ADAPTIVE FILTER FOR NOISE CANCELLATION IN AUDIO SIGNALS

Project Report submitted in partial fulfilment of the requirement for the degree of

Bachelor of Technology

in

Electronics and Communication Engineering

By

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Certificate

This is to certify that the project report entitled "Adaptive Filter For Noise Cancellation In Audio Signals", submitted by Shalini Sharma and Shruti Sharma in partial fulfillment for the award of degree of Bachelor of Technology in Electronics and Communication Engineering to Jaypee University of Information Technology, Waknaghat, Solan has been carried out under my supervision.

Signature

Date : 20-05-2010

Dr. Vinay Kumar

Sr. Lecturer

Certified that this work has not been submitted partially or fully to any other University or Institute for the award of this or any other degree or diploma

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ABSTRACT

The filters that are used to attenuate noise are usually optimized to perform extremely well when dealing with certain noise distributions. Unfortunately it is often the case that the noise corrupting the audio signal/data is not known especially if the noise is introduced while transmission due to atmospheric disturbances likes lightening etc. In such cases where the a priori of the noise corrupting the image is not available the adaptive filter plays a very important role. The adaptive filter analyzes the audio signal/data and updates its coefficients in order to minimize the noise. The adaptive filters are capable of performance far superior than the non-adaptive filters. However, the price paid for improved filtering power is an increase in filter complexity.

Commonly truncated Volterra series is used to model a large class of nonlinear systems and is particularly attractive in adaptive filtering applications because the expansion is a linear combination of nonlinear functions of the input signal. However, most of the practical applications of systems employing Volterra series expansions involve low order models whereas adaptive filters of any order can be modelled using Chebyshev polynomial. Moreover, adaptive filters based on Chebyshev polynomials yield better responses.

CHAPTER -1

ADAPTIVE FILTERS

1.1 Introduction

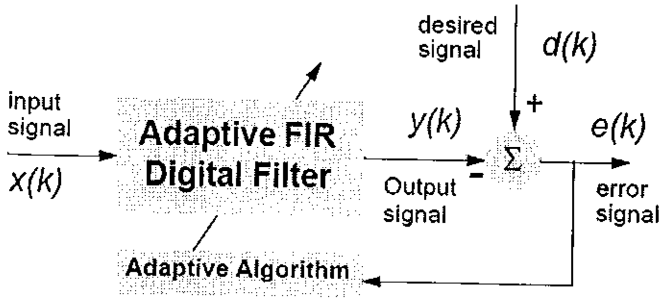
Filters are basically signal conditioners. Filters function by accepting an input signal, blocking prespecified frequency components, and passing the original signal minus those components to the output. However, some signal processing applications require that the system compensate for things outside of its direct influence. For example, when the noise in the signal is introduced while transmission, the a priori knowledge of the frequencies to be blocked is not available. This is where adaptive filtering comes into play. Adaptive filtering algorithms allow for modeling of the outside phenomena. Also, adaptive filtering can be used for analysis/synthesis of a system.

An adaptive filter is a filter that self-adjusts its transfer function according to an optimizing algorithm. Because of the complexity of the optimizing algorithms, most adaptive filters are digital filters that perform digital signal processing and adapt their performance based on the input signal. For some applications, adaptive coefficients are required since some parameters of the desired processing operation are not known in advance. In these situations it is common to employ an adaptive filter, which uses feedback to refine the values of the filter coefficients and hence its frequency response.

As the power of digital signal processors has increased, adaptive filters have become much more common and are now routinely used in devices such as mobile phones and other communication devices, camcorders and digital cameras, and medical monitoring equipment.

1.2 Block diagram

The idea behind the block diagram is that a variable filter extracts an estimate of the desired signal.The aim is to adapt the digital filter such that the input signal x(k) is filtered to produce y(k) which when subtracted from desired signal d(k), will minimise the power of the error signal e(k).



$$e(k) = d(k) - y(k)$$

$$y(k) = \text{Filter}(x(k))$$

In the block diagram, the input signal x(k) is the sum of a desired signal d(k) and the interfering noise v(k).

$x(k) = d(k) + v(k)$

The error signal or cost function is the difference between the desired and the estimated signal

$e(k) = d(k) - y(k)$

1.3Adaptive filter nomenclature

The adaptive filter can be FIR (non-recursive), IIR (recursive) or even a non-linear filter. However, most adaptive filters are FIR for reasons of algorithm stability and mathematical tractability.If the digital filter is FIR or all-zero, the adaptive system can also be called Moving Average or MA. If the digital filter is all-pole, the adaptive system can also be called Autoregressive or AR and if the digital filter is an IIR with zeros, the adaptive system can also be called ARMA.

1.4 Adaptive filter performance

The key aim of the adaptive filter is to minimise the error signal e(k). The success of this minimisation depends on the nature of the input signals, the length of the adaptive filter, and the adaptive algorithm used.

1.5 Difference between traditional digital filters and adaptive filters

The filter coefficients of a traditional digital filter do not change over time whereas, the coefficients of an adaptive filter change over time. Therefore, adaptive filters have a self-learning ability that traditional digital filters do not have.

1.6 Advantages of using adaptive filters

Compared to traditional digital filters, adaptive filters have the following advantages:

- Adaptive filters can complete some signal processing tasks that traditional digital filters cannot. For example, adaptive filters can be used to remove noise that traditional digital filters cannot remove, such as noise whose power spectrum changes over time.
- Adaptive filters can complete some real-time or online modelling tasks that traditional digital filters cannot. For example, adaptive filters can be used to identify an unknown system in online mode. Typically, adaptive filters are useful while performing real-time or online signal processing applications.

1.7 Applications of adaptive filters

Noise cancellation

Signal prediction

Adaptive feedback cancellation

Echo cancellation

CHAPTER -2

DESIGN OF CHEBYSHEV FIR FITER BASED ON ANTENNA THEORY APPROACH

In our project we have used Chebyshev polynomials for the realization of an adaptive filter. Chebyshev polynomials play an important role in antenna as well as in signal processing theory.The Dolph-Chebyshev distribution of currents feeding the elements of a linear array comprising an antenna gives a sharp main lobe and small side lobes all of which have the same power level.

Taking the case of a linear equispaced antenna array with n elements, labelled from left to right.

$$|E|=|A_0e^{j0} + A_1e^{j\psi} + A_2e^{j2\psi} + \ldots + A_{n-2}e^{j(n-2)\psi} + A_{n-1}e^{j(n-1)\psi}| \quad (1)$$

$$\psi = \beta d\cos\phi + \gamma \quad (2)$$

where,

$|E|$ =is the magnitude of far field,

$\beta = 2\pi/\lambda$,

λ is the free space wavelength,

d is the spacing between elements,

ϕ is the angle from the normal to the linear array,

γ is the progressive phase shift from left to right, and

$A_0, A_1, A_2, \ldots$ are complex amplitudes which are proportional to the current amplitudes.

If we substitute $z=e^{j\psi}$ and write Equation (1) as

$$H(z)=A_0 + A_1z + A_2z^2 + \ldots + A_{n-2}z^{n-2} + z^{n-1} \quad (3)$$

This equation represents an FIR filter .Where, H(z) is impulse response of the filter with $z=e^{j\psi}$.

$A_0, A_1, A_2, \ldots$ represents amplitudes at the corresponding frequencies.

Designing a non-adaptive FIR filter based on the antenna design.

The Chebyshev Polynomial is given by

$$T_m(x) = \cos(m\cos^{-1}x) \qquad 0<|x|<1$$

$$T_m(x) = \cosh(m\cosh^{-1}x) \qquad 1<|x| \qquad (4)$$

$$b=10^{(\text{attenuation in dB})/20} \qquad (5)$$

Taking 'm' as the order of the filter, the location of zeros, w_m, on unit circle can be calculated by the following equation

$$w_m = 2\cos^{-1}\{\cos(w_k)/\cosh(1/m \cosh^{-1} b)\} \qquad (6)$$

where, $w_k = (2k-1)\pi/2m$, and $k=0....m$.

Using the relation $z_m = e^{jw}$, we can write Equation (3) as follows

$$H(z) = (z - z_1)(z - z_2)........(z - z_m) \qquad (7)$$

Where, z_1, z_2 ...are locations of zeros, H(z) are the frequency response in z- transform domain.

Replacing z by e^{jw} and z_m's by e^{jw_m}'s in the Equation (7)

$$H(z) = (e^{jw} - e^{jw_1})(e^{jw} - e^{jw_2}).......(e^{jw} - e^{jw_m}) \qquad (8)$$

Filter Design

A Chebyshev FIR filter of order 6 with side bands 40 dB down from the pass band was designed as follows.

Following the design steps outlined from Equation (5) to Equation (8) we can say that $m = 6$,

$$b = 10^{40/20} = 100$$

and the values of the w_m's calculated by using Equation (6) are

$w_1 = 1.64$; $w_2 = 2.0958$; $w3 = 2.7739$; $w_4 = 3.5093$; $w_5 = 4.1874$; $w_6 = 4.6431$

We can write H(z) as

$$H(z) = (z - e^{j1.64})(z - e^{j2.0958})(z - e^{j2.7739})(z - e^{j3.5093})(z - e^{j4.1874})(z - e^{j4.6431})$$

The Chebyshev FIR filter of order 6 with side bands 40 dB down from the pass band has been plotted in Figure (1) and Figure (2). Similarly, a Chebyshev FIR filter of order 20 with side band 40 dB down was designed using the above mentioned procedure and plotted in Figure (3) and Figure (4). As is clear from Figure (1) and (3), that the width of the pass band decreases as the order of the filter increases, and the transition band becomes steeper. The magnitude response in dB curves shown in Figures (2) and (4) make it more clear.

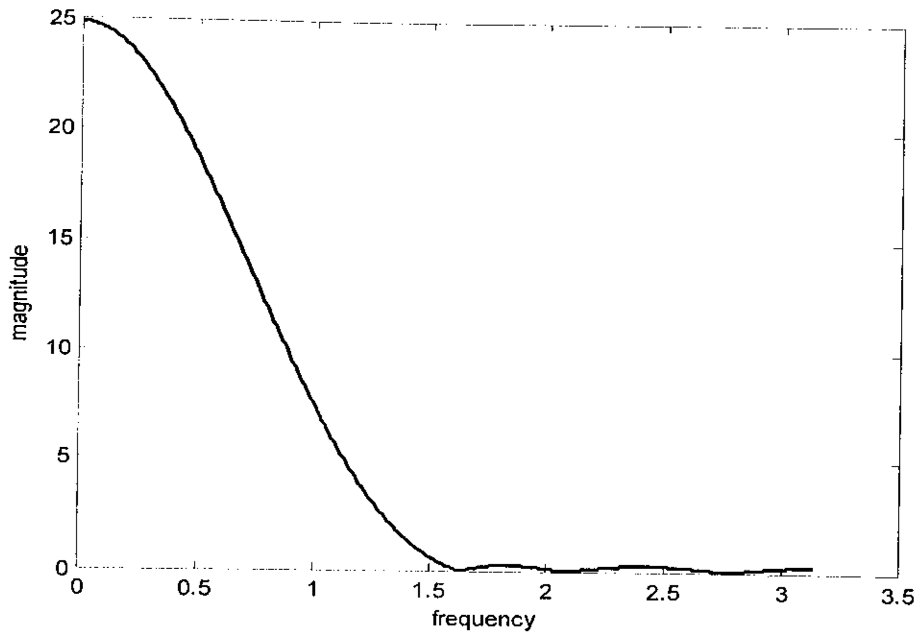


Figure 1: Magnitude Response of 6th Order FIR Filter

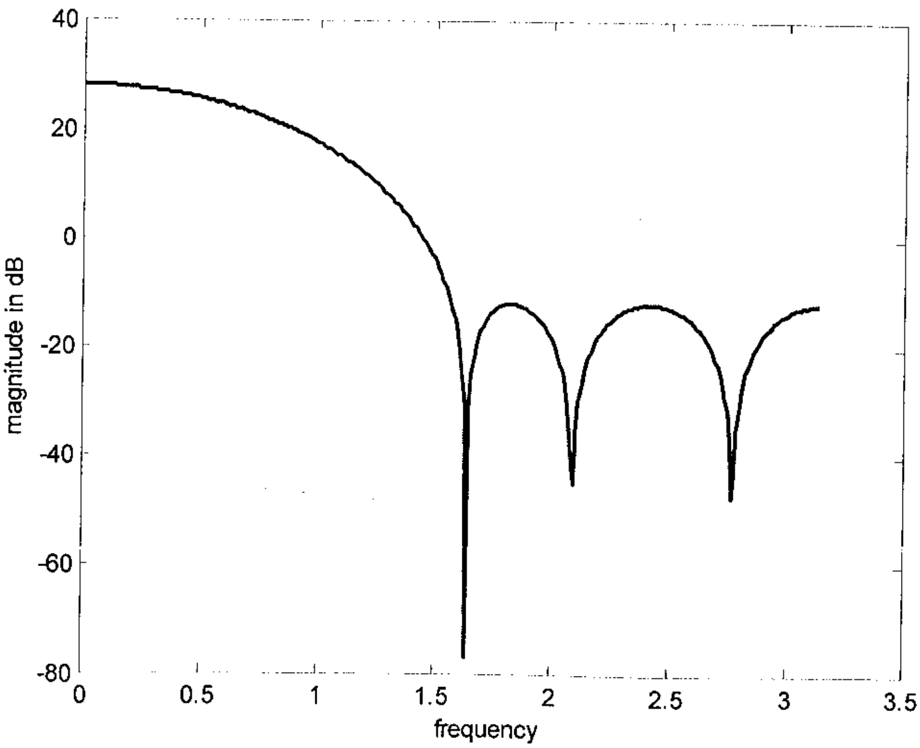


Figure 2: Magnitude Response of 6th Order FIR Filter in dB

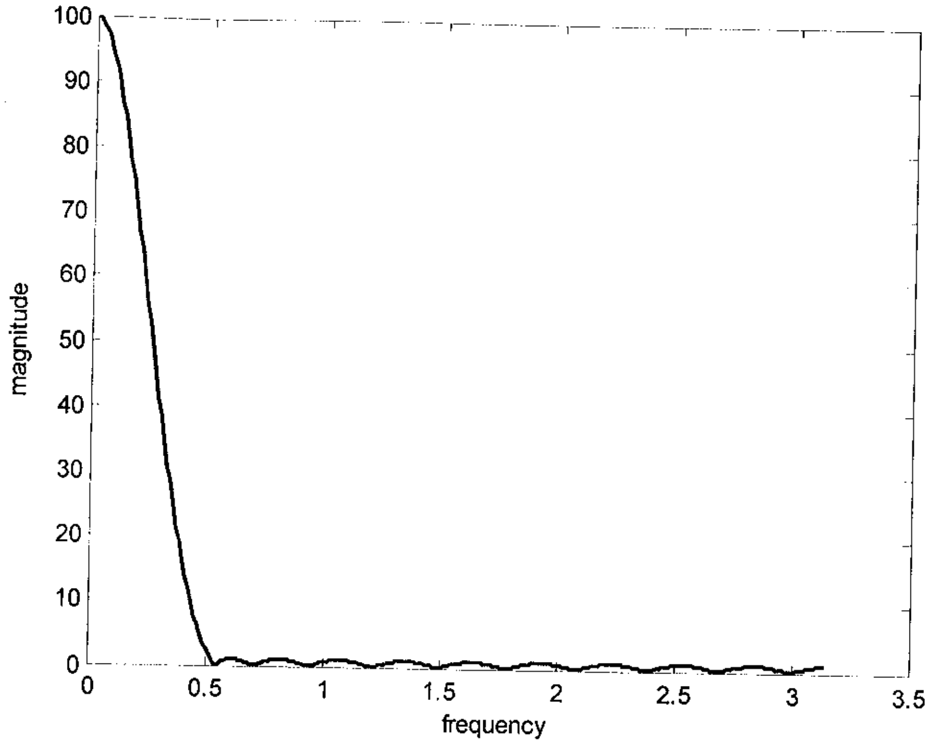


Figure 3: Magnitude Response of 20th Order FIR Filter

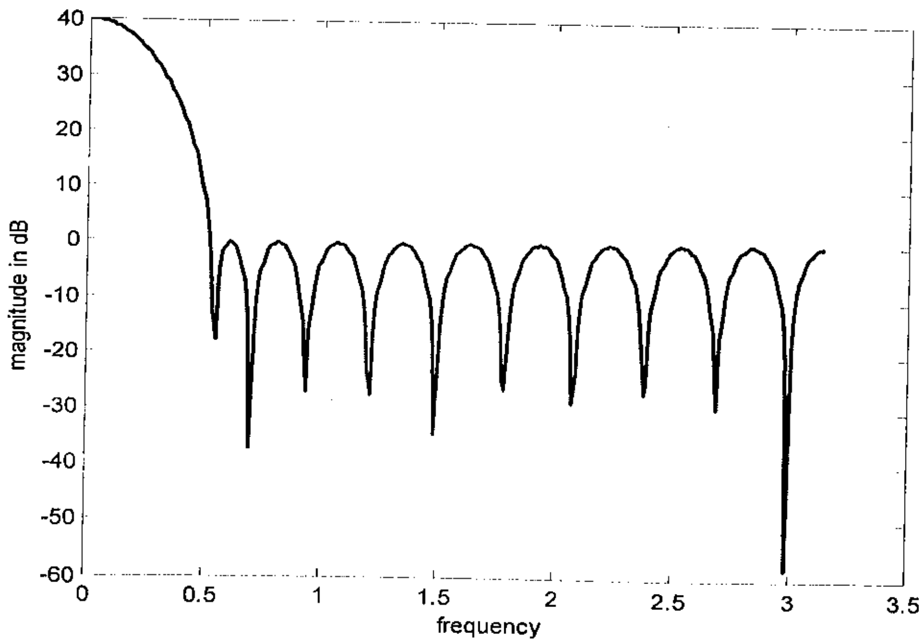


Figure 4: Magnitude Response of 20th Order FIR Filter in dB

2.1 The modified chebyshev filter

We introduced a new parameter to change the filter characteristics. In the original Chebyshev polynomial we multiplied a new parameter 'α' with parameter 'x'.

Thus, Equation (4) became

$$T_m(\alpha x) = \cos(m\cos^{-1}\alpha x) \qquad 0<|x|<1$$

$$T_m(\alpha x) = \cosh(m\cosh^{-1}\alpha x) \qquad 1<|x| \qquad (9)$$

However, multiplying 'x' with 'α' made a difference only in the stop band and the pass band remained the same.

The new location of zeros will now be given by Equation (10)

$$w_m = 2\cos^{-1}[\cos(w_k)/\{\alpha(\cosh(1/m\cosh^{-1} b))\}] \qquad (10)$$

where $w_k=(2k-1)\pi/2m$, and k=0....m.

H(z) can be written as

$$H(z) = (e^{jw} - e^{jw_1})(e^{jw} - e^{jw_2})\ldots\ldots(e^{jw} - e^{jw_m}) \qquad (11)$$

With new values of w_m's calculated using equation (10)

Plotting the magnitude response for a 6[th] order filter with the new parameter taken into account, we get the plots shown in the figures for values $\alpha>1$, $\alpha<1$ and $\alpha=1$. It is evident from the figures that the bandwidth of the filter increases in the case of $\alpha>1$. Figures (8), (9) and (10) also reveal that the side bands are further below than they were with $\alpha=1$.When $\alpha<1$ we can easily conclude from the Figures (6) and (10) that the bandwidth is reduced with an increase in the stopband level. Thus, we can change the passband frequency to some extent by using the parameter 'α'. The phase response of the original and the modified filters are linear in nature.

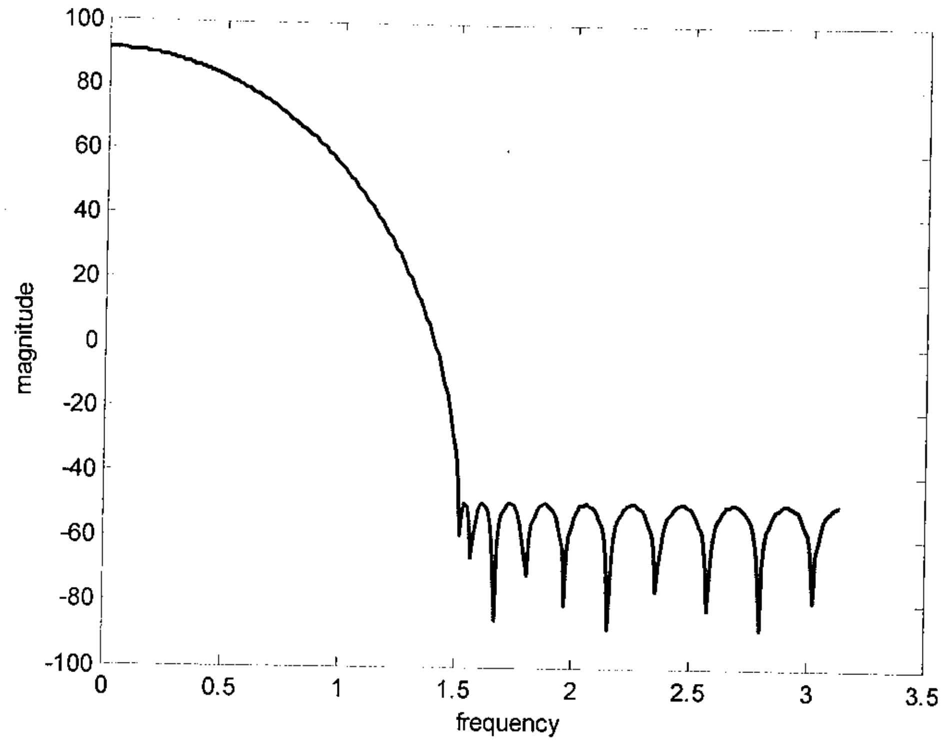


Figure 5: Magnitude Response of 20th Order FIR Filter in dB for α>1

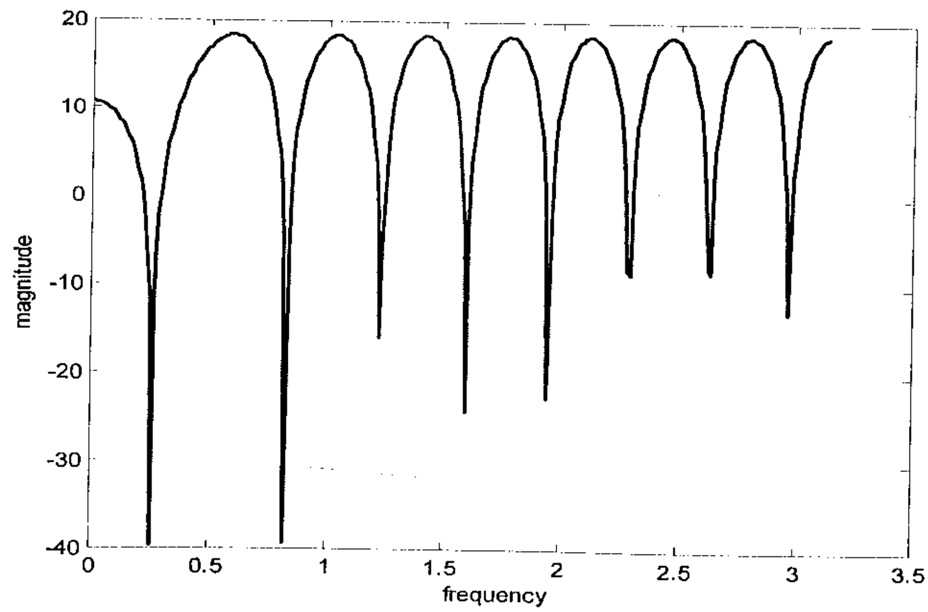


Figure 6: Magnitude Response of 20th Order FIR Filter in dB for α<1

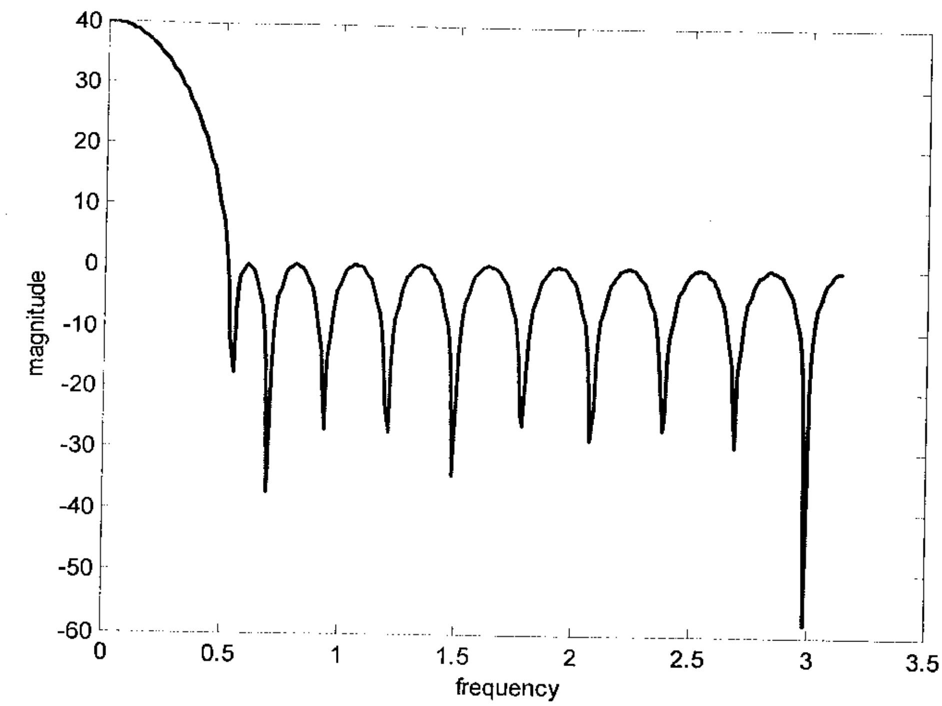


Figure 7: Magnitude Response of 20th Order FIR Filter in dB for $\alpha=1$

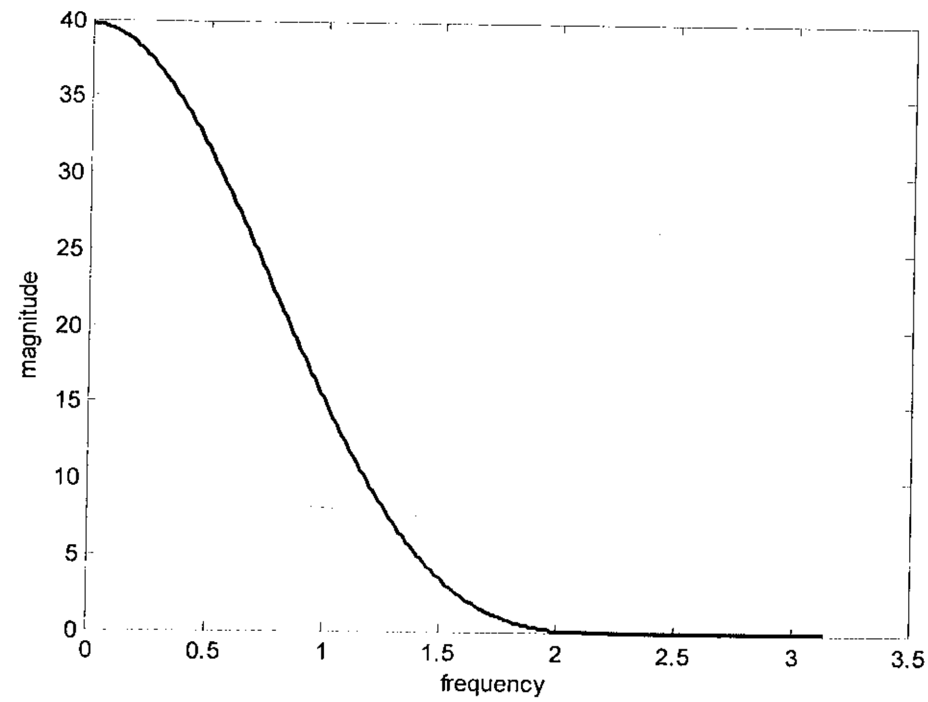


Figure 8: Magnitude Response of 6th Order FIR Filter for $\alpha>1$

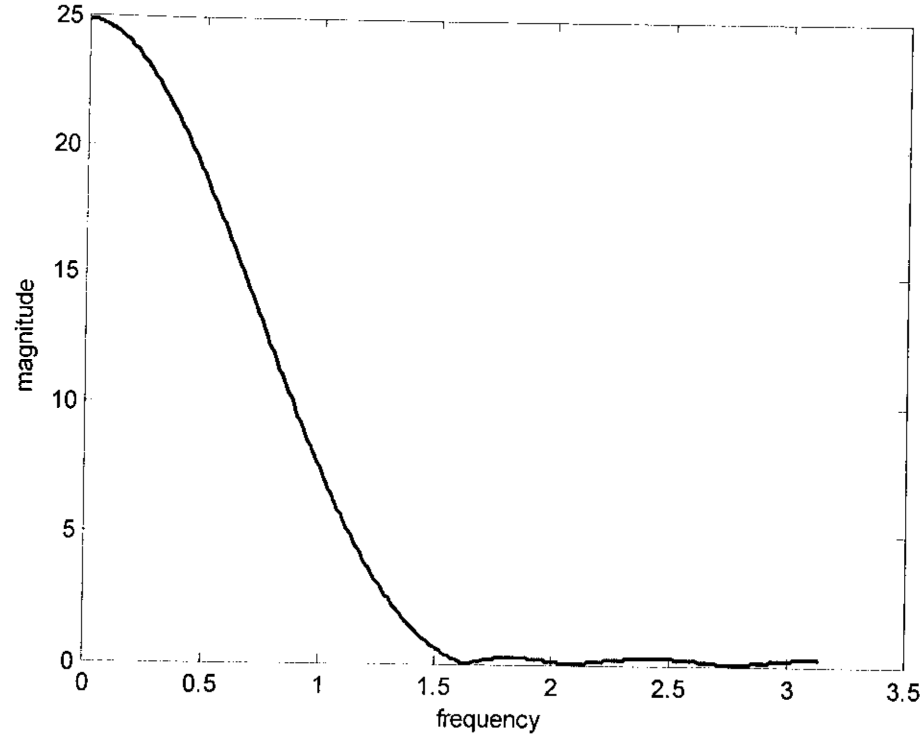


Figure 9: Magnitude Response of 6th Order FIR Filter for α=1

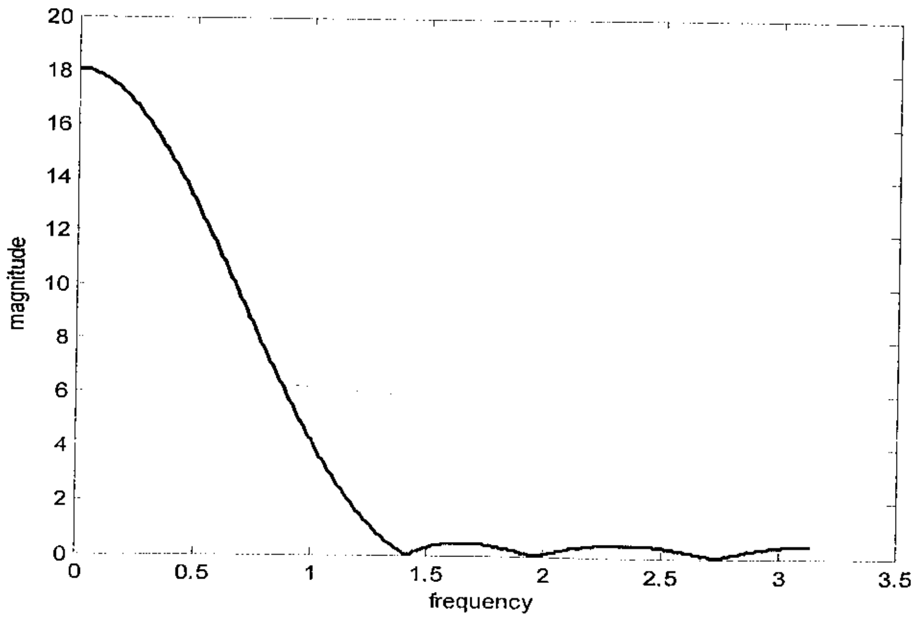


Figure 10: Magnitude Response of 6th Order FIR Filter for α<1

CHAPTER -3

OUTLINE OF THE PROJECT

PROJECT AIM

- The aim of this project is to use software techniques to automatically detect and attenuate the noisy frequencies from an audio signal.
- To develop a system that is easy to use and adaptive.
- The ultimate goal is to develop a widely acceptable piece of software. The software designed will avoid user interaction. The user will input the noisy audio signal in the software and the software which is fully automated will yield a noiseless audio signal.

SOFTWARE USED

MATLAB 7.1:- It is a high-level language with interactive environment that enables to perform computationally intensive tasks faster than with traditional programming languages such as C, C++, and FORTRAN.

SEQUENCE OF EVENTS

3.1 -1st Step

The first step comprised of designing a non-adaptive Chebyshev FIR filter based on antenna theory approach as discussed in chapter 2.The filter responses were plotted against frequency in radians. The range of the frequency axis was kept 2π because the frequency responses are periodic and will repeat after a period of 2π.

3.2 -2nd Step

After designing the filter in the first step we needed a filter that would be capable of showing translation in the frequency domain. This was implemented by introducing a small shift w_0 in H(z) as following,

$$H(z) = \left(e^{j(w-w_0)} - e^{jw_1}\right)\left(e^{j(w-w_0)} - e^{jw_2}\right)\ldots\left(e^{j(w-w_0)} - e^{jw_m}\right)$$

The results of implementing this change in the matlab code are shown in Figures (11) and (12).

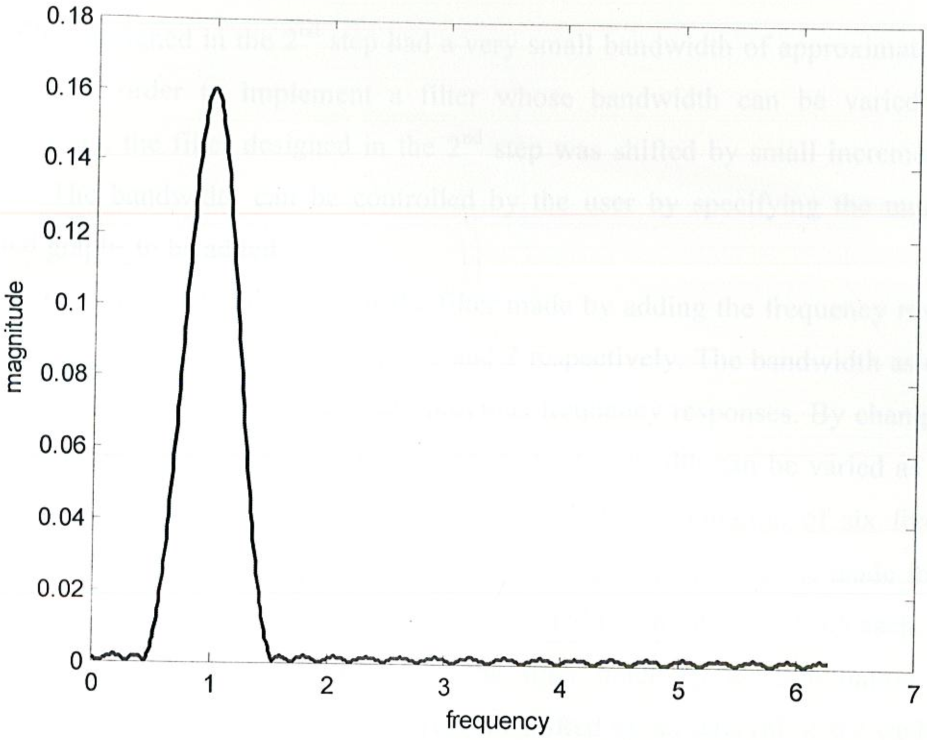


Figure 11: Magnitude Response of 20th Order FIR Filter centred at 1

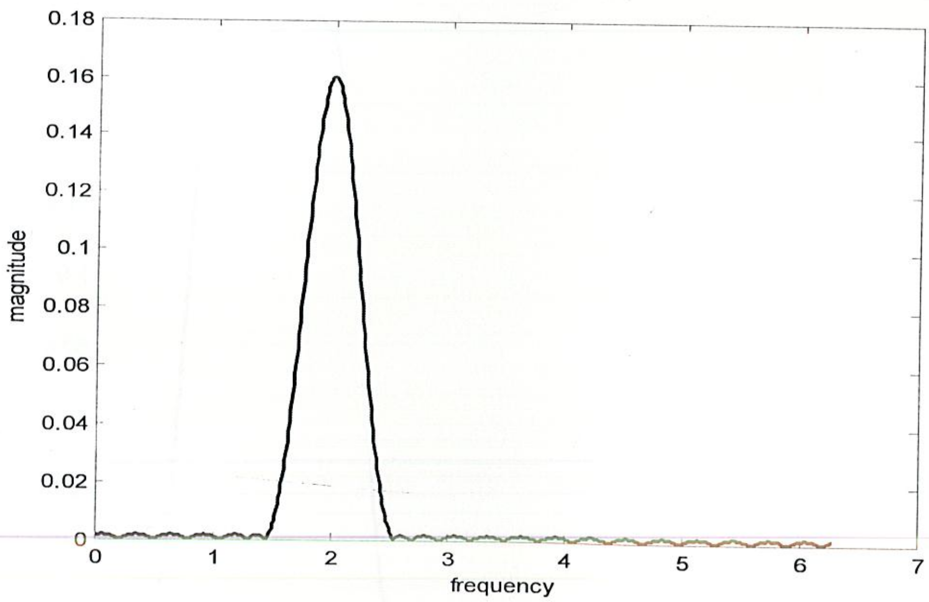


Figure 12: Magnitude Response of 20th Order FIR Filter centred at 2

3.3 -3rd Step

The filter designed in the 2nd step had a very small bandwidth of approximately 0.36 units so in order to implement a filter whose bandwidth can be varied as per requirement, the filter designed in the 2nd step was shifted by small increments and added. The bandwidth can be controlled by the user by specifying the number of shifted graphs to be added.

Figure (13) shows the response of the filter made by adding the frequency responses of a filter of order 20 centered at 1, 1.5 and 2 respectively. The bandwidth as evident is larger in this case compared to the previous frequency responses. By changing the number of frequency responses being added, the bandwidth can be varied as can be concluded from Figure (14) which is made from the summation of six frequency responses shifted by an interval of 0.5 each and Figure (15) which is made from the summation of only two frequency responses shifted by an interval of 0.5 each. Figure (16) shows the frequency response of the filter made by the summation of the frequency responses of six filters of order 40 shifted by an interval of 0.2 each. Thus filters of various orders and bandwidths can be easily implemented.

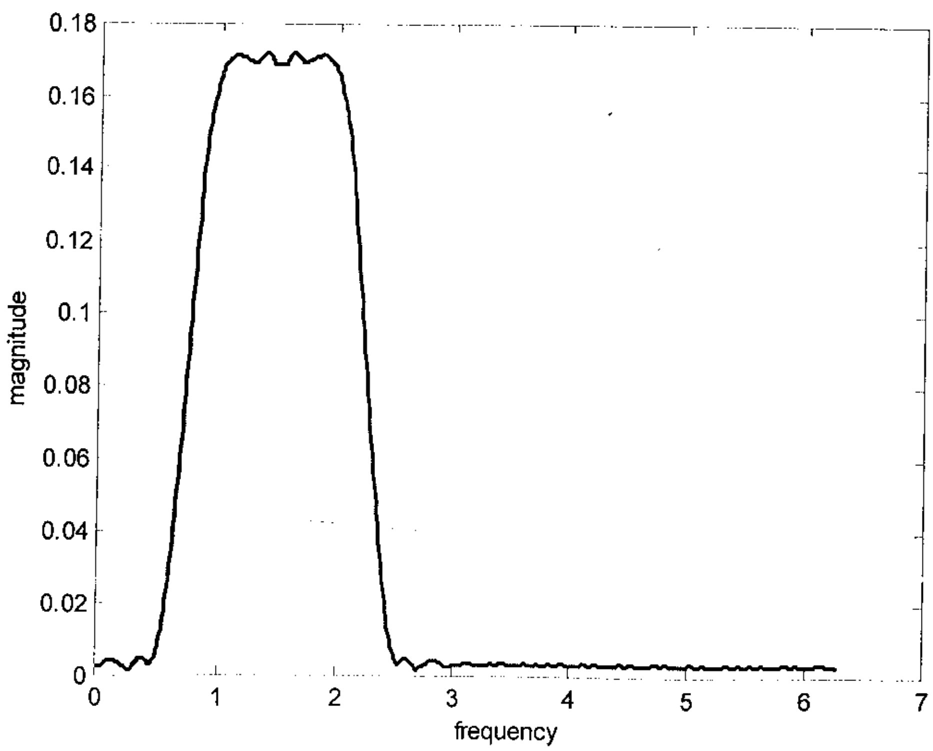


Figure 13: Magnitude Response of 20th Order FIR Filter with a bandwidth of 1.34 units

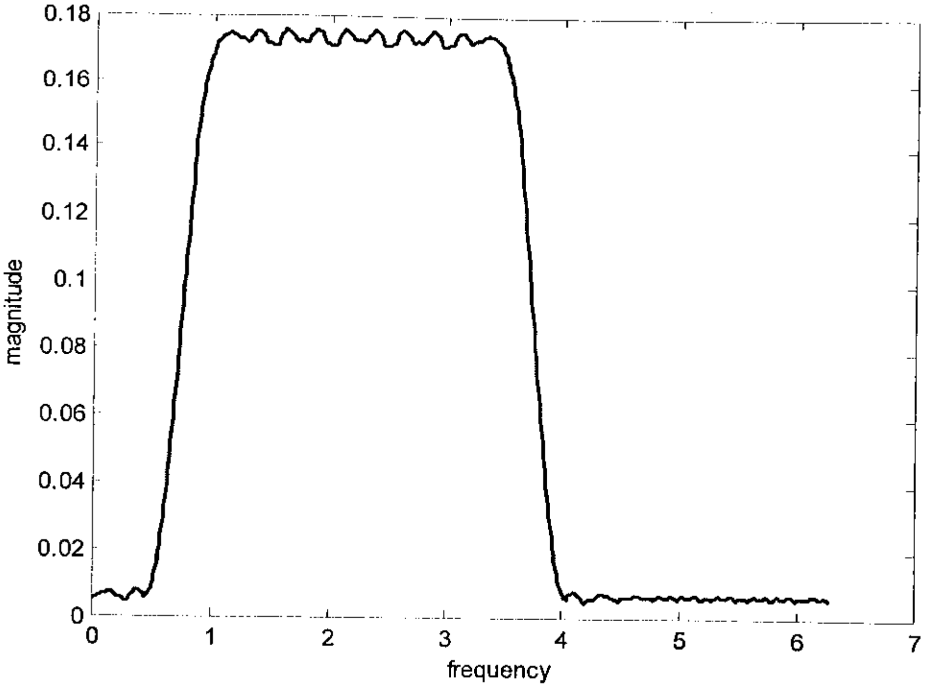


Figure 14: Magnitude Response of 20th Order FIR Filter with a bandwidth of 2.93 units

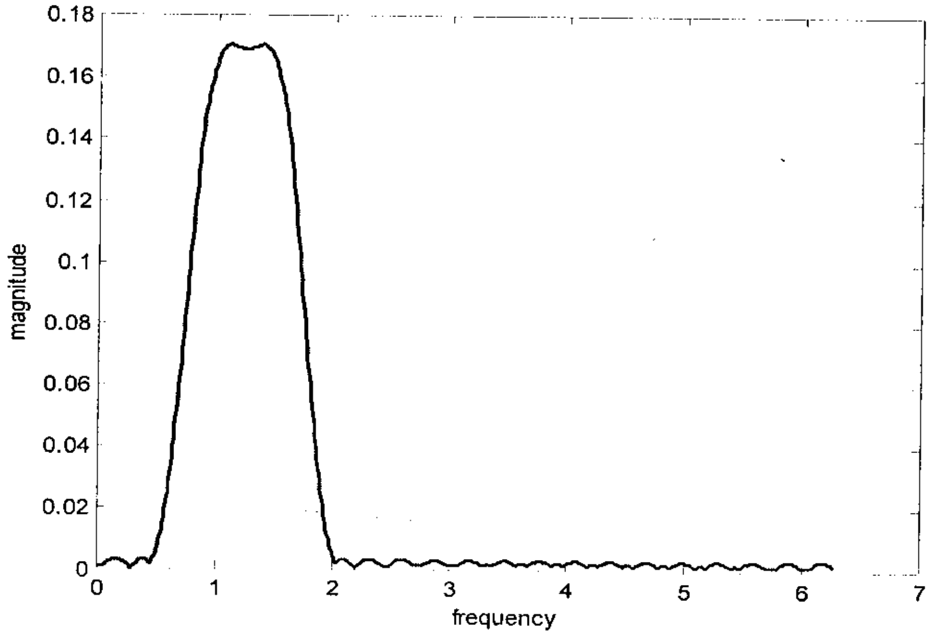


Figure 15: Magnitude Response of 20th Order FIR Filter with a bandwidth of 0.845 units

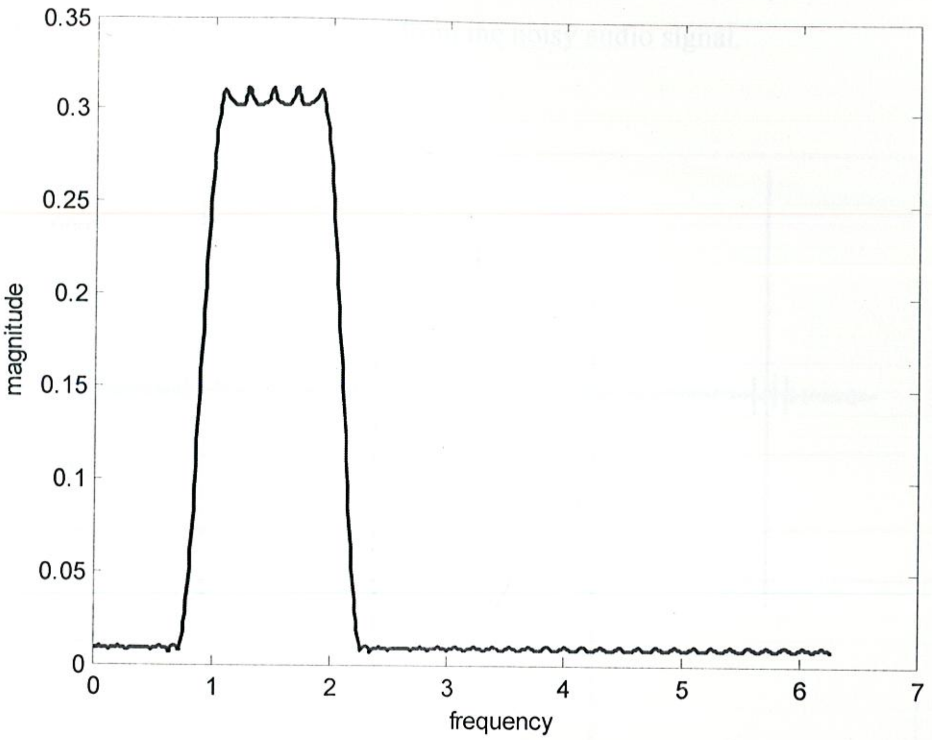


Figure 16: Magnitude Response of 40th Order FIR Filter with a bandwidth of 1.09 units

3.4 -4th Step

For the filter to be adaptive, it was necessary that the filter has an algorithm that would compute the noisy frequencies present in the signal and adjust its coefficients such that it would eliminate the noise and retain the noiseless frequencies.

Since noise present in any signal is generally of high frequency, we computed the FFT of the audio signal and further computed the positions at which a sudden spike in the plot of the FFT was observed. These spikes represented the noise present in the signal.

Figure (17) shows the FFT plot of a noisy signal and Figure (18) shows the positions of the noisy frequencies. 'p' denotes the positions of the noisy frequencies and 'c' denotes the center point between the two consecutive spikes, representing noisy frequencies. The need for calculating 'c' arose because by centering the filter shown in Figure (19) at the various values of 'c' the noisy frequencies were removed in order to yield a noiseless audio signal as represented in Figure (20). The software automatically calculates the values of 'c' and centers an appropriate filter at those values in order to remove the noise from the noisy audio signal.

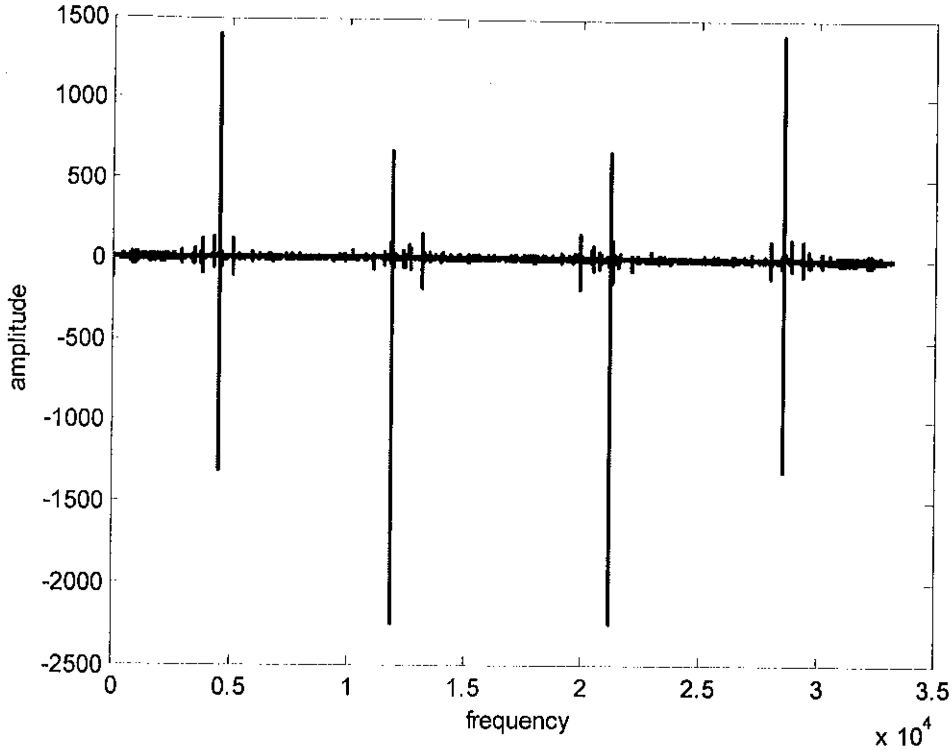


Figure 17: FFT plot of noisy audio signal

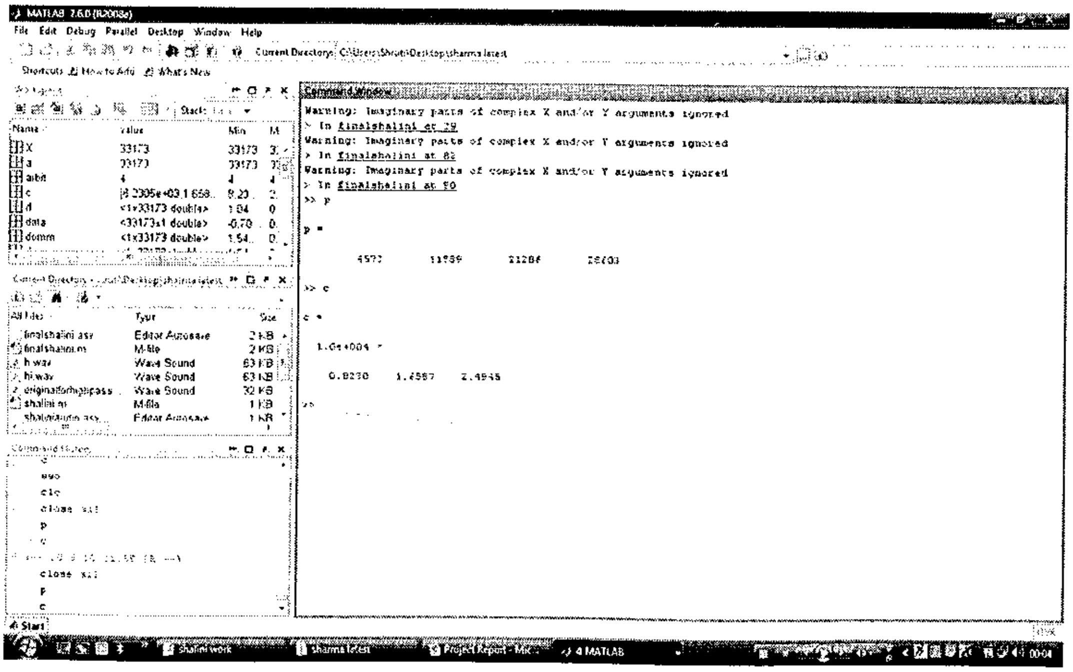


Figure 18: Output showing the noisy frequencies present in the audio signal

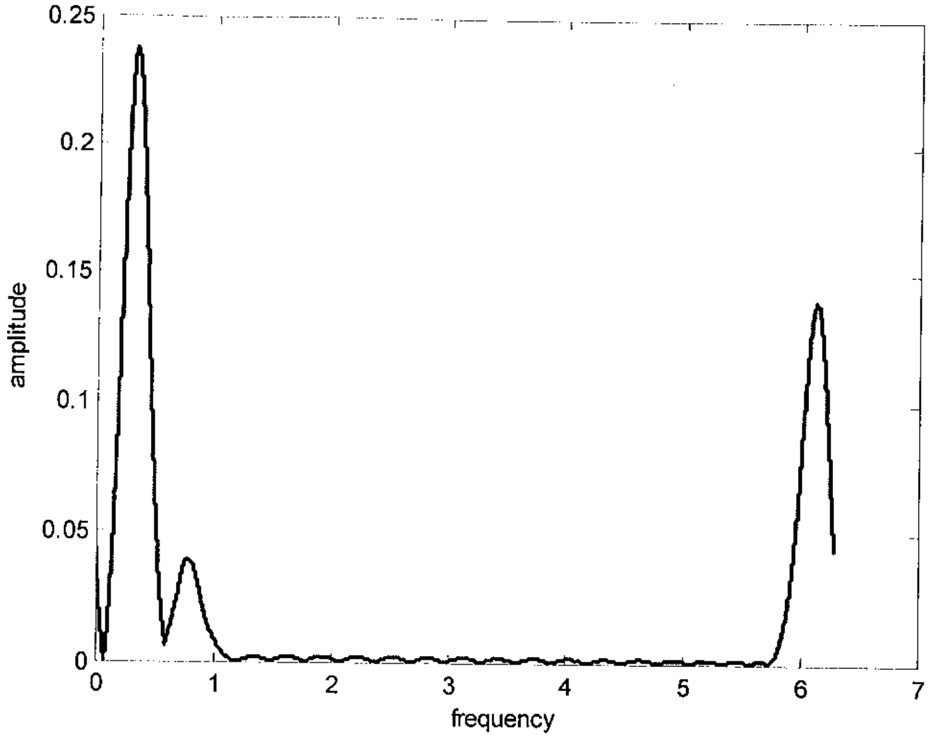


Figure 19: The filter response of the adaptive filter used for removal of noise

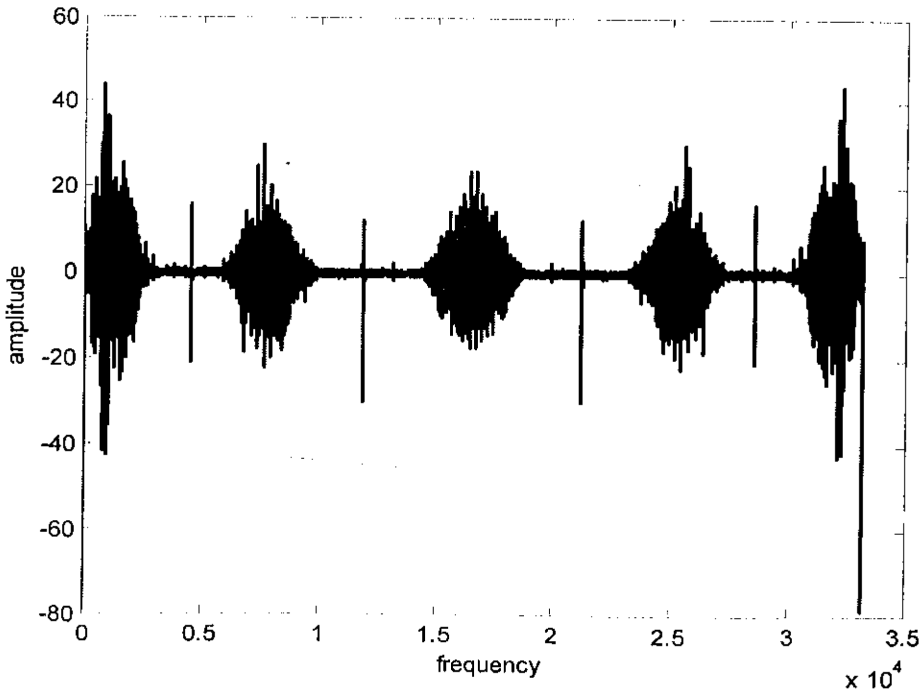


Figure 20: FFT plot of noiseless audio signal

In another simulation, the FFT plot of the noisy signal was as shown in Figure (21). Figure (22) represents the noisy frequencies distorting the audio signal. Again, the software automatically detected the noisy frequencies and based on the frequencies representing noise, adapted the filter coefficients in order to yield a filter with the filter response as plotted in Figure (23). The filter then attenuated the noisy frequencies to give a noiseless audio signal as represented in Figure (24).

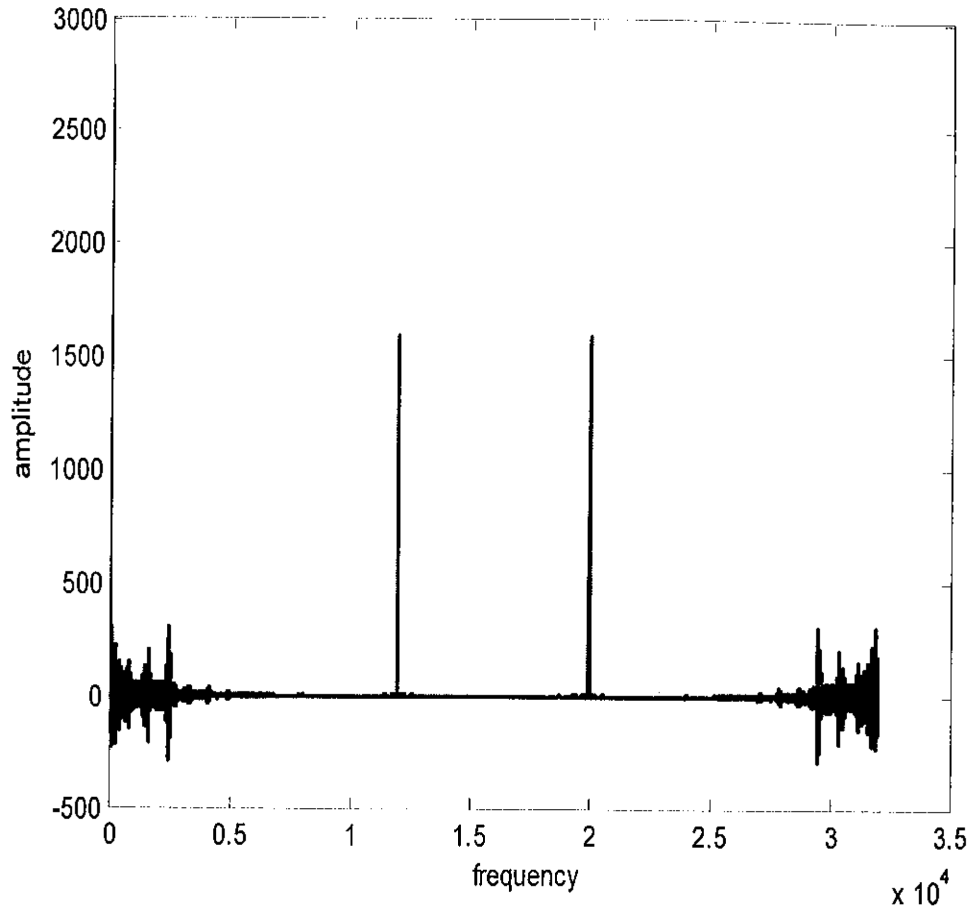


Figure 21: FFT plot of noisy audio signal

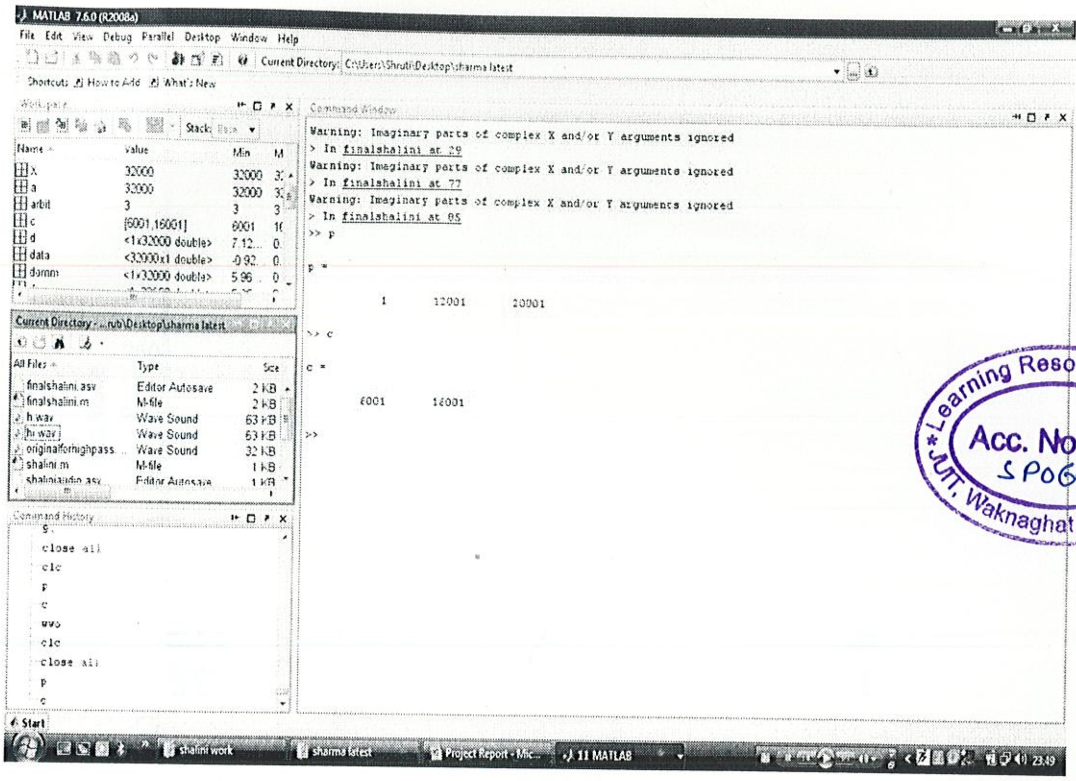


Figure 22: Output showing the noisy frequencies present in the audio signal

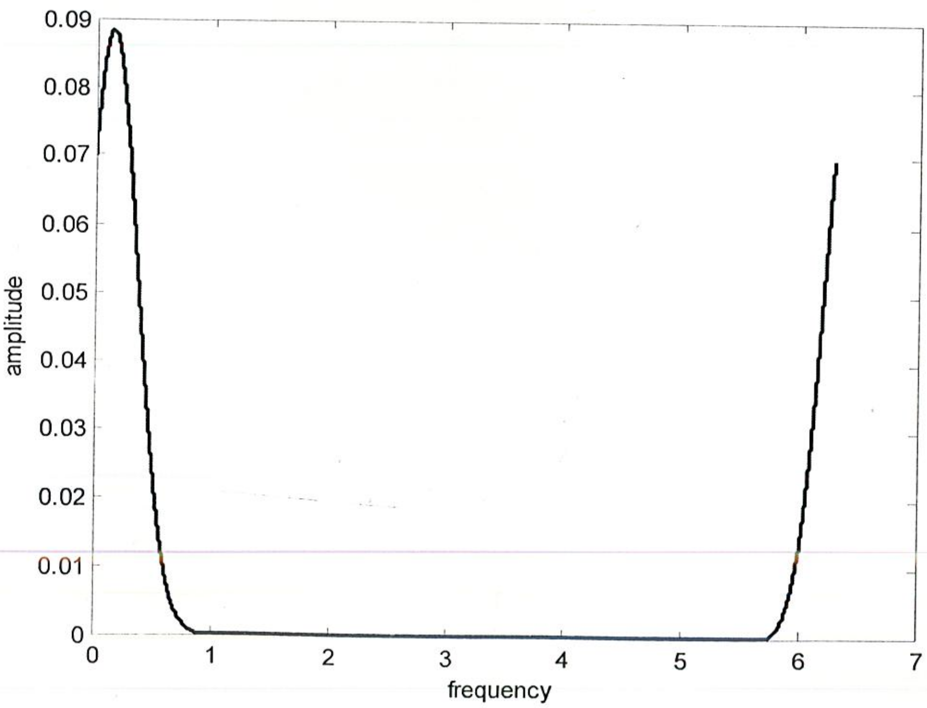


Figure 23: The filter response of the adaptive filter used for removal of noise

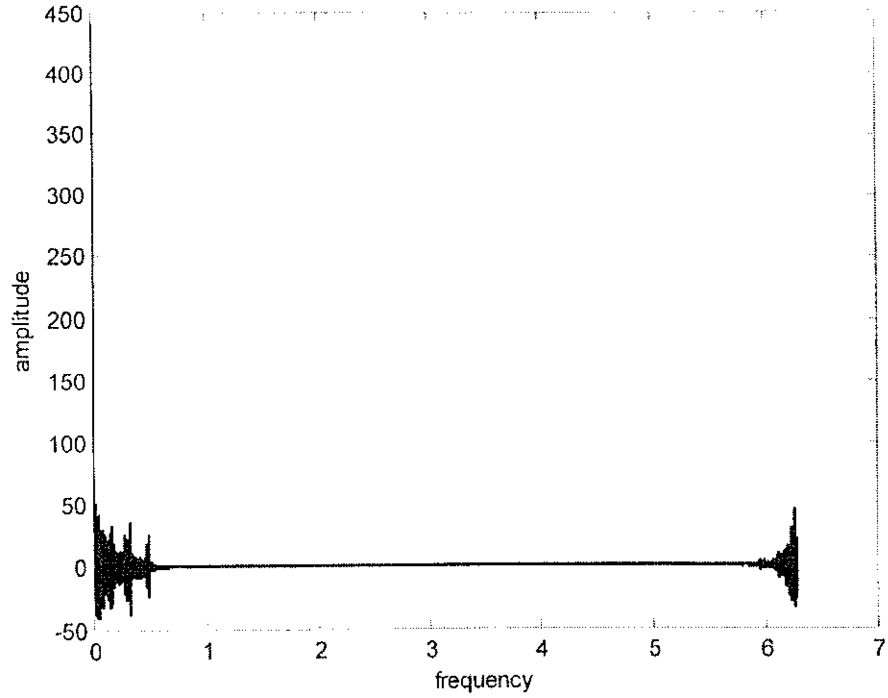


Figure 24: FFT plot of noiseless audio signal

CONCLUSION AND FUTURE WORK

The proposed method for designing an adaptive filter for noise cancellation in audio signals was implemented and tested on various audio signals. The signals showed a marked clarity after being passed through the filter.

Limitation

The adaptive filter works well only for impulsive noise. It doesn't give appropriate results for uniformly distributed noise patterns or noise introduced in the signal while recording. Figure (25) shows the FFT plot of an audio signal with uniformly distributed noise. Figure (26) represents the output of the filter in which even few frequencies of interest have been attenuated.

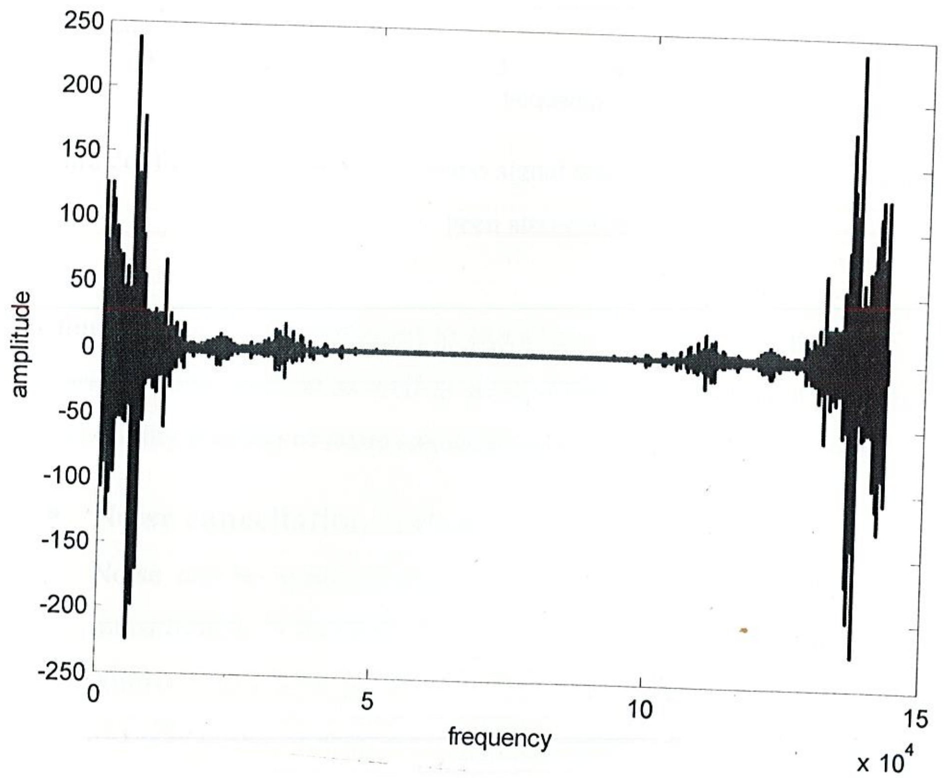


Figure 25: FFT plot of noisy audio signal

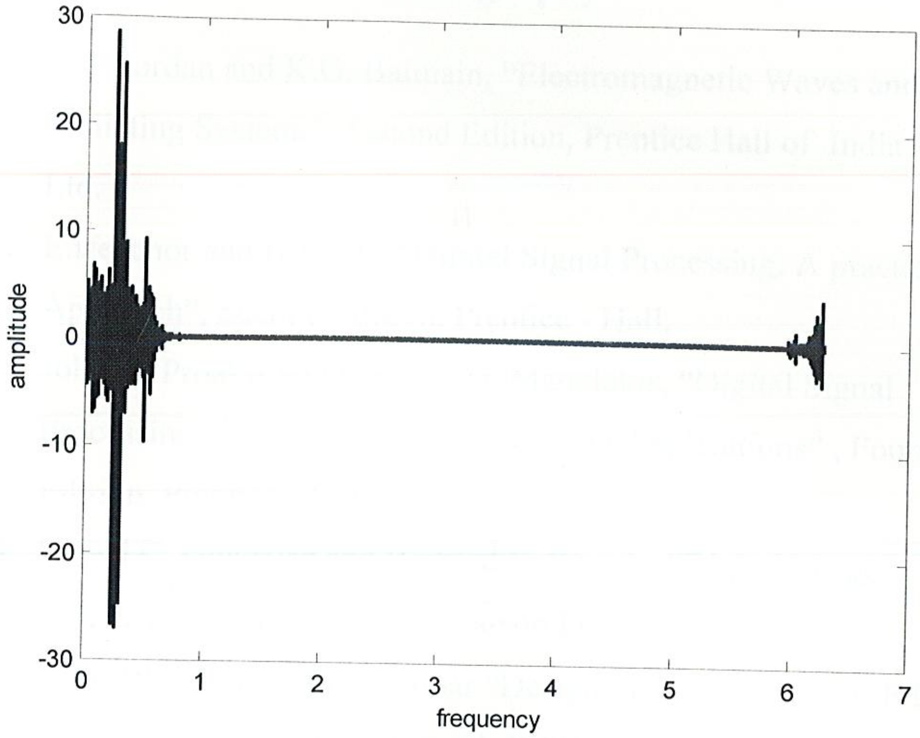


Figure 26: FFT plot of the output audio signal where a few frequencies of interest have been attenuated

For future work we have planned to extend the design so that the filter may be used for various noise patterns as well as multi-dimensional adaptive filtering techniques; such as for the purpose of noise cancellation in images.

- **Noise cancellation in images**

  Noise can be systematically introduced into images during acquisition and transmission. A fundamental problem of image processing is to effectively remove noise from an image while keeping its features intact. The adaptive algorithm may be modified in order to remove the impulse noise from images. Impulse noise is characterized by replacing a portion of an image's pixel values with random values, leaving the remainder unchanged. Such noise can be introduced due to transmission errors. The most noticeable and least acceptable pixels in the noisy image are then those whose intensities are much different from their neighbours.

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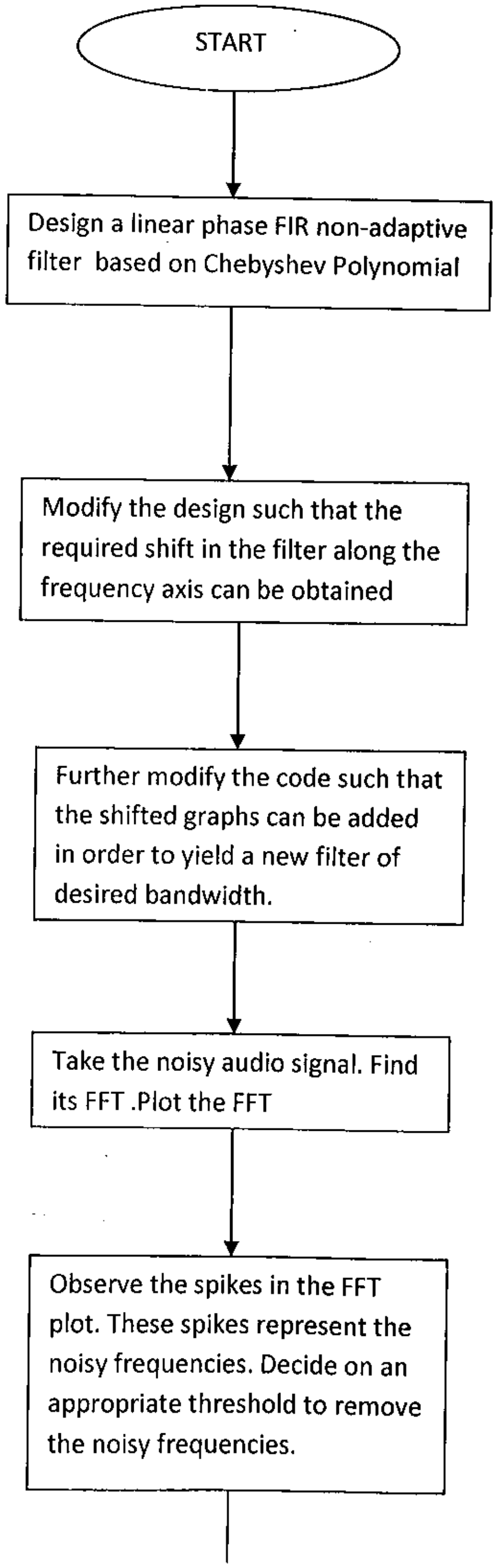
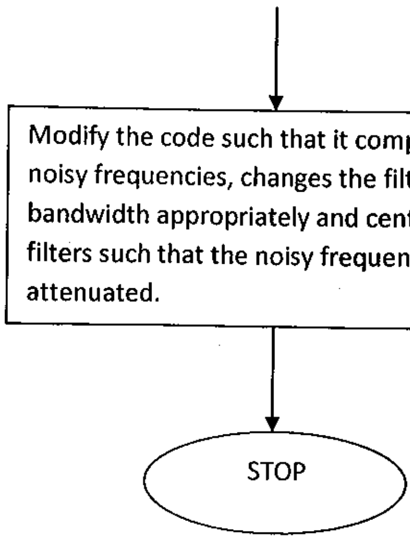
APPENDIX A

FLOW CHART




Modify the code such that it computes the noisy frequencies, changes the filter bandwidth appropriately and centres the filters such that the noisy frequencies are attenuated.
STOP

APPENDIX B

PSEUDO CODE

1) Designfir.m

- Input the order of the Chebyshev FIR filter to be designed i.e. 'm'.
- Find out the absolute value of attenuation in the stop band 'b' by using the formula $b= 10^{(\text{attenuation in dB})/20}$.
- Find the location of zeros, w_m, on unit circle by the following equation
  - $w_m = 2\cos^{-1}\{\cos(w_k)/\cosh(1/m \cosh^{-1} b)\}$
  - where $w_k = (2k-1)\pi/2m$, and k=0....m.
- Further use the values of w_m's to find out H(z) which is the frequency response in the z-transform domain.
  - $H(z) = (e^{jw} - e^{jw_1})(e^{jw} - e^{jw_2})\ldots\ldots(e^{jw} - e^{jw_m})$
- Plot the absolute value of H(z) along frequency in radian using the plot command.

2) Shiftfilter.m

- Input the order of the Chebyshev FIR filter to be designed i.e. 'm'.
- Find out the absolute value of attenuation in the stop band 'b' by using the formula $b= 10^{(\text{attenuation in dB})/20}$.
- Find the location of zeros, w_m, on unit circle by the following equation
  - $w_m = 2\cos^{-1}\{\cos(w_k)/\cosh(1/m \cosh^{-1} b)\}$
  - where, $w_k = (2k-1)\pi/2m$, and k=0....m.
- To shift the frequency response along the frequency axis use the following equation
  - $H(z) = \left(e^{j(w-w_0)} - e^{jw_1}\right)\left(e^{j(w-w_0)} - e^{jw_2}\right)\ldots\left(e^{j(w-w_0)} - e^{jw_m}\right)$ where, 'w_o' specifies the desired shift along the frequency axis.

3) Variablebandfilter.m

- Input the order of the Chebyshev FIR filter to be designed i.e. 'm'.
- Find out the absolute value of attenuation in the stop band 'b' by using the formula b= $10^{(\text{attenuation in dB})/20}$.
- Specify the number of filters whose frequency responses need to be added in order to yield the filter of desired bandwidth i.e specify 'n'.
- Specify where the resultant band pass filter has to be centred i.e. specify 'wo'.
- Add all the frequency responses and store the resultant value in the variable 'domm'.
- Plot the variable 'domm' against frequency in radians i.e. 'omega'.

4) Noisyfrequencies.m

- Import the audio file that is to be filtered.
- Find out the fft of the signal.
- Plot the fft of the signal throughout the range .
- Set a threshold to distinguish between the noisy and the noise less frequencies.
- Find the location of the noisy frequencies based on this threshold. Location of the noisy frequencies is stored in the matrix 'p'.
- Find the centre points between the two consequent noisy frequencies and store the values of the centre points in the matrix 'c'.
- The filter is automatically centred on the various values stored in 'c' such that the frequencies representing the noise are attenuated to yield the desired signal.