Denoising Real-Time Audio Signals Using Matlab Filter Techniques

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April 14, 2020

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Abstract— **In this paper we analyze real time audio signals and try to reduce the noise associated with the message signal under consideration. The main drawback of noise being present in an audio signal, is that it reduces the quality of the signal that is being transmitted within the communication system. For analysis purpose, white gaussian noise(awgn) is concatenated with the audio signal under consideration and the resulting noisy audio signal is subjected to the different filtering techniques like IIR Filter, FIR Filter, Wavelet transform techniques. The noisy audio signal is analyzed with respect to the different filter responses obtained on applying the foresaid methods. A comparative study is done between these techniques to arrive at a technique which would be the most efficient one for audio signal denoising.**

Keywords-Audio processing, Denoising, FIR filters, IIR filters, wavelets, Daubechies.

I. Introduction

The importance of noise reduction in real-time audio signals is said to be having high significance in communication. Due to this factor the noise weakens signal quality, and the recognition of audio signals becomes difficult and cause serious difficulties for the users of electronic hearing aids. A well-established method is filtering of the signal in the frequency domain or in the simplest way is analyzing the signal using different filter techniques like low pass, high pass and bandpass filters. As the voice or speech signals are not periodic these filters distort the signal more than they reduce the noise. To attenuate noise, we need more advanced methods of filtering. As it requires high advanced methods of filtering, we go for the different filtering techniques like Fourier transforms, wavelet transforms, and other methods are used in denoising the audio signals. A Fourier transform of a signal gives us the frequency composition of the audio signal. The disadvantage of Fourier transform is it is only valid within a certain Region of Convergence (ROC). So, we go for short time Fourier transform (STFT), but this method used the window analysis approach of defined size. IIR and FIR algorithms uses the Fast Fourier transform (FFT) technique for analysis of frequency spectra and signal responses. Wavelet analysis provide more detailed analysis about the signal compared to other filtering approaches.

This paper tells the readers about denoising of the audio signal. As all the audio signals are continuously affected by the different types of real-time noise such as electrostatic noise, thermal noise, channel noise, awng and etc. which creates major problem to the audio signals. Electrostatic noise which is generated due to the presence of the voltages during the design implementation and other random noises which gets added to the signal. This proposed research will solve the drawbacks of various filtering techniques which also provides unique Knowledge to the reader. The project defines a comparison between three different filtering techniques i.e. IIR filtering, FIR filtering and wavelets transforms based on real time audio signals.

II. 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 analyzed using frequency response.

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

Er. Harpal Singh[3] has used fast Fourier transform technique for performing time domain and frequency domain analysis of the signal. Mannu Singla[4] has used Butterworth filter and Chebyshev filters to reduce noise from signals with different frequency and ripple factors.

Seema rani[5] has proved more facts about FIR and IIR filters in their paper. The paper tells FIR is more stable than IIR. From the above research we can conclude that 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 and FIR filter is more stable than IIR filters.

C Mohan Rao[6] has presented a new algorithm that is the Least Mean Square (LMS) in which the awgn is added to message signal and the denoising is done to reduce the noise with minimum or no error efficiently. But it is sensitive to the scaling of its input.

B. Jai Shankar[7] has proposed the use wavelet transformation technique to denoise audio signals by dividing the signal into blocks. This technique protects each and every unique and vital features of every individual block and exposes the finest detail contributed by the grouped set of blocks. The authors Ola Ratelli, Palle, Jorgensen (2013) [8] in their book have proposed the about Discrete wavelet transform, its benefits and its functionalities. Priyanka Khattar[9] have published a paper in which denoising will be performed using wavelet transformation by comparing two wavelets families, Daubechies and Haar.

III. PROPOSED WORK

The flow of the project would be as shown in figure 1.

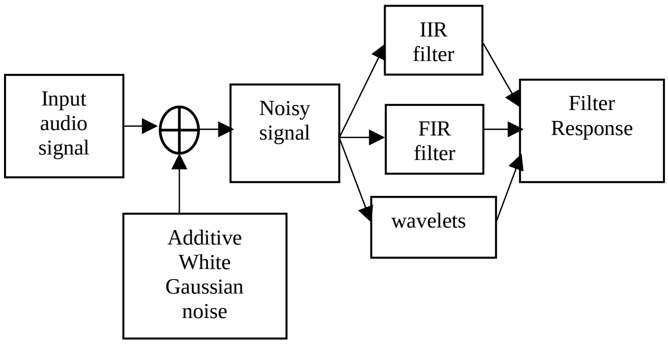

Figure.1 Block diagram of project

Using basics of noise theory and audio theory, analysis of the audio signal is carried out. The audio signal under consideration is concatenated with awgn. Using FFT, this noisy signal is converted from time domain to frequency domain is then analyzed in frequency domain by converting from time to frequency domain using Fast Fourier Transform algorithm (FFT). The following step would involve designing of the different filters taking into consideration the different parameters involved in filter design like order of the system, cutoff frequency. The filtering approach consists of normalization of signal, decomposition technique and reconstruction technique which we use in the filtering process to achieve our objectives.

A. *IIR Filter Algorithm*

The audio signal which is to be analyzed is taken as input, to this awgn is concatenated with audio signal and the resulting noisy audio signal is obtained. White gaussian noise(awgn) is preferred as it has almost constant PSD (power spectral density) and for easy analysis of the audio signal.

Now the frequency domain plot of the noisy signal is obtained by Fast Fourier Transform method and the signal is analyzed to decide the cut-off frequency. Once the cut-off frequency is determined based on the analysis, 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 and comparison with the other results obtained.

B. *Equations*

Difference Equation of IIR Filter:

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

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

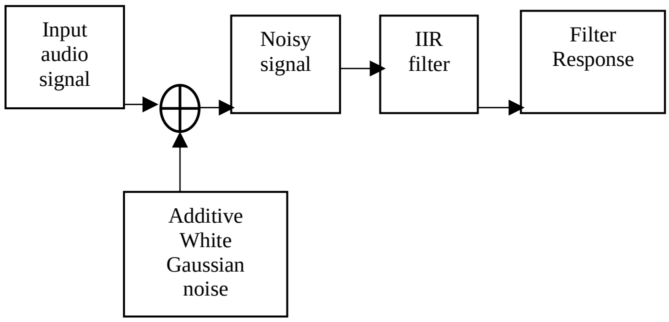

Figure 2.IIR Filtering method block diagram

C. *FIR Filter Algorithm*

The audio signal which is to be analyzed is taken as input, to this additive white gaussian noise is added and the resulting noisy audio signal is obtained. White gaussian noise is preferred as it has almost constant PSD (power spectral density) and for easy and precise analysis.

Now the frequency domain plot of the noisy signal is obtained by Fast Fourier method and the signal is analyzed with respect to the peak points to decide the cut-off frequency. Once the cut-off frequency is determined based on the analysis, the filter design process is initiated by passing the essential parameters like the order, normalized cut-off frequency, filter type (low pass filter) and appropriate windowing technique is selected. The filter coefficients (the numerator and denominator coefficients) are obtained 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 and comparison with the other results obtained.

D. *Equations*

Difference Equation of FIR filter:

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

Hamming Window function:

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

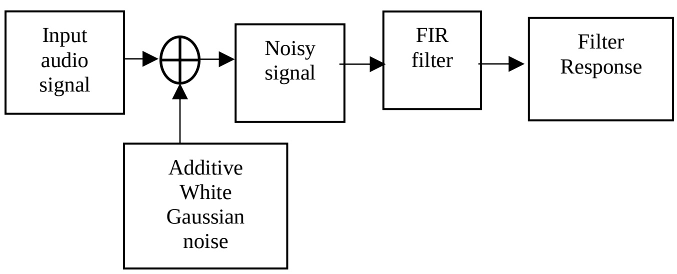

Figure 3. FIR Filtering method block diagram

E. Wavelets

Wavelet transform method can be employed for analysis of the audio signal with respect to approximation and detailed coefficients. Wavelet transforms can be classified as, continuous wavelet transform and discrete wavelet transform. We are using discrete wavelet transform technique in our project as it is more suitable for denoising of audio signals. The audio signal which is to be analyzed is taken as input, to this awgn is added and the resulting noisy audio signal is obtained. White gaussian noise is preferred as it has almost constant PSD (power spectral density) and for easy and precise analysis. This signal is passed through Level 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. Wavelet thresholding, the final step is to reconstruct the original audio signal without much loss of information. A construction process which involves using the wavelet coefficients and considering the levels of iteration, successful reconstruction of the original audio signal is obtained.

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:

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

Detailed coefficients:

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

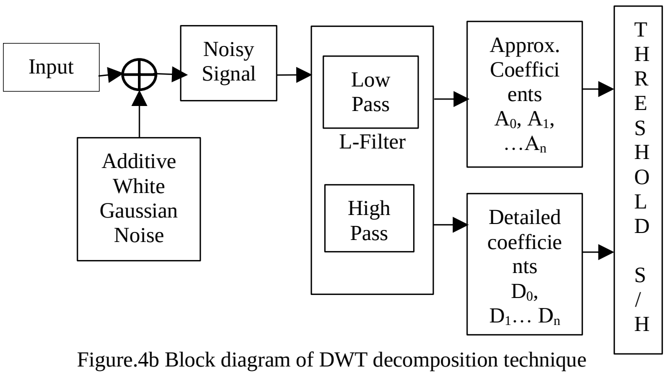

Figure.4b Block diagram of DWT decomposition technique

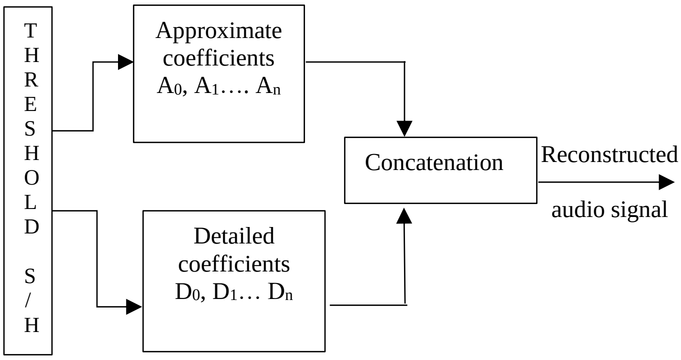

Figure.4b Block diagram of DWT reconstruction technique

IV. RESULTS AND WAVEFORMS

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

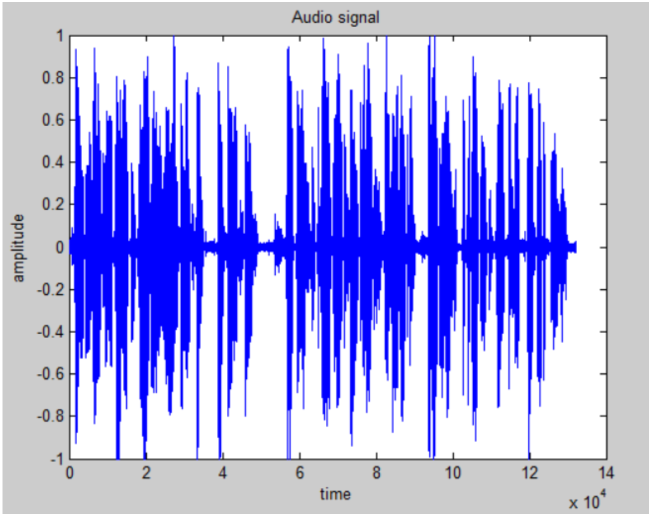

Figure.5 – Audio signal.

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

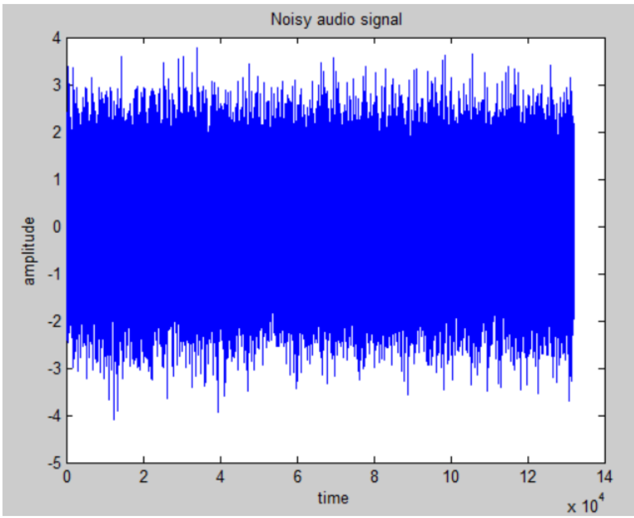

Figure.6 Noisy audio signa

The noisy signal is passed through IIR low pass filter having magnitude & phase response is as shown in figure.7a.1 &

figure.7a.2 respectively and pole & zeroes plot is as shown in figure.7b.

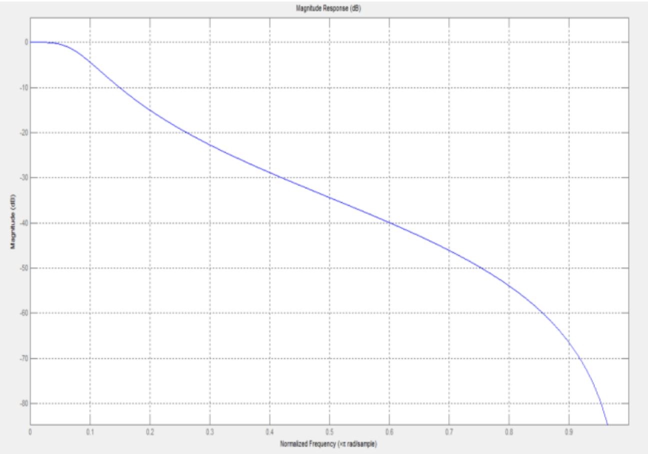

Figure.7a.1 Magnitude and phase response of IIR filter

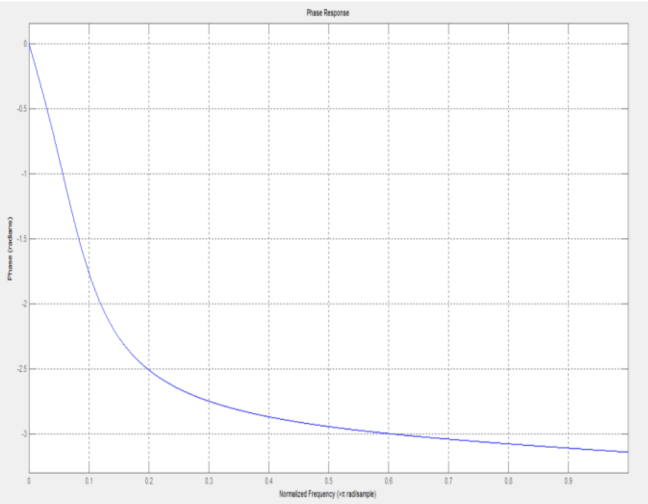

Figure.7a.2 Phase response of IIR filter

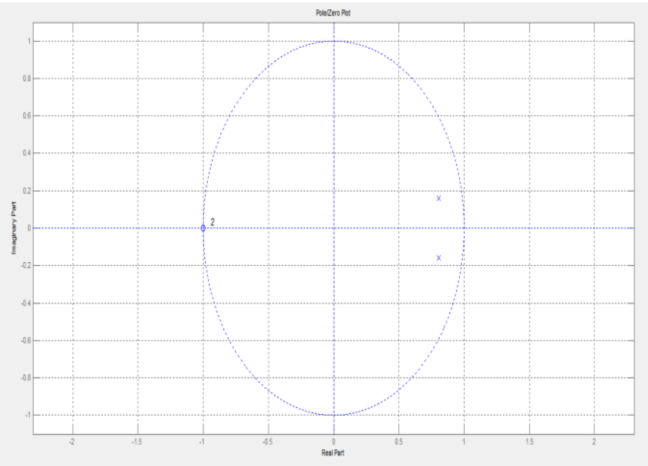

Figure.7b pole/zero plot of IIR filter

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

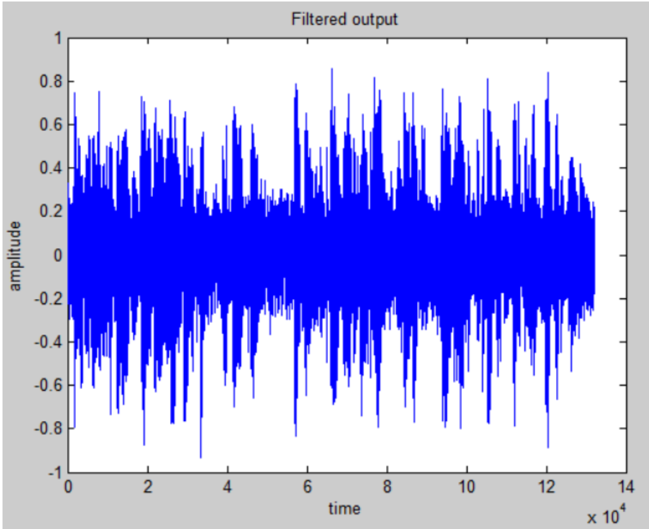

Figure8.IIR low pass filter output (denoised output)

The same noisy signal is passed through the FIR filter having magnitude & phase response as shown in figure.9a.1 & 9a.2 respectively and pole & zeroes plot as shown in figure.9b.

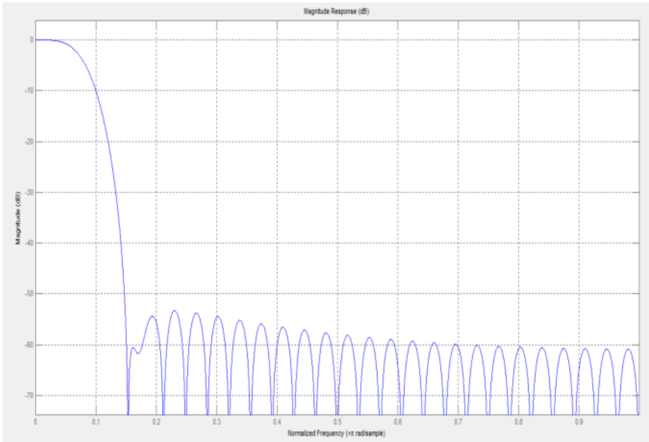

Figure.9a.1 Magnitude response of FIR lowpass filter

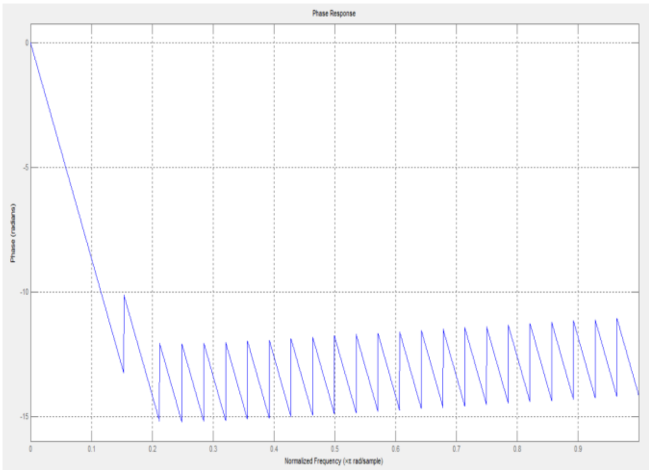

Figure.9a.2 Phase response of FIR lowpass filter

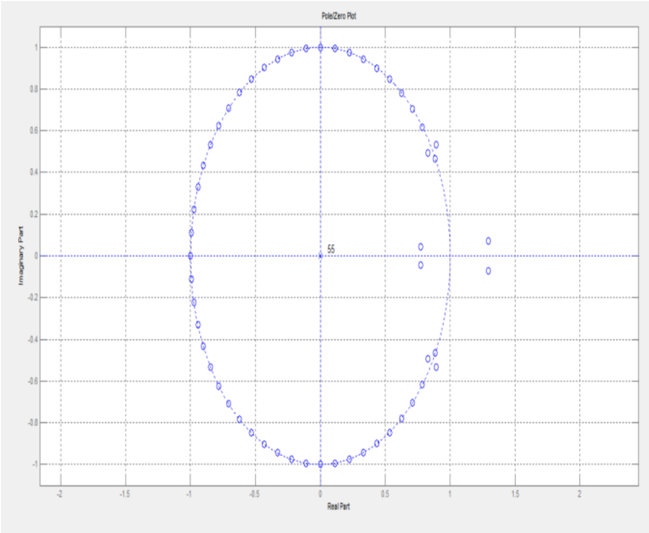

Figure.9b pole/zero plot of FIR filter

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

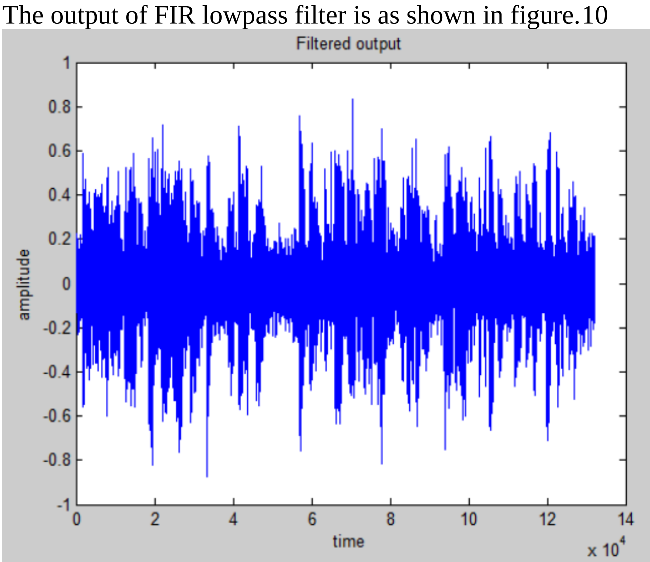

Figure.10 Output of FIR lowpass filter

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

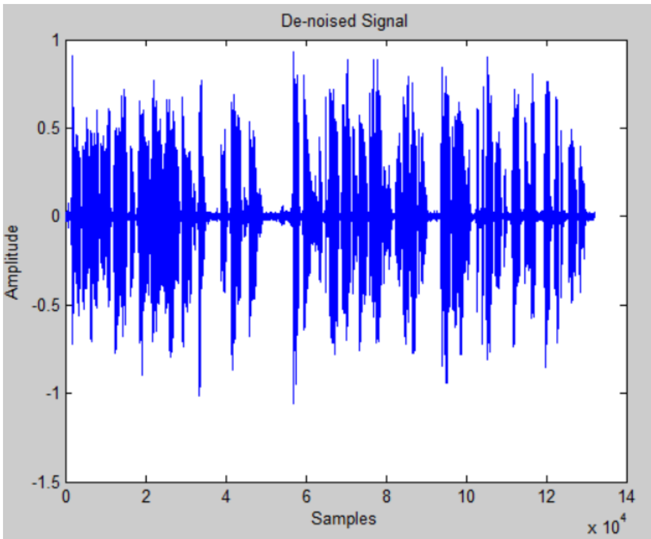

Figure.11 Denoised signal from wavelet transform method

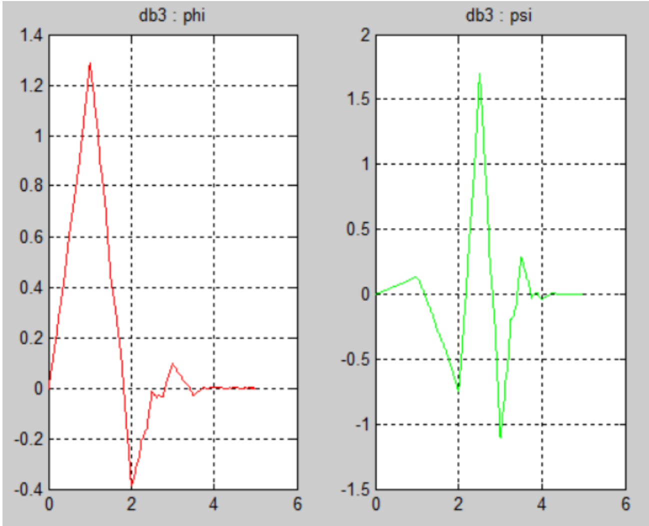

Figure.12 Scaling function-phi and Wavelet function psi of db3 wavelet

V. DISCUSSION

An analog system is said to be stable if all its poles lie in the left half of the 's' plane, but IIR filter has a both poles and zeros as we can see in figure 7b. So IIR is said to be unstable. IIR filters are difficult to implement as it has delays and distortions due to large number of poles. IIR filters are better in lower orders as IIR filters may become unstable with higher orders. As IIR filters are unstable they cannot have a linear phase as shown in the phase plot in figure.7a. Theoretically we know that IIR filters depends on present input and previous output. By the waveforms in figure.8, we observe that noise introduced is denoised to certain limit but the output audio signal will also get damped.

Due to these drawbacks we would prefer FIR over IIR as FIR filters do not have delays and distortions. Therefore, FIR filters are stable than IIR as 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. FIR filters have linear phase characteristics as shown in phase plot of figure.9a and it's 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 windows were compared to get the Filtered audio output but Hamming window was preferred as it had linear phase. FIR filters had denoised to limit much greater than IIR but the audio signal was also damped.

Due to the drawbacks we would prefer wavelets over FIR to remove noise and get the filtered signal. The wavelets used here is Daubechies wavelets have highest number N of vanishing moments with the support width 2N-1. db wave solves the problem of signal discontinuities and is applicable for continuous and discrete wavelet transforms.

db wave belongs to orthogonal family and it has daughter wavelets like db1, db2 till db45. db wavelet removes noise to get filtered output. The 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 receivers in order to avoid noise.

VI. Future scope

We have analyzed the audio signal by three filtering techniques that is IIR, FIR and wavelets which 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 information present in the sound signal from the base stations plays a major role. So, the future scope of this project is to not suppress the noise but segregate it from the audio signal and play it as store and play it 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 and play all the three as separate audio files. In this way this algorithm can be used for generating karaoke from any given song.

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