# Effective Voice Frame Shrinking Method to Enhance VoIP Bandwidth Exploitation

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***Abstract*—The traditional telecommunication system (e.g. landline telephone system) are increasingly being replaced by Voice over Internet Protocol (VoIP) systems because of the very low or free rate. However, one of the main handicaps of VoIP adoption is the inefficient bandwidth exploitation issue. A key approach to handle this issue is packet multiplexing. This article proposes a new VoIP packet payload compression method that enhances bandwidth exploitation over Internet Telephony Transport Protocol (ITTP) protocol. The proposed method is called payload shrinking over ITTP (ITTP-PS). As the name implies, the proposed ITTP-PS method shrinks the VoIP packet payload based on a certain mechanism. The ITTP-PS method has two entities, namely, sender ITTP-PS (S-ITTP-PS) and receiver ITTP-PS (R-ITTP-PS). The main function of the S-ITTP-PS entity is to shrink the VoIP packet payload, while the main function of the R-ITTP-PS entity is restoring the VoIP packet payload to its normal size. To perform the R-ITTP-PS entity function, the ITTP-PS method will reemploy the flag bits in the IP protocol header. The ITTP-PS method has been implemented and compared with traditional ITTP protocol without shrinking the VoIP packet payload. The comparison based on the VoIP packet payload shrinking ratio and isochronous calls capacity improvement ratio. The result showed that the VoIP packet payload shrinking ratio has enhanced by up to around 20%, while the isochronous calls capacity improvement ratio has enhanced by up to around 9.5%. Therefore, enhancing the VoIP bandwidth exploitation over ITTP protocol.***

***Keywords*—VoIP; VoIP protocols; ITTP protocol; payload compression; bandwidth exploitation**

## I. Introduction

The wide spreading of the Internet has produced a massive number of technologies such as Voice over Internet Protocol (VoIP) [1][2]. As the name implies, VoIP is an IP based technology to make voice calls. VoIP has become ubiquitous and progressively replaces the old public switched telephone network. The ubiquitous of VoIP resulting from i) the big number of VoIP applications that provide a low rate or free VoIP calls, ii) VoIP applications can be used by any handheld device such as mobile phone, iPad, and laptop, and iii) the useful services that can be accompanied by VoIP calls including interactive voice recognition and transfer the voicemail to email [3][4]. Nevertheless, bandwidth exploitation and quality of service are the two main issues that slow the VoIP propagation [4][5]. This article focuses on VoIP application bandwidth exploitation.

The main cause of the inefficient exploitation of bandwidth is the large header of the VoIP packet in comparison to VoIP packet payload [6][7]. On one hand, VoIP uses a codec to generate a VoIP packet's payload. Codec is hardware or software that captures analog voice signals, converts voice signals into digital signals, compresses the resulting digital data to save bandwidth, and produces a voice frame (a packet's payload). Typically, given that VoIP technology is delay-sensitive (the acceptable delay is 150 ms), the codec produces a voice frame within a short period. Accordingly, the resultant voice frame size typically ranges from 10 bytes to 30 bytes depending on the used codec. Table I shows some of the renowned voice codecs [8][9]. On the other hand, The VoIP packet header constitutes of IP protocol and media transfer protocol such as the 6-bytes internet telephony transfer protocol (ITTP) and the 12-bytes RTP protocol. In the case of ITTP, adding 26 bytes ITTP/IP header to the small payloads (10 to 30 bytes) lead to substantial bandwidth wasting between around 47% and 73% [10][11]. Fig. 1 shows the VoIP packet format when using ITTP protocol. Several approaches have been proposed by the researchers to improve VoIP bandwidth exploitation such as VoIP packet multiplexing, VoIP packet header compression, and VoIP packet payload compression [5][12][13]. This article proposes a new VoIP packet payload compression method to improve VoIP bandwidth exploitation over ITTP/IP protocols. Different from the existing methods, the proposed method focuses only on compress the voice frame without packet multiplexing. Moreover, the proposed method will not add any additional information header to the ITTP/IP packet header, because the proposed method reemploys and utilizes the existing fields of the IP protocol header.

The rest of this paper is organized as follows. Section 2 some of the researcher's effort to enhance bandwidth exploitation over RTP and ITTP protocols. Section 3 discusses the components, mechanism, and process of the proposed shrinking method. Section 4 implements the proposed method and verifies its performance. Finally, Section 5 concludes the paper.

TABLE I. RENOWNED VOIP CODECS

| **Codec Name** | **G.723.1** | **G.726** | **LPC** | **G.729** | **G.728** |
|---|---|---|---|---|---|
| Frame Size (B) | 20 | 30 | 14 | 10 | 10 |
| Bit Rate (kbps) | 5.3 | 24 | 5.6 | 8 | 16 |


Fig. 1. ITTP VoIP Packet Format.

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## II. Related Works

The researchers have exerted a considerable effort to enhance VoIP application bandwidth exploitation. This section spotlight some of these efforts over both RTP and ITTP protocols.

The first approach to enhance VoIP application bandwidth exploitation is packet multiplexing. Multiplexing approach works by grouping the VoIP packets that are sharing the same destination in one header, instead of a separate header to each packet, which highly improves bandwidth utilization. One of the best multiplexing methods over RTP protocol was proposed and patented by Vulkan, Csaba, et al. in 2014. The proposed method performs the packet multiplexing process at the sender side gateway. In which, several packets that are sharing the same destination will be grouped in one large packet with a single UDP/IP header. The packets are accumulated in the large packets until it reached a specific size or for a predefined period of time. At the receiver side gateway, the large packet is segregated, and the original packets are constructed and transmitted to their destinations [14]. Over ITTP, one of the first multiplexing methods is ITTP-Mux (ITTP-Multiplexing) which was proposed in 2015. ITTP-Mux consists of two entities, namely, stream multiplexer (S-Mux) at the sender side gateway and stream de-multiplexer (S-DMux) at the receiver side gateway. At first, the S-Mux separates the packet payload (ITTP and voice frame) from the IP header. Then, a 1-byte header is attached to the packet payload which constitutes a mini-packet. After that, several mini-packets are grouped in one IP header. Finally, the resulted multiplexed packet is transmitted to its destination. The S-DMux segregates the received multiplexed packet by inspecting the 1-byte header. Then, the S-DMux removes the 1-byte header of each mini-packet and attaches the IP header to construct the original packet. Finally, the original packets are transmitted to their destination. The performance evaluation of the ITTP-Mux showed that the bandwidth exploitation has enhanced by up to 29.1% in comparison to the traditional method [10].

The second approach to enhance VoIP application bandwidth exploitation is header compression. Sandlund and Pelletier have proposed then standardized a method that compacts the RTP/UDP/IP header from 40bytes to 2 or 4 bytes. The proposed method achieves this high compression based on two main features of the RTP/UDP/IP header. First, the majority of this header data is not changing throughout the call time. Therefore, these fields are sent when establishing the call and removed from the residual of the packets. Second, some of the remaining data in the header are incremented by steady value. Therefore, these values can be removed from the header and calculated based on a specific equation at the receiver side [15]. Sze et al. proposed another header compression method to enhance VoIP application bandwidth exploitation. However, the proposed method has been implemented with the packet multiplexing approach. It consists of the multiplexer (MUX) module at the sender side gateway and the demultiplexer (DEMUX) module at the receiver side gateway. The MUX module separate the RTP/UDP/IP header from the incoming VoIP packets, compact the RTP header, reattach the compacted RTP header to the voice frame to produce a mini-packet, combine multiple mini-packets in one large packet, and attaches the UDP/IP header to this large packet. The DEMUX module performs a reverse process of the MUX module to reconstruct the original VoIP packets and then, transmit them to their destinations. The size of the large packet is controlled based on the delay, where, if the delay reaches specific value then the multiplexing process is stopped. The performance evaluation of the proposed method showed that bandwidth exploitation has highly enhanced in comparison to the traditional method [16].

The third approach to enhance VoIP application bandwidth exploitation is packet payload compression. The payload compression can be achieved by the VoIP codec, which highly compresses the voice data during generating the voice frame. In addition, payload compression can be achieved by voice activity detection (VAD) technology, which based on the fact that one of the call ends is speaking and the other one is listening. Therefore, VAD suppresses the silence instead of sending it, which highly saves the bandwidth [3][17]. Another method to compress the packet payload, called Delta-Multiplexing, was proposed by Abualhaj et al. in 2010. However, as the name implies, the Delta-Multiplexing method has been implemented with the packet multiplexing approach. Similar to the method proposed by Sze et al. 2002, the Delta-Multiplexing method consists of two modules, Mux and DMux, that perform almost the same process of MUX and DEMUX modules, respectively. However, instead of compressing the packet header, the Mux compresses the packet payload. This is done by calculating the difference between the consecutive voice frame based on a specific mechanism, thus, reduces the voice frame size. Then, transmit this difference instead of the full voice frame, thus, enhances the bandwidth exploitation. In addition, the Delta-Multiplexing method has been added a 12-bits mini-header to the multiplexed mini-packets to be able to restore the original voice frame size and the original VoIP packet. The performance evaluation of the Delta-Multiplexing showed that bandwidth exploitation has enhanced by up to 72% in comparison to the traditional method [3]. A very similar method to Delta-Multiplexing was proposed in 2019 but over ITTP protocol. In which, several VoIP packets are multiplexed in one IP header, the same mechanism of packets payload compression is used, and a 1-byte mini header has been added to the mini-packets to be able to restore the original voice frame size and the original VoIP packet. The performance evaluation of the proposed method showed that bandwidth exploitation has highly enhanced in comparison to the traditional method of ITTP protocol [18]. This article proposes a new payload compression method to enhance bandwidth exploitation over ITTP protocol. Unlike the discussed payload compression methods, the proposed method focuses only on compact the voice frame without multiplexing. In addition, no extra header is needed to restore the original size of the voice frame, because the proposed method reemploys and utilizes the existing fields of the IP protocol header. The proposed method is called payload shrinking over ITTP (ITTP-PS). The following section discusses the proposed ITTP-PS method in detail.

## III. Payload Shrinking Over ITTP (ITTP-PS) Method

The key objective of the ITTP-PS method is to shrink the voice frame and therefore lessens its size. This leads to improves the VoIP applications bandwidth exploitation. The ITTP-PS method will be deployed at the VoIP gateway that connected to the wide area network (WAN) connection, though it can be deployed at the client end. Several advantages can be gained by deploying the ITTP-PS method at the VoIP gateway, including i) the proposed ITTP-PS method is workable with the other bandwidth exploitation approaches, such as VoIP packet multiplexing and VoIP header compression, which are typically deployed at the VoIP gateway, ii) the flexibly of using any client application from any device because it does not need to support ITTP-PS method, iii) the ITTP-PS method need to be deployed once at the VoIP gateway instead of each client, and iv) there is, typically, a plenty of bandwidth in local area network in comparison to WAN connections, where the bandwidth is limited and expensive [14][15][19][20]. Fig. 2 displays the location of the ITTP-PS method in a typical VoIP network topology. As shown in Fig. 2, the ITTP-PS method has two entities. The first entity called sender ITTP-PS (S-ITTP-PS). The main function of the S-ITTP-PS entity is to shrink the voice frame to generate a smaller one named S-F. The second entity called receiver ITTP-PS (R-ITTP-PS). The main function of the R-ITTP-PS entity is restoring the S-F to its normal size and generate an original-size voice frame named O-F. The following two subsections discuss the S-ITTP-PS and R-ITTP-PS entities, respectively.

### A. S-ITTP-PS Entity Function

The S-ITTP-PS entity performs four main functions to shrink the voice frame: i) detach the VoIP packet header (ITTP/IP) from the VoIP packet payload (voice frame), ii) apply a specific shrinking mechanism on the voice frame to produce S-F voice frame (as discussed below), iii) store the value (0 or 1) and place (start or end) of shrunk bits to be restored at the R-ITTP-PS entity (as discussed below), iv) and reattach the VoIP packet header (ITTP/IP) to the S-F voice frame. Finally, the VoIP packet is sent to the receiver's VoIP gateway. Fig. 3 demonstrates the S-ITTP-PS entity functions.

The purpose of the shrinking mechanism is to lessen the voice frame size. There are several operations performed by the shrinking mechanism to lessen the voice frame size and produce S-F voice frame. First, check whether the value of the leading bit one or zero. Second, regardless of whether the leading bit is one or zero, determine the total number of leading bits till the bit with the opposite value of the leading bit. Third, save the number of leading bits in the leading counter (L-count). Fourth, check whether the value of the trailing bit one or zero. Fifth, regardless of whether the trailing bit is one or zero, determine the total number of trailing bits till the bit with the opposite value of the trailing bit. Sixth, save the number of trailing bits in the trailing counter (T-count). Seventh, compare the L-count and T-count. If the L-count greater than or equal the T-count removes the leading bits, otherwise remove the trailing bits. These operations produce the S-F voice frame. The examples in Table II demonstrate the voice frame shrinking process at the S-ITTP-PS entity. Notably, the values of the voice frame in Table II are not real and are only for demonstration purposes.

As mentioned previously, the value (0 or 1) and place (start or end) of shrunk bits must be stored and sent to the R-ITTP-PS entity to restore the O-F voice frame. To achieve that, the ITTP-PS method will reemploy the flag bits in the IP protocol header. The three flag bits in the IP protocol header are used to control and identify fragments of the packet (if any). In the case of VoIP, the packets are not fragmented because they are typically very small, between 36-bytes to 56-bytes, when using ITTP protocol. Thus, the three flag bits are always set to zero in the case of VoIP packets. Accordingly, the S-ITTP-PS entity will use them to denote the value and place of the eliminated bits from the voice frame. Only four possibilities are needed to denote the value and place of the eliminated bits, thus, the shrinking mechanism will use only the first two bits (4 values) of the flag bits. The denotation of four values is as follows: 00 denotes shrinking the leading zeros, 01 denotes shrinking the leading ones, 10 denotes shrinking the trailing zeros, 11 denotes shrinking the trailing ones.

### B. R-ITTP-PS Entity Functions

The R-ITTP-PS entity performs four main functions to restore the S-F voice frame to its original form (O-F frame form). First, detach the VoIP packet header (ITTP/IP) from the VoIP packet payload (voice frame). Second, apply the shrinking mechanism on the voice frame to restore the S-F voice frame to the O-F voice frame (as discussed below). Third, the flag bits in the IP protocol header are set to zero to avoid misinterpretation at the receiver client. Fourth, reattach the VoIP packet header (ITTP/IP) to the O-F voice frame and send the VoIP packet to its destination. Fig. 4 demonstrates the R-ITTP-PS entity functions.

The purpose of the shrinking mechanism at the receiver end VoIP gateway is to restore the S-F voice frame to the O-F voice frame. There are two main operations performed by the shrinking mechanism to achieve this purpose. First, find the number (n) of shrunk bits from the O-F voice frame by S-ITTP-PS entity at the sender end VoIP gateway (n= O-F length – S-F length). Second, restore the S-F voice frame to O-F voice frame based on the value of the flag fields of the IP protocol header. The denotation of the flag fields is as follows: 00 denotes append n zeros at the beginning of S-F voice frame, 01 append n ones at the beginning of S-F voice frame, 10 denotes append n zeros at the end of S-F voice frame, 11 denotes append n ones at the end of S-F voice frame. The examples in Table III demonstrate the restoration of the S-F frame to the O-F frame by the R-ITTP-PS entity. These examples are the same examples in Table II.

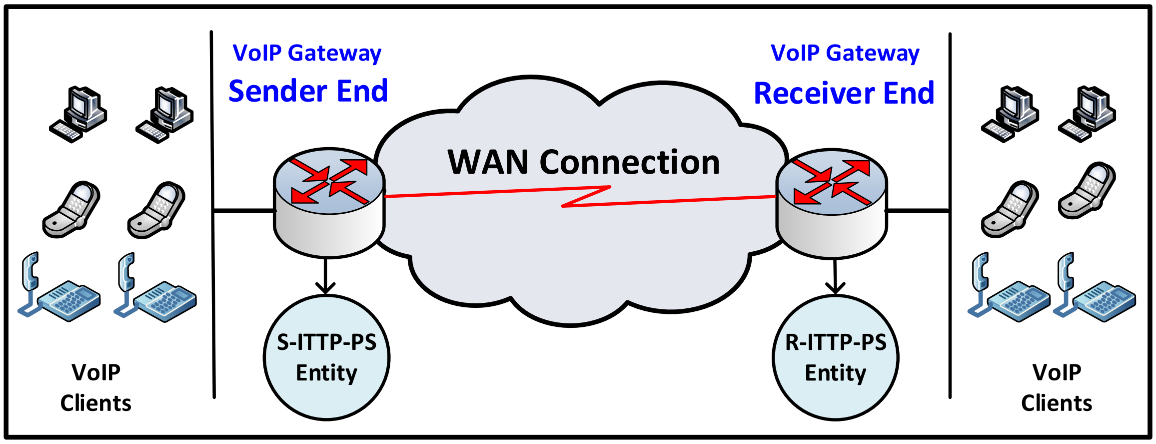


Fig. 2. Location of ITTP-PS Method's Entities.

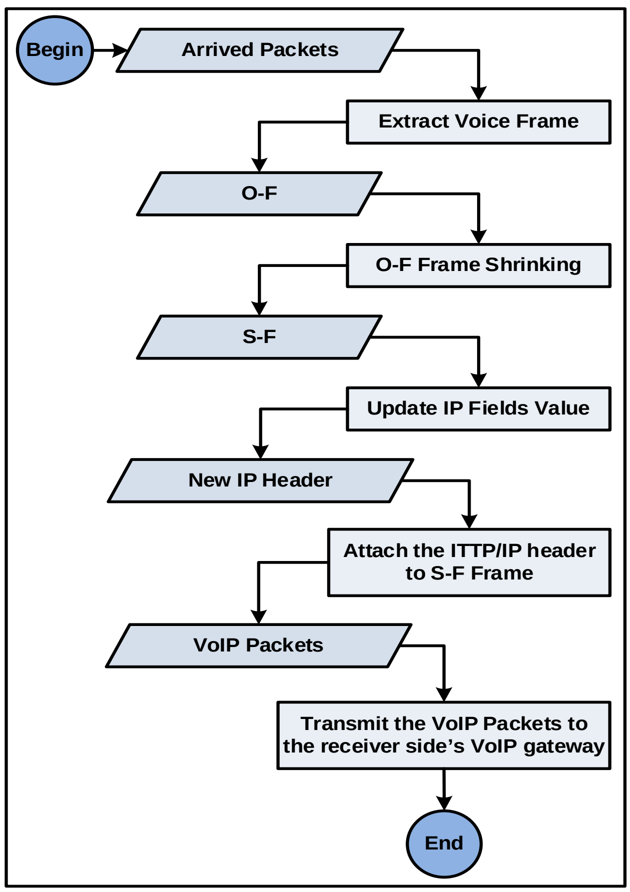


Fig. 3. S-ITTP-PS Entity Functions.

TABLE II. EXAMPLES OF SHRINKING THE VOICE FRAME

| O-F Frame | L-Count | T-Count | Action | S-F Frame |
|---|---|---|---|---|
| 0000011000101001100000 | 5 | 4 | Shrink the leading zeros because of L-Count is greater than T-Count. | 1100010100110000 |
| 0001100111100111000000 | 3 | 5 | Shrink the trailing zeros because of T-count is greater than L-Count. | 0001100111100111 |
| 000000100111001101111 | 6 | 4 | Shrink the leading zeros because of L-Count is greater than T-Count. | 100111001101111 |
| 1111110100101111110000 | 6 | 4 | Shrink the leading ones because of L-Count is greater than T-Count. | 010010111110000 |
| 11111011011011111111 | 5 | 9 | Shrink the trailing ones because of T-Count is greater than L-Count. | 111110110110 |

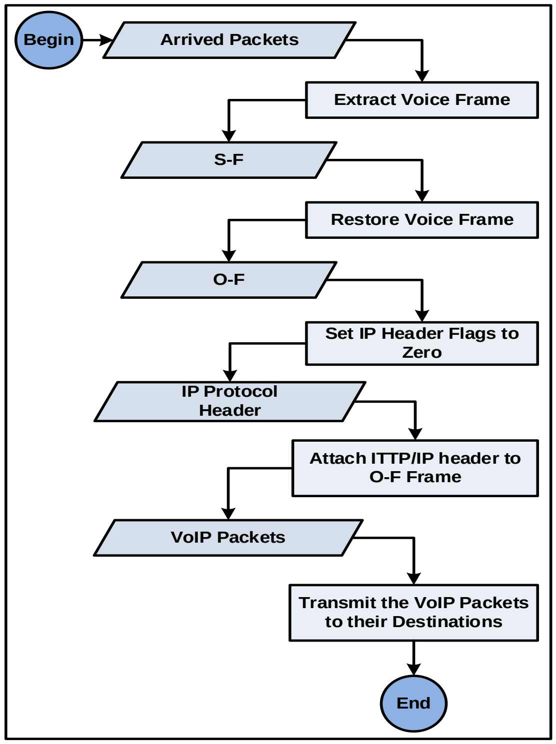


Fig. 4. R-ITTP-PS Entity Functions.

TABLE III. EXAMPLES OF RESTORING THE S-F RAME TO O-F FRAME

| S-F Frame | Flag Field Value | Action | O-F Frame |
|---|---|---|---|
| 1100010100110000 | 00 | Append n zeros at beginning of the S-F voice frame. | 000001100010100110000 |
| 0001100111100111 | 10 | Append n zeros at end of the S-F voice frame. | 000110011110011100000 |
| 100111001101111 | 00 | Append n zeros at beginning of the S-F voice frame. | 000000100111001101111 |
| 010010111110000 | 10 | Append n ones at beginning of the S-F voice frame. | 111111010010111110000 |
| 111110110110 | 11 | Append n ones at end of the S-F voice frame. | 111110110110111111111 |

## IV. THE PROPESED ITTP-PS METHOD PERFORMANCE EVALUATION

This section discusses the ITTP-PS method's bandwidth exploitation in comparison with traditional ITTP protocol without shrinking the voice frame. The bandwidth exploitation was evaluated based on two different elements, namely, voice frame shrinking ratio (FRR) and isochronous calls capacity. For the evaluation to be sensible, three different VoIP codecs (LPC, G.723.1, and G.729) will be used to investigate these two elements.

### A. Frame Shrinkig Ratio (FRR)

This subsection investigates the ITTP-PS method's FRR ratio. The FRR is the number of shrunk bits from the voice frame to the total number of bits of the voice frame without shrinking. The higher the number of shrunk bits the better the bandwidth exploitation. The FRR ratio has been investigated with ten different calls. The average FRR ratio of 20 VoIP packets of each call has been calculated separately. Fig. 5 displays the FRR of the LPC codec. The FRR ratio of the LPC codec has reached around between 12.5% and 18.5%. Fig. 6 displays the FRR of the G.723.1 codec. The FRR ratio of the G.723.1codec has reached around between 10% and 20%. Fig. 7 displays the FRR of the G.729 codec. The FRR ratio of the G.729 codec has reached around between 7% and 15.5%. Accordingly, the ITTP-PS method accomplished a noticeable bandwidth exploitation in comparison with the traditional voice frame without shrinking. Thus, the ITTP-PS method achieved it main objective of improving VoIP network bandwidth use. The three codecs accomplish different FRR ratio because the pattern of the voice frame of each codec is different.

### B. Isochronous Calls Capacity

This subsection investigates the proposed ITTP-PS method isochronous calls capacity improvement ratio. The bandwidth exploitation improvement ratio was investigated then the isochronous calls capacity improvement ratio will have similar values. The bandwidth exploitation improvement ratio has been investigated with 10 different calls. The average bandwidth exploitation improvement ratio of 20 VoIP packets of each call has been calculated separately. Fig. 8 displays the bandwidth exploitation improvement ratio of the LPC codec. The bandwidth exploitation improvement ratio of the LPC codec has reached around between 4.5% and 6.5%. Fig. 9 displays the bandwidth exploitation improvement ratio of the

G.723.1 codec. The bandwidth exploitation improvement ratio of the G.723.1codec has reached around between 5% and 9.5%. Fig. 10 displays the bandwidth exploitation improvement ratio of the G.729 codec. The bandwidth exploitation improvement ratio of the G.729 codec has reached around between 2% and 4.5%. Therefore, the isochronous calls capacity improvement ratio will have similar values. Accordingly, the ITTP-PS method accomplished a pretty good isochronous calls capacity improvement ratio in comparison with the traditional voice frame without shrinking. Thus, the ITTP-PS method achieved it main objective of improving VoIP network bandwidth use. Obviously, this feature is due to the shrinking the voice frame. The three codecs accomplish different isochronous calls capacity improvement ratio because the pattern of the voice frame of each codec is different.

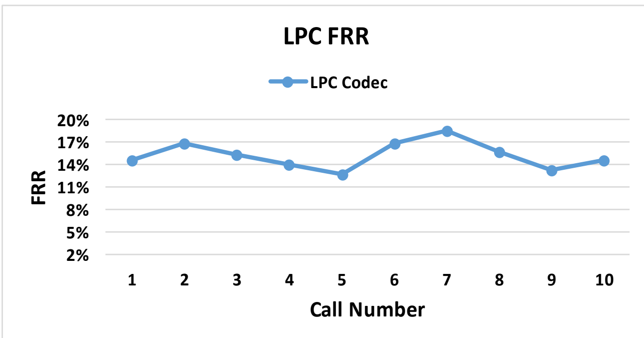


Fig. 5. FRR of the LPC Codec.

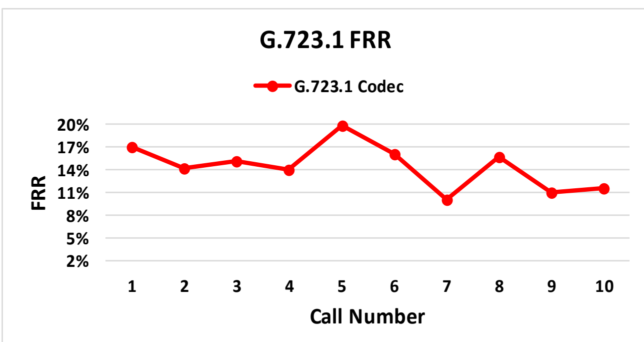


Fig. 6. FRR of the G.723.1 Codec.

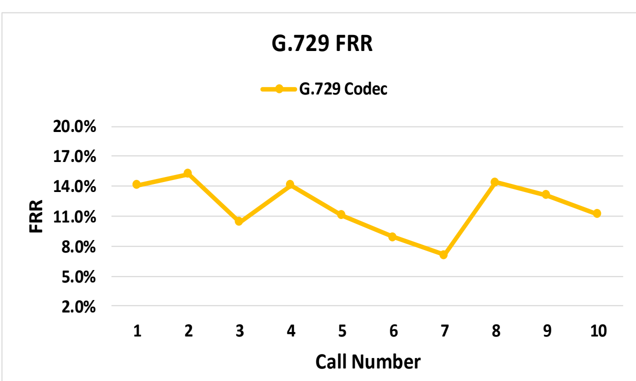


Fig. 7. FRR of the G.729 Codec.

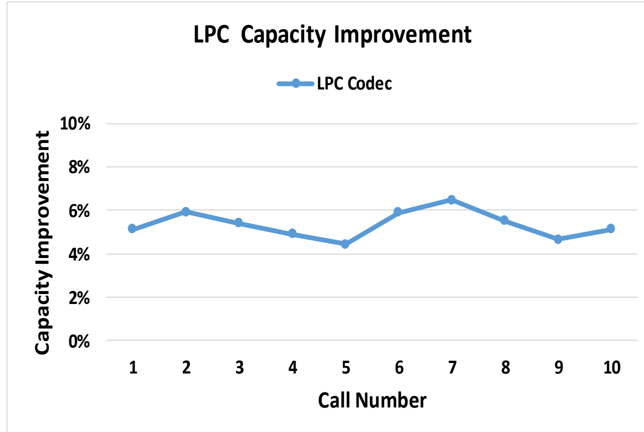


Fig. 8. Isochronous Calls Capacity Improvement Ratio of the LPC Codec.

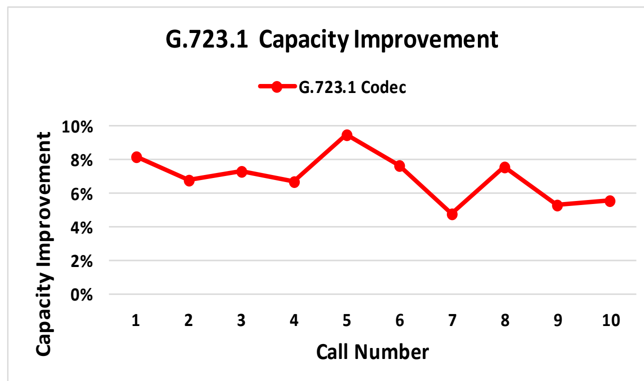


Fig. 9. Isochronous Calls Capacity Improvement Ratio of the G.7321.1 Codec.

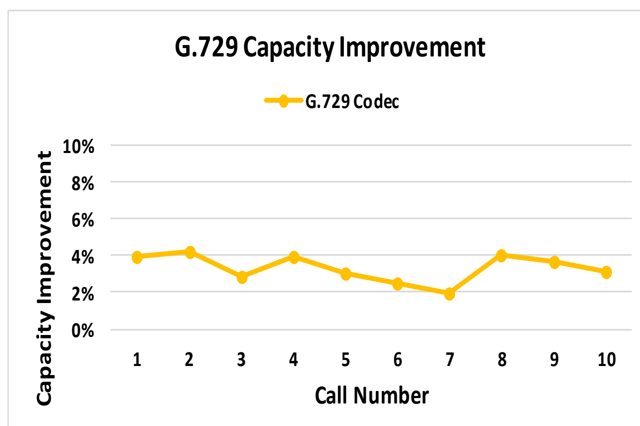


Fig. 10. FIsochronous Calls Capacity Improvement Ratio of the G.729 Codec.

## V. Conclusion

VoIP has been widely adopted by private and public sectors replacing the traditional telecommunication systems. Nevertheless, VoIP faces a serious bandwidth exploitation inefficiency problem. This article has proposed a new method, called ITTP-PS, to alleviate this problem. The main idea of the proposed ITTP-PS method is to shrink the VoIP packet payload, based on certain mechanism. The ITTP-PS method has been deployed at the VoIP gateway that is connected to the WAN connection. The sender side VoIP gateway performs VoIP packet payload shrinking using the S-ITTP-PS entity of the ITTP-PS method. The receiver side VoIP gateway restores the VoIP packet payload to its normal size using the R-ITTP-PS entity of the ITTP-PS method. The R-ITTP-PS entity achieves that by utilizing the flag bits in the IP protocol header. The ITTP-PS method has been implemented and compared with traditional ITTP protocol without shrinking the VoIP packet payload. The VoIP packet payload shrinking ratio has been enhanced by up to around 20% in the tested cases with the G.723.1 codec. In addition, the isochronous calls capacity improvement ratio has been enhanced by up to around 9.5% in the tested cases with the same codec. Eventually, the ITTP-PS has been alleviated the bandwidth exploitation inefficiency problem of VoIP applications. In the future, the proposed ITTP-PS method will be evaluated with different VoIP codecs to gain a more realistic evaluation. In addition, the ITTP-PS method will be combined with other approaches of bandwidth exploitation such as VoIP packet multiplexing.

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