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# Denoising real-time audio signals using matlab filter techniques

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## Abstract

*In this paper,denoising of real time audio signals is carried out. Our main aim was toreduce the noise associated with the audiosignal. It is very important to reducenoise being present in any signalas it reduces its quality. For analysis purpose, white Gaussian noiseis concatenatedwith the audio signal and the resulting noisy signal is subjected to the different filtering techniques like Infinite Impulse Response (IIR) Filter, Finite Impulse Response (FIR) Filter, Wavelet transform techniques. A comparative analysis is carried out with different filters, in order to find the best filter suitable for denoising the audio signal.*
***Keywords**— Denoising,FIR filters, IIR filters, Signal processing, Wavelets.*

## 1.INTRODUCTION

In communication, reduction of noise plays the major part in real-time audio signals. Presence of noise makes the signal weak and the recognition of audio signals becomes difficult. Therefore, a well-established method to reduce noise is to filter the signal in the frequency domain and analysing the signal using different filter techniques like low pass, high pass and bandpass filters. When the signals are not periodic, these filters distort the signal than reducing the noise Therefore, to attenuate noise from the signal we require more advanced filtering techniques like Fourier transforms, wavelet transforms etc. Fourier Transform of a signal gives us the frequency composition of the audio signal. The drawback of Fourier Transform is that, it is only valid within a certain Region of Convergence (ROC). Hence, we go for Short-Time Fourier Transform (STFT), which uses the window analysis approach of defined size. IIR and FIR algorithms uses the Fast Fourier transform (FFT) technique for analysing the frequency spectra and signal responses. Wavelet analysis provides more detailed analysis about the signal in comparison with other filtering approaches.

Audio signals are influenced by various types of real-time noises such as electrostatic noise, thermal noise, channel noise. The work in this paper will address the drawbacks ofvarious other filtering techniques and will help the reader to expand his knowledge in this domain. The project uses real time audio signals to establish a fine comparison among three different filtering techniques i.e. IIR filtering, FIR filtering and wavelets transforms.

## 2. BACKGROUND AND RELATED WORK

Graps[1] came up with new analysis named Fourier transform which could analyze the periodic function by creating mathematical structures that vary in scale. But the proper analysis cannot be carried out using frequency response.

Radhika Bhagat[2] has made an attempt of audio filtering using extended filters like FIR and IIR filtering techniques. They have designed different formulas and difference equations for efficient implementation of time varying filter applications.

Manoj Singla[4]has used Butterworth filter and Chebyshev filters to reduce noise from signals with different frequency and ripple factors.

Seema rani[5] showed that FIR is more stable at higher order than IIR. The error of FIR filter is less compared to IIR filters that means the output of FIR filter is very close to the desired value.

C Mohan Rao[6] has presented a new algorithmnamed Least Mean Square (LMS). In this method,additive white Gaussian noise is added to audiosignal. Later, denoising is done in order toreduce the noise with minimum or no error efficiently.

B. JaiShankar[7] has proposed the use wavelet transformation technique to denoise audio signals by dividing the signal into blocks. This technique protects every unique and vital features of every individual block and exposes the finest detail contributed by the grouped set of blocks.

The authors in [8]have proposed Discrete Wavelet Transform (DWT). They have discussed its key feature which is the temporal resolution. It captures both frequency and location information. This feature becomes one of the main advantages of DWT over Fourier transforms.

Priya Khattar[9],in their paper have carried out denoisingwavelet transformation by comparing two wavelets families, Daubechies and Haar.

## 3. PROPOSED WORK

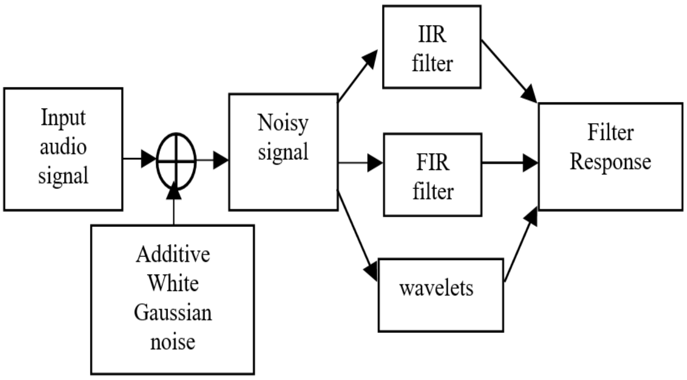


**Figure.1 Implementation flow of proposed work**

Using basics of noise theory and audio theory, analysis of the audio signal is carried out. The audio signal under consideration is concatenated with additive white gaussian noise. Using FFT, this noisy signal is converted from time domain to frequency domain. Using frequency domain representation of the signal, different filter parameters like order of the system, cutoff frequency are calculated. The filtering approach consists of normalization of signal, decomposition technique and reconstruction technique. Further the IIR, FIR and wavelet filter implementation is described.

### 3.1 IIR Filter Algorithm

The audio signal is taken as input, to which additive white Gaussian noise is added, resulting in noisy signal. White Gaussian noise is preferred as it has constant Power Spectral Density (PSD) and it is easy for analysis of audio signal.

Now the frequency domain plot of the noisy signal is obtained by Fast Fourier Transform. Further, the signal is analyzed to decide the cut-off frequency. Once the cut-off frequency is determined, the filter design process is initiated by passing the essential parameters and low pass filter is built.

The filter coefficients are obtained and the magnitude plot of the filter is plotted and analyzed. Now that the filter is designed, the normalized noisy signal is passed to low pass filter to obtain the filtered signal. The filtered signal is finally plotted and used for analysis.

### 3.2 Equations

Difference Equation of IIR Filter:

$$y(n) = -\sum_{k=1}^{N} a_k (n-k) + \sum_{k=1}^{M} b_k x(n-k) \qquad (1)$$

Wherex(n) is the input signal, y(n) is the output signal with filter coefficients a and b.

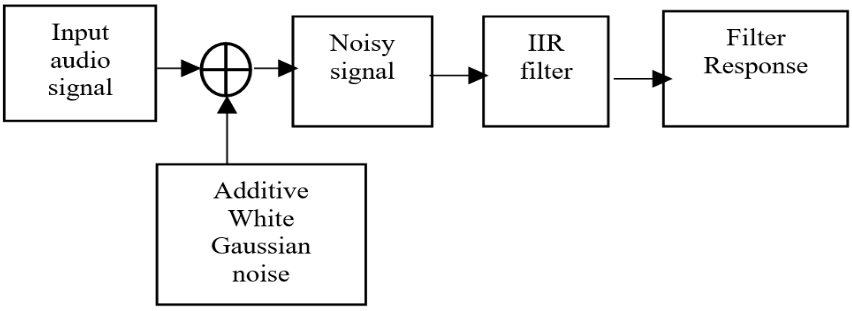


**Figure 2. Block diagram of IIR filtering approach**

### 3.3 FIR Filter Algorithm

In this algorithm, the audio signal is taken as input, to which additive white Gaussian noise is added, resulting in noisy audio signal.

Now the frequency domain plot of the noisy signal is obtained by Fast Fourier method. Further, the signal is analyzed with respect to the peak points to decide the cut-off frequency. Once the cut-off frequency is determined, the filter design process is initiated by selecting the essential parameters like the order, normalized cut-off frequency. Filter type (low pass filter) and appropriatewindowing technique is also selected. The filter coefficients are obtained,and the magnitude plot of the filter is plotted. Now that the filter is designed, the normalized noisy signal is passed to low pass filter to obtain the filtered signal. The filtered signal is finally plotted and used for analysis.

### 3.4 Equations

Difference Equation of FIR filter:

$$y(n) = \sum_{k=0}^{N-1} h(k)x(n-k) \qquad (2)$$

Where y(n)is output signal, h(k) is the unit impulse response, x(n) is input signaland k=0…N-1 where lower and upper limits on convolution sum reflects causality and finite duration characteristics of the filter.

Hamming Window function:

$$w(n) = 0.54 - 0.46\cos\left\{\frac{2\pi n}{N-1}\right\} 0 \le n \le N-1 \qquad (3)$$

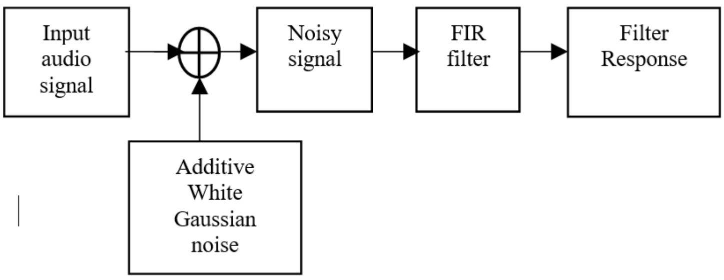


**Figure 3. Block diagram of FIR filtering approach**

### 3.5 Wavelets

Wavelet transform method can also be used for analysis of the audio signal.Wavelet transformscan be classified as continuous wavelet transform and discrete wavelet transform. We are using discrete wavelet transform technique as it is more suitable for denoising of audio signals. The audio signal which is to be analyzed is taken as input, to this additive white Gaussian noise is added and the noisy audio signal is obtained. This signal is passed throughLevel filter.

The noisy signal is decomposed into two parts, detailed coefficients and approximation coefficients.The number of levels required for decomposition generally depends upon nature of the signal.

Multilevel decomposition is done to repeat the process of decomposition so that many lower resolution components of the signal can be obtained through wavelet decomposition trees. At last, Wavelet thresholding is used to reconstruct the original audio signal without much loss of information. Therefore,construction process involves the wavelet coefficients and the levels of iterations which are required for the successful reconstruction of the original audio signal.

DWT has two functions wavelet and scaling function

Scaling function $\emptyset(t) = \sum_{n=0}^{N-1} h[n]\sqrt{2}\emptyset(2t-n)$ (4)

Wavelet function $\varphi(t) = \sum_{n=0}^{N-1} g[n]\sqrt{2}\emptyset(2t-n)$ (5)

Approximation coefficients:

$$W_{\emptyset}[j_0, k] = \frac{1}{\sqrt{M}}\sum_n f[n]\emptyset_{j_0,k}[n] \qquad (6)$$

Detailed coefficients:

$$W_{\varphi}[j,k] = \frac{1}{\sqrt{M}}\sum_n f[n]\varphi_{j,k}[n] \qquad j \geq j_0 \ (7)$$

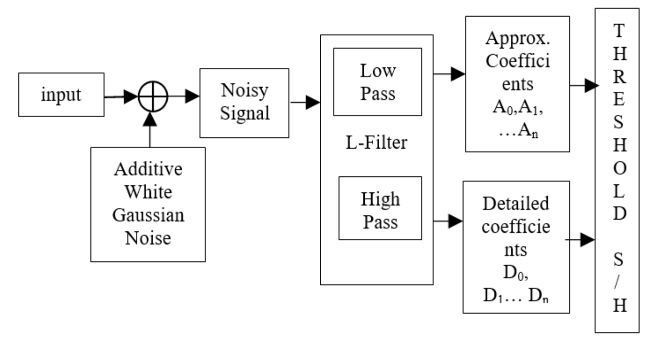


**Figure.4b Block diagram of DWT decomposition technique**

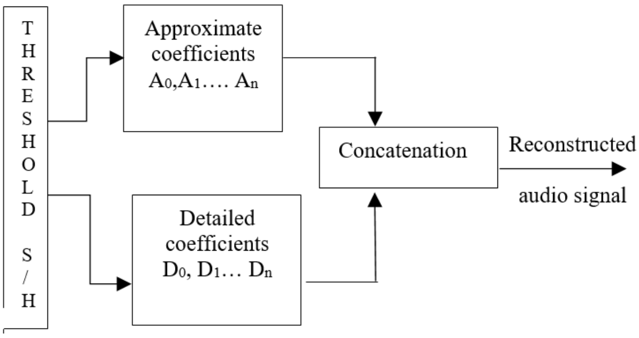


**Figure.4b Block diagram of DWTreconstruction technique**

## 4. RESULTS AND WAVEFORMS

The audio signal is imported to MATLAB and plotted as shown in figure.5

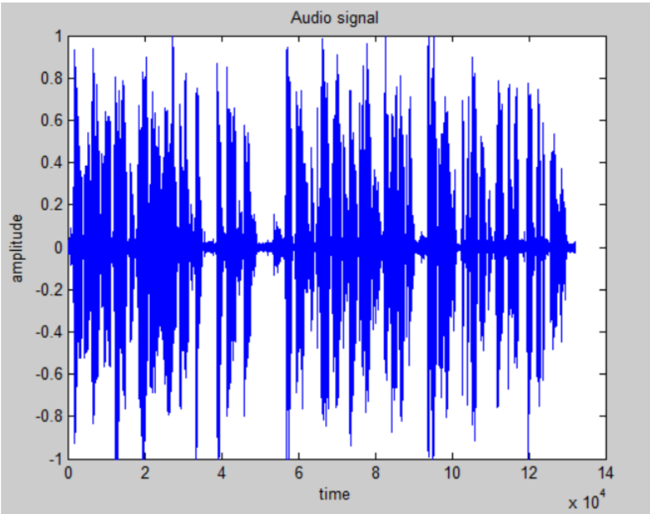


**Figure.5 – Audio signal.**

This audio signal is added with additive white gaussian noise and plotted as shown in figure.6

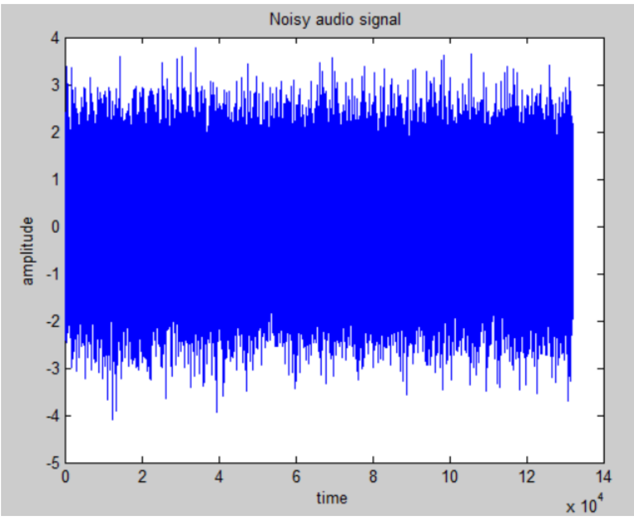


**Figure.6 Noisy Audio Signal**

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The noisy signal is passed through IIR low pass filter. Magnitude& Phase response of the filter is shown in figure.7a.1 &figure.7a.2 respectively. Also, poles & zeroes plot is shown in figure.7b.

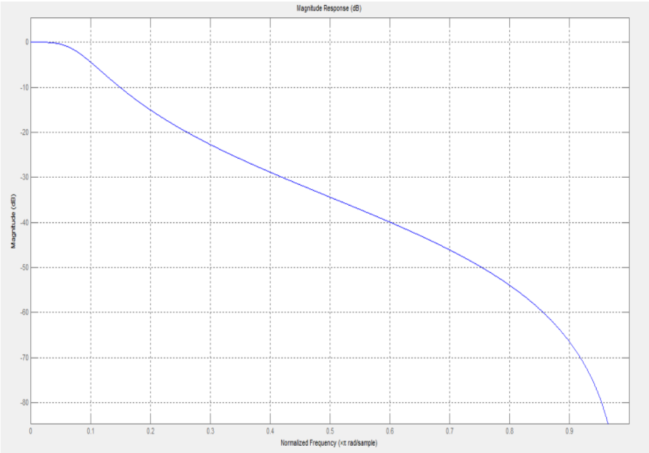

**Figure.7a.1 Magnitude Response of IIR filter**

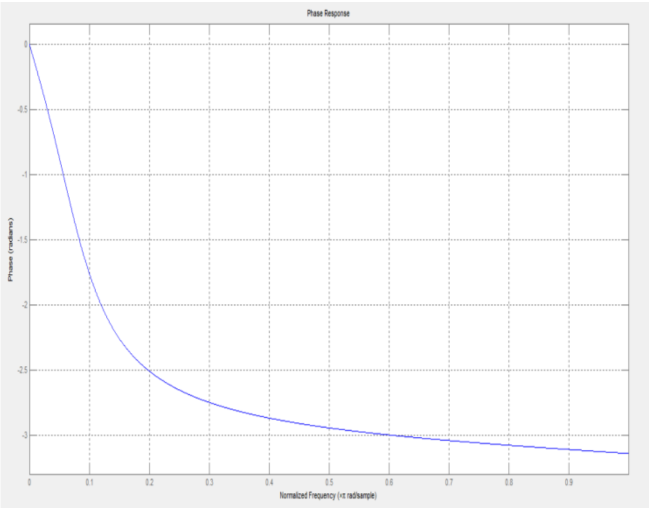

**Figure.7a.2Phase response of IIR filter**

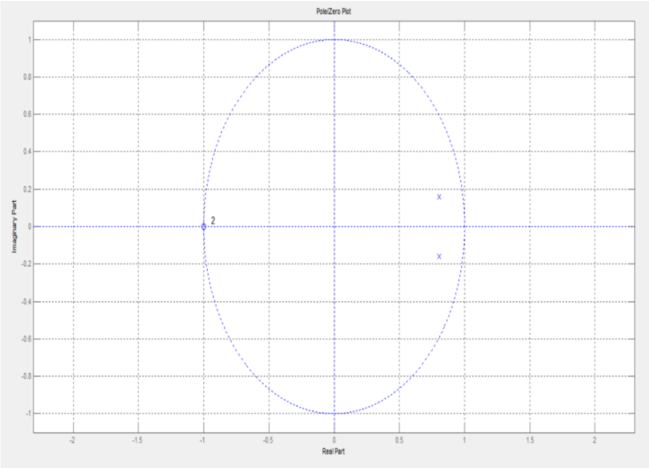

**Figure.7b poles and zeroes plot of IIR filter**

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The output of IIR lowpass filter is as shown in figure 8.

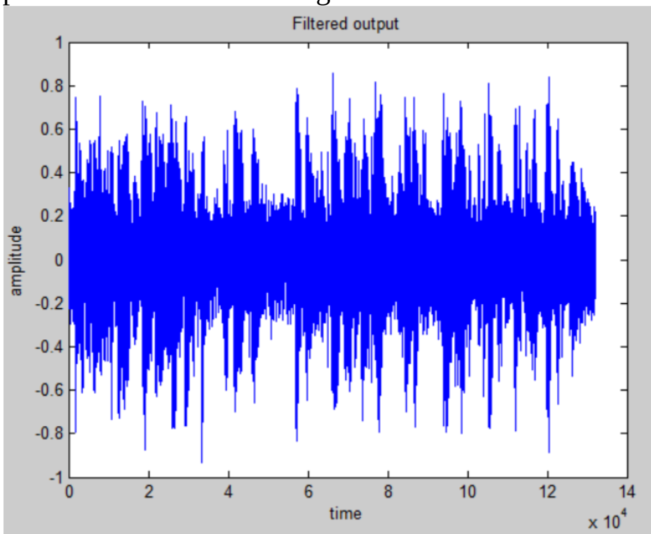


**Figure8. Denoised signal of IIR low pass filter.**

The same noisy signal is passed through the FIR filter.Magnitude& Phase Response is shown in figure.9a.1 & figure.9a.2 respectively. Pole & zeroes plot is also shown in figure.9b.

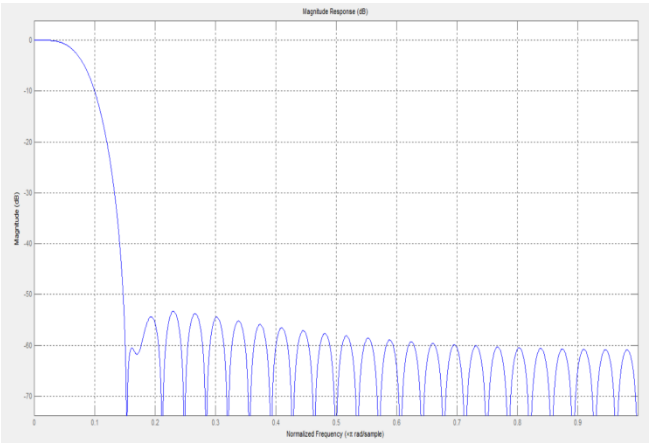

**Figure.9a.1Magnitude response of FIR lowpass filter**

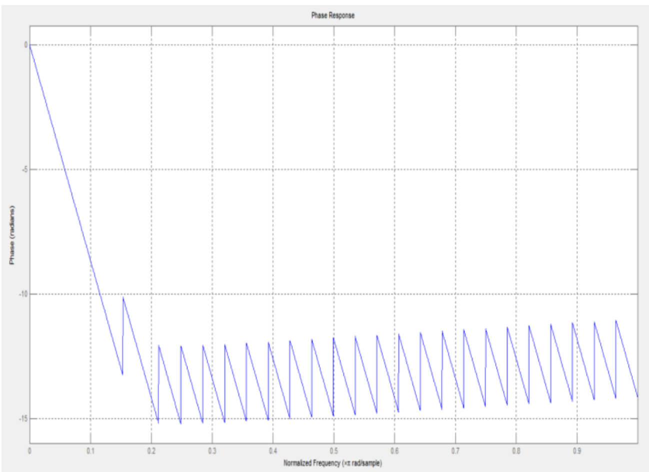

**Figure.9a.2 Phase response of FIR lowpass filter**

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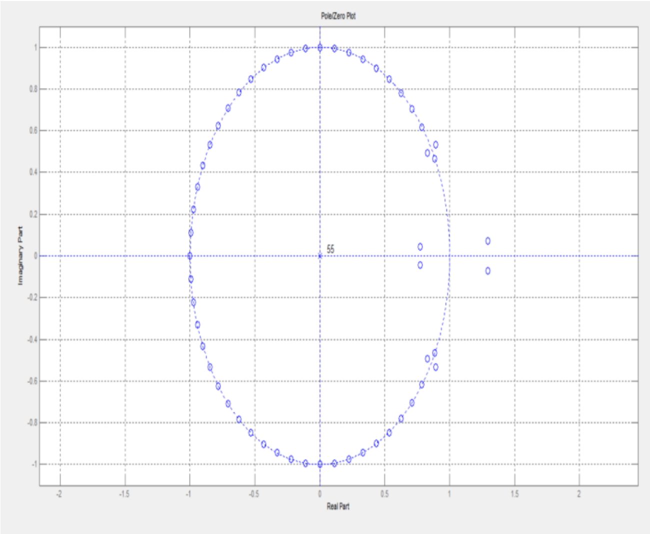

**Figure.9b pole/zero plot of FIR filter**

The output of FIR lowpass filter is as shown in figure.10

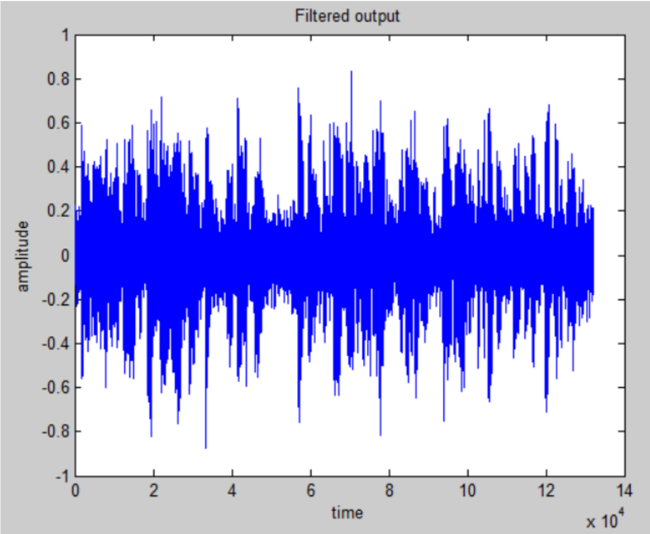


**Figure.10Denoised signal of FIR lowpass filter**

When the same noisy signal is denoised using wavelets transform technique, the output is plotted in figure.11

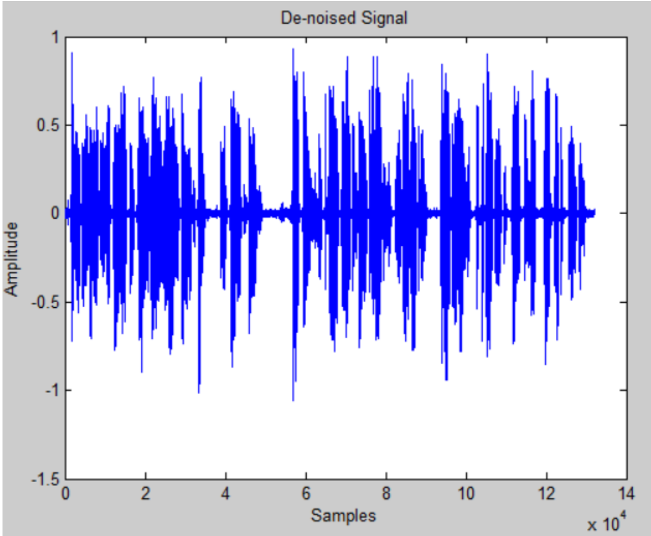


**Figure.11 Denoised signal from wavelet transform method**

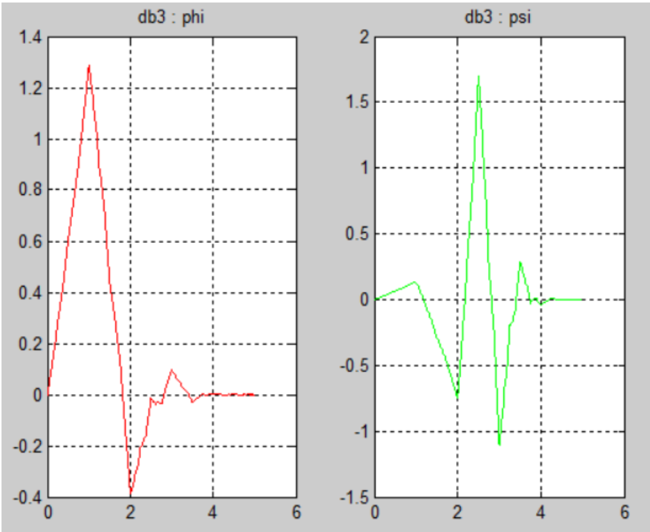


**Figure.12 Scaling function-phi and Wavelet function psi of db3 wavelet**

## 5. DISCUSSION

An analog system is said to be stable if all its poles lie in the left half of the 's'plane. Therefore, IIR is unstable as shown in figure7b. IIR filters are difficult to implement as it has delays and distortions due to large number of poles. Lower order IIR filters are better compared to higher order IIR filters. As IIR filters are unstable they cannot have a linear phase as shown in figure.7a. Theoretically we know that IIR filters depend on present input and previous output. Figure.8 shows that noise introduced is removed to certain limit, but the output audio signal is also damped at the same time.

Due to this limitation, we will prefer FIR over IIR. FIR filters do not have delays and distortions. Therefore, they are stable than IIR. FIR filters have only zeros on the unit circle in the s plane and one pole at origin as shown in figure.9b. FIR filters are better for higher orders and they maintain stability. Also, they have linear phase characteristics as shown in figure.9a. FIR filter output depends only on the present inputs. In order to avoid damping of the output audio signal we have tried various FIR filters using different windowing techniques like hamming, Kaiser and rectangular. These windowing techniques were compared in order to get the filtered audio output. It is observed that Hamming window is preferred as it had linear phase. FIR filters has denoised the signal better compared to IIR but at the same time audio signal is damped.

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Due to the drawbacks of FIR filter, wavelets are preferred over FIR to remove noise and get the filtered signal. The wavelets used here is Daubechies wavelets(db) having highest number of vanishing moments(N) with the support width of (2N-1)db. Wavelet solves the problem of signal discontinuity and is applicable for continuous and discrete wavelet transforms.

Daubechies wavelet belongs to orthogonal family and it has daughter wavelets like db1, db2 till db45. db wavelet removes noise to get filtered output. Figure.12 shows the Scaling function phi and Wavelet function psi of db3 wavelet. Low pass filters are used to avoid aliasing effect and is applicable in communication circuits as anti-aliasing filters. Bandpass filters are used to pass only certain range of frequencies which is applicable in wireless transmitters and receiversto avoid noise.

## 6. FUTURE SCOPE

In this paper, we have analyzed the audio signal by three filtering techniques -IIR, FIR and waveletswhich suppresses the noise using MATLAB functions. In some cases, the sound signal which we consider as noise may also contain some important information. For example, in military, satellite applications minute to minute information is present in the sound signal which is sent from the base stations. So,the future scope of this project aims to isolate noise from the audio signal and the denoised audio signal is generated as another signal. This project can also be used to separate the signal which have undergone stereo mixing. For example, in a song we can separate the male voice, female voice and background music which can be played as three separate audio files.

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