Enhancement of Hearing Aid System using Optimized Wolf Based Variable Bandwidth Filter

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Enhancement of Hearing Aid System using Optimized Wolf Based Variable Bandwidth Filter

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Abstract In recent times, hearing loss is a challenging task, to minimize the risks the application-based Hearing Aids (HA) system was introduced. In this research, the HA system was designed with the use of several filters to reduce the hearing loss challenges such as signal noise, etc. Besides, the filter bank scheme involves a limited quantity of sub-bands, which have similar or dissimilar bandwidths. Previously several methods are developed to enhances the HA system. Nevertheless, computational complexity and power utilization are increased when improving the filter quantity. This paper introduces a novel Wolf-based Fractional Delay Variable Bandwidth Filter (W-FDVBF) to improve the hearing obstacles by optimizing the filter parameters. Here, the wolf optimization parameters are tuned with filter bank parameters to reduce the power consumption and complexity. Moreover, the proposed W-FDVBF is implemented in MATLAB and Xilinx software, also the performance of the proposed filter was verified using Field Programmable Gate Array (FPGA) accelerator. Finally, the obtained results are compared with other existing approaches and have gained better outcomes by attaining less power and matching error. Therefore, the attained parameters are improved the HS system performance in terms of matching error, power, flip flop utilization, etc

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1 Introduction

Nowadays, hearing loss is one of the most common disorders affecting the whole world, which may happen after birth because of hereditary sickness, noise, factors, and so forth [1]. One approach to adjust the loss of affectability of the harmed ear is to fit HA. Moreover, it is harder to provide HA, when the hearing loss is more regrettable. The ear acts distinctively for delicate and loud sounds close to the hearing edges [2]. So it is vital to examine the problems of the debilitated ear to build an efficient successful hearing aid system. [3]. HA system has presented dynamic series compression and frequency range gains to balance the various hearing destructions. Hearing loss and sensor neural differences are indicated in anatomical, For instance, Presbyacusis is an age-related hearing loss [4]. It typically influences the high frequencies more than the low frequencies [5]. HA system has three fundamental parts such as an amplifier, speaker, and microphone [6]. Initially, the HA system obtains the voice signal through the microphone after that the voice signal is converted into an electrical signal with help of the microphone, then the converted signal is sent to the amplifier [7]. HA systems generally need a hearing aid that amplifies the high frequencies explicitly. The patient with hearing loss has needed specific hearing aids to improve the hearing nature. A significant unit of a computerized hearing aid comprises the advanced channels that can tune the amplitudes selectively to particular hearing loss problems [8].

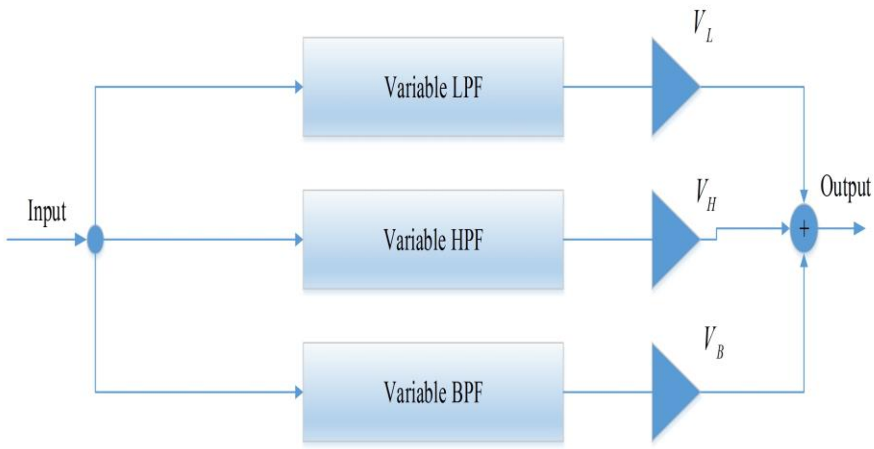

Fig. 1 Variable filter bank representation

Modern HA systems require some patterns to match the hearing loss audiogram with speech processing [9]. The variable filter bank is the arrangement of Band Pass Filter (BPF), High Pass Filter (HPF), and Low Pass Filter (LPF) [10]. Graphic equalizer and signal compression is the most important application in filter banks [11]. A decent measure of adaptability, least equipment, low force utilization, low deferral, and direct stage are the required character spasms of any advanced hearing aid [12]. The critical measure of the study is accessible on the variable filter bank channels intended for audiogram coordinating. Beginning methodologies were based on uniform sub-bands. Moreover, non-uniform variable filter banks are more qualified, with the goal that the coordinating can be accomplished with a lesser number of sub-groups, if conceivable. Moreover, the filter bank is the major influence on the whole HA system, which includes feedback cancellation, auditory compensation, and noise reduction

[13]. Moreover, the design parameter includes computation difficulty, identical prescription capacity, latency, frequency resolution, and speech quality [14]. These parameters are inclined to employ positive or negative sound effects with each other [15]. Representation of the Variable filter bank of the HA system is shown in fig. 1.

Moreover, Digital Signal Processing (DSP) has been directed to improve the hearing aid to avoid the severe effects of hearing [16]. DSP concept does not provide the physical size and power utilization restrictions of the Filter bank compared with other techniques [17]. Several approaches were utilized in the HA systems that are 32-bit Filter Bank with Sound Categorization systems [18], Speech improvement algorithms [19], sparse adaptive approach [20]. Moreover, the techniques are utilized in the HA system, but that having some limitations. Therefore, a novel optimized paradigm is developed in this research to overcome the existing issues by tuning filter parameters.

The rest of the article is ordered as follows. Sector 2 demonstrates recent associated works based on the hearing aid works, section 3 explains the system description and problem statement, the proposed filter design with optimization is described in section 4. The results and discussion about the proposed work are given in section 5, and the conclusion is detailed in section 6.

2 Related Works

Few of recently associated hearing aid works are described as follows,

Speech quality and monitoring performance is the highly desirable objective measure in the Hearing Aid (HA) system.

Salehi *et al* [21] introduced the new HA approach combined with two references free HA quality indices: hearing loss (HL) modelling and machine learning technique to forecast the quality of HA results. In HL modelling, coefficients of perceptual linear computation signals are used to predict the first feature set. Consequently, the machine learning approach and their filter bank signals are mapped in support vector regression. Therefore, the attained results are compared with existing HA quality indices like frequency shaped reference signals. Subsequently, this model achieved worse outcomes in computational complexity and accuracy.

Reshma *et al.* [22] have designed a Reconfigurable Filter (RF) bank HA system to balance the hearing losses of affected peoples. Moreover, optimization algorithms and control signals are proposed for audiogram harmonizing. Here, a reconfigurable filter bank is used 4-bit control signal to modify the sub-band techniques. Furthermore, an optimization algorithm is used to decrease the matching errors present in the newly designed FIR filter bank. Reconfigurable filter banks provide high throughput than the fixed filter bank.

Vellaisamy *et al.* [23] developed a non-uniform filter bank with a simple digital HA system; also, using a discrete Fourier transform filter bank is used as a non-uniform filter bank. The initial design of the uniform filter bank merged the new non-uniform filter bank and attained the desired structure. To achieve the sharp transition, narrow channels are used to distribute the frequency regions and adjustable gains. However, the developed HA model does not attain a practical outcome in matching error and delay in the prototype filter.

In the HA system, noise reduction and dynamic range compression strategies are commonly used algorithms. However, these algorithms cannot mitigate the signal-to-noise ratio (SNR) also; they affect the localization performance in HA systems. Hence, Llave *et al.* [24] introduced a novel HA model-based combination of noise reduction and dynamic range compression algorithms. Here, the developed algorithm enhances the SNR and maintains the acoustic characteristics of the noise element. But, the proposed design has provided lower performance in high mobility.

Naga Jyothi *et al.* [25] has introduced a combination of lower delay reconfigurable (LDR) and low complexity FIR filter is formation for HA systems. Moreover, three-level filters are designed in the proposed FIR filter and implemented in the MATLAB platform. The primary purpose is to equal a specified person audiogram, among the least fault. This proposed approach is considered based on the delay, area, and power consumption in a particular situation. Moreover, this model utilized the objectives to enhance performance in terms of attaining better outcomes in power dissipation and hardware complexity.

The Key following steps of the planned approach is summarized as follows,

- Primarily, the original speech signal is trained to the system
- Hereafter, a novel WFDVBF is designed with the support of Xilinx and MATLAB tools.
- consequently, the changing bandwidth parameter was utilized to estimate the range of frequency signal
- Moreover, the fitness of the grey wolf was updated in the designed filter bank to optimize the key parameters.
- Subsequently, the designed model is tested with an FPGA kit and the result are obtained
- Finally, the parameters of the designed model were estimated and compared with other methods in terms of delay, Matching error, power, area, and speech perception.

3 System Model and Problem Statement

The key motive of this research is to minimize the difficulties of deaf people by improving hearing aid functions.

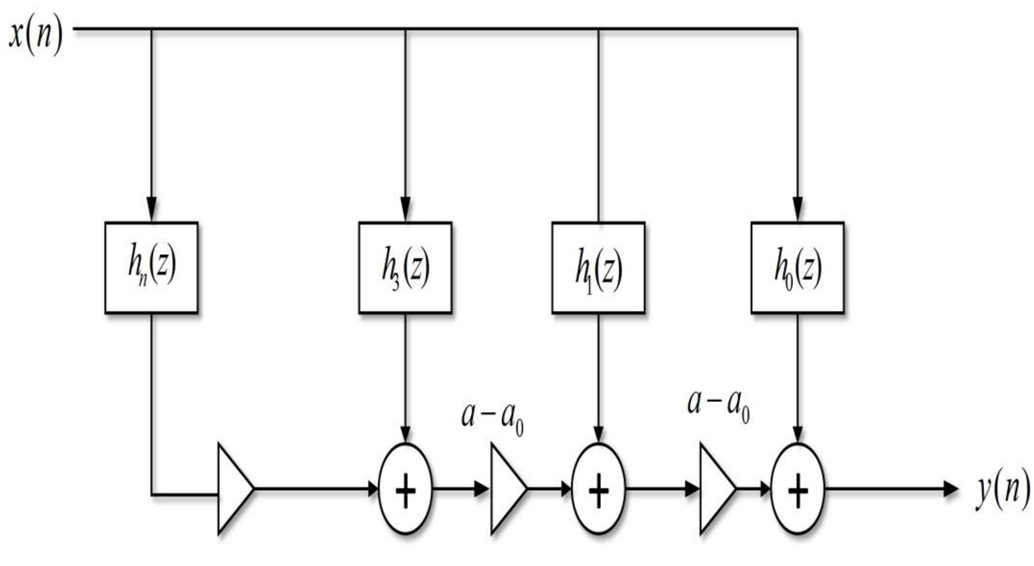

Fig. 2 VBF representation

Nowadays, all fields are digitalized with advanced technologies. But still, it is complex to use because of scalable and complicated design. Moreover, most digital-based technologies have attained less accuracy and consumed more power to run the process. As a result, the variable bandwidth-based approach was created to adjust the bandwidth dependent on the capacity of the used device. However, the changeable bandwidth filter bank might require more area and computational resources. Moreover, the block diagram of Variable Bandwidth Filter (VBF) is demonstrated in fig. 2,

In the above VBF representation frequency response and function of the system are denoted as $H(z,a)\, and\, H(e^{i\omega t},a)$ respectively. Moreover, the transfer function of the VBF sub-filter is expressed in eqn. (1),

$$H(z,a)=\sum_{j=0}^{n}(a-a_0)^k H_j(z) \tag{1}$$

Where, $H_j(z)$ represented as j^{th} sub-filter in the order of K_j includes impulse response coefficient namely $H_j(k)$, 'a' is denoted as the bandwidth of the filter and a_0 is expressed as the constant term, which is defined as $a_0=0.5(a_u-a_v)$. Here, each sub-filters are assumed as $H_j(z)$ order and type $K=K_j$ for k=0, 1, 2, .n.

Furthermore, the frequency response of the VBF sub-filter is articulated in eqn. (2),

$$H_j=e^{iK\omega\frac{t}{2}}H_{j,Z_p}(\omega t) \tag{2}$$

Where, Z_p represented as frequency response in terms of zero-phase with respect to $H_j(e^{i\omega t})$ and it is given in eqn. (3),

$$H_{j,Z_p}(\omega t)=\sum_{p=1}^{P}y_{ip}Tr\_g(p,\omega t) \tag{3}$$

This reason has motivated this research word towards this hearing aid field. The proposed variable filter bank is developed by an optimized FDS design that is reduced the matching error.

4 Proposed Methodology

The major aim of this work is to design a hearing support system for deaf people to improve their lifestyles. For that, the present article has planned to develop a novel W-FDVBF to enhance the function of the hearing aid system. Here, the parameters of the utilized filter were optimized by the grey wolf fitness. Subsequently, several metrics are calculated and compared with other techniques and have earned the finest results. The proposed framework is described in fig. 3.

Primarily, the original speech signal is taken as input, hereafter, this signal is passed through the analog to digital converter, which performs analog signal that is original signal is converted into the digital signal. Then, design the variable bandwidth filter using fractional delay structure based on the wolf optimization. Moreover, the filter parameters are optimized using wolf optimization. After that, the trained signal is given through the digital to analog converter, which performs reconstruct, the signal into the analog signal.

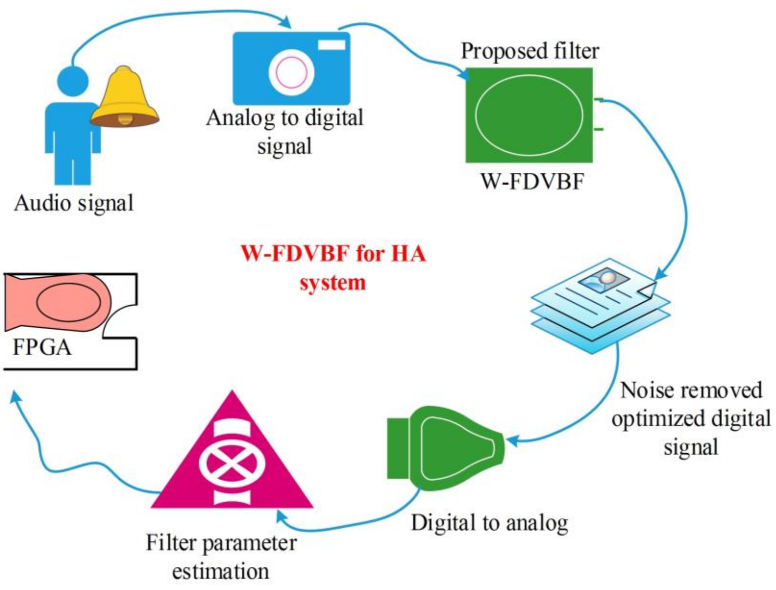

Fig. 3 Proposed methodology

4.1 Process of WFDVBF technique

The developed W-FDVBF replica is introduced for improving the HA systems. In this paper proposed variable filter bank using fractional delay structure is designed to reduce the matching error and no of multipliers. Here, VBF is used specific bandwidth and sufficient magnitude gain to reduce matching error. Moreover, VBF is provided and changed the gain to compensate for an audiogram of mild hearing loss at each frequency range. Therefore, an optimized W-FDVBF model is developed to evaluate the filter parameters based on the variable filter bank is illustrated in fig. 4. To evaluate the multi-objective problems Gray wolf optimization is developed. Moreover, it is the hunting mechanism here; it includes three parameters α, β, γ for these parameters that are used to tune the filter bank parameter.

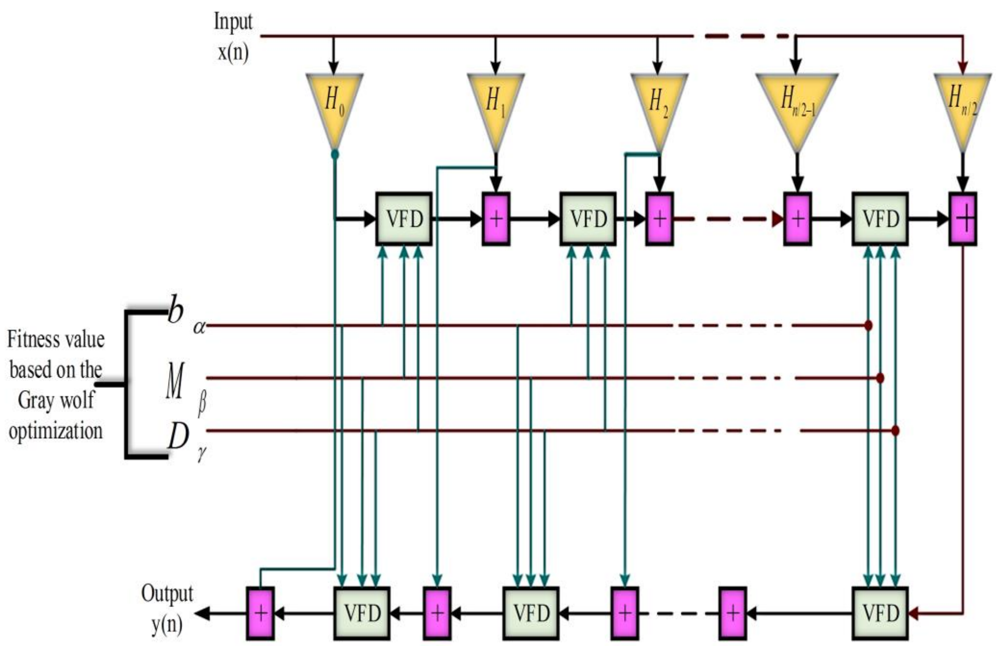

Fig. 4 Structure of proposed W-FDVBF

Thus, the overall transfer function of the proposed W-FDVBF is mentioned as eqn. (4),

$$H'(\delta, j\phi) = \{G'(\delta, j\phi)\}\{\sum_{n=0}^{n-1}\left[\sum_{m=0}^{m-1} h'(m,n)e^{-im(j\phi)}\right] j\phi^n\} \tag{4}$$

Here, the design procedure is to evaluate the $h'(m,n)$ such $H'(\delta, j\phi)$ is approximately referred as $H'_d(\delta, j\phi)$. Moreover, the frequency response of developed W-FDVBF is expressed in eqn. (5),

$$F(e^{i\omega}) = e^{-i(n/2)\omega}\prod_{n=1}^{N}\sum_{k=1}^{K}\frac{(-ia_n\omega^n)^K}{K!} = e^{-i(n/2)\omega}\sum_{P=0}^{P} b_p\omega^p \tag{5}$$

Where, b_p represented as polynomial phase element is related to the fractional delay D as $b_p = D^p$ so the frequency response of can be rewritten as in eqn. (6),

$$F(e^{i\omega}) = \sum_{p=0}^{P} D^p H'_p(e^{i\omega}) \tag{6}$$

Where, $H'_p(e^{i\omega})$ is denoted as sub-filters of farrow structure design approximation, also based on the frequency response input and output relationship is given in eqn. (7),

$$F(e^{i\omega}) = \frac{Y(n)}{X(n)} \tag{7}$$

Where, $y(n) = \sum_{i'=-\infty}^{+\infty} y(z)n^{-i'} \; and \; n = e^{i\omega}$ moreover, the sub-filter is used in a different order according to the filter requirements. Also, these different order sub-filters are established in terms of complexity. Moreover, multipliers are used to reduce the complexity. Here, the proposed design parameters are tuned into the fitness of grey wolf optimization. Thus, the transfer function is written as a function of $z \, and \, b_\alpha$ as,

$$F(z,b_\alpha) = \sum_{w=0}^{w}(b_\alpha - b_0)^w H'_w(z) \tag{8}$$

Where, $H'_w(z)$ denoted as linear phase FIR sub-filters and VBF polynomial is interpolated by w^{th} tunable change of particular bandwidth. Moreover, the difference between ideal $[F_i(z,b_\alpha)]$ and $[F(z,b_\alpha)]$ approximate frequency response is termed as matching error. Subsequently, a matching error is given in eqn. (9),

$$M_\beta(z) = F(z,b_\alpha) - F_i(z,b_\alpha) \tag{9}$$

Moreover, the filter parameter can be stated in eqns. (10) and (11).

$$1-\varphi_{pb}(b_\alpha) \leq |F(e^{i\omega t}, b_\alpha)| \leq 1+\varphi_{pb}(b_\alpha), \omega t \in [0, b_\alpha - \Delta(b_\alpha)] \tag{10}$$

$$|F(e^{i\omega t}, b_\alpha)| \leq 1+\varphi_{sb}(b_\alpha), \omega t \in [b_\alpha + \Delta(b_\alpha), \pi] \tag{11}$$

Where, $b_\alpha - \Delta(b_\alpha)$ and $b_\alpha + \Delta(b_\alpha)$ represented as the range transition width at all designed bandwidth b_α, φ_{pb} denoted as pass band ripple attenuation, φ_{sb} represented as stop band ripple attenuation. Moreover, the delay of the designed W- FDVBF is expressed in the eqn. (12),

$$D_\delta(\omega) = -\frac{d}{d\omega}\angle H'(e^{i\omega t}) \tag{12}$$

Consequently, the parameters of the developed filter bank like bandwidth, matching error, and delay, and have been tuned with the use of the GW fitness function that is declared in eqn. (13),

$$\vec{F}(t+1)+ = \frac{\vec{b}+\vec{M}+\vec{D}}{3} \tag{13}$$

Where, the filter parameters such as b, M, and D can be expressed as following eqns. (14), (15), (16),

$$\vec{b} = \vec{b}'_\alpha - \vec{R}_1(\vec{b}_\alpha) \tag{14}$$

$$\vec{M} = \vec{M}'_\beta - \vec{R}_2(\vec{M}_\beta) \tag{15}$$

$$\vec{D} = \vec{D}'_\delta - \vec{R}_3(\vec{D}_\delta) \tag{16}$$

Based on the fractional delay filter $\vec{b}'_\alpha$ designated as position alpha agent, $\vec{M}'_\beta$ denoted as the position beta agent, $\vec{D}'_\delta$ is the position delta agent, $\vec{R}_{1,}\vec{R}_2,\vec{R}_3$ is indicated as random vectors respectively. In addition workflow of the proposed W- FDVBF is demonstrated in algorithm 1.

Algorithm 1: Proposed W-FDVBF

Star

t

Initialize the optimization parameters

{

int b_α, M_β, D_δ

}

Initialize the filter parameter

{

Int x(n), y(n), b, M, D *// here, x(n) is input and y(n) is output,* $b_\alpha \Rightarrow$ *bandwidth,* $M_\beta \Rightarrow$ *matching error,* $D_\delta \Rightarrow$ *delay*

{

The design proposed FDVBF using wolf optimization

overall transfer function using eqn. (4)

$H'(\delta, j\phi)$ *// approximately referred as* $H'_d(\delta, j\phi)$

Frequency response using eqn. (5)

$$F(e^{i\omega}) = e^{-i(n/2)\omega}\prod_{n=1}^{N}\sum_{k=1}^{K}\frac{(-ia_n\omega^n)^K}{K!} = e^{-i(n/2)\omega}\sum_{P=0}^{P}b_p\omega^p$$ *// based on the sufilters*

input and output relationship using eqn. (7)

$[F_i(z,b_\alpha)] \rightarrow [F(z,b_\alpha)]$ *// matching error*

Calculate the fitness function

$b_\alpha \Rightarrow$ *Alpha agent tuned with bandwidth*

$M_{\beta} \Rightarrow$ *Beta agent tuned with matching error*

$D_{\delta} \Rightarrow$ *Delta agent tuned with delay*

While (t> max iterations)

For three search agent b_{α}, M_{β}, D_{δ}

{

Update filter parameters such as b, M, and D

}

Optimize the matching error (M), bandwidth (b) and delay (D)

End for

update optimized parameter

End while

Stop

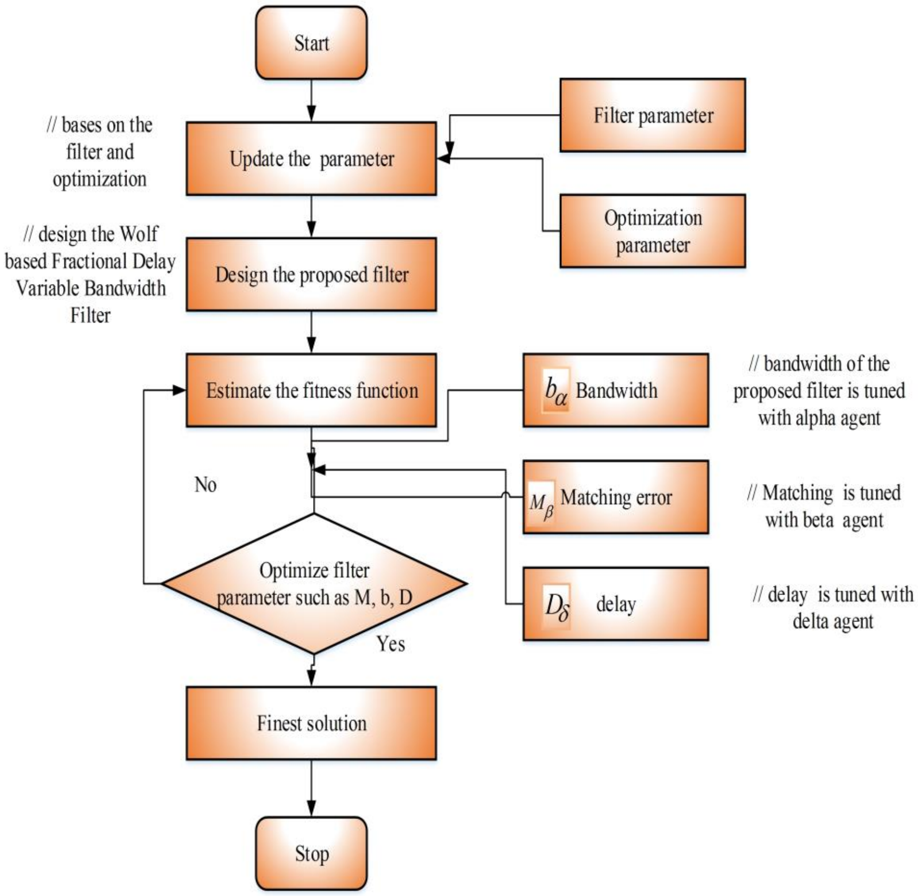

Fig. 5 Flow chart of proposed W-FDVBF

Moreover, the Flow chart of the proposed W-FDVBF is demonstrated in fig. 5. Here, initially tune the filter parameters in terms of optimization parameters.

Moreover, the schematic structure proposed W-FDVBF is demonstrated in fig. 6.

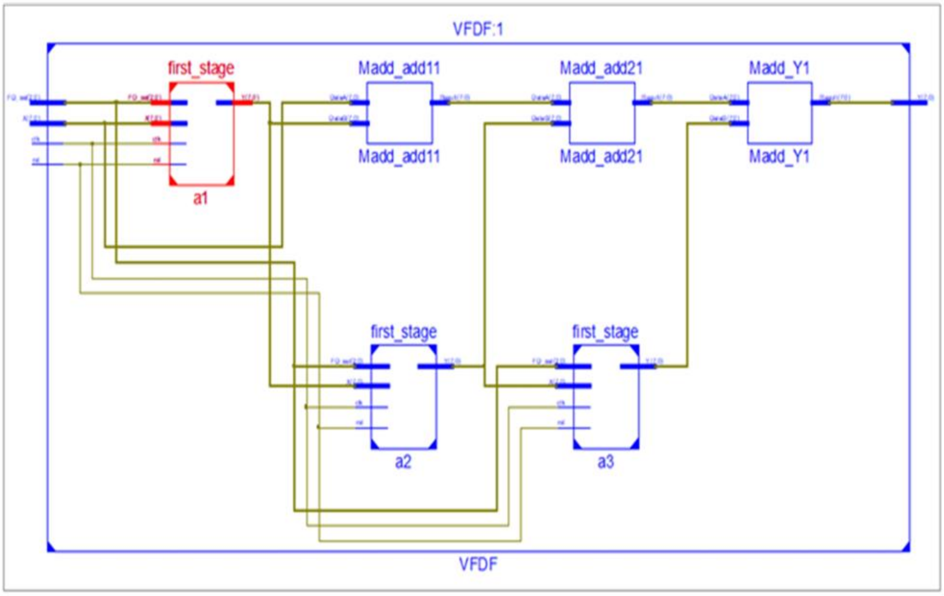

Fig. 6 Schematic structure of the proposed design

5 Results and Discussion

The proposed W- FDVBF technique is executed in the MATLAB framework and Xilinx. In this research, the HA system is developed in an FPGA kit.

The performance of the designed strategy is compared with other existing works in terms of, matching error, area, delay, no of multipliers, power, and speech perception. Initially, collect the speech signal and read to the system. Then, this speech signal is converted into a binary signal and applies to the proposed filtering techniques in the Xilinx platform. After that, apply the fitness function wolf optimization to refresh the attained outcomes from the Xilinx to achieve finest optimal speech

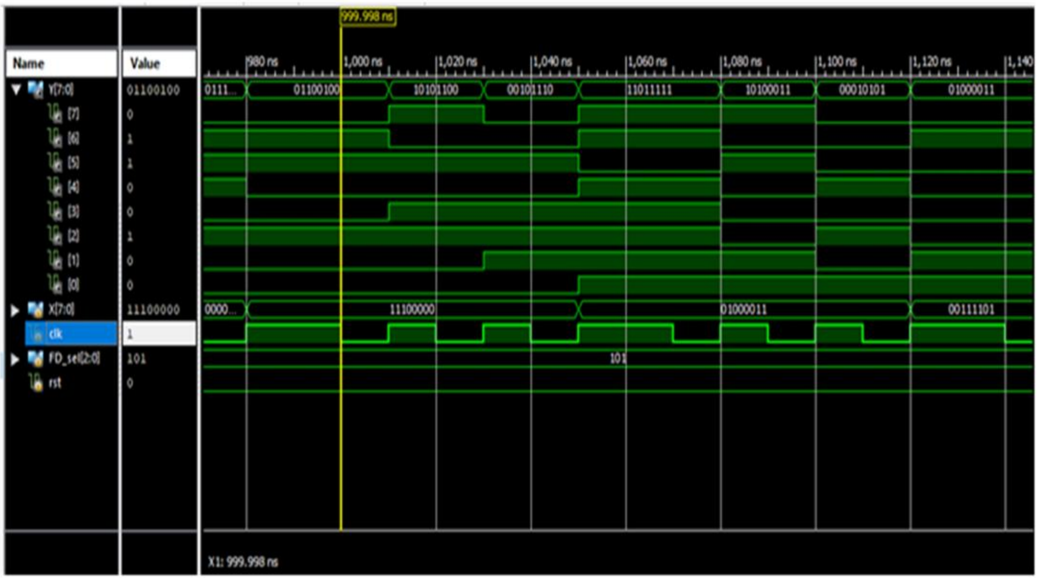

Fig. 7 FPGA module results

signal outcome.

The result of FPGA is detailed in fig. 7, which is executed in the Xilinx platform, vertex 7 families.

5.1 Case study

Generally, hearing loss is classified into four types such as mild, moderate, severe, and profound hearing loss. In this section, let us consider three design parameters like frequency, bandwidth, and gain of each band.

Table 1 Bandwidth design parameters

Band	Frequency (Hz)	Bandwidth[1]	Gain	Transition bandwidth
1	Initial 1000	1500	15	0.08465
2	800-2200	1500	52	0.0969
3	1800-2900	1500	64	0.115
4	2600-3600	1000	80	0.1344
5	3300-4300	1000	85	0.1434
6	4100-6000	1000	87	0.1456
7	5700-6200	1000	76	0.1625
8	6200-7000	1800	66	0.175
9	6400-7200	1900	54	0.1875
10	7000-7800	1500	53	0.1876

[1]Hearing frequency range at upper and lower ends

In addition, hearing loss patient data and their graphical representation is shown in fig. 8. For the testing process, initially, the tested data was trained to the system, immediately; the analog signals were converted to digital signals using the modulation model. Hereafter, the function filter has applied to

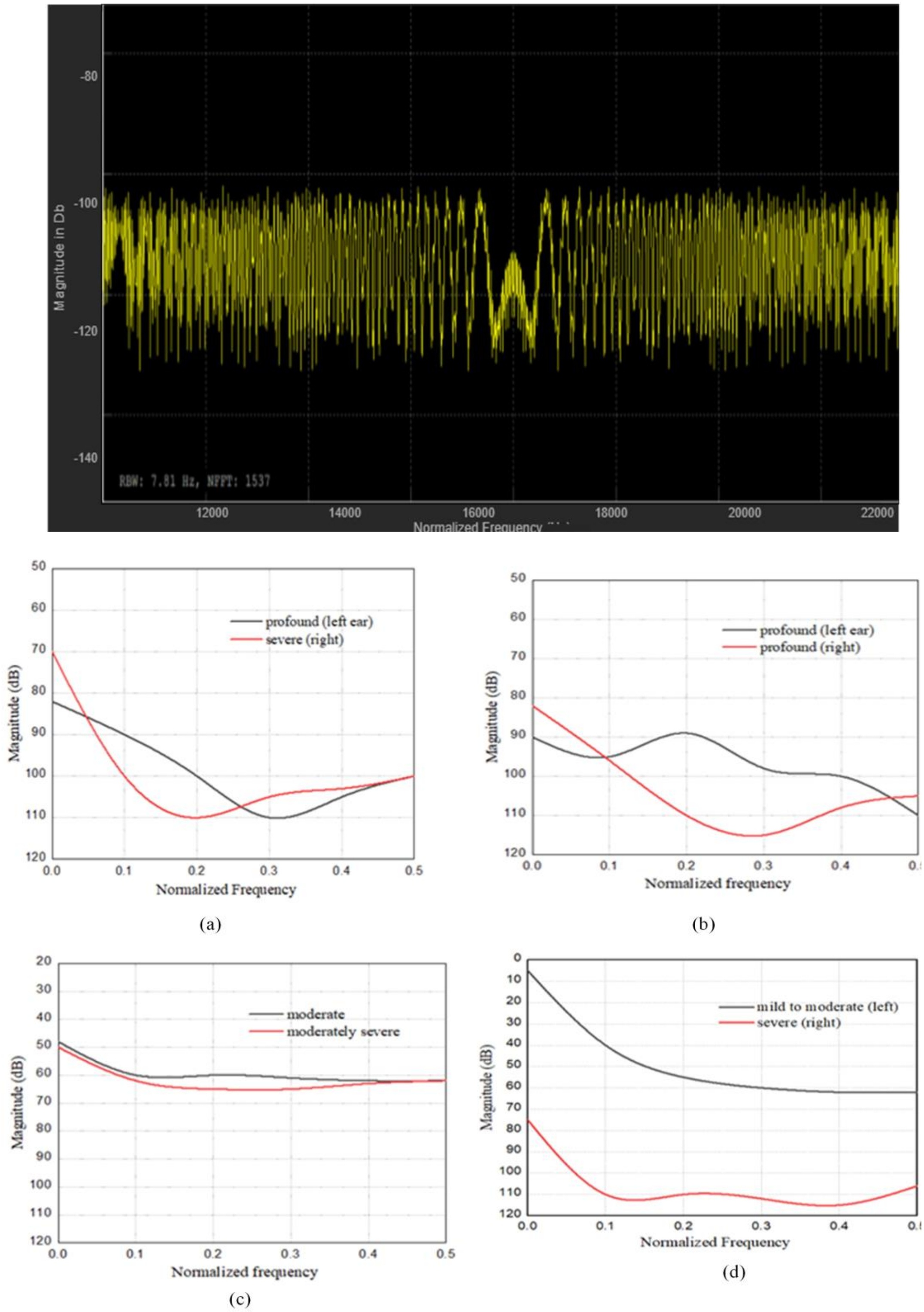

Fig. 8 Hearing loss patients data: a) the patient with severe and profound HL, b) profound HL patient data, c) moderate and moderately severe patients data, d) mild to moderate and severe HL patients data

remove the noise from the trained signal then the matching audiogram was obtained for the test data.

Let us consider passband ripple is 0.05 dB, stopband attenuation is 65dB. This condition shows the efficiency of the proposed optimized filter, while the fractional delay is structured with the variable bandwidth filter using HA systems. Moreover, the input audio signal has gained using MATLAB was described in fig. 9.

The frequency response of fractional delay variable bandwidth filter (FDVBF) is demonstrated in fig. 10; from this maximum optimum transition, width is 0.1625. Moreover, to increase the number of bands matching error is reduced based on the normalized frequency. Let us consider, the input signal having

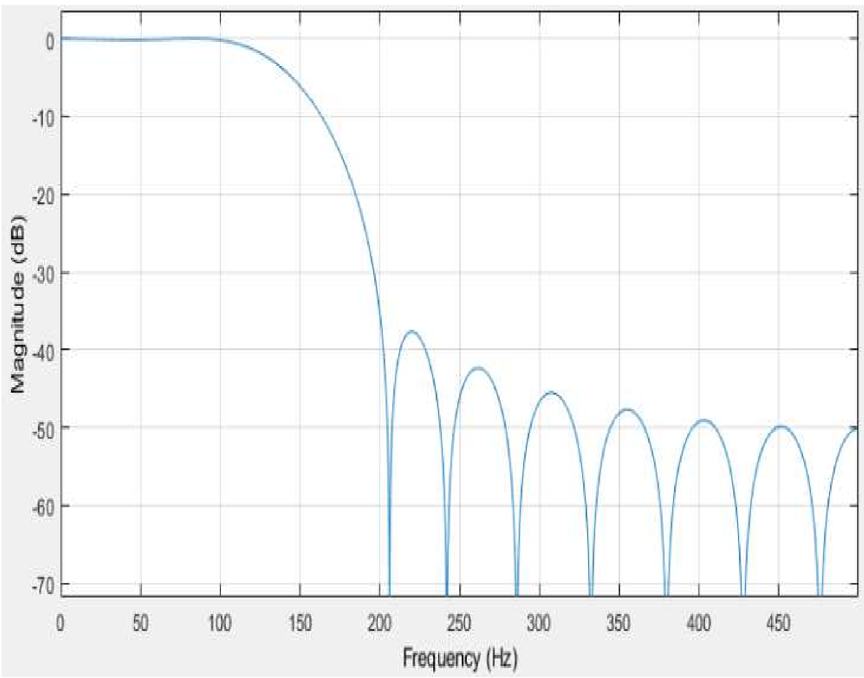

Fig. 10 Frequency response of the audio signal

frequencies 800Hz, 1000Hz, and 2200 Hz is passed through the newly designed FDVBW filter. To reduce the number of the multiplier in the FDVBW filter design, this donates towards power and area through the implementation. Moreover, the minimum order of FDVBW filter and optimal quantity of bands donates lower hardware complexity. Consequently, fractional delay structure is mainly used to providing improved tunability. Hereafter, the designed filter is matched with the set of an audiogram and compared with other hence the complexity of the hardware is low.

Initially let us consider mild to moderate hearing loss at low frequency The proposed FDVBW filter using 8 bands, 138 multipliers, and 275 adders so the maximum matching error 1.35. Secondly, let us consider mild hearing loss at all frequency ranges the designed filter utilizing 10 bands, 160 multipliers, and 319 adders hence the maximum matching error is 1.24. Moreover, hearing loss with various frequencies is demonstrated in fig. 11.

Furthermore, mild hearing loss at higher frequency the filter using 10 bands, 160 multipliers, and 319 adders for 1 band therefore the matching error is 1.86. Profound hearing loss filter design

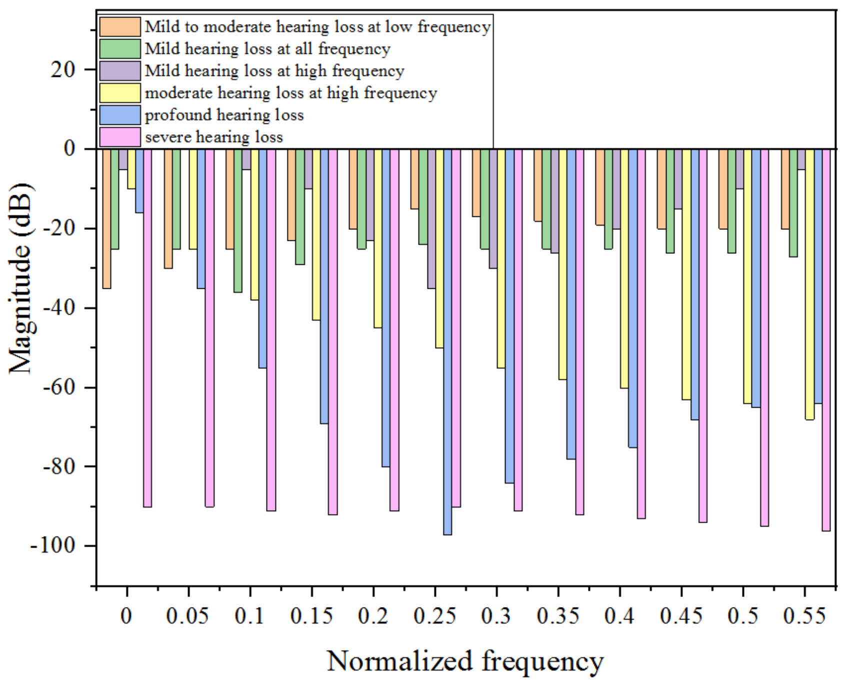

Fig. 11 Different HL data with various frequencies

parameters are 8 bands, 138 multiplier, and 275 adders, then the attained matching error is 2.51. Finally, severe hearing loss let us consider middle to high-frequency range, 160 multipliers, 10 bands, and 319 adders then the achieved matching error is 2.44. Moderate hearing loss selected multipliers 138, adders 275, and 10 bands therefore, the matching error is 1.76.

5.2 Performance assessment

Usually, in all cases, the performance of the system was verified using their parameter validation. Here also, the proficient score of the proposed filter was analyzed by validating its parameter. Moreover, in this research, the proposed filter was designed with the help of MATLAB and the FPGA model was simulated in the Xilinx platform.

5.2.1 Matching error

In this proposed novel filter, the signal is a typically irrational delay between dual input signal sampling points. Moreover, the parameter matching error was tuned with the use of an optimization approach. Hence, the optimization mechanism has provided better results, which are shown in fig. 12.

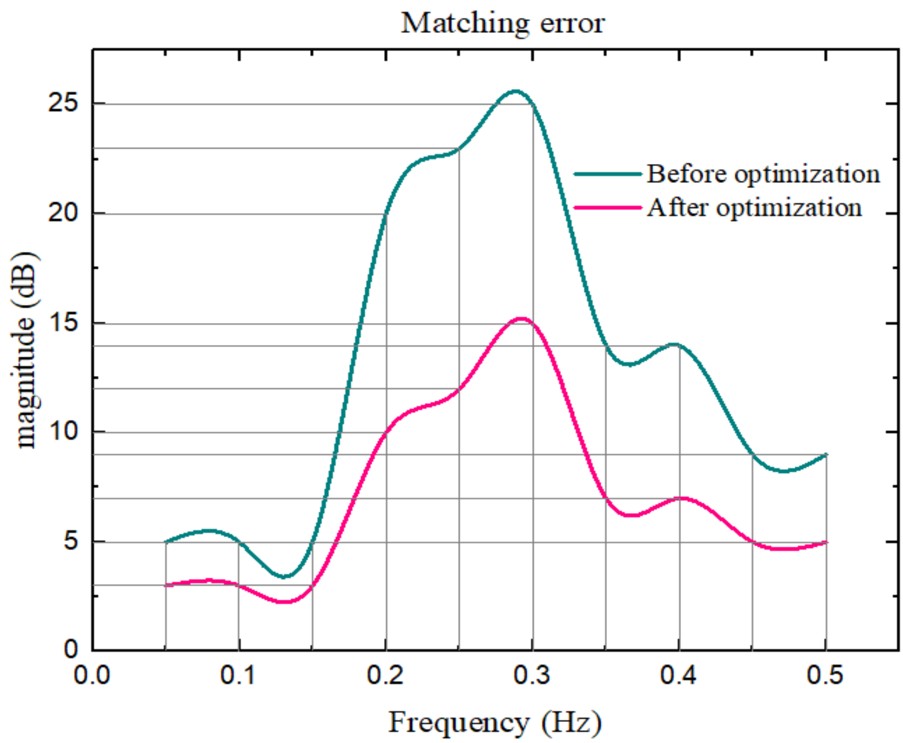

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Fig. 12 Matching error validation (before and after optimization)

aver
age matching error measure was evaluated; if the attained matching error was very high then the noise in the signal gets increased. To reduce this kind of barrier optimization strategy was incorporated in these filter applications.

- Mild hearing loss

Here, the matching error was specifically found for mild hearing defeat data. Moreover, the variance of the matching error for both left and ear was taken with dissimilar frequencies that are graphically proved in fig. 13. To find the matching error, initially, the input signal was matched with the output signal that is termed audiogram matching, which is shown in fig. 9. Here, the matching of the audiogram is measured in both sectors that are before and after optimization. The matching for mild hearing loss patients was shown in fig. 14 for both left and right ear. Once, the matching error was evaluated then the proposed filter was attempted to find the delay parameter that is shown in fig. 15.

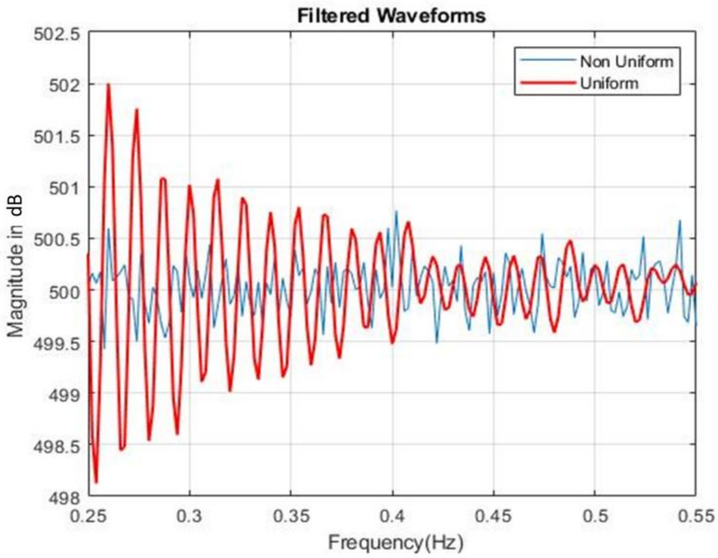

Fig. 13 Audiogram matching

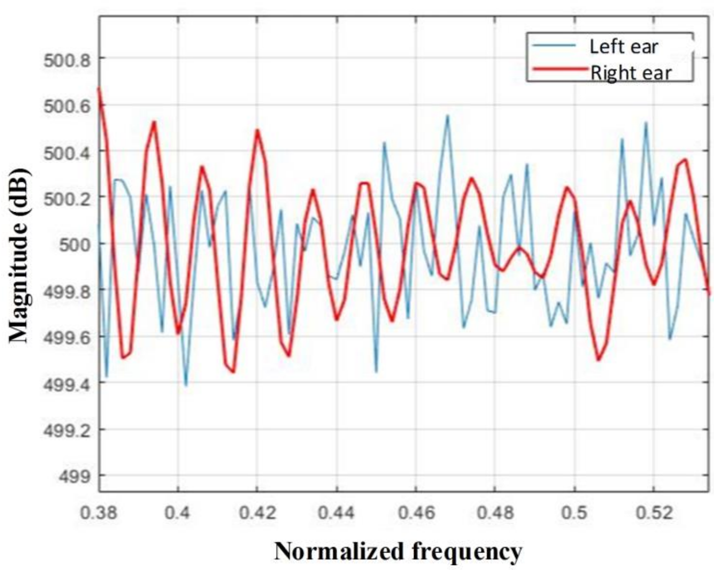

Fig. 14 Matching error of left and right ear

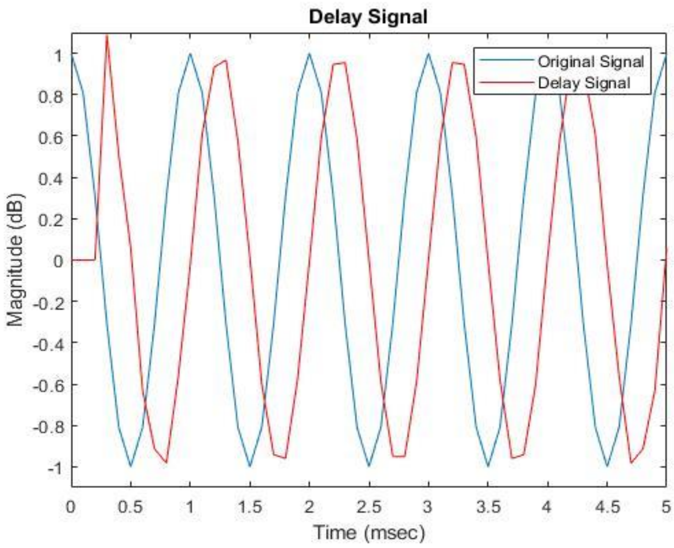

Fig. 15 Delay signal with the original signal

5.2.2 Power analysis

Power is an important factor for all digital applications, here, the consumed power by the proposed model is attained by the report that is described in fig. 16 and the comparison of power analysis is detailed in fig. 17. In comparison, the designed model consumed significantly less power than the other models under consideration. Also, the proposed design has maintained the stability of the power consumption in different frequencies. The overall comparison is shown in the table 2, here, the proposed model has attained fewer delay and area parameters than other models. Thus, the efficiency of the proposed method is verified and it is applicable in hearing aid applications.

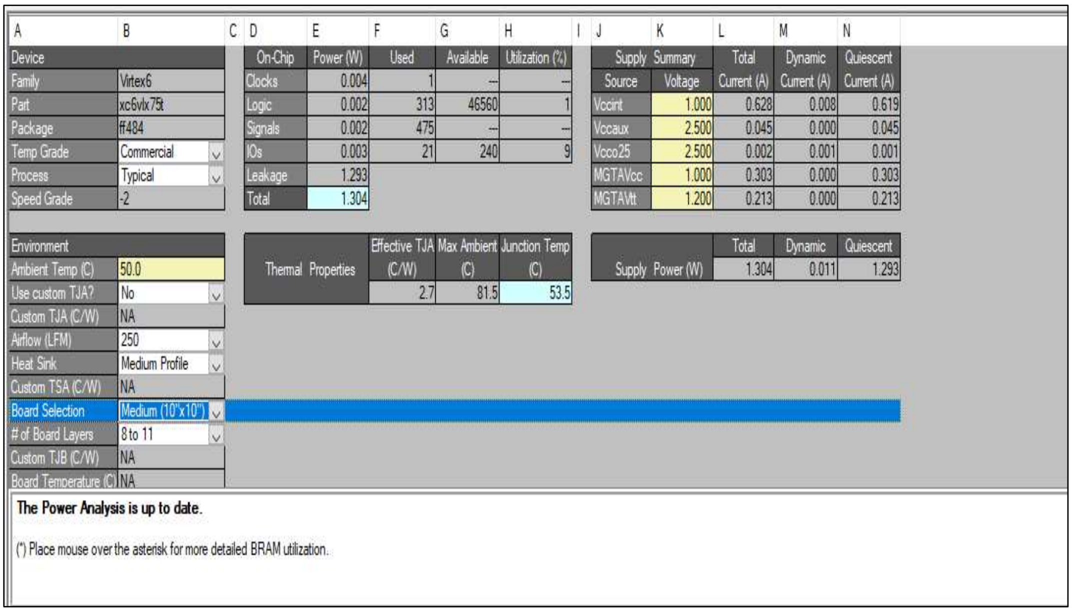

Fig. 16 Power analysis report of the proposed model

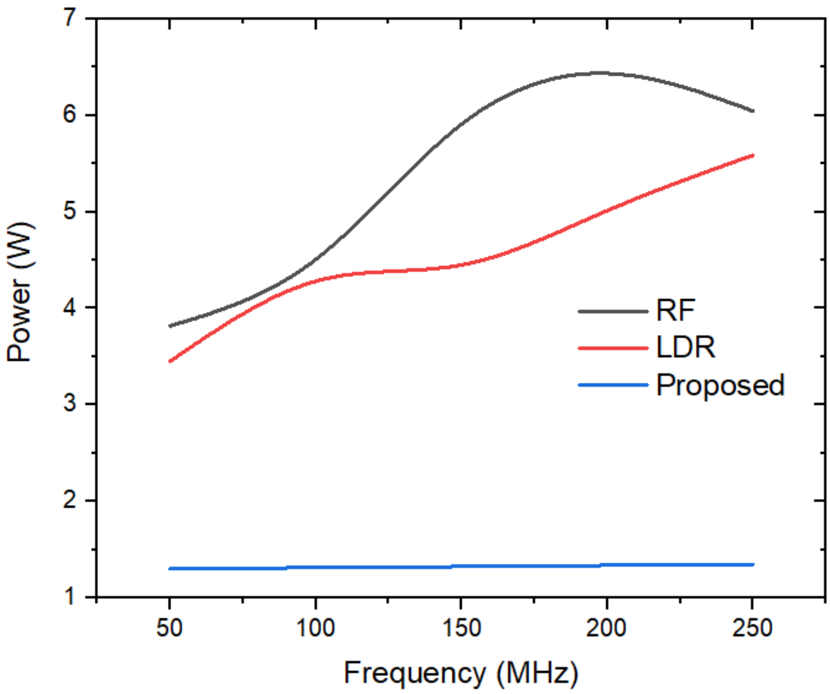

Fig. 17 Comparison of power

Table 2 Overall comparisons

Parameters[1]	RF[2]	LDR[3]	Proposed method
No. of Slice registers	325	367	224
No. of flip flops	288	267	214
No. of occupied slices	178	198	141
No. of bonded IOBs	50	60	21
No. of Slice LUTs	400	467	313
Delay(ns)	9.08	8.97	4.091

[1] Parameters that are used in filter designing

[2] It is the existing method Reconfigurable Filter

[3] It is the existing method Lower Delay Reconfigurable

6. Conclusion

An efficient technique for the design of a variable filter bank suitable for digital HA is developed in this article. The technique utilizes a fractional delay structure-based VBW filter. The chief reason for this proposed model is to enhance the hearing aid application by designing an efficient W-FDVBF. Here, the noise in the signal was reduced using the filter parameters, when the noise was reduced then the matching error was reduced. Hence, if the error was minimized then the processing signal was reached to the user with high frequency. Moreover, to validate the successive score of the proposed model the parameters of the designed filter were compared with other filter metrics and have achieved better results.

Acknowledgement

None

Compliance with Ethical Standards

1. Disclosure of Potential Conflict of Interest:

The authors declare that they have no potential conflict of interest.

2. Statement of Human and Animal Rights.

i. Ethical Approval

All applicable institutional and/or national guidelines for the care and use of animals were followed.

ii. Informed Consent

For this type of study formal consent is not required.

Data Availability Statement

My manuscript has no associated data.

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