RVSCET

R.V.S. COLLEGE OF ENGINEERING AND TECHNOLOGY

DEPARTMENT OF ELECTRONICS AND COMMUNICATION ENGINEERING

Control and Estimation of Voice Signal Using Adaptive Filter

By :

1. ABHISHEK KR. SIMRA - 182308351233
2. ADITYA AMAN - 182308351234
3. SUDHANSHU GUPTA - 182308351250

Under the sincere guidance of :

Dr.Shalini Kumari
(Asst. Professor)

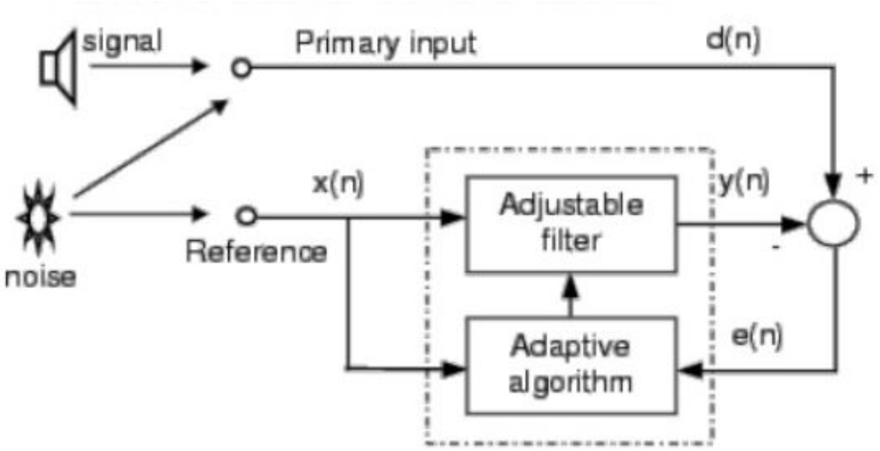
Presentation Outlines :

- Introduction
- Motivation
- Literature Review
- Objective
- Methodology/Algorithm
- Experimental Results
- Discussion of Result
- Conclusion
- Reference

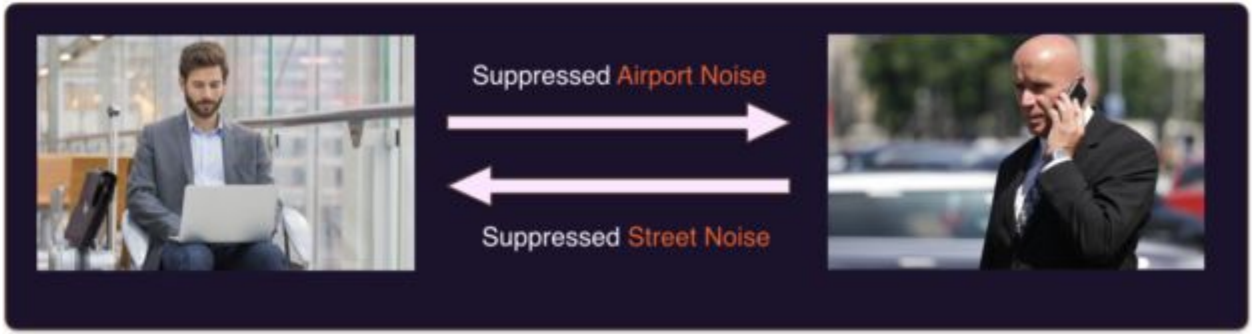
Introduction :

- **DSP :** Digital Signal Processors (DSP) take real-world signals like voice, audio, video, temperature, pressure, or position that have been digitized and then mathematically manipulate them. A DSP is designed for performing mathematical functions like "add", "subtract", "multiply" and "divide" very quickly.

- **Adaptive Filter :** An adaptive filter is a system with a linear filter that has a transfer function controlled by variable parameters and a means to adjust those parameters according to an optimization algorithm. Because of the complexity of the optimization algorithms, almost all adaptive filters are digital filters.



Motivation :



Noise has long been an enemy of human beings. It can disrupt communications, impair concentration, and even damage the auditory system if it is sufficiently loud. The traditional noise cancellation idea involves energy absorption: by strategically placing damping material close to the noise source, much of the sound energy can be converted to heat, therefore reducing the volume of the sound. The soft, porous, multi-layer material covering dance room and studio room walls can damp most of the energy of noise.

Literature Review :

Sl.No.	Author	Year of Publication	Summary
1.	S M Kuo and D.R Morgan	1999	The Acoustic Noise Control traditionally involves passive methods such as enclosures, barriers and silencers to attenuate noise. These methods are however not effective for low frequency noise. A technique to overcome this problem is Active Noise Cancellation (ANC). ANC is an electro acoustic system that cancels the primary unwanted noise by introducing a canceling “antinoise” of equal amplitude but opposite phase.
2.	G.Aishwarya, Abhishek Singh	2020	The need of adaptive filtering arises when the received signal in its course of propagation is corrupted by the noise which changes continuously. The existence of noise affects the processing and application of the signal.This review presents a survey to know the work done on different adaptive algorithms LMS, NLMS, RLS which are applied in the fields of signal processing, communication, signal control applications.
3.	Shubhra Dixit , Deepak Nagaria	2017	This paper reviews the past and the recent research on Adaptive Filter algorithms based on adaptive noise cancellation systems.. Algorithms such as LMS and RLS proves to be vital in the noise cancellation are reviewed including principal and recent modifications to increase the convergence rate and reduce the computational complexity for future implementation. The purpose of this paper is not only to discuss various noise cancellation LMS algorithms but also to provide the reader with an overview of the research conducted.

Sl.No.	Author	Year of Publication	Summary
4.	M.E. Hawley and E.D. Smshauser	1961	Considering the fact of having developed an original project without any guideline and using new programming languages and tools, the satisfaction is absolute. Even if the project has needed to be reconducted into developing a software to test different ANC approaches, the resulting application has resulted very useful to carry out the wanted experimentations. The developed tools like the experimentation selection and setup, the equalizer, the signal generator, etc. have permitted to study different aspects of the performance of some ANC approaches with different complexities.
5.	Simon Haykin and Bernard Widrow	2003	In this paper, the basic adaptive algorithm for ANC is developed and analyzed based on single-channel broad-band feedforward control. This algorithm is then modified for narrow-band feedforward and adaptive feedback control. In turn, these single-channel ANC algorithms are expanded to multiple-channel cases. Various online secondary-path modeling techniques and special adaptive algorithms, such as lattice, frequency-domain, subband, and recursive-least-squares, are also introduced. Applications of these techniques to actual problems are highlighted by several examples.

Objective :

- To control the amount of noise in voice signal with the help of Adaptive filter in MATLAB.
- To use Least Mean Square algorithm and Normalised Least Mean Square algorithm to implement Noise Cancellation in voice signal.
- To compare results from Least Mean Square algorithm and Normalised Least Mean Square algorithm to find out the better performing algorithm.


HOW
DOES IT
WORK

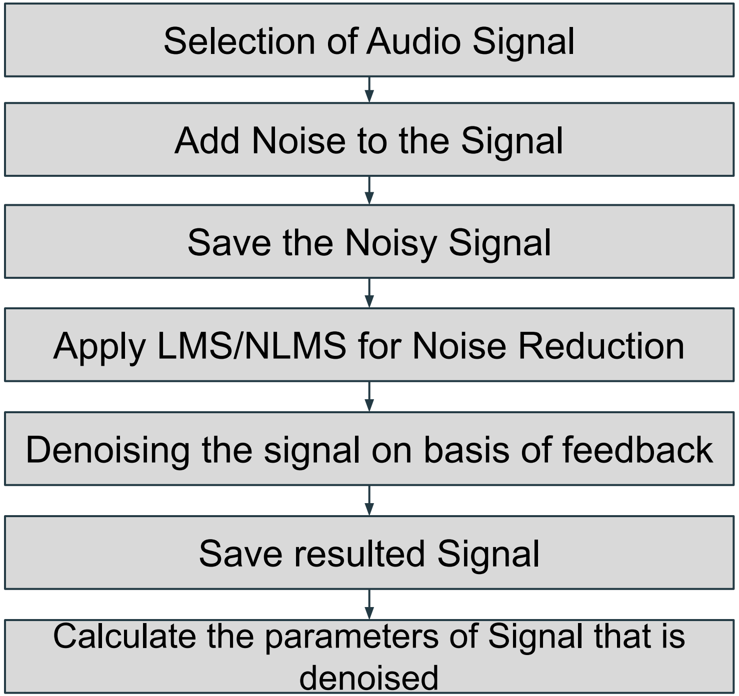
Methodology :

MATLAB & SIMULINK : The MATLAB & SIMUlINK will be used to design the prototype of the model in the simulink part, and the codes for ANC will be done in the MATLAB part, and all together will help to build the project work.

LMS Filter : The LMS filter algorithm will be used to filter the noise signal to get audio output .

NLMS Filter : The NLMS filter algorithm will be used to filter the noise signal and getting the clear audio output

Algorithm :

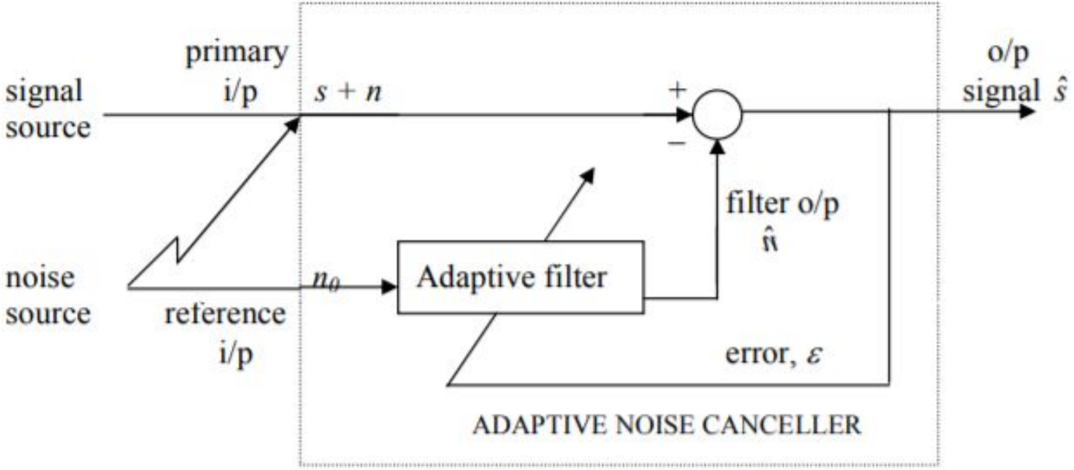

Selection of Audio Signal
Add Noise to the Signal
Save the Noisy Signal
Apply LMS/NLMS for Noise Reduction
Denoising the signal on basis of feedback
Save resulted Signal
Calculate the parameters of Signal that is denoised

Project approach :

The usual method of estimating a signal corrupted by additive noise is to pass it through a filter that tends to suppress the noise while leaving the signal relatively unchanged i.e. direct filtering.

Figure : General Filter Block Diagram

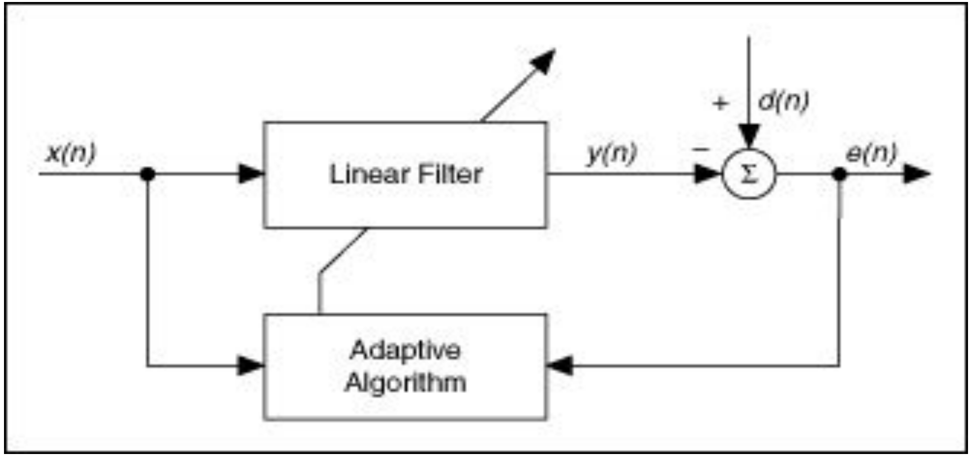
Adaptive Noise Cancellation – Principles



As shown in the figure, an Adaptive Noise Canceller (ANC) has two inputs – primary and reference. The primary input receives a signal s from the signal source that is corrupted by the presence of noise n0 uncorrelated with the signal. The reference input receives a noise n0 uncorrelated with the signal but correlated in some way with the noise n. The noise no passes through a filter to produce an output ˆn that is a close estimate of primary input noise. This noise estimate is subtracted from the corrupted signal to produce an estimate of the signal at sˆ , the ANC system output.

The Least-Mean-Square (LMS) Algorithm

The least-mean-square (LMS) algorithm as an application of the method of stochastic gradient descent that was presented in the preceding chapter. Specifically, we will expand on why the LMS algorithm is of fundamental importance in linear adaptive filtering in theory as well as application.



where $x(n)$ is the input signal to a linear filter

$y(n)$ is the corresponding output signal

$d(n)$ is an additional input signal to the adaptive filter

$e(n)$ is the error signal that denotes the difference between $d(n)$ and $y(n)$.

Least-Mean-Square (LMS) Algorithm Formula

$$\begin{pmatrix}\text{updated}\\ \text{tap-weight}\\ \text{vector}\end{pmatrix} = \begin{pmatrix}\text{old}\\ \text{tap-weight}\\ \text{vector}\end{pmatrix} + \begin{pmatrix}\text{learning-}\\ \text{rate}\\ \text{parameter}\end{pmatrix} \times \begin{pmatrix}\text{tap-}\\ \text{input}\\ \text{vector}\end{pmatrix} \times \begin{pmatrix}\text{error}\\ \text{signal}\end{pmatrix}.$$

$$w(n+1)=w(n)+\mu * x(n) * e(n)$$

Where : w(n+1)= Updated tap-weight vector

μ= Learning rate parameter (step size)

x(n)= Tap- input vector (filter input)

e(n)= Error signal

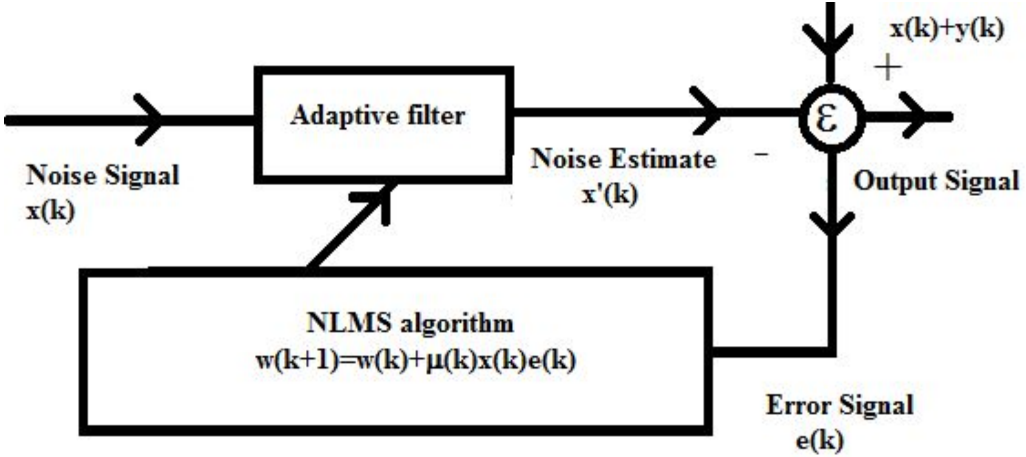
Normalized Least-Mean-Square (NLMS) Algorithm

In the traditional form of a least-mean-square (LMS) algorithm the adjustment applied to the tap-weight vector of the filter at adaptation cycle n + 1 consists of the product of three terms:

• The step-size parameter μ, which is under the designer's control.

• The tap-input vector u(n), which is supplied by a source of information.

• The estimation error e(n) for real-valued data, or its complex conjugate e*(n) for complex-valued data, which is calculated at adaptation cycle n.



Normalised Least-Mean-Square (NLMS) Algorithm Formula

$$w(n+1) = w(n) + \mu(n)x(n)e(n)$$

Where : $w(n+1)$= Updated tap-weight vector

$\mu(n)$= adaptive step size

$x(n)$= Tap- input vector (filter input)

$e(n)$= Error signal

Experimental Results

Simulation using LMS filter Approach

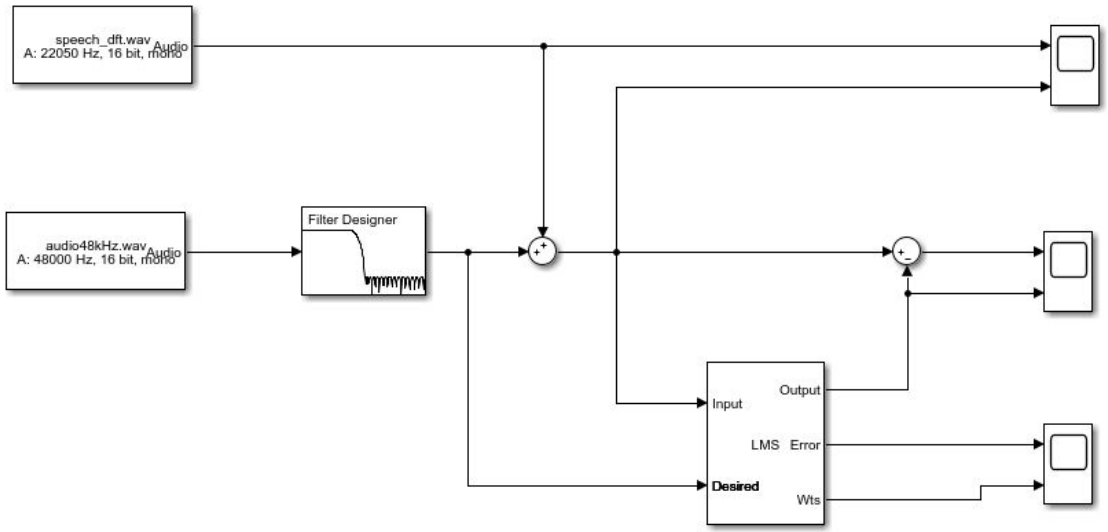


Figure : LMS Simulation Block

Respective results using LMS filter are:

Fig. shows the original audio signal which is used for the project , the original audio signal is free from all kind of noise and is fully understandable.

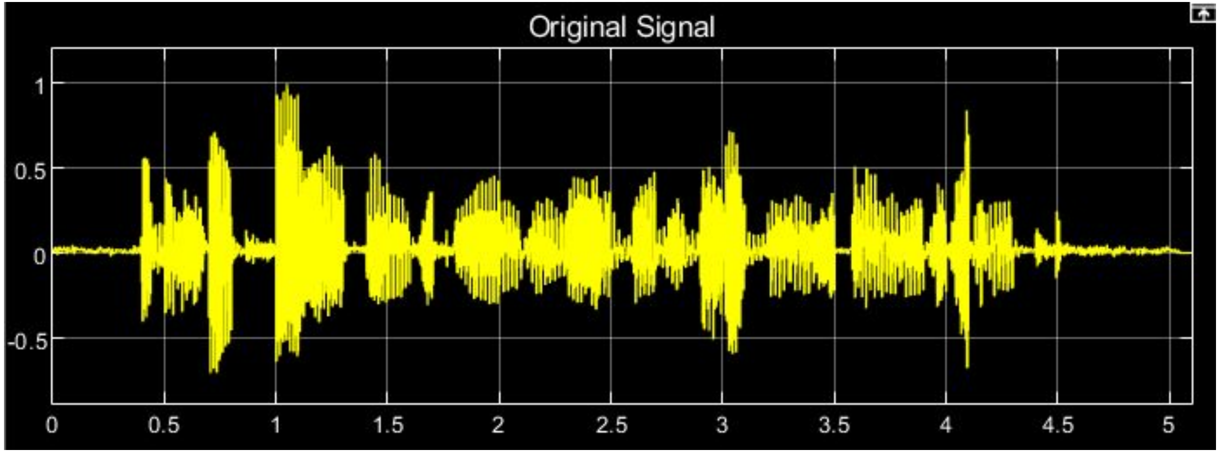


Figure : LMS Simulation Original audio signal

Fig. shows the audio signal in which the noise was mixed and the audio signal was corrupted and was made unclear to understand and makes it difficult to understand.

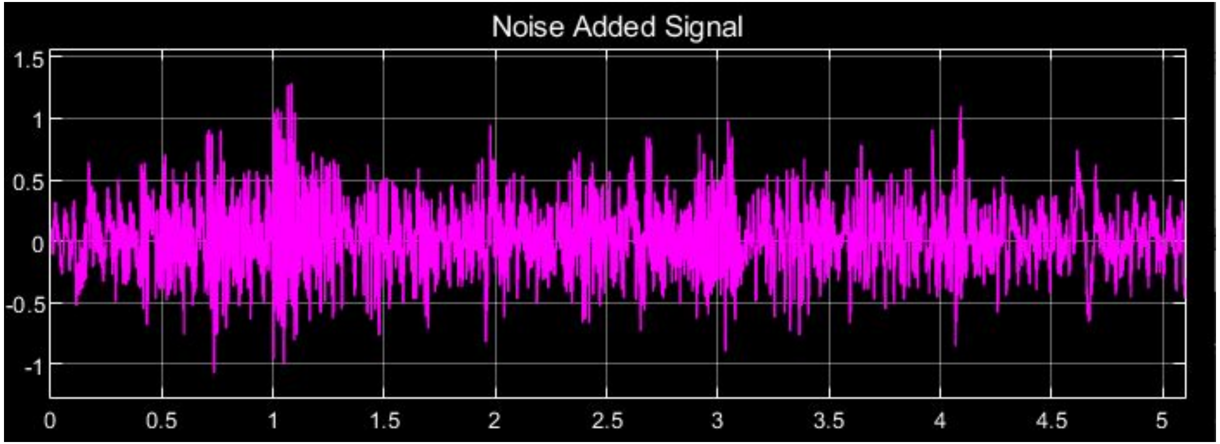


Figure : LMS Simulation Noise Added signal

Fig. shows the recovered audio signal filtered from the lms filter , but the program gets corrupted due to the non adjustable step size of the filter for different high frequency noise and the filter stops the separation of noise .

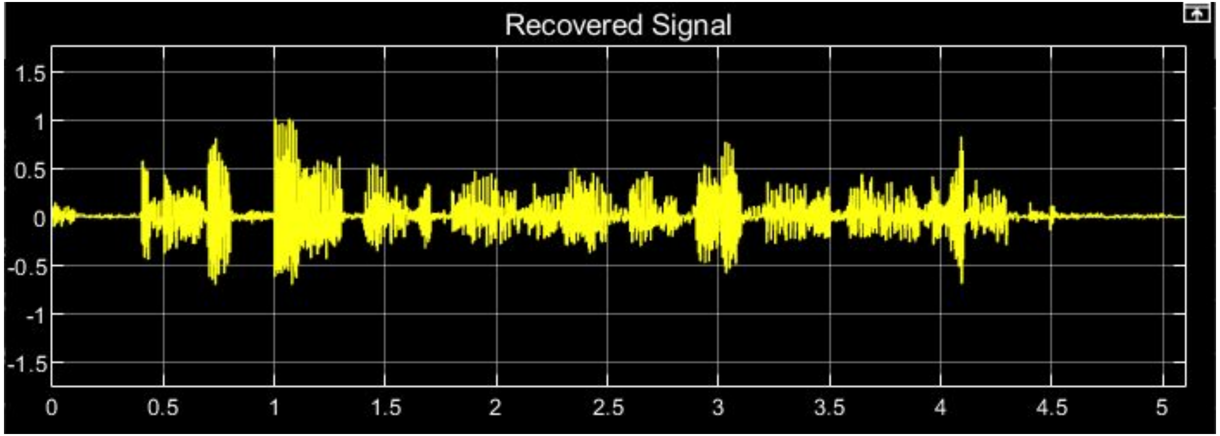


Figure : LMS Simulation Recovered signal

Fig. shows the separated noise signal filtered from the lms filter , but the program gets corrupted due to the non adjustable step size of the filter for different high frequency noise and the filter stops the separation of noise .

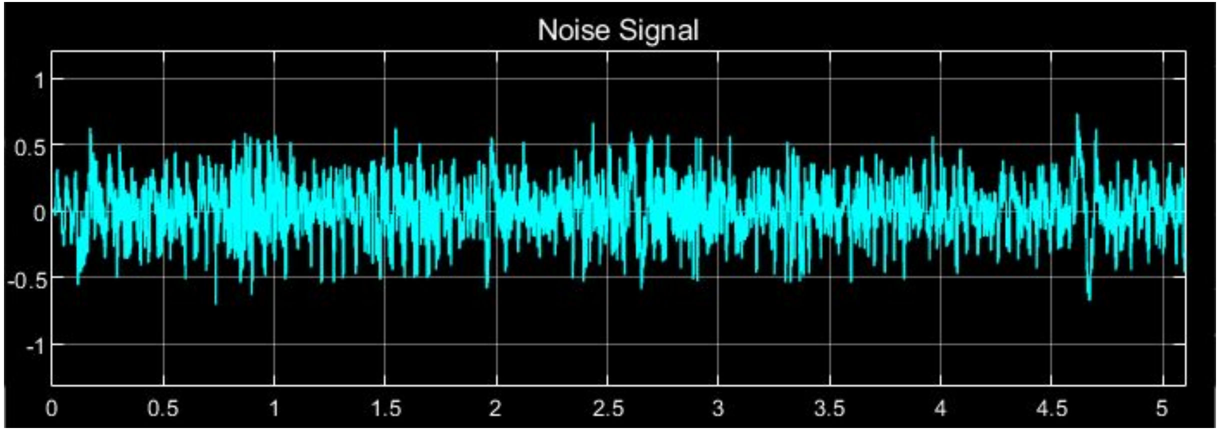


Figure : LMS Simulation Noise signal

Fig. shows the Error signal filtered from the lms filter , but the program gets corrupted due to the non adjustable step size of the filter for different high frequency noise and the filter stops the separation of noise .

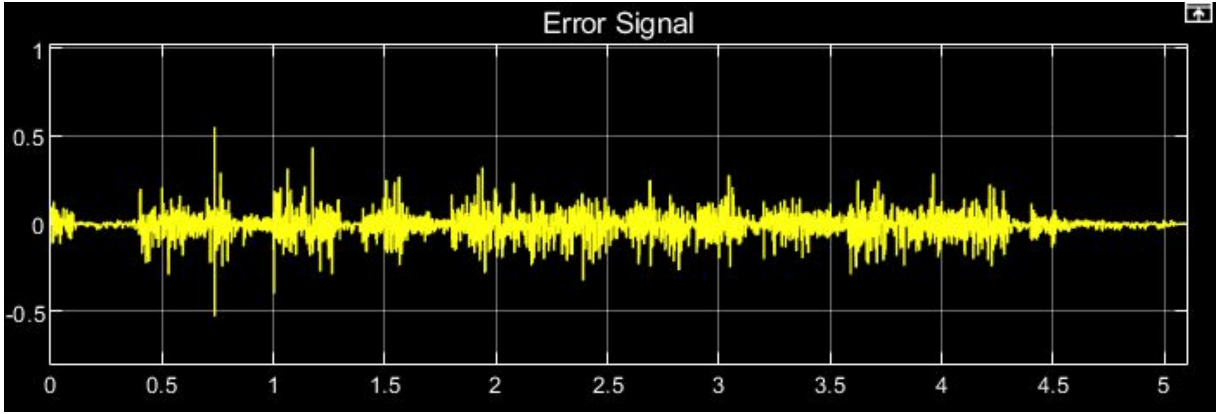


Figure : LMS Simulation Error signal

Fig. shows the weights of the lms filter , but as the program gets corrupted the filter can not adjust its weights .

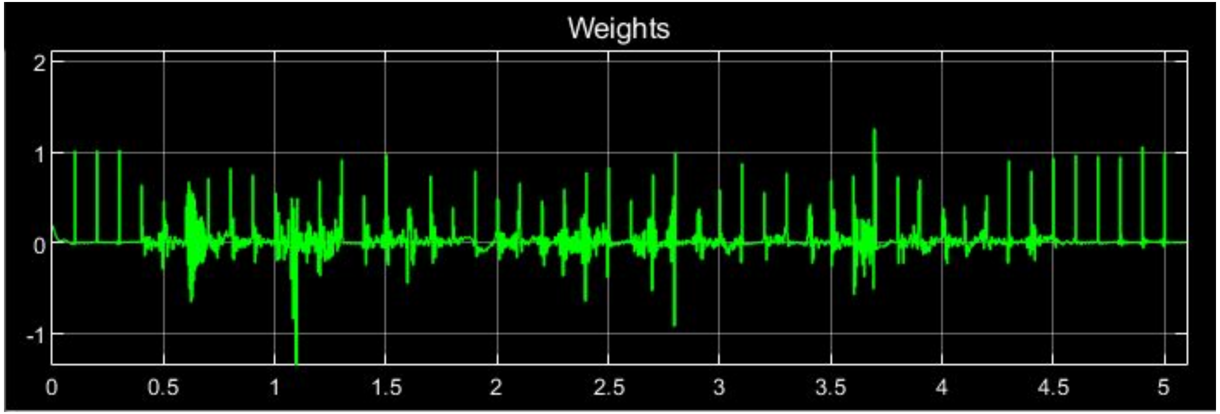


Figure : LMS Simulation Weights

Simulation using NLMS filter Approach

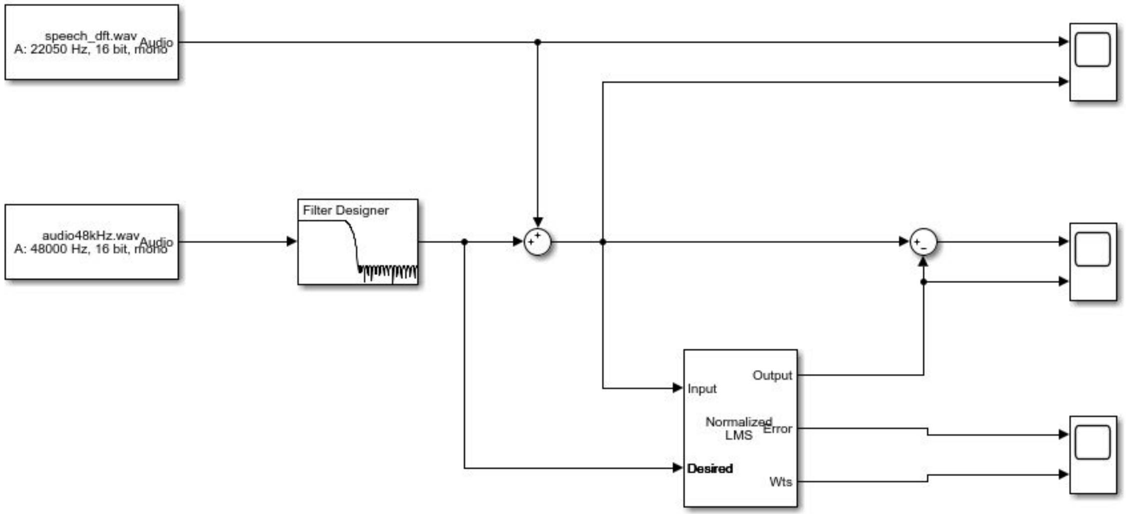


Figure : NLMS Simulation Block

Respective results using NLMS filter are

Fig. shows the original audio signal which is used for the project , the original audio signal is free from all kind of noise and is fully understandable.

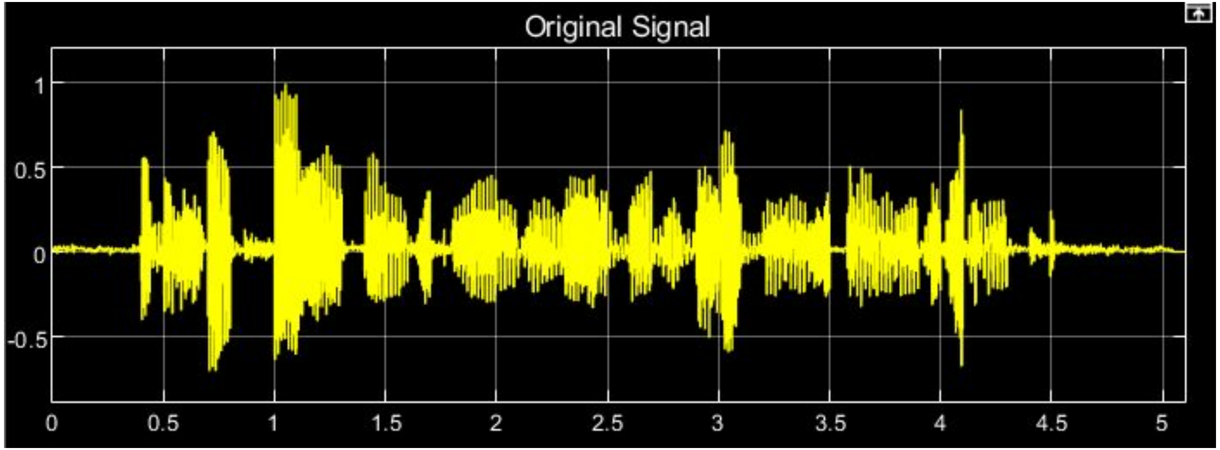


Figure : NLMS Simulation Original audio signal

Fig. shows the audio signal in which the noise was mixed and the audio signal was corrupted and was made unclear to understand and makes it difficult to understand.

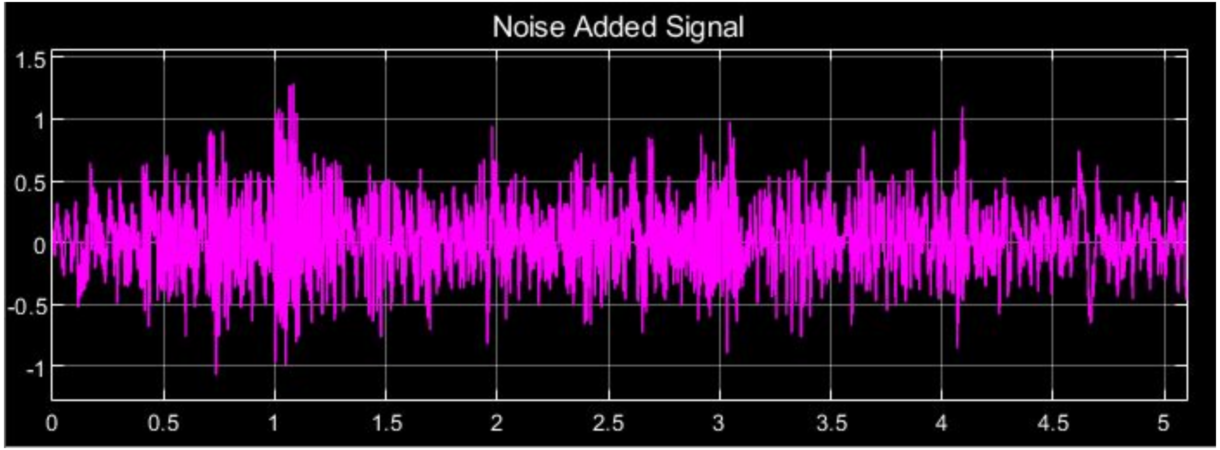


Figure : NLMS Simulation Noise Added Audio Signal

Fig. shows the recovered audio signal after passing from the NLMS filter and almost the same audio signal with almost same frequency was recovered

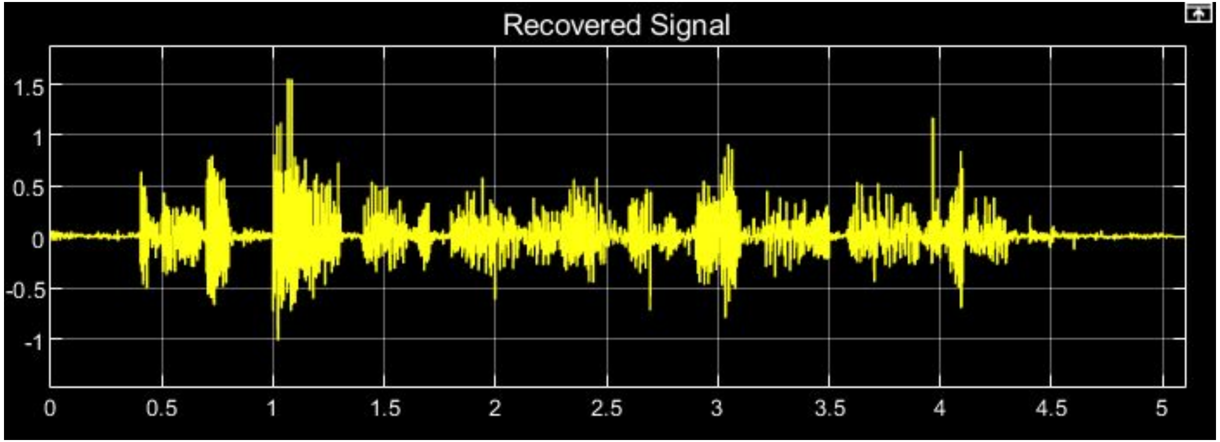


Figure : NLMS Simulation Recovered Audio Signal

Fig. shows the separated noise signal filtered from the nlms filter and almost same noise was separated.

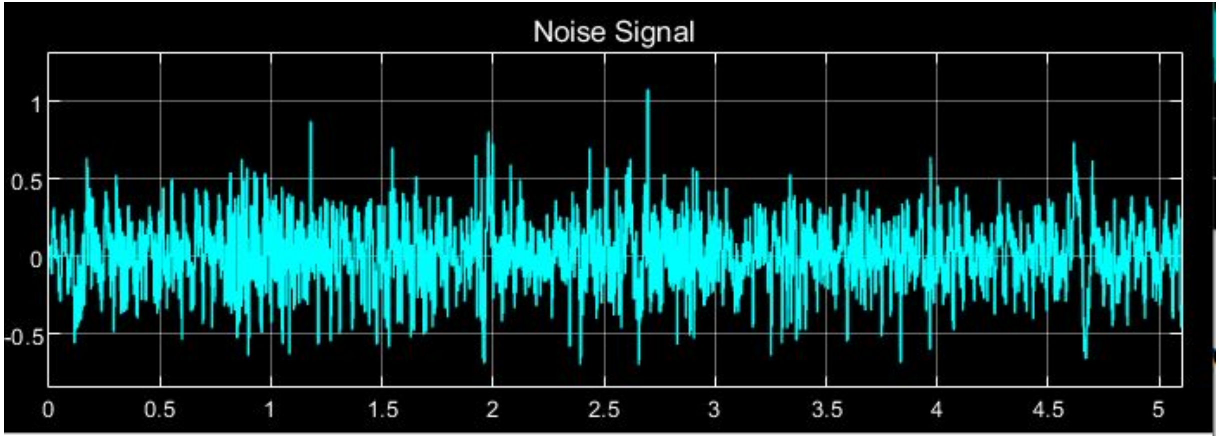


Figure : NLMS Simulation Noise Signal

Fig. shows the Error signal filtered from the nlms filter , and we can see the maximum error was seen at the point where the frequency of audio signal and noise where high , at that point the error was seen max.

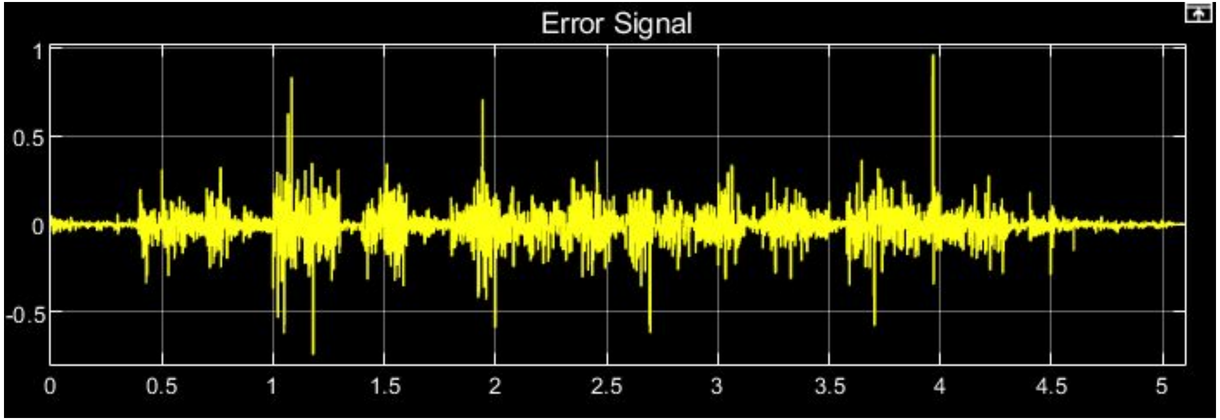


Figure : NLMS Simulation Error signal

Fig. shows the weights of the nlms filter , and we can observe that the weight was adjusted at each point and filtering of audio was done in smooth way.

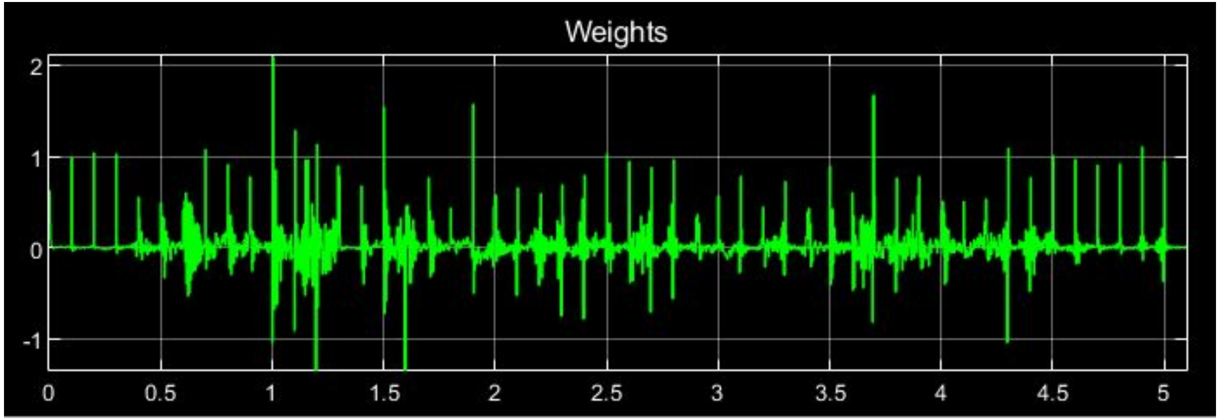


Figure 5.14 : NLMS Simulation Weights

Discussion of Result :

Sl. No.	Algorithm	Signal Power in dB	Estimated Signal Power in dB	Noise Power in dB	Estimated Noise Power in dB
01.	LMS	43.4312	41.8416	46.8142	48.1364
02.	NLMS	43.4312	46.6241	46.8142	41.6828

Table 1 : Minimum square error for different types of adaptive filter algorithm LMS and NLMS system.

Added Noise (Power Line Interface)	LMS (dB)	NLMS (dB)
Noise for input SNR	0.6515	0.6515
Noise for output SNR (After Filtering)	1.2173	0.9730

Table 2 : Improved SNR

Filter Length	LMS Filter (Time in sec)	NLMS Filter (Time in sec)
5	5.2641	6.0242
10	9.6132	10.7751
15	13.6806	15.4769
20	17.9328	23.7694

Table 3 : Comparison table for the computation time of LMS and NLMS where filter length is 5.

Conclusion :

- The fundamental algorithm of noise cancellation, Least Mean Square (LMS) algorithm and Normalised LMS algorithm is studied and enhanced with adaptive filter.

- The simulation of the noise cancellation using LMS adaptive filter algorithm and Normalised LMS algorithm is developed.

- The filtered signal is compared to the original noise-free speech signal in order to highlight the level of attenuation of the noise signal.

- On comparing the experimental results of both LMS and NLMS filters we concluded that both the processing time and SNR in case of LMS filter are better than that of the NLMS filter.

Reference:

- MathWorks
- Wikipedia
- Adaptive filter theory By Simon Haykin
- Digital Signal Processing Handbook by Douglas, S.C.
- Sushanta Mahanty and Alok Ranjan," Control And Estimation Of Biological Signals (Ecg) Using Adaptive System", International Journal of Electrical and Electronics Engineering (IJEEE) ISSN (PRINT): 2231 – 5284, Vol-2, Iss-1, 2012.
- M.E. Hawley and E.D. Smshauser, "Noise Reduction System" U.S. Patent 2 972 018, February 14, 1961.


Thank you