VHDL based Design of an Efficient Hearing Aid Filter using an Intelligent Variable-Bandwidth-Filter

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Abstract—**Filtering techniques have been elaborated in the HA field to improve signal clarity and enhance the hearing capacity of deaf people. However, public sounds are highly noisy, so filtering those signals is not an easy task. Hence, the present article has aimed to develop a novel Ant Lion based power Noise-Variable Bandwidth Filter (ALPN-VBF) for the HA applications. Here, the proposed optimized power efficient filter has incorporated several functions like de-noising and frequency tuning based on the word features. Here, the signal's noise has been removed with the maximum possible range with the help of High-pass-Filter (HPF) and low-pass filter (LPF). Finally, the developed model is tested with a few audiograms, and the filter parameters have been analyzed and compared with other models. The testing results have proved that the designed filter is better in frequency tuning and signal transmission than the previous approaches by attaining less delay and reduced power consumption rate.**

Keywords—Hearing aid system; variable bandwidth filter; audiograms; matching error; power consumption; signal filtering

I. Introduction

The response variable frequency properties are required in various telecommunications applications and software wireless applications [1]. In addition, the design of bandwidth filter with variable parameters has included adders, coefficient array, multipliers, and data array length [2]. The idea employed here reduces the relative filter bandwidth by functioning it in a lower sample rate [3]. Moreover, this process is balanced by an efficient resembling technique. Here, the down-sampled wave is treated by a fixed bandwidth and length, and the up- sampled return to the initial spectrum [4]. Hence, the combined strategy has decreased the filter's bandwidth [5]. Moreover, this approach is chiefly utilized in bandwidth reduction functions [6]. However, numerous applications may demand a higher bandwidth than the fixed-length filter's performance [7]. However, this could result in distortion of signal response outcome [8]. Besides, the non-zero bandwidth transition is the primary cause of distortion in fixed-length-filter [9]. The filter's length often depends on the bandwidth transition. In addition, to gain the undisturbed signal, reducing the filters distortion has to be minimized up to the desired level [10].

The chief advantage in using VBF is better tuning capacity based on the signaling and noise application. The resonant of frequencies have been tuned accordingly [11]. The basic function of VBF is detailed in Fig. 1. In addition, the complexity in VBF is noise removal [12]. If the data is realistic, it has taken more resources and time to de-noise the signal; also, the conventional filter has afforded the average outcomes [13]. Several filtering approaches, such as a novel key element based on cloud serve VBF [14], parallel structure filter [15], etc., have been implemented in the past. But, the appropriate filtered output was not gained, so the current research work has aimed to design a novel power-efficient, optimized filter model for HAA application. At last, the performance of the designed filter is compared with other models and has obtained the finest results. The key contributions followed in this research is described as follows:

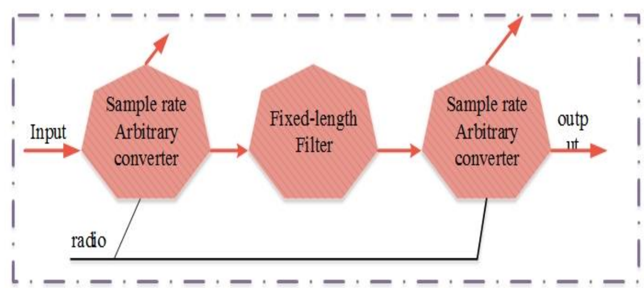

Fig. 1. Function of VBF

- Initially, different public audios were collected from the benchmark site and trained to the system.
- Consequently, a novel ALPN-VBF has been developed with suitable parameters to denoise and tune the signal frequency.
- Here, the possible level of noise signal has been reduced with the use of the HPF and LPF.
- Moreover, the desired signal range has been tuned by matching the maximum signal range in the ant lion fitness.
- Subsequently, the incorporation of the ant lion fitness in frequency tuning has afforded the finest outcome.
- Finally, the designed parameters were measured and compared with conventional paradigms in the relations of power consumption, delay, and matching error.

The current article's research arguments are organized as follows. Section II has detailed the associated works of the different filters in the HA system; the designed novel VBF filter and function is highlighted in Section III. Moreover, the working performance of the designed Filtering strategy is summarized in Section IV, and the research discussion and achievements are concluded in Section V.

II. Related Works

To enhance the tuning range of VBF, Ren et al. [14] have implemented a novel key element based on cloud services for the VBF system. The designed VBF has recorded the tuning rate between o.3nm to 1.35nm by the experimental process. Also, the measured loss of insertion is 1.39dB. Hence, the tuning function of the VBF has been maximized. Also, it is flexible to change the response rate based on applications. However, it has required more power than the conventional VBF.

The parallel structure filter has been introduced by Swamy and Alex [15] for the HA system for better signal; frequency. Here, the filter is mod- eled with the reconfigurable parameters that contain several sub-bands to process the different frequency range signals. Moreover, this filter is tested with numerous kinds of Hearing Loss (HL) people data. Hence, the tested people data contains mild to severe HL data; in many cases, the signal has reached better. However, it has consumed more power.

The corrector filters with half bands have been described for the HA system by Niveditha et al. [16] to enhance signal transmission speed. The working of the designed filter is verified with the hardware HA application. The proposed filter has earned a reduced power consumption rate and high signal transmission speed. However, it has lacked in signal clarity.

Ma et al. [17] modeled the reconfigurable filter with multi bands and modularized design structure. In addition, the multiple distortion schemes of subbands have met the frequency range of several different audiograms. The designed filter has a reduced complexity rate than the conventional noise removal filter in HA systems. However, it has required more energy to execute the process.

The notch filter has been applied in the HA system by notch filter Marcrum et al. [18] to maximize the hearing capacity of the aged hearing loss people. The key motive of this research is to measure the filtering rate by comparing the conventional filter. Hence, the filtering capacity was identified by using a different set of audio signals. However, it has required more resources to execute.

Several methods were discussed with its limitation and merits for the HA system. Hence, the common demerit in many cases is signal noise. If this problem ends, then the HA system's signal processing has been maximized. So, the current work has focused on implementing the powerful noise filter for HA applications.

III. Proposed Methodology

The key motive of this HA system application is to improve the communication and hearing capacity of deaf people. Several kinds of HA system has been designed with different efficient filter parameters. But, still, the noise in the processing signal might reduce the signal frequency. So, the present work has focused design a novel optimized VBF named ALPN-VBF for HA framework. Here, to enable the power noise mechanism, dual filters functions have been utilized that are HPF and LPF. Then to transform the optimal signal range, the maximum signal frequency range is modeled in the Ant lion fitness function. Hence, the optimization objective has been iterated repeatedly until the desired signal has been met.

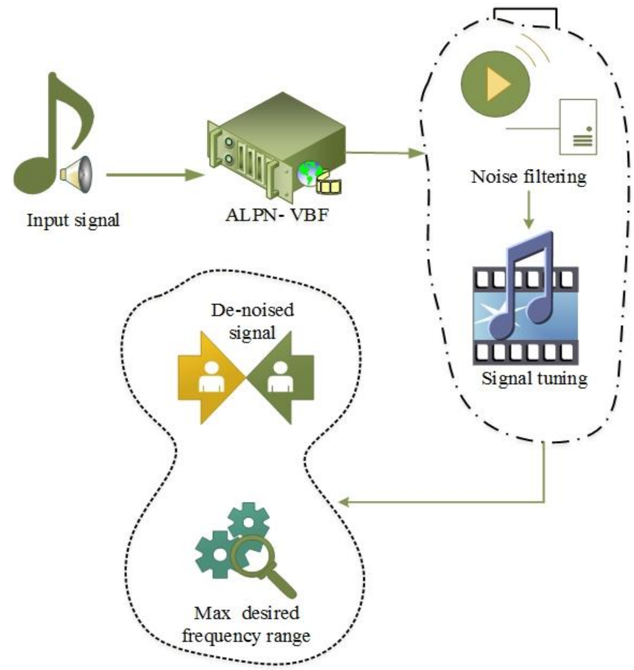

Fig. 2. Proposed architecture

The proposed architecture is detailed in Fig. 2. Finally, the working of the designed efficient VBF has been analyzed by validating the function parameters by applying any voice are sound. Subsequently, the improvement score has been measured by processing the comparison assessment.

A. Design of ALPN- VBF for HA System

The present work has developed a novel optimized intelligent ALPN-VBF for the HA system to improve the signal clarity of the HA device. Hence, this technique incorporates LPF, HPF, and Ant lion procedures. Usually, the ant lion is inspired by their hunting intelligence by trapping the ant inside the pit. Here, this fitness is utilized to fix the maximum desired frequency and tune the frequency up to the fixed frequency.

$$c = \sum_{n=1}^{x} p_n 2^{S-n} \tag{1}$$

Here, c denotes filter coefficients, world length is determined as x, the required power is represented as s and signal features are described as p. Moreover, the p_n is signal coding that is defined as $p_n = (-1,0,1)$. Hence, the filter coefficient is described in (1).

$$F(y) = \left\| G_d(\omega) - G(\omega, y) \right\|_2 \tag{2}$$

Here, the response of the zero-phase frequency is denoted as Gd and $G(\omega, y)$ is the present noise feature in the input signal In addition, $F(y)$ is the objective function signal tuning and filtering. Hence, (2) is utilized to neglect the present noise in the input signal. Moreover, the signal range has been tuned by fixing the maximum signal range and tuning the input signal during the execution of the signal processing (3).

$$A(y) = \max(0, u(y) - e_b) \tag{3}$$

Here, e_b is the detection parameter to find the equivalent fixed maximum signal range $u(y)$ and $A(y)$ is the threshold setting parameter. Now, the minimum frequency of the sound is fixed using (4),

$$\min F(y) = \beta_1 F_1(y) + \beta_2 A(y) \tag{4}$$

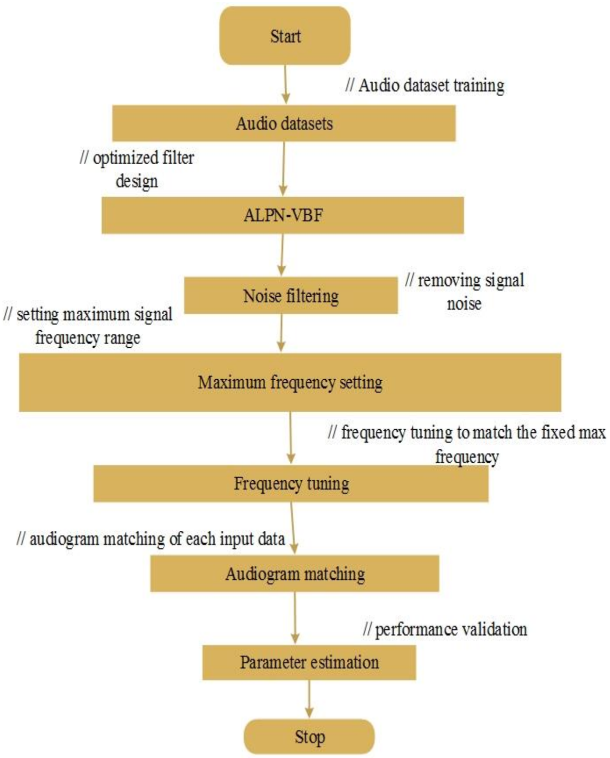

Fig. 3. Flow of novel ALPN-VBF

The design of ALPN-VBF Where, β_1, β_2 are the signal's weight coefficients. In addition, the system is trained with more different signals; each signal has specific frequency metrics. Hence, the signal selected from the total set of the collected sound signal is processed by (5).

$$\begin{cases} 1 & ifrand > \max(u(y)) \\ 0 & frand \le \max(u(y)) \end{cases} \tag{5}$$

Besides, the tuning process of the signal coefficients is detailed by (6), if the statement has been worked with 0 and 1 status. Hence, for the random selection process, 0's and 1's conditions were used.

$$T(s) = \frac{\max(u(y) + \min(u(y)}{2} \tag{6}$$

$$optimum \quad frequency = F(y), if\ T(s) < \max(u(y)) \tag{7}$$

During the signal transmission process, the optimal range signal (7) has been obtained then the optimization iteration has been stopped. Otherwise, the iteration has been continued till the desired signal is met.

The function process of the novel ALPN-VBF is executed in the way of Fig. 3. The key function of this designed filter is to filter the noise signal with a highly appropriate range and to tune frequency based on the deaf people hearing capacity.

IV. Results and Discussion

The planned research solution is implemented in the MATLAB framework, running in the windows 10 platform. Initially, benchmark data is collected from the standard sites, it contains several sounds like animals, birds, humans, etc., then a novel ALPN-VBF has been developed with desired tuning parameters. Consequently, the signal executing and frequency tuning performance have functioned.

A. Case Study and Performance Validation

To analyse the performance of the designed filter, benchmark data has been utilized. From that bird sound and public transport, sound has been processed to analyze the capacity of the noise filtration function and tuning performance.

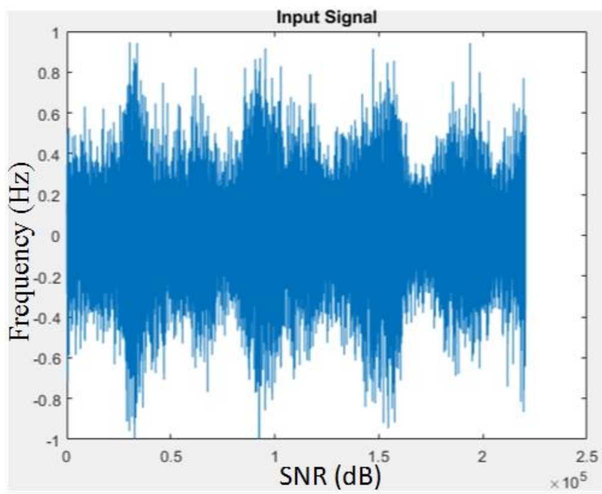

Fig. 4. Input signal frequency

The generated signal plot after importing the sound dataset is described in Fig. 4. The frequency wave of the input signal is varied based on the audio maximum frequency range.

1) Matching error: In any filtering concept, finding the signal error or mismatch became the major concern for validating the filter functions. The matching error parameter has been estimated for three audiograms.

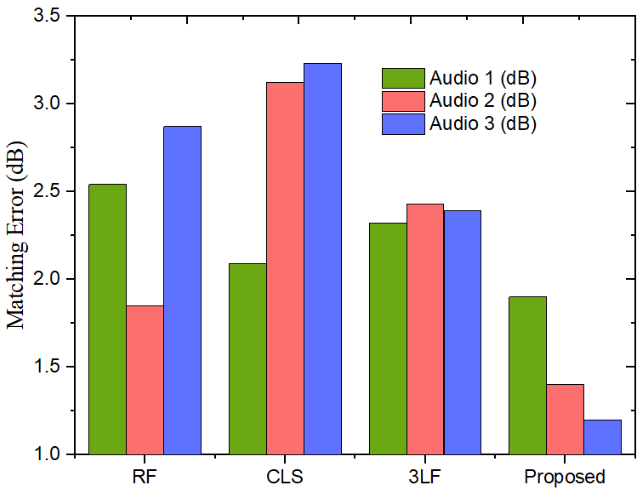

Fig. 5. Matching error assessment

In addition, to find the improvement score of the proposed model, a few existing models have been obtained, such as reconfigurable filter [19], Constrained-least-square (CLS) filter [20], and 3 level Filter (3LF) [21]. Here, the RF has obtained an error score of 2.54 dB for the first audiogram, 1.85dB for the second audiogram, and 2.87 dB for the third audiogram. Hence, evaluating these statistics, the method RF [22] has recorded the average matching error score as 2.4db. Furthermore, the CLS filter has gained the error score for the first audiogram is 2.09dB, 3.12 dB for a second audiogram, and 3.23 dB for the third audiogram. Hence, the recorded average error score by the method CLS [23] is 3dB. In addition, the method 3LF [24] has gained 2.32 dB for the first audiogram, 2.43 dB for the second audiogram, and 2.39 dB for the third audiogram. The average recorded score for matching error is 2.35dB. Considering all those existing models, the proposed approach has obtained 1.9dB for the first audiogram, 1.4dB for the second audiogram, and 1.2dB for the third audiogram. Hence, the obtained average matching error for the proposed model is 1.5dB. The matching error statistics is graphically represented in Fig. 5.

2) Delay validation: Measuring the time for the audio frequency transmission is called a delay. Mainly, the filters which have the complex function modules have required more time to execute; this causes time delay.

Moreover, the RF approach has achieved an 18.54 ms delay, the CLS filter has recorded a 7.9ms delay, 3LF has achieved the frequency delay as 12.9ms, and the proposed strategy has recorded the delay score to broadcast the signal frequency is 5ms. The delay statistics are described in Fig. 7.

3) Power Consumption: In digital signal application, power is the key metric for transferring the signal, the utilized power by the novel ALPN-VBF has been measured with Watt unit.

In the proposed research, dual filters were used: LPF and HPF, so power is a major concern for analyzing the maximum required power for HA system fitters. Besides, the reason for consuming high power is the poor filtering capacity. So, the method with less power consumption has a high noise removal rate. The CLS filter has gained the power consumption score of 0.05W; RF has obtained the power consumption score of 0.398W, and the proposed approach recorded the consumed power as 0.02W. The measurement of power utilization is described in Fig. 6 and the overall power consumption validation measurement of the proposed ALPN-VBF is tabulated in Table I.

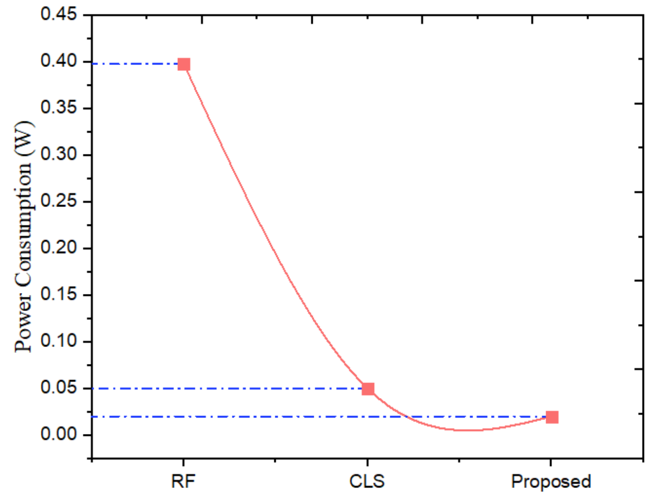

Fig. 6. Power consumption validation

TABLE I. THE OVERALL COMPARISON STATISTICS

comparison statistics					
Methods	***Matching error dB***			***Power consumption (W)***	**Delay (ms)**
	Audio1	***Audio 2***	***Audio 3***		
RF	2.54	1.85	2.87	0.398	18.54
CLS	2.09	3.12	3.23	0.05	7.9
3LF	2.32	2.43	2.39	0.03	12.9
Proposed	1.9	1.4	1.2	0.02	5

B. Discussion

The designed ALPN-VBF has recorded an outstanding performance from all the comparison assessments than the compared approach. This proved that the involvement of the combined HPF and LPF in one filter bank had provided the finest outcome.

TABLE II. OVERALL PARAMETER VALIDATION OF ALPN-VBF

Overall performance assessment		
Power consumption		0.02 W
Delay		5ms
Matching error	Audiogram 1	1.9
	Audiogram 2	1.4
	Audiogram 3	1.2

The overall metrics measurement of the proposed ALPN-VBF is tabulated in Table II. Hence, the designed filter procedure is suitable in HA application to maximize the hearing capacity of deaf people by affording the clarity sound frequency.

V. Conclusion

Improving the performance of the HA application is the most required paradigm for deaf people. Hence, this present work has attempted to design a novel optimized power filter name as ALPN-VBF for the HA system. Initially, the set of audiograms was taken and trained to the system then the filtering process was performed to de-noise the signal. Consequently, the frequency range of the signal is tuned to the maximum desired level with the help of ant lion fitness. Once the desired level frequency has been met, it has transferred to the other end. Finally, the key metrics were calculated and compared with other models. The proposed model has achieved the minimum matching error of 1.5 dB; considering the other models, it has diminished the matching error up to 1%. Besides, the recorded power consumption is 0.02W; compared to other models, it has shown an improvement rate of up to 2%. The delay measure recorded by the designed strategy is 5ms, by comparing other models, has described the improvement rate up to 12%. Hence, the designed model is suitable for the HA system.

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