Received: February 3, 2023. Revised: March 25, 2023.


International Journal of
Intelligent Engineering & Systems

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# Aquila Based Adaptive Filtering for Hearing Aid with Optimized Performance

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**Abstract:** In recent days, one of the most serious health problems in humans is hearing loss. For helping the hearing impairment, the Hearing Aid (HA) system is used, but it may lack in various aspects such as delay, matching error, power consumption, low complexity and unwanted noise. Many techniques were introduced to overcome these issues, but they still have problems. Therefore, this research uses a novel Aquila-based Adaptive Filter (AbAF) for improving the HA system with the optimized parameters of filter. Here, the incorporation of the Aquila optimizer along with the adaptive filter is considered as the key novelty of this present work. The main motive of this proposed model is to filter the noisy features more possible range for improve the HA performance. Hence, the Aquila agent removes the unwanted noise by updating the adaptive processor's filter coefficient and optimizing the filter parameters such as matching error, delay, complexity, and bandwidth and power consumption. Moreover, the suggested model is designed and executed in the MATLAB platform. Finally, the performance parameters were estimated and compared with the past studies to validate the performance improvement. The presented model scored the reduced Matching error as 0.3% and reduced power consumption as 10%, which quite lower than the compared models.

**Keywords:** Hearing aid, Adaptive filter, Optimization, Matching error, Noise signal.

## 1. Introduction

One of the most traditional forms of human communication is voice. For hard-of-hearing people, one needs to ask a hearing aid system for assistance to communicate with others. The typical hearing aid device does not delicately analyze background noises before amplifying voice signals [1]. However, when background noise is present, users constantly experience discomfort and find it challenging to interpret speech. As a result, reducing background noise and improving voice signal quality becomes crucial when building a hearing aid system. They are a little electronic device that amplifies sound and facilitates a clearer, easier understanding of speech [2]. A tiny microphone and speaker in the hearing aid capture the sound waves, boost the volume of quiet noises, and transmit them to the ear. Hearing aids are getting smaller and better due to the readily available microchips [3]. Around 10% of people worldwide experience some degree of hearing loss. But just a small percentage wear hearing aids because of several causes, including the stigma attached to using a hearing aid, not meeting the customer priority and needs and also the high pricing of the system [4]. They are small electronic instruments that maximize the sound and making it clear to hear and understand. With the use of the tiny microphone it picks the speech sound and makes it audible and dispatched it to the speaker [5]. The microphone today in the modern world got much smaller and the efficiency is increased [6]. About 10% of individual in the world's population are affected by the hearing loss. Although minimum number peoples were makes use of the HA system for the various unsatisfying reasons [7, 8].

The shift in the auditory threshold for detecting a pure tone relative to a normal ear is commonly used to quantify hearing loss [9]. Because of this, there are many different kinds of hearing aids with various features and functions to meet other demands. The study of digital hearing aids has advanced, and the devices now contain a tiny modifiable computer for amplifying millions of distinct sound signals, enhancing the hearing of those who are deaf or hard of hearing [10]. Frequency domain features are commonly used in sound processing. The audio data extracts include MFCC, support vector machines (SVM) various neural models are some of the frequently used classifiers [11]. Recent research for hearing aids introduces a new algorithm such as feedback cancellation (FBC), constriction of dynamic range (WDRC) and noise cancellation or reduction (NR) [12]. The hearing aids' flexibility and programmability are crucial because the algorithms for the HA system should be suitable for various ear impairments [13].

Although it satisfies the above problems, sometimes it is less focused on power consumption, efficient implementation and optimization for real-time operation [14]. DSP has also been used to enhance HA to reduce hearing loss's negative effects [15, 16]. Compared to other approaches, the power utilization and the surface area are limited in the DSP [17]. The filter bank with the audio categorizing ability [18], the speech improving process [19], and the approach for the adaptation of sparch [20] are some of the methodologies used in HA systems. Additionally, the HA system uses techniques. However, it has significant drawbacks. Therefore a new paradigm was designed in this research to overcome those limitations. Across the world lot of people suffers from hearing loss problem [21]. It occurs when the ability of any part of the ear is lost [22]. Several techniques are available for the voice signals improvement concerning the hearing of challenged people [23]. In this system, the entered speech signals are amplified [24]. Then the noise present in the incoming signals is removed by the filters [25]. Furthermore special solutions are activated for the flexibility of the hearing aid system for individuals under different conditions. To compare the loss of hearing patents, the HA is programmed based on their frequency specification. Here, the designed model is functioned with the help of Aquila optimizer. Hence, the filtering process is in optimal condition for every signalling feature. The filtering process was repeated continuously till the desired signal clarity is found. This iteration is processed based on Aquila best solution finding process. Hence, the presented model has scored the better filtering outcomes than the conventional filtering models. So, the signal clarity of the HA was improved, which is key advantages of the proposed model. The key contribution of the suggested AbAF is described as follows.

- Initially, the novel AbAF model is designed and the original speech signal is trained.
- Moreover, the adaptive processor's filter coefficient is updated to remove the noise
- Additionally, the fitness features of Aquila are activated to optimize the filters parameters and to update the filter coefficients.
- Subsequently, the AbAF model is tested in the MATLAB platform and the results are evaluated.
- Finally, computed the filter parameters and compared with other existing works.

The present study is organized as, the recent related studies and issues are explained in section 2, proposed strategy is given section 3, results are demonstrated in section 4 and section 5 concludes the research article.

## 2. Related works

Some of the recent works related to the hearing aid system are explained as follows,

The echo cancellation in signal processing is majorly played by the adaptive filter. So Zaman et al. [26] introduced a narrow transition band (NTB) design using distributed arithmetic. This approach is realized using the shift accumulates operations and look-up tables, and also it replaces the decimation filter with the single DA unit. It satisfies the low complexity by the proper initialization. It also achieves a less area delay product. However, it also amplifies the noise without discriminating the sound.

One of the developing fields is Acoustic signal classification which classifies the clear audio from the background noises. Therefore Mahesh and Shahana [27] designed a synthesis of filter bank (SFB) system using the low order constraints network. Here multiple features like Tonnetz, Chroma, Mel-spectrogram, Spectral contrast and MFCC are separated from the five classes of speech signals such as silence, noise, only speech and speech with noise. However, it has reported high delay.

Devis and Manuel [28] have implemented restoration audibility in HA (RAHA) for improving the HA performance. Moreover, the features and the classes are trained on the network for the different classification processes. It is efficient, consumes less memory, is suitable for the digital hearing aid system, and gained a high classification rate and low error rate. However, the addition of the pooling layer affects the working of the subsequent layers.

The existing hearing aid system does not classify desirable and undesirable noise types. So Mahesh and Shahana [29] introduced a least-square dynamic-filter (LSDF) that identifies the important noise and makes it audible using deep learning. Here instead of a classifier, a noise eliminating algorithm is used. This system initially enhances the speech and then the mapping function of all voices is studied to eliminate the other environmental noise. But, this model has reported high power usage.

Indrakanti et al. [30] implemented continuous-VBF (CVBF) framework to filter the noise of every continuous signal. Finally, the enhanced speech signal with tested with noisy features to measure the robustness of the implemented design. However, it has reported high matching error.

Deepu et al. [31] introduced an interpolated finite impulse response (IFIR) [31] to cancel feedback of acoustic based on probe signal. It consists of two adaptive filters. The impulsive algorithm based on delay adapted these two filters. The first filter receives the speaker signal, and the microphone sound is exhibited as the response. The gaussian probe processes the second filter. Further, it exchanges the coefficients to attain a decent result for the path of acoustic feedback. It gains a better accuracy of modelling, and output SNR is maintained. However, if the adaptation algorithm is not continuous or open-loop, the hearing aid's ahead path could be damaged, and the probe signal could be affected.

Devis and Manuel [32] presented octave interpolated filter (OIP) hearing aid enabling 5G IoT to transform encrypted AV to address the issues such as local privacy and eavesdropping. Here for the assurance of security, lightweight AV encryption is utilized. Also, for speech enhancement, the noisy signal of the encrypted AV is filtered by the deep learning model. The encryption process is carried out based on the techniques such as a piece-wise linear chaotic map (PWLSM), a secure hash and a Chebyshev map. This model reduces the computational complexity and minimizes real-time optimization problems. But it is complex in design because of encryption model in the interpolated techniques.

Park et al. [33] have designed analog correlator (AC) for the HA system to regulate the HA controller parameters. Here, the HA system is tested in the cadence framework. Here, the features analysis process is more exact. However, it has reported high matching error.

Recent associated articles were discussed in this section. Every HA system is suitable and efficient for any functioning model. However, the tuning function is not appropriate, which has resulted in high matching error and power consumption. Considering this, the present study has incorporated the optimizer along with the adaptive filtering. Here, the Aquila optimizer provided the finest tuning outcomes that have resulted in the most satisfactory outcome.

## 3. Proposed methodology

The digital HA is mostly helpful for people who have lost their hearing, but the HAS's primary problems are error, unwanted noise, latency, and excessive complexity. Numerous techniques have been devised to improve the HAS, but they suffer from problems such as error in matching, noise, latency, dissipation of power and huge complexity. Therefore, the present research introduced a novel Aquila-based Adaptive filter (AbAF) for the hearing aid system enhancement and the solution for the problem of matching error, unwanted noise and complexity. This work aims to provide a better improvement of lifestyles of the deaf persons. The variables of the adaptive filter are adapted by the fitness function of the Aquila. Furthermore, the performance parameters of the designed system are evaluated and then validated with the current hearing aid system to record the efficiency and improvement score. The proposed architecture is given in Fig. 1.

The proposed architecture takes the speech signal as input and is trained to the filter. The received signal is in the analog. So, transform it into a digital signal and is passed to the ADC. Then the adaptive filter is designed based on the Aquila optimization. Subsequently, the filtering parameters of the adaptive filters are optimized by the Aquila optimization after the signal is processed in the developed technique to get the clear noise removed digital signal, which performs the digital to analogue conversion.

### 3.1 Process of proposed AbAF technique

The designed AbAF is presented to enhance the HA system. Here the AF works with sufficient magnitude gain and bandwidth for the reduction of the utilization multiplier number and matching error. Moreover, AF is offered to gain an adjusted account for a voice pulse showing mild stage at all frequency band. As a result, an improved AbAF model is created to assess the parameters of the filter depending on the adaptive filter, and its structure is given in Fig. 2. To determine the noise cancellation issue, the optimization of the Aquila is utilized. The

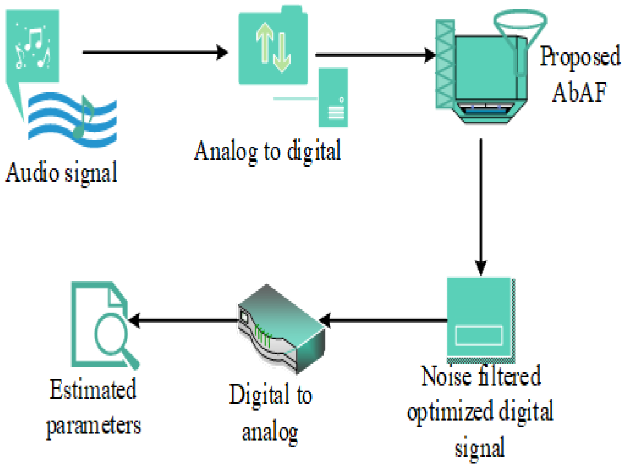


Figure. 1 Proposed methodology

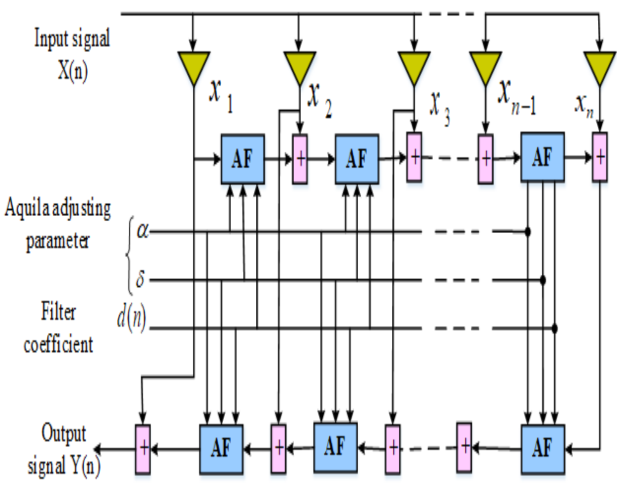


Figure. 2 Structure of the AbAF

exploitation mechanism of the Aquila is used for tuning the parameters of the filter bank. Here the exploitation adjustment parameters $\propto$ and $\delta$ are used for tuning the filter parameters.

In the proposed model, for each cycle the variation of the output and desired signal is computed. The computed difference adds the error value. Then the filter coefficient is updated by the calculated error values. The input form of the proposed filter is given in Eq. (1); this process is utilized to import the audio signal to the designed filter.

$$X(n) = [\sum_{h=1}^{n} x_h]^T \tag{1}$$

Here $X(n)$is an input signal, and $T$is signal vector transpose. Also, error waveform of the proposed filter is given in Eq. (2). By performing the Eq. (2), the error waveform of every audio signal was recognized. This error score is attained in the form of noise frequency.

$$e(n) = d(n) - Y(n) \tag{2}$$

Here $e(n)$ represents error waveform at each cycle, desired waveform is indicated as $d(n)$ and $Y(n)$ resulted output signal at each cycle. The filter coefficient vector of the filter is given in Eq. (3). The filtering level is varied based on the signalling noisy features. Hence, to tune the filtering based on the required filtering range, the filtering coefficient has to be set.

$$d(n) = [\sum_{s=0}^{n} d_s(n)]^T \tag{3}$$

The updating function of the filter coefficient based on the error signal obtained at each cycle is given in Eq. (4). To tune the noise at the maximum level, the error value has been identified at every digital pulse. Hence, the error is calculated for every cycle.

$$d(n+1) = d(n) + X(n)e(n)\mu \tag{4}$$

Here $\mu$ is the size of the step for parameters. The adaptive processor coefficients are updated in a fashion that lowers the mean square of the error signal. The mean square error (MSE) is given in Eq. (5).

$$\xi^2 = E[e(n)^2] \tag{5}$$

After updating the filter coefficient along with the optimal matching error at each cycle, the final output signal is obtained from the filter. It is the tuned signal considering the trained input signal. The output signal is in the form given in Eq. (6),

$$Y(n) = X^T(n)d(n+1) \tag{6}$$

The ideal ($\lambda_i[p, B]$) and the approximate ($\lambda[p, B]$) response of frequency variation are defined as the error in matching. Here the term B represents the bandwidth, and p is the signal's phase. Subsequently the designed filter's parameters such as error in matching, delay and bandwidth, have been optimized, and the filter's coefficients are updated with the help of the fitness function of the Aquila. It is expressed in Eq. (7).

$$F(n+1) = [Y_b(n) - \xi^2 + d(n)] \times \alpha - rand + (M + l + B) \times \delta \tag{7}$$

**Algorithm 1: AbAF**

***Start***
***{***
*Initialize audio signal*

*Convert analog signal to digital signal*
***Designing of proposed AbAF***
***{***
*Optimization paramters initialization*
*{*
*int* $\propto, \delta$
*}*
*filter parameters initialization*
*{*
*int* $X(n), Y(n), B, l, M$
*//here,* $B$ *is the bandwidth,* $l$ *is delay and* $M$ *is matching error*
***}***
*The filter coefficient vector updating using Eq. (3)*
*{*
$$d(n+1) \rightarrow \mu e(n)X(n) + d(n)$$
*// it will eliminate the noise signal completely*
***}***
*Input and output relation*
*{*
$$\lambda_i[p, B] \rightarrow \lambda[\mathrm{p}, \mathrm{B}]$$
*//computation of error in matching*
*}*
*HAS optimization*
*{*
*Fitness function evaluation*
$$F(n+1) \rightarrow [Y_b(n) - \xi^2 + d(n)] \times \alpha$$
*//the filter coefficient updating is tuned with the Aquila agent* $\alpha$
$$F(n+1) \rightarrow (M + l + B) \times \delta$$
*//delta agent is tuned with the matching error, delay, complexity and bandwidth*
*}*
***}***
***}***
***End***

Here $F(n+1)$ is the fitness process of the Aquila optimizer, $rand$ is the random value between 0 and 1, $l$, $B$ and $M$ are the terms of the filter bank such as delay, bandwidth and error in matching, and $Y_b(n)$ is the best approximate signal at each cycle of the adaptive filter.

The steps and processes presented in the designed model were detailed in Algorithm 1. The MATLAB code was executed based on these step processes, and the results were verified. The algorithm incorporated all mathematical function parameters in the pseudocode format. These processes are given step by step in Algorithm 1 and the flow diagram of the proposed model is given in Fig. 3. Based on the defined functional mathematical formulation, algorithm 1 is framed. Here, the needed parameter variables for each function process were initiated before performing the function process. Once all the variables are defined then the specific function was activated to meet the specific contributions. Here, the filtering process is tuned to the desired level by measuring the matching error percentage. The optimized machining error value was set in the Aquila optimal module then the iteration was continued till the fixed matching error is found.

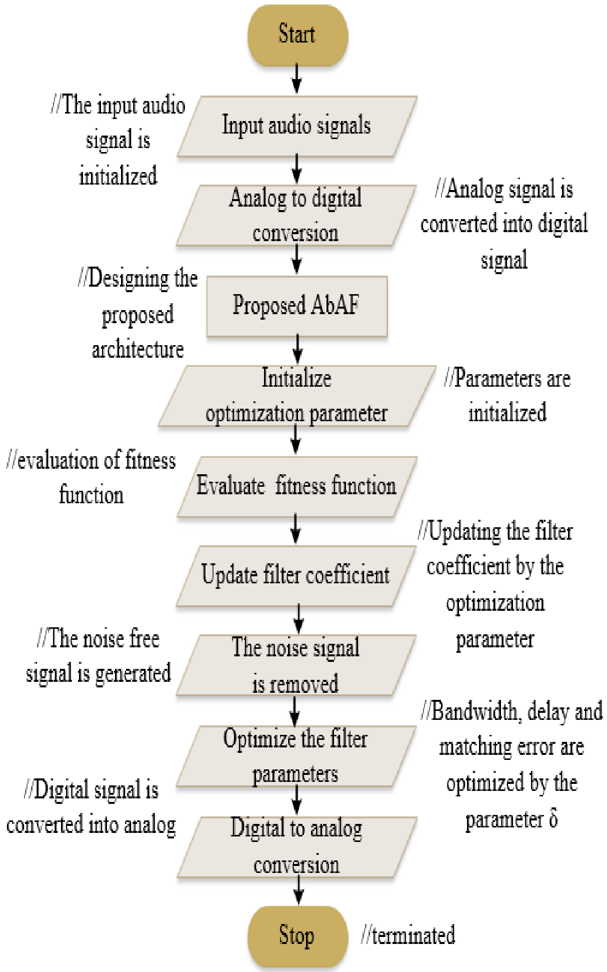


Figure. 3 Flowchart AbAF

## 4. Results and discussion

Generally, a unique adaptive filter was planned and designed to optimize and enhance the HA system's performance for impaired individuals.

Table 1. Apparatus required for implementation

| Parameters | Description |
|---|---|
| OS | Windows 10 |
| Platform | MATLAB |
| Version | R2021a |
| Input format | Audio signals |

Moreover, the proposed AbAF technique is designed and executed in the MATLAB platform. Furthermore, evaluated the proposed model's efficiency and made a comparison with the other flourishing approaches in terms of power consumption, delay, complexity, error in matching and frequency response. The apparatus required for implementation are listed in Table 1.

### 4.1 Case study

Usually, in many ways the human life affected by the loss of hearing, such as physically and emotionally, in relationships, societies, workplaces, institutions etc. It is one of the most serious health problems in humans. Therefore, a unique AbAF model is introduced to enhance the hearing aid system and satisfy deaf people. This technique removed the unwanted noise signals and reduced the complexity. Moreover, the severity of the hearing loss is classified under four conditions. They are moderate, severe, mild and normal. Initially, the input audio signal is trained to the proposed filter and is then transformed from analog to digital. These converted digital signals are injected into the proposed filter. The process using the proposed methodology for enhancing the HA system is detailed in Fig. 4.

Initially the test data is trained to the proposed model and transformed into digital signal. The proposed methodology removes the unwanted signal from the input by updating the filter coefficient of the adaptive processor. The filter parameters such as bandwidth, delay, complexity and matching error are optimized.

The updating of the filter coefficient and the optimization of the filter parameters are satisfied by evaluating the fitness function of the Aquila. Thus the suggested model gained a low rate of matching error, irrelevant noise, and delay and power consumption.

### 4.2 Performance analysis

The proposed AbAF technique is developed and implemented in the MATLAB platform version R2021a on Windows 10 OS. The efficiency of the

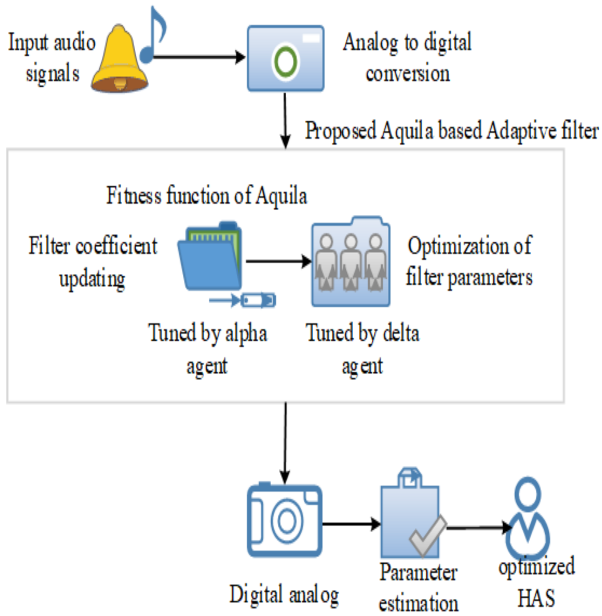


Figure. 4 Optimized HAS

proposed AbAF method is computed with the validation of the efficiency parameters such as error in matching, delay, power consumption and complexity and compared with the other current functioning techniques. The performance parameters of the suggested scheme are related with existing approaches such as NTB [26], SFB [27], RAHA [28], LSDF [29], CVBF [30], IFIR [31], OIP [32], AC [33].Here, the discussed existing studies are implemented in the same proposed platform and the performance was measured and compared with the proposed model.

#### 4.2.1. Complexity

The input size will fluctuate depending on how many resources are needed to execute MATLAB, and the complexity is represented by the notation $h(n)$. The complexity for worst case $h(n)$is represented as the largest amount of resources required for each input size, where $n$ is the input size [34]. As a result, the complexity is determined by how many times the execution of the program is replayed in the loop. Also, the complexity comparison is described in Table 2.

The evaluated complexity of the proposed AbAF technique is collated with other flourishing methods such as LSDF, OIP, NTB, and RAHA. The current design of NTB and RAHA attained 30% for 0.2 multipliers, and the OIP gained the complexity rate of 40%. Additionally, 50% is gained for LSDF, but the developed AbAF scored a low level of complexity.

Thus, the proposed AbAF technique has performed in 26% of complexity rate, which is low

Table 2. complexity rate

| Amount of multipliers | Complexity (%) | | | | |
|---|---|---|---|---|---|
| | RAHA | NTB | OIP | LSDF | Proposed |
| 0.2 | 30 | 30 | 40 | 50 | 26 |
| 0.4 | 63 | 57 | 59 | 65 | 35 |
| 0.6 | 138 | 36 | 75 | 45 | 29 |
| 0.8 | 84 | 52 | 80 | 68 | 43 |
| 0.10 | 67 | 61.11 | 92 | 74 | 59 |

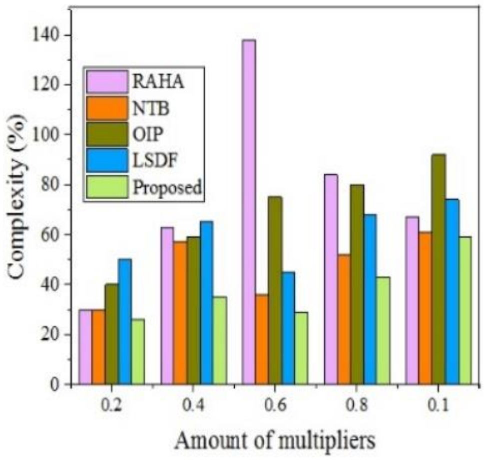


Figure. 5 Comparison of complexity

compared to another replica. Additionally, the complexity compared with the current technique has detailed in Fig. 5.

#### 4.2.2. Delay

It is the major significant term for the processing of the filter bank applied in the HA system. Here the delay value is calculated relayed on most maximum and minimum numbers [35]. The delay rate is calculated by dividing a multiplier numbers in the single band by the subfilter count in the band. It is expressed in Eq. (8),

$$Delay = \frac{N}{S} \tag{8}$$

Where N is the number of multipliers in one band and S is the subfilter's count in the full bands. Moreover, the comparison of the rate of delay is given in Table 3.

The low delay rate shows the better performance of the system. The suggested AbAF obtained a delay rate of 7% of 50 bands. For 50 bands, other current models such as CVBF scored dealy of 39ms, OIP attained 21ms, SFB gained 43ms and the IFIR ended

Table 3. Comparison of delay rate

| Number of bands | Delay (ms) | | | | |
|---|---|---|---|---|---|
| | SFB | CVBF | OIP | IFIR | Proposed |
| 10 | 18.61 | 29 | 31 | 48 | 12 |
| 20 | 13.56 | 25 | 10 | 10 | 5 |
| 30 | 34.7 | 36 | 26.6 | 13.58 | 10 |
| 40 | 56 | 45 | 18.5 | 10 | 6 |
| 50 | 43 | 39 | 21.6 | 9.75 | 7 |

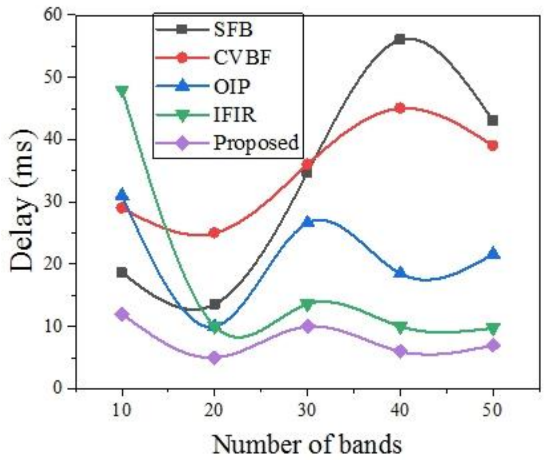


Figure. 6 Comparison of delay rate

with the 9.75ms delay. The proposed AbAF technique attained a low delay rate compared to another model.

The high delay makes it difficult to lips reading for impaired people, and the comparison for achieved delay is explained in Fig. 6.

#### 4.2.3. Matching error

The difference of the measurement the input audiogram and hearing loss patients' matching curves is termed the matching error. The computation of the value of the matching error identifies the capability of designed system of HA. Also, the computed value for the developed filter bank is listed and compared with the other existing methods in Table 4.

The proposed AbAF replica attained an error rate in matching of 0.64dB for 50 bands, and the current techniques of SFB and RAHA gained 4.12dB and 2.32dB of error in matching for 50 bands. Furthermore, the LSDF earned 3.75dB in matching error, and the CVBF attained a 1.40dB matching error rate for 50 bands. The proposed AbAF technique attained a low matching rate compared to another model.

Table 4. Matching error comparison

| Number of bands | Matching error rate (dB) | | | | |
|---|---|---|---|---|---|
| | RAHA | SFB | LSDF | CVBF | Proposed |
| 10 | 4.23 | 2.09 | 1.24 | 1.40 | 0.3 |
| 20 | 3.61 | 3.12 | 1.35 | 1.39 | 0.41 |
| 30 | 3.28 | 2.51 | 2.51 | 1.29 | 0.52 |
| 40 | 1.49 | 3.43 | 3.2 | 0.85 | 0.62 |
| 50 | 2.32 | 4.12 | 3.75 | 0.87 | 0.64 |

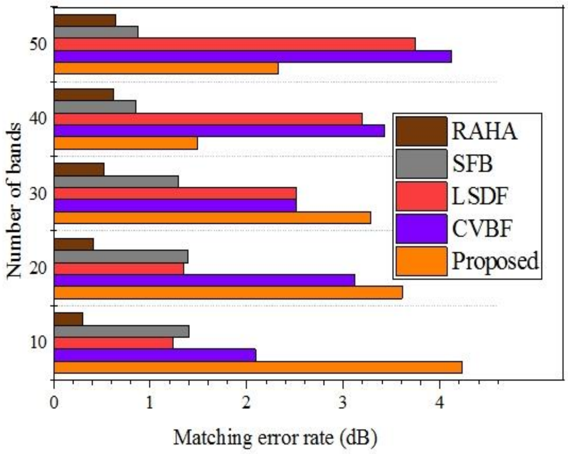


Figure. 7 Comparison of matching error rate

Table 5. Power consumption rate

| Bands rate | Power consumption (%) | | | | |
|---|---|---|---|---|---|
| | RAHA | OIP | NTB | LSDF | Proposed |
| 10 | 25 | 40 | 30 | 43 | 10 |
| 20 | 32 | 46 | 45 | 53 | 12 |
| 30 | 38 | 50 | 50 | 59 | 24 |
| 40 | 48 | 63 | 52 | 68 | 30 |
| 50 | 53 | 65 | 60 | 74 | 37 |

The comparison for the matching error rate is detailed in Fig. 7. Thus, the proposed AbAF model attained a 0.3dB error rate in matching for 10 bands. Comparing with other approaches, the developed AbAF method obtained a minimum error rate. As a result, the working of the HA system is enhanced.

#### 4.2.4. Power consumption

It is calculated by dividing the energy by the time that the filter consumed to solve the issues of the HA system, such as matching errors, delay and unwanted noise. The power consumption of the developed model is mentioned in voltage. The mathematical

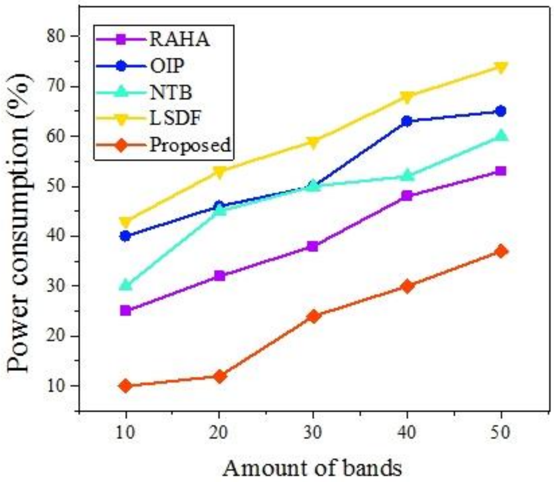


Figure. 8 Comparison of power consumption

formulation for calculating the power consumption is given in Eq. (9).

$$P = \frac{1}{t/_{E}} \tag{9}$$

Here t denotes the time the filter takes to process the signal, and E is the energy the filter brings to reduce the issues such as matching errors, delay and unwanted noise. The comparison of the power consumption is given in Table 5.

The developed AbAF technique consumed 37% of power for resolving the issues of matching error, delay and noise signal in 50 bands, and 10% of the power was consumed for 10 bands. While the other current approaches, such as RAHA, gained power rates of 53% and 25% for 50 rounds and 10 bands, OIP achieved 65% and 40%, NTB obtained 60% and 30%, and technique LSDF achieved 74% and 43% for the 50 bands and 10 bands. The power consumption results are detailed in Fig. 8.

The performance metrics of the designed AbAF technique achieved satisfying results for optimizing the HA system. The reduction of the noise signal, matching error, power consumption, complexity and delay is improved from the other current techniques. Thus the developed AbAF method is very flexible and useful for the deaf persons.

## 5. Conclusion

Reducing the matching error, delay, complexity, power consumption, and unwanted noise is the more complex process in the modern hearing aid system. Therefore, this research introduces a novel AbAF technique to resolve the above issues and enhance the HA system for the easy use of hearing loss people.

This technique improves hearing aid and reduces the risks that deaf people face. This technique optimizes filter parameters by the Aquila optimization parameter α. Also, the unwanted signals in the input audiograms are reduced by updating the filter coefficient of the adaptive processor. The Aquila parameter $\delta$ tunes this filter coefficient updating. Thus the overall working of the hearing aid system is improved by the proposed AbAF model. Moreover, the validations of the parameters are performed to identify the efficiency of the proposed model. The developed AbAF method attained a low error rate in matching, and gained only 10% in the rate of power consumption. Also the complexity is 25%; which is very low from the other approaches. In the future, a hybrid adaptive filter will be planned for satisfying the people with better outcomes

## Acknowledgements

This research was supported by the School of Electronics and Communication Engineering, Dr. Vishwanath Karad MIT World Peace University, Pune and Department of Electronics and Communication Engineering, Sinhgad College of Engineering, Vadgaon, Pune, Maharastra, India. I thank it for providing us with the capacity to conduct and complete this research

## Conflicts of interest

The authors declare no conflict of interest.

## Author contributions

Ujjwala Shitalkumar Rawandale, the first author contributed in conceptualization, methodology, software, validation, formal analysis, investigations, writing—original draft preparation, writing—review and editing and visualization. Sanjay R. Ganorkar and Mahesh T. Kolte, the second authors supervised and administered project.

## Notation

| | |
|---|---|
| $X(n)$ | Input signal |
| $T$ | Vector transpose |
| $e(n)$ | Error waveform |
| $d(n)$ | Desired waveform |
| $Y(n)$ | Resulted output |
| $\mu$ | Step parameter size |
| $\lambda[p, B]$ | Response frequency |
| $F(n+1)$ | Fitness function of Aquila |
| $rand$ | Random value |
| $l$ | Delay of filter bank |
| $B$ | bandwidth |
| $M$ | error |
| $Y_b(n)$ | Filtered best signal |
| $\alpha$ | Signal analysis parameter |
| $\xi^2$ | Filtering coefficient |

## References

[1] J. Lehmann, N. Christen, Y. M. Barilan, and I. Gannot, "Age-related hearing loss, speech understanding and cognitive technologies", *International Journal of Speech Technology*, Vol. 24, No. 2, pp. 509-16, 2021.

[2] U. S. Rawandale, S. R. Ganorkar, and M. T. Kolte, "Audiogram Study in Filter Bank Used for Hearing Aid System to Enhance the Performance", In: *Proc. of 2022 6th International Conference On Computing, Communication, Control And Automation (ICCUBEA)*, Pune, India, pp. 1-4, 2022.

[3] S. A. Nossier, J. Wall, M. Moniri, C. Glackin, and N. Cannings, "An experimental analysis of deep learning architectures for supervised speech enhancement", *Electronics*, Vol. 10, No. 1, pp. 17, 2020.

[4] Y. Yokota, A. Yamashita, S. Koyama, K. Kitano, S. Otsuka, and T. Kitahara, "Retrospective evaluation of secondary effects of hearing aids for tinnitus therapy in patients with hearing loss", *Auris Nasus Larynx*, Vol. 47, No. 5, pp. 763-8, 2020.

[5] A. Pati, R. Kumar, M. M. Dublish, and M. S. Singh, "Review Paper on Innovation of Advanced Noise Reduction Technique for Better Audibility Process", *International Journal for Research in Applied Science & Engineering Technology*, Vol. 9, No. XI, 2021.

[6] Y. Zhu, J. Zhang, Z. Zhang, G. Clepper, J. Jia, and W. Liu, "Designing an Interactive Communication Assistance System for Hearing-Impaired College Students Based on Gesture Recognition and Representation", *Future Internet*, Vol. 14, No. 7, p. 198, 2022.

[7] J. E. Ruusuvuori, T. Aaltonen, I. Koskela, J. Ranta, E. Lonka, I. Salmenlinna, and M. Laakso, "Studies on stigma regarding hearing impairment and hearing aid use among adults of working age: a scoping review", *Disability and Rehabilitation*, Vol. 43, No. 3, pp. 436-46, 2021.

[8] M. Bucholc, P. L. McClean, S. Bauermeister, S. Todd, X. Ding, Q. Ye, D. Wang, W. Huang, and L. P. Maguire, "Association of the use of hearing aids with the conversion from mild cognitive impairment to dementia and progression of dementia: A longitudinal retrospective study", *Alzheimer's & Dementia: Translational Research & Clinical Interventions*, Vol. 7, No. 1, p. e12122, 2021.

[9] E. Berninger, M. Drott, M. Romanitan, L. Tranebjærg, and S. Hellström, "Congenital Nonprofound Bilateral Sensorineural Hearing Loss in Children: Comprehensive Characterization of Auditory Function and Hearing Aid Benefit", *Audiology Research*, Vol. 12, No. 5, pp. 539-63, 2022.

[10] U. S. Rawandale, S. R. Ganorkar, and M. T. Kolte, "VHDL based Design of an Efficient Hearing Aid Filter using an Intelligent Variable-Bandwidth-Filter", *International Journal of Advanced Computer Science and Applications*, Vol. 14, No. 1, 2023.

[11] U. S. Rawandale, and M. T. Kolte, "Design, Development and Analysis of Variable Bandwidth Filter Bank for Enhancing the Performance of Hearing Aid System", *International Journal of Intelligent Engineering and Systems*, Vol. 14, No. 6, pp. 391-401, 2021, doi: 10.22266/ijies2021.1231.35.

[12] L. Gerlach, G. P. Vaya, and H. Blume, "A survey on application specific processor architectures for digital hearing aids", *Journal of Signal Processing Systems*, Vol. 94, No. 11, pp. 1293-308, 2022.

[13] S. W. Kim, M. J. Kim, and J. S. Kim, "High-performance DSP platform for digital hearing aid SoC with flexible noise estimation", *IET Circuits, Devices & Systems*, Vol. 13, No. 5, pp. 717-22, 2019.

[14] U. S. Rawandale, and M. T. Kolte, "Study of Audiogram for Speech Processing in Hearing Aid System", In: *Proc. of 2019 IEEE Pune Section International Conference (PuneCon)*, 2019.

[15] G. S. Bhat, N. Shankar, C. K. A. Reddy, and I. M. S. Panahi, "A real-time convolutional neural network based speech enhancement for hearing impaired listeners using smartphone", *IEEE Access*, Vol. 7, pp. 78421-33, 2019.

[16] K. A. Swamy, and Z. C. Alex, "Efficient low delay reconfigurable filter bank using parallel structure for hearing aid applications with IoT", *Personal and Ubiquitous Computing*, pp. 1-4, 2021.

[17] R. Rusinek, and K. Kecik, "Effect of linear electromechanical coupling in nonlinear implanted human middle ear", *Mechanical Systems and Signal Processing*, Vol. 151, p. 107391, 2021.

[18] D. A. Villamizar, D. G. Muratore, J. B. Wieser, and B. Murmann, "An 800 nW switched-capacitor feature extraction filterbank for sound classification", *IEEE Transactions on Circuits and Systems I: Regular Papers*, Vol. 68, No. 4, pp. 1578-88, 2021.

[19] H. C. Stronks, J. J. Briaire, and J. H. M. Frijns, "The Temporal Fine Structure of Background Noise Determines the Benefit of Bimodal Hearing for Recognizing Speech", *Journal of the Association for Research in Otolaryngology*, Vol. 21, No. 6, pp. 527-44, 2020.

[20] P. Kajla, and N. V. George, "Speech quality enhancement using a two channel sparse adaptive filtering approach", *Applied Acoustics*, Vol. 158, p. 107035, 2020.

[21] P. V. P. Sundar, D. Ranjith, T. Karthikeyan, V. V. Kumar, and B. Jeyakumar, "Low power area efficient adaptive FIR filter for hearing aids using distributed arithmetic architecture", *International Journal of Speech Technology*, Vol. 23, No. 2, pp. 287-96, 2020.

[22] V. S. Vivek, S. Vidhya, and P. Madhanmohan, "Acoustic scene classification in hearing aid using deep learning", In: *Proc. of 2020 International Conference on Communication and Signal Processing*, 2020.

[23] S. A. Nossier, M. R. M. Rizk, N. D. Moussa, and S. E. Shehaby, "Enhanced smart hearing aid using deep neural networks", *Alexandria Engineering Journal*, Vol. 58, No. 2, pp. 539-50, 2019.

[24] M. T. Akhtar, F. Albu, and A. Nishihara, "Acoustic feedback cancellation in hearing aids using dual adaptive filtering and gain-controlled probe signal", *Biomedical Signal Processing and Control*, Vol. 52, pp. 1-3, 2019.

[25] A. Adeel, J. Ahmad, H. Larijani, and A. Hussain, "A novel real-time, lightweight chaotic-encryption scheme for next-generation audio-visual hearing aids", *Cognitive Computation*, Vol. 12, No. 3, pp. 589-601, 2020.

[26] K. Zaman, S. S. Maghdid, H. Afridi, S. Ullah, and M. Zohaib, "Enhancement of Speech Signals for Hearing Aid Devices using Digital Signal Processing", In: *Proc. of 2020 4th International Symposium on Multidisciplinary Studies and Innovative Technologies (ISMSIT)*, 2020.

[27] V. V. Mahesh, and T. K. Shahana, "Design and synthesis of filter bank structures based on low order constrained least square and minimum phase methods for audiogram matching in digital hearing aids", *Health and Technology*, Vol. 11, No. 1, pp. 153-68, 2021.
[28] T. Devis and M. Manuel, "A low-complexity 3-level filter bank design for effective restoration of audibility in digital hearing aids", *Biomedical Engineering Letters*, Vol. 10, No. 4, pp. 593-601, 2020.
[29] V. V. Mahesh, and T. K. Shahana, "Constrained least square nonuniform dynamic filter bank for delay and Hardware efficient digital hearing aids", *Health and Technology*, Vol. 9, No. 3, pp. 355-63, 2019.
[30] R. Indrakanti, N. Haridas, and E. Elias, "High performance continuous variable bandwidth digital filter design for hearing aid application", *AEU-International Journal of Electronics and Communications*, Vol. 92, pp. 36-53, 2018.
[31] S. P. Deepu, R. Kini, and S. David, "18 band ANSI S1. 11 filter bank based on interpolated finite impulse response technique for hearing aids", *The Journal of Engineering*, Vol. 2020, No. 9, pp. 760-7, 2020.
[32] T. Devis, and M. Manuel, "Hardware-efficient auto-reconfigurable hearing aids using 3-level octave interpolated filters for auditory compensation applications", *Physical and Engineering Sciences in Medicine*, Vol. 44, No. 3, pp. 785-98, 2021.
[33] H. Park, S. Jung, and H. J. Chung, "An analog correlator based CMOS analog front end with digital gain control circuit for hearing aid devices", *Analog Integrated Circuits and Signal Processing*, Vol. 105, No. 2, pp. 157-65, 2020.