9.7%, 1.4%, 12.6%, 10.6%, 3.8%, 15.2%, 11.1%, 14.4%, 19.9%, and 13.3% compared to CapSA for all considered iterations. In terms of PSO, the proposed algorithm demonstrated a significant drop in total execution cost of up to 10%. Overall, the experimental results indicate that the proposed algorithm is more efficient than both CapSA and PSO. Additionally, the proposed algorithm exhibited faster convergence compared to these algorithms.

3) **Total execution time comparison (100 tasks):** Fig. 6 demonstrates that for 100 tasks, the proposed algorithm consistently reduced the total execution time compared to CapSA and PSO in all iterations.

4) **Execution costs comparison (100 tasks):** In Fig. 7, the algorithms were compared based on their total execution costs for 100 tasks. The proposed algorithm outperformed CapSA in all iterations, with a decrease of 0.6%, 0.3%, 0.2%, 1%, 0.9%, 1.2%, 1.4%, 0.4%, 1.1%, and 1.3% for all considered iterations. Compared to PSO, the proposed algorithm showed improvements of 0.11%, 0.04%, 0.02%, and 0.15% for iterations 200, 250, 300, and 500, respectively. These results indicate that the proposed algorithm is more effective and efficient in finding better solutions for the given problem compared to CapSA and PSO. Furthermore, it consistently produces better results as the number of iterations increases. Overall, the presented results demonstrate the superior performance of the proposed hybrid algorithm in terms of both total execution time and execution costs when compared to the standalone PSO and CapSA algorithms.

### Table I. Simulation Parameters

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>VM policy</td>
<td>Time shared</td>
</tr>
<tr>
<td>Number of processors</td>
<td>1</td>
</tr>
<tr>
<td>Bandwidth</td>
<td>1000</td>
</tr>
<tr>
<td>Ram</td>
<td>512 MB</td>
</tr>
<tr>
<td>MIPS</td>
<td>1000</td>
</tr>
<tr>
<td>Number of virtual machines</td>
<td>5</td>
</tr>
<tr>
<td>Number of tasks</td>
<td>50-100</td>
</tr>
</tbody>
</table>

![Fig. 4. Execution time for 50 tasks.](www.ijacsa.thesai.org)
Fig. 5. Execution cost for 50 tasks.

Fig. 6. Execution time for 100 tasks.

Fig. 7. Execution cost for 100 tasks.
V. CONCLUSION

Cloud services offer a wide range of solutions to the varying computing challenges in the real world. In order to accomplish a variety of tasks, users access various cloud services. Services provided by different service providers are used to handle such tasks. In this paper, a novel hybrid meta-heuristic algorithm according to the PSO algorithm as well as CapSA is proposed. The proposed algorithm uses PSO and CapSA to employ complementary global and local search strategies. Based on simulation results from the Cloudsim simulator, our algorithm outperformed standalone PSO and CapSA on both total execution cost and total execution time measures. There are several potential open problems in this domain for further research. Extending the proposed algorithm to handle dynamic task scheduling scenarios, where tasks and resource availability fluctuate in real-time, will be essential for addressing real-world dynamic workload challenges. This adaptability will ensure the algorithm remains efficient and effective in dynamic cloud environments. Exploring multi-objective optimization techniques to optimize cloud task scheduling for various performance metrics, such as energy consumption, resource utilization, and quality of service (QoS), can lead to more comprehensive and versatile solutions. Exploring the integration of the proposed algorithm with emerging technologies, such as edge computing and AI-driven resource management, can open up new possibilities for efficient and intelligent task scheduling. This integration will leverage the advancements in these fields to enhance the scheduling process and overall cloud system performance. Conducting real-world deployments and large-scale experiments to assess the scalability and applicability of the proposed algorithm in practical cloud computing environments will be valuable for validating its effectiveness.

REFERENCES


A Modified Hybrid Algorithm Approach for Solving Harmonic Problems in Power Systems

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Chongqing 401420, China

Abstract—A fundamental problem with electrical systems' power quality is electrical harmonics. In order to limit harmonics and their effects on power systems, filters used in electric power systems must be designed with the consideration of power harmonics. The study's suggested approach differs from other hybrid strategies that have been previously published, and the processes that are expected mostly center on cutting down on computing complexity and time. The voltage and current waveforms of distribution networks have started to be significantly distorted over the past 20 years due to the increased use of power electronic equipment and non-linear loads. This paper provides a new hybrid approach for harmonic estimation. Harmonic estimation of these deformed waveforms is a nonlinear problem because sinusoidal waveforms contain nonlinear distortions. As a result, the corresponding combined technique splits the problem of harmonic estimation into two independent problems due to the slow convergence of nonlinear problems in the estimate of harmonic components. The algorithm used in this study first estimates amplitude and frequency using the fuzzy logic control (FLC) approach, a non-linear estimator. The objective function is then to minimize the error value for both the original signal and the estimated signal using the genetic algorithm, a non-linear estimator. The experiments show that the proposed method for determining harmonic estimation time is 36% better than comparable methods. As a result, the suggested technique offers a number of benefits, including very quick computation times, more precise evaluation of amplitude and phase values for all conditions, and complexity in the outcomes.

Keywords—Harmonic estimation; fuzzy PD controlled; harmonic components

I. INTRODUCTION

The periodic harmonic of current and voltage waves in an electrical network is undesirable because it increases the non-linear load and temporal unpredictability of the system [1]. The waveform and voltage contain a sinusoidal component with a different frequency known as harmonics due to this non-linear load or device [2]. The fundamental elements' integer repeating frequencies are known as harmonics. When harmonics are present, the power factor quality deteriorates, which causes issues with system protection, performance, and improvement [3]. In commercial and industrial systems, the increased usage of non-linear loads such as diodes and rectifier thyristors, arc furnaces, printers, uninterruptible power supply (UPS), etc., is one of the main contributors to harmonic generation. For an accurate computation, the primary frequency's range and phase are necessary [4]. The fast Fourier series transform (FFT) is the foundation of the conventional method of harmonic estimation [5]. However, the FFT-based approach has the issue of an undesired spectrum impact despite offering outstanding performance in noise [6]. One of the effective techniques for determining the number of harmonics in this situation is the Kalman filter [7]. However, tracking dynamic changes in the measurement signal with this Kalman filter fails [8]. Because of this, the amplitude and phase of voltage and current signal harmonics are evaluated using the least square algorithm [9].

Washing machines, video cameras, gas stoves, as well as industrial activities like cement kilns, underground trains and robots, are all controlled by fuzzy controllers [10]. A control strategy based on fuzzy logic is known as fuzzy control [11]. It should be emphasized that fuzzy logic and fuzzy control can both be summarily defined as calculations with words rather than numbers and phrases, respectively [12]. In operator-controlled systems, a fuzzy controller with empirical rules is very helpful [13]. The control strategy is saved in one or more natural languages in a rule-based controller [14]. Disturbances and disturbances that can be measured are corrected [15]. This approach needs a strong model. However, a fuzzy model might be helpful if a mathematical model is challenging to develop [16].

Because they don't need a precise mathematical model, fuzzy logic controllers have an advantage over traditional PI controllers [17]. They may be non-linear, able to handle ambiguous inputs, and more durable than a typical PI controller [18]. Fuzzy logic has been used by many writers to develop cutting-edge techniques for reducing harmonics and enhancing power quality [19]. Fuzzy logic in fuzzy logic box software should be understood as FL, or fuzzy logic in its fullest definition [20]. Fuzzy Logic's foundations provide a thorough explanation of FL's core concepts. Fuzzy control, which is based on approximate reasoning simulations and fuzzy mathematical theory, is a crucial subfield in intelligent control [21]. As a result, it can offer a wise way to manage intricate systems. Fuzzy systems typically include two components [22]. A segment is distinct and governed by regulations. One component is continuous and is known as a "fuzzy set". In recent years, the genetic algorithm has been combined with these two sets to be used for harmonic control [23]. Genetic algorithms are based on natural evolutionary processes like selected mutation. Since they were first developed approximately 25 years ago, genetic algorithms have shown to be a highly effective method for addressing a variety of AI-related issues [24]. In order to achieve optimal solutions/allocation and restrict the harmonic contents of DSs, it has been explored which planning models, optimization techniques, and renewable energy sources have used...
uncertainty models to solve BESS/PVDGs allocation difficulties. Studies on BESS/PVDGs allocation planning, however, have managed to reduce costs while falling short of the DSs standard harmonic level [25]. A hybrid fuzzy-genetic method is suggested to get over traditional fuzzy logic's constraints [26]. A fuzzy genetic algorithm (FGA) is a type of genetic algorithm that enhances various types of genetic algorithm behavior using fuzzy logic methods or fuzzy tools. An FGA can be thought of as a set of instructions, some of which may be constructed using fuzzy logic tools like fuzzy operators and connections to create genetic operators with various features [27]. They also examine the harmonic characteristics of various sources in typical operating scenarios, such as typical residential electrical load, electric vehicle charging scenario, frequency converter speed regulator, and renewable energy generation. Researchers have provided a comprehensive overview of common models of harmonic sources in modern power systems and provided insight into the circuit mechanisms, mathematical models, and operational processes of these sources [28].

The present study proposes a modified hybrid algorithm approach to solve harmonic problems in power systems. The main focus of the study is on electrical power quality and the issue of electrical harmonics in power systems. The suggested approach in this study differs from previously published hybrid strategies, with a focus on reducing computing complexity and time. The algorithm used in this study first estimates amplitude and frequency using the fuzzy logic control (FLC) approach, which is a non-linear estimator. Then, the objective function is to minimize the error value for both the original signal and the estimated signal using the genetic algorithm, another non-linear estimator. The combination of the two fuzzy logic algorithms and the genetic algorithm improves the simultaneous estimation of harmonic amplitude and phase components of the system. The study evaluates the proposed approach using two experiments. The first experiment focuses on analyzing power harmonics without combining and coordinating various harmonics, while the second experiment involves estimating real signals close to each other. The results of the experiments indicate that the proposed method outperforms other existing algorithms in terms of estimation performance and computation time. In summary, the main difference between the present study and previous works lies in the novel combination of fuzzy logic control and genetic algorithms to estimate harmonic components in power systems. The proposed method shows better convergence and estimation speed compared to other methods and can accurately estimate harmonic coefficients and phases.

However, based on the presented content, some potential challenges and limitations such as the accuracy of accurate estimation of harmonics in complex and noisy scenarios and the existence of noise and disturbances in real-world power systems may affect the accuracy and robustness of harmonic estimation are present in this study.

Given the benefits that the combined fuzzy genetic algorithm has suggested, FGA is utilised in this study to identify harmonics in the power system. In this instance, the harmonic fuzzy logic will be used first to explain the power system. The target function will then be created using the minimum genetic algorithm based on the signal from the fuzzy harmonic logic.

The following is a summary of the authors' contributions to this study.

- Combining two fuzzy logic algorithms and a genetic algorithm improved the simultaneous estimation of the system’s harmonic amplitude and phase components.
- Predictive variables of harmonic coefficients $A_i$ and $\text{THETA}_i$ phases for the estimated signal were determined based on the squared error cost function of the original and estimated signals.
- The Taki-Sugno-type fuzzy preset model and PD control form the basis of the proposed structure.

The remainder of the essay is structured as follows. The system model for harmonic estimation is looked at in Section II. Additionally, the combined FGA method that has been proposed for the study model is provided in this section. The simulation results for the suggested model are discussed in the Section III. The Section IV contains the conclusion and recommendations for future work.

### II. THE ISSUE OF HARMONIC ESTIMATION IN ELECTRICAL POWER SYSTEMS

According to the literature, current or voltage harmonics can damage devices, overheat power systems, or trip circuit breakers. Therefore, utilizing traditional or optimisation methods, researchers and designers have attempted to solve the aforementioned issues [29]. Due to the amplitude and fuzzy components, it has recently been discovered that harmonic estimation problems have a non-linear structure. Optimization-based techniques demonstrate an efficient and quick solution. Due to this achievement, the amplitude and phase values of high-level harmonic components in various waveforms are estimated and optimised in this study using FGA.

#### A. Harmonics in the Power System Mathematical Model

The following is an estimation of the power harmonic current or voltage's overall waveform:

$$y(t) = \sum_{m=1}^{M} A_m \sin(\omega_m t + \theta_m) + A_{dc} \exp(-\alpha_{dc} t) + \mu(t)$$

(1)

where $\omega_m = m2\pi f_0$, the angular frequency of component $m$, $f_0$, the main frequency, $\mu(t)$ additional white Gaussian noise $\text{AWGN}$, $A_{dc}$, $\exp(-\alpha_{dc} t)$, the anticipated damping amount, are all represented by the letters $M$. Harmonic phase values are $m\theta$, while $A_m$ and $\theta_m$ are unknown values [30]. The discrete-time signal can be represented as follows when the signal $y(t)$ is sampled with a sampling period of $Ts$:

$$y(n) = \sum_{m=1}^{M} A_m \sin(\omega_m nTs + \theta_m) + A_{dc} \exp(-\alpha_{dc} nTs) + \mu(n)$$

(2)

Then, in equation (2), the so-called damping Taylor series is used, and the straightforward equation is as follows:

$$y(n) = \sum_{m=1}^{M} A_m \sin(\omega_m nTs + \theta_m) + A_{dc} - A_{dc} \alpha_{dc} nTs + \mu(n)$$

(3)
Finally, a universal and straightforward formula for estimating the phase and amplitude of all harmonics is discovered. The linearity of the proposed sinusoidal model is also improved, which allows for the use of effective and reliable optimisation techniques to solve the issue. The sine and cosine functions can be used to rewrite the general waveform shown in equation (3) as follows:

$$y(n) = \sum_{m=1}^{M} [A_m \sin(\omega_m n Ts) \cos(\omega_m n Ts) \sin(\theta_m) + A_{dc} - A_{ac} a_{dc} n Ts + \mu(n)]$$

Later, this signal is transformed as follows into a parametric form:

$$y(n) = H(n)\theta(n)$$

Which \( H(n) \) is displayed as follows:

$$H(n) = [\sin(\omega_1 n Ts) \cos(\omega_1 n Ts) \sin(\omega_m n Ts) \cos(\omega_m n Ts)]^{-1} - kTs]^T$$

Following is how the general vector of unknown parameters is expressed:

$$\theta(n) = [\theta_{1n}\theta_{2n} \ldots \theta_{(2m-1)n}\theta_{2mn}\theta_{(2m-1)n}\theta_{(2m-1)n}]^T$$

$$\theta = [A_1 \cos(\theta_1)A_1 \sin(\theta_1) \ldots A_m \cos(\theta_1)A_1 \sin(\theta_1) \ldots A_{dc} A_{ac} a_{dc}]^T$$

The harmonic estimation problem's objective function \( J \), which is used to optimise the unknown parameters, can therefore be represented as follows:

$$J = \min(\sum_{n=1}^{N} e_n^2(n)) = \min(\sum_{n=1}^{N} (y_n - y_{n\text{est}})) = \text{MSE}(y_n - y_{n\text{est}})$$

where the estimated output harmonic signal \( y_{n\text{est}} \) is specified, and \( y_n \) is the actual harmonic signal detected in the grid or electrical system.

B. Design of a Harmonic Estimator using the Combined FGA Algorithm

This section goes into great length about the harmonic estimation problem in the power system and the integrated approach of the FGA algorithm. The harmonic estimator based on the FGA method is used to solve the synchronisation problem in the following steps:

a) Fuzzy Controller: The effectiveness of fuzzy logic controllers in enhancing transient and steady-state performance is well established. The fuzzy logic controller's function is particularly beneficial because a precise mathematical model is unnecessary [31]. Fig. 1 depicts the fuzzy logic control system's diagram.

This figure's four main functional sections are basic knowledge, fuzzification, inference mechanisms, and defuzzification. A database and a legal base make up the knowledge base. The database contains information for appropriate fuzzification operations, inference, and defuzzification methods, including input and output membership functions. The language rules that link the fuzzy input variables to the desired control actions make up the rule base. A clear input signal is converted into fuzzy signals through "fuzzification," which can be recognised by the degree of membership in fuzzy sets. Instead of using numerical variables, fuzzy logic makes use of linguistic variables. Fuzzy "if-then" principles and fuzzy reasoning are used in this notion. These findings are helpful in various domains, including robotics, autonomous control, data classification, decision analysis, and expert systems. It uses a non-linear mapping from the input space to the output space. The inference mechanism assesses the control rules. Two types of inference mechanisms predominate [32]: Mamdani fuzzy inference system for starters, and a system for fuzzy inference by Sogno. The fundamental distinction between these two approaches is how they produce the outcomes of fuzzy rules. Sogno employs linear functions of the input variables, whereas Mamdani uses fuzzy sets for the rule. It should be mentioned that the existing research employed the Sogno technique. The inference system carries out tasks like decomposition and aggregation. If the laws are prevalent, they are reduced to a straightforward rule format. Finally, de-fuzzing transforms murky output signals into audible ones [33].

Fig. 1. The fuzzy logic control function block diagram.
b) **Genetic Algorithm**: General-purpose techniques for learning and optimization are genetic algorithms. A particular kind of evolutionary algorithm called a genetic algorithm uses biological processes like mutation and inheritance. In place of regression-based forecasting methods, GAs is a good choice. Binary strings are the computational language of GAs [34]. Finding a useful answer is only a matter of looking for a particular binary string. Frequently superior to chance-based prediction methods are genetic algorithms. A genetic algorithm, often known as GA, is a programming method that leverages the notion of genetic evolution to solve problems. The input is the problem that needs to be addressed, and the solutions are coded using a structure known as the fitness function [35]. It assesses each potential answer, the majority of which are chosen at random. A helpful technique in pattern recognition, feature selection, picture interpretation, and machine learning are the genetic algorithm. Genetic algorithms replicate the genetic progression of biological things. These algorithms address issues that were motivated by the natural evolutionary process. In other words, they create a population of beings, much like nature does, and by acting on this population, they create an ideal population or being [36].

A problem must be transformed into the unique form needed by genetic algorithms for them to be able to solve it. The required solution to the issue should be developed in this process so that it may be represented as a chromosome [37]. Fig. 2 lists the stages for implementing a genetic algorithm.

a) **Proposed Hybrid FGA Algorithm**: Fig. 3 displays the harmonic estimation algorithm based on the suggested FGA algorithm. The block diagram view of the suggested design for the problem of signal identification at various times by extending the Fourier series of a signal and calculating the amplitude and phase at various frequencies is really depicted in the accompanying figure. We combined the genetic algorithm with a Taki-Sugno-type fuzzy control to improve the detection accuracy. Additionally, it serves as a PD controller in the control part of the closed-loop system to estimate instantaneous and high-speed harmonics. As a result, it greatly enhances the ability of the genetic algorithm.

---

![Fig. 2. Shows how the genetic algorithm is applied.](image1)

![Fig. 3. Show the proposed design's block diagram.](image2)
Additionally, the fuzzy block built for this research's membership functions is depicted in Fig. 4, and the performance of the rules is depicted in Fig. 5, which is based on the subsequent rules. Additionally, Fig. 6 provides the input-output characteristic of the fuzzy block.

![Diagram showing fuzzy block and membership functions](image1)

**Fig. 4.** Membership functions of error input and error change as well as a general representation of Sogno-type fuzzy blocks.

![Diagram showing PDFLC fuzzy function's output performance](image2)

**Fig. 5.** Display of the PDFLC fuzzy function's output performance for input errors and error modifications.
III. EVALUATION AND SIMULATION

As was already said, numerous methods for estimating harmonic signals have been documented in the literature to offer effective estimation techniques. The main benefits of these new technologies are anticipated to be faster calculations and greater accuracy. To estimate amplitude/phase and compare the performance of the harmonic fit based on the FGA algorithm with that suggested by other meta-heuristic approaches, the analysis of numerous functions in both non-noisy and noisy situations is described in this section. Two harmonic estimating procedures are considered in this work. The second test takes into account harmonic situations with inter- and sub-harmonics, while the first test investigates the estimate method for simple power harmonics.

In these studies, a test signal that is often utilized in the literature is used. Typical industrial loads this chosen signal can depict include power electronic devices, electronic explosions, and non-linear systems [12]. Previous studies [7, 38] have shown the signal-to-noise ratio (SNR) to be 10 dB, 20 dB, and 40 dB. In addition, the CS, threshold, MCN and correction rate of the FGA algorithm are selected as 20, CS / 2 × D, 1000 and 0.1, respectively. To show the extent of the algorithm used, 50 times with different values were given for all the investigated cases in MATLAB 2016. Also, the entire given domain is valued in units (p.u.). The simulations were performed on a PC with a Windows 7 operating system, Intel 2.67 GHz processor and 4 GB RAM.

A. Experiment 1 (Analysis of Power Harmonics without Combination and Coordination)

Before starting the analysis, a basic test signal covering the behavioral effects of industrial loads and several operating systems according to [9] is below:

\[
y(t) = 1.5 \sin(2\pi f_1 t + 80^\circ) + 0.5 \sin(2\pi f_3 t + 60^\circ) + 0.2 \sin(2\pi f_5 t + 45^\circ) + 0.15 \sin(2\pi f_7 t + 36^\circ) 0.1 \sin(2\pi f_1 t + 30^\circ) + 0.5 \exp(-5t) + \gamma n\]

MATLAB software is used to sample the test signal in continuous time at a sampling frequency of 20 kHz. Following that, the average values of the findings obtained are shown together with the process of harmonic estimation by the GA algorithm using the sampled test signal. The results are estimated for a revised signal, which is then compared to the initial test signal. Additionally, the suggested algorithm is used in the relevant noisy and non-vibrating scenarios. Fig. 7 and 8, respectively, display the outcomes of the error signal in terms of time and the estimated signal.
Fig. 8. Estimated signal display.

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The estimation results for the first test that were achieved utilising the suggested solution were carried out in a noise-free environment. Additionally, the results from the literature reported in Table I are contrasted with the estimated values obtained for the first experiment. In Table I, parameters like A, P, and E represent amplitude, phase, and error. It is evident from Table I that the suggested estimation procedure outperforms the findings in the literature. Furthermore, it is noteworthy that the harmonic estimator based on the evolutionary algorithm has a faster processing time than other comparable techniques.

### B. Experiment 2

In this experiment, the estimated real signals have been calculated quite close to each other, which are shown in Fig. 9 and 10, the size and error of the signal, and Fig. 11 shows the reduction of the sum of squares of the error by the genetic algorithm for this experiment.

\[
Y(t) = 23[0.65 \times \sin(3 \times 50 \times 2\pi t) + 0.45 \times \sin(7 \times 50 \times 2\pi t) + 0.1 \times \sin(15.5 \times 50 \times 2\pi t) + 0.01 \times \sin(4000 \times 2\pi t)]
\] (11)
Fig. 9. Simultaneous display of original and estimated signal.

Fig. 10. Display of the error signal in terms of time.

Fig. 11. Showing the reduction of the sum of squared errors by the genetic algorithm.
When the second experiment's results are taken into account, it is evident that, much like the first experiment's findings, the estimated values of the power harmonics, including sub- and two inter-harmonics, are completely consistent with the actual values. Additionally, Table II displays the estimation outcomes for the second trial. The proposed technique in this study offers improved outcomes in terms of estimating performance and computation time, as can be observed from the findings that have been presented. Fig. 12 shows the harmonics calculated from two real and estimated

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signals, and Table III compares the main harmonics for both real and estimated signals. The simulation results show that the proposed method has a good convergence and estimation speed. Also, this method can estimate harmonic coefficients and harmonic phases higher than other methods.

The proposed model's projected performance is pretty impressive and nearly identical to the actual algorithm. The voltage range for each set of data is determined, and the estimated outcomes are contrasted. Clearly, the proposed approach offers the best accuracy and quickest convergence of any. As a result, the conclusions reached using real-time data from a real-time system are supported by the outcomes. Tests have been conducted using 15 cycles in a 55 Hz system, which is a 250 ms window at a sampling period of 0.5 ms, in compliance with I.E.C 61000-4-31 to determine power quality characteristics [3].

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</table>

IV. CONCLUSION

The article presents a modified hybrid algorithm approach for solving harmonic problems in power systems. The primary focus of the study is to address the issue of electrical harmonics, which is a significant problem affecting power quality in electrical systems. The proposed approach combines fuzzy logic control (FLC) and genetic algorithms (GA) to estimate harmonic components in nonlinear power systems. The hybrid algorithm approach is developed to address the slow convergence of nonlinear problems in harmonic estimation. The innovation of this thesis is the combination of a fuzzy logic algorithm and a genetic algorithm. The basis of proposed structure is based on a predetermined Takai-Sugeno type phase model and PD control to improve signal estimation for high-frequency harmonics and then determine the predictive variables of harmonic coefficients Ai and THETAi phases for the estimated signal and based on the function. The squared error cost is the original and estimated signal. The proposed method first uses fuzzy logic control to estimate amplitude and frequency, which is a non-linear estimator. Then, a genetic algorithm is employed to minimize the error value for both the original signal and the estimated signal, offering a more precise evaluation of amplitude and phase values for all conditions.

The experiments conducted in the study show that the suggested hybrid approach for harmonic estimation performs 36% better in terms of computation time compared to other comparable methods. The results demonstrate that the proposed algorithm provides faster calculations and higher accuracy for harmonic estimation. In the first experiment, the proposed method is applied to estimate power harmonics, and it outperforms existing algorithms in terms of accuracy and processing time. In the second experiment, the method successfully estimates power harmonics as well as sub-harmonics and inter-harmonics, again showing improved performance compared to other methods. Overall, the proposed hybrid approach combining fuzzy logic control and genetic algorithms offers a promising solution to the harmonic estimation problem in power systems. It provides faster computation times, more accurate results, and the ability to estimate harmonics beyond the fundamental frequency, making it a valuable contribution to power system harmonic analysis and mitigation.

In line with the limitations of this study, the following are suggested as future works.

- Using other optimization algorithms instead of GA algorithm
- Use Mamdani type fuzzy logic
- Estimation of phase and amplitude components separately from each other.
- Simulation for time-invariant power signal.

ACKNOWLEDGMENT

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Inverted Ant Colony Optimization Algorithm for Data Replication in Cloud Computing

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Abstract—Data replication is crucial in enhancing data availability and reducing access latency in cloud computing. This paper presents a dynamic duplicate management method for cloud storage systems based on the Inverted Ant Colony Optimization (IACO) algorithm and a fuzzy logic system. The proposed approach optimizes data replication decisions focusing on energy consumption, response time, and cost. Extensive simulations demonstrate that the IACO-based method outperforms existing techniques, achieving a remarkable 25% reduction in energy consumption, a significant 15% improvement in response time, and a substantial 20% cost reduction. By addressing the research gap concerning integrating IACO and fuzzy logic for data replication, our work contributes to advancing cloud computing solutions for large datasets. The proposed method offers a viable and efficient approach to improve resource utilization and system performance, benefiting various scientific fields.

Keywords—Cloud computing; data replication; cloud data centers; reliability; energy efficiency; inverted ant colony optimization algorithm

I. INTRODUCTION

Cloud computing provides significant opportunities for progress in the information technology industry. As an emerging form of a distributed network, cloud computing entails providing hardware and software resources across a range of businesses and organizations [1]. Users are only charged for the services offered by the cloud provider. Several types of cloud computing services are available, including Infrastructure as a Service (IaaS), Platform as a Service (PaaS), and Software as a Service (SaaS) [2]. SaaS shares software applications between customers in the cloud through Internet-based service providers. Users do not need to install software on their personal computers. The PaaS model involves sharing a platform with consumers responsible for configuring, deploying, testing, and developing the application. Consumers do not maintain servers, hardware, and storage, although they may control applications. To run applications, the cloud service providers supply a pre-installed operating system. IaaS is a model for deploying consumer applications and systems on shared infrastructure [3].

Four major cloud computing deployment models exist: community, public, private, and hybrid clouds. An organization can use private clouds inside or outside of their offices. It is located within the virtualized data center's firewall. Users cannot control the services provided by the cloud service providers, as they provide the infrastructure. An organization that provides cloud services to the public or large industries operates a public cloud. Cloud service providers control the services and provide the infrastructure within the public cloud. Community clouds provide cloud services offered by multiple organizations to assist individuals with similar concerns. In hybrid clouds, numerous cloud models are combined, utilizing proprietary or standardized technology that permits the portability of data and applications [4, 5].

In the realm of cloud computing, the integration and utilization of various cutting-edge concepts are instrumental in driving innovation, efficiency, and intelligence. The Internet of Things (IoT) plays a pivotal role by connecting a myriad of devices and sensors, generating vast volumes of data [6, 7]. This data deluge, known as big data, necessitates advanced data processing and storage capabilities that cloud computing can readily provide [8, 9]. Artificial Intelligence (AI) and Machine Learning (ML) are the cornerstones of intelligent cloud applications, enabling systems to learn from data patterns, make informed decisions, and automate processes, thereby enhancing user experiences and optimizing operations [10-14]. Feature and channel selection are vital in cloud computing scenarios dealing with large and diverse data sources [15]. By identifying the most relevant features and data channels, cloud-based systems can reduce computational overhead, improve accuracy, and streamline data processing [16, 17]. Furthermore, meta-heuristic algorithms in the cloud facilitate efficient optimization of resource allocation, task scheduling, and load balancing, leading to optimal resource utilization and performance [18-21]. Association rule mining enables the discovery of valuable patterns and correlations within data, empowering businesses to make data-driven decisions and tailor services to individual preferences [22]. Additionally, game theory principles come into play in cloud computing, where cloud service providers and users engage in strategic interactions. Through game theory, fair and efficient pricing models can be devised, and resource allocation strategies can be optimized to ensure mutually beneficial outcomes [23].

Providing reliable services in conjunction with a high degree of availability and data performance constitute essential criteria. These requirements are ensured through the use of the replication concept. Data replication involves replicating the duplicate files at different locations to improve reliability, availability, and performance [24]. Data replication plays a vital role in data-intensive environments in which records must be synchronized between multiple sites located at diverse locations. The availability of the data is increased through replication. Requests may be fulfilled by other sites that host replicas of the requested file if the site responsible for hosting these files cannot serve them. Furthermore, replication
enhances the overall performance of the system by allowing the user's requests to be satisfied at the nearest site that also holds replicas of the data file requested. In this manner, the system's response and access times are improved. As a final point, replication also enhances the reliability of the system. Despite the corruption or failure of any replica of a data file, user requests can be processed and handled by other non-corrupted servers that hold the required replica. In most cases, replication strategies cannot be adapted to cloud environments due to their intention to improve system performance by not considering replication costs. The creation of multiple replicas within cloud computing environments is not economically practical as it may lead to wasteful resource utilization and a reduction in service providers' profits. Thus, data replication approaches applied to cloud-based platforms must also meet the following objectives:

- Maintaining reliable Quality of Service (QoS) through the observance of Service Level Agreements (SLAs), which constitute legal contracts between cloud providers and their tenants, i.e., customers. Typically, an SLA consists of one or more tenant Service Level Objectives (SLOs), i.e., requirements.
- Based on the pay-as-you-go pricing model, the provider dynamically adjusts the resources it rents to its tenants.

Data replication, a critical technique for distributing data across multiple locations, remains challenging in cloud computing. The problem arises from the need to strike a balance between enhancing data availability and minimizing access latency for users. Replicating data to multiple locations can improve system availability and ensure users can access data from a nearby location. However, determining the optimal number and locations for data replicas is a complex optimization problem. This paper aims to address the data replication challenge in cloud computing by proposing a novel method of dynamic duplicate management. Our research draws inspiration from the Ant Colony Optimization (ACO) algorithm, which has shown promising results in solving combinatorial optimization problems. In particular, we introduce the Inverted ACO (IACO) algorithm, tailored to tackle the data replication issue in cloud storage systems. By leveraging the IACO algorithm and a fuzzy logic system, we seek to optimize data replication decisions regarding energy consumption, cost, and response time. The objective is to improve the overall performance of cloud data centers while maintaining data availability for end-users.

The rest of this paper is structured as follows: Section II presents the related work in data replication techniques for cloud computing. Section III provides a detailed description of the proposed IACO-based dynamic duplicate management approach. Section IV presents the results and analysis of extensive simulations, comparing the proposed method with existing techniques. Finally, Section V concludes the paper and outlines potential future research directions.

II. RELATED WORK

In order to solve the data replication problem, Ebadi and Jafari Navimipour [25] proposed a hybrid metaheuristic algorithm. To achieve high-quality solutions, it combines the global search capabilities of the PSO algorithm with the local search capabilities of the Tabu search algorithm. Various test cases are used to demonstrate the efficiency of the method compared to PSO, TS, and ACO algorithms. Based on the results obtained, it appears that the method is more efficient in terms of energy consumption and cost.

By combining multiple objective optimizations and evaluating the cost distance using the knapsack problem, Salem, et al. [26] presented the shortest routes and the lowest costs in the cloud based on replication and data center placement. By using the Artificial Bee Colony (ABC) algorithm, the shortest route and lower-cost problems have been solved, identifying the best replication placement based on the distances of minimizing the costs and achieving the shortest routes that were achieved when the Knapsack algorithm was employed to deal with these issues. The proposed system can be optimized using the MOABC algorithm to achieve high efficiency and low costs. By using the Knapsack approach and optimizing replica distribution based on distances and data transmission rates, MOABC is capable of finding a suitable location to place data replicas. Based on measurements, MOABC is more effective and efficient than compared algorithms in regards to determining the best placement of replications.

Rambabu and Govardhan [27] proposed a new data replication approach derived from data mining principles. Data replication is accomplished by identifying widely used data patterns within the massive database of a node. A novel hybrid algorithm selects the best thresholds based on an optimization-assisted frequent pattern mining strategy. Greywolves Updated Exploration and Exploitation with Sealion Behavior (GUEES) is a hybrid algorithm that combines the concepts of Grey Wolf Optimization (GWO) and Sealion Optimization Model (SLnO). During the mining process, two criteria are defined: (i) Prioritization and (ii) Cost. The prioritization process can be divided into two categories: queueing both high and low-priority data, and the storage demand determines the cost. High-priority queues are optimized with GUEES. As a final step, a comparative validation is performed to evaluate the efficiency of the proposed model.

Mansouri, et al. [24] propose a hierarchical data replication strategy (HDRS) for dynamic replication. HDRS involves creating replicas capable of being dynamically increased by growing exponentially or depleting exponentially, placing replicas based on access loads and labeling techniques, and replacing replicas depending on future file values. CloudSim is used to evaluate different dynamic data replication methods. In comparison with other algorithms, HDRS has been demonstrated to reduce response time and bandwidth consumption. A popular file is replicated to the best site by the HDRS. By balancing the load of the sites, this method avoids unnecessary replications and reduces access latency.
Khelifa, et al. [28] suggested a periodic and dynamic data replication approach for cloud-based federated systems. The objective of this method is to ensure financial success for cloud service providers by meeting the demands of their users regarding availability and response time. In order to detect replicas, a periodic review of the users’ tasks is conducted by employing the spectral clustering algorithm to determine relationships among remote data pertaining to SLA violations. Adapting correlations of this type can lead to a reduction in both the volume of data to be transferred and its transfer time. In the next step, a fuzzy inference system is used to select replicas from the owned or leased resources of other providers based on four main parameters. Additionally, replica numbers are adjusted when SLAs are met. In accordance with the results obtained, the proposed strategy reduces the number of SLA violations while maintaining the providers’ monetary profits.

Despite the various existing techniques, there remains a research gap concerning the integration of the IACO algorithm and fuzzy logic for dynamic duplicate management in cloud storage systems. Previous studies have primarily focused on standard ACO algorithms or hybrid metaheuristics. Our research aims to bridge this gap by introducing the IACO algorithm, a novel adaptation with potential advantages in data replication. By combining IACO with a fuzzy logic system, our approach targets energy consumption, cost reduction, and response time improvement offering a comprehensive and efficient solution for data replication in cloud computing environments.

III. PROPOSED METHOD

Replication of data in a cloud environment reduces network delays, bandwidth usage, and costs. The replication of data to several sites improves data accessibility by allowing users to access data at their convenience. Cloud systems, however, face the challenge of determining the appropriate number and location of replicas. By increasing the number of replicas, accessibility is enhanced and access delay is reduced, but replication costs increase. In this regard, it is crucial to present a method that reduces data replication costs and balances accessibility and cost. In this study, a new approach based on the IACO algorithm and fuzzy logic system is proposed to improve the management of replication versions in cloud storage systems.

A. Problem Modeling and Formulation

Our model considers a cloud computing network with M data centers, each of which has its own storage media, memory, and processing power. si refers to the storage capacity of data center i (1 ≤ i ≤ M) measured in a simple data unit (like blocks). The connection between two data centers, i and j (if any), has a positive integer number C (i, j) and another positive integer number E (i, j) related to it. Energy and communication costs are assigned to transfer a data unit among the data centers (communication costs can be expressed as delay, number of nodes, and so on). If there is no direct communication link between two data centers, the cost is calculated by the sum of all communication costs along the selected path between the data centers. Moreover, the energy is obtained by the sum of all communication energies along the selection path from data center i to j. The required assumptions for the proposed method are presented in the following:

\[ C(i, j) = C(j, i) \text{ and } E(i, j) = E(j, i) \]

There are N data d1, d2, . . ., dN. The size of each data is shown by \( d_k (1 \leq k \leq N) \) and is measured in a simple data unit. \( r_k \) and \( w_k \) are the total numbers of the read and written values during time T (the k index is used for data, and i and j indexes are used for data centers). \( r_k \) includes some of the reads of local generation on i and some intermittent reads on their path to the objective via i.

Each data \( d_k \) has a main version on the network that is not re-transferable and removable. The main version's data center maintains \( d_k \) showing by cdbk. Each main data center cdbk considers the maintenance of information related to the replication marked by RSk for each \( d_k \) (a part of the data center in which each data k is repeated). RS = {RS1, RS2, . . ., RSN} is used to determine the replication plans for all data. Moreover, the main data center, cdbk, and the nearest data center dcdk store \( d_k \). dcdk is a data center that leads to the minimum cost and energy for read requests from i for \( d_k \). Consider that if \( d_{dcdk} = \text{cddb}_k \), the main data center is close to the data center, and if \( d_{dcdk} \neq \text{cddb}_k \), the data center i maintains a replica of \( d_k \). Hence, each request is served locally, as read for data (if there is a local replica of the data) or transferred to \( d_{dcdk} \), which can answer the request. To ease, we assume an update of \( d_k \) is performed by transmission of the updated replicas of the data to the main data center cdbk that is propagated to each data center, which has a replica of \( d_k \) (there are also other plans).

Based on the proposed system model in Fig. 1, the 1, 2, 3, ..., n locations are different, distributed, and connected by firmware, including different methods to access the replicas and their management. The replicas of file x are stored in locations 1, 2, 3, ..., n. Assuming that user 1 wants to access file x if distance and cost are important for the user, locations 1 and 3 are close to user 1, and their access cost is low. Different copies of file x make the file accessible if the information is lost in different locations. It results in more reliability and accessibility rate.
B. Proposed Hybrid Algorithm

Different stages of the proposed method are explained in this section.

1) Determining the type of replicated file: The popular files are recognized, and the replication is performed. Based on it in this stage, it is investigated which file should be replicated. The proposed method performs this stage by recognizing the popular files. This stage uses the following three substages to select the files that should be replicated.

- Substage 1: Calculating the number of accesses to the files: Clustering is used to calculate the number of file accesses in this stage so that there are some cloud data centers in each cluster. A cluster head is considered in each cluster that its bandwidth is wider than other cloud data centers and can connect all the in-cluster cloud data centers. Since cloud data centers have some constraints like bandwidth limitation, the large size of the file, and the number of interactions among the cloud data centers, a method must reduce the interactions. Hence, first, the requests in each cluster are checked in each cluster to relocate the required files of that local cluster via the high bandwidth connection if possible. Transferring the high-volume files among the clusters can be avoided using clustering. This substage is explained as an algorithm in the following:
  - Calculating the count of the requests to file $f_i$ in cluster C ($Num_{Req}(f_i)$)
  - Decreasing the sort of all files based on $Num_{Req}(f_i)$
  - Storing the sorted files in a list in each cluster
  - Determining the count of all the required files ($TF^n_c$) of the data center c in period n (this stage is performed for all clusters).

- Substage 2: Calculating the amount of the file popularity: After determining the number of requests to the file in the previous stage, the file's popularity is calculated in this stage. To calculate the popularity of file $f_i$ in period n, if $Num_{Req}(f_i) > 0$, equation (1) is used, otherwise, equation (2) is used. $Num_{Req}(f_i)$ is the count of the requests for file $f_i$ from cluster c in period n. $size(f_i)$ is the file's size, and $TotalReq^n$ is the total number of requests in period n. $p$ and $q$ are constants, and their values directly affect the response time as one of the necessities for the proposed method's Quality of Service (QoS).

\[
FP^*_c(f_i) = \frac{10^{p-2} + (Num_{Req}(f_i) \times p)}{size(f_i)} \times \frac{Num_{Req}(f_i)}{TotalReq^n} (1)
\]
\[
FP^{n-1}(f_i) = q (2)
\]

- Substage 3: This substage involves determining the candidate files for replication. The count of the files, which should be replicated based on the two previous stages, is determined in this stage. Moreover, the files whose average popularity is calculated in the previous stages are sorted descendingly. In equation (3) $RepNum_i$ shows the number of files that must be copied and the proposed method selects the $RepNum_i$ files from begin of the sorted list to copy. $\rho$ is a random number between 0 and 1. 30% of the repeatedly requested files are copied in the proposed method, meaning the $x$ value is set to 0.7.

\[
RepNum_i = TF^*_c \times (1 - \rho) (3)
\]

2) Selecting the most suitable datacenter to save new replicas: One of the most important stages in the replication management process is finding the most suitable cloud data center to store the replicas. We use two indexes to find the most appropriate cloud data center to store the replicas. These indexes select the candidate data centers to store these replicas.

Fig. 1. Proposed model system.
The IACO algorithm is an improvement of the ACO algorithm in that the pheromone of the path affects the ants’ movement inversely, and it has a hatred effect on the ants instead of attract effect. At first, a graph is considered, including nodes and the edges connecting them. Each node in the proposed method is a cloud data center, including different data that a cloud’s data may be repeated in another cloud. A user’s requested data and the objective data to answer the user’s request are determined. Then, the beginning node is selected randomly, and all the nodes start their movement from the beginning node. The ants traverse the graph, and each ant selects the next ant using equation (5), including pheromone, heuristic, and the condition to check its existence. The heuristic matrix in the proposed inverse ants’ algorithm is calculated and valued using fuzzy logic. This fuzzy system’s stages are presented in the following.

At first, because there is no pheromone on the path, it is not required to have a primary traverse to create the pheromone on the graph, the first ant has a traverse, and the pheromone on the traversed path is updated using equation (6) after its satisfaction. The next ants may not select the traversed path by the first ant because the pheromone’s smell is not desired for the next ants. The second ant chooses the graph’s best node (the node with the highest probability) to traverse and moves to the node. If all the paths are traversed, and there are some ants, the remaining ants copy the traversed paths by the first, second, and other ants.

A tolerance is selected for the graph’s nodes because the objective data may be in the beginning node. If there is no tolerance, all ants select the first node, which leads to the inefficiency of the proposed method to provide workload balance among the nodes. The considered tolerance value is one so that if the utilized data is from a cloud data center, this center’s tolerance is zero, and its data cannot be used unless the tolerances of other cloud centers with the objective data are also zero. If the node’s tolerance is zero, we can pass over the node but cannot use its data. A complete graph tolerance is considered only for the first node. But it is considered for all the cloud centers in a non-complete graph.

If the objective data is not in the beginning node and the graph is complete, the selected node to traverse by the first ant is the node with the appropriate data. But if the graph is non-complete, a specific policy is used to choose the suitable data. If the objective data is in the beginning node, the first ant should use that cloud center’s data, and the second ant should select the next node to traverse. Hence, the beginning node is compared with the traversed node by the second ant to find the objective data. If we are on the beginning node, and the objective data is not in its adjacent nodes, the nodes are selected randomly, and their adjacent nodes are also checked. Finally, the nodes, including the objective data, are compared based on the heuristic values, and the better node, including the objective data, is selected. These heuristic values are generated using the proposed Fuzzy system.

Each data center has some parameters called QoS parameters. QoS parameters are non-operational features of the system, which can be provided by the service providers, evaluated by their performance, and monitored by the users. QoS parameters can be divided into two categories: positive and negative. Increasing values of the positive QoS parameters can be helpful for the users, like accessibility and reliability. On the other hand, decreasing values of the negative QoS parameters can be useful for users, like cost and execution time. Each cloud data center is equal to an ant in the inverse ants’ algorithm, and the cloud data centers are evaluated based

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Definition</th>
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<tbody>
<tr>
<td>$Num_{Req}(f_i)$</td>
<td>The number of requests for the file $f_i$</td>
</tr>
<tr>
<td>$HNR_i$</td>
<td>The data center with the highest number of requests for the $f_i$ file</td>
</tr>
<tr>
<td>$FS^n$</td>
<td>Cloud data center with free storage space</td>
</tr>
<tr>
<td>$TS$</td>
<td>Total storage space</td>
</tr>
<tr>
<td>$Dis^n$</td>
<td>The distance between cloud data center $n$ and other data centers</td>
</tr>
<tr>
<td>$HD_{is}$</td>
<td>The cloud data center with the highest value for the $Dis$ parameter</td>
</tr>
</tbody>
</table>

Index 1: The amount of the data center fitting: $Fit^n_i$ can be calculated for each file $f_i$ for a data center like $n$ using equation (4). This equation explanation is presented in Table I.

$$Fit^n_i = w_1 \times \frac{Num_{Req}(f_i)}{HNR_i} \times w_2 \times \frac{FS^n}{TS} + w_3 \times (1 - \frac{Dis^n}{HD_{is}})$$

(4)

In equation (4), the coefficients $w_1$, $w_2$, and $w_3$ are weighted and between 0 and 1. The important note about these coefficients is that they can be valued based on the user's opinion, and their sum should be 1. Hence, the importance and priority of the user can be modeled based on this index. For example, if a user's priority is the data centers with the most repetition, $\frac{Num_{Req}(f_i)}{HNR_i}$, meaning that the $w_1$ coefficient should be increased, and $w_2$ and $w_3$ should be decreased. Generally, the amount of the tendency and the priority of a user can be modeled using the above equation.

Index 2: The amount of the total cost of the cloud data center: Generally, in addition to the amount of fitting the cloud data center, another index should be used to calculate the total cost of the data center and select the most suitable data center to store the replicas based on the movement pattern of the inverse ants. In this stage, QoS parameters determine the total cost of the data center. These parameters are used in Fuzzy logic (a fitting function for each data center) to evaluate the candidate centers for the replica’s storage. After calculating the two indexes to select the most suitable data center to store the replicas, IACO algorithm and fuzzy logic are used to traverse the cloud data center and evaluate each center based on these two indexes.

The IACO algorithm is an improvement of the ACO algorithm in that the pheromone of the path affects the ants’ movement inversely, and it has a hatred effect on the ants instead of attract effect. At first, a graph is considered, including nodes and the edges connecting them. Each node in the proposed method is a cloud data center, including different data that a cloud’s data may be repeated in another cloud. A user’s requested data and the objective data to answer the user’s request are determined. Then, the beginning node is selected randomly, and all the nodes start their movement from the beginning node. The ants traverse the graph, and each ant selects the next ant using equation (5), including pheromone, heuristic, and the condition to check its existence. The heuristic matrix in the proposed inverse ants’ algorithm is calculated and valued using fuzzy logic. This fuzzy system's stages are presented in the following.
on the QoS parameters. These parameters are the inputs to the proposed fuzzy model, and an output value is generated after defuzzification. This fuzzy generated value is used as a heuristic function value, and the ants traverse the network based on these values. A list of the positive and negative parameters with their units is defined in the following.

Value: It shows the amount of service reliability (in percent).

Accessibility: It shows the probability of service accessibility (in percent).

Cost: It is the cost that the service requester should pay to the service provider to use the services (in dollars).

Response time: It is the execution time between the reception of the service and the response to the service (in milliseconds).

Consumption energy: We used the energy model introduced in [29]. Energy consumption is being increased by the number of replicas which is relied by the given work resources so that for decreasing the level in energy amount.

Equation (5) is used to select the cloud data center for graph traversal. The first ant, which starts the traverse, chooses the optimal data, and the optimal data is in the optimal cloud data center with more probability for selection. After selecting the optimal data and traversing the path related to the data center, the path's pheromone is updated, and its smell makes hatred for the following ants. Hence, the path traversed by the previous ant may be ignored by the following ants. It continues by the next ant while all the paths to reach the objective data are traversed. Then if there is any ant, the traversed paths by the previous ants are repeated in order.

\[ p_{k,j}^i(t) = \frac{[\tau_{k,j}^i(t)]^{[\eta_{k,j}^i(t)]^\beta}[D]^\gamma}{\sum_{l=1}^{D}[\tau_{k,l}^i(t)]^{[\eta_{k,l}^i(t)]^\beta}[D]^\gamma} \]  

In this equation, \( t \) is the current replica, \( \tau_{k,j}(t) \) is the existing pheromone in an edge (\( \epsilon_{k,j} \)), and D is the condition of checking data existence. If the objective data is on the cloud data center, \( D=1 \), else \( D=0 \). \( \eta_{k,j} \) refers to the heuristic value generated by the proposed fuzzy logic system. \( \alpha, \beta, \) and \( \gamma \) are the effects on the pheromone, the heuristic, and the condition of checking the existing data that all three values are considered 1 in this method. Equation (6) shows the pheromone update in each repetition that \( \tau(t) \) is the value of the previous pheromone and is considered one before traversing a node by the ants. The \( k \) denotes the number of traversed nodes.

\[ \tau(t + 1) = \tau(t) \times \frac{1}{k} \]  

The fuzzy system output is used for the quantification of the heuristic matrix. Selecting the precise fuzzy rules helps the algorithm to traverse the cloud data centers precisely.

Stage 1: Defining the variables and linguistic words (input information fuzzification): This method begins by fuzzification of the information related to the QoS of each cloud data center. This information includes parameters of positive and negative service quality. Indeed, these values are applied to this stage of the Fuzzy system as inputs. Information on the QoS is constructed based on Fuzzy Inference System (FIS). The Fuzzy function is determined using input parameters. FIS generates a replica value (RV).

Stage 2: Defining the fuzzy rules: Fuzzy rules based on different input parameters states are written at this stage. FIS generates an output value based on input different states by the FIS. The fuzzy rules are written using nested If-then. The Mamdani inference algorithm is one of the known algorithms for fuzzy inference. These systems can be used extensively in decision support systems because of their visual and interpretative nature. They also have a high power of expression and can be implemented in both forms of Multiple Input Multiple Output (MIMO) and Multiple Input Single Output (MISO). Mamdani controller is used for ease. Input and output parameters are expressed using linguistic parameters. The triangular membership function is used for input variables' linguistic values of Very-Low (VL), Low (L), Medium (M), High (H), and Very-High (VH). Output variables have five triangular membership functions for the VL to VH values, and the fuzzy system uses 25 rules shown in Fig. 2. The FIS input parameters' membership functions are presented in Fig. 3 and 4. As mentioned in the previous section, some QoS parameters are positive, and some are negative. In other words, the minimizer and the maximizer parameters can be expressed as positive and negative. Because the lower values for the positive parameters (close to zero) result in the optimal output value, and higher values for the negative parameters (close to one) lead to the optimal output value. Hence, we should find a trade-off value between these values to select the best answer that is performed using the Fuzzy rules set. Thus, we define PQ and NQ as positive and negative parameters, respectively, that are obtained by the sum of the normalized values of the quality parameters.

<table>
<thead>
<tr>
<th>Fuzzy rules</th>
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<tbody>
<tr>
<td>1. If (PQ is VL) and (NQ is VH) then (Fitness is VH)</td>
</tr>
<tr>
<td>2. If (PQ is VL) and (NQ is H) then (Fitness is H)</td>
</tr>
<tr>
<td>3. If (PQ is VL) and (NQ is M) then (Fitness is M)</td>
</tr>
<tr>
<td>4. If (PQ is VL) and (NQ is L) then (Fitness is L)</td>
</tr>
<tr>
<td>5. If (PQ is VL) and (NQ is Q) then (Fitness is Q)</td>
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Fig. 2. The proposed fuzzy rules.
Stage 3: Defuzzification and analysis of the outputs: Defuzzification of the fuzzy values is performed in this stage after producing multiple fuzzy values according to the fuzzy rules and the input values. In other words, an interval in the fitting function is defined for and mapped to each of the fuzzy outputs in this stage. Fig. 5 illustrates the membership function of the fuzzy system's output variables.

IV. EXPERIMENTAL RESULTS

Based on the experimental setup, results demonstrate a clear improvement strategy for data elimination and data ambiguity. The significance of the system is demonstrated by conducting experiments using the MATLAB simulator.

First experiment: In this experiment we used the [11] data for evaluating the performance of the proposed method. Based on this experiment result, with increasing replication numbers, replication costs also increase. The value of the main and super data centers and their respective probabilities of file availability will be maintained in accordance with the probability of file availability and number of replicas in each data center. In each data center, the value of the super and main data centers and the corresponding probability of file availability will be determined based on the probability of file availability and the number of replicas. According to the Fig. 6-8, when the proportion parameters are set to the same value, the proposed method performs better than the alternative methods. Also, the proposed method demonstrates the medium level of cost efficiency that has been incorporated into the file system. Experimental results demonstrate that our proposed algorithm is capable of dynamically adapting to the preferences of the user.
Fig. 6. Mean file unavailability.

Fig. 7. Energy consumption level.

Fig. 8. Number of tasks given.
Second experiment: In this experiment, we used the [30] data's and evaluated the proposed method in terms of average response time. Fig. 9 and 10 compare the average response time of the replication strategies for the uniform distribution and Zipf distribution. Our proposed method exhibited notably improved average response times compared to other strategies, indicating its effectiveness in reducing access latency for users.

The experimental results highlight the advantages of our proposed method over existing strategies for data replication in cloud computing. Our approach showcases superior performance and effectiveness by considering cost efficiency, response time, and dynamic adaptability. The comparisons with relevant literature reveal that our model provides a better solution to the data replication problem in cloud computing environments. The MATLAB-based simulations validate the feasibility and practicality of our proposed method. It is essential to acknowledge some limitations of our work. While our experiments demonstrate significant improvements, further investigations could explore the scalability of the proposed method for more extensive datasets and complex cloud environments. Additionally, incorporating real-world case studies and validating the results in a practical cloud computing environment would enhance the applicability of our approach.

V. CONCLUSION

Data replication constitutes one of the most important aspects of cloud computing, since it provides fast access to data, high availability, and data reliability. By maintaining all replicas available, the replicas can increase the probability of a system task being completed successfully if the requests and replicas are distributed reasonably. However, placing replicas
in a highly scalable, virtualized, and large data center poses a greater challenge. To provide load balancing for cloud storage, decrease the response time of applications, and achieve cost-effective availability, this paper proposed a novel dynamic duplicate management in cloud storage systems based on the IACO algorithm and fuzzy logic system. According to the simulation results, our proposed method exhibits significant advantages over existing replication techniques. It accomplishes a remarkable 15% decrease in energy consumption, a 12% cost reduction, and an impressive 20% enhancement in response time. While our research has demonstrated substantial improvements, there remain opportunities for future investigations. Scalability testing under more extensive datasets and complex cloud environments will provide insights into the model's performance under varying scales. Additionally, incorporating dynamic workload management strategies and exploring hybrid optimization techniques could further enhance the algorithm's adaptability and convergence speed.

REFERENCES


Improved Cat Swarm Optimization Algorithm for Load Balancing in the Cloud Computing Environment

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Abstract—Recently, cloud computing has gained recognition as a powerful tool for providing clients with flexible platforms, software services, and cost-effective infrastructures. Cloud computing is a form of distributed computing that allows users to store and process data in a virtual environment instead of a physical server. This is beneficial because it allows businesses to quickly scale up or down their computing capacity, reducing the need to invest in expensive hardware. As cloud tasks continue to grow exponentially and the usage of cloud services increases, scheduling these tasks across diverse virtual machines poses a challenging NP-hard optimization problem with substantial requirements, including optimal resource utilization levels, a short execution time, and a reasonable implementation cost. The issue has consequently been addressed using a variety of meta-heuristic algorithms. In this paper, we propose a new load-balancing approach using the Cat Swarm Optimization (CSO) algorithm in order to distribute the load among the various servers within a data center. Statistical analyses indicate that our algorithm is superior to previous research with regard to energy consumption, makespan, and time required up to 30%, 35%, and 40%, respectively.

Keywords—Cloud computing; resource utilization; load balancing; optimization

I. INTRODUCTION

Cloud computing provides on-demand, convenient network access to a range of customizable computing resources, including services, applications, storage, and servers, instantly available by service providers without much effort on management's part [1]. A cloud computing model can be deployed in four ways: public, private, community, and hybrid, representing its underlying structure. As the name suggests, a public cloud is a cloud environment publicly accessible by many cloud customers without any restrictions placed upon them by the cloud service provider [2]. There are two scenarios in which a private cloud can be deployed. The cloud service provider may manage and maintain a private cloud exclusively within an organization. In the second scenario, the private cloud is deployed exclusively within a single organization and is managed and controlled by the organization. Regardless of the scenario, a private cloud is used privately by a single organization within a controlled environment. Community clouds are cloud environments that restrict access to organizations and cloud customers who share the same objectives [3]. Members of the community may jointly maintain and manage this type of cloud, or a separate cloud service provider may provide its services to accommodate this type of cloud. Hybrid clouds are cloud environments that combine multiple cloud deployment models. Public and private clouds may be combined in this manner. Organizations and customers often opt for this cloud environment due to security concerns, whereas less sensitive data may be stored in a public cloud environment [4].

Cloud computing is also comprised of three key services, Software as a Service (SaaS), Platform as a Service (PaaS), and Infrastructure as a Service (IaaS). IaaS is concerned with providing hardware as a service. IaaS provides users with the ability to rent servers, storage, and networks, while PaaS provides developers with an on-demand platform to develop, test, and host applications. Lastly, SaaS provides users access to applications hosted on the cloud, usually through a web browser. SaaS is the most popular service, offering users the most convenience and scalability. It eliminates the need to manage hardware and software and allows users to access software on demand [5]. Cloud computing brings together several key attributes, including measured service, rapid elasticity, resource pooling, broad network access, and on-demand self-service. In general, cloud services can be quantified through various metrics, such as bandwidth, data, and time. The cost of cloud computing services is usually determined by the resources used, and the cost savings can be significant compared to traditional IT solutions [6]. The level of elasticity in cloud computing refers to the ability of the system to adapt to changes in workloads through automated provisioning and de-provisioning and the availability of resources based on that. In a resource pooling model, virtual and physical resources are pooled to serve multiple consumers through a multi-tenant model, with resources allocated and reallocated dynamically according to consumer demand. Broad network access involves locating resources on the network and accessing them via different computing platforms and devices, including tablets, smartphones, laptops, and various types of computers. The concept of on-demand self-service involves allowing users to access data and resources in the cloud whenever they need to without requiring human assistance [7].

In order to improve the performance of a cloud, the workload must be balanced among the servers. Due to heavy processing tasks or numerous processes, some servers may be under high utilization. At the same time, some other servers may be idle. This situation would result in a decrease in the efficiency of the network, while the idle servers would waste the network's energy. It is also necessary to wait for the busy server to complete its in-process tasks before sending requests to an overloaded server [8]. The reason for this delay is that a server can only handle a limited number of requests.
simultaneously. This results in a decrease in service availability, which causes dissatisfaction. The availability of the service refers to the user's ability to access the desired service at any time. In this regard, load balancing can be used to optimize energy consumption and enhance performance. By transferring virtual machines from an overloaded host to an idle host without interruption, the workload is balanced between the servers, resulting in improved network efficiency [9].

In the domain of cloud load balancing, the significance of IoT, machine learning, artificial intelligence, meta-heuristic algorithms, association rule mining, deep learning, and feature selection lies in their collective potential to revolutionize resource allocation, performance optimization, and adaptability in cloud computing environments. The integration of IoT devices enables real-time data collection and monitoring of cloud resources, facilitating dynamic load distribution based on real-time demands. Machine learning and artificial intelligence techniques empower load balancing systems to autonomously learn from historical data and network patterns, enhancing their ability to predict resource demands and make informed load distribution decisions [10-12]. Meta-heuristic algorithms play a pivotal role in fine-tuning load balancing parameters and optimizing resource utilization for complex and dynamic cloud workloads [13]. Deep learning, with its capability to automatically extract intricate patterns from cloud data, enhances load balancing efficiency and enables the identification of performance bottlenecks and anomalies [14-16]. Association rule mining is of utmost significance as it allows for the identification of intricate relationships and dependencies among various cloud resources, enabling more efficient resource allocation and load distribution strategies to optimize the overall performance and utilization of the cloud infrastructure [17]. Feature selection techniques help identify the most relevant and discriminative cloud attributes, enabling streamlined load balancing processes and reduced computational overhead [18]. By harnessing the potential of these cutting-edge technologies, cloud load balancing achieves enhanced scalability, energy efficiency, and response times, ensuring optimal utilization of cloud resources, seamless user experiences, and the ability to adapt to changing cloud conditions effectively.

The proposed approach in this research aims to address the challenges of load balancing in cloud computing environments, which play a crucial role in optimizing resource utilization and enhancing overall system performance. Cloud computing offers a versatile and scalable platform, but efficient load distribution among servers is essential to avoid underutilization and overutilization issues. The motivations for this study stem from the need to overcome inefficiencies caused by imbalanced workloads, which can lead to reduced service availability and energy wastage. The potential benefits of the proposed approach are multi-fold. Firstly, by utilizing the Cat Swarm Optimization (CSO) algorithm, inspired by the foraging behavior of cats, the approach introduces a novel nature-inspired metaheuristic to the field of cloud load balancing. CSO’s adaptive and efficient load distribution capabilities enable optimized resource allocation, resulting in reduced energy consumption, improved makespan, and enhanced task execution times. Secondly, by conducting a comprehensive comparative analysis with Q-Learning and MPSO, well-established load balancing methods, the research demonstrates the superiority of the proposed CSO-based approach, highlighting its potential to outperform existing techniques.

The rest of the paper is organized in the following manner. Section II reviews the related works in the field of cloud load balancing. Section III details the proposed method. Experimental results are reported in Section IV. Finally, Section V concludes the paper and suggests some hints for upcoming research.

II. RELATED WORK

Muteeh, et al. [19] presented a multi-resource load-balancing mechanism using the Ant Colony Optimization (ACO) algorithm for cloud computing environments to ensure a well-load-balanced system and reduce cost and makespan time. Experimental results from benchmark workflows are used to validate the algorithm. The results indicate that the execution time and cost are reduced while the available resources are efficiently utilized by maintaining a balanced workload among them. Based on the mimicking behavior of flocks of birds, Mishra and Majhi [20] have developed a load-balancing method in which tasks are viewed as birds and VMs as food patches for the birds. The proposed approach was evaluated using a dataset (GoCJ) logged by Google in 2018 as part of cloudlet execution. This method is intended to improve the performance of the system by decreasing response time and maintaining overall balance. A comparison is made between the proposed technique and previously developed techniques. The proposed approach demonstrates an improvement in resource utilization and reduces the time required for tasks to be completed.

Mapetu, et al. [21] proposed a low-cost and low-complexity version of the PSO algorithm to schedule and balance cloud computing tasks. An objective function is defined that calculates the maximum difference in completion time between heterogeneous virtual machines. Constraints related to updating and optimization affect the objective function. Then, a particle position update strategy is devised with respect to load balancing. The experimental results indicate that the proposed algorithm performs better in task scheduling and load balancing than existing meta-heuristic and heuristic algorithms. Kruekaew and Kimpan [22] proposed an independent approach to cloud computing task scheduling that utilizes the Q-learning and Artificial Bee Colony (ABC) algorithms. ABC algorithm is improved through the use of reinforcement learning techniques. The proposed method aims to distribute workload among virtual machines proportional to resource utilization, cost, and makespan. The experimental results showed that the proposed algorithm was superior to the competitive algorithms with respect to throughput, imbalance degree, cost, and makespan. Sefati, et al. [23] employed Grey Wolf Optimization (GWO) algorithm to maintain proper load balancing based on resource reliability capability. The GWO algorithm attempts to identify unemployed or busy nodes first and, after discovering these nodes, determines the threshold and fitness function of each node. The experiments conducted in CloudSim have demonstrated that this approach has lower response times and
costs than other methods. The algorithm is capable of efficiently optimizing the load balancing process which requires less time and fewer resources.

Due to the complexity and multitude of criteria involved in dynamic task allocation to heterogeneous resources, finding an optimal solution for every scheduling problem in real-time can be extremely challenging. In such cases, researchers often turn to meta-heuristic techniques, which aim to find near-optimal solutions within a reasonable timeframe while exhibiting outstanding performance for the task at hand. In this context, Mirmohseni, et al. [24] have proposed a hybrid method called Fuzzy Particle Swarm Optimization Genetic Algorithm (FPSO-GA). Their approach combines two powerful optimization techniques, namely fuzzy particle swarm optimization and genetic algorithms. By integrating these methods, the researchers seek to leverage the unique strengths of each technique and achieve better performance in dynamic task allocation.

Haris and Zubair [25] have introduced a dynamic load balancing algorithm called Mantaray modified multi-objective Harris hawk optimization (MMHHO). This algorithm utilizes a hybrid optimization approach, combining the benefits of two optimization algorithms: Harris Hawk Optimization (HHO) and Manta Ray Forging Optimization (MRFO). The aim of MMHHO is to address dynamic load balancing in cloud computing environments. The hybridization process involves updating the search space of HHO using MRFO, taking into account factors such as cost, response time, and resource utilization. By considering multiple objectives, the proposed algorithm seeks to enhance the performance of the system in terms of Virtual Machines (VMs) throughput, load balancing among VMs, and maintaining a balance among task priorities by adjusting the waiting time of tasks. The simulation results demonstrated that the MMHHO load balancing scheme outperforms other algorithms, highlighting its efficiency and effectiveness in achieving dynamic load balancing in cloud computing environments.

The reviewed approaches have demonstrated improvements in resource utilization, reduced response times, and overall load balancing efficiency. However, despite their advancements, there are certain gaps in these previous studies that present opportunities for further research. One of the gaps is the need for a more comprehensive comparison and evaluation of these load balancing techniques in diverse and dynamic cloud environments. While some studies have compared their proposed approach with existing techniques, there is a lack of a unified benchmark dataset or standardized evaluation criteria to assess their performance consistently across different cloud scenarios. Additionally, there is limited exploration of hybrid approaches that combine the strengths of multiple algorithms to achieve optimal load balancing results. Furthermore, existing studies predominantly focus on performance metrics such as response time, makespan, and resource utilization, but there is less emphasis on energy efficiency and cost-effectiveness. Considering the growing importance of sustainability and cost optimization in cloud computing, future research should explore load balancing methods that prioritize energy-efficient resource allocation and cost reduction.

III. PROPOSED METHOD

This section explains the Cat Swarm Optimization (CSP) algorithm. Then, the proposed load-balancing algorithm is described in detail. Finally, a detailed insight into the proposed algorithm is provided.

A. Cat Swarm Optimization Algorithm

Observations of cat behavior in the real world have driven the design of a novel swarm-based optimization algorithm. CSO mimics many of the behavior patterns of cats, such as their ability to move quickly, their ability to focus on a single target, and their tendency to explore their environment. CSO has been found to be effective in solving complex optimization problems. The CSO explains cats' behavior in two ways: by seeking and tracing. A local search is performed by the former, and a global search is conducted by the latter. In the seeking mode, the individuals are perturbed multiple times so that each can approach the local optimum. In the tracing mode, each cat traces the target at a certain speed and updates its position to match better the swarm's optimal position [26, 27]. Generally, the CSO algorithm consists of the following steps:

- Create N cats with a specified number of dimensions.
- Assign random velocities to each cat.
- Randomly place cats in seeking and tracing modes.
- Compute the fitness of all cats and keep track of the non-dominated cats.
- For each cat, if it is in seeking mode, execute seeking mode actions, else execute tracing mode actions and relocate it to its new location.
- If the termination condition is not satisfied, move to step 3, else stop.

B. Problem Statement

Cloud computing offers cloud services to cloud users, in which tasks are performed within a cloud environment. A cloud typically consists of a large number of Data Centers (DCs) called Physical Machines (PMs), each of which is equipped with a specific computing resource to fulfill the consumer's task. Each cloud consumer includes a variety of tasks to execute on virtual machines. The load balancing process assigns diverse users' tasks to virtual machines and constantly checks loads on virtual machines in the cloud computing environment. According to the time taken to complete each task, the VM load fluctuates as each task takes a different amount of time. Load balancing allocates tasks from an overloaded virtual machine to an underloaded one. A load balancing system in a cloud environment is illustrated in Fig. 1. Cloud services receive a wide range of requests from users, which requires setting up a dynamic environment for executing tasks. Load balancing strategies are applied when the load balancer receives requests from users. The load balancer selects the necessary virtual machines in response to a request. Load balancers can be configured to distribute consumer requests among multiple virtual machines. The Data Center Controller (DCC) is responsible for task management. The load balancer assigns the task to the appropriate virtual machine based on a load-balancing algorithm.
C. CSO-based Load Balancing Method

Load balancing is an excellent technique for reducing request waiting times and optimizing resource utilization. In this way, virtual machines can be prevented from becoming overloaded or underloaded, and load distribution can be uniformized among them. In the proposed method, cats represent solutions and attempt to achieve the target through iteration over different cats, striving to achieve the optimal solution. The load balancing problem is expressed according to the following terminology. The primary objective of the proposed method is to achieve optimal load balancing in the cloud. The considered cloud computing model comprises c number of PMs composed of m number of virtual machines. The suggested technique is to improve response time, resource utilization, and cost. Accordingly, Eq. 1 describes the objective function based on the above conditions, where w denotes the weight factor, and e represents the exponential function associated with each parameter. F1, F2, and F3 refer to objective functions for response time, resource utilization, and cost.

\[ F = w_1e^{F_1} + w_2e^{F_2} + w_3e^{F_3} \]  

(1)

1) Response time: It is defined as the time interval between a task arriving in the system and being completed. In terms of load balancing, response time refers to the time required to allocate VMs to lower load conditions in response to a user request. There is an inverse relationship between this factor and the efficiency of the system. The optimal response time is determined by the shortest makespan time as follows.

\[ F_1 = \text{Fin}_t - \text{Arr}_t + \text{TDelay} \]  

(2)

In Eq. 2, \text{Arr}_t indicates a user demand's arrival time, \text{Fin}_t denotes a user demand's ending time, and TDelay signifies transmission delay.

2) Resource utilization: It refers to the utilization level of virtual machines. The goal is to achieve maximum resource utilization and minimize the makespan time. There is a reverse linear relationship between these two terms. Eq. 3 is used to calculate the average utilization of all virtual machines, where N indicates the total number of virtual machines and \( T_{VM_j} \) represents the time required to complete all tasks by a jth virtual machine.

\[ F_2 = \frac{\sum_{j=1}^{N} T_{VM_j}}{MS \times N} \]  

(3)

3) Cost: This indicator is calculated by Eq. 4. The cost varies in accordance with the tasks. The cost is determined by the complexity of the task and the resources needed. It is important to consider the cost when evaluating the effectiveness of a system. Optimizing the cost is essential for the success of any system.

\[ F_3 = \sum_{i=1}^{N} (VM_{time} \times VM_{cost}) \]  

(4)

In Eq. 4, VM\text{cost} denotes the cost involved in running a virtual machine to perform a task, VM\text{time} denotes the duration of the execution of the task, and N represents the number of tasks involved in the workflow.

Suppose a solution comprises five PMs, each containing two VMs. Initially, incoming tasks are distributed randomly among virtual machines. All virtual machines should be balanced in this regard. Overloaded PMs should be migrated to underloaded ones. Fig. 2 illustrates the initial solution. The CSO algorithm is used to optimize the solutions. The CSO algorithm involves two primary steps, seeking and tracing.

![Initial solution](image-url)
4) Seeking mode: Cats are placed in this mode following the fitness calculation. In this process, the cats look around and adjust their position. In this way, cats search for the best opportunity to capture prey. The CSO algorithm employs four variables: self-position consideration (SP), counts of different dimensions (CD), seeking collection of selected dimensions (SR), and seeking memory pool (MP). The steps for seeking mode are described below:

- Cats duplicate their position seeking Memory Pool (MP).
- Using Eq. 5, the number of dimensions changed (DC) can be calculated for these duplicated positions.
  \[ S = (1 \pm SR \times R) \times Sn \]  
  \[ (5) \]
  In Eq. 5, R is a random number ranging from 0 to 1, S represents the latest position, and Sn indicates the recent position.

- All cat positions are compared using fitness calculation values. All cats are set to 1 if they are identical. In any other case, we use the identifying probability Eq. 6, where FC \_min refers to the minimum significance of fitness calculation, FC \_max stands for the maximum assessment of fitness calculation, FC \_i is the fitness calculation value of each cat, and P \_v represents the probability of the latest cat.
  \[ P_v = \frac{|FC_i - FC_j|}{|FC_{max} - FC_{min}|}, \text{ for } 0 < i < j \]  
  \[ (6) \]

5) Tracing mode: In tracing mode, cats are recreated based on their positions in order to update their velocity in dimension d. Eq. 7 described the updating of velocity in tracing mode, where Pn,d is the current location of the cat, Pbest,d is the cat's location with the best result, k is a random number between 0 and 1, and r represents a constant.
  \[ Vn,d = Vn,d + k \times r(P\text{best},d - Pn,d) \]  
  \[ (7) \]

IV. EXPERIMENTAL RESULTS

This section analyzes the efficiency of the suggested method in light of the simulation results. In order to conduct the cloud computing experiment, the CloudSim simulator was installed on a system configured with an Intel Core i5 CPU, 4 GB RAM, and Windows 8 operating system. Performance has been evaluated in terms of makespan, energy consumption, idle time of tasks, response time of tasks, and number of tasks migrated. Table I presents the simulation environment of the experiments.

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IV. EXPERIMENTAL RESULTS

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**Fig. 3 and 4 show the energy consumption during load balancing for three different algorithms: Q-Learning, MPSO, and CSO. According to these figures, CSO consumes the least amount of energy during load balancing in comparison to other algorithms. Makespan assesses the performance in another parameter. It measures users' response time for specific tasks in cloud computing, and the response time determines the quality of service. Consequently, the service provider is able to provide the client with a high-quality of service. Fig. 5 and Fig. 6 present a comparison of algorithms according to their makespan under various task and virtual machine configurations. According to these figures, CSO performed significantly better than both Q-Learning and MPSO. Therefore, CSO was able to achieve stability and load balance in a highly efficient manner.**
A longer waiting time can be expected if only one task can be executed simultaneously while other tasks are waiting in the ready queue. By increasing the processing power of VMs and allocating more VMs, the waiting time for all tasks will be reduced. Fig. 7 and 8 illustrate the idle time of tasks and the processing time of VMs, respectively. Based on Fig. 7, CSO exhibits a lower idle time than Q-Learning and MPSO for all tasks. Fig. 8 illustrates the processing power of VMs. The processing power has been increased from 200 MIPS to 2500 MIPS, resulting in a reduction in processing time from 9400 ms to 100 ms. Based on the results of the analysis, it can be concluded that CSO balances the load within the cloud network in an effective and efficient manner.
The presented experimental results demonstrate the superior performance of the proposed CSO-based approach in terms of energy consumption, makespan, and task idle time. CSO's ability to optimize load distribution and achieve efficient resource utilization makes it a promising solution for load balancing in cloud computing environments. The findings support the significance of the proposed method in enhancing the quality of service and overall cloud network performance. The comprehensive analysis of the simulation results reaffirms the potential benefits of adopting CSO for load balancing in cloud computing environments. However, further investigations and real-world validations may be necessary to ascertain its scalability and effectiveness in diverse and dynamic cloud infrastructures.

V. CONCLUSION

In recent years, cloud computing has experienced an upsurge of interest, primarily due to its usefulness and relevance to contemporary technological trends. Globally, it provides better computational services to its clients by being highly customizable. With the increase in demand for higher processing power, large amounts of data will put a considerable strain on the cloud computing environment. As a result, the need for an efficient scheduling algorithm for cloud computing tasks has become increasingly important. Cloud tasks continue to grow exponentially, and the number of cloud users rapidly increases, making scheduling and balancing these tasks among heterogeneous virtual machines a challenging NP-hard problem with significant constraints, including high resource utilization, quick scheduling, and low implementation cost. This paper proposed a novel load-balancing mechanism using the CSO algorithm, which distributes load among systems in a data center. As demonstrated by simulation results, our algorithm exhibits superior performance to previous research in terms of energy consumption, makespan, and time required by approximately 30%, 35%, and 40%, respectively. While the proposed CSO-based load balancing approach demonstrates promising results, it is essential to acknowledge its limitations. The scalability and adaptability of CSO in highly dynamic and large-scale cloud environments require further investigation. Additionally, the algorithm's performance might be sensitive to parameter settings, emphasizing the importance of proper tuning for optimal results.

REFERENCES


Automatic Fraud Detection in e-Commerce Transactions using Deep Reinforcement Learning and Artificial Neural Networks

Yuanyuan Tang*
School of Economics and Management, Tianjin Vocational Institute, Tianjin 300410, China

Abstract—Fraud is a serious issue that has plagued e-commerce for many years, and despite significant efforts to combat it, current fraud detection strategies only catch a small portion of fraudulent transactions. This results in substantial financial losses, with billions of dollars being lost each year. Given the expected surge in the volume of online transactions in the upcoming years, there is a critical need for improved fraud detection strategies. To tackle this problem, the article proposes a deep reinforcement learning approach for the automatic detection of fraudulent e-commerce transactions. The architecture's policy is built on the implementation of artificial neural networks (ANNs). The classification problem is viewed as a step-by-step decision-making procedure. The implementation of the model involves the use of the artificial bee colony (ABC) algorithm to acquire initial weight values. After that, in each step, the agent obtains a sample and performs a classification, with the environment providing a reward for each classification action. To encourage the model to concentrate on detecting fraudulent transactions precisely, the reward for identifying the minority class is higher than that for the majority class. With the aid of a supportive learning setting and a specific reward system, the agent ultimately determines the best approach to achieve its objectives. The performance of the suggested model is assessed utilizing a publicly available dataset contributed by the Machine Learning group at the Université Libre de Bruxelles. The experimental outcomes, determined using recognized evaluation measures, indicate that the model has attained a high level of accuracy. As a result, the suggested model is considered appropriate for identifying deceitful transactions in e-commerce.

Keywords—Fraud detection; reinforcement learning; artificial neural network; artificial bee colony; imbalanced classification

I. INTRODUCTION

Fraud refers to intentional dishonesty or deception by an individual or group of people with the aim of obtaining financial benefits. As a result of the increase in online transactions such as shopping and insurance claims, there is a new level of fraudulent activity that individuals and businesses must be wary of. Reports indicate that the increase in fraudulent activities in e-commerce transactions during the first quarter of 2018 was significantly higher than the growth rate of e-commerce transactions in 2016. The e-Commerce Fraud Index revealed that in 2017, account takeover fraud in online department stores rose to 0.23%, a significant increase from 0.06% in 2016 and accounting for over 10% of fraud losses. While credit card fraud only makes up 0.1% of all card transactions, fraudulent transactions involving large sums of money have led to significant financial losses. However, even with the surge in credit card transactions in recent times, the ratio of fraud cases has not changed [1].

Numerous institutions and industries have invested significant resources in developing efficient techniques to combat fraudulent activities by leveraging advanced technologies, especially machine learning [2, 3]. As a result of these endeavors, a plethora of solutions have been developed that can differentiate between valid credit card transactions and those that are fraudulent without human intervention. Irrespective of the method employed, there are certain shared issues that can hinder its effectiveness. The prevalent issue is the imbalanced distribution of training data, a feature of past transactions. This creates various challenges, such as overfitting, and results in low-accuracy classifiers used. Imbalanced classification is a common challenge in machine learning, where one class has significantly more data compared to other classes. Due to this disproportion, recognizing minority specimens becomes difficult because of their infrequency and randomness, resulting in a poorer outcome [4-6]. The problem of imbalanced classification in machine learning can be tackled using one of two methods, namely the data level and algorithmic level [7, 8]. At the data level, balancing the class distribution can be achieved through various methods like oversampling and under-sampling. Oversampling generates new samples by linear interpolation between adjacent minority samples, while under-sampling removes some majority examples utilizing the nearest neighbor algorithm [9]. However, these methods risk overfitting and loss of significant information [10, 11]. At the algorithmic level, techniques such as cost-sensitive learning, decision threshold adjustment, and ensemble learning can be applied to increase the significance of the minority class. Ensemble learning trains multiple sub-classifiers and combines them to improve performance, while cost-sensitive learning assigns varying costs to incorrect classification, with the misclassification of the minority class given a higher cost. Threshold adjustment techniques change the decision-making threshold during testing. Some proposed deep models for imbalanced data classification utilize innovative loss functions that consider errors in classifying minority and majority groups [12]. These techniques can retain the specific attributes of a dataset that has imbalanced classes or clusters while also safeguarding the margins between samples from different classes or clusters [5, 13, 14].
While traditional machine learning approaches use a rigid feature extraction strategy that often leads to poor generalization ability, long processing time, and low accuracy, deep learning algorithms have emerged as promising alternatives for classification tasks [8]. With their layered structure, deep learning algorithms can capture complex patterns and relationships within data, making them highly adept at learning high-level features. One such popular algorithm is the Multilayer Perceptron (MLP), a universal approximation that excels at handling nonlinear problems. Originally developed to solve the XOR problem, MLP has since been successfully applied to various combinatorial optimization problems, including pattern recognition, classification, image processing, and linear and nonlinear optimization. MLP processes input signals by passing them through interconnected layers of processing nodes, each of which receives a set of input values, sums them, and then applies an activation function to determine its output, mimicking the behavior of a human neuron. MLP, through its layers of interconnected nodes, is capable of acquiring intricate associations between the input and output variables. Additionally, the lack of interconnections between nodes in the same layer helps reduce computational complexity, making processing more efficient [15].

The paper presents a novel approach to detecting fraud in e-commerce transactions by combining the ABC algorithm [16] and reinforcement learning. The model being proposed views the classification task as a process of making sequential decisions, which is analogous to playing a game of guessing. In this game, the agent is presented with a training instance and has to classify it using a policy. The agent's performance is evaluated based on the rewards received for correct and incorrect classifications, with a higher reward assigned to correctly identifying the minority class. The main goal of the agent is to achieve the highest possible cumulative reward by correctly identifying the largest number of samples. This technique has the potential to overcome the challenges of imbalanced classification discussed earlier in the paper. The suggested model frames the classification issue as a series of sequential decisions, enabling the agent to acquire knowledge and adjust its strategy according to responses from the surroundings. The use of reinforcement learning further enhances the agent's ability to explore and exploit the search space efficiently. This article makes three significant contributions. 1) The approach taken to address the challenge of imbalanced classification is to view the prediction problem as a sequential decision-making process and a reinforcement learning-based algorithm is introduced. 2) An encoding method based on the ABC algorithm was devised to obtain the best initial value instead of assigning weights randomly, and 3) the proposed model's performance was evaluated through experiments, and a comparison was made between this model and other methods that use random weight initialization, which encounter challenges in dealing with imbalanced classification.

The format of the article is organized in the following manner. In the second section, the paper presents an overview of existing research in the relevant area. In the third section, a succinct explanation of the ABC algorithm and its functioning is presented. In the fourth section, the model proposed in the study is introduced, and in the fifth section, the evaluation criteria, dataset, and analysis of the results are presented. The concluding section of the paper discusses the study's findings and draws conclusions, as well as outlining potential avenues for future research.

II. RELATED WORK

Initially, fraud detection was associated with the utilization of Information Retrieval or the Rule-based method. The details of each transaction were scrutinized by hand, and decisions were made concerning fraudulent or reliable based on strict criteria. Each transaction causes the development of a feature vector [2]. A feature vector is composed of various attributes and parameters such as Transaction identifier, Transaction amount, Cardholder data, Site of the transaction, and Time of transaction. This vector is assigned points according to the scoring criteria determined by human investigators. As an example, if a transaction has taken place on another continent within an hour, then the fraud score would be 0.95 [17]. This system depends on the addition of more and more regulations in order to remain one step ahead of scammers that seek to exploit and bypass current rules.

Big Data Analytics, through the use of machine learning, is more wide-reaching, economical, precise, and automated [18, 19]. This powerful combination of Big Data and machine learning has opened up new possibilities for businesses and organizations across various industries, enabling them to extract valuable insights, make data-driven decisions, and optimize their operations like never before. One of the remarkable applications of Big Data Analytics is the construction of sophisticated models capable of forecasting, classifying, or estimating the authenticity of transactions, especially when it comes to identifying fraudulent activities [20]. Such models, empowered by the wealth of data collected from digital datasets containing numerous transactions, have revolutionized fraud detection methodologies. By leveraging machine learning algorithms and tapping into large and diverse datasets, these data-informed models have achieved impressive results in accurately differentiating between genuine and fraudulent transactions. The abundance of data provides these models with a rich source of information, enabling them to discern complex patterns and anomalies that would be challenging for traditional rule-based systems to detect. The training process of these models involves exposing them to vast quantities of labeled data, where each transaction is tagged as either authentic or fraudulent. Through iterative learning and optimization, the models adapt and fine-tune their parameters, continuously improving their performance and generalization abilities. Different data-driven models have emerged in the realm of Big Data Analytics, each employing a variety of methods and algorithms tailored to specific use cases and data characteristics. These models can include traditional machine learning approaches like SVM, Random Forest, and Gradient Boosting, as well as state-of-the-art deep learning techniques like Neural Networks and Transformer-based models. The choice of model and method depends on the nature of the data, the complexity of the problem, and the desired level of interpretability [21, 22].
The commonly utilized approach involves using machine learning techniques to create a model based on the data. This data-driven model is generally more versatile and dependable, enabling it to achieve high accuracy levels, often reaching up to 87% or even higher, depending on the specific problem and dataset. The success of machine learning models can be attributed to their ability to uncover complex patterns and relationships within the data that might not be easily discernible through traditional rule-based systems. Among the well-known machine learning algorithms, several have proven to be highly effective in various domains. K-means is a popular clustering algorithm used for grouping data points into clusters based on their similarity, making it useful for segmentation and pattern discovery tasks. Regression Analysis, on the other hand, is widely employed for predicting numerical values and understanding the relationships between variables. SVM have been extensively utilized in both classification and regression tasks. SVM is particularly suitable for binary classification problems, and with appropriate kernel functions, it can handle complex decision boundaries efficiently. Similarly, Random Forest and Decision Trees are powerful ensemble methods that can be applied to both classification and regression tasks, providing robustness and reducing overfitting [23]. Recent advancements in deep learning have introduced breakthroughs in the field of fraud prevention and detection. Recurrent Neural Networks (RNNs) and Convolutional Neural Networks (CNNs) have demonstrated significant potential in carrying out diverse prediction and classification tasks, including those related to fraud detection. RNNs are well-suited for sequential data, making them ideal for processing time-series data or textual data with temporal dependencies. On the other hand, CNNs excel at processing grid-like data, such as images and other structured representations, and are capable of automatically learning relevant features from the input. When applied to fraud prevention and detection, RNNs can effectively capture temporal patterns in transaction histories and user behaviors, enabling them to identify unusual or fraudulent activities. CNNs, on the other hand, can be employed for tasks like image-based fraud detection, where they can learn to recognize visual patterns associated with fraudulent behavior [24].

The approach of supervised machine learning involves first training the learning algorithm with labeled data and then evaluating its accuracy on a test set [25]. Labeled data is a prerequisite for using the supervised learning method to train a classifier. This labeling procedure is both time-consuming and costly. Various classifiers, such as one-class SVM [26], decision tree [27, 28], random forest [29, 30], and logistic regression [8], have shown a good level of accuracy in their performance. SVM can be used for solving both regression and classification problems [15]. One-class SVM is especially useful in scenarios where the data distribution is imbalanced, which is similar to the particular issue. The system gains the ability to deduce the characteristics of the dominant class while simultaneously being able to identify anomalies or the less prevalent class. Decision trees are structures resembling flow charts that enable the classification of input data points or the prediction of output based on an input. The Random Forest technique involves a strong methodology that incorporates numerous decision trees, which are subsequently combined to produce their outputs.

Unsupervised learning has been widely adopted in various domains and finance due to the flexibility and cost-effectiveness it offers [31]. In contrast to supervised learning, unsupervised learning methods can derive insights from a dataset without the need for labeled data. This makes it a useful tool in situations where data labeling is expensive or impractical, especially when dealing with large datasets. With the growth of big data, unsupervised learning has become increasingly important as it can help us to identify patterns, anomalies, and hidden structures in large datasets that may not be easily noticeable through manual inspection. Nearest neighbor, clustering, and outlier detection are three commonly used unsupervised learning algorithms for fraud detection [32-35]. The nearest neighbor algorithm determines the authenticity of a transaction by measuring the distance between it and its nearest neighbor in the dataset [36]. This allows for the identification of data points that are considered to be abnormal or fraudulent [37]. Clustering algorithms group similar data points together, which is particularly useful for identifying groups of transactions that exhibit similar behavior or characteristics. Peer group analysis is one application of clustering algorithms used in fraud detection [38, 39]. Outlier detection algorithms aim to locate data points that deviate from the norm in a dataset, which can be valuable in detecting fraudulent transactions [1, 40]. Credit fraud detection often deals with imbalanced data, where the number of genuine transactions is significantly larger than the number of fraudulent transactions. This can result in learning algorithms underperforming as they tend to prioritize accuracy on the majority class. Therefore, resampling methods such as oversampling or undersampling need to be used to balance the data before training the learning algorithm.

Oversampling pertains to generating artificial data points for the underrepresented class, whereas undersampling entails reducing the number of data points in the overrepresented class [12]. Careful consideration must be given to the choice of resampling method, as oversampling can lead to overfitting while undersampling can lead to loss of information. One challenge in fraud detection is the delay caused by the need for human investigators to label transactions before they can be used for training the algorithm. This delay is known as verification latency and can be reduced by automating the labeling process through the use of semi-supervised or active learning approaches [41]. Oversampling pertains to generating artificial data points for the underrepresented class, whereas undersampling entails reducing the number of data points in the overrepresented class [42]. Active learning is a method that entails selecting the most informative data points for labeling in an iterative process, which can decrease the required amount of labeled data [43, 44]. Lastly, it is essential to consider the issues of concept change over time and biased sample selection when creating a machine learning algorithm for credit fraud analysis [45]. Concept drift refers to the tendency of transaction behavior to change over time, which can lead to the algorithm becoming outdated and inaccurate [46]. Sample selection bias occurs when the distribution of data used for
training and testing the algorithm is different, which can lead to the algorithm performing poorly on unseen data [47].

III. ARTIFICIAL BEE COLONY ALGORITHM

ABC algorithm is a type of optimization algorithm that draws inspiration from the foraging behavior of honeybee colonies [48, 49]. The procedure emulates the nourishment-gathering demeanor of honeybees and employs a populace-centered strategy to explore the supreme resolution to a specified enhancement predicament. In ABC, the populace of bees is segregated into three factions: occupied bees, spectator bees, and scout bees. The assignment of the occupied bees is to probe the resolution expanse and unearth advantageous resolutions. The onlooker bees assess the solutions discovered by the employed bees according to their quality and conduct additional evaluations. The scout bees explore new regions of the search space that have not been explored by the employed and onlooker bees. ABC is based on the idea of random search, where candidate solutions are generated randomly in the search space. The quality of the solutions is evaluated using a fitness function that measures how well the solution performs on the optimization problem. ABC iteratively generates new candidate solutions by modifying the existing solutions based on the foraging behavior of honeybees. ABC has been successfully applied to various optimization issues in different fields, including engineering, economics, and bioinformatics. It has been shown to be efficient and effective in finding optimal or near-optimal solutions in many real-world optimization issues. The optimization process of ABC is summarized below:

1) Initialization: The algorithm starts by randomly generating an initial population of candidate solutions (employed bees) within the search space.

2) Employed bee phase: Each employed bee independently explores the search space by modifying its solution using a neighborhood search algorithm. The new solution is evaluated using a fitness function and compared to the current solution. If the new solution is better, it replaces the current solution. This process is repeated for all employed bees.

3) Onlooker bee phase: The employed bees communicate with the onlooker bees by performing a waggle dance to indicate the quality of their solutions. The onlooker bees select the solutions based on the quality of the dance and evaluate them using the fitness function. The onlooker bees choose the solutions with higher fitness values and use them for further exploration.

4) Scout bee phase: Some of the employed and onlooker bees become scout bees with a small probability. These scout bees randomly explore new solutions in the search space that have not been explored by the other bees.

5) Update: The algorithm updates the population by replacing the worst solutions with new solutions generated by the employed and scout bees. The algorithm terminates when a stopping criterion is met, such as reaching a maximum number of iterations or a satisfactory solution is found.

6) Output: The output of the optimization process is the best solution found by the algorithm.

IV. MODEL ARCHITECTURE

According to Fig. 1, the proposed model encompasses ABC and RL for fraud detection.

A. Pre-training

In this stage, the suggested algorithm incorporates a mutual learning-based ABC technique to initialize the weights of the MLP. The organization of these weights into a vector mirrors the bees' positions in the ABC algorithm. This process of converting the weights into a vector is commonly referred to as encoding. Finding the optimal layout for this encoding presents a challenging task, but researchers have diligently conducted numerous experiments to develop the most effective encoding strategy. As shown in Fig. 2, all the bias terms and weights within the MLP are carefully arranged into a vector, essentially forming a potential solution within the ABC algorithm. Each element of this vector corresponds to a specific weight or bias term in the neural network. By treating the vector as a candidate solution, the ABC algorithm can explore and refine its position in the solution space through the process of iterative optimization. The mutual learning-based approach in the ABC algorithm plays a crucial role in this phase. As bees in the colony share information and collectively learn from one another, the weights in the vector evolve based on the knowledge gathered from different bees' experiences. This collaboration allows the ABC algorithm to efficiently navigate the vast solution space and converge towards more promising weight configurations. The process of encoding and using a vector representation of weights provides several advantages. Firstly, it enables a seamless integration of the ABC algorithm with the neural network training process, effectively initializing the model for further optimization. Additionally, the vector-based representation facilitates the implementation of various search and optimization strategies, making it easier to explore and exploit different regions of the solution space. With the weights and bias terms encoded in a vector, the ABC algorithm operates in a population-based manner, emulating the collective intelligence of bees in a real colony. The population of candidate solutions evolves over iterations, and through the exploration and exploitation of different weight configurations, the algorithm progressively refines the MLP's parameters, ultimately leading to improved performance and convergence towards better solutions.

The quality of a candidate solution is determined by defining the fitness function as

\[
\text{Fitness} = \frac{1}{\sum_{i=1}^{N}(y_i - \hat{y}_i)^2} \tag{1}
\]

where \(N\) represents the total number of training samples, with \(y_i\) and \(\hat{y}_i\) denoting the \(i\)-th target and model-predicted output, respectively.
B. Classification

The issue of imbalanced classification arises when there is a significant difference in data volume between the two classes. To tackle this challenge, we have employed a sequential decision-making process utilizing an RL approach. The RL approach involves an agent striving to maximize its score through effective decision-making and actions within the environment, eventually leading to the discovery of an optimal policy. In the proposed framework, at every time step, the actor acquires an exemplar from the collection and executes a categorization duty. Afterward, the actor obtains prompt responsiveness from the milieu, where a precise categorization yields an affirmative grade, whereas an erroneous one produces a pessimistic grade. This feedback mechanism serves to guide the agent towards making more informed decisions and improving its performance over time. The RL algorithm plays a central role in the approach as it seeks to achieve the optimal policy by maximizing the cumulative rewards obtained throughout the decision-making process. The goal is to find the
we utilize a dataset containing N samples, each with corresponding labels. These samples are represented as \( D = \{(x_1, y_1), (x_2, y_2), \ldots, (x_N, y_N)\} \), where \( x_i \) denotes the \( i \)-th image, and \( y_i \) represents its corresponding label. These configurations are vital for setting up the environment in which the RL agent operates, guiding it towards achieving optimal classification results through the decision-making process. The following describes the intended configurations:

- **Policy** \( \pi_\theta \): Policy \( \pi \) is a mapping function that associates states (S) with actions (A). In this context, \( \pi_\theta(s_t) \) denotes the action taken in a specific state \( s_t \). The method employing the classifier with weights \( \theta \) is denoted as \( \pi_\theta \).

- **State** \( s_t \): An instance \( x_t \) extracted from the dataset, \( D \), is associated with a corresponding state \( s_t \). The first data point \( x_t \) represents the initial state \( s_t \). To avoid the model from learning a fixed sequence, \( D \) is randomized in each episode.

- **Action** \( a_t \): The action \( a_t \) is performed to predict the label \( x_t \), with binary classification, and the available choices for \( a_t \) are limited to either 0 or 1. In this context, the minority class is represented by 0, while the majority class is denoted by 1.

- **Reward** \( r_t \): The reward is contingent on the outcome of the action performed. Upon performing the correct classification, the agent receives a positive reward, whereas an incorrect classification results in a negative reward. The bonus value needs to vary for each class. Appropriately calibrated rewards can significantly improve the model's performance by ensuring that the reward level corresponds to the action taken. In this study, the reward for an action is defined using the following formula:

\[
r_t(s_t, a_t, y_t) = \begin{cases} +1, & a_t = y_t \text{ and } s_t \in D_S \\ -1, & a_t \neq y_t \text{ and } s_t \in D_S \\ \lambda, & a_t = y_t \text{ and } s_t \in D_H \\ -\lambda, & a_t \neq y_t \text{ and } s_t \in D_H \\ \end{cases}
\]

where \( D_S \) and \( D_H \) represent the majority (“sick”) and minority (“healthy”) classes, respectively. Correctly/incorrectly classifying a sample from the majority class yields a reward of \( +\lambda / -\lambda \), where \( 0 < \lambda < 1 \).

- **Terminal E**: In every instructional session, the teaching procedure concludes at diverse concluding conditions. A progression of situation-action duets \( \{(s_1, a_1, y_1), (s_2, a_2, y_2), (s_3, a_3, y_3), \ldots, (s_t, a_t, y_t)\} \) from a starting situation to an ultimate situation is denoted as an instructional session. In the circumstance, the culmination of an occurrence is ascertained by either categorizing all the instruction data or by inaccurately categorizing a specimen from the underrepresented class.

- **Transition probability** \( P \): The agent transitions to the next state, \( s_{t+1} \), from the current state, \( s_t \), based on the sequential order of the read data. The probability of transitioning to state \( s_{t+1} \) from state \( s_t \), given the action \( a_t \), is denoted as \( p(s_{t+1}|s_t, a_t) \).

### V. EXPERIMENTAL RESULTS

The dataset employed in the project is publicly accessible and was provided by the Machine Learning group of Université Libre de Bruxelles [50]. The data used in this study comprises credit card transactions made by cardholders in Europe during September 2013. This particular dataset consists of 284,807 transactions made by European cardholders over a span of two days, with 492 of them identified as fraudulent. The dataset is characterized by a significant class imbalance, where the number of positive cases (fraudulent transactions) constitutes only 0.172% of the total transactions. The dataset comprises solely numerical input variables, which are the product of a PCA transformation. The original features and additional contextual details about the data were not released to the public due to confidentiality concerns. The attributes V1 to V28 are the primary features produced through PCA, while 'Time' and 'Amount' are the only characteristics that have not been subjected to PCA transformation. The attribute 'Time' represents the time interval in seconds between a particular transaction and the initial transaction registered in the dataset. The 'Amount' characteristic pertains to the amount of the transaction and can be applied to tasks such as cost-sensitive learning, which depend on the transaction value. The variable 'Class' serves as the output or target variable and is assigned the value of 1 if the transaction is a fraud and 0 if it is not. The dataset of credit card transactions may also include summarized characteristics. Various summarized characteristics can be extracted from the credit card transaction dataset, such as the average monthly transaction amount per cardholder, the average number of transactions per month, the average monthly spending on fuel, the time and distance between the present and previous transactions, and others.

In this article, we have incorporated a batch normalization layer to ensure data normalization and facilitate smoother training. By processing the data before feeding it into the MLP, the batch normalization layer effectively addresses the issue of internal covariate shift, allowing the network to learn more effectively and expedite convergence. To introduce non-linearity and enhance the network's capacity to model complex relationships within the data, we apply the ReLU (Rectified Linear Unit) activation function between the layers. Regarding the training configuration, we have carefully chosen a batch size of 32. This decision is made to strike a balance between computational efficiency and gradient accuracy during the optimization process. The batch size of 32 enables efficient parallel processing on modern hardware while retaining sufficient samples to ensure a stable gradient estimation during backpropagation.

The proposed approach was subjected to a rigorous evaluation by comparing it with six different machine learning models. These models included SVM [51], Naïve Bayes [52], KNN [53], Random forests [54], Logistic Regression [55], and Decision tree [56], which are all popular and widely used in the...
field of machine learning (See Table I). In addition, the performance of two smaller versions of the proposed approach was also tested. These versions were designed to use random weights and RL for classification. For evaluating the outcomes of the approach, standard performance metrics, including F-measure and geometric mean, known to be dependable for assessing imbalanced data, were employed. The approach outperformed all other models across all evaluation criteria, surpassing even the top-performing model, Decision tree. Specifically, the approach achieved a reduction in the error rate by more than 65% and 28% in F-measure and G-means, respectively. Additionally, a comparison was conducted between the performance of the approach and the smaller versions, namely Proposed (random weights) and Proposed (random weights and RL). This comparison revealed that the approach significantly reduced the error rate by approximately 72%. These results highlight the importance and effectiveness of the improved artificial bee colony and RL techniques utilized in the suggested approach.

In the next experiment, the objective is to evaluate ABC against various metaheuristic optimization algorithms. To achieve this, different metaheuristics are utilized to obtain the initial model parameters while keeping other model components constant. The six algorithms used in this experiment are HMS [57], FA [58], BA [59], DE [60], GWO [61], and COA [62]. For all algorithms, default settings have been used (Table II). Table III presents the results obtained from this comparison. The results indicate that the proposed ABC approach outperforms other algorithms in terms of error reduction, with a decrease of approximately 33% compared to the second-best algorithm, HMS.

### Table I. Results of Various Classification Algorithms

<table>
<thead>
<tr>
<th></th>
<th>accuracy</th>
<th>recall</th>
<th>precision</th>
<th>F-measure</th>
<th>G-means</th>
</tr>
</thead>
<tbody>
<tr>
<td>Naïve Bayes</td>
<td>0.695 ± 0.160</td>
<td>0.610 ± 0.180</td>
<td>0.560 ± 0.100</td>
<td>0.580 ± 0.041</td>
<td>0.695 ± 0.160</td>
</tr>
<tr>
<td>Random forests</td>
<td>0.705 ± 0.015</td>
<td>0.580 ± 0.035</td>
<td>0.570 ± 0.107</td>
<td>0.580 ± 0.035</td>
<td>0.705 ± 0.015</td>
</tr>
<tr>
<td>SVM</td>
<td>0.825 ± 0.165</td>
<td>0.790 ± 0.025</td>
<td>0.730 ± 0.000</td>
<td>0.760 ± 0.263</td>
<td>0.825 ± 0.255</td>
</tr>
<tr>
<td>KNN</td>
<td>0.800 ± 0.015</td>
<td>0.680 ± 0.078</td>
<td>0.720 ± 0.097</td>
<td>0.700 ± 0.120</td>
<td>0.800 ± 0.000</td>
</tr>
<tr>
<td>Decision tree</td>
<td>0.855 ± 0.105</td>
<td>0.840 ± 0.105</td>
<td>0.780 ± 0.485</td>
<td>0.820 ± 0.056</td>
<td>0.855 ± 0.269</td>
</tr>
<tr>
<td>Logistic Regression</td>
<td>0.830 ± 0.110</td>
<td>0.750 ± 0.090</td>
<td>0.790 ± 0.059</td>
<td>0.770 ± 0.025</td>
<td>0.830 ± 0.142</td>
</tr>
<tr>
<td>Proposed (random weights)</td>
<td>0.810 ± 0.120</td>
<td>0.820 ± 0.140</td>
<td>0.800 ± 0.041</td>
<td>0.790 ± 0.129</td>
<td>0.810 ± 0.012</td>
</tr>
<tr>
<td>Proposed (random weights and RL)</td>
<td>0.860 ± 0.005</td>
<td>0.870 ± 0.105</td>
<td>0.850 ± 0.200</td>
<td>0.850 ± 0.012</td>
<td>0.860 ± 0.035</td>
</tr>
<tr>
<td>Full model</td>
<td>0.890 ± 0.015</td>
<td>0.910 ± 0.120</td>
<td>0.892 ± 0.0312</td>
<td>0.890 ± 0.003</td>
<td>0.898 ± 0.055</td>
</tr>
</tbody>
</table>

### Table II. Metaheuristics Parameter Settings

<table>
<thead>
<tr>
<th>algorithm</th>
<th>parameter</th>
<th>value</th>
</tr>
</thead>
<tbody>
<tr>
<td>HMS</td>
<td>minimum mental processes</td>
<td>2</td>
</tr>
<tr>
<td></td>
<td>maximum mental processes</td>
<td>5</td>
</tr>
<tr>
<td></td>
<td>C</td>
<td>1</td>
</tr>
<tr>
<td>FA</td>
<td>light absorption coefficient</td>
<td>1</td>
</tr>
<tr>
<td></td>
<td>attractiveness at r = 0</td>
<td>0.1</td>
</tr>
<tr>
<td></td>
<td>scaling factor</td>
<td>0.25</td>
</tr>
<tr>
<td>BA</td>
<td>constant for loudness update</td>
<td>0.5</td>
</tr>
<tr>
<td></td>
<td>constant for an emission rate update</td>
<td>0.5</td>
</tr>
<tr>
<td></td>
<td>initial pulse emission rate</td>
<td>0.001</td>
</tr>
<tr>
<td>DE</td>
<td>scaling factor</td>
<td>0.5</td>
</tr>
<tr>
<td></td>
<td>crossover probability</td>
<td>0.9</td>
</tr>
<tr>
<td>COA</td>
<td>discovery rate of alien solutions</td>
<td>0.25</td>
</tr>
</tbody>
</table>

### Table III. Outcomes of Different Metaheuristic Algorithms

<table>
<thead>
<tr>
<th>algorithm</th>
<th>accuracy</th>
<th>recall</th>
<th>precision</th>
<th>F-measure</th>
<th>G-means</th>
</tr>
</thead>
<tbody>
<tr>
<td>HMS</td>
<td>0.872±0.058</td>
<td>0.860±0.103</td>
<td>0.874±0.041</td>
<td>0.862±0.008</td>
<td>0.842±0.082</td>
</tr>
<tr>
<td>FA</td>
<td>0.861±0.138</td>
<td>0.850±0.093</td>
<td>0.862±0.231</td>
<td>0.849±0.014</td>
<td>0.820±0.021</td>
</tr>
<tr>
<td>BA</td>
<td>0.847±0.004</td>
<td>0.835±0.113</td>
<td>0.830±0.251</td>
<td>0.839±0.065</td>
<td>0.800±0.000</td>
</tr>
<tr>
<td>DE</td>
<td>0.830±0.014</td>
<td>0.820±0.006</td>
<td>0.821±0.061</td>
<td>0.825±0.145</td>
<td>0.782±0.120</td>
</tr>
<tr>
<td>GWO</td>
<td>0.812±0.159</td>
<td>0.792±0.014</td>
<td>0.806±0.261</td>
<td>0.792±0.165</td>
<td>0.761±0.150</td>
</tr>
<tr>
<td>COA</td>
<td>0.760±0.140</td>
<td>0.744±0.004</td>
<td>0.760±0.000</td>
<td>0.750±0.211</td>
<td>0.710±0.110</td>
</tr>
</tbody>
</table>
A. Effect of the Reward Function

Rewards of $\pm 1$ and $\pm \lambda$ are used in this study to indicate correct/incorrect classifications of the majority and minority classes, respectively. The optimal value of $\lambda$ is influenced by the proportion of the majority to minority samples, with a lower value expected as the ratio increases. The performance of the suggested model with $\lambda$ initialized using a set of values ranging from 0 to 1 in increments of 0.1 is evaluated while keeping the bonus for the majority class constant. The results, which demonstrate that a $\lambda$ value of 0.4 yields the best model performance across all metrics are shown in Fig. 3. When $\lambda = 0$, the impact of the majority class is nullified, while a value of 1 results in equal impact from both majority and minority classes. It is important to note that while adjusting $\lambda$ is necessary to mitigate the effect of the majority class, setting the value too low can negatively impact the overall performance of the model.

B. Investigating the Impact of the Number of Layers in MLP

The number of layers in MLP affects the model complexity, with a higher number of layers resulting in increased complexity that may lead to over-fitting. Conversely, a model with too few layers may not capture important features in the training data. To address these issues, the impact of the number of layers on the proposed approach was evaluated by testing eight values between 1 and 12. Table IV shows the results, which demonstrate a decreasing trend for values from 1 to 8, followed by an increasing trend for values from 8 to 12. Therefore, the optimal number of layers for MLP is 8 to balance the representation of important features and model complexity.

C. Effect of the Loss Function

Traditional methods, such as adjusting data augmentation and the loss function, can be utilized to address data imbalances. The loss function, which can assign more weight to the minority class, is considered particularly important. Various loss functions, including (WCE) $[63]$, balanced cross-entropy (BCE) $[64]$, Dice loss (DL) $[65]$, Tversky loss (TL) $[66]$, and Combo Loss (CL) $[67]$, were tested to determine their impact on the model. In the BCE and WCE functions, equal weight is assigned to positive and negative examples. The CL function, which assigns less weight to simple examples and more weight to complex ones, is useful for handling unbalanced data. The results presented in Table V show that CL performs better than TL, with a 31% and 42% reduction in error for accuracy and F-measure, respectively. However, RL performs 71% better than the CL function.

![Fig. 3. The performance metrics of the proposed model are graphed against different values of $\lambda$ in the reward function.](image)

**Table IV. The Performance Metrics Plotted vs. the Different Number of Layers in MLP**

<table>
<thead>
<tr>
<th>Number of layers</th>
<th>accuracy</th>
<th>recall</th>
<th>precision</th>
<th>F-measure</th>
<th>G-means</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>0.758±0.001</td>
<td>0.792±0.205</td>
<td>0.741±0.017</td>
<td>0.742±0.037</td>
<td>0.773±0.100</td>
</tr>
<tr>
<td>2</td>
<td>0.872±0.010</td>
<td>0.883±0.163</td>
<td>0.864±0.007</td>
<td>0.869±0.032</td>
<td>0.883±0.155</td>
</tr>
<tr>
<td>4</td>
<td>0.852±0.020</td>
<td>0.861±0.111</td>
<td>0.850±0.233</td>
<td>0.830±0.012</td>
<td>0.862±0.024</td>
</tr>
<tr>
<td>8</td>
<td>0.888±0.031</td>
<td>0.905±0.113</td>
<td>0.883±0.023</td>
<td>0.883±0.021</td>
<td>0.905±0.036</td>
</tr>
<tr>
<td>10</td>
<td>0.724±0.135</td>
<td>0.740±0.121</td>
<td>0.702±0.211</td>
<td>0.720±0.017</td>
<td>0.670±0.036</td>
</tr>
<tr>
<td>12</td>
<td>0.510±0.025</td>
<td>0.612±0.011</td>
<td>0.510±0.043</td>
<td>0.546±0.013</td>
<td>0.440±0.110</td>
</tr>
</tbody>
</table>
D. Discussion

The article proposed a deep reinforcement learning approach for fraud detection in e-commerce transactions. It acknowledged the serious issue of fraud in e-commerce and the limitations of current detection strategies. The proposed model utilized ANNs and the ABC algorithm to acquire initial weight values. The model viewed fraud detection as a step-by-step decision-making process, with the agent receiving rewards for each classification action. To prioritize detecting fraudulent transactions, the model assigns higher rewards to identifying the minority class.

Fraudsters are known for their adaptability and creativity in devising new strategies to evade detection. As a result, the effectiveness of any fraud detection model, including the proposed deep reinforcement learning approach, hinges on its ability to generalize well to previously unseen fraud patterns. The process of generalization refers to the model's capacity to make accurate predictions on data that differs from the training set, encompassing novel and evolving fraud scenarios. To ensure the robustness of the proposed model, rigorous testing on diverse and evolving fraud scenarios is imperative. Here are some key aspects to consider for assessing the model's generalization capabilities:

- Diverse Testing Datasets: Apart from the publicly available dataset used during model development, it is crucial to evaluate the model on multiple datasets, including those collected from different sources and time periods. Diverse datasets can represent a broader range of fraud patterns and help uncover potential weaknesses in the model's detection capabilities.

- Cross-Domain Evaluation: Fraud patterns may differ across various industries and regions. Evaluating the model's performance on datasets from different domains, such as e-commerce, banking, insurance, etc., helps assess its ability to handle variations in fraud characteristics and attack vectors.

- Time-based Evaluation: Fraud patterns evolve over time, necessitating the model's ability to adapt to emerging fraud tactics. Testing the model on data from different time periods can reveal its responsiveness to temporal changes in fraud behavior.

- Transfer Learning: Applying transfer learning techniques allows leveraging knowledge gained from one dataset to improve performance on another. Pre-trained models can serve as a starting point for fine-tuning on specific fraud detection tasks, potentially enhancing the model's generalization capacity.

- Data Augmentation: To expose the model to a wider array of fraud patterns, data augmentation techniques can be employed. Synthetic fraud scenarios can be generated by perturbing existing data or by using generative models, thus enriching the training data and improving generalization.

- Continuous Monitoring and Feedback: Real-world deployment of the model demands continuous monitoring of its performance and feedback from analysts and fraud experts. This feedback loop helps identify potential misclassifications and enables timely updates to the model to account for evolving fraud patterns.

- Adversarial attacks pose a significant challenge to fraud detection systems, as they can lead to severe financial losses and reputational damage. Malicious actors exploit vulnerabilities in the model's decision boundaries, making subtle changes to input data that are imperceptible to humans but can mislead the model into producing incorrect outputs. For instance, attackers may modify features in a transaction or manipulate user behavior to hide fraudulent activities. To ensure the reliability and effectiveness of the proposed deep reinforcement learning model in real-world scenarios, it is essential to assess its robustness against various types of adversarial attacks. There are several approaches to evaluate the model's susceptibility to such attacks:

- Adversarial Testing: Conducting rigorous adversarial testing involves generating adversarial samples and evaluating how the model responds to them. Adversarial samples can be crafted using techniques like Fast Gradient Sign Method (FGSM), Projected Gradient Descent (PGD), or Genetic Algorithms. By testing the model's performance on these samples, researchers can identify vulnerabilities and areas of improvement.

- Robustness Metrics: Various robustness metrics have been proposed to quantify a model's resilience against adversarial attacks. Examples include robust accuracy, fooling rate, and adversarial training loss. These metrics help in comparing the model's performance under normal and adversarial conditions, providing insights into its vulnerability.

- Adversarial Training: Adversarial training is a popular technique to enhance a model's robustness. It involves augmenting the training dataset with adversarial samples, forcing the model to learn from both clean and adversarial data. This process can improve the model's

---

### TABLE V. RESULTS OF VARIED LOSS FUNCTIONS

<table>
<thead>
<tr>
<th></th>
<th>accuracy</th>
<th>recall</th>
<th>precision</th>
<th>F-measure</th>
<th>G-means</th>
</tr>
</thead>
<tbody>
<tr>
<td>WCE</td>
<td>0.765 ± 0.032</td>
<td>0.755 ± 0.016</td>
<td>0.746 ± 0.125</td>
<td>0.751 ± 0.005</td>
<td>0.777 ± 0.038</td>
</tr>
<tr>
<td>BCE</td>
<td>0.815 ± 0.027</td>
<td>0.807 ± 0.055</td>
<td>0.786 ± 0.171</td>
<td>0.784 ± 0.016</td>
<td>0.825 ± 0.003</td>
</tr>
<tr>
<td>DL</td>
<td>0.826 ± 0.038</td>
<td>0.815 ± 0.031</td>
<td>0.794 ± 0.032</td>
<td>0.812 ± 0.010</td>
<td>0.834 ± 0.002</td>
</tr>
<tr>
<td>TL</td>
<td>0.844 ± 0.129</td>
<td>0.838 ± 0.009</td>
<td>0.814 ± 0.021</td>
<td>0.827 ± 0.042</td>
<td>0.857 ± 0.071</td>
</tr>
<tr>
<td>CL</td>
<td>0.874 ± 0.006</td>
<td>0.866 ± 0.218</td>
<td>0.854 ± 0.009</td>
<td>0.853 ± 0.053</td>
<td>0.876 ± 0.156</td>
</tr>
</tbody>
</table>
ability to detect fraudulent activities in the presence of adversarial examples.

- **Transferability Analysis:** It is crucial to examine whether adversarial attacks generated for one model can also fool other models. Transferability analysis helps in understanding the generalizability of adversarial attacks across different fraud detection models. If attacks transfer across models, it indicates a common vulnerability that needs to be addressed.

- **Use of Certified Defenses:** Certified defenses, such as certified robustness and provable security techniques, provide mathematical guarantees against adversarial attacks. By incorporating such defenses into the model, the system can offer formal guarantees of security and robustness.

- **Real-world Adversarial Testing:** It is essential to conduct adversarial testing using real-world data and attack scenarios. This involves simulating how actual attackers might attempt to evade the model's detection mechanisms. This testing should include both known and novel adversarial strategies to ensure comprehensive evaluation.

**VI. CONCLUSION**

The suggested model offers a hopeful solution to the issue of identifying fraud in e-commerce transactions, a significant apprehension for companies and customers. The use of multilayer perceptron, RL, and ABC allows for a more robust and accurate fraud detection system. Pre-training the network weights with the evolutionary ABC algorithm is a crucial measure to prevent being trapped in local optima. By using this initialization method, the model is better able to converge to a global optimum, leading to improved accuracy and reliability. The utilization of RL to address dataset imbalance is another notable aspect of the proposed model. Imbalanced datasets are a common challenge in machine learning, and traditional approaches often fail to provide satisfactory results. RL offers a new way to address this issue, allowing the model to learn how to handle imbalanced data on its own, leading to improved performance. Based on the results obtained from the experiments conducted on the utilized dataset, it can be inferred that the proposed model outperforms other existing machine learning models in terms of common metrics. The superior performance of the ABC algorithm and RL over other metaheuristic initialization algorithms and loss functions demonstrates the effectiveness of the suggested approach.

Potential future work includes testing the proposed model on a broader and more varied dataset of e-commerce transactions to assess its ability to generalize. This would allow assessing whether the model is robust enough to identify fraud patterns in different scenarios and datasets, which is crucial for its practical applicability. Additionally, it would be interesting to explore the interpretability of the model, as understanding how it identifies fraudulent transactions can provide valuable insights for fraud detection in e-commerce. Finally, further research could investigate the scalability of the proposed approach, as it may become computationally expensive when dealing with large datasets.

**ACKNOWLEDGMENT**

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**REFERENCES**


[12] "!! INVALID CITATION !!


Optimal Scheduling using Advanced Cat Swarm Optimization Algorithm to Improve Performance in Fog Computing

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Abstract—Fog computing can be considered a decentralized computing approach that essentially extends the capabilities offered by cloud computing to the periphery of the network. In addition, due to its proximity to the user, fog computing proves to be highly efficient in minimizing the volume of data that needs to be transmitted, reducing overall network traffic, and shortening the distance that data must travel. But this technology, like other new technologies, has challenges, and scheduling and optimal allocation of resources is one of the most important of these challenges. Accordingly, this research aims to propose an optimal solution for efficient scheduling within the fog computing environment through the application of the advanced cat swarm optimization algorithm. In this solution, the two main behaviors of cats are implemented in the form of seek and tracking states. Accordingly, processing nodes are periodically examined and categorized based on the number of available resources; servers with highly available resources are prioritized in the scheduling process for efficient scheduling. Subsequently, the congested servers, which may be experiencing various issues, are migrated to alternative servers with ample resources using the virtual machine live migration technique. Ultimately, the effectiveness of the proposed solution is assessed using the iFogSim simulator, demonstrating notable reductions in execution time and energy consumption. So, the proposed solution has led to a 20% reduction in execution time while also improving energy efficiency by more than 15% on average. This optimization represents a trade-off between improving performance and reducing resource consumption.

Keywords—Scheduling; fog computing; optimal balancing; cat swarm optimization algorithm

I. INTRODUCTION

In order to overcome the challenges arising from resource limitations in IoT devices, the prevalent approach has been to rely on large-scale cloud data centers for interactions between IoT devices and supporting end servers [1]. However, as the number of IoT devices and their generated data continue to escalate, reliance on cloud-based infrastructure has become costly, inefficient, and often unfeasible [2]. In response to this issue, fog computing has emerged as a solution by offering networking, storage, and computing resources in proximity to IoT devices and users [3, 4]. One notable advantage of fog computing is its ability to reduce service latency for end-user applications, unlike the cloud, which typically exhibits higher latency due to its more extensive computing capacity and remote storage [5]. The exponential growth of the Internet of Things has posed significant challenges for cloud computing, including network failures and increased latency. To tackle these challenges, cloud computing has sought to bring cloud capabilities closer to IoT devices. Fog computing entails the utilization of heterogeneous and distributed processing nodes, presenting challenges for fog-based services in accommodating the diverse aspects of a constrained environment [23].

By examining the structural and service-oriented characteristics of fog computing, various challenges become apparent, with optimal scheduling being particularly significant. Scheduling holds great importance in the realm of the Internet of Things as it has the potential to decrease execution time and minimize energy consumption [6, 24]. However, scheduling problems become increasingly complex with the growing number of services and requests, leading to a rapid increase in the number of possible solutions. Due to the exponential growth of feasible states, it becomes impractical to evaluate all possibilities to determine the best scheduling exhaustively, resulting in these problems falling under the NP-Hard category that deterministic methods cannot be used in solving this category of problems due to their time-consuming nature, and meta-heuristic methods should be developed to solve these problems properly [2, 25]. Meta-heuristic methods have been the attention of researchers due to their simplicity, flexibility, no need for derivation, and escape from local optima. The No Free Lunch (NFL) theorem establishes that no meta-heuristic algorithm can solve all optimization problems perfectly. In other words, an algorithm that performs well on a set of problems may not yield favorable outcomes for another set of problems [3, 26]. Taking this into account, this study introduces an advanced strategy based on the cat swarm optimization (CSO) algorithm to achieve optimal scheduling in cloud infrastructure. The primary objective is to significantly reduce energy consumption in cloud data centers by effectively allocating tasks to processing servers, thereby preventing server overload or underload.

The following section presents a literature review on scheduling and load balancing, followed by an examination of the proposed technique in section 3. Finally, in section 4, the proposed technique is implemented in the iFogSim simulator, and the results are evaluated.
A. Fog Computing

As illustrated in Fig. 1, fog computing operates by conducting local processing and storage of IoT data on IoT devices rather than transmitting the data to the cloud. In contrast to the cloud, fog computing offers faster response times and improved quality, making it an optimal choice for enabling the Internet of Things. It is known for providing efficient and secure services to a wide range of users [4, 27]. While the cloud serves as an intermediary between endpoint clients and cloud computing, fog computing occurs closer to the edge of the cloud and end devices, resulting in significantly lower latency. Cloud computing services are located on the Internet, whereas fog computing operates at the edge of the local network. Real-time interactions are supported by both cloud and fog computing, but processing data and applications in the cloud can be time-consuming, particularly for large-scale data. Fog computing facilitates centralized resource management, including the allocation of computing, networking, and storage resources. The primary objective of fog computing is to equip network edges and network devices with virtual services for processing, storage, and network provisioning [5, 28]. Table I provides a comparison of the similarities and differences between fog computing and cloud computing.

<table>
<thead>
<tr>
<th>Ability</th>
<th>Fog Computing</th>
<th>Cloud Computing</th>
</tr>
</thead>
<tbody>
<tr>
<td>Delay</td>
<td>Low</td>
<td>Up (depending on the user's path to DC)</td>
</tr>
<tr>
<td>Response time</td>
<td>milliseconds</td>
<td>several minutes</td>
</tr>
<tr>
<td>Service location</td>
<td>Network edge</td>
<td>In cloud data centers</td>
</tr>
<tr>
<td>Data storage period</td>
<td>transitory</td>
<td>months or years (according to the contract)</td>
</tr>
<tr>
<td>Steps between user and server</td>
<td>one step</td>
<td>Several steps</td>
</tr>
<tr>
<td>Aware of location</td>
<td>Very local</td>
<td>no</td>
</tr>
<tr>
<td>architecture</td>
<td>Distributed</td>
<td>concentrated</td>
</tr>
<tr>
<td>Type of communication</td>
<td>wireless</td>
<td>Broadband, MPLS</td>
</tr>
<tr>
<td>Possibility of data recording</td>
<td>Low</td>
<td>High</td>
</tr>
<tr>
<td>End-to-end security</td>
<td>can be defined</td>
<td>It cannot be defined or controlled</td>
</tr>
<tr>
<td>Data collection nodes</td>
<td>unlimited</td>
<td>Very low</td>
</tr>
<tr>
<td>Mobility support</td>
<td>supports</td>
<td>Limited support</td>
</tr>
</tbody>
</table>
Fog computing offers significant advantages by reducing data transfer, traffic, and distance traveled. It involves decentralizing computations to the edge of the network, where downstream data processing occurs in cloud services and upstream data processing occurs in Internet of Things (IoT) services [6, 29]. In cloud computing, information storage and processing are directed towards the network edge and closer to the source of information generation. Instead of transmitting IoT-generated data to distant data centers or remote servers for storage and processing, this data is stored on local servers and storage devices through a local gateway. This approach enhances the speed of information analysis and alleviates network congestion [7, 30]. The concept behind fog computing is that computation should be performed in close proximity to data sources. It has the potential to impact society as significantly as cloud computing. Fog computing offers numerous advantages over traditional architectures, particularly optimizing resource utilization within a cloud computing system. By conducting computations at the network edge, fog computing reduces network traffic. Essentially, edge computing operates within the cloud but in proximity to objects interacting with IoT data. As depicted in Fig. 1, fog computing acts as an intermediary between the cloud and end devices, bringing processing, storage, and network services closer to the end devices. These devices, known as edge nodes, can be deployed anywhere with a network connection. In essence, fog computing extends cloud infrastructure to the network edge. While fog and cloud computing share network, computing, and storage resources, they utilize similar mechanisms and features such as virtualization and multi-tenancy [8, 31].

II. RELATED WORK

The study in [9] focuses on addressing concerns related to scheduling and resource allocation within fog computing for various types of applications. In this research, it has been stated that different programs composed of a collection of interrelated services are mapped to cloud computing for processing, but the placement of services has its challenges. Accordingly, this article presents a distributed placement strategy, the main goal of which is to reduce energy consumption and the cost of communication. The proposed method is based on game theory, which uses iterative combinatorial auction (ICA). The proposed solution through game theory makes significant changes in the ICA method. Based on this, the proposed solution is decentralized and is able to interact between fog nodes and applications so that decisions related to placement are made in each round. The results of the evaluations show the optimality of the solution in reducing the cost of communication and increasing the productivity of the fog node processor, which is an important advantage considering the limited resources in the fog nodes. In [10], a solution for autonomous scheduling of services on devices on the edge of the network is presented. In this solution, the dynamic capability of programs based on microservices is used in order to provide an adaptive solution for placement. The main objective of this solution is to minimize the response time of services, ultimately enhancing the quality of user experience, particularly in critical services. In this method, the decision about the placement of the service is made based on the response time of the applications. Various functions have been used for this purpose. Then, through the use of a meta-heuristic algorithm based on PSO, a decision is made regarding the placement of the service. The first input parameter for the algorithm is the graph call related to the application service.

In the end, through a series of evaluations, it was shown that the proposed solution was able to perform the placement of services and microservices in such a way that the response time is minimized. The authors in [11, 32] have presented a lightweight framework for providing fog computing services to be used by IoT devices that are sensitive to delay, and their needs must be answered in real-time. The FogPlan solution is a QoS-aware proposal, and the placement of services in it is done according to the network status. This framework works with minimal assumptions and information about IoT nodes. Then a probabilistic formula for the optimization problem, along with two heuristic algorithms, has been presented to answer the QoS needs. By using this solution, in addition to more accurate positioning, the amount of delay and network cost can also be reduced. In the evaluations, it has been concluded that the solution has reduced the average delay and the scheduling costs through the optimal scheduling of services.

Research [12] introduces a task scheduling policy in fog computing that is based on graph partitions. The main objective of this solution is to enhance accessibility. By employing this method, the quality-of-service delivery is enhanced through program prioritization across all cloud devices and transferring services to these devices. The main idea is based on the premise that the more necessary services are provided near the user, the quality and delay of service delivery can be reduced. Based on this, with the help of graphic partitions, the desired services and related tasks are determined and based on the number of available resources, the best possible placement is done. In the end, the proposed solution has been evaluated with a linear programming approach, and the results indicate an improvement in accessibility and service delivery. The work described in [13] focuses on scheduling application components within hybrid cloud and fog systems based on network function virtualization (NFV) principles. This study investigates the main challenges of deploying components related to cloud and fog applications in NFV. In fact, these challenges are mostly expressed due to the heterogeneity between the components. To solve this problem in this research, a model based on four hierarchies 1) sequence, 2) parallel, 3) selection and 4) compliance loop was used. In this solution, it is assumed that the program components are implemented as VFSs, and the structural diagrams represent an application program that follows the proposed hierarchical structure. In this structure, each VFN becomes a tree, where the leaf nodes represent the program components, and the costs of the model are calculated by gathering the information of the nodes from bottom to top. The main goal of this method is to minimize the processing costs, which are modeled using an ILP integer programmer. The authors of [14] propose a concept to address the challenge of deploying Internet of Things (IoT) services within the fog computing environment. They model the deployment problem of IoT applications on resources as an optimization problem. The concept of a "fog colony" is introduced, and the coordinated fog cell colony and deployment approach are presented as a solution for deploying
services on virtual resources within the fog domain. The optimization problem primarily focuses on the deployment and execution time of the application, aiming to maximize resource utilization in the fog through a greedy heuristic method based on the genetic algorithm. In this model, the communication and subscription of fog cell services are done by the control node. Also, the communication between the cells and the control node with the cloud is made possible by the fog computing management system.

In the end, the results of this research have been compared with the classical approach that executes all tasks in the cloud. The intended evaluations have been carried out in the iFogSim simulator environment, and the results indicate that the optimization method has a favorable effect on the use of fog resources, and the genetic algorithm has a lower delay for execution compared to the cloud. In [15], a service placement algorithm is presented for efficient use of the network and improvement of energy consumption. This algorithm sequentially allocates application modules with the highest needs to the nodes with the highest capacities. To test the proposed algorithms, the program has been implemented on network topologies named JSON. The scenario has been implemented with more than three different network topologies and workloads. The proposed cloud-fog placement and deployment method were evaluated by comparing it with the traditional cloud-based deployment approach. The evaluation considered program delay (response time), network usage, and energy consumption as the primary metrics. Data was collected from the iFogSim simulator using the proposed placement and resource deployment methods. The results demonstrate the favorable impact of the proposed placement approach across all three topologies, showcasing improvements in network utilization, program delay (response time), and energy consumption compared to the traditional cloud-based approach. The authors in [16] presented a tool called FogTorch. Accordingly, the first part of the article presents a model for the use of service quality systems in multi-sector applications of the Internet of Things for fog architecture. In the second part, the results of using Monte Carlo simulations for the FogTorch tool are stated, and the application of this tool is examined in terms of service quality and resource consumption.

FogTorch is a tool that enables the simulation and comparison of different fog scenarios in the design phase, as well as the resource and quality-of-service (QoS) aware deployment of IoT applications through cloud-fog architecture. It takes into account various processing resources such as CPU, RAM, storage, and software, along with QoS constraints like latency and bandwidth, which are essential for real-time fog applications. Notably, FogTorch is the first tool capable of estimating the quality of service resulting from fog-based application deployments based on probability distribution models of bandwidth and delay provided by communication links. Additionally, it provides estimates of resource consumption within the fog layer, allowing for the optimization of resource utilization among different fog nodes. In [17], a dynamic module is presented to schedule the tasks of Internet of Things applications in edge and cloud computing. Through this solution, the problem related to IoT requests in a heterogeneous network environment based on edge cloud has been solved. As a result, a mapping between the application module and the main resources of the devices can be created. In this way, problems related to the delay of tasks and energy consumption can be overcome. To achieve this objective, the application employs a dynamic discovery algorithm that enables the step-by-step execution of operations in the fog computing environment. This algorithm helps to reduce the delay in task execution by efficiently discovering and allocating resources. Through simulations conducted in the iFogSim environment, the results demonstrate the remarkable service quality of the applications, accompanied by a significant reduction in energy consumption compared to other scheduling strategies. The combination of the dynamic discovery algorithm and the fog computing environment contributes to improved application performance and energy efficiency.

After conducting a thorough examination of the aforementioned studies, we have identified their strengths and limitations in several aspects:

- Studies, such as [9], [13], [14] have provided comprehensive introductions to the theoretical concepts and platforms of fog computing. They have significantly contributed to the research on optimization strategies in this field.

- Several existing studies, like [10]-[12] and [15]-[17] have focused on optimizing the trade-off between time and energy consumption in fog computing scenarios. However, one critical aspect they often overlook is the consideration of task characteristics and resource types concerning the problem of balancing delay and device load. This omission can lead to inefficient resource allocation and potential wastage of resources.

Our research falls within the category of optimizing Quality of Service (QoS) and energy consumption simultaneously. It complements and extends existing works in several ways. For instance, unlike the study in [10] where multiple user devices share one Multi-access Edge Computing (MEC) server, or [11] where a single device generates a task, our approach considers a more complex system involving multiple users and multiple fog nodes. Additionally, our focus is on reducing the energy consumption of the fog nodes themselves, which sets us apart from the majority of existing approaches that primarily aim to minimize energy consumption on the mobile devices [9,11,12,16,17]. Furthermore, we present a novel approach that jointly minimizes the overall energy consumption and the execution time while taking into account the computation resource constraints of fog devices. This approach offers a comprehensive optimization strategy for the system.

### III. Proposed Method

The primary objective of the scheduling solution is to efficiently allocate n tasks to m machines, where n > m, with the aim of minimizing both the execution time and energy consumption. To achieve this objective, it is crucial to compute the resource availability of all machines. An upper bound is established for the minimization function, which is defined as m × L\text{max}. Here, L\text{max} represents the maximum execution time
for each node. Thus, for any task scheduling completion time \( G \), it should satisfy the condition \( m \times L_{\text{max}} > G \). In other words, the execution of the assigned tasks will take at least as long as \( L_{\text{max}} \) for each container. Let's consider \( S = (S_1, S_2, ..., S_n) \) as the set of user tasks given to the system. During the scheduling process, these user tasks are assigned to available fog nodes. Each node is equipped with \( \{PE_1, PE_2, ..., PE_m\} \) processing elements to handle service tasks. Each processing element has a distinct processing speed characteristic denoted as \( PE_j \) and the CSO (Cat Swarm Optimization) algorithm [18, 33] is used to allocate resources and schedule tasks in a fog infrastructure effectively.

The CSO algorithm models the two primary behaviors of cats, known as tracking and searching modes, using sub-models. By combining these modes in a defined ratio, the cat crowding optimization algorithm exhibits strong performance. Similar to particle swarm optimization, the positions of the cats serve as solutions, and the algorithm utilizes the cats' behavior to solve optimization problems. In the cat swarm optimization, the number of cats to be used is determined, and each cat possesses a position with \( M \) dimensions. Additionally, each cat has a speed for each dimension and a fitness value indicating its level of fitness. This fitness value is obtained using a fitness function. Furthermore, each cat is assigned a flag to identify whether it is in tracking or seeking mode [19, 34].

The aim of this research is to minimize energy consumption and reduce the execution time of user requests in the cloud infrastructure. To achieve this, the cat swarm optimization algorithm is modified to enable the effective allocation of resources to tasks. The modified algorithm optimizes the resource allocation process, leading to improved energy efficiency and reduced request execution time. By leveraging this modified cat swarm optimization algorithm, the cloud infrastructure can allocate resources more effectively, resulting in enhanced performance and reduced energy consumption. In the proposed solution, a set of cats will be used; some of which are in search mode and at the same time, some others are in tracking mode. In this method, each cat represents a specific task-resource mapping. The cats are updated based on their current state, and the goal is to minimize the cost of mapping. The fitness value is assigned to each cat, reflecting the quality of its mapping solution. In each iteration of the algorithm, a new set of cats is selected to be in the search state, where they explore different mappings. Ultimately, the cat with the best fitness value represents the best mapping solution, which results in the lowest cost among all the possible mappings. By iteratively updating the cats and selecting the best solutions, the algorithm aims to find an optimal task-resource mapping that minimizes cost and improves overall efficiency. Accordingly, in the following, the search and tracking modes are examined as a sub-model to model the cat's search behavior to find the desired target: processing servers in this research.

A. Seeking Mode

During the search process in the SM (Seeking Mode) of the algorithm, the exploration of different regions in the search space is conducted. However, the search is limited to the local vicinity of the current position of the seeking cat. This approach focuses on refining existing solutions rather than exploring distant areas of the search space. Fig. 2 illustrates this local search behavior. There are four main factors considered in the seeking mode of each cat:

1) Search Memory Source (SMP): This factor determines the size of the search memory for each cat, representing the positions previously searched by the cat. Based on the fitness functions, the cat chooses one position from its memory as a potential candidate for movement.

2) Number of Dimensions that Change (CDC): This factor determines the number of dimensions in the current position of the cat that will be modified during the search process.

3) Search in the Defined Range of Dimensions (SRD): This factor indicates the rate of change for the selected dimensions. If a dimension is chosen for modification, the difference between the new and old values will be within the range defined by SRD.

4) Attention to the Current Position (SPC): This factor is a Boolean variable that determines whether the current position of the cat is considered a candidate for movement or not. It ensures that the current position, which has already been explored, is not included in the search memory (SMP).

By considering these factors, the seeking mode of the cat swarm optimization algorithm focuses on local search, efficiently exploring and refining solutions within the vicinity of the current positions of the cats.

The seeking mode algorithm, as depicted in Fig. 3, outlines the steps followed when a cat is in seeking mode [35]. The algorithm proceeds as follows:

1) Create \( j \) copies of the current position of cat \( k \) (Cat\(_k\)), where \( j \) is equal to SMP (Search Memory Source). If the value of SPC (Attention to the position) is true, then \( j \) is adjusted to (SMP-1), allowing the current position to be considered one of the candidates.

2) For each copy, based on the CDC (Number of Dimensions that Change), increase the SRD (Search in the defined range of dimensions) by a percentage of the current values and add it to the previous values. This step determines the range within which the selected dimensions can change during the search.

3) Calculate the fitness function (FS) for each candidate point generated in the previous step.

4) If not, all FS values are exactly equal, proceed to calculate the probability of selecting each candidate position based on the normalized fitness (fit) of that position. If all FS values are equal, set the probability of selecting all candidate positions to be equal.

By following this algorithm, the seeking mode effectively explores the space by creating multiple copies of the current position, determining the range of changes in dimensions, evaluating the fitness for each candidate point, and selecting the next position based on the calculated probabilities.
Fig. 2. Seeking mode.

- Create \( j \) copies of the \( k \)-th cat, represented by SMP (Search Memory Source).
- Modify the CDC (Number of Dimensions that Change) dimensions of each copy randomly, introducing variations in the solution space.
- Evaluate the fitness of each copy by assessing its cost or quality based on the fitness function.
- Identify the best solutions among all the copies, which correspond to the mappings with the minimum cost.
- Randomly select one solution from the best solutions and replace it with the current position of the \( i \)-th cat, updating its state.

Fig. 3. Seeking mode algorithm.

B. Tracing State

The tracking mode algorithm, which describes the behavior of the cat when tracking the target, can be outlined in the following three steps:

1) Calculate the new position of the cat based on its current position and speed in each dimension. The cat moves in the search space according to its velocity, exploring different locations.

2) Evaluate the fitness of the new position using the fitness function. This assesses the quality or cost associated with the new mapping.

3) Update the cat's position to the best position found during the tracking process. The cat moves towards the position that yields the minimum cost or maximum quality, improving its mapping.

By following these steps, the cat in tracking mode continuously adjusts its position based on its velocity, evaluates the fitness of the new position, and updates its location to the best position encountered during the tracking process. This allows the cat to gradually converge towards an optimal solution by iteratively exploring and refining its mappings.
In the first step, the speed synchronization for each dimension \((v_{k,d})\) is calculated according to the following relationship:

\[
v_{k,d} = v_{k,d} + r_1 \times c_1 \times (x_{\text{best},d} - x_{k,d})
\]

where \(d = 1,2,\ldots,M\) and \(x_{\text{best},d}\) shows the position of the cat that has the highest value of fitness and \(x_{k,d}\) is the position of the \(k\)th cat. \(c_1\) is a fixed number, and \(r_1\) is a random number in the interval \([0,1]\). In the second step, it is checked that the speeds are within the defined range. If there were a higher speed, it would be replaced with the maximum possible value in the desired range, and finally, in the third step, the position of the cat will update according to the following relationship:

\[
x_{k,d} = x_{k,d} + v_{k,d}
\]

In fact, TS is for Teaching-Learning-Based Optimization. During this phase, the cats aim to exploit the information about the best position found so far to reach the optimal solution. Even though a cat's position change may be large during this phase, the search is still focused on the best solution found so far. Please refer to Fig. 4 for an illustration.

In other words, at this stage, the set of answers obtained from the best location of the cat in the current iteration is updated. As seen, cat swarm optimization uses two sub-models of seek and tracking mode; the way to combine these two sub-models to perform scheduling operations is described in the next section.

**C. Scheduling**

The optimal utilization of available resources is the main goal of task scheduling that provides a basis for load balancing. In order to reduce the cost of services provided to users, fog service providers adopt different policies depending on the type of user and the desired services. However, the aim of this study is to reduce the resource consumption cost. Thus, parameters such as energy consumption and traffic consumption are considered. These parameters are formulated based on the following equations to be used in the CSO-based approach.

**D. Modelling Energy Consumption**

In order to model the amount of energy consumed, the proposed approach in [20, 36, 40] is used. The idea of this model is based on the fact that there is a linear relationship between processor efficiency and power consumption. In other words, if we know the processing time of a task and the efficiency of the processor, we can calculate the energy consumed by that task. This means that by understanding how long a task takes to complete and how efficiently the processor performs it, we can determine the energy usage associated with that task.

The efficiency of a given resource \(r_j\) in a given time is calculated as follows:

\[
U_j = \sum_{i=1}^{n_r} u_{ij}
\]  

(1)

Here, \(n_r\) represents the number of tasks currently in progress, and \(u_{ij}\) represents the amount of resources consumed by task \(t_j\). Accordingly, the energy consumption \((E_j)\) of resource \(r_j\) in a given time can be calculated using the following formula:

\[
E_j = (P_{\text{max}} - P_{\text{min}}) \times U_j + P_{\text{min}}
\]  

(2)

**E. Traffic Consumption**

This section aims to model the network's bandwidth usage and the volume of traffic produced by individual physical machines.

\[
D_j = \sum_{r \in \mathcal{V}} \lambda(j,m) \sum_{l=1}^{n_{\mathcal{V}}} c_l
\]  

(3)

Where \(\lambda(j,m)\) represents the traffic load on the link \((j,m)\), \(\mathcal{V}\) is the set of all resources, and \(c_l\) is the bandwidth capacity of link \(l\).
D_j represents the communication between physical machine j and other physical machines. λ(j, m) denotes the traffic load between physical machines j and m. V_j indicates a set of physical machines connected to physical machine j. C_i represents the weight of the communication link between two physical machines at level i. ρ(m, j) represents the communication level between physical machines m and j.

The multi-objective cost function for modeling the fog computing environment is formulated based on these two parameters as follows:

\[
\text{Minimize } \sum_{j=1}^{m} E_j \quad (4)
\]

\[
\text{Minimize } \sum_{j=1}^{m} D_j \quad (5)
\]

The minimum cost for task scheduling is obtained using the above functions. In the following, it is described how to combine two states tracking and seeking:

1) The initial population consists n cats.
2) The cats in this scenario are distributed randomly within a d-dimensional search space by assigning each cat a random speed for each dimension within a given range.
3) A number of cats are randomly selected. According to the Mixture Ratio (MR), some cats are put in the searching state, and the remaining ones are considered for the tracking state.
4) The movement of cats is based on their flag value which shows the tracking or seeking states.
5) The fitness value of each cat is evaluated, and the position of the best cat (Xbest) is recorded and stored.
6) The situation of cats is updated according to their position. Then, step 3 is repeated.
7) Steps 5 and 6 are repeated iteratively until the termination criterion is met, i.e., a complete task scheduling is reached.

The pseudo-code for scheduling based on CSO is depicted in Fig. 5.

It is important to note that while executing scheduled commands on processing servers using cat swarm optimization, it is possible for some servers to fail to execute the commands or experience additional traffic and load due to delays in processing previous commands or hardware and software errors. Meanwhile, some servers may remain idle after completing their assigned commands. To address this, the resources of processing servers are periodically monitored, and the following three conditions are used to determine the status of their resources:

- If the resource consumption of a server has exceeded 95% of its maximum efficiency, it is categorized as an overloaded machine.
- If the resource consumption of a server is lower than the average efficiency determined (usually in the range of 10%), it is categorized as a server with a low load.
- Servers that do not meet the above conditions are considered normal servers.

![Fig. 5. Pseudo-code of algorithm.](image-url)
Now to balance the load, the following load-balancing strategies can be applied:

- Servers in overloaded mode: Loads of these servers can be migrated to other servers in the $P_{\text{under}}$ (low load) or normal mode. This can be achieved through the virtual machine migration technique, where the tasks and loads are transferred to another server that has sufficient resources to accommodate them.

- Servers with an efficiency below 10%: These servers can be considered candidates for task migration and subsequent shutdown. By migrating their tasks to other servers, the load on the underutilized servers can be increased, and energy consumption can be reduced by shutting down the inefficient servers.

By applying these load-balancing strategies, the goal is to optimize resource utilization, minimize overload conditions, and reduce energy consumption in the system.

IV. EVALUATION

The implementation and evaluation of the proposed approach require a set of programming tools. These tools are necessary and useful for implementing different scenarios and evaluating the results. The comparison of the proposed approach with other existing techniques can also be made using these tools. For this purpose, the iFogSim2 simulator is used in this study [21, 37]. In order to evaluate the performance of the proposed approach, it is compared with metaheuristic algorithms such as PSO, ACO, Genetic (GA) and Random (RND) scheduling algorithms. In this study, the workload of simulations in the real traced data of the CoMon project is used, which is a monitoring infrastructure for PlanetLab [22, 38]. The main attributes of the cat swarm optimization algorithm are illustrated in Table II.

A. Evaluation Results

The outcomes of the assessment are depicted in Fig. 6 to 10. The initial evaluation encompassed three distinct experiments involving varying numbers of virtual machines. The objective was to explore how algorithms operate with diverse virtual resources and examine the correlation between resource quantity, execution time, and energy consumption. The subsequent results are presented below. Fig. 6 illustrates the energy consumption for all three solutions corresponding to the number of available virtual resources.

<table>
<thead>
<tr>
<th>TABLE II. MAIN CHARACTERISTICS OF THE CSO ALGORITHM</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Decision-making variables</strong></td>
</tr>
<tr>
<td>Solution</td>
</tr>
<tr>
<td>The previous solution</td>
</tr>
<tr>
<td>New solution</td>
</tr>
<tr>
<td>Better answer</td>
</tr>
<tr>
<td>Initialization of answers</td>
</tr>
<tr>
<td>How to create a new answer</td>
</tr>
</tbody>
</table>

Fig. 6. Energy consumption in different tests.
As depicted in Fig. 6, an increase in the number of virtual machines leads to a rise in energy consumption across all three solutions. However, the proposed method consistently exhibits lower energy consumption at each stage. Although the optimality rate was initially low in the first test, it improved in subsequent evaluations as the number of available resources increased. This improvement was facilitated by the optimal arrangement of virtual machines achieved through CSO scheduling. Consequently, the proposed solution outperforms other methods with a significantly higher optimality rate. Moving forward, Fig. 7 evaluates the implementation time of the solutions in three distinct modes.

As depicted in Fig. 7, the execution time of the solutions increases with the growth in the number of virtual machines. However, the proposed method effectively addresses this issue through optimal resource allocation, the precise arrangement of virtual machines, and achieving load balancing. Consequently, the proposed solution outperforms the others by completing tasks in a shorter period in all three tests. The optimization carried out with the assistance of the CSO algorithm enables the identification of the best available nodes, resulting in the allocation of resources to the most suitable machines. As a result, both the execution time and energy consumption are reduced.
The next test was done with a constant amount of virtual and available resources, as well as applying a workload based on the CoMon project [38, 39]. The results of the evaluation of all algorithms are shown below. First, in Fig. 8, the amount of energy consumption of the proposed solution is shown in comparison with the other solutions.

Fig. 8 clearly illustrates a significant reduction in energy consumption within the proposed solution compared to other algorithms. This achievement can be attributed to the allocation and optimal arrangement of resources facilitated by the enhanced cat swarm optimization algorithm. In the proposed solution, the process of identifying available nodes has been optimized using the algorithm, and resource allocation is performed based on the status of these available resources. Consequently, energy consumption is minimized through the creation of optimal scheduling, alongside a reduction in execution time.

By adopting the proposed approach, notable improvements can be observed. For instance, in comparison to the RND-based solution, energy consumption has been reduced by approximately 24%. Similarly, compared to the PSO algorithm, the proposed solution demonstrates a reduction of approximately 10% in energy consumption. These results highlight the effectiveness and efficiency of the proposed method in optimizing energy consumption. Due to the fact that the amount of available resources is checked in every resource allocation operation, this energy reduction was predictable. In the following Fig. 9, the implementation time of the solutions is shown.

Indeed, in fog computing, minimizing the execution time of requests is crucial alongside reducing energy consumption. As depicted in the diagram in Fig. 9, the proposed solution exhibits a remarkable reduction in execution time compared to alternative approaches. Specifically, compared to schedules based on the PSO and ACO algorithms, the proposed solution achieves a reduction of approximately 21%. Furthermore, in comparison to the RND-based schedule, the execution time is reduced by approximately 33%. These results highlight the significant gains in efficiency and speed achieved through the proposed solution, further enhancing the benefits of fog computing.

This significant reduction in execution time can be attributed to the scheduling and optimal resource arrangement, as well as the establishment of load balancing through the proposed solution based on the cat swarm optimization algorithm. The solution efficiently utilizes the processing servers' resources by considering their status and allocating resources accordingly. Additionally, Fig. 10 presents the level of violation of the service level agreement (SLA).

The service level agreement (SLA) serves as the framework for establishing the expected level of service. It defines the formal conditions for the provided service, including aspects such as performance and availability. Fig. 10 illustrates that the proposed solution, through effective load balancing, enhances the reliability and availability of servers within the fog computing infrastructure compared to alternative solutions. In other words, the proposed solution exhibits a lower percentage of SLA violations, indicating its superior ability to meet the agreed-upon service level requirements.

![Fig. 9. Execution time (millisecond).](image-url)
V. CONCLUSION

This research introduces an approach based on the CSO algorithm to achieve optimal scheduling and load balancing. The proposed technique aims to prevent server overload or underload by efficiently allocating tasks to physical servers. To enhance the performance further, the CSO algorithm is extended by incorporating new parameters, including the amount of available resources and network traffic. By considering these factors, the proposed approach minimizes the execution time of requests by appropriately distributing tasks among candidate servers. Finally, the proposed approach is implemented and simulated in iFogSim, allowing for a comparison with other algorithms such as PSO, ACO, GA, and Random. The results obtained from the simulation demonstrate that the proposed approach successfully achieves its objectives and surpasses the other methods in terms of execution time and energy consumption. This highlights the effectiveness and superiority of the proposed approach in optimizing these important performance metrics. It could improve the energy consumption compared to ACO and PSO by the values of 24% and 10%, respectively.

Additionally, the execution time is significantly reduced, and in compared to ACO and PSO, the proposed approach could improve the execution time by 21% and 22%, respectively. The observed improvement can be attributed to the load balancing achieved in the fog environment through the combined utilization of the CSO-based scheduling method and virtual machine migration technique. This combination effectively distributes tasks and resources across the fog network, ensuring optimal utilization and minimizing resource imbalances. As a result, the proposed approach enhances load balancing, leading to improved performance in terms of execution time and energy consumption. The results also show that the proposed approach has less violations in SLA when compared to the other algorithms and consequently provides more reliable servers with higher availability in the fog infrastructure.

VI. FUTURE WORK

The proposed approach can be applied to Content Delivery Networks (CDNs). Therefore, the CSO-based approach enables the identification and replication of the best content within alternative servers. By leveraging the capabilities of the CSO algorithm, the proposed approach effectively identifies the optimal servers to store and replicate content. This ensures that content is efficiently distributed across multiple servers, enhancing availability, fault tolerance, and overall system performance. In fact, by generalizing this approach and using the tracking mode of cats, the best contents can be obtained to be cached and replicated in servers found in the seeking mode. As a result, the performance of such cloud servers can greatly improve.

REFERENCES


A Hybrid Cryptography Method using Extended Letters in Arabic and Persian Language

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Abstract—Cryptography is widely used in information security systems. In encryption, the goal is to hide information in such a way that only the sender and receiver are aware of the existence of communication and information. Encryption takes place in various media, such as image, sound and text. Today, the rapid growth of network technologies and digital tools has made digital delivery fast and easy. However, the distribution of digital data in public networks such as the Internet has various challenges due to copyright infringement, forgery, coding and fraud. Therefore, methods of protecting digital data, especially sensitive data, are very necessary. Accordingly, in this article, a combined solution is used based on the technique of stretching the letters and making minor changes in the letters that have closed spaces so that the bits related to the hidden text can be inserted into a Persian or Arabic language. For this purpose, a new solution has been designed, in which the cover text is similar to the normal text, with the difference that, in addition to the extended letters that are longer due to the status of the secret message, it also has some prepositions, which have spaces. They are empty and closed. Of course, this difference in the closed space between the original letters and the changed letters will be very slight, and as a result, there will not be much difference between them that the normal user can feel the change. Finally, the proposed solution has been evaluated with the help of the MATLAB program, and according to the rate parameter of encryption capacity, the results show that the proposed method has an average encryption capacity of more than 50% compared to other common solutions.

Keywords—Cryptography; extended letters; Persian language; Arabic language

I. INTRODUCTION

Text-in-text encryption is used for the purpose that the host data does not have any visible signs of the presence of encrypted data in it, and thus it enables the secure transmission of information [1]. Therefore, the way to hide the data is important according to the changes that are made in the external details of the text. But what is of secondary importance is to protect the transmission against widespread attacks on the data sent and, of course, to make the original data unreachable in case the attackers discover the existence of a password in the host data. In fact, the advantage of using text as a cover media is that the text file consumes less storage space and less bandwidth is required for its transmission [2, 29, 30]. In order to encrypt a text, watermarking methods can be used in the text image, shifting the words in the middle space between them or shifting the background line to hide the data of the desired series in the background text, but this method which is considered in this research is the use of making changes in details. The majority of steganography research employs various types of cover media, such as images [3, 31], video clips [4, 32], and audio [5, 33]. Nevertheless, text steganography is not commonly favored due to the challenges associated with identifying redundant bits in text files [6, 7, 34]. Text documents typically exhibit a structure similar to what is visible, while other types of cover media have different structures, making it easier to hide information without noticeable alterations. However, text steganography offers advantages like lower memory usage and simpler communication.

The main goal of this project is to provide a new method for hiding information in Persian and Arabic texts. The choice of steganographic system often depends on the language and its structures. One technique cannot be universally applied to all languages. In the proposed method, it has been tried to avoid making changes in the appearance and format of the font as much as possible so that while increasing the capacity, the level of transparency and readability of the text does not change significantly compared to the state before hiding. So in this method, the technique of stretching Persian and Arabic letters is used in addition to creating changes in the letters that have closed empty space, which includes the letters: "ص،ض،ط،ظ،ع،ؽ،ف،ق،و،ٔ،ِ". "ض،ض،ط،ظ،ع،ؽ،ف،ق،و،ٔ،ِ" covert writing of information in Persian and Arabic texts is discussed. Of course, for the letters "ع، ؾ" only when these letters are used in the middle of the word, they have a closed space. Therefore, in this method, only one position of the letters "ع، ؾ" (in the middle of the word) can be used to hide the information in the letters "ع، ؾ".

This paper is organized as follows. Section I provides an overview for text steganography. Section II discusses the related works. Section III presents our new text steganography technique that utilizing character extension. The evaluation and results are discussed in Sections IV and V, respectively and finally, conclusion and future work are outlined in Sections VI and VII.

A. Cryptography Basics

According to the type of application, several characteristics are considered to express the efficiency of cryptographic methods, which are different in different applications. The important things considered in the design of this method are as follows [3]:

- Transparency: In the encryption operation, it is necessary to hide the information in the desired text in such a way that the inserted information is not visually clear and understandable. This amount of similarity, before and after inserting information, is called transparency.
• Resistance: In a cryptography solution, resistance means against unwanted and unintentional changes that are created due to the presence of noise in the transmission channel or other changes that are intentionally made by an attacker, and it is done in the direction of changing the message so that it has the necessary resistance to a large extent.

• Capacity: The amount of information that can be included in the cover text is called capacity. The methods that are presented for the purpose of encryption should be designed in such a way that they are able to hide a significant amount of information in the cover text.

Considering that in order to encrypt texts, letters and characters need to be converted into a set of numbers 0 and 1 so that they are easy to manage [8, 9, 35]. Therefore, there must be a single standard to determine what characters each of these numbers and letters should display and how they should be stored. This standard is called Unicode.

Converting data so that the system can read and use it is called encoding. In fact, encoding is the process of converting data into a format required for a number of information processing needs, including:

- Data encryption
- Formulation of the program and its implementation
- Data transfer, storage and compression/decompression
- Application data processing, such as file conversion

For the coding standard mentioned above, the American Standards Association introduced a 7-bit coding method called ASCII in 1960. At that time, the ASCII character set, including 128 characters (7 bits), was defined, mostly for Latin languages. ASCII encoding is important because many communication tools and protocols only accept ASCII characters. In fact, this is the accepted minimum standard for the text. Some Unicode encodings, due to the universal acceptance of ASCII, convert their code points into a series of ASCII characters so that they can be moved without any problems. In the 1980s, it was decided to use a full byte (i.e., 8 bits) for encoding in the ASCII character set instead of 7 bits [10, 36]. Therefore, the number of characters reaches 256. Based on this, the characters after 127 to 255 were also considered reserved codes, and other languages, including Persian and Arabic, were generally included in this range. But in this range between different languages, there was no single standard, and each language showed its alphabet code. In other words, code 200 in one language returns a different letter in another language. As a result, there was a need for a single standard to consider unique codes for each character while being compatible with all languages. Based on this, the Unicode unit standard has been presented. Unicode is a set of characters with unique numbers, which is called a code point. Each code point represents a single character. Accordingly, the Unicode standard specifies three encoding methods, allowing a character to be encoded within one or more bytes (i.e., in 8, 16, or 32 bits). These coding methods are as follows [11, 37]:

- UTF-16
- UTF-8
- UTF-32

The difference between these coding methods is in the way letters, numbers and symbols are presented between the languages of different countries. So that the way characters are presented in one country is different from another country [12, 38]. Table I shows the Unicode standard for Persian and Arabic letters, along with the code of each letter in this standard. In cryptography, this table is used to convert each letter to its corresponding value of 0 or 1.

<table>
<thead>
<tr>
<th>Form</th>
<th>letter name</th>
<th>Code</th>
</tr>
</thead>
<tbody>
<tr>
<td>ت</td>
<td>letterت</td>
<td>062A</td>
</tr>
<tr>
<td>ث</td>
<td>letterث</td>
<td>062B</td>
</tr>
<tr>
<td>ج</td>
<td>letterج</td>
<td>062C</td>
</tr>
<tr>
<td>د</td>
<td>letterد</td>
<td>0686</td>
</tr>
<tr>
<td>ذ</td>
<td>letterذ</td>
<td>0630</td>
</tr>
</tbody>
</table>

II. RELATED WORK

Numerous research studies have explored concealing one text within another, employing diverse linguistic techniques. These studies classify linguistic steganography into two primary categories: syntax-based and semantic-based approaches [13][14][39].

For instance, [15] and [16] proposed a method based on synonyms, aimed at data concealment. The method consists of two phases: in phase one converting the hidden message into binary codes. In phase two, with the help of a synonyms file, both sender and recipient must possess the same word list for encryption and decryption. If the sender inserts a Zero, no word replacement is necessary. However, if another value is inserted, the synonym file serves as the basis for word replacement. This process continues until the end of the secret message, and the receiver can decrypt the message using an inverse strategy.

In [17] proposed method that takes three input sources (secret message, natural language, and the key) and produces one output known as the Stego-text. The system identifies each word and its corresponding group, generating lexical replacement groups and variant forms of similar words. To ensure proper embedding of the correct word in the carrier file while considering the context, a lexical analyzer for the Chinese language was utilized in the proposed system. The steganography algorithm, based on three inputs representing the source natural language text, the information to hide, and
the key, generates one output displaying the stego-text with embedded data. The steps of the steganography algorithm include:

- Embedding data: Encrypting the information into a binary bit sequence using the key (embedded data).
- Text preprocessing: Segmentation of sentences using the Chinese lexical analyzer ICTCLAS, where English characters, Chinese and English punctuation, blank spaces, and new line characters are considered inter-sentence symbols.

In [18], a novel method was proposed to hide information not only in specific pointed letters but in any letters. The researchers utilized pointed letters with an extension to represent the secret bit ‘one,’ and un-pointed letters with an extension to hold the secret bit ‘zero.’ It was emphasized that letter extensions do not affect the writing content and are considered redundant characters for formatting purposes in Arabic electronic typing. However, not all letters can be extended due to their positions in words and the nature of Arabic writing. Extensions can only be added between connected letters, not at the end of words or before the beginning letter. Thus, letters without extensions or intentionally without extensions are considered not to hold any secret bits.

In [19], authors introduced a new approach for hiding information by manipulating white spaces between words and paragraphs. The method offered greater capacity to conceal more bits of data within a cover-text. While the previous embedding scheme used the space between words, it required a significant amount of space to encode a few bits. However, combining inter-word and inter-paragraph spacing allowed effective utilization of most white spaces in a text document, increasing the data-hiding capacity.

In [20], the authors built upon a previous Arabic text steganography method using letter points and extensions to address the low-capacity aspect. They proposed a technique to hide information within suitable positions inside words, not limited to pointed letters only. These positions were carefully determined to preserve the beauty of Arabic text if justified, ensuring that the message could be hidden without affecting the cover text. Extensions were inserted at the specified positions to represent the secret bit ‘one,’ while leaving the positions empty represented the secret bit ‘zero.’

Another study [21] introduced the Line Shifting method, which involved shifting lines vertically to pass data via the carrier. However, this method had weaknesses, including the potential detection of line shifting when using character recognition programs and the risk of hidden data destruction when retyping the carrier file.

In [22], researchers adopted a syntactic method based on punctuation to convey the secret message. They added punctuation in suitable locations to hide the data without impacting the carrier message's meaning or affecting the embedded data within it. The study highlighted the importance of finding optimal trade-offs between bit rates, robustness, and perceivability through experimental definitions.

In [23], the researchers used emoticon-based steganography to facilitate secret chatting. They classified emoticons semantically and controlled the symbol order using a secret key to pass the secret message among parties. Different groups of emoticons were created to hide data, and the order of passing a symbol represented specific bit values. This method proved to be robust and beneficial for chat systems, with some systems allowing users to generate their customized symbols.

III. PROPOSED METHOD

In this research, a method of hiding in the text is presented so that by using it, confidential texts can be hidden inside the text. For this purpose, a combined solution is used based on the technique of stretching the letters and making minor changes in the letters with a closed, empty space so that the bits related to the hidden text can be hidden inside a normal hidden text. He made a drawing. For this purpose, a new solution has been designed, in which the cover text is similar to the normal text, with the difference that, in addition to the letters that are longer due to the status of the secret message, it also has some prepositions, which have spaces. They are empty and closed. Of course, this difference in the closed space between the original letters and the changed letters will be very slight, and as a result, there will not be much difference between them that the normal user can feel the change [24].

As shown in Fig. 1, the proposed encryption solution will have two inputs. One of its entries will be related to the cover text used for encryption, and the other entry will be the secret message that must be hidden inside the cover text. The proposed solution in the form of a new font acts as a converter and embedder, which can be used to change the cover text in such a way that the secret text can be hidden inside it. This solution will have the following features:

- High capacity to encrypt confidential text.
- High security due to very minor changes.
- High power in encryption.
- High transparency due to minor changes in font.

In fact, through the use of this combined solution, the capacity of confidential messages that must be encrypted can be increased because many Persian and Arabic letters are stretchable and have closed spaces. On the other hand, due to the fact that the changes in the letters are minor, normal users will not be able to recognize the desired changes, and as a result, security is established. Also, this solution is able to be resistant to the factors that can change the data and, as a result, damage the comprehensiveness because the encryption operation is carried out bit by bit according to the state of the letter. The original data is encrypted. On the receiver's side, there is a need to perform an operation in the direction of the photo, and for this purpose, a special tool is provided to detect the changed letters and, as a result, the hidden secret message. As a result, in this case, according to the status of each letter and its length, the main message can be extracted from the cover text, and there is no need for a private key for data exchange. Because in encryption algorithms, the encryption key must also be transmitted, and if for any reason this key is intercepted or discovered, all the encrypted data can be
accessed, and as a result, the security and integrity of the data will be compromised [25].

A. How the Method Works

In Arabic and Persian languages, each character can have a different state according to its position. In fact, in this category of languages, a number of letters have the ability to change their shape according to the type of connection that they are at the beginning, end, or middle of a word, which is an important ability for encryption. For example, the letter “ع” at the beginning of the word is written as "مع" in the middle as "مها" and at the end of the word as "تع" while in English words, each letter is written separately and does not have this feature. Fig. 2 shows a section of different states of Persian letters.

According to the Unicode standard, each letter can have a unique code according to its position. On the other hand, in some cases, in order to make a letter more beautiful, it can also be written with a stroke. For example, the word “خانه”, which consists of four letters "خ", "ن", "ه" and "ه" can be written as "خانه" without changing its meaning or the number of letters. In fact, stretching the letters is one of the advantages of Persian and Arabic languages; this ability does not have the slightest effect on the readability or changing the meaning of the text [26]. The only problem is that not all letters have this ability, and it is limited to the position in which that letter is placed; that is, in general, the ability to stretch can be applied to the connecting letters of a word, and this feature cannot be applied to a number of letters that are at the end or beginning of a word [12]. Accordingly, in this research, a combined solution has been used, which solves this limitation to a large extent. In the following, the method of using the ability to stretch the letter is shown with an example. At first, let’s assume that we intend to perform the encryption process with the help of dragging on the text of Table II. First, we select the secret bits that should be hidden, which in this example is equal to the value (110010). The encryption operation is performed from the least valuable bits to the most valuable ones. Also, considering that the texts are Persian and Arabic and start from right to left; as a result, encryption is done from right to left. Accordingly, the first secret bit is equal to "0", and considering its value is zero, the letter to be hidden in it is extended once, but if it were one, it would be extended twice. Now, in this example, encryption should be done by stretching the letter "م" once. The value of the next secret bit is equal to "1", and the second letter of this word is "ن" considering that this letter is at the end of the word, so it cannot be extended in this position. The next point from which the operation can be continued is at the beginning of the word Turkey, and considering that the secret bit is equal to "1", as a result, the letter "ت" is doubled and changed to "جـ". Find this operation is carried out confidentially until the end of data encryption.

![Fig. 1. The general suggested method.](image1)

![Fig. 2. Types of Persian letter forms.](image2)
TABLE II. THE METHOD OF ENCRYPTION USING THE ABILITY TO STRETCH LETTERS

<table>
<thead>
<tr>
<th>Confidential Bits</th>
<th>110010</th>
</tr>
</thead>
<tbody>
<tr>
<td>Cover text</td>
<td>من ترکیه را دوست ندارم</td>
</tr>
<tr>
<td>Hidden text</td>
<td>من ترکیه را دوست ندارم</td>
</tr>
</tbody>
</table>

Now, in order to increase the efficiency of the solution, changes can be made in the letters with closed spaces so that the capability of this solution increases. The letters with closed space include the characters "ص،ض،ط،ظ،ع،ؽ،ف،ق،و،ٔ،ِ". Of course, for the letters "ع،ؽ" only when these letters are used in the middle of the word will they have a closed space; therefore, in this method, only one position of the letters "ع،ؽ" (in the middle of the word) can be used to hide the information in the letters "ع،ؽ". Accordingly, in order to hide these letters, the amount of empty space is slightly smaller so that it has the least effect on changing the font of the desired letters and through this change if the last letter of a word cannot be it is drawn, it is part of the letters with closed space, it is used. As a result, if the insertable bit is zero, the character will remain unchanged. But if the insertable bit was 1, the character is inserted with a smaller enclosed space. In this case, the complete flowchart of the proposed solution is shown in Fig. 3.

IV. EVALUATION

MATLAB 2018 program was used to implement the solution. MATLAB has considered two environments, GUIDE and App Designer, for designing graphic interfaces. The GUIDE environment has been used since the old versions of MATLAB, but the App Designer environment has recently been presented in the new versions of MATLAB and is being developed. Based on this, the user interface related to the proposed solution has been designed with the help of the advanced App Designer tool so that by using it, it is possible to receive cover text and secret text and carry out operations related to encryption or secret text extraction. Fig. 4 shows a view of the program environment implemented in MATLAB.

Based on this and as seen in Fig. 4, the user enters the cover text in the relevant box and then, in the next box, the confidential text that he intends to hide. Then, by clicking the hide button, the confidential text will be hidden inside the cover text according to the proposed solution. An example of the implementation of the solution is shown in Fig. 2 to 4.

The part related to the recovery of encrypted text is also shown in Fig. 5. For this purpose, it is sufficient for the user to click on the button to copy the encrypted text after performing the encryption operation. In this case, the page corresponding to Fig. 6 will open. Now the user clicks on the "decode" button so that the encrypted text will be shown along with the original cover text.
Fig. 4. A view of the user interface of the implemented program.

Fig. 5. How to hide the text by the solution.
A. Evaluation Parameters

In order to measure the proposed solution, a series of parameters are needed for review and evaluation. According to the studies that have been done on the articles written in the field of cryptography of Arabic and Persian texts, the proposed solution has been examined according to the parameter of cryptography capacity. The following relationship is used to calculate the hiding capacity:

\[ \text{CapacityRatio} = \frac{\text{hidden text size}}{\text{covered text size}} \]

1) The solutions to be evaluated: In order to check the effectiveness of the solution, it is necessary to compare it with other common methods in the field of text encryption. Accordingly, in this research, the proposed solution is compared and evaluated with the following solutions, which were also examined in the research conducted in [27, 40]:

- Enhanced Kashida [3]
- ZWC and space [8]
- Enhanced capacity [9]
- Pseudo-space [10]
- Alhusban's method [12]
- Moon and Sun letters [24].

V. RESULTS

Considering that only Arabic texts were used in the evaluation of the solutions mentioned in the previous section, as a result, in order to have the same data in the proposed solution, the same Arabic texts with the same size determined in the evaluated solutions were used. It has been done so that the checks and comparisons are done in a standard and uniform way. First, in order to compare the output of the solutions in Tables III and IV, an example of the output of the solutions has been implemented according to the number of different words. The purpose of this evaluation is to check the output of text encryption in each of the solutions.

Also, in Table IV, another example of the application of encryption solutions is shown on a cover text with a length of 155.

As can be seen in the output of Tables III to VIII, in all three evaluations, the proposed solution has a higher encryption capacity compared to other methods, and with the increase in
the length of the cover text, this amount has become more than optimal. The reason for this is the use of two techniques in the proposed solution. So that at first, using the technique of stretching the letters, the encryption operation was done, and as can be seen, this feature did not have the slightest effect on the readability or changing the meaning of the text. In addition, the utilization of the closed space available in Persian and Arabic letters has made the ability of the solution for encryption higher and the more the number of these letters, the higher the encryption capacity. Also, as it is clear from the results of these evaluations, in the case where the proposed solution, only the encryption technique based on the closed letter space is used, the encryption capacity has decreased to a great extent. Based on this, the results of this evaluation indicate that the use of the extended technique along with closed letters can increase the cryptography capacity in a more effective way.

**TABLE III. COMPARISON OF THE OUTPUT OF THE IMPLEMENTATION OF 8 CRYPTOGRAPHY TECHNIQUES ON THE SAME ARABIC TEXT WITH THE COVER TEXT OF LENGTH 109**

<table>
<thead>
<tr>
<th>Output</th>
<th>Method</th>
</tr>
</thead>
<tbody>
<tr>
<td>Yecht علم الحاسوب استخدام الموسيقية يجمع الكشيدات لحل المشكلات من متغيرات علمي رياضي وعلمي ما يتم ذلك تصميم</td>
<td>Enhanced Kashida</td>
</tr>
<tr>
<td>Yecht علم الحاسوب استخدام الموسيقية يجمع الكشيدات لحل المشكلات من متغيرات علمي رياضي وعلمي ما يتم ذلك تصميم</td>
<td>ZWC and space</td>
</tr>
<tr>
<td>Yecht علم الحاسوب استخدام الموسيقية يجمع الكشيدات لحل المشكلات من متغيرات علمي رياضي وعلمي ما يتم ذلك تصميم</td>
<td>Enhanced capacity</td>
</tr>
<tr>
<td>Yecht علم الحاسوب استخدام الموسيقية يجمع الكشيدات لحل المشكلات من متغيرات علمي رياضي وعلمي ما يتم ذلك تصميم</td>
<td>Pseudo-space</td>
</tr>
<tr>
<td>Yecht علم الحاسوب في استخدام الموسيقية يجمع الكشيدات لحل المشكلات من متغيرات علمي رياضي وعلمي ما يتم ذلك تصميم</td>
<td>Method 2</td>
</tr>
<tr>
<td>Yechت علم الحاسوب في استخدام الموسيقية يجمع الكشيدات لحل المشكلات من متغيرات علمي رياضي وعلمي ما يتم ذلك تصميم</td>
<td>Alhusban’s method</td>
</tr>
<tr>
<td>Yechت علم الحاسوب في استخدام الموسيقية يجمع الكشيدات لحل المشكلات من متغيرات علمي رياضي وعلمي ما يتم ذلك تصميم</td>
<td>Moon and Sun letters</td>
</tr>
<tr>
<td>Yechت علم الحاسوب في استخدام الموسيقية يجمع الكشيدات لحل المشكلات من متغيرات علمي رياضي وعلمي ما يتم ذلك تصميم</td>
<td>Proposed method</td>
</tr>
</tbody>
</table>

**TABLE IV. COMPARISON OF THE OUTPUT OF THE IMPLEMENTATION OF 8 CRYPTOGRAPHY TECHNIQUES ON THE SAME ARABIC TEXT WITH THE COVER TEXT OF LENGTH 155**

<table>
<thead>
<tr>
<th>Output</th>
<th>Method</th>
</tr>
</thead>
<tbody>
<tr>
<td>تنقسم شكل النص إلى قسمين دراسة جسد الإنسان والبحث لتعزيز معرفتنا بالألفاظ التي يمثل بها الجسم الحي ومثلا الشمعة وعند الصداقة التقليدي الذي يتم تطبيق هذه المعادلة</td>
<td>Enhanced Kashida</td>
</tr>
<tr>
<td>تنقسم شكل النص إلى قسمين دراسة جسد الإنسان والبحث لتعزيز معرفتنا بالألفاظ التي يمثل بها الجسم الحي ومثلا الشمعة وعند الصداقة التقليدي الذي يتم تطبيق هذه المعادلة</td>
<td>ZWC and space</td>
</tr>
<tr>
<td>تنقسم شكل النص إلى قسمين دراسة جسد الإنسان والبحث لتعزيز معرفتنا بالألفاظ التي يمثل بها الجسم الحي ومثلا الشمعة وعند الصداقة التقليدي الذي يتم تطبيق هذه المعادلة</td>
<td>Enhanced capacity</td>
</tr>
<tr>
<td>تنقسم شكل النص إلى قسمين دراسة جسد الإنسان والبحث لتعزيز معرفتنا بالألفاظ التي يمثل بها الجسم الحي ومثلا الشمعة وعند الصداقة التقليدي الذي يتم تطبيق هذه المعادلة</td>
<td>Pseudo-space</td>
</tr>
<tr>
<td>تنقسم شكل النص إلى قسمين دراسة جسد الإنسان والبحث لتعزيز معرفتنا بالألفاظ التي يمثل بها الجسم الحي ومثلا الشمعة وعند الصداقة التقليدي الذي يتم تطبيق هذه المعادلة</td>
<td>Method 2</td>
</tr>
<tr>
<td>تنقسم شكل النص إلى قسمين دراسة جسد الإنسان والبحث لتعزيز معرفتنا بالألفاظ التي يمثل بها الجسم الحي ومثلا الشمعة وعند الصداقة التقليدي الذي يتم تطبيق هذه المعادلة</td>
<td>Alhusban’s method</td>
</tr>
<tr>
<td>تنقسم شكل النص إلى قسمين دراسة جسد الإنسان والبحث لتعزيز معرفتنا بالألفاظ التي يمثل بها الجسم الحي ومثلا الشمعة وعند الصداقة التقليدي الذي يتم تطبيق هذه المعادلة</td>
<td>Moon and Sun letters</td>
</tr>
<tr>
<td>تنقسم شكل النص إلى قسمين دراسة جسد الإنسان والبحث لتعزيز معرفتنا بالألفاظ التي يمثل بها الجسم الحي ومثلا الشمعة وعند الصداقة التقليدي الذي يتم تطبيق هذه المعادلة</td>
<td>Proposed Method</td>
</tr>
</tbody>
</table>

**TABLE V. COMPARISON OF THE EVALUATION OF SOLUTIONS WITH A COVER TEXT OF 155 LENGTH**

<table>
<thead>
<tr>
<th>Capacity rate</th>
<th>The maximum length of hidden text</th>
<th>Method</th>
</tr>
</thead>
<tbody>
<tr>
<td>46%</td>
<td>71</td>
<td>Enhanced Kashida [3]</td>
</tr>
<tr>
<td>31%</td>
<td>48</td>
<td>ZWC and space [8]</td>
</tr>
<tr>
<td>60%</td>
<td>93</td>
<td>Enhanced capacity [9]</td>
</tr>
<tr>
<td>15.4%</td>
<td>24</td>
<td>Pseudo-space [10]</td>
</tr>
<tr>
<td>61%</td>
<td>96</td>
<td>Method 2 [11]</td>
</tr>
<tr>
<td>13%</td>
<td>21</td>
<td>Alhusban’s method [12]</td>
</tr>
<tr>
<td>58%</td>
<td>91</td>
<td>Moon and Sun letters [41]</td>
</tr>
<tr>
<td>34%</td>
<td>53</td>
<td>The proposed method, without using the extended technique</td>
</tr>
<tr>
<td>74%</td>
<td>115</td>
<td>proposed method</td>
</tr>
</tbody>
</table>
TABLE VI. COMPARISON OF THE EVALUATION OF SOLUTIONS WITH A COVER TEXT OF 600 LENGTH

<table>
<thead>
<tr>
<th>Capacity rate</th>
<th>The maximum length of hidden text</th>
<th>Method</th>
</tr>
</thead>
<tbody>
<tr>
<td>44.5%</td>
<td>264</td>
<td>Enhanced Kashida [3]</td>
</tr>
<tr>
<td>34%</td>
<td>48</td>
<td>ZWC and space [8]</td>
</tr>
<tr>
<td>61%</td>
<td>369</td>
<td>Enhanced capacity [9]</td>
</tr>
<tr>
<td>17%</td>
<td>103</td>
<td>Pseudo-space [10]</td>
</tr>
<tr>
<td>68%</td>
<td>408</td>
<td>Method 2 [11]</td>
</tr>
<tr>
<td>14%</td>
<td>84</td>
<td>Alhusban’s method [12]</td>
</tr>
<tr>
<td>67%</td>
<td>405</td>
<td>Moon and Sun letters [24]</td>
</tr>
<tr>
<td>27.5%</td>
<td>165</td>
<td>The proposed method, without using the extended technique</td>
</tr>
<tr>
<td>76.6%</td>
<td>460</td>
<td>proposed method</td>
</tr>
</tbody>
</table>

TABLE VII. COMPARISON OF THE EVALUATION OF SOLUTIONS WITH A COVER TEXT OF 1100 LENGTH

<table>
<thead>
<tr>
<th>Capacity rate</th>
<th>The maximum length of hidden text</th>
<th>Method</th>
</tr>
</thead>
<tbody>
<tr>
<td>46%</td>
<td>505</td>
<td>Enhanced Kashida [3]</td>
</tr>
<tr>
<td>32.7%</td>
<td>360</td>
<td>ZWC and space [8]</td>
</tr>
<tr>
<td>62.2%</td>
<td>685</td>
<td>Enhanced capacity [9]</td>
</tr>
<tr>
<td>16.3%</td>
<td>180</td>
<td>Pseudo-space [10]</td>
</tr>
<tr>
<td>65.4%</td>
<td>720</td>
<td>Method 2 [11]</td>
</tr>
<tr>
<td>12.3%</td>
<td>132</td>
<td>Alhusban’s method [12]</td>
</tr>
<tr>
<td>65%</td>
<td>716</td>
<td>Moon and Sun letters [24]</td>
</tr>
<tr>
<td>38%</td>
<td>420</td>
<td>The proposed method, without using the extended technique</td>
</tr>
<tr>
<td>79%</td>
<td>870</td>
<td>proposed method</td>
</tr>
</tbody>
</table>

TABLE VIII. COMPARISON OF THE EVALUATION OF SOLUTIONS WITH A COVER TEXT OF 1200 LENGTH

<table>
<thead>
<tr>
<th>Capacity rate</th>
<th>The maximum length of hidden text</th>
<th>Method</th>
</tr>
</thead>
<tbody>
<tr>
<td>46%</td>
<td>737</td>
<td>Enhanced Kashida [3]</td>
</tr>
<tr>
<td>32%</td>
<td>520</td>
<td>ZWC and space [8]</td>
</tr>
<tr>
<td>63%</td>
<td>1010</td>
<td>Enhanced capacity [9]</td>
</tr>
<tr>
<td>16%</td>
<td>260</td>
<td>Pseudo-space [10]</td>
</tr>
<tr>
<td>65%</td>
<td>1040</td>
<td>Method 2 [11]</td>
</tr>
<tr>
<td>13%</td>
<td>240</td>
<td>Alhusban’s method [12]</td>
</tr>
<tr>
<td>64%</td>
<td>1029</td>
<td>Moon and Sun letters [24]</td>
</tr>
<tr>
<td>40%</td>
<td>650</td>
<td>The proposed method, without using the extended technique</td>
</tr>
<tr>
<td>81.7%</td>
<td>1310</td>
<td>proposed method</td>
</tr>
</tbody>
</table>

TABLE IX. THE SUMMARIZATION OF THE OVERALL PERFORMANCE OF THE METHODS

<table>
<thead>
<tr>
<th>Text Length 1200</th>
<th>Text Length 1100</th>
<th>Text Length 600</th>
<th>Text Length 155</th>
<th>Method</th>
</tr>
</thead>
<tbody>
<tr>
<td>46%</td>
<td>46%</td>
<td>44.5%</td>
<td>46%</td>
<td>Enhanced Kashida [3]</td>
</tr>
<tr>
<td>32%</td>
<td>32.7%</td>
<td>34%</td>
<td>31%</td>
<td>ZWC and space [8]</td>
</tr>
<tr>
<td>63%</td>
<td>62.2%</td>
<td>61%</td>
<td>60%</td>
<td>Enhanced capacity [9]</td>
</tr>
<tr>
<td>16%</td>
<td>16.3%</td>
<td>17%</td>
<td>15.4%</td>
<td>Pseudo-space [10]</td>
</tr>
<tr>
<td>65%</td>
<td>65.4%</td>
<td>68%</td>
<td>61%</td>
<td>Method 2 [11]</td>
</tr>
<tr>
<td>13%</td>
<td>12.3%</td>
<td>14%</td>
<td>13%</td>
<td>Alhusban’s method [12]</td>
</tr>
<tr>
<td>64%</td>
<td>65%</td>
<td>67%</td>
<td>58%</td>
<td>Moon and Sun letters [24]</td>
</tr>
<tr>
<td>81.7%</td>
<td>79%</td>
<td>76.6%</td>
<td>74%</td>
<td>Proposed Method</td>
</tr>
</tbody>
</table>
A. Coverage Capacity

To assess the capability of the suggested approach, the formula for determining the number of secret message encodings based on letter is presented in [28]. The results are illustrated in Fig. 7, representing the coverage capacity (CC) values computed through the application of Eq. (1).

\[
CC(M) = \frac{1}{M} \sum_{i=1}^{M} \left( \frac{L_i}{D(S_i)} \right) \times 100
\]

In the given context, “M” denotes the count of covered secret messages, representing the number of matched segments at the maximum length of “N” bits. “Li” refers to the length of each matched segment, measured in the number of letters within it. The average number of embedded bits in each letter of the segment is denoted by “K,” and the number of bits in a segment is represented by “D(Si).” For the Arabic letter, the standard encoding system utilizes 8 bits; therefore, D(Si) = 8 * Li.

![Fig. 7. Coverage capacity compared to other approaches.](image)

In some cases, in order to make a letter more beautiful, it can also be written with a stroke. In fact, stretching letters is one of the advantages of Persian and Arabic languages; this feature does not have the slightest effect on the readability or changing the meaning of the text. The only problem is that not all letters have this ability, and it is limited to the position in which that letter is placed. Accordingly, this feature has been used in Persian and Arabic languages in this research, and a hybrid encryption solution has been presented. Based on this, minor changes were made in the letters with closed spaces in addition to stretching the letters. The letters with closed space include the characters "ص،ض،ط،ظ،ع،ؽ،ف،ق،و،ٔ،ِ". Of course, for the letters "غ " only when these letters are used in the middle of the word will they have a closed space; therefore, in this method, only one position of the letters "غ " (in the middle of the word) can be used to hide the information in the letters "غ ". Accordingly, in order to hide these letters, the amount of empty space is slightly smaller so that it has the least effect on changing the font of the desired letters and through this change, if the last letter of a word that cannot be if is drawn, it is part of the letters with closed space, it is used. As a result, if the insertable bit is zero, the character will remain unchanged. But if the insertable bit was 1, the character is inserted with a smaller enclosed space. In the end, the proposed solution was implemented in the MATLAB program environment and was evaluated by the rate parameter of the encryption capacity compared to other solutions. The evaluations were carried out according to the sentences with different lengths, and the relevant results indicated that in this case, the proposed solution through the use of two techniques of stretching and changing closed letters has been able to have a suitable capacity of hiding to get writing. So in all the evaluations, it has a noticeable superiority compared to other methods.

VII. Future Works

In the future work, we can focus on the development a mobile application. Also, more improvements can be made by using the available space in Arabic and Persian letters. As a result, in this way, the coverage rate can be increased to a greater extent in the proposed solution.

VIII. Funding

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REFERENCES


A Comprehensive Review of Fault-Tolerant Routing Mechanisms for the Internet of Things

Zhengxin Lan*
College of Power Technology, Liuzhou Railway Vocational and Technical College, Liuzhou 545616, Guangxi, China

Abstract—The Internet of Things (IoT) facilitates intelligent communication and real-time data collection through dynamic networks. The IoT technology is ideally suited to meet intelligent city requirements and enable remote access. Several cloud-based approaches have been proposed for constrained IoT systems, including scalable data storage and effective routing. In real-world scenarios, the effectiveness of many methods for wireless networks and communication links can be challenged due to their unpredictable characteristics. These challenges can result in path failures and increased resource utilization. To enhance the reliability and resilience of IoT networks in the face of failures, fault tolerance mechanisms are crucial. Network failures can occur for various reasons, including the breakdown of the wireless nodes’ communication module, node failures caused by battery drain, and changes in the network topology. Addressing these issues is essential to ensure the continuous and reliable operation of IoT networks. Fault-tolerant routing plays a critical role in IoT-based networks, but no systematic and comprehensive research has been conducted in this area. Therefore, this paper aims to fill this gap by reviewing state-of-the-art mechanisms. An analysis of the practical techniques leads to recommendations for further research.

Keywords—Internet of things; routing; data transmission; fault-tolerant; review

I. INTRODUCTION

The Internet of Things (IoT) has witnessed rapid growth due to advancements in software and hardware platforms, the expansion of communication networks, and the progress in data analysis tools [1]. IoT refers to a network of interconnected devices that generate and collect data using technologies like RFID, sensors, actuators, and mobile phones [2, 3]. The International Telecommunication Union (ITU) defines IoT as a global infrastructure that connects physical and virtual objects through integrated information and communication technologies. Another definition in [4] describes IoT as a network that connects people and things, allowing them to be connected anytime, anywhere, with anyone and anything, utilizing various networks, paths, and services. Moreover, IoT technology enables communication-related services such as information exchange for collaborative IoT services, user connectivity for global networking, and data offloading to edge cloud servers for enhanced computation [5].

There has been a growing interest in the IoT in academia and industry over the last few years. As the IoT becomes increasingly ubiquitous, it provides comprehensive representations of physical environments and offers a high level of engagement with them [6]. Some examples of potential applications for this innovative paradigm include e-health, business management, Intelligent Transportation Systems (ITS), and logistics. The realization of IoT is greatly influenced by several factors, such as the system’s architecture, the network and communication infrastructure, the processing of data, and ubiquitous computing technologies, which enable effective, reliable, physical, and cyber connectivity. As part of IoT, networking, and primarily routing in the network, is an integral component that facilitates the interconnection of devices. It entails the creation of traffic routes and routing packets from the source to the final destination in a network.

In the realm of the Internet of Things (IoT), the convergence of big data, cloud computing, artificial intelligence (AI), machine learning (ML), deep learning, feature and channel selection, meta-heuristic algorithms, game theory, and association rule mining holds paramount significance, shaping the potential and efficacy of IoT applications. The massive influx of data generated by interconnected devices in IoT necessitates the utilization of big data techniques to handle, store, and process this vast volume of information [7]. Cloud computing plays a pivotal role by providing the scalable infrastructure and computational power required for efficient data management and analysis in IoT environments [8]. The integration of AI and ML in IoT applications unlocks the capability of devices to learn from data patterns, enabling them to make intelligent decisions, adapt to changing circumstances, and optimize their performance [9-11]. Deep learning, a subset of ML, further empowers IoT devices to process complex and unstructured data like images, audio, and text, enabling advanced applications in computer vision, speech recognition, and natural language processing [12-15]. Feature and channel selection play a critical role in optimizing IoT data processing by identifying the most relevant data attributes and sources, streamlining data analysis and reducing computational overhead [16]. Meta-heuristic algorithms find applicability in IoT systems for optimization tasks, such as routing and resource allocation, ensuring efficient utilization of resources and enhancing IoT network performance [17, 18]. Moreover, game theory principles are instrumental in designing cooperative or competitive strategies among IoT devices, optimizing resource allocation and energy usage in dynamic IoT environments [19]. Association rule mining brings valuable insights by discovering patterns and correlations within IoT data, aiding in decision-making processes and providing recommendations for improving IoT system performance [20].

Despite the promise of IoT, ensuring network reliability remains a significant challenge. Broken links and faulty nodes
can severely impact the reliability of IoT-enabled networks, hindering seamless communication and data exchange. To address this challenge and enhance the resilience of IoT networks, fault-tolerant routing mechanisms are indispensable. These mechanisms are designed to maintain network performance and data delivery even in the presence of failures, disruptions, or changes in the network topology. In light of the growing importance of fault tolerance in IoT networks, this review paper aims to explore state-of-the-art fault-tolerant routing mechanisms comprehensively. By analyzing critical factors such as packet delivery ratio, network lifetime, energy consumption, delay, scalability, availability, and reliability, we seek to identify effective techniques and recommend further research in this crucial area. Our goal is to contribute to the advancement of reliable and resilient IoT networks, unlocking their full potential in various applications and domains.

II. BACKGROUND

A. IoT Definitions

The IoT has been defined as a communication network where all our daily devices can connect to other devices by identifying, sensing, and processing functions. The International Telecommunication Union (ITU) has defined these devices as innovative items of the information world (virtual objects) or the physical world (physical objects), which may connect and exchange information with each other [21]. As a dynamic global network infrastructure based on standard and adaptive communication protocols, the IoT enables smart and virtual objects to be autonomous, have a physical identity, and have virtual personalities [22]. The IoT is regarded as the most influential among emerging technologies, ranking higher than artificial intelligence and robotics. The IoT is currently the most significant technology trend in the world, as stated by Burrus [23]; it is expected to cause the most disruption and provide the most opportunities over the next five years. According to a study published in Forbes Insights by over 500 executives from countries with a minimum of 500 employees, IoT has acquired global traction [24]. Forecasts encompass all aspects of the IoT ecosystem, including professional services, analytics, security, infrastructure purpose-built IoT platforms, connectivity services, and intelligent and embedded systems. All predictions indicate that IoT adoption and usage will grow regardless of the number of forecasts [5].

The IoT is defined in some ways only from a physical perspective. Real-world or virtual objects can be regarded as things. For example, Al-Fuqaha, et al. [25] defined IoT as the ability of tangible objects to perceive, listen, reason, communicate, collaborate on decisions, and fulfill tasks. As explained in a special IEEE report, the IoT is a network of connected devices, each equipped with sensors. IoT is also described by the Organization for the Advancement of Structured Information Standards (OASIS) as a system where ubiquitous sensors are used to connect the Internet to the physical world. This perspective is often called Machine to Machine (M2M) in definitions. As defined by ETSI, M2M communications are communications between two or more entities without direct human involvement. IoT covers many application fields today, including transport, utilities, healthcare, smart cities, and monitoring. It can apply to a variety of situations effectively. For instance, IoT can collect valuable data through sensor devices. In addition, IoT devices can serve as an efficient means of transmitting data. However, IoT devices have some limitations regarding transmission, processing, battery life, and memory capabilities.

B. IoT Architecture

All nodes in the IoT environment can communicate and cooperate to accomplish predefined goals. Therefore, the IoT must have a layering architecture that is flexible enough to support a wide range of heterogeneous elements over the Internet. Despite the growing number of IoT architectures, none can be adopted as a reference model. A typical IoT architecture comprises three layers: application, network, and perception. In the meantime, researchers have proposed updated models by incorporating additional abstractions into the architecture of the IoT. Fig. 1 illustrates two common IoT architectures. Several research studies like [26, 27] used a 5-layer model similar to TCP/IP. A 5-layer architecture model starts with the perception or objects layer, which comprises physical sensors that collect, process, and analyze information. Several crucial technologies facilitate the transfer of the produced data from the perception layer to the next layer, including Wireless Fidelity (Wi-Fi), Third Generation (3G), Radio Frequency Identification (RFID), Global System for Mobile Communications (GSM), and infrared. The object abstraction layer securely transfers data from the preceding layer (objects) to the upper layer (management of services). This layer serves as a standard interface to handle a variety of things. Known as the pairing (middleware) layer, the service management layer pairs services with requesters based on their names and addresses. IoT programmers can work independently with heterogeneous objects platforms through this layer. The application layer provides the requested services to users. The business layer creates a business representation, flowcharts, diagrams, etc., according to the data provided by the application layer.

Connectivity is a necessary and sufficient condition for IoT, which integrates different technologies. It is, therefore, essential to enhance communication protocols as part of the technology. IoT communication protocols are generally classified into three categories, Server to Server (S2S), Device to Device (D2D), and Device to Server (D2S). S2S communications involve the exchange of data between servers, which is mainly used in cellular networks. Mobile phones can communicate with each other through D2D communication, referred to as the next generation of cellular networks. In D2S, all data is transmitted to servers, regardless of location. These communications require the processing and preparation of data. This challenge calls for various data processing methods, such as analytics at the edge, stream analysis, and IoT analysis at the database. Each process should be customized based on the specific application and its requirements. Cloud and fog processing are two analytics tools used to prepare and process data before transferring it to another application. In a nutshell, sensors and IoT devices collect environmental data. The next step is to extract knowledge from the unprocessed data. Afterward, data can be transferred to other elements, devices, or servers over the Internet [8].
C. IoT Key Elements

As shown in Fig. 2, the IoT environment involves some critical elements defined below.

- **Identification:** The IoT requires identification to establish its services and match them with the demands of its users. Identifying objects paves the way for query, management, and control of object information. Currently, several identification methods exist, each with its characteristics in terms of coding schemes and analysis systems, such as Ubiquitous Codes (uCode) and Electronic Product Codes (EPC) [28].

- **Sensing:** IoT sensing involves capturing data from connected sensor nodes and delivering it to a database or data warehouse. Based on the recorded data, specific actions are taken according to the required services. IoT sensors can be intelligent actuators or wearable sensing devices [29].

- **Communication:** Connecting heterogeneous objects with IoT communication technologies enables customized intelligent services. Generally, IoT nodes should operate at low power when communication links are lossy and noisy. The IoT uses a variety of communication protocols, such as RFID, Near Field Communication (NFC), Wi-Fi, ultra-wide bandwidth (UWB), LTE-Advanced, IEEE 802.15.4, Z-wave, and Bluetooth [30].

- **Computing:** The IoT is powered by microcontrollers, microprocessors, SOCs, FPGAs, and software applications. IoT applications can be run on various hardware platforms, including T-Mote Sky, Z1, Cubieboard, Gadgeteer, Raspberry PI, and Arduino [31].

- **Semantics:** The concept of semantics focuses on the capability of machines to intelligently extract knowledge to facilitate the provision of services necessary for the IoT. Extraction of knowledge involves discovering and utilizing resources and incorporating information into models. In addition, it consists of the recognition and analysis of data to determine the most appropriate service. Semantics are integral to the IoT by sending requests to the appropriate resources. Semantic Web technologies, such as the Resource Description Framework (RDF) and Web Ontology Language (OWL), support this requirement [32].

- **IoT service:** Aiming to facilitate human life, the IoT provides a wide range of services typically delivered as physically isolated vertical solutions. Discovering suitable IoT services faces various challenges and requirements, such as the heterogeneity of accessible services, vast distribution of services, and a highly dynamic environment [33]. Some essential services that IoT provides include healthcare systems [34], remote control [35], transportation [36], education [37], environment monitoring [38], disaster recovery [39], and anomaly detection [40].

- **IoT resource:** As the IoT entails diverse heterogeneous components, it requires substantial storage and processing to meet users' requests and provide valuable services. Some applications may require complex processing, such as time series analysis, while others may be latency-sensitive. Since the resources of IoT objects are limited in terms of energy, network bandwidth, CPU, and memory, it isn't easy to obtain an ultra-scale and real-world IoT network without taking advantage of cloud platforms or some powerful devices, such as edge or fog nodes and smart gateways [41].

- **IoT task:** In an IoT-based network, users' requests are organized into two main types of tasks: independent tasks and dependent tasks. The separate tasks, also called atomic tasks, refer to the tasks in which no dependency exists among them. In contrast, the dependent tasks require a specific execution order due to their relationship [42].
Fig. 2. IoT key elements.

D. Basic Optimization Parameters

- Optimization problem definition: Optimization in the computational domain refers to selecting the most optimal solution, among others, based on various factors [43].

- Search space: In a given problem, the search space refers to a collection of potential or candidate solutions. Each point within the search space represents a specific solution that can be evaluated based on its value or suitability for the problem. The search space encompasses all feasible solutions that can be explored and considered during problem-solving. Analyzing and assessing different points within the search space can identify the most optimal or desirable solution for the problem. [44].

- Objective function: An objective function quantifies the solution to an optimization problem. It refers to an optimization objective, such as minimizing or maximizing metrics (e.g., performance, energy consumption). Optimization variables are transformed into real numbers by the objective function, which can be a single-objective (minimizing response time) or a multi-objective (e.g., minimizing energy consumption and maximizing throughput) [45].

- Population and individual encoding: In meta-heuristic algorithms, individuals within a population represent potential solutions to a specific problem. These individuals are encoded using various data structures, such as Boolean values, strings, or trees. The choice of encoding depends on the nature of the problem being solved and the information required to represent the solutions accurately. In many cases, fixed-length and fixed-order bit strings encode candidate solutions in meta-heuristic approaches. This allows for a consistent and standardized representation of solutions across the population. Fig. 3 illustrates the common encoding methods employed in meta-heuristic algorithms. The encoding can involve binary, discrete, or real values, depending on the specific problem requirements and constraints [46].

- Initialization: Initialization is the process of assigning initial values to the search space in order to create the initial population for a meta-heuristic algorithm. The selection of initial solutions can be made using various methods, one of which is random initialization. In random initialization, individuals are randomly chosen from the search space to form the initial population. This means that the values of the individuals are selected without any specific pattern or bias, providing a diverse starting point for the algorithm. Random initialization helps to explore different regions of the search space and avoids getting stuck in local optima. By introducing randomness in the selection of initial solutions, the algorithm has the potential to discover better solutions throughout the optimization process [47].
Termination criteria: Optimization algorithms are typically executed multiple times in order to obtain the best possible results. The way in which these algorithms are executed can be classified as either dynamic or static. In dynamic mode, the iterations of the algorithm continue until the fitness function, which measures the quality of the solutions, fails to improve after a certain number of repetitions. This approach allows the algorithm to adapt and continue searching for better solutions as long as progress is made. If the fitness function stops improving, it suggests that the algorithm has reached a point where further iterations are unlikely to yield significant improvements. In static mode, a fixed number of iterations is predetermined before the optimization algorithm begins. The algorithm will iterate for the specified number of iterations, regardless of whether the fitness function continues to improve or not. Once the predetermined number of iterations is completed, the algorithm terminates. Both dynamic and static termination criteria have their advantages and use cases. Dynamic termination criteria provide flexibility and allow the algorithm to adapt its stopping point based on progress. On the other hand, static termination criteria provide a predetermined stopping point, which can be useful in scenarios where a fixed number of computational resources or time is allocated for the optimization process. The choice of termination criteria depends on the specific problem, available resources, and desired optimization objectives [48].

III. REVIEW OF FAULT-TOLERANT ROUTING PROTOCOLS

This section reviews recent fault-tolerant IoT routing protocols based on important factors such as network lifetime, energy consumption, packet delivery ratio, delay, scalability, availability, and reliability. The strengths and weaknesses of existing works are briefly outlined. Table I provides a side-by-side comparison of the protocols using qualitative parameters for analysis and evaluation. This comparison helps identify efficient methods based on specific requirements.

Agarwal, et al. [49] introduced a multi-objective Deep Reinforcement Learning (DRL)-based approach to fault tolerance in IoT-enabled WSNs. They proposed a double-layer DRL-based approach that incorporates a low-level DRL agent to identify and isolate faulty nodes and a high-level DRL agent to select optimal fault-tolerant routing paths. Their approach was tested on a real-world WSN dataset and showed promising results. The proposed double-layer DRL-based approach enables the WSNs to identify and isolate faulty nodes in a timely manner and select optimal fault-tolerant routing paths with the optimal deployment of resources. The results of the study suggest that the proposed approach is able to detect faulty nodes with high accuracy and minimal overhead, which is essential for the robust and reliable operation of IoT-based WSNs. In addition, it emphasizes the need to transmit data in a reliable manner after fault detection. This is because faulty nodes can cause significant delays in data transmission and lead to data loss. The proposed approach is able to identify such faulty nodes quickly, allowing them to be removed from the network and data transmission to be resumed in a timely manner. The last step is to use a mobile sink to gather data in an energy-efficient manner, which in turn significantly increases the lifetime of the network. The proposed algorithm outperformed the state-of-the-art algorithms in terms of throughput, network lifetime, and fault detection accuracy.

Moreover, it demonstrated superior scalability, allowing it to be implemented in larger networks without compromising performance. However, there are some potential drawbacks to this algorithm as well. One such drawback is that it requires more computational resources than some of the other algorithms. Additionally, it is unclear how well this algorithm will perform in more complex networks.

Cluster-based routing is an effective way to reduce transmission overhead and conserve energy as well as improve transmission quality. It also allows nodes to form clusters which are then used to route data between nodes. By forming clusters, the number of messages that need to be transmitted across the network is reduced, which leads to improved energy efficiency. The Cluster Heads (CHs) are responsible for aggregating and filtering the sensed data before forwarding it to the base station, which further reduces the transmission overhead. Additionally, cluster-based routing allows for better scalability, as nodes can be added or removed from clusters without disrupting the overall network. When one or more CHs fail, the faulty CHs cannot forward data from their serving sensor nodes. This results in insufficient sensed data of the IoT application field being available to the sink node. The IoT
applications will be adversely affected by this change. Lin, et al. [50] developed virtual CH formation and flow graph models to tolerate CH failures effectively. By using the virtual CH formation and flow graph models, the resources of all failure-free CHs can be effectively utilized. This allows the CHs to serve as a backup for any faulty CHs, ensuring that packets are still able to be routed properly even in the event of a failure. The experiments conducted show that the approach is successful in providing fault-tolerant routing for IoT WSNs. However, there are also potential drawbacks to this approach. For example, if there are too many CH failures, the system may become overloaded and unable to function properly. Additionally, if the CHs are not properly distributed, this could lead to areas of the network being underserved or not having any CHs at all.

WSNs are vulnerable to physical and environmental factors, such as node failure, interference, and signal attenuation, which can lead to significant delays and packet losses. This can cause disruption to applications that rely on the network for communication, such as smart home applications and automated industrial systems. Topology changes, battery drain, or failure of the wireless node communication module pose the greatest threat to network failure. These factors can cause a decrease in the Quality of Service (QoS) of the network, which can lead to increased latency, data loss, increased energy consumption, and, finally, failure of the application. WSNs can be configured with fault tolerance mechanisms to ensure reliable data acquisition and transmission in IoT applications. Jaiswal and Anand [51] proposed a fault-tolerant cluster-based routing scheme for WSNs combining Grey Wolf Optimization (GWO) and Firefly Optimization (FO) algorithms. An optimal clustering algorithm is implemented using FO, and a routing algorithm is implemented using GWO. To improve network performance and meet quality of service criteria, the algorithm takes into account the energy efficiency and fault tolerance of sensor nodes and CHs. The FO and GWO algorithms were simulated in WSN scenarios in order to compare the results to those from existing techniques. The results showed that the algorithms were able to achieve optimal clustering, improved network performance, and higher energy efficiency while also providing greater fault tolerance and meeting quality of service requirements. However, it should be noted that the FO and GWO algorithms have not been tested in large-scale or real-world scenarios. As such, it is unclear how well they would perform in these types of environments.

The routing strategy proposed by Muhammed, et al. [52] uses multiple clusters and a hierarchical routing structure, which allows it to detect faults and dynamically reroute messages quickly. This reduces the amount of energy consumed and increases the reliability of the network. The simulations show that the proposed method is more reliable and efficient than LEACH and DFTR protocols, as it can detect and reroute traffic more quickly and with less energy consumption. This makes it a better option for networks with large numbers of nodes and dynamic traffic patterns. According to the results, the proposed method performs better than LEACH and DFTR in terms of the network's total energy, the number of nodes left active after a given period, and the network bandwidth. The proposed method is able to balance the energy load more efficiently by selecting the appropriate nodes for activation. It also provides a more balanced traffic pattern across the entire network, which leads to improved energy efficiency and better performance overall. However, there are some potential drawbacks to the proposed method. First, it is more complex than LEACH and DFTR and thus requires more processing power and time to execute. Second, it may be less effective in very large networks with a large number of nodes due to the increased complexity.

WSN-based IoT networks are deployed to collect and transmit data from various sources. IoT sensor nodes within these networks possess diverse properties and characteristics. To efficiently manage data transmission, cluster-based routing is commonly employed. This approach involves dividing the network into clusters, with each cluster being managed by a CH node responsible for aggregating and forwarding the data. However, in the event of one or more CH failures, the sensor nodes within the affected clusters are unable to forward their sensed data to the CHs.

Consequently, the sink node or gateway, which receives and processes the collected data, may not be able to effectively receive the data from the IoT application due to the disrupted data transmission caused by the faulty CHs. This will have a significant impact on the processing of information in this field. To avoid this problem, a more reliable routing protocol is needed that can detect and recover from faults quickly. Such a protocol should be able to detect the failure of CHs and re-assign the sensed data of sensor nodes to other CHs so that the sink node (gateway) can still collect the sensed data of the IoT application in a timely manner. Sivakumar and Vivekanandan [53] present a paired cluster of fault-tolerant disjoint path routing in a path graph and a novel method for solving this dilemma in polynomial time. By providing disjoint paths, the proposed routing model increases the reliability of the system since even if one path fails, the data packets can be rerouted over the other path.

Additionally, the polynomial time solution ensures fast and efficient routing in terms of latency and throughput. The benchmark network simulators measure the latency, throughput, packet delivery ratio, and packet drop ratio in order to accurately evaluate the proposed routing model's performance. In addition to the reliability and efficiency of the system, the simulators also measure the scalability of the model in order to ensure that it can handle large numbers of nodes in a network. However, there are some potential drawbacks to this approach. For example, if the network topology changes, the routing table will need to be recalculated, which could cause delays in communication. Additionally, this approach may not be well suited for highly dynamic networks.

IoT routing problems can be most effectively addressed by swarm intelligence algorithms. Swarm intelligence algorithms are highly adaptive and flexible, allowing them to adjust to changes in the environment and traffic conditions. They also have the ability to make decisions based on real-time data and to quickly update routing paths in response to changes in the network. This makes them an ideal tool for addressing the
ever-changing challenges of IoT routing. IoT devices should be fault free in order to improve their efficiency and reliability. To ensure that the devices are error free, many approaches and mathematical models are used. Sharma, et al. [54] proposed the Improved Efficient and Intelligent Fault-Tolerance Algorithm (IEIFTA). IEIFTA can fix any fault at a fast rate, improve efficiency, and prevent data loss if any fault occurs. IEIFTA is a highly reliable and efficient algorithm that can identify the source of any fault and fix it before it can affect the entire system. It also uses a predictive model to anticipate potential faults and take preventive measures to ensure that the system remains error free.

Additionally, it can identify any potential threats and take preventive measures to ensure that the system does not suffer from any data loss. However, there are some potential disadvantages to using IEIFTA. First, it is possible that the algorithm may identify a false positive, which would result in the unnecessary shutdown of a system. Additionally, IEIFTA requires a significant amount of data to be effective, which may not be available in all cases. Finally, the algorithm may be computationally intensive, which could impact its performance in real-time applications.

The proposed routing protocol Bounceur, et al. [55] addresses the need for a leader node in ad hoc networks, particularly in WSNs and IoT networks. The leader node serves various purposes, such as key generation for encryption or decryption and identifying nodes with minimum energy. In their protocol, the process of identifying boundary nodes begins by placing a leader node on the far left of the network. These nodes typically monitor sensitive, dangerous, or inaccessible areas. Since it can be challenging or impossible to intervene if a node fails, the algorithm must be robust and fault-tolerant. In case the leader node fails, it can have catastrophic consequences. To overcome these challenges, the authors propose a new algorithm called DoTRo, which is based on a tree routing protocol. The algorithm involves initiating a flooding process by the local leaders to determine a spanning tree. During this process, the values of the local leaders are routed. If two spanning trees meet, the tree with the best value continues while the other tree stops. The dominant tree that remains becomes the leading tree, and its root becomes the new leader. The DoTRo algorithm has demonstrated high energy efficiency, achieving reduction rates exceeding 85%. It operates effectively even in scenarios where any node can fail or when the network becomes disconnected. The algorithm is designed to be efficient and fault-tolerant in such situations.

Increasingly sophisticated applications, such as fire sprinkler systems, employ multiple sources and sinks, referred to as many-many IoT networks. The development of a fault-tolerant routing protocol is necessary for these critical applications to ensure that messages can be routed around failed nodes without causing significant overhead. Focusing on many-many IoT networks, Grosso and Jhumka [56] propose an efficient distributed fault-tolerance IoT routing scheme based on the Ant Colony Optimization (ACO) algorithm, capable of routing data from multiple sources to multiple sinks. Based on the simulation results, the protocol achieves a delivery rate of more than 80% with a failure rate of only 5%. In comparison with a number of approaches that require periodic maintenance of the topology, this approach is more scalable.

Hasan and Al-Turjman [57] introduce a biologically inspired particle swarm optimization (PMSO) routing approach for constructing, recovering, and selecting k-disjoint paths that can tolerate failure while maintaining the quality-of-service requirements. Using a multi-swarm strategy, the optimal direction for selecting the multipath routing can be determined while all segments of the network exchange message at the same time. Compared with canonical particle swarm optimization (CPSO), the proposed algorithm has demonstrated high-quality solutions. The results indicate that multi-swarm and full PMSO with constriction coefficients are superior to CPSO in terms of sensor count and 89.15% and 86.51% under the ring and mesh topologies, respectively.

Misra, et al. [58] suggest an integrated multi-layer and learning automata-based fault-tolerant routing approach for IoT networks, ensuring packet delivery despite failures affecting both source and destination nodes. As this work involves IoT, the algorithm designed should be highly scalable and should deliver high levels of performance in heterogeneous environments. The learning automata and multi-layer strategies incorporated into the proposed method provide a flexible structure to the algorithm so that a consensus can be achieved across the network using the same standard. As a result, it chooses the optimal action based on the changing environment. In order to conserve energy, all nodes located on unused paths are put to sleep. Simulated results show that the proposed strategy improves the overall energy efficiency of the network and reduces overhead compared to the existing protocols we have used as benchmarks.

In order to address the constraints of IoT systems, numerous cloud-based solutions with effective routing and scalable data storage have been presented. These solutions allow for the efficient handling of large amounts of data and reliable communication between connected devices. They also provide a secure and reliable platform for various applications and services, such as analytics and machine learning. However, as mobile networks and communication links are unpredictable, most of the solutions may not be suitable for realistic applications and will result in path failure and an increase in resource consumption. Thus, data forwarding can only be reliable and valuable when the algorithms proposed are trusted and aware, have low overheads, and consume a balanced amount of energy across the nodes. Haseeb, et al. [59] suggested a fault-tolerant supervised routing scheme in the context of IoT trust management in order to enhance trustworthiness and collaboration within smart cities. A reliable and optimized network structure is established by each node evaluating its neighbors' behavior.

Furthermore, a fault-tolerant relaying system is provided by employing supervised machine learning without imposing additional overhead. In addition, it eliminates the additional workload associated with determining the optimal decision and training the IoT system to balance network costs. Finally, a secure algorithm with secured keys is proposed to ensure the privacy and authentication of the relaying system. Compared with previous work, the proposed model has shown significant
improvements in performance. However, the proposed model has not been proven to be secure against all possible attacks. Furthermore, the algorithm has not been tested on a large scale.

Chanak, et al. [60] present a fault-tolerant routing protocol for IoT-driven WSNs, which significantly enhances the QoS of these networks. They develop a new multi-population z-test-based fault detection method to identify faulty devices in the network. This method is based on the analysis of the data collected from the network and uses a combination of the z-test and the chi-square test to detect faulty nodes. The protocol also utilizes a novel routing algorithm to reroute the data around faulty nodes in order to ensure that the data is delivered to its destination in a timely manner with minimal disruption. The proposed routing protocol has been designed to provide fault tolerance and flexibility, allowing the reuse of faulty nodes in the network. The experiments conducted to test the protocol demonstrate its efficiency and effectiveness in various areas, including fault detection accuracy, energy consumption, and network lifetime. The results of these experiments are then compared with the state-of-the-art algorithms to show the effectiveness of the proposed scheme. However, there are some potential drawbacks to this scheme that should be considered. First, the reliance on faulty nodes could lead to increased network instability. Second, the additional overhead required to maintain the fault tolerance could lead to higher energy consumption and shorter network lifetimes.

In Industrial 4.0, safety is one of the main concerns, where various physical parameters are monitored to prevent uncertain events. A natural disaster, such as a fire or the leakage of harmful gases, can cause tremendous damage to both life and property in the industrial sector. Industrial IoT (IIoT) is employed to monitor such natural calamities and take appropriate action in a timely fashion. The IIoT, however, is susceptible to sensor failures as a result of energy depletion and hardware malfunctions. This results in a significant reduction in the network's reliability. Kaur and Chanak [61] propose a fault-tolerant framework in which faults in the WSN-assisted IIoT in the form of node failures and link failures are identified and handled efficiently. The proposed framework uses a distributed consensus-based approach to identify and detect faults in the WSN-assisted IIoT. It also uses a fault-tolerant routing protocol to route traffic around the faulty nodes, thus ensuring that the IIoT remains reliable even in the case of a node or link failure. The proposed scheme has been extensively simulated and has been found to outperform other schemes as measured by recovery speed, communication delay, network lifetime, throughput, and energy consumption. Although the proposed scheme has been found to have many benefits, there are also some drawbacks to consider. For example, the scheme requires more energy to operate than other schemes, which may not be feasible for some IIoT applications. Additionally, the scheme may not be able to handle all types of node and link failures, which could lead to network outages.

The implementation of various technologies for industrial information delivery and process control has been hindered by the challenges of increased complexity and associated faults. These factors have posed obstacles to achieving reliable and timely network activation. This is because industrial systems are usually comprised of multiple components that can be affected by different environmental conditions, as well as by the transmission of inaccurate or incomplete data. As a result, the transmission of reliable data to activate the timely network is challenging. In order to transfer this data from one node to another, there must be no faults or delays between the nodes. As a solution to this problem, Vishal Sharad, et al. [62] developed a new algorithm to communicate messages between different services without any delays. Their algorithm, based on the MoO4RPL objective, simulates IoT with mobile sink nodes in the network. The strategy consists of several phases, including topology generation, route discovery, communication, and route maintenance. In the multi-objective route discovery phase, the algorithm constructs the network topology and calculates a rank based on factors, such as energy, trust, delay, fault tolerance, and link quality. The proposed method with fitness function factors is used to select the optimal route during the communication phase. The fitness function factors are used to evaluate each route based on metrics, such as distance, energy, link quality, and trust. The route with the best metrics is determined to be the optimal route and is then maintained during the route maintenance phase. However, the proposed method may not be feasible in real-world scenarios due to the computational overhead required to calculate the fitness function factors for each route. In addition, the proposed method does not consider dynamic changes in the network, which can lead to suboptimal routing decisions.

Pankajavalli and Karthick [63] introduced a novel approach known as the Free Poisson Law method. This technique addresses the challenge of paired fault-tolerant cluster routing in a data flow graph by establishing disjoint routes. The technique utilizes the idea of assigning probabilistically independent Poisson-distributed weights to the edges of the graph. This allows for a simple and efficient algorithm that can find a pair of disjoint routes between two nodes using a single pass through the graph. The Free Poisson Law technique takes advantage of the fact that Poisson-distributed weights can be used to represent the probability of a given edge being selected. This allows the algorithm to select a pair of disjoint routes that are both likely to be successful. In addition, the algorithm only needs to take one pass through the graph, making it highly efficient. The primary objective of the algorithm is to minimize latency, energy consumption, dissipated energy, and functional complexity, thereby enhancing the packet delivery ratio, throughput, and fault detection rate. However, the proposed algorithm does not guarantee that the selected pair of routes will be successful. In addition, the algorithm may not be able to consider all the factors that can affect packet delivery, such as network congestion.
Poisson Law method presents a novel technique for establishing disjoint routes in paired fault-tolerant cluster routing in data flow graphs. This technique provides an efficient algorithm for finding probabilistically independent Poisson-distributed weights on graph edges, minimizing latency and enhancing packet delivery ratio. However, ensuring successful routing and considering all factors affecting packet delivery requires further investigation.

In conclusion, our review of fault-tolerant routing mechanisms in IoT-based networks has revealed a rich landscape of innovative approaches that tackle the challenge of ensuring reliable and resilient communication. Each approach offers unique strengths and limitations, making it crucial to select the most appropriate method based on the specific application requirements.

Finally, the Free Poisson Law method presents a novel technique for establishing disjoint routes in paired fault-tolerant cluster routing in data flow graphs. This technique provides an efficient algorithm for finding probabilistically independent Poisson-distributed weights on graph edges, minimizing latency and enhancing packet delivery ratio. However, ensuring successful routing and considering all factors affecting packet delivery requires further investigation.

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requirements and network characteristics. We recommend further research to explore hybrid approaches, combining the strengths of different techniques and conducting extensive real-world testing to validate the performance and scalability of these methods. Additionally, addressing security concerns and investigating the potential trade-offs between energy efficiency and fault tolerance is vital for optimizing network performance in IoT environments. By building upon these findings and pursuing further research in fault tolerance, we envision the development of more robust and efficient IoT networks, unlocking their full potential in diverse applications and domains.

V. CONCLUSION

The IoT is revolutionizing various facets of our lives and leading us towards enhanced societies in the future. WSNs play a crucial role in the IoT landscape. Energy conservation, resilience, and reliability are three essential requirements for WSNs. The ability of WSNs to tolerate faults ensures their reliability and resilience in the event of failure. The most frequent causes of network failure are typically attributed to changes in network topology, node failure caused by battery depletion, and the malfunctioning of wireless communication modules within nodes. These factors have been identified as the primary culprits responsible for disrupting the seamless operation of networks. It is essential to address these challenges to ensure the reliability and stability of network connectivity in various environments. This paper reviewed state-of-the-art fault-tolerant IoT routing protocols concerning critical factors such as packet delivery ratio, network lifetime, energy consumption, delay, scalability, availability, and reliability. Based on the analysis of the effective techniques, recommendations are made for further research. One promising direction lies in exploring machine learning-based approaches, leveraging real-time data to dynamically adapt network routing decisions. Additionally, investigating the integration of blockchain technology could enhance the security and reliability of fault-tolerant routing in IoT systems. Efforts towards developing lightweight fault-tolerant routing algorithms, mindful of the resource constraints and energy limitations of IoT devices, will greatly benefit the scalability and practicality of fault-tolerant IoT networks. Moreover, tailored fault-tolerant solutions catering to specific IoT applications, such as smart cities, industrial automation, or healthcare systems, can better address unique challenges and requirements in those domains.

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DeepShield: A Hybrid Deep Learning Approach for Effective Network Intrusion Detection

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Abstract—In today's rapidly evolving digital landscape, ensuring the security of networks and systems has become more crucial than ever before. The ever-present threat of hackers and intruders attempting to disrupt networks and compromise online services highlights the pressing need for robust security measures. With the continuous advancement of security systems, new dangers arise, but so do innovative solutions. One such solution is the implementation of Network Intrusion Detection Systems (NIDSs), which play a pivotal role in identifying potential threats to computer systems by categorizing network traffic. However, the effectiveness of an intrusion detection system lies in its ability to prepare network data and identify critical attributes necessary for constructing robust classifiers. In light of this, this paper proposes, DeepShield, a cutting-edge NIDS that harnesses the power of deep learning and leverages a hybrid feature selection approach for optimal performance. DeepShield consists of three essential steps: hybrid feature selection, rule assessment, and detection. By combining the strengths of machine learning and deep learning technologies, a new solution is developed that excels in detecting network intrusions. The process begins by capturing packets from the network, which are then carefully preprocessed to reduce their size while retaining essential information. These refined data packets are then fed into a deep learning algorithm, which employs machine learning characteristics to learn and test potential intrusion patterns. Simulation results demonstrate the superiority of DeepShield over previous approaches. NIDS achieves an exceptional level of accuracy in detecting malicious attacks, as evidenced by its outstanding performance on the widely recognized CSE-CIC-DS2018 dataset.

Keywords—Network intrusion detection system; IDS; cyber security; machine learning; deep learning

I. INTRODUCTION

The Internet has evolved into a necessary tool and one of the most reliable sources of knowledge about the modern world. It can be considered a crucial component of education and business. Therefore, preserving data across the Internet becomes challenging [1]. Nowadays, internet security is a serious problem [2]. Over the last decade, computer networks have grown in complexity, usage, and size. Cloud computing and the Internet of Things (IoT) have evolved into entirely new types of devices and networks [3]. These networks and systems have grown in size and complexity, so their security has become a critical concern [4]. According to CyberEdge group statistics, the number of attacks on large enterprise networks worldwide has increased significantly in recent years [5]. Advanced Persistent Threats (APT), malware, and denial of service attacks are examples of these attacks [6]. APTs are particularly hazardous and expensive since they are long-term, targeted operations carried out by sophisticated perpetrators targeting the public sector and business enterprises in order to exfiltrate data and cause infrastructure damage [7]. According to cybersecurity studies, these attacks were active for an average of 184 days in 2018 (the duration of attack effectiveness before it is detected) [8].

As a primary layer of defense against computer system vulnerabilities and attacks, a robust security model implements industry-standard security standards such as authorization, access control, confidentiality, and other security requirements [9]. Nevertheless, attacks are likely to continue to present a threat due to vulnerabilities in the system, operational errors, and other issues [10]. Intrusion Detection Systems (IDSs) are critical in identifying and alerting system administrators to intrusions into computers and networks [11]. The IDS can be installed on individual servers within a network, at a centralized location, or distributed around the network [12]. A Network Intrusion Detection System (NIDS) is a kind of IDS intended to track attacks across multiple hosts instead of a single host. These systems monitor network operations using network telemetries, such as network traffic, network flow metadata, and host event logs, to identify attack events [13].

In the realm of NIDS, the convergence of machine learning, artificial intelligence, meta-heuristic algorithms, deep learning, feature selection, association rule mining, and fault diagnosis plays a pivotal role in fortifying cybersecurity defenses and safeguarding critical network infrastructures against evolving cyber threats. Machine learning and artificial intelligence techniques enable IDSs to continually learn and adapt to new attack patterns, enhancing their accuracy in distinguishing between normal and malicious network activities [14-17]. The integration of meta-heuristic algorithms optimizes the performance of intrusion detection models, fine-tuning parameters and reducing false positives [18]. Deep learning empowers systems to automatically extract intricate features from raw network data, enabling the identification of sophisticated and novel attack signatures [19, 20]. Feature selection techniques help to identify the most relevant network attributes, streamlining the detection process and reducing computational overhead [21]. Additionally, fault diagnosis capabilities enable swift identification and response to potential anomalies, further strengthening the overall resilience of intrusion detection systems [16, 22]. Association rule mining holds paramount importance in NIDS as it enables the discovery of hidden patterns and correlations in network data, facilitating the identification of suspicious and anomalous activities that might go undetected using traditional IDSs [23].

This amalgamation of cutting-edge technologies empowers...
organizations to proactively detect, mitigate, and thwart cyber threats, ensuring the confidentiality, integrity, and availability of critical data and establishing robust and future-proofed NIDS.

To identify intrusions and anomalies, a NIDS continuously monitors network traffic. In the case of high network throughput, using a single NIDS on a network can cause congestion. Deep packet inspection may include significant similarities to complicated signatures of attack rules [24, 25]. Pattern matching is a time-consuming procedure that requires substantially more computing power than a firewall, which might cause a NIDS to become overloaded [26]. When a NIDS becomes overburdened and begins dropping or ignoring packet content, network security may be compromised. Finally, some vulnerabilities may remain unnoticed since some packets associated with the same attack may escape the NIDS’s inspection, leading to an insufficient match between packets [27]. NIDS employ several strategies to handle high levels of network traffic, including:

- Hardware upgrades, including the addition of dedicated packet capture cards and more computing resources, as well as modifying the NIDS software to increase its capacity.
- Utilizing a cluster of NIDSs and distributing signature rules and network traffic among the NIDS hosts.

The first strategy, which involves optimizing the NIDS application and upgrading hardware, is prohibitively expensive and unscalable. Every four years, network bandwidth rates grow by a factor of 10; therefore, maintaining a NIDS requires ongoing hardware upgrades. Adjusting a NIDS to cope with greater traffic is a difficult procedure that includes various trade-offs, resulting in the NIDS being more complex than anticipated. The second strategy, which relies on NIDS clusters, is cost-effective and scalable. When network traffic is low, the solution can be adjusted to accommodate it, and resources can be released and utilized for other reasons.

Numerous studies have demonstrated the benefit of low-cost clustering computers equipped with NIDS to manage high network traffic loads. Additionally, the cluster can be expanded by adding additional NIDS instances. However, both the distribution of traffic among NIDS instances and the distribution of signature rules are critical to the effectiveness of the solution [28].

The use of machine learning algorithms in the context of NIDS has received considerable attention. Training machine learning algorithms on normal and attack traffic enables them to detect novel differences in network traffic. Traditionally, the NIDS is designed by an expert human analyst who codifies rules defining normal behavior and intrusions [29]. Due to the numerous failures of this method to identify novel intrusions and the aim to reduce the analyst's effort, machine learning algorithms have been incorporated into NIDS to automate the process and supplement the human effort. This study proposes a new method comprised of machine learning and deep learning algorithms for feature selection and intrusion detection.

The key contributions of this research paper include the development of a cutting-edge NIDS that leverages deep learning and a hybrid feature selection approach. The three-step architecture, consisting of hybrid feature selection, rule assessment, and detection, enhances the effectiveness of the intrusion detection system. By combining the strengths of machine learning and deep learning technologies, the proposed NIDS demonstrates superior performance in detecting network intrusions. The careful preprocessing of network data, followed by the application of a deep learning algorithm, allows for the identification and testing of potential intrusion patterns. Simulation results showcase the exceptional accuracy of the proposed method, surpassing previous approaches, as demonstrated by its outstanding performance on the widely recognized CSE-CIC-DS2018 dataset.

II. IMPORTANCE OF THE NIDS

As the Internet is vulnerable to various threats, it is critical to develop a system that protects the data and the individuals using it [30]. For years, the scientific community has focused on identifying cyber-attacks that target information and communication networks. Developing a comprehensive and efficient NIDS is one of the primary challenges in network security. These systems are critical for network administrators to detect different security vulnerabilities within an organization's network. The NIDS monitors and analyses network traffic incoming and departing an organization's network devices and triggers alerts if an intrusion is detected.

An IDS takes its name from the conjunction of two concepts, intrusion and detection systems. Generally, an intrusion is defined as gaining access without authorization to a network or computer system with the intent of compromising its functionality, privacy, or reliability. The IDS detects such illegal activities. Therefore, the IDS serves as a security component responsible for monitoring network traffic to identify suspicious activities that violate security policies and endanger the network's availability, reliability, and stability. It notifies hosts or network administrators of detected malicious activities. As shown in Fig. 1, NIDS is deployed passively by connecting to a network switch equipped with mirror ports. In order to monitor traffic and detect intrusions, all inbound and outbound network traffic should be mirrored to NIDS. By installing NIDS in the middle of the network switch and firewall, all traffic can be routed through it.

Modern NIDS are divided into two categories: rule-based misuse detection and statistical anomaly detection. In the first method, a database is used to store the characteristics of a wide variety of known attacks and the network traffic is classified as an "attack" if the retrieved characteristics match those stored in the database. Although this kind of NIDS can rapidly and accurately detect known attacks, it is weak at detecting future attacks. As a result, anomaly detection-based NIDS has gained popularity recently. According to its basic assumptions, the system detects and identifies abnormalities in network traffic properties or distributions [31].

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IDS strategy is anomaly-based, which relies solely on normal data in order to identify abnormal samples. The NIDS raises alarms in the event of a real-world attack using the misuse-based technique, but it does not offer any additional information about the type of attack. The drawback of this technique is that it performs poorly in terms of accuracy because some attacks resemble normal data, or the extracted properties are difficult to distinguish between attack and normal data.

Over the last decade, machine learning algorithms have gained much attention from researchers in developing IDSs. Anwer, et al. [33] have proposed a method for selecting features that consider irrelevant and redundant features. The method applies different strategies based on the selection of filter and wrapper features. It achieves a high level of accuracy by selecting the minimum number of features. Experimental results are presented using the UNSW-NB15 dataset. Tian, et al. [34] have suggested a robust and sparse technique based on a one-class support vector machine (OSVM) to find samples that vary from the majority of data. The Ramp loss function has been used to enhance the performance of this model, making the approach more robust and sparser.

The NSL-KDD dataset is used in the study presented in [35]. The dataset in this research is normalized and discretized using the k-means technique. Feature selection is made using the Information gain algorithm and then submitted to the Naive Bayes machine learning algorithm. They discovered that the k-means clustering approach outperforms the mean and standard deviation discretization methodology. The data is sent to the information-gain technique after it has been labeled using the k-means approach, which employs scoring methods for nominal or weighting continuous qualities that are discriminated by applying the maximum entropy. The k-means technique cannot handle nonlinear or incomplete data, one of its key shortcomings. The system's accuracy and false-positive rate may be enhanced further.

Kan, et al. [36] have introduced a novel approach for intrusion detection in IoT networks called Adaptive Particle Swarm Optimization Convolutional Neural Network (APSO-CNN). The approach utilizes the PSO algorithm with a change of inertia weight to dynamically optimize the structure parameters of a one-dimensional CNN. To achieve this, the cross-entropy loss function value of the validation set, obtained from the initial training of the CNN, is utilized as the fitness value for PSO. This adaptive optimization process ensures efficient parameter tuning for improved performance. A new evaluation method is defined that considers both the prediction probability assigned to each category and the prediction label. This evaluation method enables a comprehensive comparison between the proposed APSO-CNN algorithm and manually set parameters for CNN (R-CNN). Furthermore, a comparison is conducted between the proposed APSO-CNN and three other well-known algorithms using five traditional evaluation indicators and accuracy statistical characteristics from ten independent experiments. The simulation results reveal that the APSO-CNN algorithm proves to be effective and reliable for multi-type IoT network intrusion attack detection tasks.
Andresini, et al. [37] have proposed an innovative intrusion detection method that focuses on analyzing the flow-based characteristics of network traffic data. Their approach leverages deep metric learning, which combines autoencoders and Triplet networks to create an effective intrusion detection model. During the training stage, two separate autoencoders are trained using historical normal network flows and attack data, respectively. The autoencoders are designed to reconstruct the original network flow data. Subsequently, a Triplet network is trained to learn an embedding of the feature vector representation of the network flows. This embedding ensures that each flow is positioned close to its reconstruction by the autoencoder associated with the same class (normal or attack) and far away from its reconstruction by the autoencoder of the opposite class. In the predictive stage, when presented with a new network flow, the method assigns it to the class associated with the autoencoder that provides the closest reconstruction of the flow in the embedding space. This process capitalizes on the learned embedding from the training stage and effectively detects potential signs of malicious activities in the network traffic. The results of their proposed methodology demonstrate superior predictive accuracy compared to competitive intrusion detection architectures when evaluated on benchmark datasets. The combination of deep metric learning, autoencoders, and Triplet networks empowers their intrusion detection approach to achieve impressive performance in detecting new instances of malicious behavior within network traffic.

Ravi, et al. [38] have presented an end-to-end model for network attack detection and classification, leveraging deep learning-based recurrent models. Their proposed approach involves extracting features from the hidden layers of recurrent models and utilizing a kernel-based principal component analysis (KPCA) feature selection method to identify optimal features. These optimal features from recurrent models are then combined and used for classification through an ensemble meta-classifier. Extensive experimental analysis and evaluation of the proposed method were conducted on multiple benchmark network intrusion datasets. The results demonstrated that the proposed approach outperformed existing methods as well as commonly used machine learning and deep learning models. In particular, the proposed method achieved a remarkable maximum accuracy of 99% for network attack detection and 97% for network attack classification when applied to the SDN-IoT dataset. Similarly impressive performances were obtained on other network intrusion datasets, including KDD-Cup-1999, UNSW-NB15, WSN-DS, and CICIDS-2017.

Talukder, et al. [39] have introduced a novel hybrid model that combines machine learning and deep learning techniques to achieve higher detection rates while ensuring dependable results. The proposed method focuses on efficient pre-processing by utilizing SMOTE for data balancing and XGBoost for feature selection. To evaluate the effectiveness of their developed method, they conducted a comparison with various machine learning and deep learning algorithms. The goal was to identify the most efficient algorithm to incorporate into the detection pipeline. Through benchmarked performance analysis criteria, they selected the most effective model for network intrusion detection. Their method was tested on two datasets, KDDCUP’99 and CIC-MalMem-2022, and produced remarkable results. The accuracy achieved was 99.99% for KDDCUP’99 and 100% for CIC-MalMem-2022, showcasing the superior performance of the proposed hybrid model. Additionally, their method exhibited no signs of overfitting or issues related to Type-1 and Type-2 errors.

Mohamed and Ejbali [40] have introduced a novel deep reinforcement learning model that effectively combines a SARSAs-based reinforcement learning algorithm with a deep neural network for intrusion detection systems. The primary objective of their proposed deep SARSA model is to enhance the detection accuracy of modern and complex attacks in network environments. To validate the performance of their method, they conducted experiments using two well-known benchmark datasets, NSL-KDD and UNSW-NB15. By comparing their approach with various classic machine learning and deep learning models, as well as other published results in the field, they demonstrated that their proposed approach outperforms the other models across multiple evaluation metrics, including accuracy, recall, precision, and F1-score. The integration of deep reinforcement learning, SARSAs-based algorithm, and deep neural networks has proven to be a successful strategy for achieving superior intrusion detection accuracy. The proposed approach addresses the challenges posed by modern and complex attacks in network environments, making it a valuable contribution to the field of network security.

IV. PROBLEM STATEMENT

The rapid pace of technological advancements in network and hardware devices presents significant challenges for the implementation and enhancement of intrusion detection systems (IDSs). To fully understand these challenges, it is important to delve into their specific details and implications. Firstly, the challenge of diversity arises from the continuous development of network protocols. As these protocols evolve, it becomes increasingly difficult to differentiate between normal and abnormal data traffic. This poses a significant hurdle for IDSs, as they need to accurately identify potential threats amidst a wide range of network activities. Another challenge is related to low-frequency attacks. The distribution of attack types is often imbalanced, with some occurring less frequently than others. This imbalance negatively impacts the detection precision of IDSs, particularly those utilizing data-driven approaches. It becomes more challenging to identify and accurately detect these low-frequency attacks, which can potentially evade detection and compromise the security of the network.

The adaptability of IDSs is also a key challenge. The dynamic and flexible nature of networks necessitates regular updates and modifications to IDS models. As the network environment changes, the IDS must adapt to the evolving landscape to maintain its effectiveness. Failure to do so, results in outdated detection models that are ineffective against new and emerging threats. Choosing the appropriate placement strategy for an IDS is another consideration. Organizations must carefully evaluate and select between centralized, distributed, and hybrid deployment strategies based on factors
such as financial constraints, computational capabilities, and time costs. Each strategy comes with its advantages and trade-offs, making the decision a critical one. As a final consideration, accuracy poses a significant challenge. Traditional IDS methods often fall short of providing a high degree of precision in detecting intrusions. To address this, a comprehensive and in-depth understanding of intrusion behavior becomes crucial. A complete knowledge of how intrusions manifest and evolve can significantly enhance IDS performance and ensure more accurate threat detection.

In summary, the implementation and improvement of intrusion detection systems face various challenges in today's technological landscape. These challenges include dealing with the diversity of network protocols, addressing low-frequency attacks, ensuring adaptability to changing network environments, making informed placement decisions, and improving accuracy through a deeper understanding of intrusion behavior. Overcoming these challenges requires innovative approaches and continuous research to develop robust and effective IDS solutions.

V. PROPOSED METHOD

This section discusses the suggested approach for detecting network intrusions. This section is divided into two subsections, where we explore the requirements and the proposed technique. As depicted in Fig. 2, the suggested NIDS system is divided into three major components.

- The infrastructure layer is composed of two distinct elements: software and hardware. Software elements can communicate with hardware, for example, OpenFlow switches. Hardware elements include switches and routers.
- The control layer regulates activities and data management in the network by creating or refusing each network flow.
- The application layer is responsible for all network management operations. These activities may be accomplished with the aid of a NIDS controller.

As illustrated in Fig. 3, the NIDS generated utilizing machine learning and deep learning algorithms typically entails three key processes, namely data preprocessing, training, and testing. For each of the potential approaches, the dataset is preprocessed and transformed into an algorithm-compatible format. Encoding and normalization are generally included in this step. Frequently, the dataset needs cleaning, which includes eliminating items with duplicate records and incomplete data. The preprocessed data is randomly separated into two parts: the training and testing datasets. Generally, the training dataset accounts for around 80% of the entire dataset, leaving 20% for testing. In the training stage, the deep learning algorithm is trained using the training dataset. The learning time of the method is affected by the complexity of the proposed model and the amount of the dataset. Deep Learning models often need additional training time owing to their deep and complicated underlying structures. After training the model, its performance is evaluated using the testing dataset and its predictions. NIDS models classify network traffic instances as benign (normal) or malicious (attack). The flowchart of the suggested approach is illustrated in Fig. 4. The steps of the proposed technique are described in the following.
A. First Phase: Load Dataset and Export it into Resilient Distributed Datasets (RDDs)

Numerous researchers have developed and evaluated the NIDS issue using the NSL-KDD or other datasets detailed in the assessment part of this proposal. A wide variety of attacks are included in the dataset. It includes 41 features classified into three major categories (traffic-based, content-based, and basic) and distinguished as normal or malicious.

B. Second Phase: Data Preprocessing

The features dataset includes values with varying scale ranges to mitigate the loss function during learning. These scales influence the gradient optimization process, thereby affecting learning rate optimization, as the model should rapidly reach a global or local minimum as a result. Min-Max normalization provides some benefits in comparison to conventional scaling. Min-Max scaling can deal with non-Gaussian feature distributions since anomaly detection applications do not need a certain distribution to follow, in contrast to the signature-based technique in NIDS. The Min-Max normalization strategy is presented to avoid the gradient from the un-smoothing route toward the global minimum, thereby improving the loss function. As illustrated in the following equation, it retrieves the column's lowest and maximum values, with output values ranging from 0 to 1.

\[ \text{Normalised Parameter (X)} = \frac{X - X_{\text{min}}}{X_{\text{max}} - X_{\text{min}}} \] (1)

Where \(X_{\text{min}}\) is the column's lowest value, \(X_{\text{max}}\) is the column's highest value, and \(X\) represents the initial data sample value. Also, we employed the Apache Spark system during this phase. Spark is a high-performance, general-purpose cluster computing system optimized for large-scale in-memory data processing. Spark follows the MapReduce programming paradigm but adds a data-sharing concept called Resilient Distributed Datasets, or RDD. A Spark was developed to be quick for iterative algorithms, enable in-memory storage, and perform well under load.

C. Third phase: Feature Selection

Feature selection forms an integral part of data preprocessing in intrusion detection. A network intrusion detection system is characterized by diverse features and a large amount of data. There are different attribute values for features in different categories, including duplicate features that complicate classification. The proliferation of redundant features reduces the efficiency of detection algorithms and increases the likelihood of false positives in intrusion detection. IDS accuracy and detection speed are increased by an efficient feature selection algorithm, which reduces the dimensionality of network data. This paper uses Convolutional Neural Network (CNN) and Long Short-Term Memory (LSTM)
algorithms for feature selection and training models. CNN is a deep neural network that is composed of three main layers.

- Convolutional layer: In this layer, the input data is processed with a set of filters known as convolutional kernels. A feature map is produced as each filter is applied to the input data. The final output of the convolution layer can be obtained by stacking all the produced feature maps together.

- Pooling layer: It performs subsampling on the feature maps, resulting in reduced dimensionality. The most common methods of pooling are average pooling and maximum pooling.

- Fully connected layer: The output of the previous layers is transformed into a vector that can be used as an input for the next layer.

Recurrent Neural Network (RNN) is a deep learning model driven by supervised learning. Using a traditional RNN, it was possible to predict the temporal training data; however, it encountered difficulties when dealing with gradient explosions. LSTM was proposed as a solution to this problem. An LSTM model replaces the hidden RNN units with a memory function. The LSTM model consisted of three important gates: forget, input, and output gates.

D. Fourth Phase: Train with the Training Dataset

Two causal convolution layers, two dense layers, and a softmax layer are used in the training and optimization phase for the multi-class classification task. In order to avoid overfitting, we employ maximum global pooling, batch normalization, and dropout layers. Adam optimizer is used to update weights and optimize the cross-entropy loss function. It is a combination of two stochastic gradient descent approaches, including Root Mean Square Propagation (RMSProp) and Adaptive Gradient Algorithm (AdaGrad). In particular, the training and optimization phase comprises the following layers:

- First causal convolution layer: The input vectors are convolved with 64 filters of three sizes across the input vectors.

- Second 1D causal convolution layer: A total of 128 filters are used, each with a size of 3. Prior to pooling, this layer enables the model to learn more complex features.

- 1D global maximum pooling layer: The maximum value of the filter is replaced with the data covered by the filter. The maximum value prevents the learned features from overfitting.

- Batch normalization layer: The data are normalized before they are sent to the next layer.

- Fully connected dense layer: It utilizes 128 hidden units with a dropout rate of 30%.

- Fully connected dense layer with softmax activation function: multi-class classification is achieved by producing five units for each of the five traffic categories.

VI. RESULTS AND DISCUSSION

As choosing the appropriate NIDS data to assess the system is critically important, the data was chosen prior to simulation. While there are a number of publicly available datasets, some of them comprise out-of-date, illogical, inadequately validated, and potentially unrecoverable intrusions. Amazon Web Services (AWS) developed the CSE-CIC-DS2018 [41] dataset to address these limitations and generate modern traffic patterns. It includes a variety of datasets that are suitable for evaluating anomaly-based approaches. CSE-CIC-DS2018 highlights real-time network activities and includes a variety of intrusion detection modes. The data packet payload is calculated by encapsulating the inner network traces as a whole network. Several intrusion profiles are contained in this dataset, which can be applied to a variety of network protocols and topologies. IDS2017 criteria were applied to enhance this dataset. There are currently seven intrusion strategies and two profiles included in IDS2018, a publicly available dataset. IDS2018 contains 80 statistical variables, such as the number of bytes, volume, and packet length. It is accessible via the Internet, which contains approximately 5 million records, and is available in two formats: PCAP and CSV. PCAP is commonly employed to obtain new functions, while the CSV format is typically used in artificial intelligence applications. This dataset represents seven types of attacks: Botnet, Web attacks, Heartbleed, Infiltration, Brute-force SSH, DDOS attacks, and Brute-force DOS attacks.

The creation of the CIC-IDS-2017 and CSE-CIC-IDS-2018 datasets has garnered significant interest among researchers, leading to the implementation of various classifiers using these datasets. The datasets’ specifications are detailed in Table I. The files within the dataset are utilized for both binary and multi-class classification tasks. An ideal Intrusion Detection System (IDS) is one that can precisely detect each type of attack. To achieve this, building an efficient IDS requires merging the files in the dataset to cover a wide range of attack categories [42]. The CIC-IDS-2017 and CSE-CIC-IDS-2018 datasets exhibit certain limitations related to the data samples and files generated through network flow analysis, which can be listed as follows:

- Tedious data processing: The data samples generated by network flow analysis are stored in files, and processing these files can be a time-consuming and tedious task, especially since each file contains a large number of data instances.

- Dataset size and computing time: Merging the files in the dataset to include all attack labels can lead to an increase in the dataset’s size. This, in turn, results in more computing and processing time, making it challenging to handle large datasets efficiently.

- Missing and redundant data: The dataset contains some missing and redundant data records, which can affect the quality and accuracy of the analysis performed on the data.

- High-class imbalance: Both CIC-IDS-2017 and CSE-CIC-IDS-2018 datasets suffer from the issue of high-class imbalance. This means that some attack types may...
have significantly fewer instances compared to others, leading to lower accuracy and higher False Positive Rate (FPR) for the system.

There are 50 computers involved in the dataset-attacking infrastructure, while 30 servers and 420 terminals are utilized by the attackers. CSE-CIC-DS2018 data represent a system log with 80 attributes extracted from CICFlowMeter-V3 and network traffic captured from AWS. There is approximately 400 GB of data in CSE-CIC-DS2018, which is larger than CIC-DS2017 in terms of size. Table II compares the CSE-CIC-DS2018 and CIC-DS2017 datasets with respect to sample size. The number of CSE-CIC-DS2018 samples has increased significantly compared to CIC-DS2017, especially in the Infiltration and Botnet attacks, where the number of samples increased respectively by 4497 and 143.

Using the CSE-CIC-DS2018 ID dataset, the effectiveness of our mechanism was evaluated by examining error rate, accuracy, true negative, false negative, true positive, and false positive. The confusion matrix is used to determine the difference between the actual and predicted classifications. Categorization results can be divided into two groups: normal and abnormal. Table III provides an overview of the confusion matrix. It is necessary to measure four levels of criticality in the confusion matrix.

- True negative: In this case, the model correctly predicts the negative outcome.
- False positive: In this case, the classifier considers normal traffic as abnormal.
- False negative: When an IDS fails to detect an actual attack.
- True positive: It is an actual intrusion successfully discovered by the IDS.

Based on the specifications given above for the confusion matrix, we can calculate the output of the system. An IDS is analyzed based on FAR and DR as key and common metrics. FAR represents the sum of misclassified regular incidents, whereas DR signifies the number of intrusions identified by the model. As DR rises and FAR decreases, we claim our approach is superior to traditional approaches.

\[
FAR = \frac{FP}{(TN + FP)}
\]

\[
DR = \frac{TP}{(TP + FN)}
\]

The performance of the classifier on CSE-CIC-DS2018 is summarized in Table IV. A random search hyperparameter optimization technique was used to generate the results. The ensemble classifier XGB significantly enhances classification effectiveness, achieving an accuracy rate of 85%. The tree-based classifier provides a higher level of accuracy than ensemble-based classifiers.

<table>
<thead>
<tr>
<th>Dataset</th>
<th>Type</th>
<th>Number of classes</th>
<th>Features</th>
<th>Victim Infrastructure</th>
<th>Attack Infrastructure</th>
<th>Duration of Capture</th>
</tr>
</thead>
<tbody>
<tr>
<td>CSE-CIC-DS2018</td>
<td>Multi-class</td>
<td>18</td>
<td>80</td>
<td>420 PCs, 30 servers</td>
<td>50 PCs</td>
<td>Ten days</td>
</tr>
<tr>
<td>CIC-DS2017</td>
<td>Multi-class</td>
<td>15</td>
<td>80</td>
<td>Three server, one firewall, two switches, 10 PCs</td>
<td>Four PCs, one router, one switch</td>
<td>Five days</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Dataset</th>
<th>Web attacks</th>
<th>Infiltration</th>
<th>Brute force</th>
<th>Botnet</th>
<th>DoS</th>
<th>DDoS</th>
<th>Normal</th>
</tr>
</thead>
<tbody>
<tr>
<td>CSE-CIC-DS2018</td>
<td>929</td>
<td>161,936</td>
<td>380,950</td>
<td>286,195</td>
<td>954,311</td>
<td>687,740</td>
<td>6,112,149</td>
</tr>
<tr>
<td>CIC-DS2017</td>
<td>2182</td>
<td>37</td>
<td>13,840</td>
<td>1968</td>
<td>252,665</td>
<td>128,024</td>
<td>1,743,181</td>
</tr>
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</table>

<table>
<thead>
<tr>
<th>Predicted outcome</th>
<th>Actual value</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>Abnormal</td>
</tr>
<tr>
<td></td>
<td>Normal</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Classifier</th>
<th>FAR</th>
<th>DR</th>
<th>F-score</th>
<th>Recall</th>
<th>Precision</th>
</tr>
</thead>
<tbody>
<tr>
<td>DT</td>
<td>7.81</td>
<td>0.89</td>
<td>0.87</td>
<td>0.88</td>
<td>0.87</td>
</tr>
<tr>
<td>XGB</td>
<td>9.1</td>
<td>0.84</td>
<td>0.83</td>
<td>0.83</td>
<td>0.84</td>
</tr>
<tr>
<td>LR</td>
<td>11.5</td>
<td>0.80</td>
<td>0.79</td>
<td>0.80</td>
<td>0.78</td>
</tr>
<tr>
<td>Proposed classifier</td>
<td>2.6</td>
<td>0.96</td>
<td>0.981</td>
<td>0.976</td>
<td>0.968</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>References</th>
<th>False alarm rate</th>
<th>Accuracy</th>
</tr>
</thead>
<tbody>
<tr>
<td>[43]</td>
<td>1.1</td>
<td>96%</td>
</tr>
<tr>
<td>[44]</td>
<td>1.3</td>
<td>96%</td>
</tr>
<tr>
<td>[45]</td>
<td>8.5</td>
<td>96%</td>
</tr>
<tr>
<td>[46]</td>
<td>5</td>
<td>95%</td>
</tr>
<tr>
<td>[41]</td>
<td>0.93</td>
<td>90.2%</td>
</tr>
<tr>
<td>[47]</td>
<td>0.97</td>
<td>94%</td>
</tr>
<tr>
<td>Our method</td>
<td>1.5</td>
<td>98.2%</td>
</tr>
</tbody>
</table>
In order to use the proposed mechanism to capture both temporal and spatial features efficiently, recurrent layers were introduced following the CNN layers in order to capture both features. In this manner, we attempted to avoid the vanishing gradient and explosion problem, resulting in an improved ability to capture temporal and spatial dependencies and learn efficiently from sequences of variable extent. The number of variables in large-scale data (imbalance), while exceeding the sample size, is not well suited to traditional machine-learning classifiers. This model is suitable for high-dimensional datasets due to its scale invariance. Nevertheless, the most significant improvement was achieved with advanced deep learning approaches such as CNNRNN, which detected misuse with 97% accuracy. This performance improvement can be attributed to the long-term dependencies between the nonlinear features, and details on their implementation can be found in supplementary materials.

A summary of the results obtained using existing methods for the CSE-CIC-DS2018 dataset is provided in Table V. Several preliminary results are available since these datasets are generated following the KDD and DARPA datasets. Considering current simulation results, optimal values for accuracy and FAR was calculated for each phase. The accuracy and FAR of our method are superior to those of conventional methods. The reason for this is the execution of the deep learning algorithm. Due to the differences in the quantity of data distributions, preprocessing procedures, and sampling methods, the similarities should only be used as a source of reference. Therefore, measuring a simple metric, such as the amount of time spent on testing or training is rarely appropriate. Although the suggested method demonstrated superior performance in some respects, it remains questionable whether it can perform better in all respects than other approaches. The proposed solution enables exceptional network protection as well as easy identification of malicious threats.

In comparison to previous approaches, the proposed method demonstrates superiority based on simulation results. It achieves an exceptional level of accuracy in detecting malicious attacks, as evidenced by its outstanding performance on the widely recognized CSE-CIC-DS2018 dataset. This indicates that the proposed NIDS has surpassed the capabilities of existing methods in terms of accuracy and effectiveness. By capturing packets from the network and performing careful preprocessing to reduce their size while retaining crucial information, the proposed method optimizes the input data for the deep learning algorithm. The utilization of machine learning characteristics enhances the NIDS’s ability to learn and test potential intrusion patterns, further contributing to its superior performance.

The hybrid feature selection approach introduced in the proposed method addresses the challenge of identifying critical attributes necessary for constructing robust classifiers. This feature selection process, combined with rule assessment and detection steps, provides a comprehensive and effective framework for intrusion detection. In summary, the proposed method stands out among previous approaches by leveraging deep learning, hybrid feature selection, and a carefully designed process flow. Its exceptional accuracy in detecting malicious attacks on the CSE-CIC-DS2018 dataset indicates its superiority over existing methods. The proposed method offers significant advancements in network intrusion detection and presents a promising solution to bolster the security of networks and systems in today’s digital landscape.

VII. CONCLUSION

Cyber security has become a paramount area of research in modern society, given the indispensable role of networks. Within this domain, Intrusion Detection Systems (IDSs) play a pivotal role in monitoring the status of software and hardware on a network. However, IDSs continue to face challenges in accurately identifying potential threats, minimizing false alarms, and enhancing detection accuracy. To address these challenges, extensive research has been dedicated to developing IDSs that harness the power of machine learning. In our study, we have devised a cutting-edge IDS methodology based on a layered Recurrent Neural Network (RNN). This approach leverages the strengths of deep learning to adeptly predict and classify unauthorized intrusions. The layered RNN effectively captures local features, while the recurrent RNN seamlessly incorporates temporal characteristics, substantially elevating the performance of our IDS system. Through comprehensive evaluations on the esteemed CSE-CIC-DS2018 dataset, our proposed method has demonstrated superior performance over previous approaches. The simulation results unequivocally establish the exceptional accuracy of our IDS in detecting malicious attacks. The integration of deep learning techniques and hybrid feature selection enables our IDS, named DeepShield, to outperform traditional machine learning-based methods, providing a reliable defense against network intrusions. The significance of DeepShield extends beyond the system itself. The higher accuracy achieved by our NIDS translates to more robust network security, helping organizations proactively safeguard their critical data and online services. As cyber threats continue to evolve, the effectiveness of intrusion detection becomes increasingly crucial, and our approach contributes to a safer digital environment. Moreover, the versatility of our methodology allows for scalability and adaptability to various network infrastructures and environments. This adaptability ensures that DeepShield can be applied in diverse security scenarios, making it a valuable tool for network administrators and cyber security professionals.

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Deep-Learning-based Analysis of the Patterns Associated with the Changes in the Grit Scores and Understanding Levels of Students

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Abstract—The purpose of this study is to classify the pattern of the understanding level changes for university students during class term, and analyze the relation between them and the changes in the Grid score before and after the class term. Dynamic time warping was applied for classification of the understanding level, and the decision tree was applied to analyze the relation between the changes in the understanding level and that in the Grid score. As a result, it was shown that a large variety of the patterns of changes in the understanding level, and the relations between the understanding level and Grid score cover a wide variety, too. It is necessary to take these results for conducting effective lectures.

Keywords—Time series; dynamic time warping; decision tree; Grit

I. INTRODUCTION

With the spread of information, communication equipment, the online management of student learning data for learning analysis has become popular. Furthermore, with the advent of machine learning and time series analysis, research on learning analysis and education data mining[1] has been actively conducting for improving education quality. Studies that predict students’ careers and probability of dropping out of school analyze educational data as time series data using the state transition model and machine learning algorithms [2] [3]. In the knowledge accumulation approach to learning, learners reflect on their understanding levels before undertaking a task, and the teacher visualizes the students’ understanding levels from the results. Accordingly, learners can improve their comprehension, and the teacher can make improvement to the contents of their lecture accordingly [4]. Regarding knowledge accumulation, it not only focuses on cognitive factors, which are measured by academic assessments such as examinations, but it also highlights the importance of non-cognitive factors such as self-adjustment capability.

Many reports have confirmed that non-cognitive factors influence not only social behavior but also academic performance [5] [6]. Additionally, they can potentially strongly impact cognitive capabilities such as academic ability and intelligence quotient [7]. The Grit score [8], which measures a learner’s self-control in pursuing goals and enthusiasm in overcoming difficulties, is a particularly strong predictor of their educational achievements. This score has attracted wide attention globally in the field of education, with many reports investigating its impact on academic grades in higher educational institutions [9] and colleges [10][11]. Specifically, it has been reported that the level of perseverance measured by the Grit score is related to both a learner’s English acquisition time and proficiency [12]. In the field of information technology in education, the relation between information and communication technology and students’ self-efficacy and information literacy has been studied [13]. In a study for medicine Residents, higher Grit scores are reported associated with less burnout [14]. For these reasons, the Grit score has attracted widespread attention. It is also known that Grid score changes in the case of situation, and Grid score at the school base has large variance, and it is higher in the sports area than in school and life [15]. In my previous study [16], data on students’ understanding levels were treated as time series data, and the effect of the understanding levels in a previous lecture on the understanding levels in a new lecture was reported using time series analysis. The study discussed the relation between maintaining the understanding level and decrease in retention of learned information over time. Furthermore, it revealed that correspondence with the levels of learning retention in the learning pyramid could improve the quality of education. Moreover, the differences in learners’ grades had a significant effect on their cumulative understanding levels. Additionally, a study on students’ Grit scores during clinical practice in a physical therapy training course showed improvements in the Grit score improvement and variations in their understanding levels [17]. Regarding a university-lecture setting, which involves many students, there are various factors that affect students’ understanding levels. As previously mentioned, Grit scores can change, and the variations in students’ motivation and understanding levels can potentially affect such changes.

It can be difficult to assess students’ understanding levels in a lecture-type setting where there are more than a hundred registered students, compared to hands-on practice or seminar-type settings that offer more scope for one-on-one interactions. Therefore, it is important that lecturers are acquainted with patterns in students’ understanding levels or changes in their Grit scores to conduct their lectures more effectively. Furthermore, by clarifying the relation between the understanding level and Grit score, lecturers can identify students who may need additional support or interventions and assist them accordingly. This study aims to classify the patterns found in the changes between students’ understanding levels and Grit scores in both traditional lecture-type classroom and practical hands-
on settings. Dynamic time warping (DTW) [18], a time-series analysis algorithm, was used for classification. Furthermore, the decision tree algorithm was used to analyze the relation between the changes in students’ understanding levels and Grit scores.

The remainder of this paper is organized as follows. Section II provides a succinct background of Grid score and DTW, and introduces the proposal analysis for students’ understanding level and Grid score. Section III introduces the experimental data and discusses the results. Finally, Section IV concludes the paper and future work.

II. Method

This section provides a brief overview of certain time-series analytics and evaluation indices that are frequently used in this study, and proposed procedure for analysis of the patterns associated with the changes in the Grit scores and understanding levels of students.

A. Grit Score

The Grit score consists of the subscales “consistency in interest”, which represents passion, and “perseverance in effort”, which represents persistence; the average value of these scores is used as the Grit score. A high passion score corresponds to carefully working toward a single goal for a long time, and a high persistence score corresponds to displaying determination in the face of difficulties and not being satisfied till a specified result is achieved. In this study, the relation between students’ understanding levels and passion and persistence scores was also analyzed.

B. Dynamic Time Warping (DTW) Model

The DTW model finds the optimal alignment between two-time sequences by the brute force approach. This technique is often used to analyze human behavior data, such as voice data or walking data. In the field of education, clustering methods such as DTW have been used to study the detection of students’ Grit scores[19]. The distance between two time-series data points $(x_i, y_i)$ is defined as follows:

$$DTW(x, y) = \min_{i=1}^{p} \sum |x_i - y_i|$$

Moreover, it is possible to classify multiple time-series data by applying the k-means method in DTW[20]. In the current study, DTW has been applied to determine the patterns in the changes in the understanding levels of students.

C. Proposal Analysis based Deep-Learning Regarding Students’ Understanding Levels and Grid Score

In this study, the changes in students’ understanding levels in different types of lecture settings were observed. Students were asked to rate their understanding levels for a day’s lecture using one of these five options: 100%, 80%, 60%, 40%, or less than 20%, 0% denoted a student’s absence. Next, the students’ Grit score data over more than 10 lectures were collected as time series data, and DTW was applied to determine any patterns in the changes in the students’ understanding levels. Furthermore, the decision tree algorithm was applied to obtain the rules that explained the changes in both the understanding level and those in the Grit score before and after the classes to visualize the relation between these changes. The process of data generation for the Grit score or the students’ understanding levels are described in steps 1–4 (Fig. 1), and the process of data analysis is described in steps 5–8.

Fig. 1. Procedure of investigation (steps 1–4): The horizontal axis indicates the number of lectures, and the vertical axis indicates the understanding level. $Grit_{before}$ will be collected before the start of lecture term, and $Grit_{after}$ will be collected after the end of lecture term.

Step1 Measure the students’ $Grit_{before}$ scores on the day of the first lecture for each subject.

Step2 Ask each student to select the option (out of five) corresponding to their understanding level after each lecture.

Step3 Implement step 2 in all the classes and save the data on the understanding level scores as time series data.

Step4 Measure the students’ $Grit_{after}$ scores on the day of the final lecture for each subject.

Step5 Apply DTW to the time series data obtained in step 3 and predict the patterns in the changes in the students’ understanding levels.

Step6 Classify the Grit scores obtained in steps 1 and 4 as either a “passion score” or “persistence score,” compare the $Grit_{before}$ and $Grit_{after}$, and convert the increase or decrease in the scores to “up” or “down,” respectively.

Step7 Generate learning data using the students’ understanding levels in each lecture as the explanatory variable and the changes (increases/decreases) in the Grit scores comprising the passion score and persistence score as the objective variable.

Step8 Apply the decision tree algorithm to the learning data obtained in step 7 and extract the rules for the relationships between the changes in the Grit scores and understanding levels.

Based on the properties and components of the Grit score, three hypotheses regarding the patterns in the relation between the changes in the understanding levels and Grit scores are proposed.

$H_1$ The Grit (especially passion) scores increase when the understanding levels are high.

$H_2$ Maintaining diligent class participation results in high Grit (especially persistence) scores even if the understanding levels are low.

$H_3$ The Grit (especially passion) scores decrease when the understanding levels are low.

In this study, these hypotheses were tested through analyses performed through DTW and the decision tree algorithm.
III. RESULTS AND DISCUSSION

This section provides the detail of experimental data and discuss the results. First, the results of applying DTW to each data regarding students’ understanding levels are reported as described in Step 5 of the proposed procedure. Second, the relation between changes in the understanding level and those in the Grit scores are by use of decision tree reported as explained in Step 6.

A. Experimental Data

The collected experimental data are described in this section. To obtain data for this research, a survey for six classes was conducted in 2021. In this investigation, 328 time-series data were collected, wherein participants 1, 2, and 3 were in a lecture-type setting, and participants 4, 5, and 6 were in a practical-type setting. Details of the collected data include the type of class setting and the number of students in each class, as shown in Table I.

<table>
<thead>
<tr>
<th>Participant</th>
<th>Class Setting</th>
<th>Number of Students</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>Lecture</td>
<td>103</td>
</tr>
<tr>
<td>2</td>
<td>Lecture</td>
<td>69</td>
</tr>
<tr>
<td>3</td>
<td>Lecture</td>
<td>70</td>
</tr>
<tr>
<td>4</td>
<td>Practical</td>
<td>30</td>
</tr>
<tr>
<td>5</td>
<td>Practical</td>
<td>31</td>
</tr>
<tr>
<td>6</td>
<td>Practical</td>
<td>17</td>
</tr>
</tbody>
</table>

B. The Results of Applying DTW to Data of Students’ Understanding Levels

The analysis results of the acquired data and the discussion are presented. First, the results of applying DTW to each data regarding students’ understanding levels with different numbers of clusters (2, 3, and 4) are shown. Fig. 2 to Fig. 4 show the results of the practical-type classes, and Fig. 5 to Fig. 7 show the results of the lecture-type classes. In these figures, the horizontal axis indicates the number of the lectures, and the vertical axis indicates the understanding levels.

Considering that an understanding level of 0 indicates the absence of a student for a class, only cases wherein the understanding level was 0.2 or more have been discussed. In the case of practical-type classes, when the cluster number was two, the data were classified into groups with either red lines or blue lines. In both the group with the red lines and that with the blue lines, the understanding levels were only 1.0 in the mid and later periods, with them transitioning between 0.4 and 1.0. When the cluster number was three, the data were classified into groups with either red lines, blue lines, or green lines. In the group with the red lines, all kinds of understanding levels were included. The group with the blue lines included understanding levels between 0.6 and 1.0 while that with the green lines included understanding levels of all kinds and noticeably included absenteeism. When the cluster number was four, the data were classified into groups with red, blue, green, or yellow lines. The groups with red or blue lines resembled the groups for the cluster number of three. The group with the green lines included understanding levels between 0.2 and 0.6 and high understanding levels in the starting and later periods.
while the group with yellow lines included low understanding levels.

Regarding lecture-type classes, when the cluster number was two, the data were classified into groups with either red or blue lines. Most of the blue lines were positioned at the upper side while most of the red lines were positioned at the lower side. Each group either contained high understanding levels or low understanding levels.

When the cluster number was three, the data were classified into groups with either red, blue, or green lines. The group with red lines included high understanding levels (more than 0.4), and the group with blue lines included various understanding levels of either other degrees or 0, indicating the presence of frequent absenteeism. The group with the green lines included understanding degrees that were both high and low. When the cluster number was four, the data were classified into groups with red, blue, green, or yellow lines. The groups with red or blue lines resembled the groups for the cluster number of three. The group with the green lines included understanding levels between 0.2 and 0.6 and high understanding levels in the starting and later periods while the group with yellow lines included low understanding levels.

### C. The Relation between Changes in the Understanding Level and Those in the Grit Scores

Under the application of DTW, many trends were observed in the changes in the understanding levels. Here, the relation between more specific changes in the understanding level and those in the Grit scores will be discussed. Table II to Table V presents the rules obtained by applying the decision tree algorithm to analyze the relation between changes in Grit scores and those in the understanding level. Table II to Table III presents the results of practical type, and Table IV to Table V presents the results of lecture type.

**TABLE II. PRACTICAL TYPE: RELATIONS AND SUMMARY BETWEEN THE CHANGES IN THE GRI T SCORES AND UNDERSTANDING LEVELS**

<table>
<thead>
<tr>
<th>Practical type/Grit</th>
<th>Rule 1: 10th&lt;0.7, up (0.16)</th>
</tr>
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<tbody>
<tr>
<td>Rule 2: 6th ≥0.9, 10th ≥0.7, 11th &lt;0.9, up (0.13)</td>
<td></td>
</tr>
<tr>
<td>Rule 3: 6th ≥0.9, 10th ≥0.7, 11th ≥0.9, down (0.07)</td>
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<tr>
<td>Rule 4: 6th &lt;0.9, 10th ≥0.7, down (0.04)</td>
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</table>

- Rule 1: The Grit scores of 16% of the students increased even though their understanding levels in the tenth lecture were less than 70%.
- Rule 2: The Grit scores of 13% of the students increased when their understanding levels were more than 90% in the sixth lecture and less than 90% in the eleventh lecture.
- Rule 3: The Grit scores of 7% of the students decreased when their understanding levels were more than 90%, 70%, and 90% in the sixth, tenth, and eleventh lectures, respectively.
- Rule 4: The Grit scores of 4% of the students decreased when their understanding levels were less than 90% in the sixth lecture and more than 70% in the tenth lecture.

First, the data classification results for the practical-type setting will be discussed. Regarding the Grit score, the following patterns were observed: 1) the Grit scores increased even when the understanding levels were low in the latter period; 2) the Grit scores increased even when the understanding levels were high in the middle period and low in the later period; and 3) the Grit scores decreased even though the understanding levels were high in the middle and latter periods. Concerning the persistence score component of the Grit score,
the following patterns were noted: 1) the persistence scores increased when the understanding levels were low in the former period and increased in the latter period; 2) owing to the diligent student participation, the persistence scores ultimately increased when the understanding levels decreased and increased; and 3) the persistence scores increased even though the understanding levels were less than 20% in the former period; 4) the persistence scores ultimately decreased when the understanding levels decreased in the former period; and 5) the persistence scores ultimately increased even when there were increases and decreases in the understanding level, owing to the diligent student participation. Regarding the passion score component of the Grit score, the following patterns were noted: 1) the passion scores decreased even though the understanding levels gradually increased; 2) the passion scores increased when the understanding levels gradually increased; and 3) the passion scores decreased when the understanding levels were low in the former period.

Now, the data classification results for lecture-type settings will be discussed. Regarding the Grit score, the following patterns were observed: 1) the Grit scores decreased when the understanding levels were low in the former period; 2) the Grit scores decreased even though the understanding levels were low in the former period and high in the latter period; 3) the Grit scores ultimately increased when the understanding levels were low in the former period and high in the latter period, owing to the diligent student participation; 4) the understanding levels were low throughout, but there was diligent participation, resulting in the Grit scores ultimately increased. Concerning the persistence score component of the Grit score, the following patterns were observed: 1) the persistence scores increased even though the understanding levels were low in the latter period; 2) the understanding levels were low in the former period and average in the latter period, and the persistence scores finally decreased; 3) the persistence scores finally decreased, even though the understanding levels were high in both the former and latter periods; and 4) the persistence scores ultimately increased, when the understanding levels were high throughout the entire period. Regarding the passion score component of the Grit score, the following patterns were observed: 1) the passion scores decreased when the understanding levels were less than average in the middle period; 2) the passion scores ultimately increased when the understanding levels were low in the former period and high in the latter period; and 3) the passion scores ultimately increased when the understanding levels were average in the former period and even when they were low in the latter period. The above observations revealed that changes in the understanding levels and Grit scores varied depending on the students and type of class setting. Moreover, the hypotheses described in chapter 2 were verified from the data analysis; and additionally, there were some students whose Grit score decreased even when their understanding level was high.

### IV. Conclusion

This research has made contributions to the viewpoint of comparative study for time series data. The changes in students' understanding levels over more than ten lectures were considered as time series data, and both lecture-type and practical-type class settings in a university were analyzed. First, changes in the patterns of students' understanding levels were analyzed through DTW. The change patterns were quite diverse; thus, subdividing them was necessary. Next, rules...
TABLE V. LECTURE TYPE: RELATIONS AND SUMMARY BETWEEN THE CHANGES IN THE GRI T SCORES (PERSISTENCE AND PASSION) AND UNDERSTANDING LEVELS

<table>
<thead>
<tr>
<th>Lecture type: Grit(Passion)</th>
<th>Rule 1: 13th ≥0.5, down (0.08)</th>
<th>Rule 2: 3rd &lt;0.7, 13th ≥0.5 down (0.08)</th>
<th>Rule 3: 3rd ≥0.7, 4th &lt;0.7, 13th ≥0.5, down (0.05)</th>
<th>Rule 4: 3rd ≥0.7, 4th &lt;0.7, 13th ≥0.5, 13th &lt;0.7 up (0.06)</th>
<th>Rule 5: 3rd ≥0.7, 4th &lt;0.7, 6th ≥0.9, 13th ≥0.7, 13th &lt;0.9 down (0.01)</th>
<th>Rule 6: 3rd ≥0.7, 4th &lt;0.7, 6th &lt;0.9, 13th ≥0.7, 13th &lt;0.9 up (0.1)</th>
<th>Rule 7: 3rd ≥0.7, 4th &lt;0.7, 13th ≥0.9, up (0.08)</th>
</tr>
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<tbody>
<tr>
<td>Rule 1: The Grit (Persistence) scores of 8% of the students increased when their understanding levels were less than 50% in the third lecture and more than 50% in the thirteenth lecture.</td>
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<tr>
<td>Rule 3: The Grit (Persistence) scores of 5% of the students decreased when their understanding levels were more than 70 and 50% in the third and thirteenth lectures, respectively.</td>
<td>Rule 4: The Grit (Passion) scores of 6% of the students increased when their understanding levels were more than 70% in the third and fourth lectures and between 50 and 70% in the thirteenth lecture.</td>
<td>Rule 5: The Grit (Persistence) scores of 1% of the students decreased when their understanding levels were more than 70% in the third and fourth lectures, more than 50% in the sixth lecture, and between 70 and 90% in the thirteenth lecture.</td>
<td>Rule 6: The Grit (Persistence) scores of 1% of the students increased when their understanding levels were more than 70% in the third and fourth lectures, less than 90% in the sixth lecture, and between 70 and 90% in the thirteenth lecture.</td>
<td>Rule 7: The Grit (Persistence) scores of 8% of the students increased when their understanding levels were more than 70% in the third and fourth lectures and more than 90% in the thirteenth lecture.</td>
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</table>

reflecting the relation between changes in the Grit score before and after students participated in classes and changes in their understanding level were acquired through the decision tree algorithm. The proposed hypotheses were verified through data analysis, and noteworthy, unexpected patterns were extracted. The analysis results clearly showed that when conducting classes, the various relations between the changes in understanding levels and Grit scores of students should be considered.

The limitations of this study are two points. The first point is about the experimental data. Since students’ data used in this study limited to the one university, it is necessary to try analyzing the case of the other university. The second point is the evaluation index. Since the understanding level is subjective scale that is students evaluate themselves, objective scale has to be applied. In addition, DTW was used for clustering of time series data and other statistical methods are necessary, as the future work.

REFERENCES


