ODA BULTUM UNIVERSITY

INSTITUTE OF TECHNOLOGY

DEPARTMENT OF ELECTRICAL AND COMPUTER ENGINEERING

(Communication System Engineering)

Semester Project Submitted To Electrical and Computer Engineering Department

Tittle: Design and Simulation of Noise Reduction System by Using Adaptive Filter

By: **ID No**

1. ROBA DUGDA ……………..…………………..……………. 0150/11
2. NEGA GETNET…………………………...………………..0144/11
3. FIKRU RAMSO …………………………...……………….0076/11
4. BEKELE TAREKEGN ……………………..………………0033/11

Advisor: MS. SELAM .T

Submission Date: June, 2022

.

Chiro; Ethiopia

DECLARATION

We do here is that the project titled "DESIGN AND SIMULATIN OF NOISE REDUCTION BY USING ADAPTIVE FILTER" is submitted to the school of electrical and computer engineering at ODA BULTUM INSTITUTE OF TECHNOLOGY, under the advisor of SELAM.T. This project is our work and was not presented elsewhere

Name	Signature	Date
1. ROBA DUGDA	__________________	June/22/2022
2. NEGA GETNET	__________________	____________
3. FIKRU RAMSO	__________________	____________
4. BEKELE TAREKEGN	__________________	____________

As an advisor of this group, I declare that I have advised their work thought the course of this project and all works included in this project document is their work and no plagiarism takes place. I assure all this with my signature.

	Signature	Date
Ms. SELEM .T	___________________	__________________

ACKNOWLEDGEMENT

For our project to arrive at this stage, many individuals give us their forwarding contribution since the beginning. First we would like to thank our friends who gave an important point about this project and they also helped us by showing the direction to do this project on our title design and simulation of noise reduction system by using adaptive filter. Next we want to thank our teachers especially, our advisor MS.SELAM.T for their truly help and guide us from the beginning of the project and we also thank some other lectures for helping us to do the project. And next we would like to thank department of Electrical and Computer Engineering who create the chance for us to do this project. Finally, and foremost we would like to thank our God who helped us from starting to ending process successfully this project.

ABSTRACT

In practical application, the statically characteristics of signal and noise are usually unknown or can't have been learned so that we hardly design fix coefficient digital filter.in allusion to this problem, the theory of adaptive filter and adaptive noise cancellation are researched deeply. According to the recursive least mean square algorithm realizes the design and simulation of the adaptive algorithm in noise reduction and analyze the result to prove its performance is better than the use of fixed filters designed by conventional methods. In any communication system noise always had been a major area of concern. Noise signals affect the transmitted signals during transmission. Most of project designed before have their own limitation and also only seen in large company when it's applicable.

Therefore, the main objective of this project is design of noise reduction by using adaptive filter to promote accurate solutions and a timely convergence to that solution to implement and effective to use locally. In order to achieve our objective, we have used mat lab and RLMs (Recursive Least Mean Square) algorithm. In simulation result the system minimize the noise from noisy signal. Also the system plots the spectral analysis of the original signal, noise signal that have reduced noise. To sum up, this system will be useful and applicable in many areas.it can be used at place in which noise is difficult problem such as conference hall and music stage and in different communication way.

Key words: Adaptive Filter, Noise Reduction, Adaptive Algorithm, RLS Algorithm

Table of Contents

LIST OF FIGURE

LIST OF ACRONYM

RLMS- Recursive Least Mean Square

LMS - Least Mean Square

MSE- Mean Square Error

FIR-Finite Impulse Response

IIR - Infinite Impulse Response

ANC - Adaptive Noise Reduction

SNR- Signal to Noise Ratio

MPC- Mean Power Calculation

DSP- Digital Signal Processing

UNSYS-Unknown System

CHAPTER ONE

INTRODUCTION

1.1 Background

In the process of digital signal processing, often to deal with noise or time varying signals. Noise consists of unwanted waveforms that can disturb our communication. Noise can be internal or external to the system. Noise is defined as unwanted, annoying, unpleasant loud disruptive to hearing. Our ears are excellent at telling us what noise is. Most commonly, noise is an annoying tone that causes mild to major discomfort or irritation. From a physics standpoint, noise is indistinguishable from sound, as both are vibrations through a medium, such as air or water. The difference arises when the brain receives and perceives a sound. So if the noise is not reduced from our system there is no a way to communicate clearly. Noise can have complicated mixture of different frequencies and amplitudes. What's worse is it can even change from time to time. In such cases, it is difficult to write precise equations and often unnecessary to model noise. Even if we succeed in writing good enough approximations, it would become very difficult to build filters that would suppress this noise. With the advent of Digital Signal Processing algorithms, and availability of fast computing power has made it possible.

From starting of using different way of communication there is also the designing of noise reduction system. For the past many years, noise redaction using adaptive filters design has been an active area of scholarly research and innovative implementations. In this century there is a huge improvement of noise reduction mechanisms that goes parallels with technology. Noise cancellation technology is a growing field that aims to cancel or at least minimize unwanted signal and get noise free information between transmitter and receiver.

There are several system of noise reduction. For example, design and implementation of adaptive filtering algorithm for noise reduction in speech signal, de-noising or noising reduction mechanisms using Mean Power Calculation (MPC) for voice communication, Noise Cancellation using Adaptive Filtering in ECG Signals and Noise cancellation using least mean squares adaptive filter. Adaptive Filter is one of the digital filter parameter that that adjusts the filter parameter of

The present moment, to adapt to the unknown signal and noise, or overtime changing statically properties, in order to achieve optimal filtering [1].

Adaptive system has “self-regulation ’’and “tracking “capacities. Adaptive filter can be divided in to linear and nonlinear adaptive filter .nonlinear adaptive filter has more signal processing capabilities. However, due to the on linear adaptive filter more complicated calculations; the actual use is the linear adaptive filter. Performance of the filter is measured with different parameter like convergence rate; mean square error etc.

Our system is noise reduction system using adaptive filter. In the process of digital signal processing, often to deal with some unforeseen signal, noise, or time varying signals, if FIR and IIR filter of fixed coefficients cannot achieve optimal filtering, we must design adaptive filters to the change of signal and noise. Adaptive filter is that uses a filter parameter of moment ago to automatically adjust the filter parameter of the present moment, to adapt to the statically properties of that the signal and unknown noise or random changes, in order to achieve optimal filter. Adaptive noise cancellation is widely used to improve the Signal to Noise Ratio (SNR) of a signal by removing noise from the received signal. Availability of different Digital Signal Processing algorithms, made it possible for us to use the digital treatment of noise cancellation effectively and in a simple way. In this project we tried to implement adaptive noise cancellation on different voice signals.

1.2 Statement of Problem

Currently information plays significant role in our daily activity and there is different way of communication that is used to interchange the information for many problem within our communication way and also there is misunderstanding between transmitter and receiver this is caused by unwanted signal that tends to disturb the transmission and processing of signals in communication systems and over which have incomplete control. In addition to this noise is difficult problem such as conference hall and music stage and in different communication way. Generally, most of project designed before is only seen in large company when it’s applicable. So, the reason that adaptive filtering design techniques must be implemented is to promote accurate solutions and a timely convergence to that solution and implement to use locally.

1.3 Objectives

1.3.1 General objective

- ✓ The main objective of this project is to design noise reduction system by using adaptive filter.

1.3.2 Specific objectives

- ✓ To design different systems that iteratively reduce the noise from the noisy signal such as adaptive system identification, adaptive inverse system each having respective function to reduce the noise.
- ✓ To design and analyze circuit diagram of adaptive system identification.
- ✓ To design and analyze circuit diagram of adaptive noise cancellation.
- ✓ To design and simulate time response of noisy signal by using mat lab program.
- ✓ To converge the error signal that is interfered on the input signal by comparing it to the desired signal.

1.3 Significance of the Study

- ✓ Knowing the importance of adaptive filter for reduction of nose signal from the noisy signal.
- ✓ To get aware of how to program noise reduction using adaptive filter in mat lab.
- ✓ Overcoming the difficulties of noise reduction using a digital filtering method called direct filtering that are not adapted the environment to filter out the noise and other DSP filtering mechanisms.
- ✓ Reducing noise from noisy signal without knowing the characteristics of the signal.

1.4 Scope of the Project

The scope of this project will cover to design of noise reduction system by using adaptive filter in the area of communication system.

This project contains the noise reduction of communication by using adaptive filter that uses rlms algorithm that written using MATLAB.

Data will be simulate using adaptive algorithm to reduce noise in a communication signal. The requirements of the projects are that it should be simple and understandable to be used by interested user.

CHAPTER TWO

LITERATURE REVIEW

2.1 Brief Overview

The innovated technology always gives the updated and versatile trend in every field. In past years we perceived that every task was difficult due to the backward technology but now a day's technology provides a new direction in every field of electronic communication, agriculture, business, medical etc. This unit explain about the evolution in technology in the field of noise reduction in communication.

Noise Cancellation using Adaptive Filtering in ECG Signals is done in [2].

The project is done to reduce the noise interference in the ECG signals. Some of the most common examples of noise that the ECG filter would need to remove in order to give useful results includes power line interference, motion artifacts, muscle contraction, electrode contact noise and interference caused due other electronic equipment. This project is good because it shows the simplicity of LMS algorithm and ease of implementation, evident from above make this algorithm better in many real time systems to improve the SNR and to reduce the noise of signal. But the limitation is that The ECG (electrocardiogram) is a system that provides patient's critical heart activity. The adaptive electrocardiogram is designed to reduce noise caused by external systems and body artifacts only. So, it only applicable for medical service.

The other project done on noise redaction topic is that the simulation of the noise cancellation using RLMS adaptive filter algorithm. In this project the noise corrupted speech signal and the engine noise signal are used as inputs for RLMS adaptive filter algorithm. The filtered signal is compared to the original noise-free speech signal in order to highlight the level of attenuation of the noise signal. This project is well done that the result shows that the noise signal is successfully canceled by the developed adaptive filter and outcome implies that the filtered signal is approaching the noise-free speech signal. But it has the limitation in terms of frequency range on

Which it's applicable that it removes the noise signal at lower frequencies range (0Hz to 5000Hz) [3].

Noise cancellation using least mean squares adaptive filter is done in [4].

The system implemented for the removal of constant energy, wide bandwidth noise (that has little to no abrupt changes), such as noise from a plane or car engine, noise from air vents, or any constant “room noise.” The project good performance is it’s able to successfully remove much of the noise added to the input signal. When implementing a microphone for measuring room noise, the system removed much of the constant energy, wideband noise elements interfering with the music played through the headphones. However, the limitations of this system come when there is some narrow band noise, such as people talking, sharp sounds like clapping, and other narrow band non-periodic sounds

Design and Development of Noise Cancellation System for Android Mobile Phones is another work done before on noise cancellation (redaction) topic**.** This paper describes the design and development of Noise Cancellation System (NCS) to cancel the background noise from the speech signal of speaker in android mobile phones. NCS system has been implemented using an adaptive filter, which changes its filter characteristic according to the change in behavior of the input signal to minimize the noise in signal. Least Mean Square (LMS) algorithm has been used for adapting the filter weights. The good quality of this work is that the quality of the required signal can be further improved by enhancing the adaptive filter algorithm. Even if it has its own good thing it is only worked for android mobile phone [5].

As briefly explained above in this unit the project done before on noise reduction remove much of the noise added to the input signal and in many of them quality of the required signal can be further improved by enhancing the adaptive filter algorithm. But they have their own limitation that the only work for limited purpose, they are complex, work on limited frequency range, the signal used as input is generated by user’s code. However, in our project we used recorded audio signal as input and it is good option to use locally when compared to other by explained condition with above listed project. So it has big role in noise redaction system in communication.

CHAPTER THREE

SYSTEM DESIGN AND METHODOLOGY

3.1 Methodology

This project is done on design of adaptive filters for redaction of noise in communication signal and how adaptive filters work and some of the applications where they can be useful. As the signal into the filter continues, the adaptive filter coefficients adjust themselves to achieve the desired result, such as identifying an unknown filter or canceling noise in the input signal. In the figure below, the shaded box represents the adaptive filter, comprising the adaptive filter and the adaptive recursive least squares (RLS) algorithm. Block diagram that shows the Inputs and Output of a Generic RLS Adaptive Filter noise reduction using adaptive filter and this can be set as: [6]

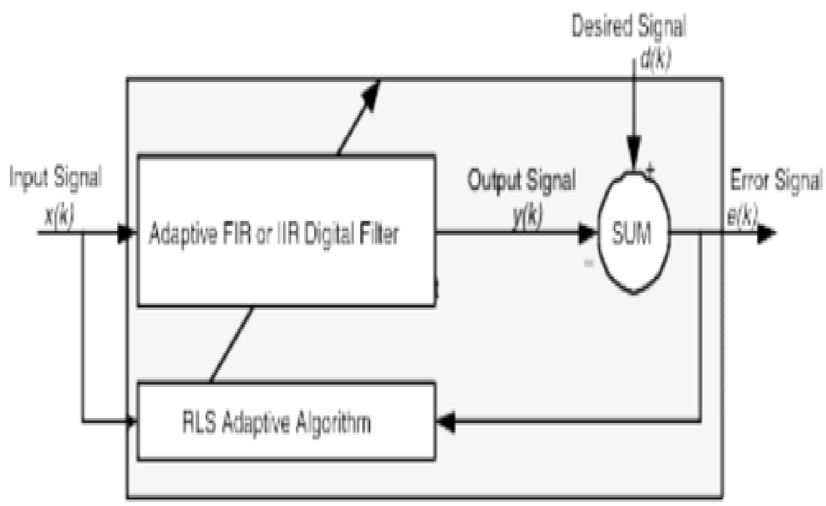

Figure 3.1: Block diagram that shows the Inputs and Output of a Generic RLS Adaptive Filter

The next figure provides the general adaptive filter setup with inputs and outputs

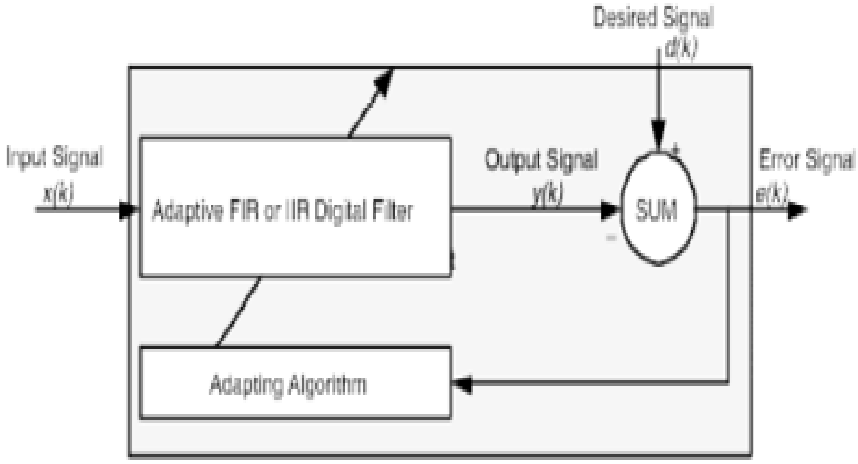

Figure 3.2: Block Diagram Defining General Adaptive Filter Algorithm Inputs and Outputs.

An adaptive filter designs itself based on the characteristics of the input signal to the filter and a signal that represents the desired behavior of the filter on its input. Designing the filter does not require any other frequency response information or specification. To define the self-learning process the filter uses, you select the adaptive algorithm used to reduce the error between the output signal y (k) and the desired signal d(k). When the RLS performance criterion for e(k) has achieved its minimum value through the iterations of the adapting algorithm, the adaptive filter is finished and its coefficients have converged to a solution. Now the output from the adaptive filter matches closely the desired signal d(k). When you change the input data characteristics, sometimes called the filter environment, the filter adapts to the new environment by generating a new set of coefficients for the new data. Notice that when e(k) goes to zero and remains there you achieve perfect adaptation, the ideal result but not likely in the real world. The adaptive filter functions in this toolbox implement the shaded portion of the figures, replacing the adaptive algorithm with an appropriate technique. To use one of the functions, you provide the input signal or signals and the initial values for the filter.

Analysis of Adaptive Filtering Algorithms

The characteristics of the transmission paths or signals are not known and are unpredictable, so filtering and subtraction are controlled by an adaptive process. Hence an adaptive filter is used that is capable of adjusting its impulse response to minimize an error signal, Adaptive Noise Canceling makes possible attainment of noise rejection levels that are difficult or impossible to achieve by direct filtering. The adjustment of the filter weights, and hence the impulse response, is governed by an adaptive algorithm. The most commonly used adaptive algorithm is RLS, LMS, and NLMS but from these algorithms the concerning algorithm for this project is recursive least mean square algorithm (RLMs) algorithm. [7]

Recursive Least Mean Square Algorithm

The recursive least mean square (RLS) adaptive filter is an algorithm that recursively finds the filter coefficients that minimize the weighted linear least square cost function that relates to the input signals. This is in contrast to other algorithms such as the least mean square (LMS), normalized least mean square (NLMS) that aims to reduce the mean square error. In a derivation of RLS, the input signal is considered deterministic, compared to most of its competitors, the RLS exhibits extremely fast convergence, however, these benefits come at the cost of computational complexity.

3.2 System Components and Operations

Adaptive Filter

An adaptive filter is a digital filter that has self-adjusting characteristics. It is capable of adjusting its filter coefficients automatically to adapt the input signal via an adaptive algorithm. Adaptive filters play an important role in modern digital signal processing (DSP) products in areas such as telephone echo cancellation, noise cancellation, equalization of communications channels, biomedical signal enhancement, active noise control (ANC), and systems. It is also a device dedicated to model the relationship between two signals in real time in a computationally iterative manner. Adaptive filters are often realized either as a set of program instructions running on a processing device such as a specific Digital Signal Processing chip(ASIC), or as a set of logic operations implemented in a field programmable gate array (FPGA).

Different systems in adaptive filter

Adaptive System Identification

One common adaptive filter application is to use adaptive filters to identify an unknown system, In the figure, the unknown system is placed in parallel with the adaptive filter. This layout represents just one of many possible structures. The shaded area contains the adaptive filter system. Using an Adaptive Filter to Identify an Unknown System.

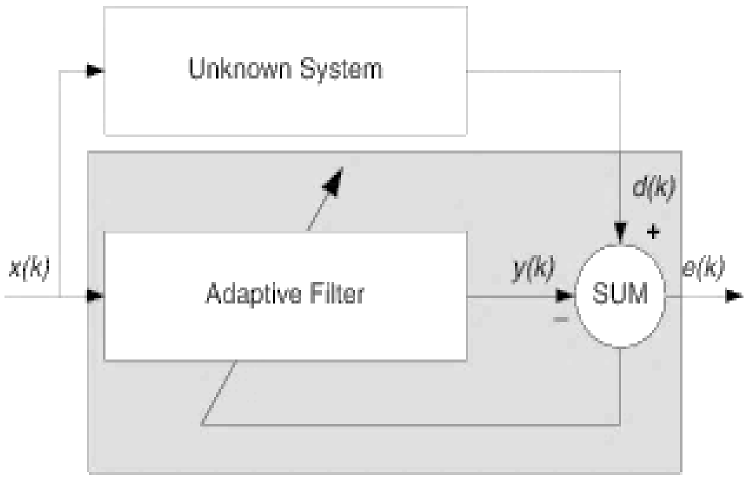

Figure 3.3: system identification system block

Clearly, when e (k) is very small, the adaptive filter response is close to the response of the unknown system. In this case, the same input feeds both the adaptive filter and the unknown Noise Cancellation

In noise cancellation, adaptive filters let you remove noise from a signal in real time. Here, the desired signal, the one to clean up, combines noise and desired information. To remove the noise, feed a signal n'(k) to the adaptive filter that represents noise that is correlated to the noise to remove from the desired signal. [8]

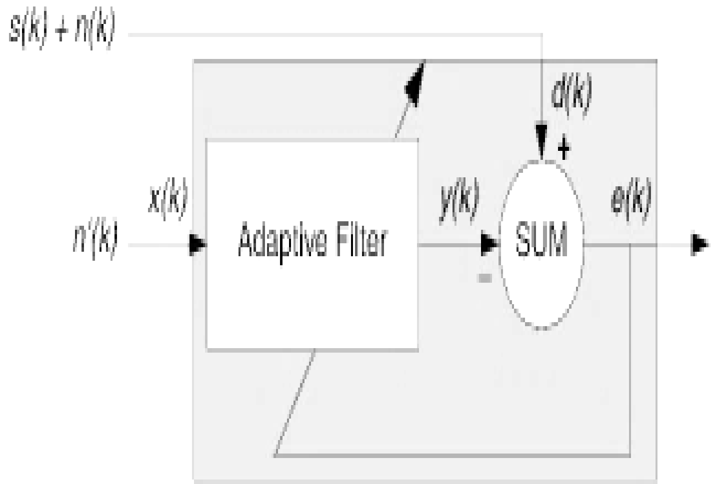

Figure 3.4: noise cancellation system block

A linear Prediction system

Predicting signals requires that you make some key assumptions. Assume that the signal is either steady or slowly varying over time, and periodic over time as well.

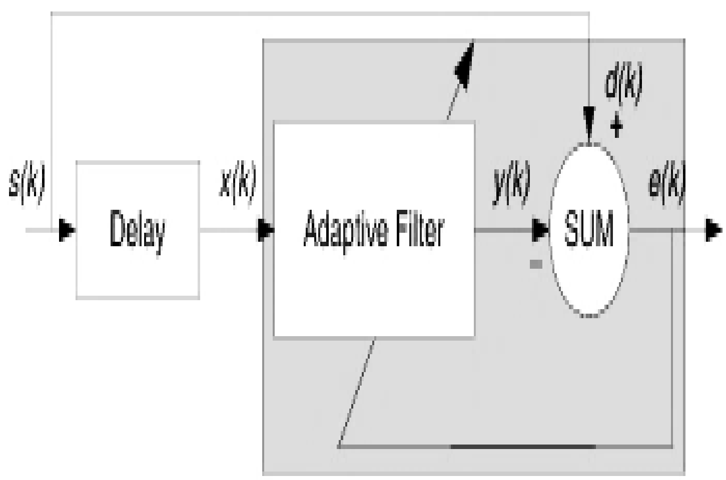

Figure 3.5: linear prediction system block

Accepting these assumptions, the adaptive filter must predict the future values of the desired signal based on past values. When s(k) is periodic and the filter is long enough to remember previous values, this structure with the delay in the input signal, can perform the prediction. You might use this structure to remove a periodic signal from stochastic noise signals. Finally, notice that most systems of interest contain elements of more than one of the four adaptive filter structures. Carefully reviewing the real structure may be required to determine what the adaptive filter is adapting to. Also, for clarity in the figures, the analog-to-digital (A/D) and digital-to analog (D/A) components do not appear. Since the adaptive filters are assumed to be digital in nature, and many of the problems produce analog data, converting the input signals to and from the analog domain is probably necessary.

Adaptive Inverse System Identification

By placing the unknown system in series with your adaptive filter, your filter adapts to become the inverse of the unknown system as e(k) becomes very small. As shown in the figure the process requires a delay inserted in the desired signal d(k) path to keep the data at the summation synchronized. Adding the delay keeps the system causal. Determining an inverse response to an Unknown system.

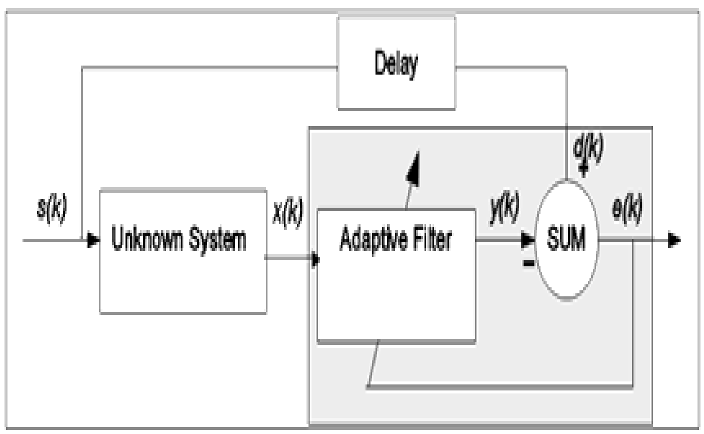

Figure 3.6: adaptive inverse system blocks

The adaptive system identification is primarily responsible for determining a discrete estimation of the transfer function for an unknown digital or analog system. The same input x (n) is applied to both the adaptive filter and the unknown system from which the outputs are compared. The output of the adaptive filter y (n) is subtracted from the output of the unknown 2 system resulting in a desired signal d (n). The resulting difference is an error signal e (n) used to manipulate the filter coefficients of the adaptive system trending towards an error signal of zero.

Performance Measures in Adaptive Systems

There are different performance measures of adaptive system. For example, convergence rate, minimum mean square error, computational complexity and stability can be listed. This shows that the convergence rate can only be considered in relation to the other performance metrics, not by itself with no regards to the rest of the system.

Convergence rate

The convergence rate determines the rate at which the filter converges to its resultant state. Usually a faster convergence rate is a desired characteristic of an adaptive system. Convergence rate is not, however, independent of all of the other performance characteristics. There will be a tradeoff, in other performance criteria, for an improved convergence rate and there will be a decreased convergence performance for an increase in other performance.

Minimum Mean Square Error

The minimum mean square error (MSE) is a metric indicating how well a system can adapt to a given solution. A small minimum MSE is an indication that the adaptive system has accurately modeled, predicted, adapted and/or converged to a solution for the system. A very large MSE usually indicates that the adaptive filter cannot accurately model the given system or the initial state of the adaptive filter is an inadequate starting point to cause the adaptive filter to converge.

Computational Complexity

Computational complexity is particularly important in real time adaptive filter applications. When a real time system is being implemented, there are hardware limitations that may affect the performance of the system. A highly complex algorithm will require much greater hardware resource than simplistic algorithm.

Stability

Stability is the most important performance measure for the adaptive system. By the nature of the adaptive system, there are very few completely asymptotically stable systems that can be realized.

Robustness

The robustness of a system is directly related to the stability of a system. Robustness is a measure of how well the system can resist both input and quantization noise.

Filter Length

The filter length of the adaptive system is inherently tied to many of the other performance measures. The length of the filter specifies how accurately a given system can be modeled by the adaptive filter.

CHAPTER FOUR

RESULT AND DISCUSSION

4.1 Adaptive system identification

Some of the results of nose reduction using Adaptive filter is discussed below as: A fourth order system was created using the unknown system Mat lab program, see figure 7. Using the adaptive system identification program a 7th order system was initialized to zero and the algorithm was performed. In under 200 iterations, see figure 2, the coefficients were determined with an overall error of 3.4486e-006. The coefficients of the system were estimated as shown below. Estimated coefficients = [1.0000, 0.2790,-1.4488,-1.0700, 0.4678, 0.8100,-0.0000, 0.0000].

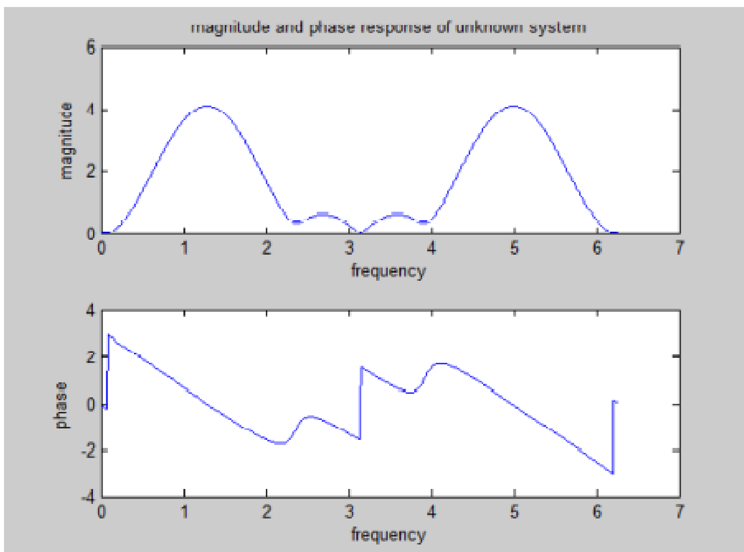

Figure 4.1: Magnitude and Phase Response of the Unknown System

The figure shown below shows that the mean square error that shows how well the adaptive filter can adapt the unknown system and reduce the level of noise with a number of iteration.

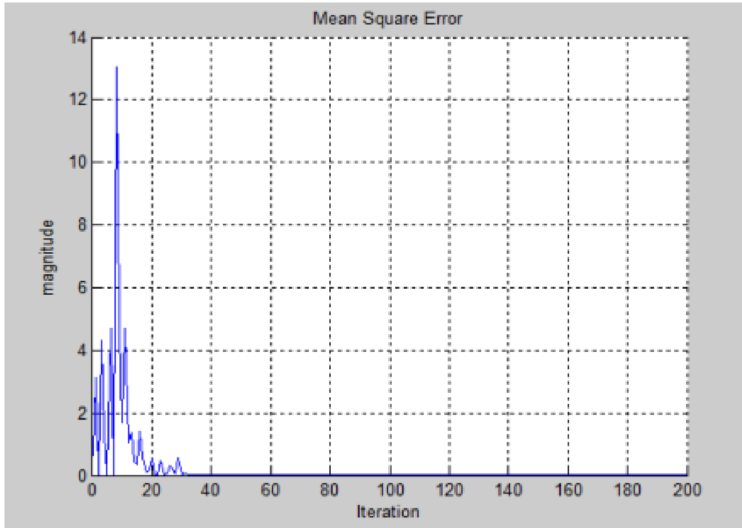

Figure 4.2: MSE verses Number of Iterations.

4.2 Adaptive noise cancellation results

The adaptive noise cancellation system was given an input of a noisy, frequency varying sine wave, and a frequency response of nosy signal. After 1000 iteration of the filter the noise considerably reduced having an output of filter signal with time and frequency equivalent and analyzed the mean square error that shows how an adaptive filter can reduce the noise level with the increasing of an iteration number.

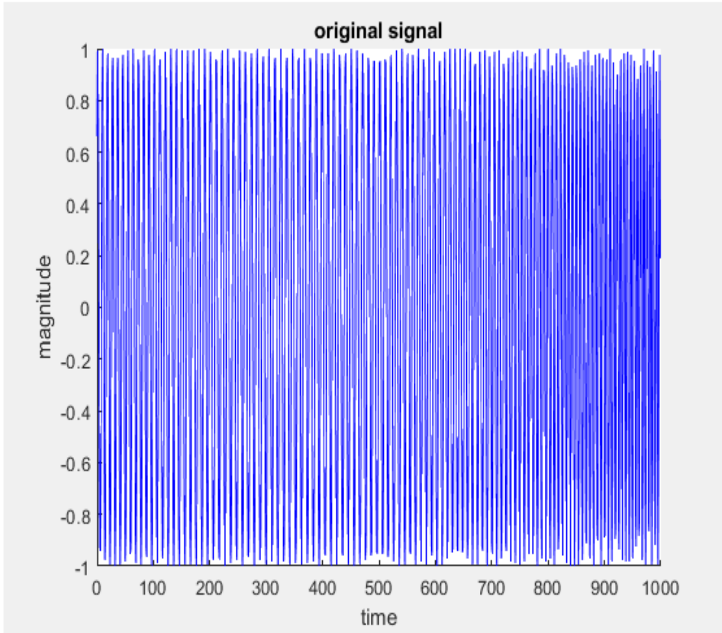

Figure 4.3: Original input signal

From the performance of the adaptive filter that it filter the input noise enter in to the input signal is random white Gaussian which is undeterminstic.

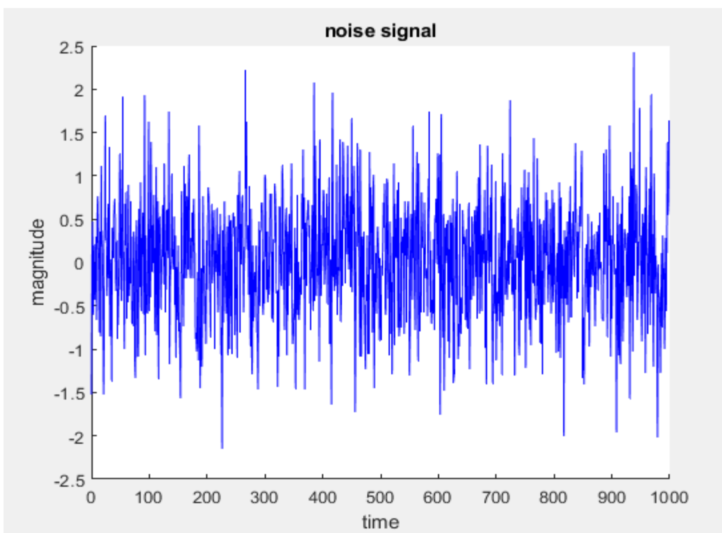

Figure 4.4: White Gaussian Noise signal

The figure shown below is the noisy signal that consists of the frequency varying sine wave and the input noise.

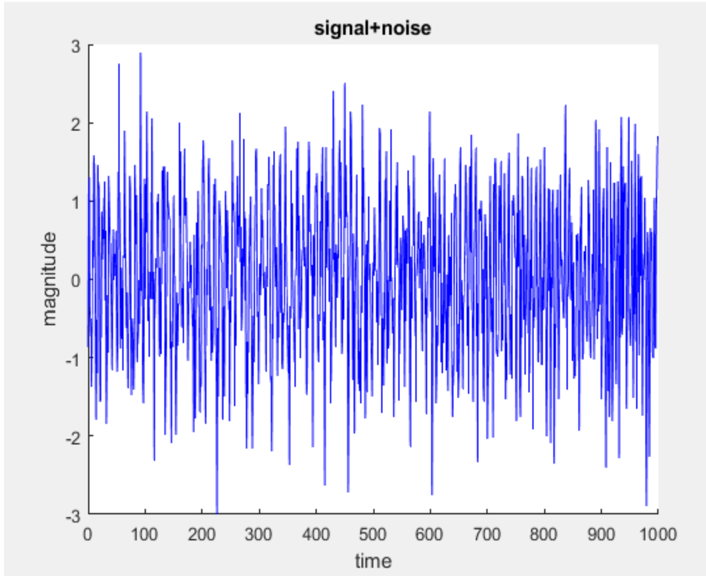

Figure 4.5: Time Response of a Noisy Signal

In order receive the transmitted signal to the receiver side we have to apply filtering so after applying the adaptive filter the noisy signal is recovered as Mach as signal but little difference because of the nature of the error can't be zero.

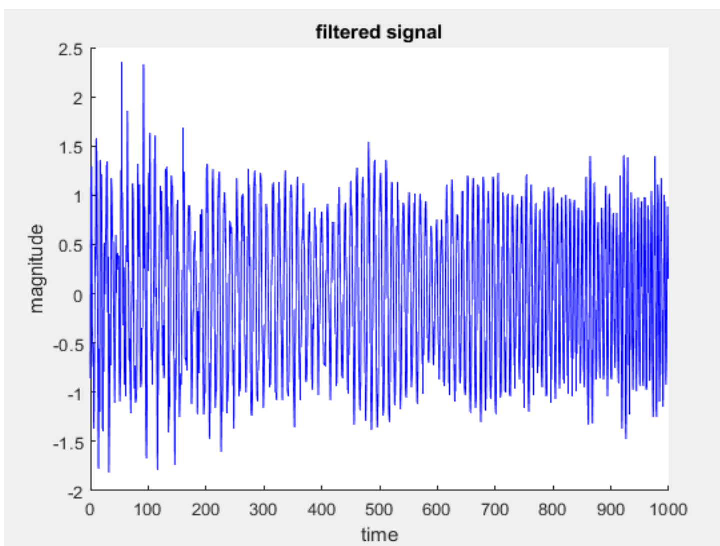

Figure 4.6: Time Response of the Filtered Signal.

Since signals in time domain is infinite we have to express it in to the frequency domain that shows the signal with visualized time.

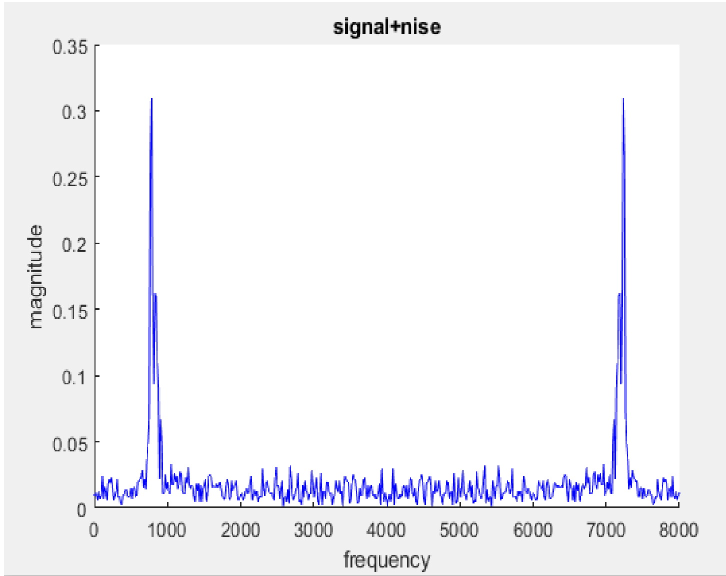

Figure 4.7: Frequency Response of a Noisy Signal.

The figure shown below is the frequency response of filtered signal that equivalently expresses the time response the filtered signal with its advantage of frequency domain.

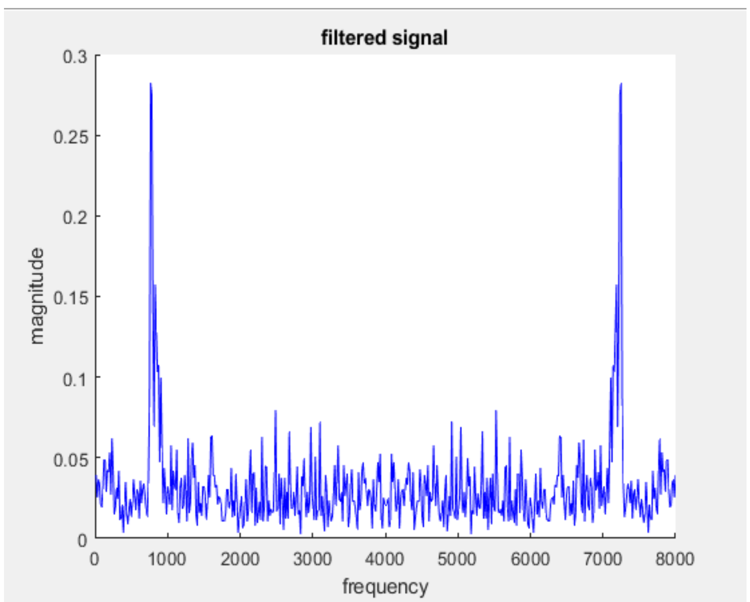

Figure 4.8. Frequency Response of the Filtered Signal.

After a number of iteration is takes place the mean square error is minimum shows that the adaptive filter is effective to reduce the noise is shown in the figure below.

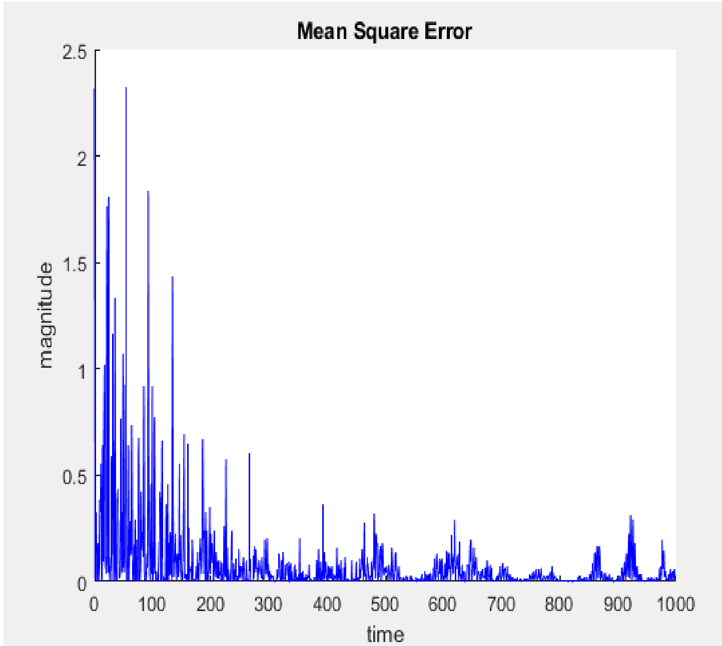

Figure 4.9: MSE verses the Number of Iterations

4.3 Adaptive linear predicting coding results

For this system Time Response of a Noisy Signal shows the input to the adaptive linear predictive coding algorithm.

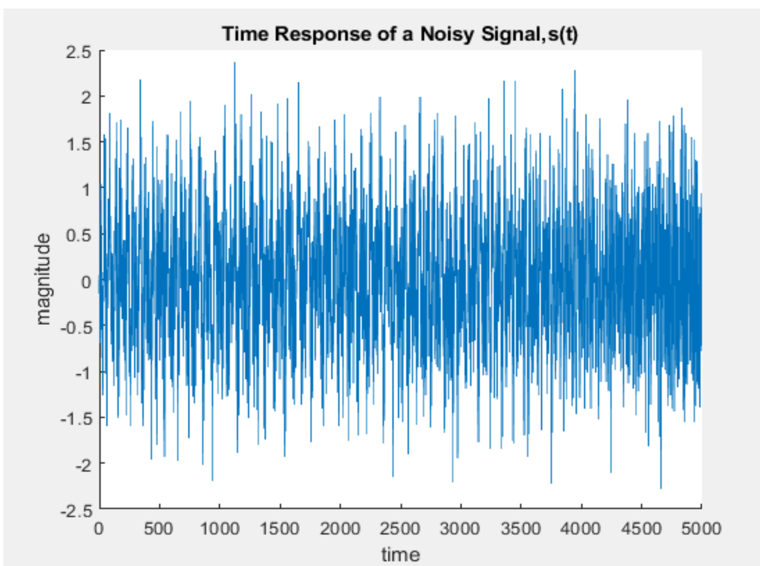

Figure 4.10: Time Response of a Noisy Signal

Time Response of the Filtered Signal shows the output, y (t) of the adaptive LPC system representing the noise cancellation output, where it can be seen that the output is converging to a noiseless system. And the time response of the error signal is shown below as.

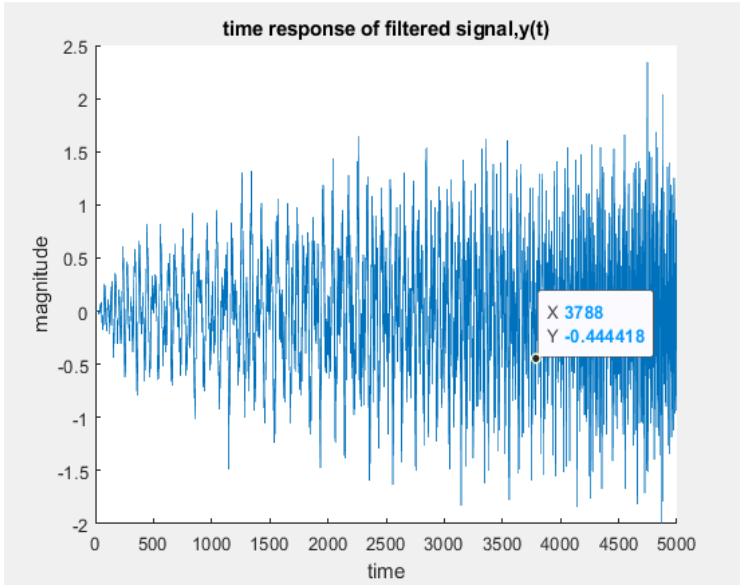

Figure 4.11: Time response of the filtered signal

The figure shown below shows the time response of the error signal

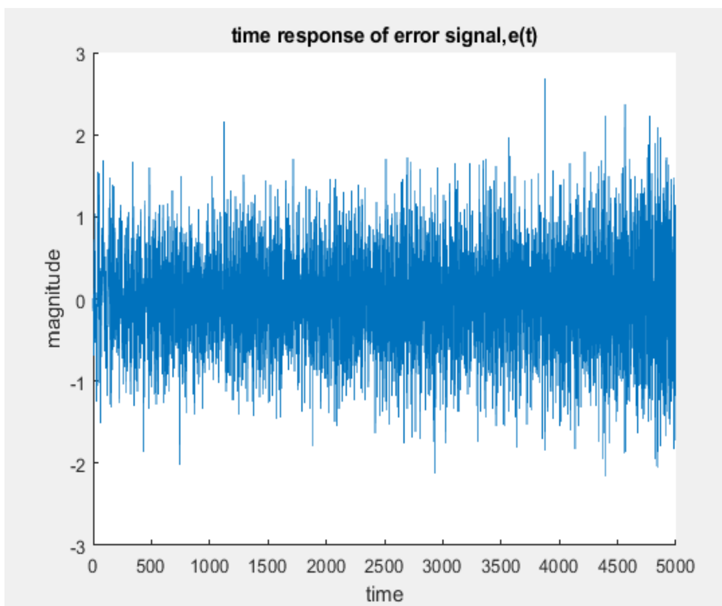

Figure 4.12: Time Response of the Error Signal.

The frequency responses of each of the above inputs and outputs are given as an output of Frequency Response of a Noisy Signal S (W), Frequency Response of the Filtered Signal, and Frequency Response of the Error Signal as below shown.

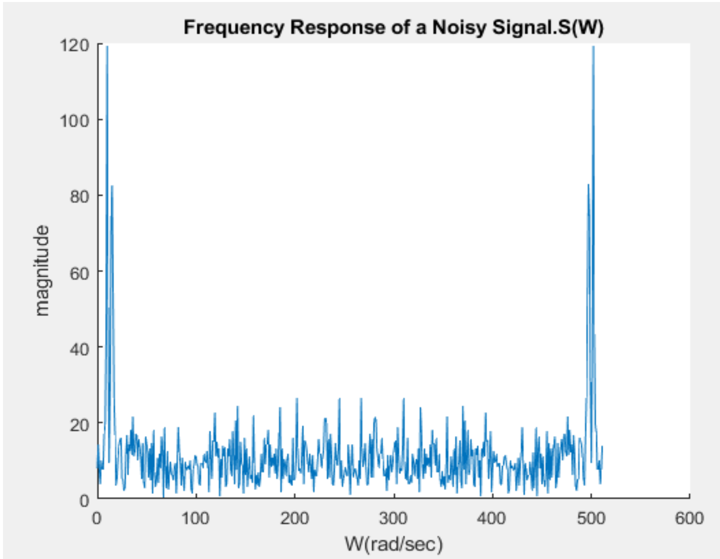

Figure 4.13: Frequency Response of a Noisy Signal.

The figure shown below is the frequency response of filter signal

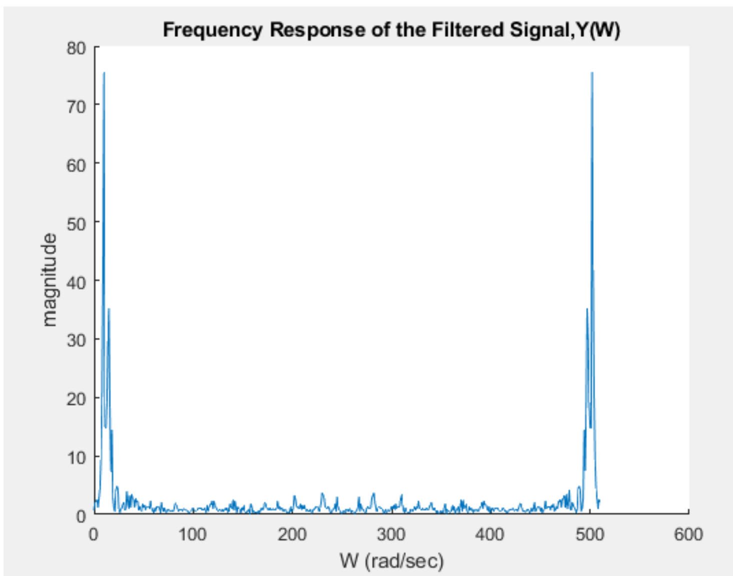

Figure 4.14: Frequency Response of the Filtered Signal.

The figure shown below is the frequency response that equivalently expresses the time response of the error signal.

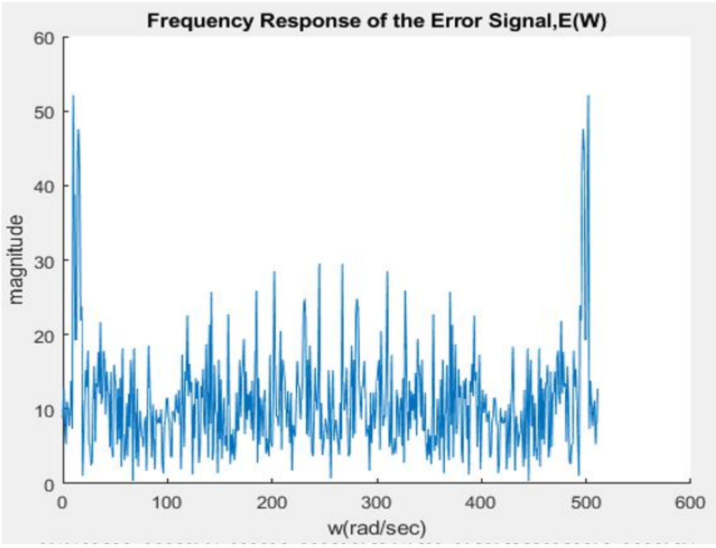

Figure 4.15: Frequency Response of the Error Signal

4.4 Adaptive inverse system

In this system the frequency response of unknown system shown in figure 22. The Adaptive system algorithm is performed on this system and results Frequency Response of the Adaptive System. Frequency Response of the Corrected System. Shows the combination of the 2 responses in an attempt to equalize the frequency response of the original unknown system the three outputs are shown below as:

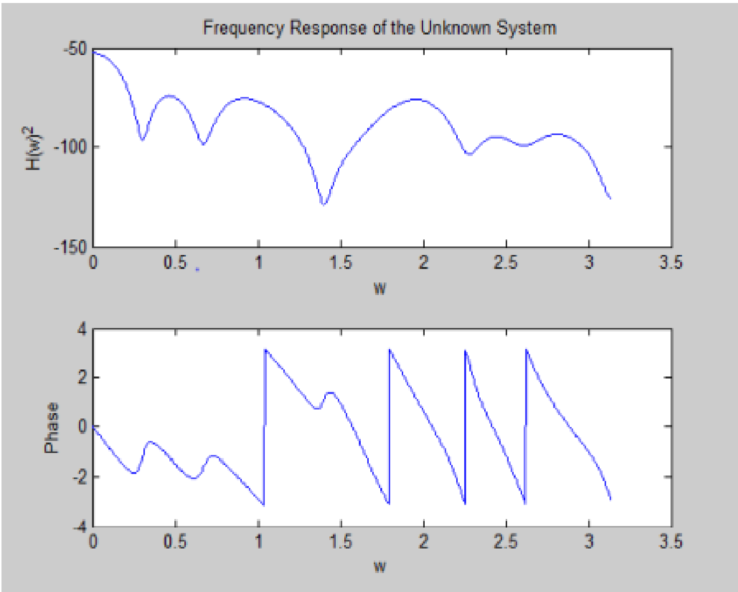

Figure 4.16: Frequency Response of the Unknown System.

The frequency response of adaptive system is applied to correct of the frequency response of the unknown system that is not deterministic to the inputs.

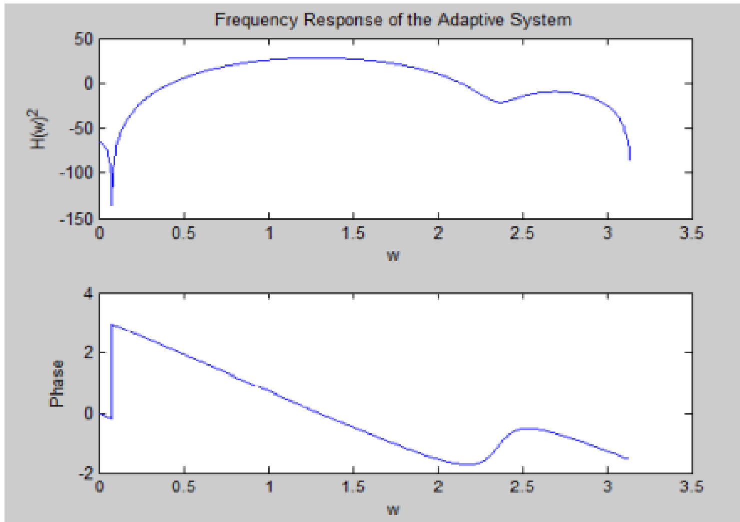

Figure 4.17: Frequency Response of the Adaptive System.

The frequency response of the corrected system is the combination of the unknown system and adaptive system of the frequency responses.

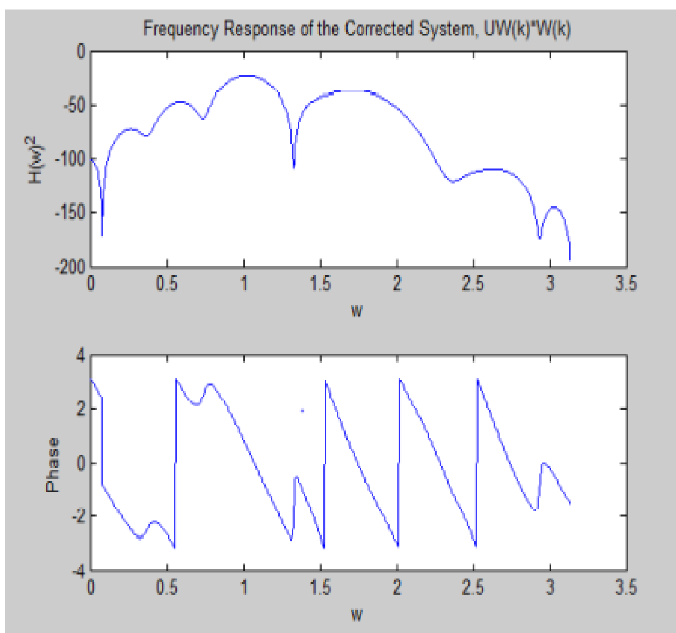

Figure 4.18: Frequency Response of the Corrected System.

CHAPTER FIVE

CONCLUSION AND RECOMMENDATION

5.1 Conclusion

In this project adaptive noise redaction has been successfully designed and simulated. The designed system can perform the noise reducing or minimization in simple and clear way. Therefore, the project is understandable. Also the algorithm used is easy and it can be worked by simple way. This makes the project preferable and used for many proposes. Going through the planning, flow process, design and software simulation, the system has been good one; the chapter's one up to five has actually tried as much as possible to explain strongly almost all what is involved in the simulation of this project.

To sum up this project is done by using mat lab. The audio signal is generated by recording any voice. The spectral analysis of the signal is showed and the noise is randomly added to signal then the spectral analysis of signal plus noise is plotted finally there is spectral showed the noise free signal. After the complete design of the system, the expected result and the actual result was very close. So, to implement such an idea will be useful and it is important to use in different applications area.

5.2 Recommendation

The possible modification that can be made in the project for improving the performance of reducing the noise from the noisy environment. The application can be extended for the noise reduction in the speech for hearing aids in the noisy environment like crowed noise, car noise aircraft noise etc. With modified RLS algorithm convergence speech can be increased as per the real time requirement fast algorithm can be developed. The one who can do for the future must consider noise reduction using adaptive filter not only simulating in software for subjective case but also for practical application in real time by hardware.

REFERENCES

[1] S.I.A.Sugiyama, "an adaptive noise canceller low signals distortion," *IEEE Trans.Signal processing,* vol. vol. 47, p. pp. 665 , mar 1999.

[2] B. S.Rajvansh Sehamby, " “Noise Cancellation using Adaptive Filtering in ECG Signals”," *International Journal of Bio-Science and Bio-Technology, ,* Vols. Vol.8,, November,2016..

[3] L.-e.Ia-Haw Lee, “simulation of the noise cancellation using LMS adaptive filter algorithm”, Tunku Abdul Rahman University College, , May, 2017..

[4] S. P.Jonathan Cedarleaf, “Noise cancellation using least mean squares adaptive filter”, University of Rochester, Department of Electrical and Computer Engineering,, May, 2017..

[5] S. D.Ravikanth, "“Design and Development of Noise Cancellation System for Android Mobile Phones”," *School of Advanced Studies of Bangalore,,* Vols. Vol.11, , Apr, 2012..

[6] S. Haykin, "Adaptive Filter Theory," *Prentice Hall, Upper Saddle River, New Jersey.*

[7] M. L. M. D. G. Honig, "Adaptive Filters, Structures, Algorithms and Applications," *Kluwer Academic Publishers, Boston,* 1984.

[8] S.I.A.Sugiyama, "an adaptive noise canceller low signals distortion," *IEEE Trans.Signal processing,* vol. vol. 47, p. pp. 665 , mar 1999.

APPENDIX

MATLAB CODE

```
% Adaptive System Identification
clear
INPUTVEC = 0;
mu= .1;
len = 8;
lin = 200;
xin = randn(lin,1);
mse = 0;
x = 0*(1:len)';
w = 0*(1:len);
d = double(x(1));
y = w*x;
e = d-y;
mse(1) = e^2;
w = w + mu*e*x';
for j = 1:lin
for n = len:-1:2
x(n) = x(n-1);
end
x(1) = xin(j);
d = double(x(1));
y = w*x;
e = d-y;
mse(j+1) = e^2;
w = w + mu*e*x';
end
figure(1)
hold on
[H,f] = freqz(w,1,512,'whole');
subplot(2,1,1),plot(f,abs(H),'b')
title('magnitude and phase response of unknown system')
xlabel('frequency')
ylabel('magnitude')
subplot(2,1,2),plot(f,angle(H),'b')
xlabel('frequency')
ylabel('phase')
num = 0:length(mse)-1;
figure (2)
hold on
grid on
title('Mean Square Error')
ylabel('magnitude')
xlabel('Iteration')
plot(num,mse,'b')
disp('Estimated coeficients w =')
```

```
disp(w')
disp('Final error e =')
disp(abs(e))
% Adaptive noise cancellation
clear
mu= .01;
len = 40;
lin = 1000;
n1 = 0.7*randn(lin,1);
n0 = n1;
mse = 0;
for n=1:lin
s(n) = sin(2*pi*(n)/abs(10*sin(pi*(1000+n)/3000)));
end
sn = s'+n1;
x = 0*(1:len)';
w = 0*(1:len);
y = w*x;
e = n0(1)-y;
mse(1) = e;
w = w + mu*e*x';
for j = 1:lin
for n = len:-1:2
x(n) = x(n-1);
end
x(1) = n0(j);
y = w*x;
e = sn(j)-y;
mse(j) = e;
w = w + mu*e*x';
end
f=0:length(s)-1;
W = fft(w,512);
figure(3)
hold on
title('original signal')
xlabel('time')
ylabel('magnitude')
plot(f,s,'b')
num = 0:length(mse)-1;
f=0:length(n1)-1;
W = fft(w,512);
figure(4)
hold on
title('noise signal')
xlabel('time')
ylabel('magnitude')
plot(f,n1,'b')
num = 0:length(mse)-1;
f = 0:length(sn)-1;
```

```
W = fft(w,512);
figure(5)
hold on
title('signal+noise')
xlabel('time')
ylabel('magnitude')
plot(f,sn,'b')
num = 0:length(mse)-1;
figure (6)
hold on
title('filtered signal')
xlabel('time')
ylabel('magnitude')
plot(num,mse,'b')
t = 0:8000/(length(W)-1):8000;
figure (7)
hold on
title('signal+nise')
xlabel('frequency')
ylabel('magnitude')
plot(t,abs(fft(mse,512))./512,'b')
figure(8)
hold on
title('filtered signal')
xlabel('frequency')
ylabel('magnitude')
plot(t,abs(fft(sn,512))./512,'b')
figure(9)
hold on
title('Mean Square Error')
xlabel('time')
ylabel('magnitude')
plot(f,(s-mse).^2,'b')
disp('Estimated coeficients w =')
disp(w')
disp('Final error e =')
disp(abs(e))
snr = 10*log10(sum(abs(fft(s)).^2)/sum(abs(fft(sn-s')).^2))
snrf = 10*log10(sum(abs(fft(s)).^2)/sum(abs(fft(mse-s)).^2))

% Adaptive Linear Predictive Coding
mu= .0007;
len = 100;
lin = 5000;
n0 = .5*randn(lin,1);
a = 0;
mse = 0;
s = 0;
for n=1:(lin);
s1(n) = 1*sin(2*pi.*n/abs(50*sin(pi*(2500.+n)/10000))).*sin(2*pi*n/200);
```

```
end
s = s1+n0';
x = 0*(1:len)';
w = 0*(1:len);
for j = 1:lin
y = w*x;
a(j) = y;
for n = len:-1:2
x(n) = x(n-1);
end
x(1) = s(j);
e = x(1)-y;
mse(j+1) = e;
w = w + mu*e*x';
end
f = 0:length(a)-1;
figure(10)
hold on
title('time response of filtered signal,y(t)')
xlabel('time')
ylabel('magnitude')
plot(f,a)
l = 0:len-1;
num = 0:length(mse)-1;
figure (11)
hold on
title('time response of error signal,e(t)')
plot(num,mse)
xlabel('time')
ylabel('magnitude')
W = fft(s,512);
figure (12)
hold on
title('Frequency Response of a Noisy Signal.S(W)')
t = 0:length(W)-1;
plot(t,abs(W))
xlabel('W(rad/sec)')
ylabel('magnitude')
figure (13)
hold on
title('Time Response of a Noisy Signal,s(t)')
t = 0:length(s)-1;
plot(t,s)
xlabel('time')
ylabel('magnitude')
W = fft(a,512);
figure (14)
hold on
title('Frequency Response of the Filtered Signal,Y(W)')
t = 0:length(W)-1;
```

```
plot(t,abs(W))
xlabel('W (rad/sec)')
ylabel('magnitude')
W = fft(mse,512);
figure (15)
hold on
title('Frequency Response of the Error Signal,E(W)')
t = 0:length(W)-1;
plot(t,abs(W))
xlabel('w(rad/sec)')
ylabel('magnitude')

% Adaptive Inverse System
INPUTVEC = 0;
mu= .005;
len = 20;
lin = 5000;
mse = 0;
x = 0*(1:len)';
w = 0*(1:len);
x1 = 0*(1:len)';
n0 = .1*randn(lin,1);
n = 0:lin-1;
xin = n0'+sin(2*pi*n./(sin(2*pi*(n+1)/11000)));
for j = 1:lin
for n = len:-1:2
x1(n) = x1(n-1);
end
x1(1) = double(x(1));
b(j) = x1(1);
y = w*x1;
a(j) = y;
e = x(1)-y;
mse(j+1) = e^2;
w = w + mu*e*x1';
for n = len:-1:2
x(n) = x(n-1);
end
x(1) = xin(j);
end
w'
[H,f] = freqz(w,1,512);
figure(16)
hold on
subplot(2,1,1), plot(f,20*log(abs(H)),'b')
title('Frequency Response of the Unknown System')
ylabel('H(w)^2')
xlabel('w')
subplot(2,1,2), plot(f,angle(H),'b')
ylabel('Phase')
```

```
xlabel('w')
num = 0:length(a)-1;
figure (17)
hold on
grid on
title ('y(n)')
plot(num,a,'b')
xlabel('n,ieration')
ylabel('magnitude')
figure (18)
hold on
grid on
title ('x(n)')
plot(num,xin,'b')
xlabel('n,iteration')
ylabel('ampliude')
figure(19)
w1 = [1 0.2790 -1.4488 -1.0700 0.4678 0.8100];
hold on
[H,f] = freqz(w1,1,512);
subplot(2,1,1), plot(f,20*log(abs(H)),'b')
title('Frequency Response of the Adaptive System')
ylabel('H(w)^2')
xlabel('w')
subplot(2,1,2), plot(f,angle(H),'b')
ylabel('Phase')
xlabel('w')
figure(20)
hold on
grid on
w2 = conv(w1,w);
[H,f] = freqz(w2,1,512);
subplot(2,1,1), plot(f,20*log(abs(H)),'b')
title('Frequency Response of the Corrected System, UW(k)*W(k)')
ylabel('H(w)^2')
xlabel('w')
subplot(2,1,2), plot(f,angle(H),'b')
ylabel('Phase')
xlabel('w')
disp('Final error e =')
disp(abs(e))
% Unknown System Function
function [dn] = double(xn)
global INPUTVEC
uw = [1 0.2790 -1.4488 -1.0700 0.4678 0.8100]; % System
Coefficients
len = length(uw);
INPUTVEC(1) = xn;
if length(INPUTVEC) ~= len
INPUTVEC(len) = 0;
```

```
end
dn = uw*INPUTVEC';
for i = len:-1:2
INPUTVEC(i) = INPUTVEC(i-1);
End
% Unknown System Function 2
function dn = double(xn)
global INPUTVEC
uw = [1 0.2790 -1.4488 -1.0700 0.4678 0.8100]; % System
Coefficients
len = length(uw);
if length(INPUTVEC) ~= len
INPUTVEC(len) = 0;
end
dn = uw*INPUTVEC';
for i = len:-1:2
INPUTVEC(i) = INPUTVEC(i-1);
end
end
```