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Paper on Frequency based audio Noise Reduction using Butter Worth, Chebyshev & Elliptical Filters

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Abstract :- Noise from the audio signal is removed by using Audio Noise Reduction System. Audio noise reduction systems uses filters for removal of noise. Filters are the manipulation of the amplitude and/or phase response of a signal according to their frequency. These are the basic components of all signal processing and -telecommunication systems. There are two kinds of filters- fixed and tunable. Fixed filters are those in which passband frequencies and stopband frequencies are fixed whereas in case of tunable filters, passband and stopband frequencies are variable. These frequencies can be changed according to the requirement of the applications. Tunable digital filters are widely employed in telecommunications, medical electronics, digital audio equipment and control systems. This is the basic need for removal of noise from the audio signal.

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I. INTRODUCTION

Audio noise reduction system is the system that is used to remove the noise from the audio signals. Audio noise reduction systems can be divided into two basic approaches. The first approach is the complementary type which involves compressing the audio signal in some well-defined manner before it is recorded (primarily on tape). The second approach is the single-ended or non-complementary type which utilizes techniques to reduce the noise level already present in the source material—in essence a playback only noise reduction system [18]. This approach is used by the LM1894 integrated circuit, designed specifically for the reduction of audible noise in virtually any audio source. Noise reduction is the process of removing noise from a signal. All recording devices, both analogue or digital, have traits which make them susceptible to noise. Noise can be random or white noise with no coherence, or coherent noise introduced by the device's mechanism or processing algorithms. Their is a Active noise control (ANC), also known as noise cancellation, or active noise reduction (ANR), is a method for reducing unwanted and unprocessed sound by the addition of a second sound specifically designed to cancel the first[26]. Sound is a pressure wave or we can say sound is the analog signals that are processed according to their frequency, which consists of a compression phase and a rarefaction phase. A noise-cancellation speaker emits a sound wave with the same amplitude but with inverted phase (also known as anti phase) to the original sound. The waves combine to form a new wave, in a process called interference, and effectively cancel each other out - an effect which is called phase cancellation.

Modern active noise control is generally achieved through the use of analog circuits or digital signal processing. An Adaptive algorithms are designed to analyze the waveform of the background no neural noise, then based on the specific algorithm generate a signal that will either phase shift or invert the polarity of the original signal. This anti phase is then amplified and a transducer creates a sound wave directly proportional to the amplitude of the original waveform, creating destructive interference [8]. This effectively reduces the volume of the perceivable noise. The transducer emitting the noise cancellation signal may be located at the location where sound attenuation is wanted (e.g. the user's ear/any music/headphone sound). This requires a much lower power level for cancellation but is effective only for a single user.

Types of Noises:

There are many types and sources of noise or distortions and they include:

1. Electronic noise such as thermal noise and shot noise,
2. Acoustic noise emanating from moving, vibrating or colliding sources such as revolving
Machines, moving vehicles, keyboard clicks, wind and rain,
3. Electromagnetic noise that can interfere with the transmission and reception of voice, image and data over the radio-frequency spectrum,
4. Electrostatic noise generated by the presence of a voltage,
5. Communication channel distortion and fading and
6. Quantization noise and lost data packets due to network congestion.

Signal distortion is the term often used to describe a systematic undesirable change in a signal and refers to changes in a signal from the non-ideal characteristics of the communication channel, signal fading reverberations, echo, and multipath reflections and missing samples [10]. Depending on its frequency, spectrum or time characteristics, a noise process is further classified into several categories:

1. **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.
2. **White noise:** purely random noise has an impulse autocorrelation function and a flat power spectrum. White noise theoretically contains all frequencies in equal power.
3. **Colored 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.
4. **Narrowband noise:** It is a noise process with a narrow bandwidth such as 50/60 Hz from the electricity supply.

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5. **Transient noise pulses:** Consist of relatively long duration noise pulses such as clicks, burst noise etc.
6. **Impulsive noise:** Consists of short-duration pulses of random amplitude, time of occurrence and duration.

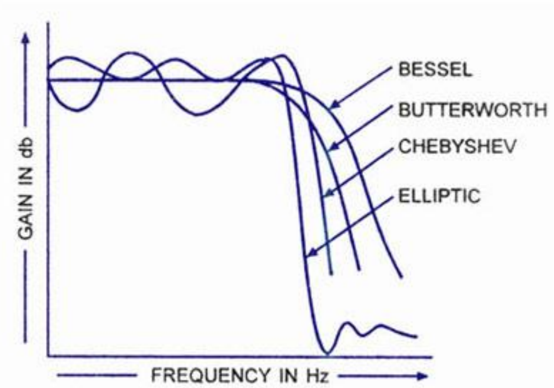
II. Active vs. passive noise control

Noise control is an active or passive means of reducing sound emissions, often for personal comfort, environmental considerations or legal compliance. Active noise control is sound reduction using a power source. Passive noise control is sound reduction by noise-isolating materials such as insulation, sound-absorbing tiles, or a muffler rather than a power source [1].

Active noise canceling is best suited for low frequencies. For higher frequencies, the spacing requirements for free space and zone of silence techniques become prohibitive. In acoustic cavity and duct based systems, the number of modes grows rapidly with increasing frequency, which quickly makes active noise control techniques unmanageable. Passive treatments become more effective at higher frequencies and often provide an adequate solution without the need for active control [3].

III. FILTER

Filters are networks that process signals in a frequency-dependent manner. The basic concept of a filter can be explained by examining the frequency dependent nature of the impedance of capacitors and inductors [17]. Filters have many practical applications. A simple, single-pole, low-pass filter (the integrator) is often used to stabilize amplifiers by rolling off the gain at higher frequencies where excessive phase shift may cause oscillations. A simple, single-pole, high-pass filter can be used to block dc offset in high gain amplifiers or single supply circuits[8]. Filters can be used to separate signals, passing those of interest, and attenuating the unwanted frequencies. There are a large number of texts dedicated to filter theory. An ideal filter will have an amplitude response that is unity (or at a fixed gain) for the frequencies of interest (called the pass band) and zero everywhere else (called the stop band). The frequency at which the response changes from passband to stopband is referred to as the cutoff frequency[19].



Frequency Response Curve

IV. LITERATURE SURVEY

Previous Research work on **"Audio Noise Reduction Using Butter Worth, Elliptical & Chebyshev Filters"** in the literature survey starting from 2010 to November 2013 was studied which helped me to complete my work and enhance my knowledge. I studied so many papers and some of them are given below:

C Mohan Rao, Dr. B Stephen Charles, Dr. M N Giri Prasad[2013] have presents a new adaptive filter whose coefficients are dynamically changing with an evolutionary computation algorithm and hence reducing the noise. This algorithm gives a relationship between the update rate and the minimum error which automatically adjusts the update rate. When the environment is varying, the rate is increased while it would be decreased when the environment is stable and the computation complexity of adaptive filter can be significantly reduced. In the simulation, additive white Gaussian noise is added to the randomly generated information signal and efficiently reduced this noise with minimum or no error by using evolutionary computation with Least Mean Square (LMS) algorithms. Adaptive Noise Cancellation is an alternative way of cancelling noise present in a corrupted signal [18].

K.P. Obulesu1 P. Uday Kumar [2013] have studied the audio signals are synthetic signals, in which music or speech, are often corrupted by noise during recording and transmission. Speech enhancement is a long standing problem with numerous applications ranging from hearing aids, to coding and automatic recognition of speech signals etc. and assume that the noise is additive and statistically independent of the signal. Audio denoising procedures are designed to attenuate the noise and retain the signal of interest. Reduction of noise from audio signals has two methods, Diagonal & Non Diagonal audio denoising algorithms. In this paper, Non diagonal method is used in which Block parameters are automatically adjusted to the nature of the audio signal by minimizing a Stein estimator which is calculated analytically from noisy signal values. This Block thresholding method eliminates "musical noise" by grouping Time-frequency coefficients in blocks before being attenuated[15].

Matheel E. Abdulmunim, Rabab F. Abass [2013] have presented the digital videos are often corrupted by a noise during the acquisition process, storage and transmission. It made the video in ugly appearance and also affect on another digital video processes like compression, feature extraction and pattern recognition so video denoising is highly desirable process in order to improve the video quality. There are many transformation for denoising process, one of them are Fast Discrete Wavelet Transform(FDWT) and framelet transform (Double-Density Wavelet Transform) which is a perfect in denoising process by avoiding the problems in the other transformations. In this paper we propose a method named Translation Invariant with Wiener filter (TIW) this method is proposed to solve the shift variance problem and use this method to denoised a noisy video with Gaussian white noise type. It is applied 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)[1].

Raghavendra Sharma, Vuppuluri Prem Pyara [2013] 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 [21].

S. N. Sampat, Dr. C. H. Vithalani [2013] have presented the denoising of one dimensional signal using threshold is one of the major applications of wavelet transform. Quadrature Mirror Filter bank method of wavelet transform has many advantages like support for all major orthogonal wavelets, dyadic resolution and adequate retention of energy. Determination of threshold type and threshold value is one of the important tasks in threshold based denoising techniques. Denoising of audio signal is a subjective matter and remains a valid challenge. In this paper, a noisy speech wav file having additive white Gaussian noise is used for denoising to demonstrate features of two stage hard threshold, soft threshold and customized threshold denoising using Quadrature Mirror Filter bank method of wavelet transform . The second stage of denoising uses neighborhood concept where in a set of three wavelet coefficients, threshold is applied to any wavelet coefficient on the bases of the value of the other two neighborhood wavelet coefficients. Eight different denoised files are generated. Various parameters are measured and compared [19].

B. JaiShankar and K. Duraiswamy [2012] have introduced the noises present in communication channels are disturbing and the recovery of the original signals from the path without any noise is very difficult task. This is achieved by denoising techniques that remove noises from a digital signal. Many denoising technique have been proposed for the removal of noises from the digital audio Signals. But the effectiveness of those techniques is less. In this paper, an audio denoising technique based on wavelet transformation is proposed [8].

B. Jai Shankar, K.Duraiswamy [2012] have presented the noises present in signals are difficult to recover using the traditional methods. Now wavelet transform is used for denoising techniques. The thresholding both hard and soft are used in wavelet transform. The technique exposes each and every finest details contributed by the grouped set of blocks and also it protects the vital and unique features of every individual block. The blocks are filtered and replaced in their original positions from where they are detached. Their implementation results reveal that the proposed technique achieves a state-of-the-art denoising performance in terms of signal-to-noise ratio[9].

Eric Martin [2012] have introduces an adaptive audio block thresholding algorithm. The denoising parameters are computed according to the time-frequency regularity of the audio signal using the SURE (Stein Unbiased Risk Estimate) theorem. The author studied unlike the diagonal estimators, the adaptive audio block thresholding algorithm based on a non-diagonal estimator is very much elective with white noise. However there are some defects. The sounds which are like a white Gaussian noise will be deleted. For instance, it's impossible to hear cymbals from a drum kit after a denoising [12].

J. Jebastine, Dr. B. Sheela Rani [2012] In this paper the author describes the development of an adaptive noise cancellation algorithm for effective recognition of speech signal and also to improve SNR for an adaptive step size input. An adaptive filter with Fast Block Least Mean square Algorithm is designed for noise free audio (speech/music) signals. The signal input used is a audio speech signal which could be in the form of a recorded voice. The filter used is adaptive filter and the algorithm used is Fast Block LMS algorithm. A Gaussian noise is added to this input signal and given as a input to the Fast Block LMS [10].

Kai Siedenburg, Monika D¨orfler [2012] in this paper the author considers the denoising problem from the viewpoint of sparse atomic representation. A general framework of time-frequency soft-thresholding is proposed which encompasses and connects well-known shrinkage operators as special cases. In particular, the ground breaking idea of exploiting signal sparsity in the framework of redundant representations is extended to incorporate knowledge about structural properties of the observed signals. Convergence of the corresponding algorithms is numerically evaluated and their performance in denoising real-life audio signals is compared to the results of similar existing approaches. The novel approach is competitive with respect to signal to noise ratio and improves the state of the art in terms of perceptual criteria[23].

Rajeev Aggarwal, Jai Karan Singh Vijay, Kumar Gupta [2011] In this paper the author describe the Discrete-wavelet transform (DWT) based algorithm are used for speech signal denoising. Here both hard and soft thresholding are used for denoising. Analysis is done on noisy speech signal corrupted by babble noise at 0dB, 5dB, 10dB and15dB SNR levels. Output SNR (Signal to Noise Ratio) and MSE(Mean Square Error) is calculated & compared using both types of thresholding methods. Soft thresholding method performs better than hard thresholding at all input SNR levels. Hard thresholding shows a maximum of 21.79 dB improvement whereas soft thresholding shows a maximum of 35.16 dB improvement in output SNR[2].

Romain Serizel, Marc Moonen [2010] – has presented the combined active noise control and noise reduction schemes for hearing aids to tackle secondary path effects and effects of noise leakage through an open fitting. Such leakage contributions affect the noise signals. The result of these signals appears to have a non-negligible impact on the final signal-to-noise ratio. The author studied a noise-reduction algorithm and an active noise control system in cascade may be efficient as long as the causality margin of the system is large enough. A Filtered-x Multichannel Wiener Filter is presented and applied to integrate noise reduction and active noise control. The cascaded scheme and the integrated scheme are compared experimentally with a Multi channel Wiener Filter in a classic noise reduction framework without active noise control, where the integrated scheme is found to provide the best performance [21].

Guoshen Yu, Stéphane Mallat [2008] have studied the removing noise from audio signals requires a non diagonal processing of time-frequency coefficients to avoid producing "musical noise." Non diagonal time-frequency estimators are more effective than diagonal estimators to remove noise from audio signals because they introduce less musical noise. These non diagonal estimators are derived from a time-frequency SNR estimation performed with parameterized filters applied to time-frequency coefficients. This paper introduces an adaptive audio block-thresholding algorithm that adapts all parameters to the time-frequency regularity of the audio signal. The adaptation is performed by minimizing a Stein unbiased risk estimator calculated from the data [26].

V. PROBLEM DEFINITION

The problem undertaken for the dissertation is ***"Audio Noise reduction using Butterworth, Chebyshev & Elliptical filters"***. 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.

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, different filters are used to denoised the audio signals and enhance speech and audio signal quality.

This project entitled ***"Audio Noise reduction using Butterworth, Chebyshev & Elliptical filters"*** 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 Techniques.
- ➢ To Implement the Exiting Various Techniques studied as in literature review such as Butter worth filter.
- ➢ Study and Analyze the Results Being Obtained such as frequency, time signal and phase angle.
- ➢ Noise reduction technology is aimed at reducing unwanted ambient sound, and is implemented through different methods.

VI. OBJECTIVES OF WORK

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

VII. METHODOLOGY

This dissertation is 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. 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, pass-band frequency, stop-band frequency, pass-band 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.

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VIII. RESULTS

This shows the implementation results of the dissertation work.Their are different figures that shows how the signals are processed and how the system tools works in MATLAB.

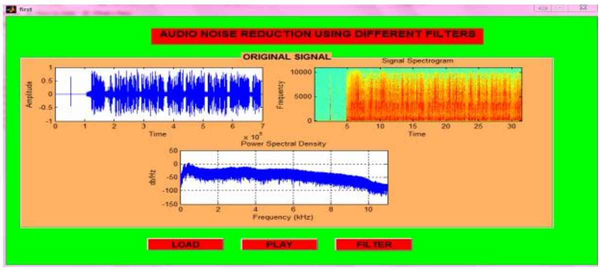

Figure 5.1: Initial window for uploading the wave signal

The figure 5.1 is used to upload the original wave signal that we wanted to enhance. In this window signal is displayed amplitude vs. time and frequency vs. time and noise vs. frequency. The wave signal is plotted on different axes.

Figure 5.2: Butterworth filter with low pass frequency and 12th orders

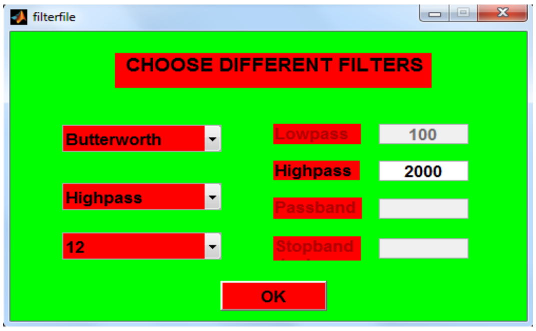

Figure 5.3: Butterworth filter with high pass frequency and 12th orders

The figure 5.2 and figure 5.3 shows the Butterworth filter with low and high frequency with 12th order to enhance the noisy signals. In these windows different value of frequency with different order is used for enhancing the signals.

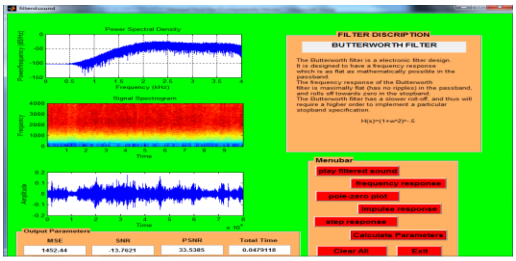

Figure 5.4: Shows the butter worth filter output

Figure 5.5 shows the Butterworth filter output when we are using the high frequency to enhance the signals. In this window the output is shown in the form of graphs and signals. Here the signal is plotted power spectrum vs. frequency and frequency vs. time and amplitude vs. time.

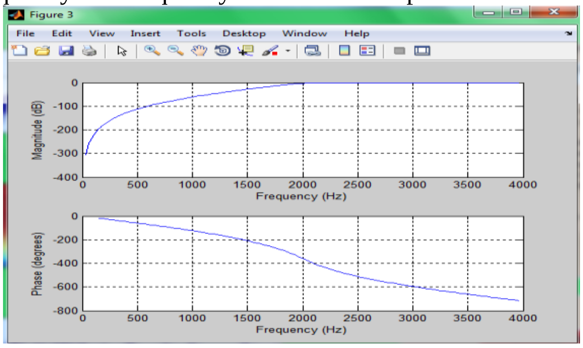

Figure 5.5: Frequency response for butter worth filter output

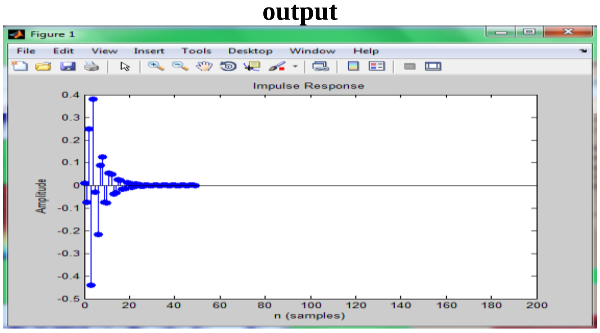

Figure 5.6: Impulse response for butter worth filter output

Figure 5.5 and figure 5.6 shows the frequency response and Impulse response for butter worth filter, when we are using the Butterworth filter with low frequency and high frequency with 12th order degree.

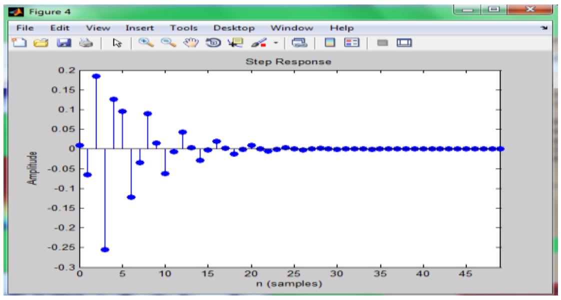

Figure 5.7: Step response for butter worth filter output with high frequency

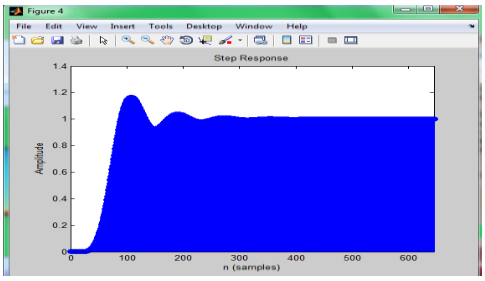

Figure 5.8: Step response for butter worth filter output with low frequency

Figure 5.7, figure 5.8 shows the step response for butter worth filter, when we are using the Butterworth filter with low frequency and high frequency with 12th order degree.

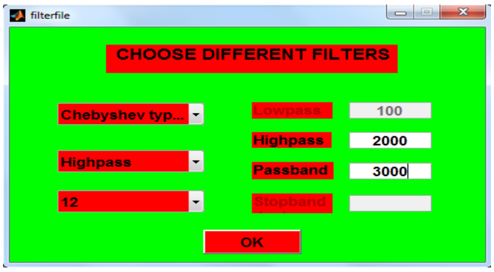

Figure 5.9: Chebyshev type-1 filter with high frequency and 12th orders

The Figure 5.9. Shows the Chebyshev type-1 filters with band stop frequency and 8th order to enhance the noisy signals. In these windows low pass, high pass and band pass frequency is used to enhance the signals.

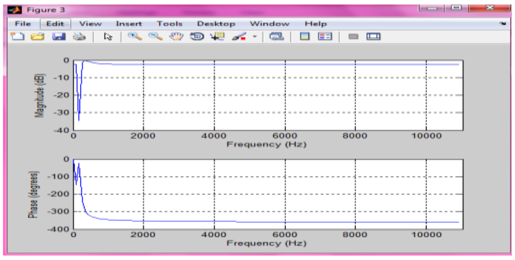

Figure 5.10: Frequency response for Chebyshev type-1 filter with 12th orders

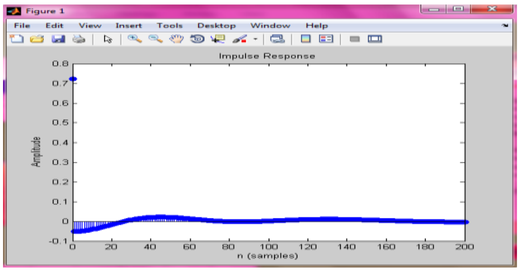

Figure 5.11: Impulse response for Chebyshev type-1 filter with 12th orders

The Figure 4.10 and figure 4.11 Shows the frequency response and impulse response for Chebyshev type-1 filters with 12th order to enhance the noisy signals. In these signal is plotted with phase and amplitude and time according to the input frequency for audio signals.

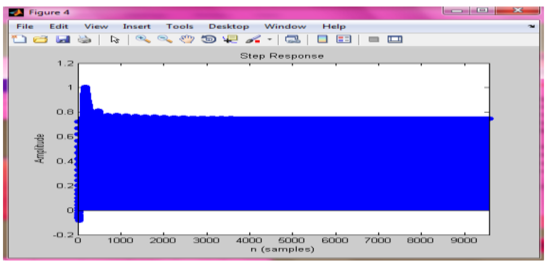

Figure 5.12: Step response for Chebyshev type-1 filter with 12th orders

Figure 5.12 shows the step response Chebyshev type-1 filter with 12th orders when we are using the Chebyshev type-1 filter with low frequency ,high frequency and band pass with 12th order degree.

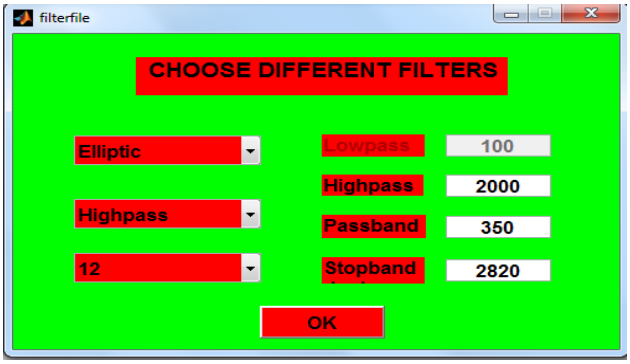

Figure 5.13: Elliptic filter with high pass frequency and 12th order

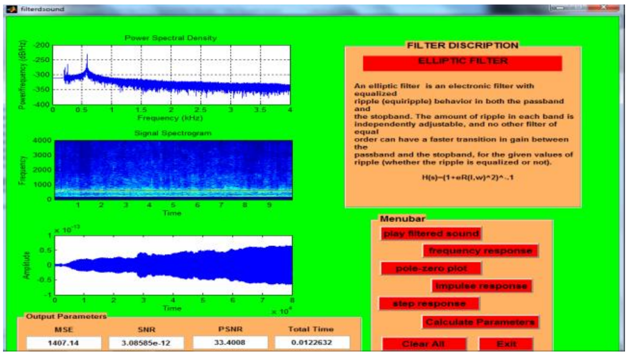

Figure 5.14: Shows the Elliptic filter output

Figure 5.14 shows the Elliptic filter output when we are using the high frequency is used to enhance the signals. In this window the output is shown in the form of graphs and signals. Here the signal is plotted power spectrum vs. frequency and frequency vs. time and amplitude vs. time.

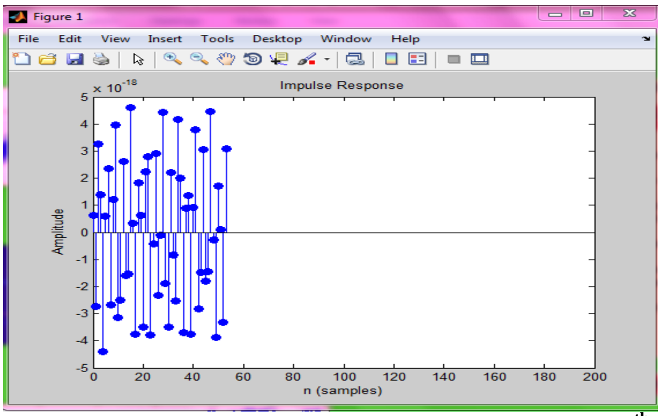

Figure 5.15: Impulse response for Elliptic filter with 12th orders

Figure 5.15 shows the impulse response for elliptic filter with 12th orders when we are using the high frequency to enhance the signals.

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Table 5.1 : PSNR value for different Filter

Name of Signal	Butterworth Filter (12th order)	Chevshevby Type-1 Filter (12th order)	Elliptic Filter (12th order)
1N.wav	39.6232	34.5001	34.5001
2N.wav	47.6887	34.5001	34.5001
3N.wav	34.813	34.5001	34.5001

Table 5.2 : SNR value for different Filter

Name of Signal	Butterworth Filter (12th order)	Chevshevby Type-1 Filter (12th order)	Elliptic Filter (12th order)
1N.wav	-512.309	0	0
2N.wav	-1318.86	5.09375	1.14292
3N.wav	-31.2863	0	0

Table 5.3 : MSE value for different Filter

Name of Signal	Butterworth Filter (12th order)	Chevshevby Type-1 Filter (12th order)	Elliptic Filter (12th order)
1N.wav	222.453	68.3802	68.3802
2N.wav	1424.93	68.3802	68.3802
3N.wav	73.488	68.3802	68.3802

Table 5.4 : Total Time Elapsed value for different Filter

Name of Signal	Butterworth Filter (12th order)	Chevshevby Type-1 Filter (12th order)	Elliptic Filter (12th order)
1N.wav	0.0166216	0.00898936	0.00970516
2N.wav	0.0260432	0.00871712	0.0276344
3N.wav	0.016078	0.0157941	0.00822272

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