REVIEW: AUDIO NOISE REDUCTION USING FILTERS AND DISCRETE WAVELET TRANSFORMATION

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ABSTRACT-our applications include noise propagation problem in industrial air handling systems, noise in aircrafts and tonal noise from electric power, as well as isolation of vibration from which noise is one kind of sound that is unexpected or undesired . The noise related problem can be divided into non-additive noise and additive noise. The non-additive noise includes multiplier noise and convolution noise, which can be transformed into additive noise through homomorphism transform. The additive noise includes periodical noise, pulse noise, and broadband noise related problems. There are many kinds of broadband noise, which may include heat noise, wind noise, quantization noise, and all kinds of random noise such as white noise and pink noise. Statistical relationship between the noise and speech i.e. uncorrelated or even independent noise, and correlated noise (such as echo and reverberation). In acoustics applications, noise from the surrounding environment severely reduces the quality of speech and audio signals. Therefore, basic linear Filters are used to denoise the audio signals and enhance speech and audio signal quality. Our main objective is to reduce noise from system which is heavily dependent on the specific context and application. As, we want to increase the intelligibility or improve the overall speech perception quality. Such as SNR, PSNR, MSE and the Time to reduce the noise for noisy signals for removing noise.
Keywords: Chevshevby Type-1 Filter, butterworth filter, elliptic filter, MSE,SNR, PSNR

I. INTRODUCTION

Channels are the control of the adequacy and/or stage reaction of a sign as indicated by their recurrence. These are the essential segments of all sign handling and - telecom frameworks. There are two sorts of channels altered and tunable. Altered channels are those in which pass band frequencies and stop band frequencies are settled while if there should arise an occurrence of tunable channels, pass band and stop band frequencies are variable. These frequencies can be changed by necessity of the applications. Tunable computerized channels are broadly utilized in information transfers, medicinal hardware, advanced sound gear and control frameworks. These channels are otherwise called variable advanced channels. Tunable computerized channels are utilized as a part of telecom framework in the front end of a beneficiary to choose a specific band of frequencies. In medicinal gadgets, tunable score channels are utilized to smother the electrical cable obstruction [11]. The bases for the outline of the tunable advanced channels are the ghastly change [3]. It is fundamentally used to change the qualities of a channel to meet new details without rehashing the channel outline system. This adjustment is finished by changing a Low pass(LP) advanced channels to Low pass(LP) channels with distinctive cutoff frequencies or to a High pass(HP), Band pass(BP) or Band stop(BS) channels. The variable Band pass (BP) and Band stop (BS) channels are utilized to kill and recover some limited band signals. Sound clamor lessening framework is the framework that is utilized to expel the commotion from the sound signs. Sound clamor decrease frameworks can be Variable band pass and band stop channels are indicated with high precision and autonomous tuning qualities.

Audio Noise Reduction

separated into two fundamental methodologies. The principal methodology is the correlative sort which includes compacting the sound flag in some very much characterized way before it is recorded (basically on tape). The second approach is the single-finished or non-correlative sort which uses strategies to diminish the clamor level officially show in the source material—fundamentally a playback just commotion lessening framework [3]. This methodology is utilized by the LM1894 coordinated circuit, composed particularly for the diminishment of perceptible commotion in practically any sound source. Commotion decrease is the procedure of expelling clamor from a sign. Every single recording gadget, both simple or advanced, have attributes which make them vulnerable to clamor. Clamor can be irregular or background noise no soundness, or lucid commotion presented by the gadget's component or handling calculations. Their is an Active clamor control (ANC), otherwise called commotion undoing, or dynamic commotion lessening (ANR), is a system for decreasing undesirable and natural sound by the expansion of a second stable particularly intended to cross out the first. Sound is a weight wave or we can say sound is the simple flags that are prepared by recurrence, which comprises of a pressure stage and a rarefaction stage. A commotion undoing speaker transmits a sound wave with the same adequacy yet with upset stage (otherwise called against stage) to the first solid. The waves consolidate to frame another wave, in a procedure called impedance, and viably cover one another - an impact which is called stage wiping out. Current dynamic clamor control is by and large accomplished through the utilization of simple circuits or advanced sign preparing. An Adaptive calculations are intended to break down the waveform of the foundation no neural commotion, then in light of the particular calculation create a flag that will either stage move or reverse the extremity of the first flag. This hostile to stage is then opened up and a transducer makes a sound wave specifically corresponding to the abundancy of the first waveform, making ruinous obstruction [8]. This adequately decreases the volume of the recognizable commotion. The transducer emanating the clamor crossing out sign may be situated at the area where sound constriction is needed (e.g. the client's ear/any music/earphone sound). This obliges a much lower force level for retraction yet is successful just for a solitary client

Types of Noises

1. There are numerous sorts and wellsprings of commotion or mutilations and they include: Acoustic clamor exuding from moving, vibrating or impacting sources, for example, spinning
2. Machines, moving vehicles, console snaps, wind and downpour, Electromagnetic commotion that Electronic clamor, for example, warm commotion and shot clamor,
3. can meddle with the transmission and gathering of voice, picture and information over the radio-recurrence range,
4. Electrostatic commotion produced by the vicinity of a voltage,
5. Correspondence channel twisting and blurring and
6. Quantization commotion and lost information parcels because of system blockage.
7. Signal twisting is the term regularly used to depict an efficient undesirable change in a sign and alludes to changes in a sign from the non-perfect qualities of the correspondence channel, sign blurring resonations, reverberation, and multipath reflections and missing examples [10].

Contingent upon its recurrence, range or time attributes, a commotion procedure is further characterized into a few classifications:

White noise: purely random noise has an impulse autocorrelation function and a flat power spectrum. White noise theoretically contains all frequencies in equal power.

Band-limited white noise: Similar to white noise, this is a noise with a flat power spectrum and a limited bandwidth that usually covers the limited spectrum of the device or the signal of interest. The autocorrelation of this noise is sinc-shaped.

Narrowband noise: It is a noise process with a narrow bandwidth such as 50/60 Hz from the electricity supply.

Coloured noise: It is non-white noise or any wideband noise whose spectrum has a non flat shape. Examples are pink noise, brown noise and autoregressive noise.

Impulsive noise: Consists of short-duration pulses of random amplitude, time of occurrence and duration.

Transient noise pulses: Consist of relatively long duration noise pulses such as clicks, burst noise etc.

Filter

Channels are systems that procedure motions in a recurrence subordinate way. The essential idea of a channel can be clarified by inspecting the recurrence subordinate nature of the impedance of capacitors and inductors [1]. Channels have numerous commonsense applications. A straightforward, single-post, low-pass channel (the integrator) is frequently used to balance out intensifiers by moving off the increase at higher frequencies where over the top stage movement may bring about motions. A basic, single-post, high-pass channel can be utilized to piece dc counterbalance in high pick up intensifiers or single supply circuits [1]. Channels can be utilized to isolated signs, passing those of interest, and lessening the undesirable frequencies. There are **Basic Linear Design**

The functional complement to the low-pass filter is the high-pass filter. Here, the low frequencies are in the stop-band, and the high frequencies are in the pass band. Figure shows the idealized high-pass filter [11]. In the event that a high-pass channel and a low-pass channel are fell, a band pass channel is made. The band pass channel passes a band of frequencies between a lower cutoff recurrence, f l, and an upper cutoff recurrence, f h. Frequencies underneath f l or more f h are in the stop band. A romanticized band pass channel is demonstrated in Figure 1.1 [16]. A supplement to the band pass channel is the band-reject, or score channel. The admired channels characterized above, sadly, can't be effectively manufactured. The move from pass band to stop band won't be prompt, however rather there will be a move area. Stop band lessening won't be unbounded [8].

Low pass filter

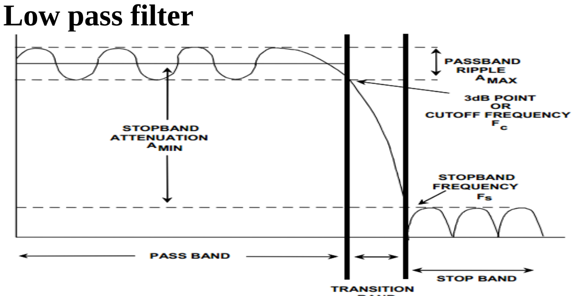


Figure 1.2 Low Pass Filter [16]

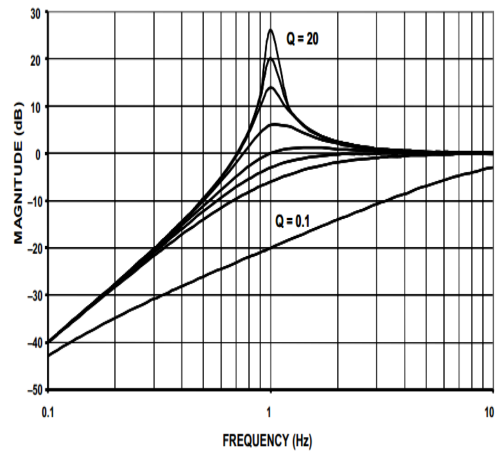


Figure 1.3 Frequency Response of Butterworth

Band-Pass Filter

Changing the numerator of the low pass prototype to $H_0\omega_0$ will convert the filter to a band-pass function. The transfer function of a band-pass filter is then:

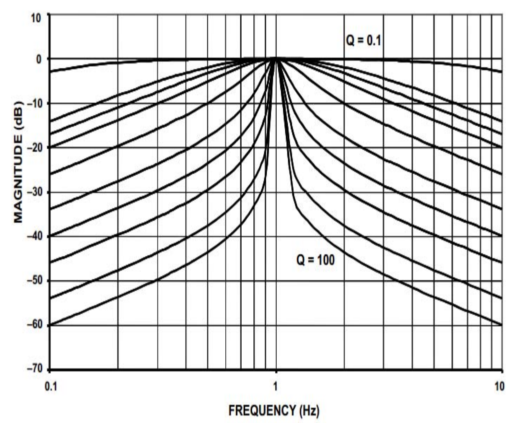


Figure 1.4 Band-Pass Filter

Band-Reject (Notch) Filter

By changing the numerator to $H_0(S^2+\omega_z^2)$, we convert the filter to a band-reject or notch filter. As in the bandpass case, if the corner frequencies of the band-reject filter are separated by more than an octave (the wideband case), it can be built out of separate low- pass and high-pass sections. We will adopt the following convention: A narrow-band band-reject filter will be referred to as a notch filter and the wideband band-reject filter will be referred to as band-reject filter.

Butterworth Filter

The Butterworth channel is the best trade off in the middle of weakening and stage reaction. It has no swell in the pass band or the stop band, and due to this is at time called a maximally level channel [16]. The Butterworth channel accomplishes its levelness to the detriment of a generally wide move district from pass band to stop band, with normal transient attributes. The standardized shafts of the Butterworth channel fall on the unit circle (in the s plane) [3].The posts are dispersed equidistant on the unit circle, which implies the points between the shafts are equivalent. Given the post locations, H0, andQ0 (or Q) can be resolved. These qualities can then be use to focus the part estimations of the channel. The configuration tables for uninvolved channels use recurrence and impedance standardized channels. The Butterworth channel is standardized for a - 3 dB reaction at H0*Q0 = 1[3]. The Butterworth channel is the best trade off in the middle of lessening and stage reaction. These qualities can then be used to focus the part estimations of the channel. The configuration tables for latent channels use recurrence and impedance standardized channels. The Butterworth channel is standardized for a - 3 dB reaction at H0.

Elliptical Filter

The already specified channels are all-shaft outlines, which imply that the zeros of the exchange capacity (foundations of the numerator) are at one of the two extremes of the recurrence go (0 or ∞). For a low-pass channel, the zeros are at f =∞ . In the event that limited recurrence exchange capacity zeros are added to posts an Elliptical channel (now and then alluded to as Cauer channels) is made. This channel has a shorter move area than the Chebyshev channel on the grounds that it permits swell in both the stop band and pass band [16]. It is the expansion of zeros in the stop band that causes swell in the stop band yet gives a much higher rate of lessening, the most workable for a given number of shafts. There will be some "bounceback" of the stop band reaction between the zeros. This is the stop band swell. The Elliptical channel likewise has debased time space reaction. Since the posts of an elliptic channel are on a circle, the time reaction of the channel looks like that of the Chebyshev[3].

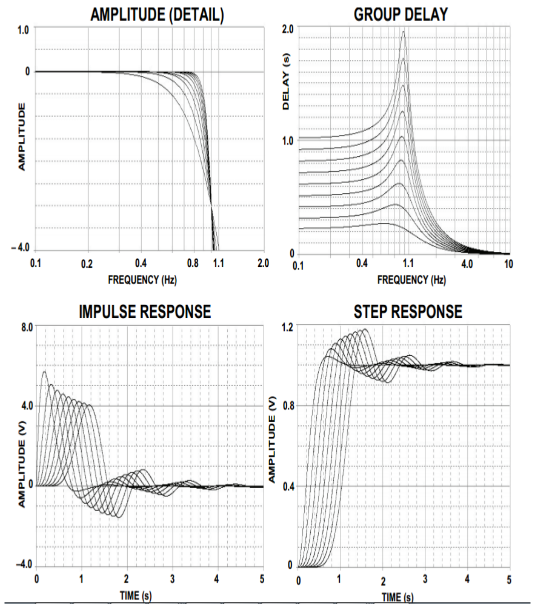


Figure 1.5 Steps and Impulse Response

Chebyshev Type1 Filter

Chebyshev1 filters have a narrower transition region between the passband and the stopband. The sharp transition between the passband and the stopband of a chebyshev filter produces smaller absolute errors and faster execution speeds than a Butterworth filter. The poles of chebyshev1 filter lies on an ellipse [16]. Ripple increase (band), the roll-off becomes sharper (good). The chebyshev filter is completely defined by three parameters-cut-off frequencies, number of poles and passband ripples. The chebyshev response is a mathematical strategy for achieving a faster roll off by allowing ripple in the frequency response. The chebyshev response is an optimal trade- off between parameters.

Wav

Waveform Audio File Format (WAVE, or all the more ordinarily known as WAV because of its filename extension),(also, yet once in a while, named, Audio for Windows) is a Microsoft and IBM sound record design standard for putting away a sound bit stream on PCs. It is an utilization of the Resource Interchange File Format (RIFF) bit stream position system for putting away information in "pieces", and along these lines is likewise near to the 8SVX and the AIFF organization utilized on Amiga and Macintosh PCs, respectively [17]. It is the primary arrangement utilized on Windows frameworks for crude and ordinarily uncompressed sound. The typical bit stream encoding is the direct heartbeat code regulation (LPCM) group.[2]

Spectral density

In factual sign handling and material science, the ghostly thickness, power ghastly thickness (PSD), or vitality unearthly thickness (ESD), is a positive genuine capacity of a recurrence variable connected with a stationary stochastic procedure, or a deterministic capacity of time, which has measurements of force per hertz (Hz), or vitality per hertz [16]. It is regularly called essentially the range of the sign. Instinctively, the ghostly thickness measures the recurrence substance of a stochastic process and aides recognize periodicities. [7]

Thesis Outline:

This thesis report has been divided in to the following chapters ·

- "**Introduction**", it includes the information about audio, wavelet, different types of filters like Butterworth filter, Chebyshev filter and elliptic filter and overview of dissertation report. It also includes the problem definition and Objective of the dissertation work.
- "**Review of Literature**", it provides overview of the work done in this area.
- **"problem definition" and "objective of this work".**
- "**Methodology**", explain the explicit algorithm and flowcharts. It includes the tool used to implement the research work.
- "**Results and Discussion**", gives the detail about result obtained and discussion about visual results.
- "**Conclusion and Future Work**", gives the detail of future work that can be done on this de-noising.
- **Publications**

References & Appendices

II. RELATED WORK

Previous Research work on **"Audio noise reduction using filters and Discrete Wavelet Transformation"** in the literature survey starting from 2010 to November 2013 was studied. I studied so many papers.

Komal Singla et.al [2014] have presented the sound noise reduction. Speech signal analysis is one of the important areas of research in multimedia applications. Digital filters effectively reduce the unwanted higher or lower order frequency components in a speech signal. The additive noise includes periodical noise, pulse noise, and broadband noise related problems. The noise generated by the engine is one kind of periodical noise while the one generated from explosion, bump, or discharge is pulse. There are many kinds of broadband noise, which may include heat noise, wind noise, quantization noise, and all kinds of random noise such as white noise and pink noise. In acoustics applications, noise from the surrounding environment severely reduces the quality of speech and audio signals. Therefore, basic linear filters are used to denoise the audio signals and enhance speech and audio signal quality. Our objective is of a noise reduction system with heavily dependent on the specific context and application as to increase the intelligibility or improve the overall speech perception quality which aimed to reduce unwanted ambient sound by implementing through different filters.[1]

Nishan Singh and Dr. Vijay Laxmi [2014]: have described Speech signal analysis is one of the important areas of research in multimedia applications. Discrete Wavelet technique is effectively reduces the unwanted higher or lower order frequency components in a speech signal. Wavelet-based algorithm for audio de-noising is worked out. We focused on audio signals corrupted with white Gaussian noise which is especially hard to remove because it is located in all frequencies. We use Discrete Wavelet transform (DWT) to transform noisy audio signal in wavelet domain. It is assumed that high amplitude DWT coefficients represent signal, and low amplitude coefficients represent noise. Using thresholding of coefficients and transforming them back to time domain it is possible to get audio signal with less noise. Our work has been modified by changing universal thresholding of coefficients which results with better audio signal. In this various parameters such as SNR, Elapsed Time, and Threshold value is analyzed on various types of wavelet techniques alike Coiflet, Daubechies, Symlet etc.[2]

Manjeet Singh et.al [2014] have discussed Digital filters effectively reduce the unwanted higher or lower order frequency components in a speech signal. In this paper the speech enhancement is performed using different digital filters .In this real noisy environment is taken into consideration in the form of Gaussian noise. The Time domain as well as frequency domain representation of the signal spectra is performed using Fast Fourier transformation technique. MATLAB in built functions are used to carry out the simulation. Gaussian type noise is added using in-built function randn () and keyboard noise is added as a second speech file to the original speech signal. The filters remove the lower frequency components of noise and recover the original speech signal. It is also observed that keyboard noise is typical to remove as compared to Gaussian type but these filters performed well to get sharper spectra of original speech signal.[3]

Prateek Basavapur Swamyet.al [2013]: have stated the noise cancellation technique to remove impulsive noises that commonly corrupt speech signals. A discrete wavelet transform is applied on the corrupted speech signal to obtain the approximation and detail coefficients. Reconstruction is done using only the detail coefficients. A threshold depending on the signal statistics is applied on the reconstructed signal to detect information of the time occurrence of impulsive noise. Based on the number of samples at a stretch that are corrupted, an adaptive filter with a variable size window is applied on the corrupted speech signal to remove the impulse noise. Evaluations of the proposed system show that the intelligibility.[4]

C Mohan Rao et.al [2013]: presented another versatile channel whose coefficients are alterably changing with a developmental reckoning calculation and consequently diminishing the commotion. This calculation gives a relationship between the redesign rate and the base blunder which naturally modifies the upgrade rate. At the point when nature is fluctuating, the rate is expanded while it would be diminished when the earth is steady and the reckoning multifaceted nature of versatile channel can be fundamentally lessened. In the reenactment, added substance white Gaussian commotion is added to the arbitrarily produced data sign and proficiently decreased this clamor with least or no slip by utilizing transformative calculation with Least Mean Square (LMS) calculations. Versatile Noise Cancelation is an option method for scratching off commotion show in an undermined sign. [5]

K.P. Obulesul et.al [2013] have concentrated on the sound signs are engineered signs, in which music or discourse, are regularly debased by commotion amid recording and transmission. Discourse upgrade is a long standing issue with various applications running from portable amplifiers, to coding and programmed acknowledgment of discourse signs and so on and accept that the clamor is added substance and factually autonomous of the sign. Sound denoising methods are intended to constrict the clamor and hold the sign of hobby. Lessening of commotion from sound signs has two techniques, Diagonal & Non Diagonal sound denoising calculations. In this paper, Non askew technique is utilized as a part of which Block parameters are naturally changed in accordance with the way of the sound flag by minimizing a Stein estimator which is computed systematically from loud flag values. This Block thresholding strategy takes out "musical commotion" by gathering Time-recurrence coefficients in pieces before being attenuated.[6]

Matheel E. Abdulmunim et.al [2013] have introduced the advanced features are regularly tainted by a clamor amid the obtaining procedure, stockpiling and transmission. It showed up furthermore influence on another advanced feature procedures like pressure, highlight extraction and example acknowledgment so feature denoising is profoundly attractive process keeping in mind the end goal to enhance the feature quality. There are numerous change for denoising process, one of them are Fast Discrete Wavelet Transform(FDWT) and framelet change (Double-Density Wavelet Transform) which is an immaculate in denoising process by evading the issues in alternate changes. In this paper we propose a system named Translation Invariant with Wiener channel (TIW) this strategy is proposed to take care of the movement fluctuation issue and utilize this technique to denoise a loud feature with Gaussian repetitive sound. It is connected with Two Dimensional Fast Discrete Wavelet Transform (2-D FDWT), Three Dimensional Fast Discrete Wavelet Transform (3-D FDWT), Two Dimensional Double Density Wavelet Transform (2-D DDWT) and Three Dimensional Double Density Wavelet Transform (3-D DDWT). [7]

Anju et.al [2012] have discussed a direct design of Infinite Impulse Response filters (IIR filters) which minimizes group delay without changing the magnitude response of filters. In this paper

Butterworth and Chebyshev1 lowpass filters are designed by using allpass filters. The design specifications are passband and stopband frequencies and passband ripple and stopband attenuation. In this paper MATLAB programming is used for implementation of proposed algorithm. Experimental results show that the proposed method can effectively optimize the group delay of the designed Butterworth and chebyshev1 low pass filters.[8]

Raghavendra Sharma et.al [2012] In this author studied a robust DWPT based adaptive bock algorithm with modified threshold for denoising the sounds of musical instruments shehnai, dafli and flute is proposed. The signal is first segmented into multiple blocks depending upon the minimum mean square criteria in each block, and then thresholding methods are used for each block. All the blocks obtained after denoising the individual block are concatenated to get the final denoised signal. The discrete wavelet packet transform provides more coefficients than the conventional discrete wavelet transform (DWT), representing additional subtle detail of the signal but decision of optimal decomposition level is very important. When the sound signal corrupted with additive white Gaussian noise is passed through this algorithm, the obtained peak signal to noise ratio (PSNR) depends upon the level of decomposition along with shape of the wavelet. Hence, the optimal wavelet and level of decomposition may be different for each signal. The obtained denoised signal with this algorithm is close to the original signal [9].

B. Jai Shankar et.al [2012] have presented the clamors show in correspondence channels are irritating and the recuperation of the first flags from the way with no commotion is exceptionally troublesome errand. This is accomplished by denoising methods that expel commotions from a computerized sign. Numerous denoising procedure have been proposed for the expulsion of commotions from the advanced sound Signals. However, the adequacy of those strategies is less. In this paper, a sound denoising system in light of wavelet change is proposed. [10]

K. Durai swamy et.al [2012] have exhibited the clamors display in signs are hard to recuperate utilizing the conventional techniques. Presently wavelet change is utilized for denoising methods. The thresholding both hard and delicate are utilized as a part of wavelet change. The procedure uncovered every single finest subtle elements contributed by the gathered arrangement of squares furthermore it secures the indispensable and interesting elements of each individual piece. The pieces are separated and supplanted in their unique positions from where they are confined. Their usage results uncover that the proposed strategy accomplishes a cutting edge denoising execution regarding sign to-clamor ratio.[11]

J. Jebastine et.al [2012] In this paper the creator depicted the advancement of a versatile clamor wiping out calculation for successful acknowledgment of discourse sign furthermore to enhance SNR for a versatile step size information. A versatile channel with Fast Block Least Mean square Algorithm is intended for commotion free sound (discourse/music) signals. The sign information utilized is a sound discourse signal which could be as a recorded voice. The channel utilized is versatile channel and the calculation utilized is Fast Block LMS calculation. A Gaussian clamor is added to this info flag and given as an information to the Fast Block LMS. [12]

D K Vishwakarma et.al [2012] have discussed a simple and novel approach for de-noising of the Audio Signals i.e. non-stationary signal using statistical distribution function at different sub-band level of coefficients. The performance of wavelets are analysed under various thresholding techniques. Non stationary signals are continuous in nature consequently we use 1D Discrete Wavelet Transform which gives us a better time- frequency localization as compared to the spectral analysis of Fourier Transform. The coefficients of wavelet are modelled on the basis of Heavy Tailed Distribution function which gives a valuable and stable representation against Gaussian distribution function in filtering noise components from the signal. We have used the statistically independent White Noise of magnitude 5db to make the noisy audio signal. The performance is numerically assessed in terms of Signal-to-Noise ratio (SNR) and Mean-Square error(MSE) terms. The Coiflets wavelet in combination with Neighboring Coefficients with Level-Dependent Threshold Estimator shows superior performance in comparison to other wavelets.[13]

III. WHY TO STUDY THIS

The issue embraced for the exposition is "Sound Noise lessening utilizing Different Filters and DWT". The Current applications include noise propagation problem in industrial air handling systems, noise in aircrafts and tonal noise from electric power, as well as isolation of vibration from noise is one kind of sound that is unexpected or undesired . The noise related problem that I have studied can be divided into non-additive noise and additive noise. The non-additive noise includes multiplier noise and convolution noise, which can be transformed into additive noise through homomorphism transform. The additive noise includes periodical noise, pulse noise, and broadband noise related problems. The noise generated by the engine is one kind of periodical noise while the one generated from explosion, bump, or discharge is pulse noise problem that I have studied in literature survey. There are many kinds of broadband noise, which may include heat noise, wind noise, quantization noise, and all kinds of random noise such as white noise and pink noise. Therefore, basic linear filters are used to denoise the audio signals and enhance speech and audio signal quality.

Objectives of Work

This dissertation entitled **"Audio Noise reduction using Different Filters and discrete wavelet transformation"** aims for the following objectives:

- The objective of a noise reduction system is heavily dependent on the specific context and application. In some scenarios, for example, we want to increase the intelligibility or improve the overall speech perception quality.

- Study of noise cancellation discrete wavelet transformation with their types and thresholding and filters.

- Study and Analyze The Results Being Obtained.

- Noise reduction technology is aimed at reducing unwanted ambient sound, and is implemented through two different methods.

- To improve the results from the existing techniques.

There are different types of parameters are calculated that is SNR, PSNR, MSE and the Time to reduce the noise for noisy signals.

IV. Methodology

The dissertation removes noise from the audio signal. It is based upon GUI (graphical user interface) in MATLAB. It is an effort to further grasp the fundamentals of MATLAB and validate it as a powerful application tool. There are basically different files. Each of them consists of m-file and figure file. These are the programmable files containing the information about the filter and figure files are the way to analyze the given audio and enter the various filter related data. Now open these files in the Matlab individually. Now run the first file and then filter file and then filtered sound file to remove the noise.

In this work we will firstly upload the sound in the format .wav in the given window. Listen the sound which will appear to be noisy .In the GUI we will take the filter button and when click on the filter sound button than a new window will open named filter sound. Then choose the desired filter for denoising and enter its various parameters namely type, order, passband frequency, stopband frequency, passband ripple and stop band ripple. Then click on ok than a new window filtered sound will open. This window shows filtered sound along with the graphs and various details about the filters like Transfer function, Step impulse and Frequency response. Study three graphs of the noisy sound namely spectrogram power spectral density and amplitude vs time .

Graphical User Interface (GUI)

MatLab provides Graphical User Interface Development Environment (GUIDE).A MatLab tool used to create GUI's. Decide between using GUIDE or writing the code from scratch GUI's give the user a simplified experience running a program. Associates a "function(s)" with components of the GUI.GUI should be consistent and easily understood. Provide the user with the ability to use a program without having to worry about commands to run the actual program.

Components of GUI

1. Push button.
2. Edit text.
3. Static text.
4. Slider.
5. Checkbox.
6. Pop-up-menu.
7. Radio button.
8. Panel.
9. List box.
10. Button group.
11. ActiveX control.
12. Toggle button.
13. List box.

This chapter contains the stepwise, detailed of two proposed algorithms that are followed while denoising audio signals using filters and wavelet transforms. For better and easy understanding, a complete flowchart of the proposed algorithms has been shown at the end of this chapter.

Algorithm for Filters

Step 1: Start the program.

Step 2: Load the wave signal that you have to denoise.

Step 3: Play the wave original signal

Step 4: Apply filters

Step 5: Choose different types

Step 6: Butter worth Filter i.e. low pass filter, high pass filter, band pass filter, band reject filter

Step 7: Chevbyshev type-1 i.e. low pass filter, high pass filter, band pass filter, band reject filter

Step 8: Elliptic Filter i.e. low pass filter, high pass filter, band pass filter, band reject

Step 9: Choose the order of the filter and enter different types of frequency and press ok button

Step 10: Check the impulse response, step response, zero plots, and frequency response and play the denoised signal

Step 11: Calculate the PSNR, SNR and MSE parameters

Step 12: Stop

The above algorithm shows the working of filters that is used to remove the noise from the audio signals.

V RESULTS

The Chapter result and & analysis include the information about the implementation results that is calculated after reducing the noise from signals using filters.Their are different figures that shows the different working of the filters.In this there is starting window that is used to upload the input wave signal and browse the original signal. for comparing the noise reduction of the signals with different filters i.e. Butterworth filter,Chebysve filter and elliptic filter.

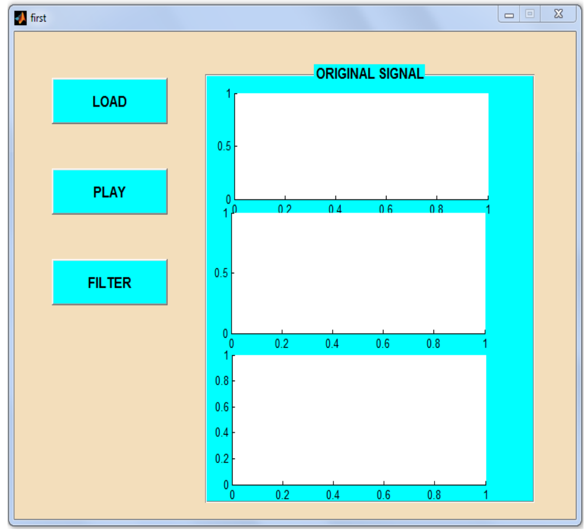


Fig. 5.1 Starting window of the audio signal

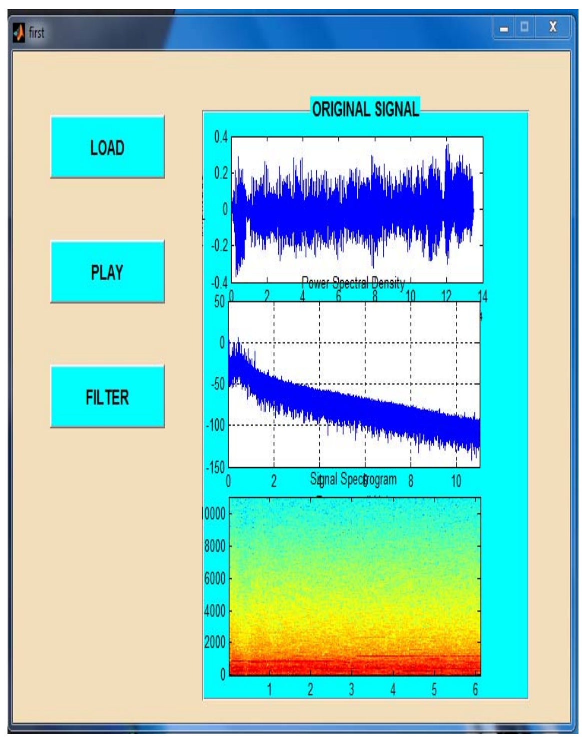
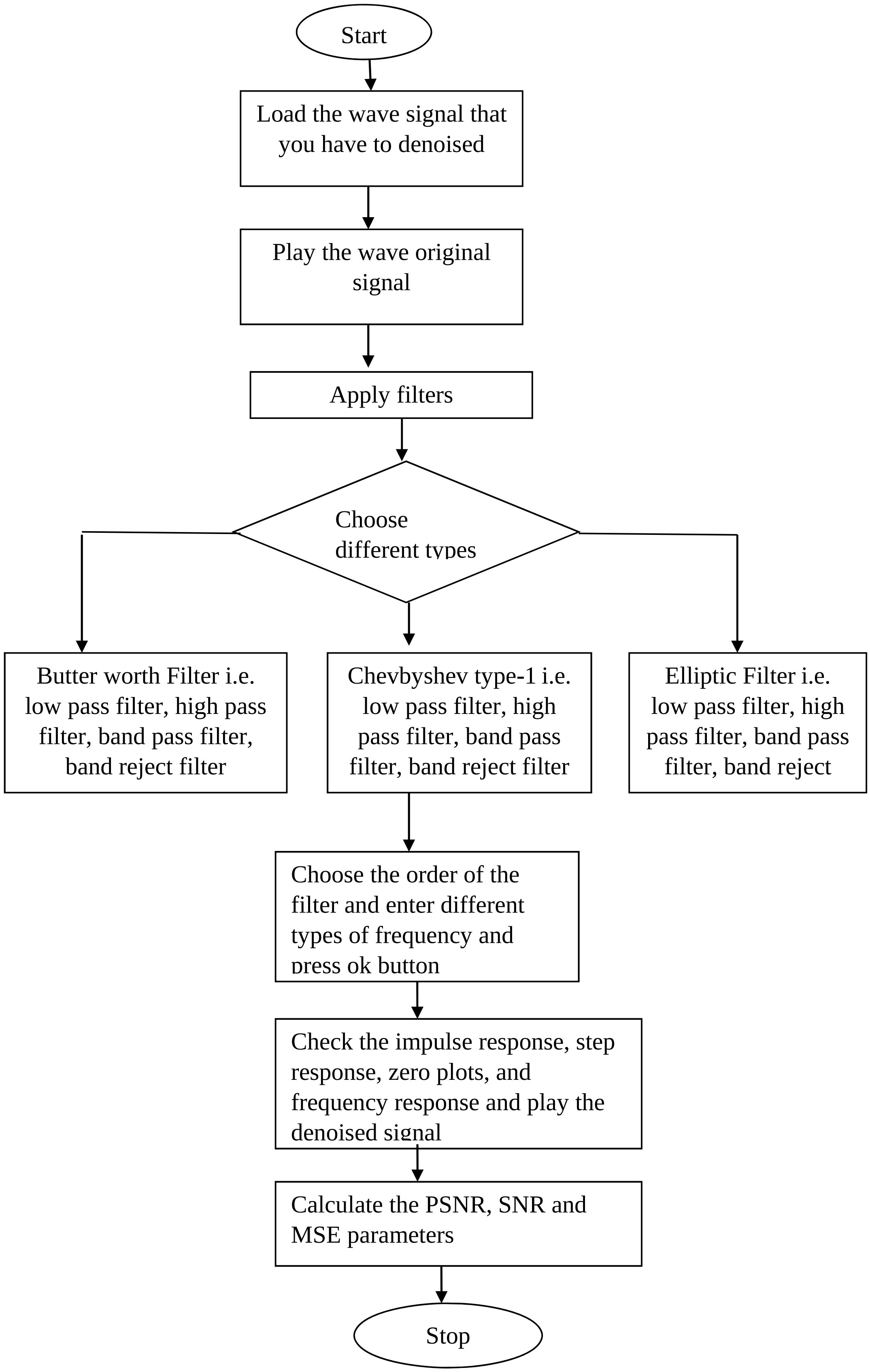


Fig.5.2 Load the input wav signal

Flowchart of Proposed Algorithm for filters



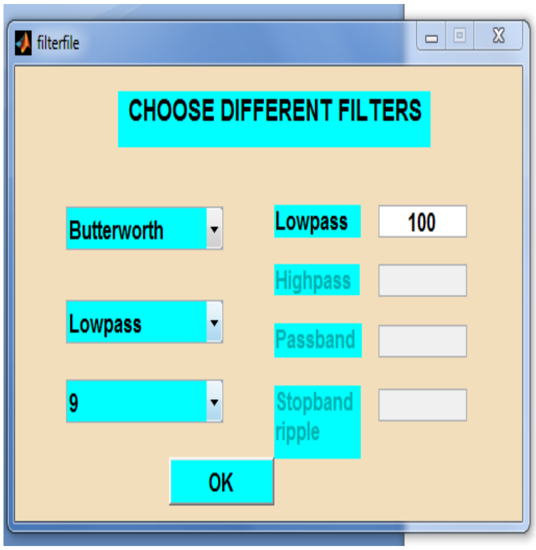


Fig.5.3 Choose Filter Type

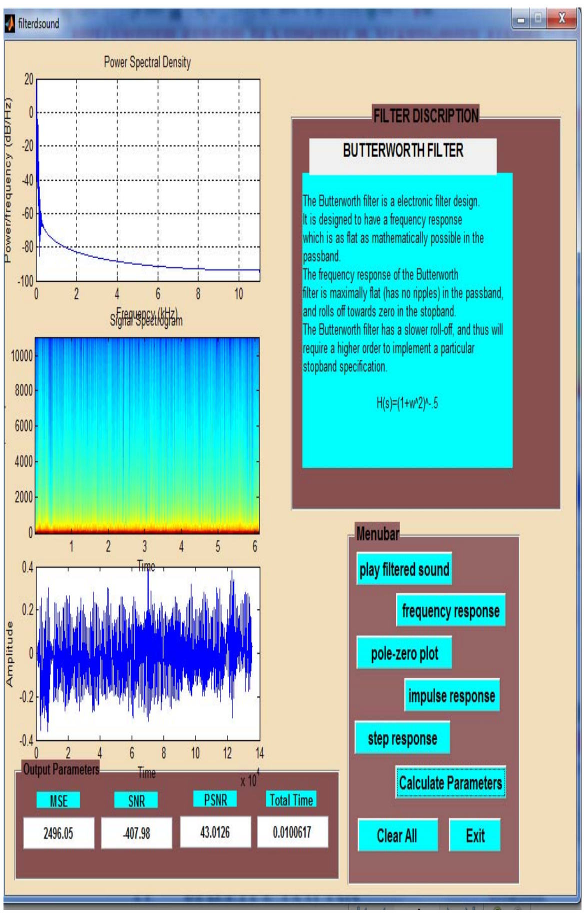
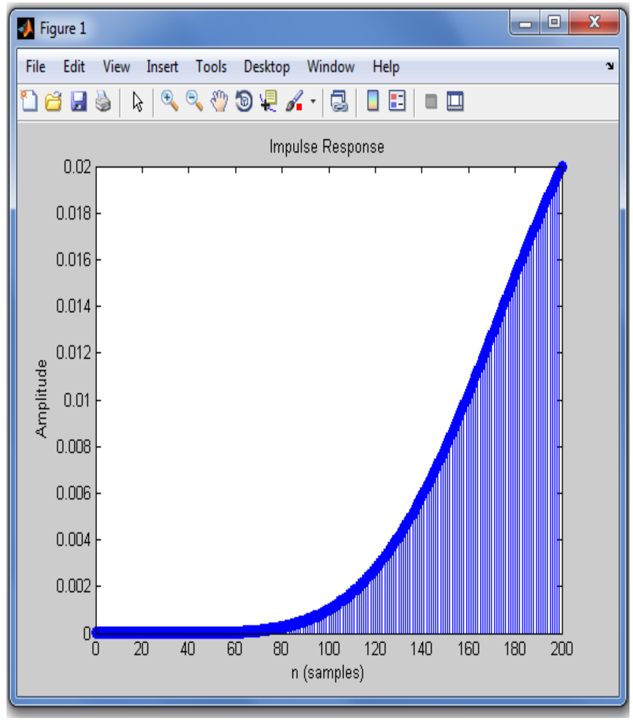


Fig.5.4 Butterworth filter Output for Noise Reduction

Fig. 5.5 Impulse Response



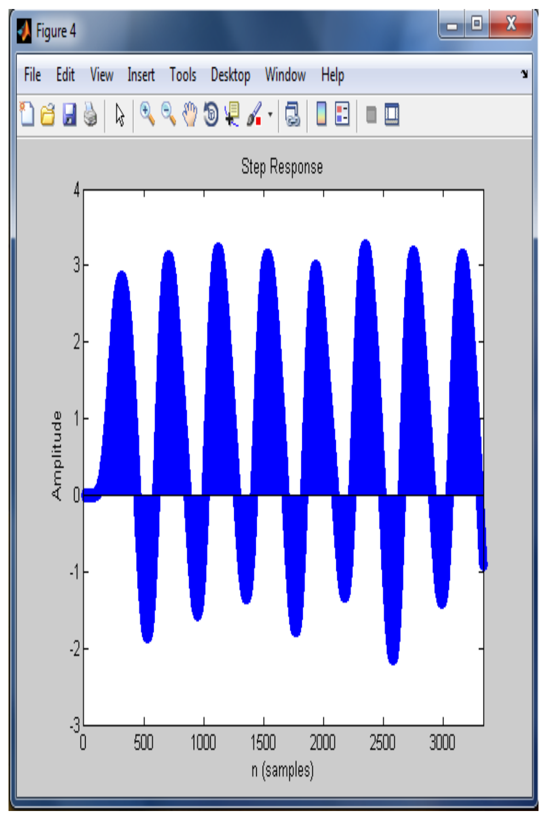


Fig. 5.6 Step Response

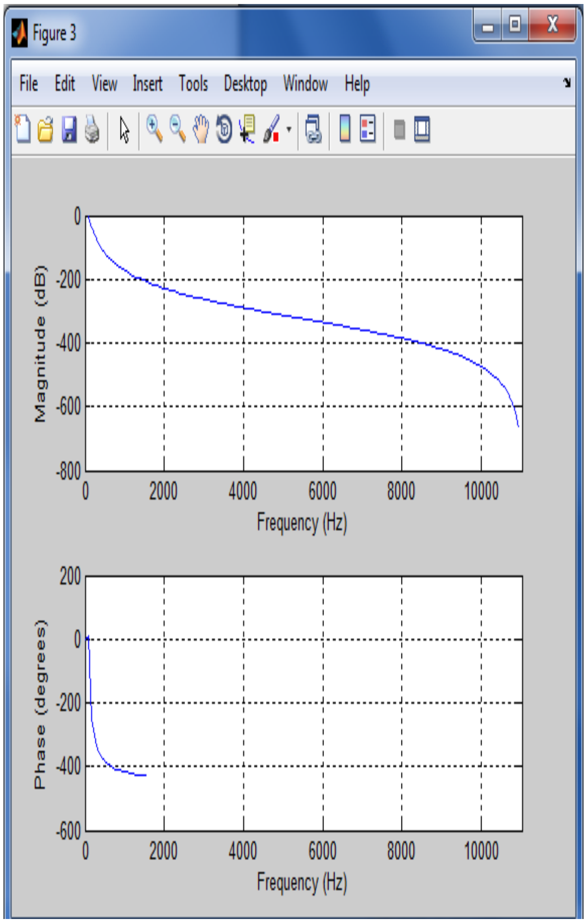


Fig. 5.7 Frequency Response

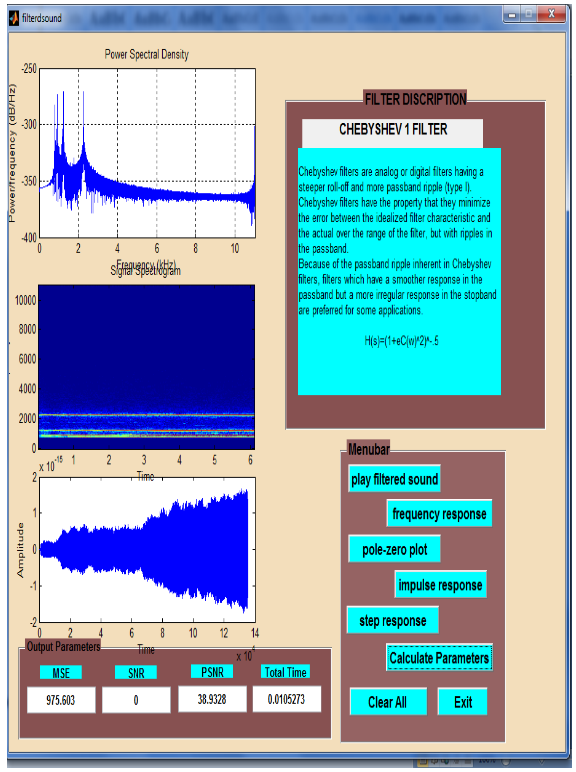


Fig. 5.9 Chebyshev filter Output

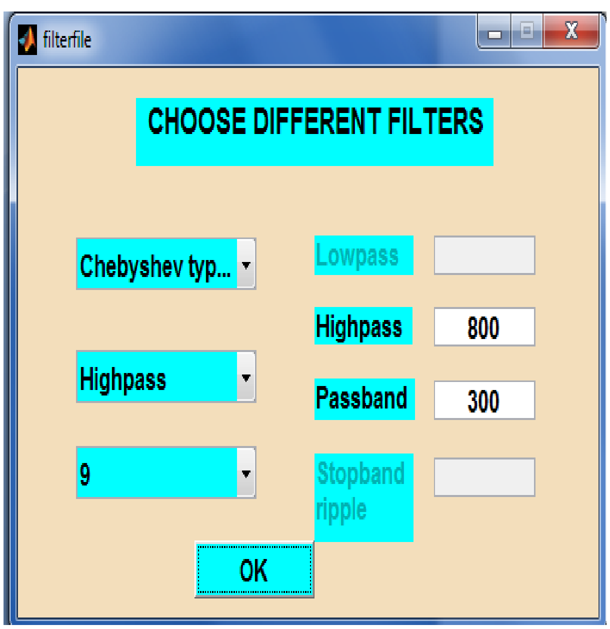


Fig. 5.8 Choose Filter type

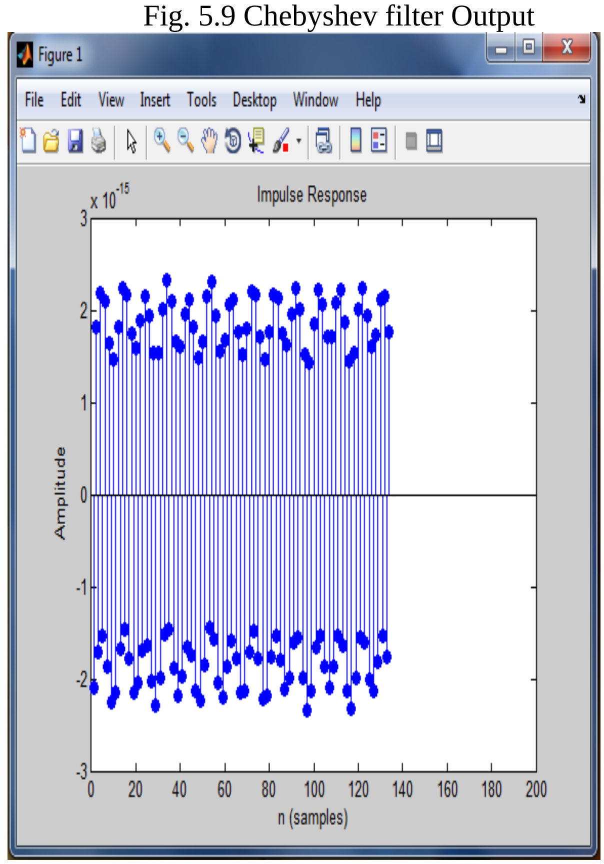


Fig.5.10 Impulse Response

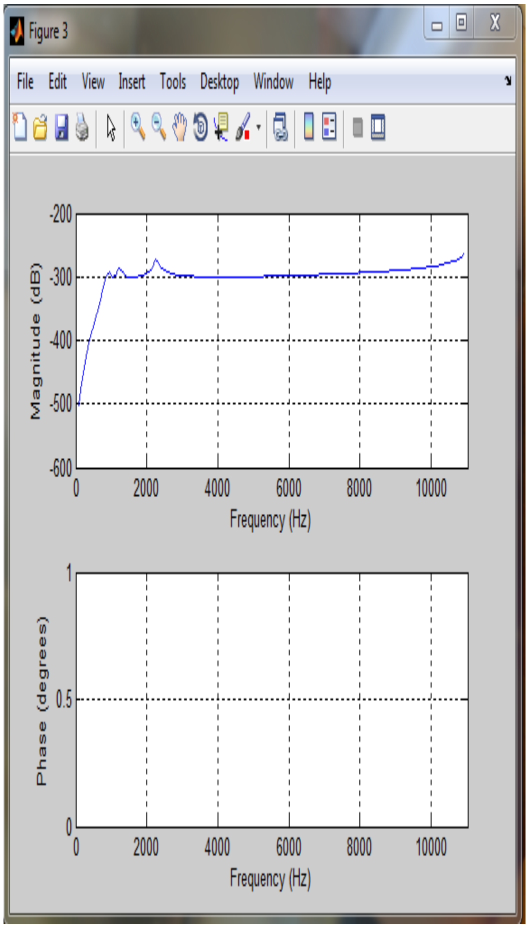


Fig.5.11 Frequency response

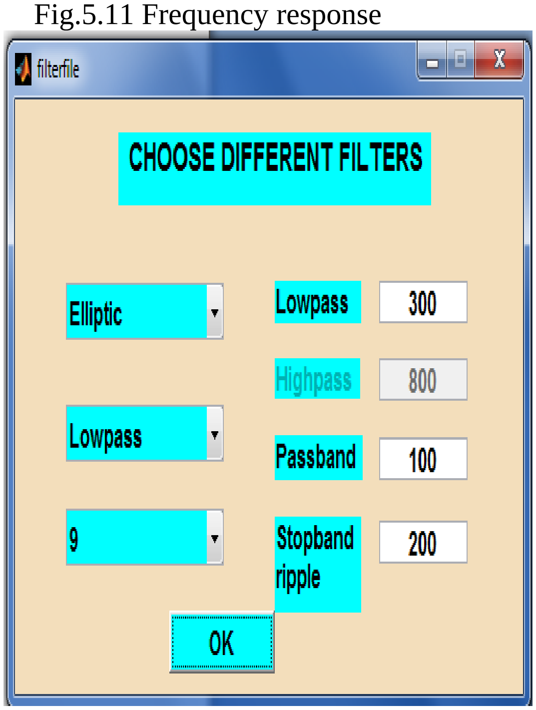


Fig. 5.12 select filter type

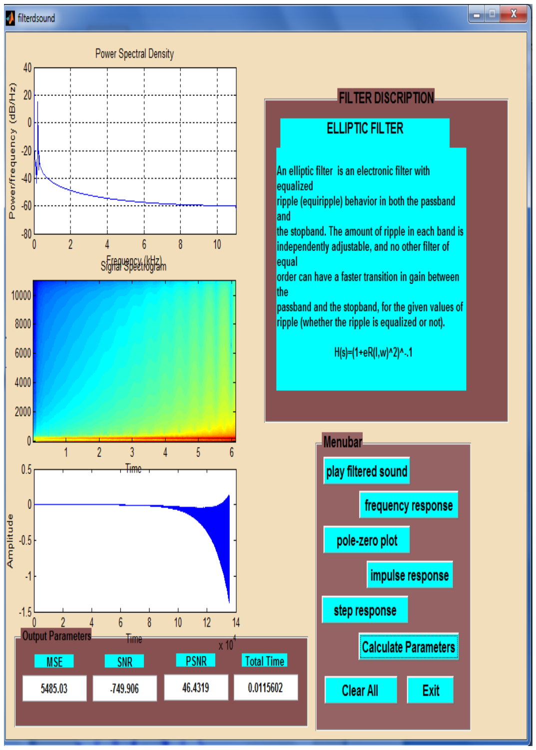


Fig. 5.13 elliptic filter Output

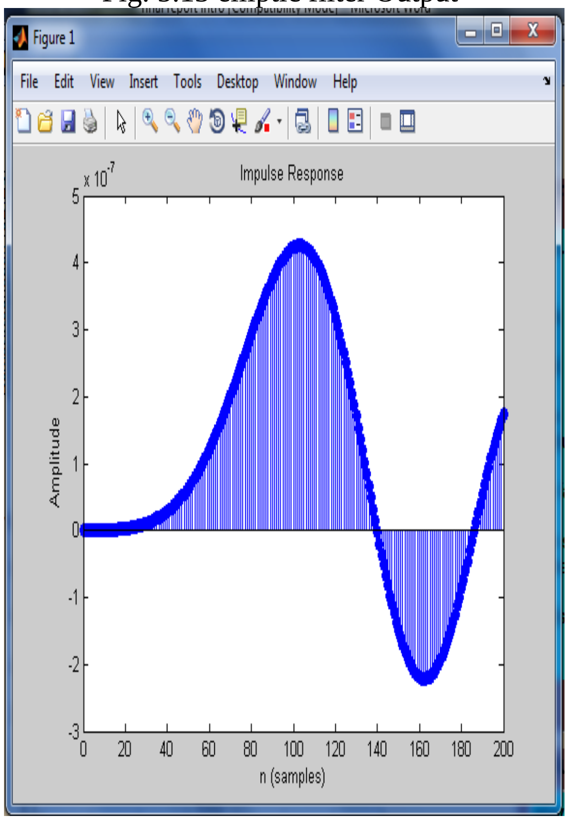


Fig. 5.14 Impulse Response

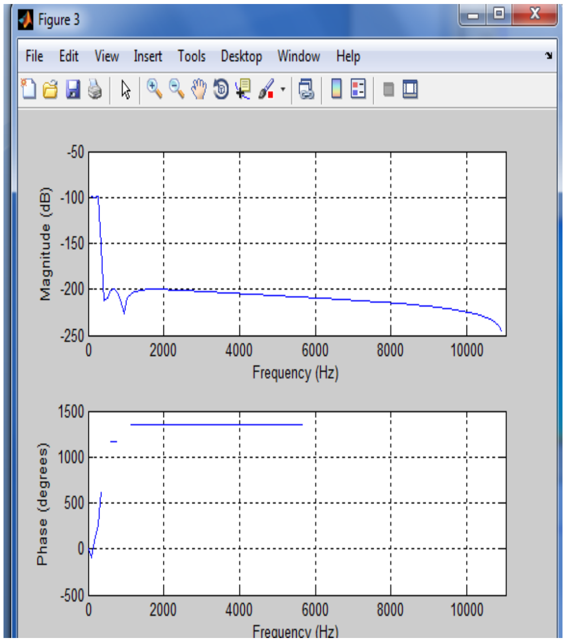


Fig. 5.15 Frequency Response

Table 5.1: RESULTS

PSNR value for different filters

Name of signal	**Butterworth filter(9th order)**	**Chebyshev Type-1 Filter (9th order)**	**Elliptic Filter (9th order)**
Music1.wav	43.0126	38.9328	46.4319
Airport.wav	36.2884	33.4008	33.4008
1N.wav	50.4234	50.4235	50.4236

SNR value for Different Filters

Name of Signal	**Butterworth Filter(9th order)**	**Chebyshev Filter(9th order)**	**Elliptic Filter (9th order)**
Music1.wav	-407.98	0.000000000	-749.206
Airport.wav	-178.829	-0.000119208	-123.764
1N.wav	0.0084448	-.00010493	0

MSE value for Different Filters

Name of Signal	**Butterworth Filter(9th order)**	**Chebyshev Filter(9th order)**	**Elliptic Filter (9th order)**
Music1.wav	2496.05	975.603	5485.03
Airport.wav	103.219	1407.14	1407.13
1N.wav	52764.7	52765.8	52765.8

Total Time Elapsed value for different filter

Name of Signal	**Butterworth Filter (9th order)**	**Chebyshev Filter(9th order)**	**Elliptic Filter (9th order)**
Music1.wav	0.0100617	0.01055273	0.0115602
Airport.wav	0.0876448	0.00653539	0.00636064
1N.wav	0.154365	0.0174491	0.018838

TABLE 5.2 : RESULTS

PSNR value for different filters

Name of signal	**Butterworth filter(10th order)**	**Chebyshev Type-1 Filter (10th order)**	**Elliptic Filter (10 th order)**
Music1.wav	39.236	38.9328	788.824
Airport.wav	106.764	34.5001	33.4008
1N.wav	51.9849	50.4235	51.5646

SNR value for Different Filters

Name of Signal	**Butterworth Filter(10 th order)**	**Chebyshev Filter(10th order)**	**Elliptic Filter (10th order)**
Music1.wav	-30.3149	0.01267	-74989.1
Airport.wav	-193.492	0.00000	0.000153767
1N.wav	-156.136	0.000000001579	-114.112

MSE value foe Different Filters

Name of Signal	Butterworth Filter(10th order)	Chebyshev Filter(10th order)	Elliptic Filter (10th order)
Music1.wav	1046.14	975.603	951390.00
Airport.wav	106.764	68.38002	1407.14
1N.wav	75594.2	52765.8	68622.1

Total Time Elapsed value for different filter

Name of Signal	Butterworth Filter (10th order)	Chebyshev Filter(10th order)	Elliptic Filter (10th order)
Music1.wav	0.0103228	0.0113433	0.010499
Airport.wav	0.0485549	0.0088586	0.00649714
1N.wav	0.00865536	0.0174043	0.015991

TABLE 5.3: RESULTS

PSNR value for different filters

Name of signal	Butterworth filter(11th order)	Chebyshev Type-1 Filter (11th order)	Elliptic Filter (11 th order)
Music1.wav	39.0467	38.9382	570.85
Airport.wav	36.5367	34.5001	802.723
1N.wav	52.8252	50.4235	50.4236

SNR value for Different Filters

Name of Signal	Butterworth Filter(11th order)	Chebyshev Filter(11th order)	Elliptic Filter (11th order)
Music1.wav	-11.387	-0.000276709	-53195.7
Airport.wav	-203.665	0	-77522.3
1N.wav	-240.171	0.0000	-0.0030384

MSE value for Different Filters

Name of Signal	Butterworth Filter (11th order)	Chebyshev Filter (11th order)	Elliptic Filter (11th order)
Music1.wav	1001.52	975.603	153095.0056
Airport.wav	109.294	68.38002	68.3502
1N.wav	91732.5	52765.8	52767.8

Total Time Elapsed value for different filter

Name of Signal	Butterworth Filter (11th order)	Chebyshev Filter(11th order)	Elliptic Filter (11th order)
Music1.wav	0.0104597	0.0100501	0.010722
Airport.wav	0.01488845	0.0641522	0.076181
1N.wav	0.0159895	0.0156152	0.0118814

TABLE 5.4

COMPARISON TABLE:

Name of signal	Butterworth Filter(PSNR)	Chebyshev Filter (PSNR)	Elliptic Filter(PSNR)
Music1.wav	E= 36.2673	E=33.3970	E=33.4022
	P=43.0126	P=38.9328	P=46.4319
	P=39.236	P=38.9321	P=788.824
	P=39.0467	P=38.9382	P=570.85
Airport.wav	E= 32.2801	E= 30.8981	E= 30.8912
	P=36.2884	P=33.4008	P=33.4008
	P=106.764	P= 34.5001	P=33.4008
	P=36.5367	P=34.5001	P=802.723
1N.wav	E=35.8952	E= 34.4867	E= 34.4881

	P=50.4234	P=50.4235	P= 50.4236
	P=51.9849	P=50.4235	P= 51.5646
	P=52.8252	P=50.4235	P= 50.4236

Audio Noise Reduction using DWT:

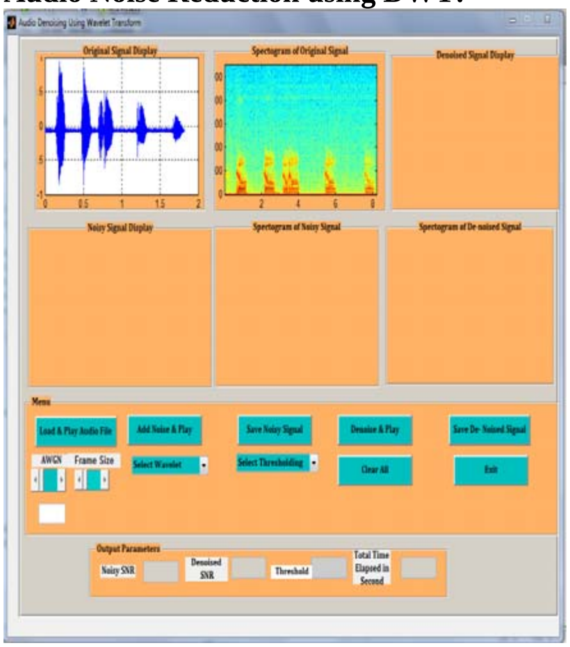


Fig. 5.16 Load the original signal

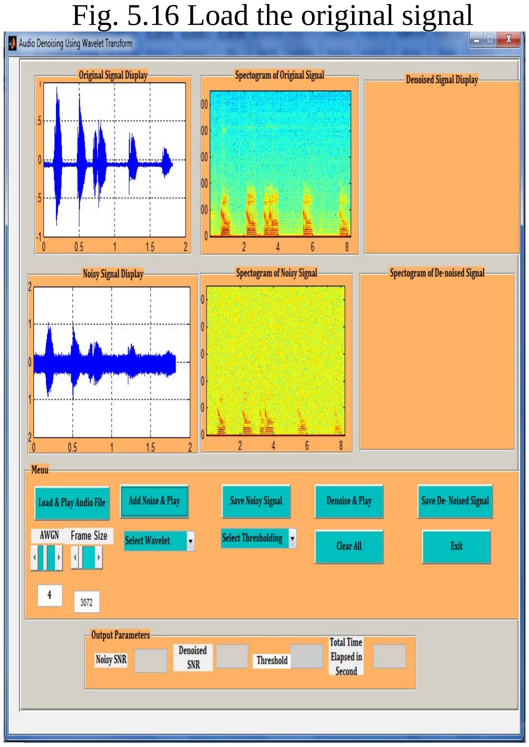


Fig.5.17 Play the noisy signal by adding noise

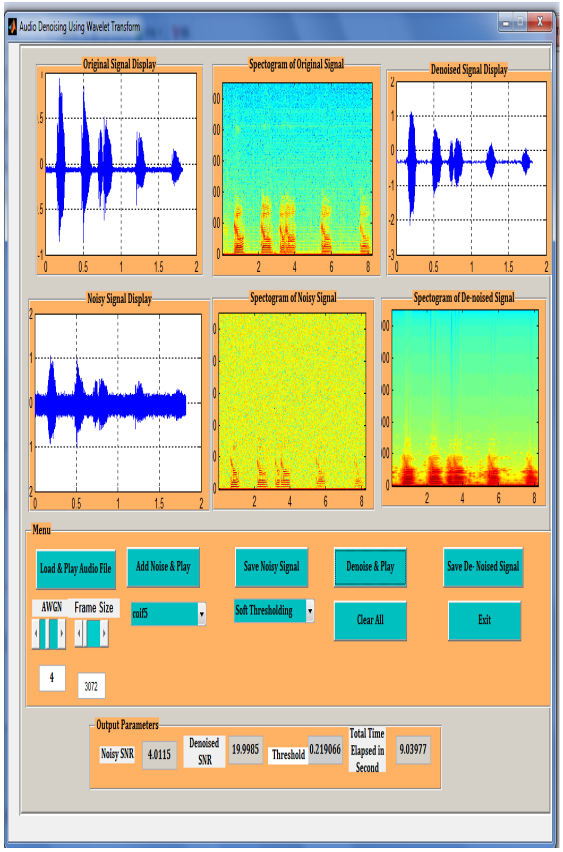


Fig. 5.18 Denoised signal

TABLE 5.5: RESULTS SOFT THRESHOLDING

Sr. No.	Signal name	Types of wavelets	Noisy SNR	Denoised SNR	Threshold	Total time Elapsed
1.	123.wav	Coif 5	4.0115	19.9985	0.219066	9.03977
2.	Music.wav	Db 9	4.00741	20.1332	0.265959	8.91066
3.	Music1.wav	Sym 4	4.00221	22.7372	0.35509	21.6675

TABLE 5.6: RESULTS
HARD THRESHOLDING

Sr. No.	Signal Name	Types of wavelets	Noisy SNR	Denoised SNR	Threshold	Total time Elapsed
1.	123.wav	Coif 5	4.00529	19.4761	0.230166	8.15582
2.	Music .wav	Db 9	4.00422	18.9089	0.174128	6.33789
3.	Music 1.wav	Sym 4	4.01249	18.8797	0.172271	6.35753

TABLE 5.7
COMARISON TABLE: SOFT THRESHOLDING

Sr. No.	Signal name	Denoised SNR
1.	123.wav	E=7.8471
		P=19.9985
2.	Music.wav	E=7.1884
		P=20.1332
3.	Music1.wav	E=7.6209
		P=22.7372

TABLE 5.8
HARD THRESHOLDING

Sr. No.	Signal name	Denoised SNR
1.	123.wav	E= 6.3363
		P=19.4761
2.	Music.wav	E=6.8802
		P=18.9089
3.	Music1.wav	E=6.6905
		P=18.8797

VI. CONCLUSION

This chapter concludes the work in this thesis in terms of the various parameters that have been considered while denoising audio using various methods of filters ad DWT. Here it also provides with a look up in the future scope of our work area.

The different filters with different frequencies are used to remove noise. It can be concluded that for different center frequencies, order of the filter always remains same. By changing its center frequencies filters are being tuned to different frequencies. We have designed Butterworth filter, Chebyshev filter type-1 and elliptic filter to remove the noise. We have designed these filters with different type's i.e. low pass, high pass, band pass and band reject and the order of these filters are same. By using this we can get the better results of de-noising, especially for low level noise. During different analysis we found that soft thresholding is better than hard thresholding because soft thresholding gives better results than hard thresholding. Higher threshold removes noise well, but the part of original signal is also removed with the noise. It is generally not possible to filter out all the noise without affecting the original signal. We can analyze the denoised signal by signal to noise ratio (SNR), mean square error (MSE), Threshold values and elapsed time analysis.

FUTURE WORK

Future work might involve a real time implementation of the system so that the maximum noise is reduced form the audio signals and videos. In the future anybody can extent the order of the different filters and works on higher amplitude signals. They can calculate the efficiency of the filters that they have to implement. In the DWT we are using coif and sym4 with hard and soft threshold but in the future different types of wavelet is implemented with different types of thresholding techniques or hybrid techniques is designed with the help of filters and wavelets and thresholding techniques. Other things in future the results may be improved in the filters and DWT technique.

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[1] Shalu Rani, Rajbhupinder Kaur "REVIEW: AUDIO NOISE REDUCTION USING FILTERS AND DISCRETE WAVELET TRANSFORMATION**"** Journal of The International Association of Advanced Technology and Science (JIAATS), ISSN-5563-1682,Vol. 16 | June 2015