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De-noising of Signal using FIR Filters for Hearing Aids

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***Abstract**: In Today's era health related issues are increasing day by day and one of them is hearing loss. Hearing aids have done wonders and helped people in this situation. Hearing aid is a small device that makes the sound loud and clear for the person who is unable to listen, communicate and could not participate actively in day to day activities. A hearing aid can help people to listen in different environments such as loudspeakers in the background, a loud train horn and different noisy situations etc. But the speech signals are degraded and corrupted by different types of noises such as train noise, car engine noise, loud speaker and traffic noise at different SNR levels like -2db, 5db, 10db. These speech signals need to be improved by removing the unwanted disturbances or interference called noise so that enhanced signal can be obtained with good quality for voice communication in hearing aids. This research work proposes the noise reduction technique using FIR filters for hearing aids. The approach of threshold will be applied to remove empty bands or unwanted noise from the noisy signal. The threshold based technique de-noises the input signal efficiently. The proposed approach is implemented in MATLAB and results are analyzed in terms of SNR (signal to noise ratio) and MSE (mean square error).*

***Index Terms**: DWT, FIR Filter, Hearing aids, MSE, Signal processing, SNR.*

I. INTRODUCTION

Due to different kinds of environments the raw input or speech signal has a very high chance of getting contaminated with various kinds of interferences and noises. Noise is defined as an unwanted or undesired signal which gets mixed with the original signal and yields distorted signal. As speech signal gets contaminated due to different types of noises causing difficulties for the listener and degrading the performance in speech processing tasks such as speech recognition, speaker identification, hearing aids many other speech related tasks [1]. The degraded signal needs to be enhanced by removing the additive noise to improve its quality so that clear and clean de-noised signal can be used in the hearing aids.

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Interferences are added during the acquisition of the signal and its transmissions. Additive noises can be power line transmission noise that is noise due to power supply, explosion noise, thermal noise, environmental noise, background noise , noise generated due to machines etc. [2]. Hence the de-noising of the signal becomes very important for an effective speech communication analysis for hearing aids. Many techniques have been used for this purpose. A hearing aid only amplifies the input signal which allows the person to listen voice or speech but that signal is not noise-free signal. So to remove the noise from the signal Digital FIR filters are used in this research. Hearing loss can be different for every person at different level as shown in the given Table I [3]. To improve the signal quality different approaches have been performed by many researchers. Our approach uses DWT [4] (Discrete wavelet Transform) method which uses a small wave, where energy is concentrated at a specific point and it transform means representing signals into a different form, then the signal is decomposed using EMD [5] method which helps in extracting functions called IMFs [6] (intrinsic mode functions) from the signal. A signal is uncorrelated into number of IMFs but these contain noise. In the single IMF various modes of oscillations get mixed so another efficient de-noising method is needed. Sometimes EEMD is applied called Ensemble EMD which takes mean over wide range of IMFs value and then added to the white noise, the process is repeated for several cycles , each time EMD is applied until average IMF is yielded [7]. At other times higher order IIR filters or FIR filters are taken into consideration. A new de-noising signal method is implemented which uses two approaches together, improved EMD and the DWT based thresholding technique [4]. When the signal gets uncorrelated and decomposed into numbers of IMFs then IMFs can be grouped into two categories: - 1) High-frequency group 2) Low frequency group. The first few IMF contains high frequency artifacts which contributes to major section to noise and results in the loss of large part of information and hence their de-noising is attained by first un-correlating them, applying wavelet operation and then de-noising them. Noise cancellation model is used for de-noising of these IMFs or window technique is applied to remove the noise part. These obtained de-noised IMFs are added to those previous remaining noise free IMFs which reconstructs the noise free signal. Mean square error (MSE) and the signal to noise ratio (SNR) are taken as the performance parameters to analyze the performance of proposed technique with the existing.

In Fig.1 the block diagram of hearing aid is shown.

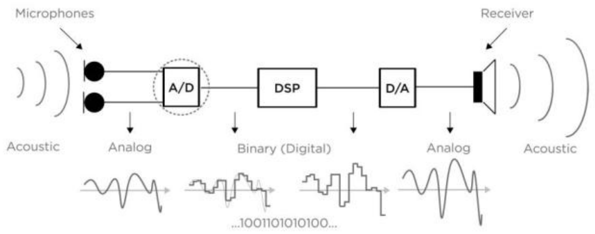

Fig. 1 Hearing aid block diagram.

Table. I Different Levels of Hearing Loss in humans [3]

Level	Types of hearing loss	Lower limit	Upper limit
1	Normal	-10 dB	26 dB
2	Mild	27 dB	40 dB
3	Moderate	40 dB	70 dB
4	Severe	70 dB	90 dB

This research paper is based on threshold based signal filtering technique. In the section number two, the literature survey is presented on the signal filtering technique. In the section number three, the research methodology is shown and in the section number four result and discussion is presented in this paper.

II. LITERATURE REVIEW

Raghu Indrakantia, et.al reviewed a less complicated, less stoppage recomposing digital filter framework for hearing aids. The major purpose of this study was the matching of a specified human audiogram with least amount of faults [8]. The elevated estimation difficulty in the traditional standardized and non-standardized filter bank methods shifted the intend model towards changeable filter arrangement. A Farrow arrangement relied devise was utilized for realizing the subjective sample rate converter. This approach used a differentiator based mock-up for the variation of bandwidth in continuous manner. The projected framework also considered the significance of energy, region, and impediment in such kind of dangerous situation. The comprehensive comparisons were carried out with the accessible literature on audiogram matching. The tested results demonstrated that the proposed structure was very spirited in comparison with several other approaches. The proposed design showed good performance not only in error matching but also with respect to region, energy distribution and hardware intricacy. Qiuju Li, et.al proposed a new de-noising approach. This approach introduced a guided filter shape image processing domain for the de-noising of audio signal [9]. The selection of a guided filter was fundamental. In this study, a wavelet de-noising filter was utilized for the attainment of direction signal. The attainment of authentic signal was found complicated for audio signal. This affected the recognition of the optimal wavelet de-noising outcome. It was identified that the attainment of direction signal proved very difficult for researchers. It was also recognized that Guided filter with enhanced direction signal were not only better in de-noising but these filters also preserved the pointed fraction of the audio signal. Different investigations demonstrated that the new de-noising approach, which utilized wavelet de-noising in the initial phase with the new strengthened evaluation technique for getting the improved direction signal and also used guided filter for the de-noising of audio signal, was extremely effective audio signal de-noising technique. Jianming Yu, et.al suggested the use of LMD (Local mean Decomposition) and EMD (Empirical Mode Decomposition) method for the de-noising of rational tremor indication [10]. In this study, the obtained outcomes were investigated and compared with the de-noising presentations of the two techniques. The investigational outcomes demonstrated that both approaches had the abilities of signal de-noising, self-reliance etc. These approaches also improved the eminence of signals with noise concurrently. The performance of these two algorithms was evaluated through correlation coefficient (NC) and signal to noise Ratio (SNR). Equivalent information indicated that mechanism attained in the putrefaction of the seismic indication utilizing LMD showed better correlation level in comparison with EMD. In the meantime, the filtered indication which possessed superior SNR value, the performance of LMD was somewhat better than that of the conventional EMD by means of de-noising for seismic indications. Salim Lahmiri, et.al presented a new method for physiological signal de-noising based on the vibrational mode decomposition (VMD), the discrete wavelet transform (DWT), and constrained least squares (CLS) optimization [11]. This approach was compared to others based on empirical mode decomposition (EMD) and DWT thresholding of the obtained intrinsic mode functions (IMFs) and residue, followed by the un-weighted summation of the results. The comparisons were performed with two EEG signals from the left and right cortex of a rat, and one ECG signal from a human subject. Using the signal-to-noise ratio and mean squared error as performance metrics, the results show strong evidence of the superiority of the VMD-DWT-CLS approach over the standard EMD-DWT. It was concluded that using CLS in the final reconstruction stage and ignoring the residue may bring significant improvement to the de-noising process. Mengjiao Wang, et.al stated a new approach to de-noise chaotic signals based on ensemble empirical mode decomposition (EEMD). The EEMD technique was first used to decompose the noisy chaotic signal into the so-called intrinsic mode functions (IMFs) [12]. A criterion was proposed to determine which modes were used to reconstruct the de-noised signal. Computer simulations were used to demonstrate the effect of the method. The results were compared with the signal-filtering approach based on empirical mode decomposition (EMD-based). It was found that the method proposed in paper performs better than the EMD-based approach. Kai Liu, et.al used wavelet packet transform to de-noise the torpedo magnetic fuse signal, and the waveform close to the real signal was obtained [13]. Both theoretical and simulation results show that the WPT de-noising method can effectively eliminate background noise existed in torpedo magnetic fuse signal, and has the following advantages: high precision, no distortion, retain the original characteristics of the signal, etc., and can be used to extract weak signal in the background of strong noise, so it is an ideal means of magnetic fuse signal de-noising.

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In the process of de-noising, how to select the wavelet packet basis and re-adjust the threshold is the key to the de-noising quality good or bad, and should combined with the characteristics of the signal to select the wavelet packet basis, the threshold and the layer of wavelet packet decomposition, etc. Jose L. San Emeterio et.al studied about both synthetic and experimental ultrasonic and A-scans was used. Synthetic ultrasonic traces have been generated using an approximate speckle model which includes frequency dependent attenuation and scattering [14]. Ultrasonic signals acquired from a test block made of austenitic steel have also been de-noised. Results obtained using a Cycle-Spinning (CS) implementation of the stationary wavelet transform are compared with those obtained using with the Discrete Wavelet Transform (DWT), using soft thresholding and two decomposition level dependent threshold selection rules (Universal and SURE). It was shown that both DWT and CS de-noising procedures yield very bad results when using Universal thresholds. It was also noticed that CS de-noising using SURE thresholds was an effective approach to de-noise ultrasonic signals with low initial SNR, providing very good results for both synthetic and experimental A-scans.

III. RESEARCH METHODOLOGY

This research work is related to the de-noising of the analog signal using signal processing techniques. The proposed methodology gives enhanced quality of the analog signal and effective signal de-noising using the FIR filter. As the signal de-noising is done using thresholding technique that is defining the threshold value for empty bands and removing them using the different steps. The techniques used for de-noising are explained step by step that include signal segmentation, IMF reconstruction, derivative, DWT (discrete wavelet transform), Transformation and the final output we analyze is the de-noised signal. The final output received is the ECG de-noised signal for the proposed method can be measured in the terms of SNR (signal to noise ratio) and MSE (mean square error) in the performance metrics in comparison of previous approach. The below given steps describes the signal processing techniques used for de-noising in the five steps:

A. EMD or Signal Decomposition

In the very first step, the original input signal as shown in the Fig. 2 is decomposed into IMF (intrinsic mode function) as shown in the Fig. 3. IMF is basically the sum of original signal which have two properties, firstly the number of extrema and the number of zero crossings are equal or they might differ by one and second one the upper and the lower envelope are symmetric. This EMD (empirical mode decomposition) is decomposing of the signal into AM/FM modulated components by the sifting algorithm and these components are IMFs.

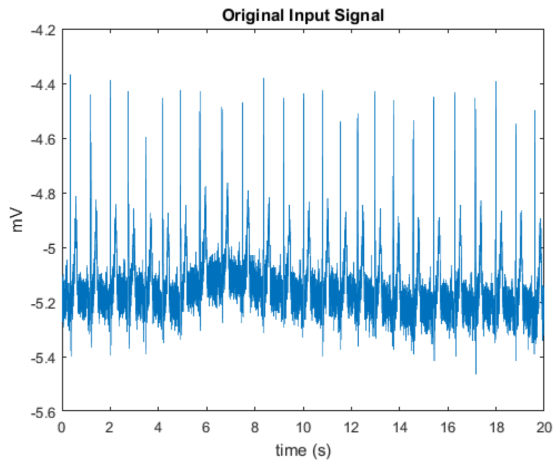

Fig. 2 Original noisy signal taken as input signal

It aims at presenting the signal via number of IMFs and the residual. These components called IMFs gives the oscillations present in the signal or data. Once the decomposition is done, the original signal can be written as:

$$x(t)=\sum_{i=1}^{L}\varphi_i(t)+r(t) \qquad (1)$$

where L is the total number of extracted IMFs.

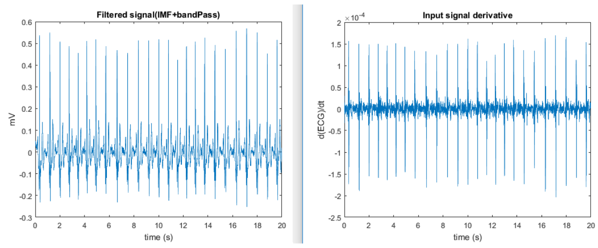

Fig. 3 Decomposition of original input signal into IMFs.

B. IMF Reconstruction

The second step is reconstruction or combining the IMFs as shown in the Fig. 4 in which it selects only those IMFs who have maximum fluctuations. After reconstructing them, it allows them to pass through the band pass filter. In this research work, the Butterworth FIR filter is used for the reconstruction of signal and it's de-noising. It is the first operation performed by system; it is applied to reduce the processing time. We consider that IMFs 4, 5 and 6 possess important information in this research. These filters are considered easy and steady. The window technique is the easiest form of FIR filter intends technique. In this technique, all frequencies less than cut off frequency are transferred with accord amplitude while other frequencies are not permitted. The various windows utilized in this approach are Rectangular Window, Hanning window, Hamming window, and Blackman window. High pass filter and Low pass filters are structured with the help of these windows. These windows provide cut off frequency of 3Hz and 100Hz correspondingly.

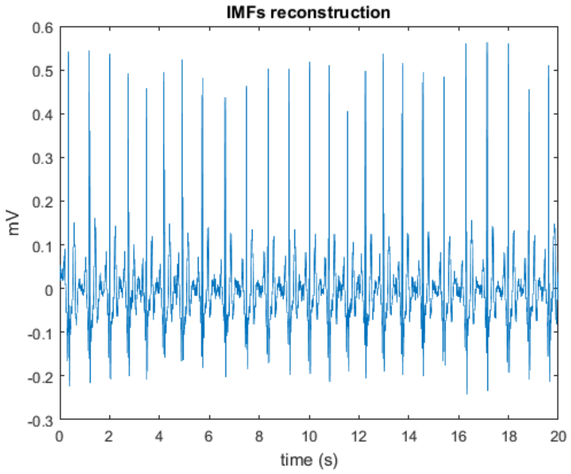

Fig. 4 IMFs reconstruction of the signal

C. Derivative of the signal

The third step is derivative part. There is a correlation matrix present, from which we select a region, there are number of points in that region out of which one point will correlates with all other points. The point which will not correlate or will not match with the point matching all other region is recognized as noise or unwanted disturbance. Thus the noise will recognized by derivative. After this, the noisy ECG signal travels by these band-pass filters for the removal of unwanted noise. These kinds of filters are used for the minimization of mean squared error with the main input signal (speech signal) and the orientation input (noise with ECG signal). The below given Fig. 5 shows the filtered signal and derivative of input signal.

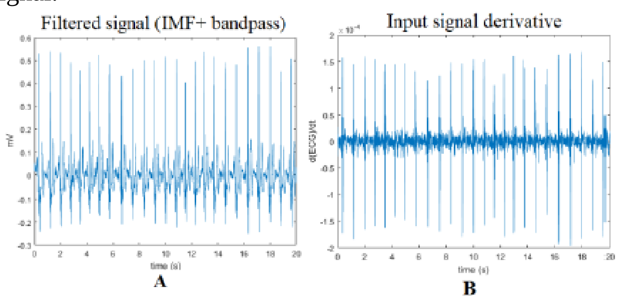

Fig.5 (A) Filtered signal (IMF +band pass). (B) Input signal derivative.

D. DWT (Discrete wavelet Transform)

The next is fourth step in research work that is discrete wavelet transform based de-noising method. In this technique the signal is decomposed at a level where thresholding value is applied to extract the wavelet coefficients. It is used to define threshold value of the empty band which we need to remove. It is known as thresholding technique which is used in FIR filter for de-noising the noisy input signal. The high order frequency noise is removed from IMFs components and it allows only low order frequencies IMFs components. The signal obtained from the input signal can be de-noised with the help of adaptive noise canceller. Adaptive filter is used in this noise canceller which is a linear filter and it cancels the noise. It is known as adaptive because it adapts its parameters according to the algorithm used in it. The Recursive least squares algorithm is utilized in this filter. This approach is considered very effective for de-noising. The signal which is analog has noise or contain empty band which affect quality of the signal. The threshold based technique is applied which can remove empty band from the signal. The below given Fig.6 shows the decomposed input signal with the original signal along with the level-2 and level-3 wavelet coefficients.

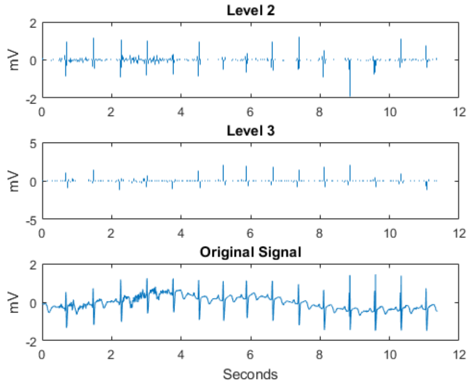

Fig. 6 Decomposed input signal with the original signal along with the level-2 and level-3 wavelet coefficients.

As wavelet coefficients are extracted after threshold value is applied to choose whether the frequency is to be passed or not.

E. Transformation

In this last step the transformed signal is obtained as shown in the Fig. 7, to get the final output as the signal will be reconstructed by combining those IMFs from which noise is been removed and we obtain the de-noised signal as final output using thresholding technique. When the empty bands or noise get removed from the signal it leads to increase the quality of the signal which can be used for voice communication. The threshold based technique is proposed in this research work for the signal de-noising. In the threshold based technique, the threshold value is calculated in the signal based on the signal frequency. The frequencies in the signal which is above the threshold based is considered as noisy frequencies and other are considered as non-noisy signals. Below is the flowchart for the proposed methodology explained step by step in Fig.8 showing the de-noising of input signal of this research work.

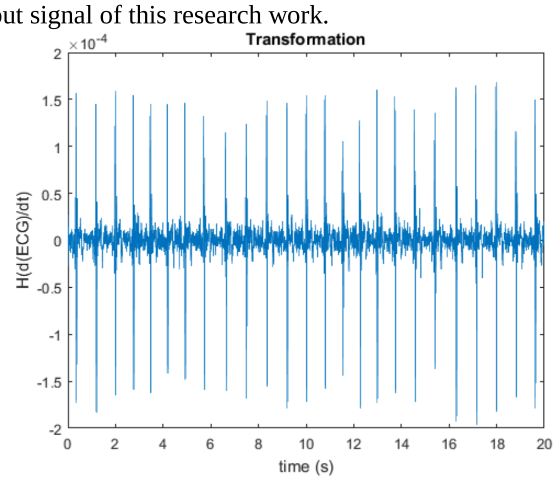

Fig. 7 Transformation of the signal

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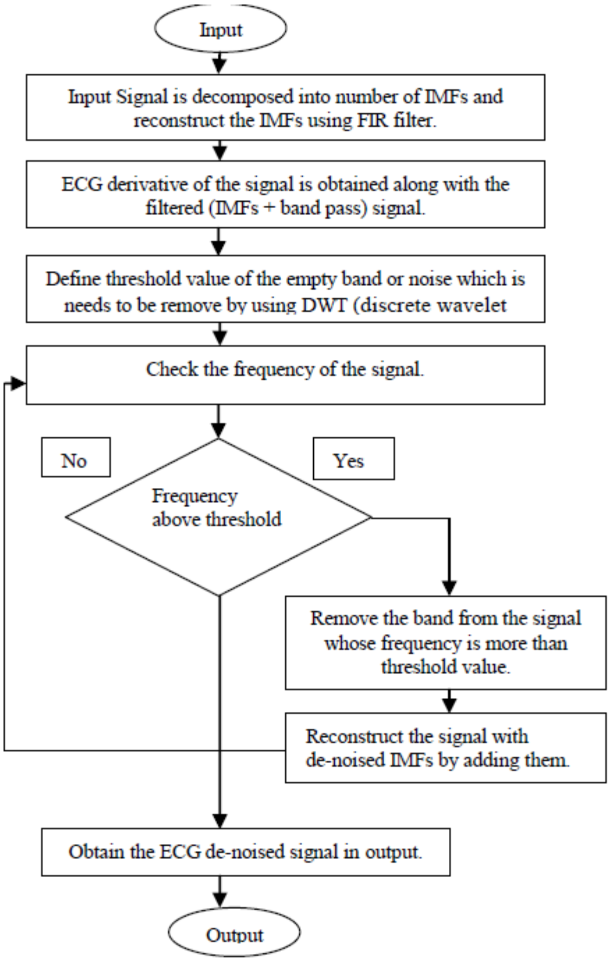

Fig. 8 Flowchart for proposed methodology

IV. RESULTS AND DISCUSSION

To de-noise the noisy signal the proposed methodology is implemented in MATLAB. The Final output gives the de-noised signal and results are analyzed in terms of SNR (signal to noise ratio) and MSE (mean square error). The de-noised output signal is shown in Fig. 9, the technique of adaptive filter is applied which is a linear filter used to cancel the noise and finally the signal is obtained that is the de-noised signal which is used for hearing aids in speech communication.

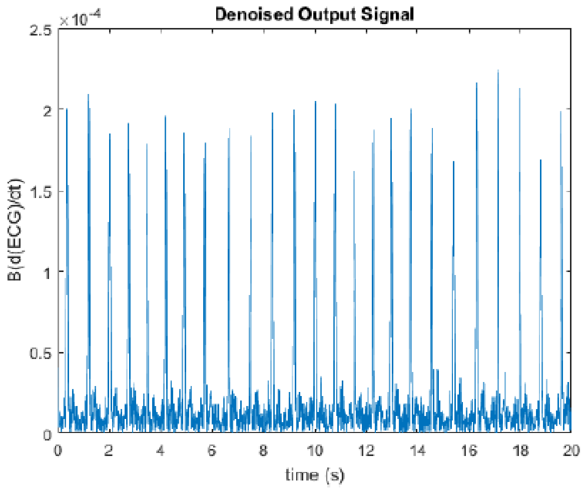

Fig. 9 Final Output De-noised output signal

The graph shown in Fig. 10, with the increasing SNR (signal to noise ratio) value the RMSE value is reducing in the proposed technique which shows efficiency of the implemented method. It is analyzed that proposed technique has improved SNR over the different signal interval. The mean square error value is reduced

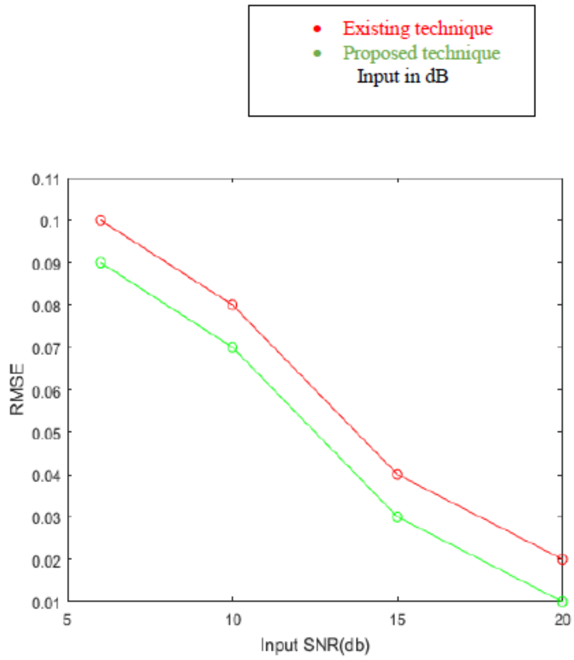

Fig. 10 Increase value of SNR with respect to RMSE

Table. II RMSE analysis with respect to SNR

Input SNR in dB	Existing technique RMSE value	Proposed technique RMSE value
6	0.1	0.09
10	0.08	0.07
15	0.04	0.03
20	0.02	0.01

The above given Table. II explain the values plotted for the graph that shown in the Fig. 10 that the SNR level increases with the reducing value of RMSE.

In the Fig. 11, the mean square error (MSE) of existing and proposed technique is compared for the performance analysis. It is analyzed that proposed technique has less MSE over the different signal interval. The overall MSE value achieved by this method decreases by 5% and thus improving the efficiency.

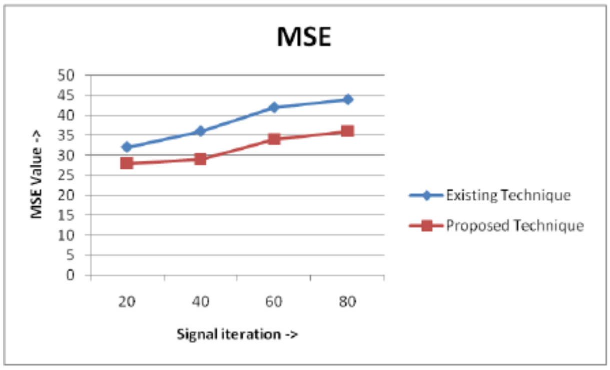

Fig. 11 MSE Analysis

Table. III MSE analysis

Signal iteration	Existing technique (MSE)	Proposed technique (MSE)
20	32	28
40	36	29
60	42	34
80	44	36

The above given Table. III shows the comparison of MSE analysis of the existing and proposed technique. Thus the overall performance is improved and more efficient.

V. CONCLUSION

Due to the addition of unwanted noises such as electronic noise in the form of thermal noise and explosion noise, the audio noise originating from touching, vibration or crashing origin like rotating equipment's, steering automobiles, keyboard clack and other surrounding noises including loud sirens, noise produced by machines, whistling noise, motor bikes, loud volumes from music concerts etc. which creates difficulty for the person using hearing aids to listen and communicate. So it is necessary to remove these noises from the signals which can be done using adaptive noise canceller as implemented in this research. The various types of filtering techniques are designed so far to remove noise from the signal. In this research work, a new de-noising signal method is implemented which uses two approaches together, improved EMD and the DWT based thresholding technique. The de-noising of signal is achieved using digital FIR filter. The noisy signal is decomposed into number of IMFs and those IMFs components are reconstructed using band-pass filters. The DWT (discrete wavelet transform) method in which threshold will be applied to remove empty bands or noise from the signal and the final output is obtained by summing up de-noised IMFs. The proposed technique is implemented in MATLAB which gives the de-noised signal that has about 5 percent less MSE (mean square error) with respect to increasing level of SNR (signal to noise ratio) as compared to existing technique which is useful for the hearing aids.

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