© Universiti Tun Hussein Onn Malaysia Publisher's Office

IJIE

http://penerbit.uthm.edu.my/ojs/index.php/ijie
ISSN : 2229-838X e-ISSN : 2600-7916

The International Journal of Integrated Engineering

Acoustic Feedback Noise Cancellation in Hearing Aids Using Adaptive Filter

D. N. S. Ravi Kumar[1], K. Reddy Madhavi[2], N. Praveen[3*], Hari Hara P. Kumar[4], Ganganagunta Srinivas[3], M. V. Raju[5]

[1]Sathyabama Institute of Science & Technology, Chennai-119, Tamilnadu, INDIA

[2]Sree VidyanikethanEngineering college, Tirupati, INDIA

[3]University of Technology and Applied Sciences, North Al Sharqia region, Sultanate of Oman, OMAN

[4]Sri Satya Sai University of Technology and Medical Sciences, Sehore, Madhya Pradesh, INDIA

[5]Vignan's Foundation for Science Technology and Research Vadlamudi, Guntur, Andhra Pradesh, INDIA

*Corresponding Author

DOI: https://doi.org/10.30880/ijie.2022.14.07.004
Received 00 Month 2000; Accepted 01 Month 2000; Available online 31 December 2022

Abstract: To enhance speech intelligibility for people with hearing loss, hearing aids will amplify speech using gains derived from evidence-based prescriptive methods, in addition to other advanced signal processing mechanisms. While the evidence supports the use of hearing aid signal processing for speech intelligibility, these signal processing adjustments can also be detrimental to hearing aid sound quality, with poor hearing aid sound quality cited as a barrier to device adoption. In general, an uncontrolled environment may contain degradation components like background noise, speech from other speakers etc. along with required speech components. In this paper, we implement adaptive filtering design for acoustic feedback noise cancellation in hearing aids. The adaptive filter architecture has been designed using normalized least mean square algorithm. By using adaptive filters both filter input coefficients are changeable during run-time and reduce noise in hearing aids. The proposed design is implemented in matlab and the simulations shows that the proposed architecture produces good quality of speech, accuracy, maintain stable steady state. The proposed design is validated with parameters like Noise Distortion, Perceptual Evaluation of Speech Quality, Signal to Noise Ratio, and Speech Distortion. The feedback canceller is implemented in MATLAB 9.4 simulink version release name of R2018a is used for validation with Echo Return Loss Enhancement (ERLE). The ERLE of the NMLS is reduced when the filter order is increases. Around 10% of the power spectrum density (PSD) is less when compared with existing designs.

Keywords: Beam forming, speech intelligibility, sound quality, speech enhancement, speech communication

Corresponding author: praveenn@ict.edu.om

1. Introduction

During the last decades, digital hearing aids have replaced their analogue predecessors. Whereas the size of hearing aids decreased with time by using of digital circuits and the capabilities for signal processing has been increased.

Nomenclature is included if necessary
DSP Digital Signal Processing
LMS Least Mean Square
SNR Signal to Noise Ratio
RLS Recursive least Square
BTE Behind the Ear
ITE Inside the Ear
CIC completely in the Channel
BAHA Bone Anchored Hearing Aid
PA Public Address
ERLE Echo Return Loss Enhancement
PSD Power Spectrum Density
SNRI Signal to Noise Ratio Improvement

A wide variety of signal-processing strategies has been developed for hearing aids in order to compensate for different aspects of hearing loss [1-3]. Straight-forward amplification can largely compensate for deficits in the outer and middle ear (conductive hearing loss), but compensation for deficits in the cochlea (hearing loss) requires a more sophisticated approach. Hearing loss causes not only reduced sensitivity for soft sounds, but also a reduced dynamic range as well as a reduced spectral and temporal resolution. The resulting distortions in the perceived sound make it more difficult to understand speech, especially in noisy environments. For a given level of background noise, a listener with hearing loss needs a higher speech level than a normal hearing listener to obtain the same performance, even if the sound is amplified. In noisy situations the hearing aid should therefore amplify speech more than background noise to improve the signal-to-noise ratio (SNR) [4].

Hearing loss can be partially compensated through the use of a hearing aid. It is an electro acoustic device designed to amplify and modulate sound and is shown in Fig.1. Previous hearing aids are based on analogue technology, have large fixed frequency responses while allowing emphasis of high or low frequencies, whose spectrum cannot always match the hearing loss. To overcome the deficiencies of analogue technology, Digital signal processing (DSP) devices would come to offer the best solution for hearing aids [5-9]. In several places, people with a noise exposure above 85 dB limit are requested to wear hearing protectors. In some environments such as air craft's, helicopters, and other industrial work places, to communicate with other worker/person while protecting their hearing, workers need hearing protectors with speech enhancement capabilities.

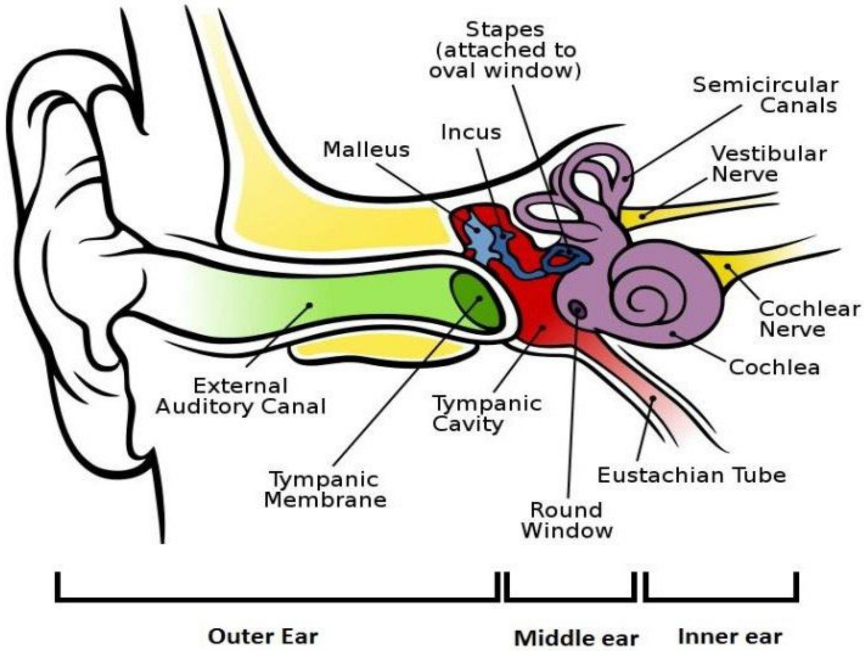


Fig. 1 - Anatomy of hearing aids

2. Literature Survey

In recent years, several hearing aid gadgets have been designed. In paper [5] author proposed decimation filter for hearing aids using basic Distributed Arithmetic (DA) based Finite Impulse Response (FIR) filter which is more effective but the design occupies more area and power con- sumption. In the article [6], Haw-Jing described reusable DA based FIR filter architecture for hearing aids. The design has the memory based filter, so it also occupies more area. In this author used the digital filter and matlab simulink tool is used for the designing of the digital DA based FIR filter for hearing aid application. In this multiplier less DA filter is used with 15MHZ frequency signal as input [7], Feedback cancellation adaptive filter has been present, which employs two types of open loop identifications. In first algorithm second order signal statistics is used for up-dation of weights, where as in the second algorithm higher order statistics is implemented for the updating of the filter coefficients.

Schepker et.al proposed Null-steering beam former-based feedback cancellation for multi-microphone hearing aids with incoming signal preservation. In this he used different beam formers for cancellation of noises occurred in microphones of the hearing aid [8].

In paper [9], the author implemented least mean square (LMS) or recursive least square (RLS) algorithm is used for and it has been shown that limiting estimate will be biased if there exists an error in the model. To overcome this problem, an extra signal is used to the output of the hearing aid to remove the non linearity of the hearing aid. Frequency domain adaptive filtering is used for this architecture.

Tran et.al proposed a switched algorithm based adaptive feedback cancellation using pre-filters in Hearing Aids. In this the author has been pre arranged the filters before the hearing aids to reduce the noises when entering into the double talk detector [10].

2.1. Types of Hearing Aids

There are many types of hearing aids, which vary in power, circuitry and size. Hearing aids are mainly divided into two groups:

1. Implanted hearing aids.
2. External hearing aids.

An overview of different types of hearing aids is shown in Fig.2. External hearing aids are subdivided into two subgroups: Body worn Instruments and Ear worn Instruments. Body worn hearing aids. This was the first type of hear- ing aid consists of a case, an ear mold and attachment wire. The case contains amplifier section, controls and battery. This case is about the size of pack of playing cards and carried on the body or in a pocket. The ear mold contains a miniature loudspeaker. In spite of its size constraints, body worn hearing aid provides large amplification, long battery life. It available for lower prices in the market compared to other aids [11-13]. Ear worn hearing aid is the most common hearing aid, used by the majority of hearing patients. Four types of ear worn hearing aids can be identified. They are:

2.1.1. Behind the Ear (BTE)

This type of aid consists a case behind the pinna, an ear-mold and connection between them. The case contains the controls, battery, electronic equipment, microphones and the loudspeaker. Sound is directed from the hearing aid, through the tubing, and through the ear-mold to the eardrum. The sound from the aid can be routed either acoustically or electrically to the ear. If the sound is routed acoustically, a plastic tube is used to deliver the sound from the loudspeaker to ear-mold, while if the sound is router electrically then the speaker is placed in the ear-mold.

2.1.2. In the Ear (ITE)

This type of aid is smaller than BTE and perfectly fits in the outer ear bowl. The hearing aid case is made out of hard plastic. Due to its size, ITE hearing aid allow for optional manual features such as a volume control, program button, or telephone switch. Feedback is possible in ITE due to closeness of microphone and the receiver. Earwax and moisture are the problems for this type of hearing aids.

2.1.3. In the Channel (ITC)

This type of aid fills only the bottom half of the external ear. It is smaller than the ITE hearing aid but slightly larger than completely in the channel (CIC) hearing aid. It is more discrete than the ITE hearing aid and more suitable for mild to moderately severe hearing loss due to its size. Like ITC, earwax and moisture are the same problems [14, 15].

2.1.4. Completely in the Channel (CIC)

This type of hearing aid is the smallest of custom hearing aid and it is practically invisible to an observer. CIC hearing aid fits deep inside the ear canal. CIC hearing aid is available in analogue and digital technology. Due to the small size, there is no option for directional microphones and volume controllers. These hearing aids are for the people with mild to moderate hearing loss [16, 17].

Implanted hearing aids are in turn sub divided into two sub groups: Destructive and non-destructive. In destructive hearing aids, electrodes are placed inside the cochlea of the patient surgically. Sounds are transmitted to these electrodes across the skin, bone and cartilage by an FM radio signal. This type of treatment is suitable for patients with severe hearing loss [12]. The surgical procedures are irreversible in destructive hearing aid. In non-destructive implanted hearing aids, instruments are relying on conventional bone conduction and direct bone conduction. A conventional bone conduction hearing aid works by conducting, or carrying, sound through the temporal bone. The person hears sound the when the vibrations of the sound are transmitted directly from the vibrating part of the bone conduction hear- ing aid through temporal bone to the cochlea, missing out the outer and middle ears. Such type of arrangement may cause pain, headache, skin irritation and eczema. Bone Anchored Hearing Aid (BAHA) is the developed version of bone conducted hearing aids. In this, skin-penetrating titanium screws are implanted behind the ear and the bone conductor is attached to this titanium screw. User comfort and the fidelity are increased with this BAHA. The surgical procedures are reversible in this type. The other type of non-destructive implanted hearing aid is the middle ear implant. This type of hearing aid converts sound waves into mechanical vibrations. The middle ear implant excites the chain directly via a small exciter [18,19].

The main objective of the paper is to attenuate noise/interference and also enhance the source speech signal in any hands-free communication device (in this report-hearing aid) under various noisy environments. In this paper, the speech enhancement will acquire from normalized adaptive filter. The pro- posed work will compare the different techniques on the basis of parameters: Signal-to-Noise Ratio Speech Distortion and Noise Distortion. This paper will also make the acoustic feedback cancellation in the hands-free equipment.

The rest of this paper organized with Section.2 describes the literature survey of the acoustic feedback cancellation for hearing aid. Section.3 reveals the about the public address system using acoustic feedback cancellation. Section.4 describes the implementation of proposed acoustic feedback cancellation for hearing aids. The matlab results are shown in section.5. Finally section.6 concludes the paper.

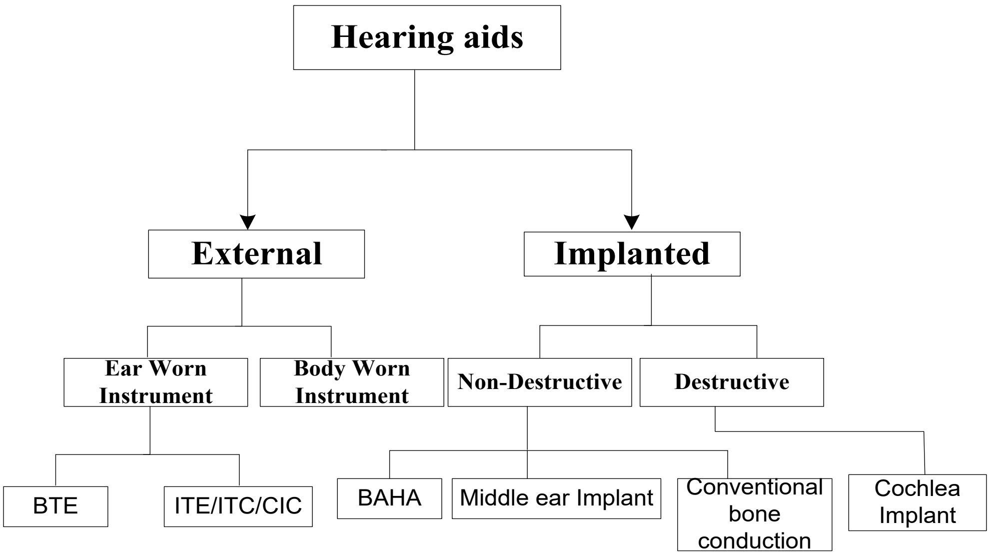


Fig. 2 - Types of hearing aids

3. Public Address System with Acoustic Feedback Cancellation

People amplify their voices in various situations by using public address (PA) systems. In most of the situations, acoustic paths exist between the speaker and the addressing person. Fig.3. shows the acoustic feedback path in PA system. Acoustic feedback is a considerably serious problem in sound amplification systems. It is often referred to as howling, whistling, screeching or squealing. Acoustic feedback may arise either whenever an acoustical, electrical coupling exists between a microphone and a loudspeaker or when the signal in the feedback loop grows underhandedly. Since the squealing is usually very loud, it is unpleasant.

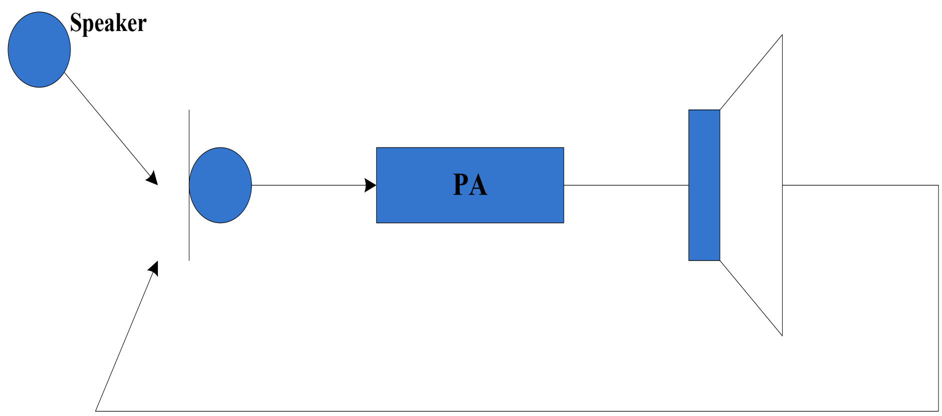


Fig. 3 - Public Address (PA) system with acoustic feedback path

Let us consider with an example that acoustic feedback is a normal problem in hearing aids, because loud speaker and microphone positions are nearer to each other. The portion of the sound coming out of the speaker is taken by the microphone, amplified and then delivered again to speaker. This process continues until the hearing aid goes into audible feedback oscillations. Because of this feedback oscillation, maximum amplification in hearing aid is limited to some extent. This becomes a problem for the hearing aid user who typically needs to maximize the audibility and gain from the hearing aid. To overcome this proposed work will work properly.

The proposed AEC system consists of three important blocks, namely

1. Doubletalk detector
2. Adaptive filter
3. Nonlinear processor
4. Proposed Acoustic Echo Cancellation (AEC)

4.1. Doubletalk Detector (DTD)

In the presence of far-end signal, it is very important to know that the near- end speech signal is exits or not. It is also important to predict when the adaptation of the filter would stop. The situation, where both the far-end signal and near-end signal are present is called as double-talk. In double-talk situation, the error signal has both near-end signal and echo estimation error. While updating the filtering coefficients with this error signal, the final result tends to diverge. Double-talk detector is the solution to overcome this problem. There are several methods of DTD such as Geigel, Benesty and Normalized Cross-Correlation. In this paper, Normalized Cross-Correlation method is used to detect the presence of double talk. This algorithm computes the decision static depending on the relations of microphone signal and error signal.

4.2. Adaptive Filter

It is most important block and it plays a vital role in the acoustic echo cancellation. It estimates the echo path for getting a replica of echo signal.

4.3. Nonlinear Processor (NLP)

It is used for partly or completely cancels the residual signal in the absence of near-end speech signal. Removing of the residual signal will also cancels the any existing acoustic echo. The non-linear processor is a device with a defined suppression threshold level in which signals having a level detected:

1. Below the threshold are suppressed.
2. Above the threshold are passed (although the signal can be distorted).

The non-linear processor functions only during single talk situations. The non-linear processor attenuates the residual echo that could not be cancelled by the adaptive filter. The nonlinear processor (NLP) is required for completely or partly cancels the residual signal in the absence of near-end speech signal. By removing the residual signal will cancel any occurring acoustic echo. The NLP will gradually cancel the signal and insert a form of comfort noise to give the impression to far-end. The NLP as well as the adaptive filter need an accurate estimation from the DTD to operate efficiently.

4.4. Normalized Least Mean Square (NLMS)

Normalized Least Mean Square (NLMS) is actually derived from Least Mean Square (LMS) algorithm. The requirement to derive NLMS algorithm is that the input signal power changes in time, which will affect the convergence rate in LMS algorithm. Small signals will slow down the convergence rate and loud signals will increase the convergence rate. To overcome this, the step size parameter in LMS should be normalized.

In this algorithm the filter output and the error value are computed for each cycle. The error value is obtained by the difference of the desired output and the filter output. Obtained error value is used for updating the filter coefficient. The input equation X (k) is written as:

$$X(k) = \left[x(k), x(k-1).......x(k-K+1)\right]^T \qquad (1)$$

Where *X(k)* is the input signal at time 'k' and 'T' is the transpose of the vector. The output signal *Y(k)* of an adaptive filter can be represent as

$$Y(k) = X^T(k)d(k) \qquad (2)$$

Where d(k) is the filter coefficient vector and it is represented as

$$d(k) = \left[d_0(k), d_1(k), d_2(k), ...d_{(K-1)}(k)\right]^T \qquad (3)$$

The Windrow Hoff LMS algorithm for filter coefficient updating can be represented as:

$$d(k+1) = d(k) + \mu e(k)X(k) \qquad (4)$$

Where *e(k)* is the error signal, μ is the step size parameter which determines the convergence speed and accuracy of the filter. The error signal can be obtained by the difference between desired signal and the output signal. The equation is given as

$$e(k) = D(k) - Y(k) \qquad (5)$$

Where *D (k)* is the desired signal. The input signal of the filter *X (k)* is nourishing into the delay line and then shifted to right for each sampling period. Fig.4 shows the block diagram of the proposed AEC

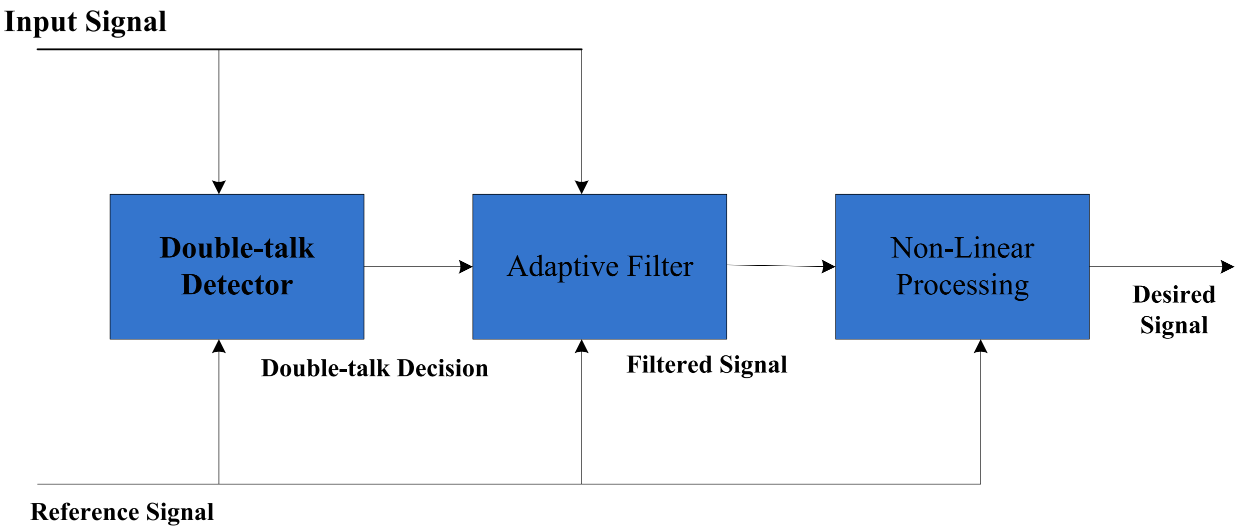


Fig. 4 - Proposed acoustic echo cancellation

When we observe fig.4, the input signal and reference are given to the Double talk detector. In this both signals are compared and the noise signal is passed to the adaptive filter. The adaptive filter will remove the noise and passed towards the non-linear processing. In this any amplification is needed then amplified and finally reach the desired signal.

5. Result Analysis

The implementation of proposed acoustic feedback cancellation in adaptive filters for hearing aids is carried out in MATLAB 9.4 simulink version release name of R2018a is used. Fig.5. shows the matlab simulink diagram of the proposed architecture. The noise double talk decision signal is passed to adaptive filter. The input sweep signal with 4.8k HZ frequency and 16 bit input samples are given to the adaptive filter and the magnitude and frequency response for the adaptive filter is shown in Fig.6. The noise signal in the sweep signal can be removed by using feedback filter. It is important to find whether near-end signals are also present with the far-end signals. This operation can be performed by Double-Talk-Detector (DTD). Once it detects the double talk, it consequently stops the echo cancellation adaptation [14]. Adaptive echo cancellation is used to adapt the echo path impulse response and synthesize the replica echoes and Non-linear processor is used to remove the residual echoes. In this thesis, one far-end signal and one near-end signal are given to feedback cancellation system. Normalized Least Mean Square (NLMS) algorithm is used to cancel the echo signal and gives the desired output signal. PESQ and Echo Return Loss Enhancement (ERLE) are calculated for the output. Based on these values performance of feedback cancellation can be estimated.

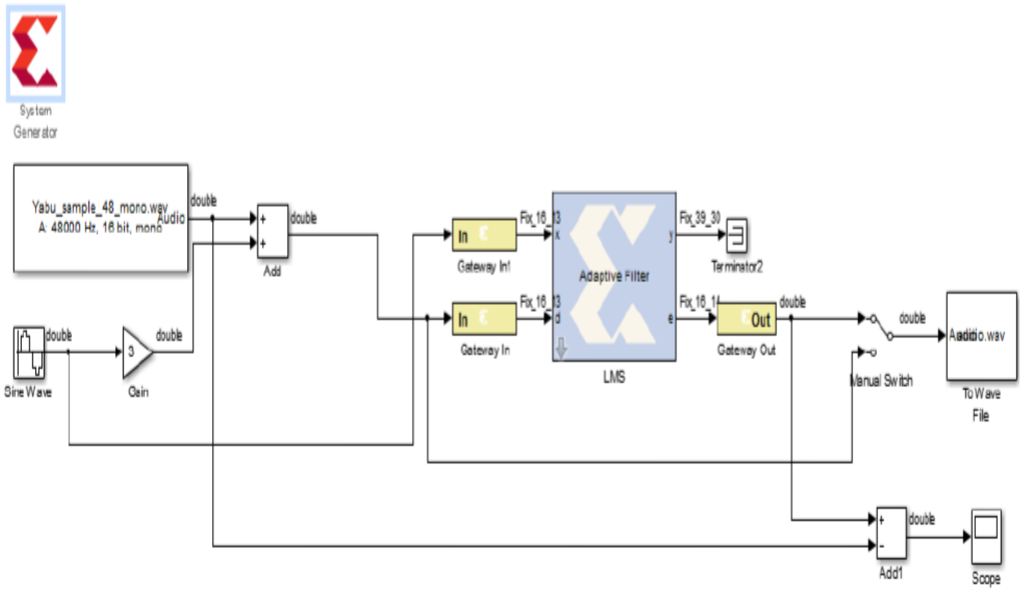


Fig. 5 - Proposed adaptive filter design for echo cancellation for hearing aids

5.1 Test Data

5.1.1 Speech Signals

The speech signals used for this thesis have 16 kHz sampling rate and each speech signal have span of 6-7 seconds. Two male voices and one female voice are used for the test. In male voices, one is used as main speech signal and other is used as interference. The power spectral densities of all the speech signals are shown in fig.7.

5.1.2. Noise Signals

Different noises sampled at 16 kHz are used in this thesis. Noises used are babble noise, wind noise, restaurant noise and white noise. Fig.8. shows the power spectral plots of all noises used. All the results are taken for one speech signal as the source and one noise signal as the disturbing noise. The input SNR value is scaled to different values such as 0dB, 5dB, 10dB, 15dB and 20dB.

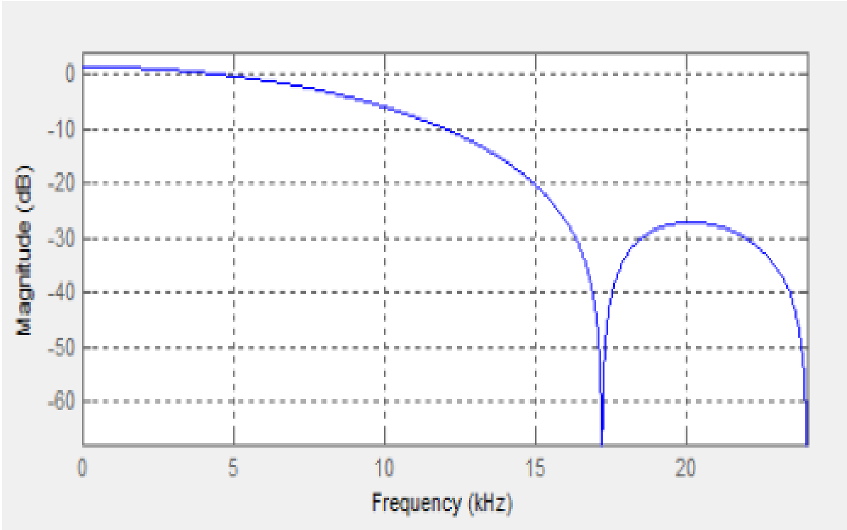


Fig. 6 - Magnitude vs frequency plot

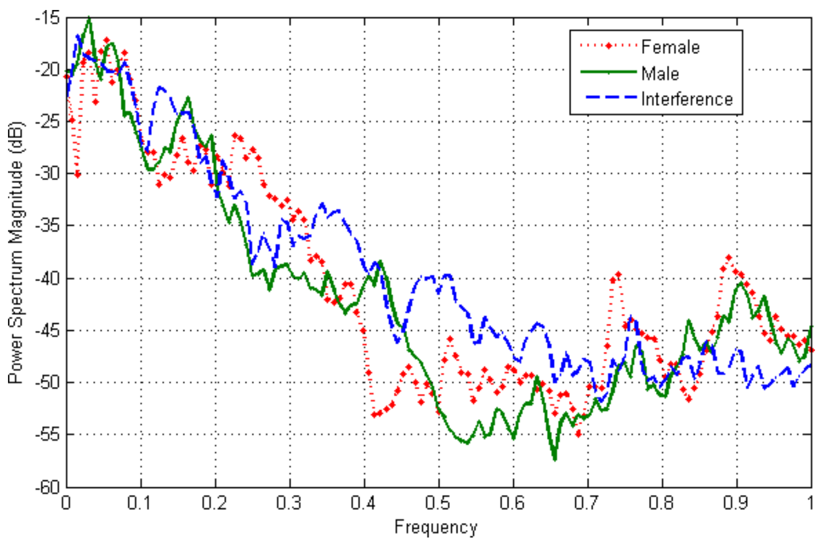


Fig. 7 - PSD plot for different frequency responses

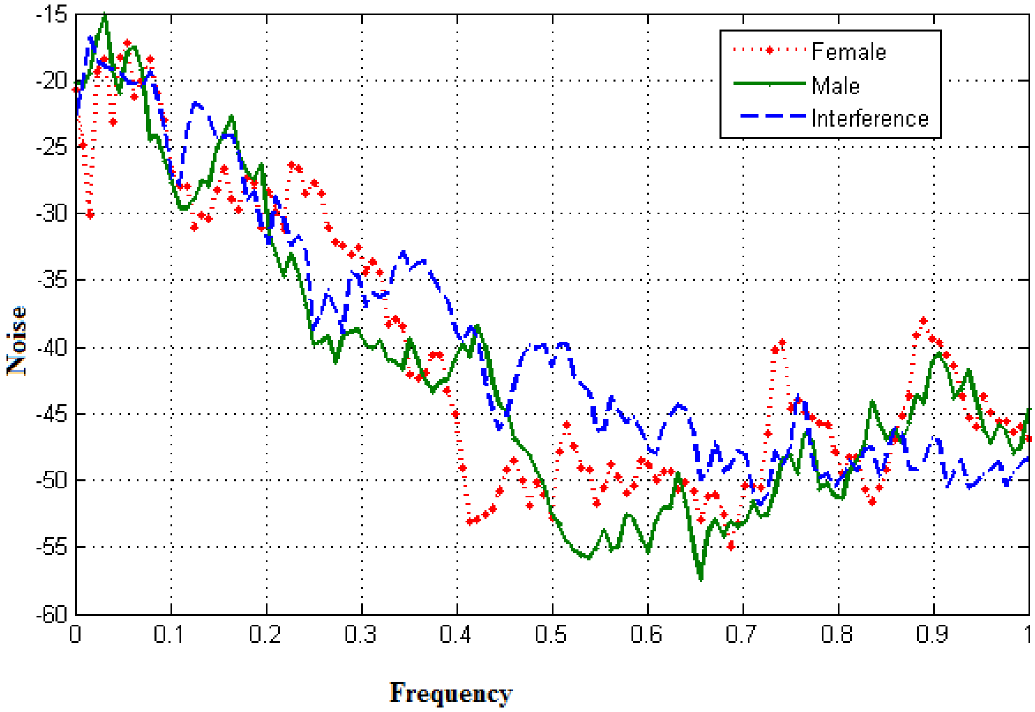


Fig. 8 - PSD plots different noise signals

In this paper, acoustic echo cancellation is also implemented in MATLAB. A simple NLMS algorithm with is used for this AEC purpose. One speech signal and one echo version of corresponding speech signal are taken for AEC system. The echo signal is generated from room impulse response and it is added with random noise. The combined echo signal with noise is used as input signal for AEC system. For simple computations, the NLMS operation is performed for five filter orders. For every filter order the error signal, adaptive filtered output and ERLE are noted. The Table.1. Shows the ERLE values for different filter orders.

The average ERLE for all the orders is around 18.05 dB. The NLMS algorithm gives very small estimated error and also large average ERLE value. So it is one of the best adaptive algorithms recommended for acoustic echo cancellation.

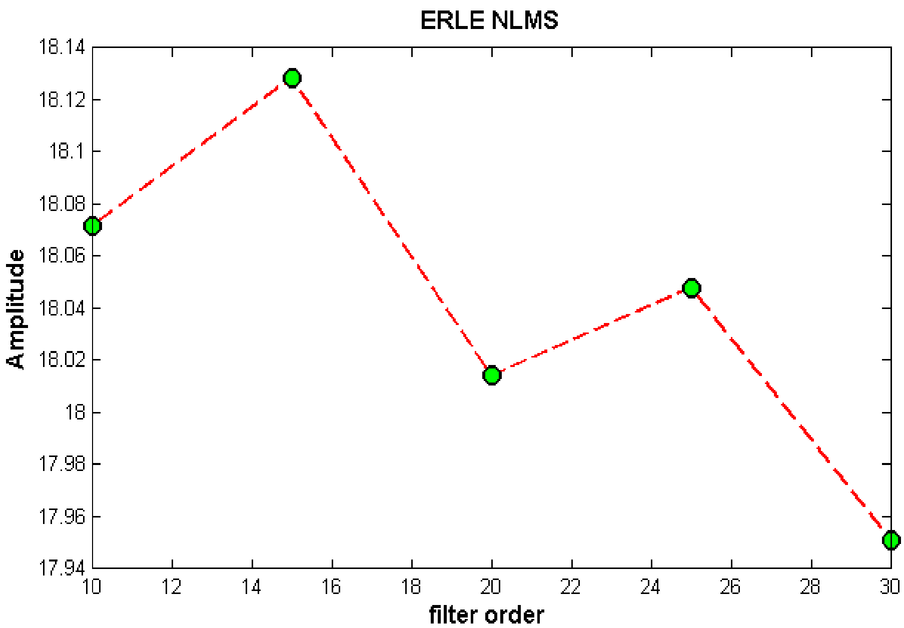
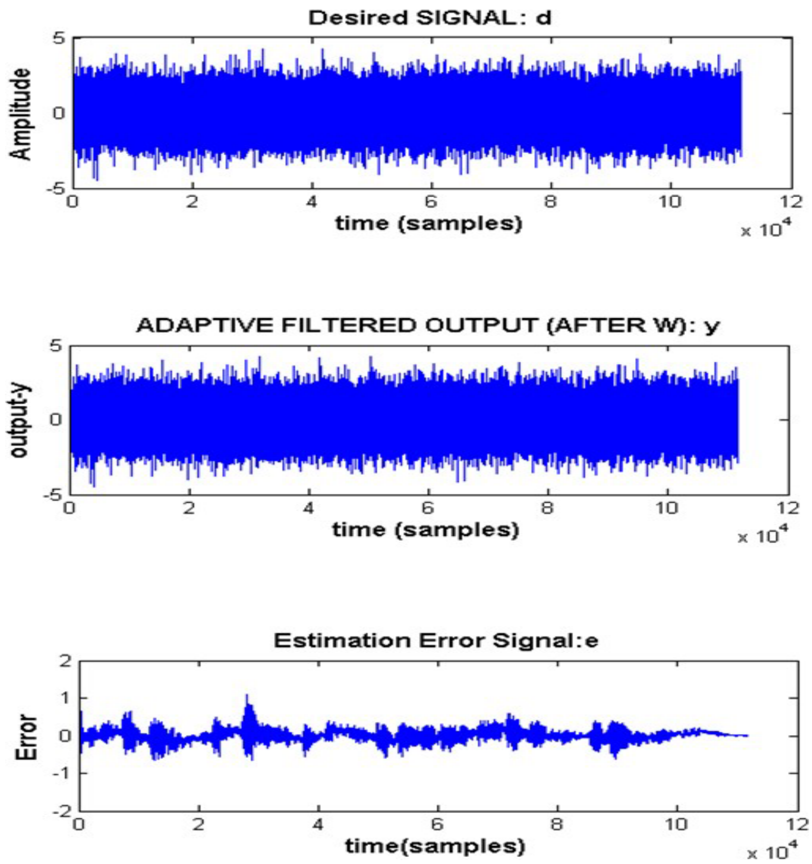


Fig. 9 - ERLE plot for NLMS algorithm

Table 1 - ERLE values for different filter orders

Filter Order	ERLE
10	18.0716
15	18.1286
20	18.0143
25	18.0479
30	17.9513

In general, ERLE of NLMS is larger than the LMS. Fig.9 shows ERLE plot for NLMS algorithm NLMS function used in AEC provides estimated error, adaptive filtered output and filter weights. In order to get estimated error as echoed speech signal, we have to take the desired signal as random noise. The adaptive filter output is the output after the filter coefficients gets multiplied with the input signal. So after multiplying with filter coefficients filter output nearly closed to desired signal. The Fig.10 -shows the signals in NLMS: desired signal, output signal and error signal. By giving random noise as desired signal and echo with random noise as input signal for AEC, we can get error as echo signal. So we can say that NLMS algorithm provides best result for echo cancellation.



6. Conclusion

In this paper the proposed acoustic echo cancellation system will remove the noises occurring in the hearing aids. The simulation results shows excellent performance of the proposed method and it can be promising choice for the practical hearing aids. NLMS algorithm provides ERLE of 18.05 dB and it is large value when compared with LMS adaptive algorithm. Also NLMS algorithm provides very less computational complexity when compared with other existing works. When compared with the existing design around 10% of less power spectrum density is present by using the proposed work. In the future, it would be interesting to investigate the performance of the proposed method for music signals which are considered as a challenge for the present hearing aids. Furthermore, a theoretical analysis of the proposed method is a task for the future work.

References

[1] Stenger, A., Trautmann, L., & Rabenstein, R. (1999, March). Nonlinear acoustic echo cancellation with 2nd order adaptive Volterra filters. In *1999 IEEE International Conference on Acoustics, Speech, and Signal Processing. Proceedings. ICASSP99 (Cat. No. 99CH36258)* (Vol. 2, pp. 877-880). IEEE.

[2] Siqueira, M. G., & Alwan, A. (2000). Steady-state analysis of continuous adaptation in acoustic feedback reduction systems for hearing-aids. *IEEE Transactions on Speech and Audio Processing*, *8*(4), 443-453.

[3] FanChiang, Y., Wei, C. W., Meng, Y. L., Lin, Y. W., Jou, S. J., & Chang, T. S. (2014). Low complexity formant estimation adaptive feedback cancellation for hearing aids using pitch based processing. *IEEE/ACM transactions on audio, speech, and language processing*, *22*(8), 1248-1259.

[4] Siqueira, M. G., & Alwan, A. (2000). Steady-state analysis of continuous adaptation in acoustic feedback reduction systems for hearing-aids. *IEEE Transactions on Speech and Audio Processing*, *8*(4), 443-453.

[5] Teutsch, H., & Elko, G. W. (2001, October). An adaptive close-talking microphone array. In *Proceedings of the 2001 IEEE Workshop on the Applications of Signal Processing to Audio and Acoustics (Cat. No. 01TH8575)* (pp. 163-166). IEEE.

[6] Westerlund, N., Grbic, N., & Dahl, M. (2006). *Subband Adaptive Feedback Control in Hearing Aids with Increased User Comfort*.

[7] Brandstein, M., & Ward, D. (Eds.). (2001). *Microphone arrays: signal processing techniques and applications*. Springer Science & Business Media.

[8] Schepker, H., Nordholm, S. E., Tran, L. T. T., & Doclo, S. (2019). Null-steering beamformer-based feedback cancellation for multi-microphone hearing aids with incoming signal preservation. *IEEE/ACM Transactions on Audio, Speech, and Language Processing*, *27*(4), 679-691.

[9] Yermeche, Z. (2007). *Soft-Constrained Subband Beamforming for Speech Enhancement* (Doctoral dissertation, Blekinge Institute of Technology).

[10] Jyothi, G. N., Sanapala, K., & Vijayalakshmi, A. (2020). ASIC implementation of distributed arithmetic based FIR filter using RNS for high speed DSP systems. *International Journal of Speech Technology*, *23*(2), 259-264.

[11] Tran, L. T. T., & Nordholm, S. E. (2021). A Switched Algorithm for Adaptive Feedback Cancellation Using Pre-Filters in Hearing Aids. *Audiology research*, *11*(3), 389-409.

[12] Arlinger, S. (2003). *Hearing aids for adults: Benefits and costs*. Swedish Council on Health Technology Assessment (SBU).

[13] Välimäki, V., & Laakso, T. I. (2001). Fractional delay filters—design and applications. *Nonuniform Sampling*, 835-895.

[14] Naik, R., Stojcevski, A., Vibhute, V., & Singh, J. (2004, December). Implementation of magnitude estimation algorithm for hearing aid. In *IEEE International Workshop on Biomedical Circuits and Systems, 2004.* (pp. S1-3). IEEE.

[15] Kates, J. M., & Arehart, K. H. (2015). The hearing-aid audio quality index (HAAQI). *IEEE/ACM transactions on audio, speech, and language processing*, *24*(2), 354-365.

[16] Bernardi, G., van Waterschoot, T., Wouters, J., & Moonen, M. (2018). Subjective and objective sound-quality evaluation of adaptive feedback cancellation algorithms. *IEEE/ACM Transactions on Audio, Speech, and Language Processing*, *26*(5), 1010-1024.

[17] Spriet, A., Doclo, S., Moonen, M., & Wouters, J. (2008). Feedback control in hearing aids. In *Springer Handbook of Speech Processing* (pp. 979-1000). Springer, Berlin, Heidelberg.

[18] Drusin, M. A., Lubor, B., Losenegger, T., & Selesnick, S. (2020). Trends in hearing rehabilitation use among vestibular schwannoma patients. *The Laryngoscope*, *130*(6), 1558-1564.

[19] Banakis Hartl, R. M., & Jenkins, H. A. (2020). Implantable Hearing Aids: Where are we in 2020?. *Laryngoscope Investigative Otolaryngology*, *5*(6), 1184-1191.

[20] NagaJyothi, G., & Sridevi, S. (2020). High speed low area OBC DA based decimation filter for hearing aids application. *International Journal of Speech Technology*, *23*(1), 111-121.