Performance Analysis of LMS Based Active Noise Cancellation Algorithms for Seminar Hall

Ms. Pallavi A. Pawar[1], Prof. Mrs. S. V. sagavkar[2]

[1]M. Tech. Student, Department of Electronics and Telecommunication Engineering, AMGOI, Vathar
[2]Professor, Department of Electronics and Telecommunication Engineering, AMGOI, Vathar

---

Abstract - *An active noise control (ANC) system based on adaptive filter theory was developed in the 1980s.It is important in a seminar or conference room it is needed to maintain silence. Active Noise Cancellation (ANC) technology is popular low frequency noise reduction. High frequency is reduced using passive technique.. But inside low frequency noises produced from fan, air conditioning system, cough, whispering etc. can be disturbing while a seminar or conference is going on. In this project we will make a seminar room silent using Active Noise Cancellation (ANC) method using different LMS based algorithm and analyses its performance.*

Key Words: MATLAB, Adaptive Filter, LMS algorithm.

Introduction :

Noise is considered as an unwanted signal which is the result of distorted or interfered signal being communicated. It is a form of random signal. Noise cancellation is reducing the effect of noise occurred in the final output of any system

Types of noises are **-** 1) Passive Noise **2**) Active Noise

In passive noise cancellation method high-quality absorbing materials used. Passive techniques such as enclosures, barriers, and silencers used to reduce high frequency noise. The concept of impedance change used in passive silencers. Reactive silencers are used as mufflers on internal combustion engines; these passive silencers are popular for their high attenuation over a broad frequency range. Passive method is ineffective for low frequency noise control and very expensive.

In active noise cancellation (ANC) superposition principle is used to reduce the noise. And we comparatively studied LMS, NLMS algorithm. Our thesis is also to study performance analysis of active noise cancellation using LMS based algorithms.

Literature review:

Different types of noises present in the environment these noises are removed by using active noise cancellation system using different LMS based algorithm.

In this paper noise reduction headset is designed for the audio and communication application. For more accurate noise cancellation adaptive feedback noise control technique is used. In this system single microphone is used per ear cup, it produce low power consumption, cheaper and integration is easy.in active noise cancellation super position principle is used, because of this primary noise is not present during operation because it cancel by secondary noise. In this system integrate the feedback ANC with receiving audio input. By using FxLMS algorithm adaptive filter coefficients are updated. The remaining noise picked by the error microphone it is used to synthesize the primary noise foe updating filter coefficient. The advantages of integrated feedback ANC system are good estimation if true residual noise without interfering with the audio signal, large step size can be used in adapting filter. The adaptive feedback ANC system is more accurate for noise cancellation since the microphone is placed inside the ear cup of headset. The audio signal can be used to drive both on line and offline modeling of the secondary path transfer function.[1]

In this paper, in yacht cabin high frequency noises can be reduced by high absorbing materials but in yacht low frequency noises such as engines, air conditioning and electrical generator.in yacht the noise signals are time variant due to change in cruising speed or sea condition. Frequency behavior in terms of amplitude and phase cannot be considered stationary. Each yacht is exclusive and specifically developed for the final user .each environment is developed differently with several architectural constrains. Analyze environment under bedroom and aim is quite the area closed to the bed pillow to make sleeping more comfortable. Due to long length of yacht feed forward control cannot use. The error microphone is placed in the head of bed. Subwoofers are used to generate anti-noise and microphones analyze the noise produced by yacht. In this system FxLMS algorithm is used for psychoacoustic correction. [2]

In this paper, there are different noises under seminar hall produced from fan, air conditioning, cough, shoe noise etc.. These noises create the disturbance while seminar is going on. To reduce such annoying noises by using ANC system with FxLMS algorithm. The conventional LMS algorithm has slow convergence rate, when the environment is changed quickely,it is not work very well.to avoid drawback of LMS algorithm FxLMS algorithms implemented. The several microphones are used to increases the performance of ANC system.[3]

In this paper to implement the adaptive algorithm like LMS&NLMS in frequency domain.in frequency domain adaptive filtering using adaptive algorithm. Frequency domain can be done by taking Fourier transform of input signal and independent weight coeffient.The adaptive filter suppress the noise without changing the input signal.in LMS algorithm changed its behavior due to changing the step size parameter. Less step size the convergence time is high and it leads the instability of the adaptive filter. In NLMS algorithm standard equation of step size is used using two constant. In time domain result of these algorithms is less SNR and more noise in output signal.so implement LMS and NLMS in frequency domain increase the signal to noise ratio around 8-9 times. It will give good quality of reconstructed signal as compared to time domain.[4].

In this system ECG is a recording of the electrical activities of the heart .the requirement of the ECG signals are accurate. Even small distortions in ECG waveforms impair the understanding of the patient heart condition. But due to some instrumentation faults some noises get added in ECG signal.to remove non-stationary signal like ECG various adaptive algorithms are used.in this system RLS and LMS algorithm are used to remove ECG noise. The filtered signals are compared by using different parameter like signal to noise ratio, minimum square error, percentage root mean square difference, plots of power spectral density. Using these performance criteria it has been observed that RLS algorithm has removed all type of noises more effectively but it is more complex.[5]

In this paper ANC system with online secondary path modeling uses modified FxLMS algorithm for adapting noise control and VSS LMS algorithm is used in the secondary path modeling filter. In FxLMS algorithm slow convergence speed introduce delay in secondary path.in modified FxLMS algorithm two extra filters are used one for the secondary path modeling .another for the control filter. The secondary path modeling filter generate error signal and control filter is used to avoid adaption using FxLMS algorithm. Control filter uses LMS algorithm and upper bound step size parameter is large than FxLMS algorithm. Larger step can be selected for faster convergence. step size is varied accordance with the power of error signal. In this paper improve the performance of online secondary path identification in single channel feed forward ANC system without using third adaptive filter. [6]

In this paper FxLMS algorithm is applied for narrowband noise. Variable step size is used to improve the convergence speed and reduce noise .In variable step size FxLMS algorithm narrowband noise is taken as sources and it is controlled by FxLMS algorithm. This system controls harmonic source by adaptively filtering a synthesized reference signal.varible step size function is used in this algorithm. The step size contains the parameter a and B .the a parameters control the shape and speed of step size of the algorithm. But B control the value range of the function. a and B are the function of error ratio. When the ratio is large the error change is large and VSFxLMS algorithm will be in the convergence stage. If the error ratio is small the step size is small and VSFxLMS algorithm will be in the steady state. The performance is analyzed by using the convergence rate parameter. Variable step size FxLMS with narrow band noise produce better noise reduction and convergence speed when compared with the fixed step FxLMS algorithm.in this technique the feedback is used from cancelling loudspeaker to reference microphone.[7]

The Adaptive Noise Canceller:

Adaptive noise canceller is used to remove the various noises and improve the signal quality. Noise cancelling depends upon difference of noise from a received signal which is controlled in an adaptive filter. The block diagram of an adaptive filter used in the adaptive noise canceller is given in Fig. 1. [5]

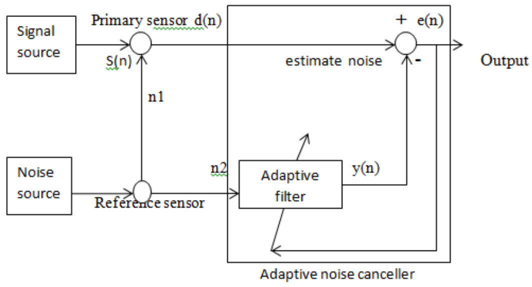

Fig 1.Adaptive Noise Canceller

The primary sensor receives the corrupted signal d (n), which contains the signal from signal source s(n) and additive noise from noise sources n1(n). The signals s(n) and n1(n) are uncorrelated. The primary input of the adaptive noise canceller is written as.[5]

$$d(n)=s(n)+n1(n) \quad \text{........ (1)}$$

The reference sensor receives a noise n2 (n) which is uncorrelated with signal s (n) but is correlated with noise signal produced from noise sources n1 (n). The reference signal is given by the adaptive filter to produce an output y(n), which is the estimate of noise. This y(n) is given

$$y(n)=\sum_{k=0}^{M-1} w_k v_2(n-k) \quad \text{......... (2)}$$

Here, wk is the adjustable filter coefficients. The difference between the filter output y(n) and primary signal d(n) . The error signal e(n) equation is ,

$$e(n) = d(n) - y(n) \quad \text{............ (3)}$$

By substituting Eq. (1) into Eq. (3) we get,

$$e(n) = s(n) + v_1(n) - y(n) \quad \text{......... (4)}$$

By using this error signal to update the filter coefficients of the adaptive filter. From Eq. (4) the noise component present in the system output is n1 (n) – y (n). The adaptive filter attempts to minimize the average power of the error signal e(n). [5]

Methodology:

Our project is to make a seminar hall silent using different adaptive control algorithms and to evaluate the performance of these algorithms. Active Noise Cancellation (ANC) is a method for reducing unwanted noise. ANC is achieved by introducing a canceling "anti-noise" signal through secondary sources. These secondary sources are interconnected through an electronic system using a specific signal processing algorithm for the particular cancellation scheme

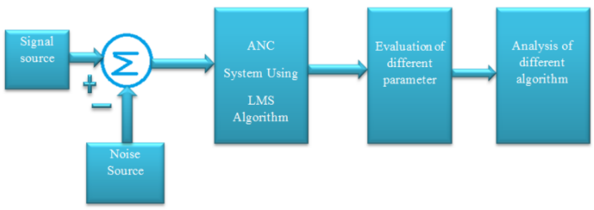

Fig 2. Block Diagram

Proposed Work: -

The proposed work will be Phase 1: We get the primary noise signal through the Reference microphone then it goes to the ANC. which analyses it. Evaluate performance parameter of LMS Based Algorithm. Phase 2: ANC creates an anti-noise through a cancelling loudspeaker. Phase 3: The remnant noise is then picked up by the error microphone and is transferred to the ANC as a kind of feedback .Our aim is to minimize the error signal. Phase 4: Analysis performance of different LMS based algorithm.

Least Mean Square (LMS) Algorithm :

LMS algorithm developed by Widrow and Hoff for the approach of noise cancellation. The least mean squares (LMS) method is used for noise cancellation adaptive filter. This method is used gradient descent in this algorithm which estimates a time varying signal. It identifies the minimum; the order to minimize the error is then computed by adjusting the filter coefficient. In order to find the divergence of a function used by the gradient of the del-operator, which in this case is the error with respect to the n-th coefficient. The error signal is that the difference between the desired d(n) and output y(n).

The elements of LMS equation are following

Weights evaluation:

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

Filtering output:

$y(n) = y(n) = w(n) * x(n-i)$ ...(2)

Error estimation:

$e(n) = d(n) - y(n)$..... (3)

Quality of the noise cancellation affected by the filter order. To eliminate noise faster using a high order filter.

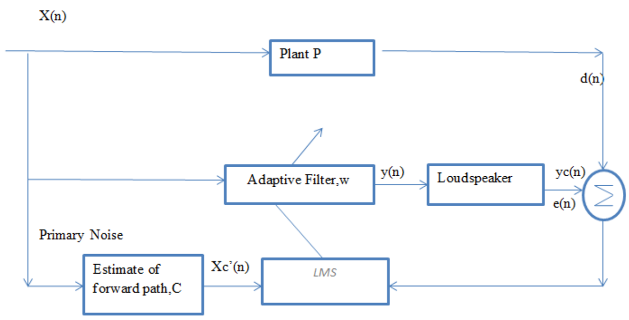

Fig 3. Block diagram of ANC system using the LMS algorithm.

Normalized Least Mean Square (NLMS) Algorithm:

A general form of the adaptive filter is shown in Fig. 2. Where d(n)is a primary input signal, y(n)is the actual output of a programmable digital filter driven by a reference input signal x(n), and the error e(n)is the difference between d(n) and y(n). The function of the adaptive algorithm is to adjust the digital filter coefficients to minimize the mean-square value of e(n).A technique to adjust the convergence speed is the Normalized LMS (NLMS)algorithm.

The NLMS is shown as follows:

w(n+ 1) =w(n) +μ(n)x(n)e(n)

μ(n) is adaptive step size which is calculated from the power and the step size of the filter.

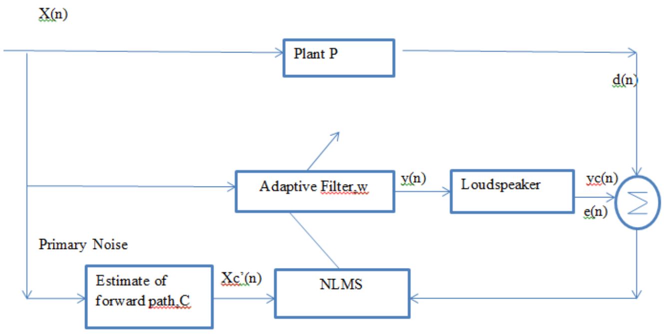

Fig 4. Block diagram of ANC system using NLMS algorithm

Filtered X Least Mean Square (FXLMS) Algorithm:

The most popular adaptation algorithm used for ANC applications is the FxLMS algorithm, which is a modified version of the LMS algorithm the schematic diagram for a single-channel feed forward ANC system using the FxLMS algorithm is in figure 4. Here, P(z) is primary path between the reference noise source and the error microphone and S (z) is the secondary path following the ANC adaptive filter W(z). The reference signal X (n) is filtered through S (z), and produce as anti-noise signal y'(n) at the error microphone. This anti-noise signal added with the primary noise signal d (n) to create a silence zone in the error microphone. The error microphone measures the residual noise e (n), which is used to adapt W (z) to minimize the noise at error microphone. the secondary path S (z) between the output of the controller and the output of the error microphone. The filtering of the given reference signals denoted to be x (n) A through the secondary-path model S (z) is demanded by the fact that the output y (n) of the adaptive controller W (z) is filtered by using the secondary path S (z).

The expression for the residual error e (n) is computed using equation (1)

e(n) = d(n) -y'(n) (1)

Where y' (n) is the controller output

y(n) filtered through the secondary path S (z). The y' (n) and y(n) are computed using equation (2) and (3)

y'(n) = s T (n)y(n) (2)

y(n) = w T (n) x(n) (3)

Where the weight vector is defined to be w(n) = [wo (n) w1 (n) .... wL_1 (n)f the reference signal . Is defined to be,x(n) = [x(n)x(n -1) .... x(n -L + 1)] this reference signal is picked by the reference microphone and s(n) is impulse response of secondary path S (z). It is assumed that there is no feedback from secondary loudspeaker to reference microphone.

FxLMS Update equation for the coefficients of w(z) is given in equation (4).

wn+1 = wn + ue{n)x'{n) (4)

Where x' (n) is reference signal of x(n) filtered through secondary path model S(z) is given in equation (5)

x'{n) = j/ (n)x{n} (5)

The FxLMS algorithm emerges to be very indulgent of errors made in the valuation of S (z).

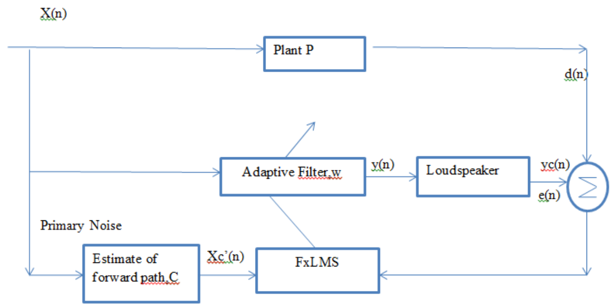

Fig 5. Block diagram of ANC system using the FxLMS algorithm.

Fast Block Least Mean Square (FBLMS):

For some application like adaptive noise cancellation needs adaptive filters with a large filter length. LMS algorithm applied to the adaptive filter take a long time to complete the filtering and coefficients updating process. This may cause difficulties in these applications because the adaptive filter is used to filter the input signals. In such problems, use the fast block LMS algorithm. The Fast block LMS algorithm used the fast Fourier transform (FFT) to transform the input signal x(n) to the frequency domain . Using frequency domain it also updates the filter coefficients. This updates can save computational resources. The fast block LMS algorithm differs from the standard LMS algorithm in the following ways: The size of block is equal as the filter length. The filter coefficients are updated by sample by sample basis in the standard LMS algorithm. The multiplication operation needed for the fast block LMS algorithm is less than the standard LMS algorithm. If both the filter length and block size are N, the standard LMS algorithm requires N(2N+1) multiplications, whereas the fast block LMS algorithm requires only (10Nlog2N+26N) multiplications.

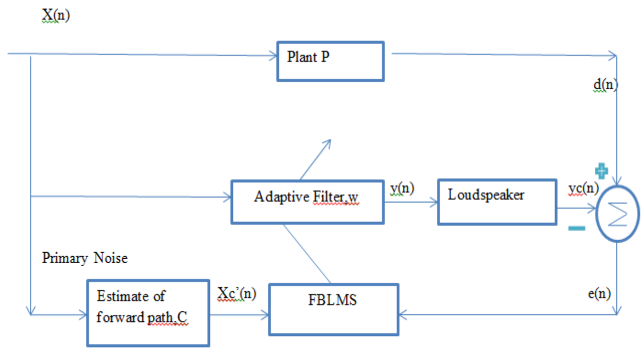

Fig 6. Block diagram of ANC system using the FBLMS algorithm

Summary of equation:

Equation	Least Mean Square Algorithm(LMS)
The output of the adaptive filter	$y(n) = \sum_{i=0}^{N-1} w(n)x(n-i) = \mathbf{w}^T(n)\mathbf{x}(n)$
Weights evaluation:	$w(n+1) = w(n) + \mu * e(n) * x(n-i)$
Filtering output:	$y(n) = w(n) * x(n-i)$
Error estimation:	$e(n) = d(n) - y(n)$

Equation	Normalized Least Mean Square Algorithm(NLMS)
The output of the adaptive filter	$y(n) = \sum_{i=0}^{N-1} w(n)x(n-i) = \mathbf{w}^T(n)\mathbf{x}(n)$
Weights evaluation:	$w(n+1) = w(n) + \mu(n)x(n)e(n)$
Filtering output:	$y(n) = w(n) * x(n)e(n)$
Error estimation:	$e(n) = d(n) - y(n)$

Equation	Filtered –X Least Mean Square Algorithm(FxLMS)
Weights evaluation:	$W(n+1) = w(n) + \mu\, x'(n)e(n)$
Filtering output:	$e(n) = d(n) + X_{Tr}(n)W(n)$
Error estimation:	$e(n) = d(n) - s(n) * [\mathbf{w}^T(n)\mathbf{x}(n)]$

Equation	Fast Block Least Mean Square Algorithm(FbLMS)
Weights evaluation:	$eF(n) = FFT\ 0$ $e(n)$ $wF(n+1) = wF(n) + 2\mu(n) x* F(n)\ eF(n)$
Filtering output:	$\mathbf{y}(n) = [y(nL)\ y(nL+1)\ \cdots\ y(nL+L-1)]^T$
Error estimation:	$\mathbf{e}(n = \mathbf{d}(n) - \mathbf{y}(n)$

Software Implementation

There various parameters that affect the performances of the voice signal like background noises such as ac, fan, whispering, sneezing etc. In this current thesis, we conducted two experiments in experiment we examined our approach by using an speech input signal X(n), with Additive stationary and non-stationary noise for different parameters. Noise is added with clean speech signal and it measured an error microphone. And noise is measured at reference microphone. The error signal is feedback signal and it is given to LMS based Algorithm.

According to noise LMS algorithm modify the signal parameter and it is given to the adaptive filter .Adaptive filter adjust the weight and produce the attenuated noise signal. This signal is given to loudspeaker. We use 44100Hz sampling frequency. The low path impulse response in the near physical environment is assumed to be of different order for different adaptive filters like LMS, NLMS ,FxLMS and FBLMS .This speech signal is sent into the room where it is convolved with the noise and external noise is added near the microphone and this is sent back to the adaptive filter algorithm. Where the noise cancellation takes place.

The various parameters which are compared in the experiments are given as follows:

1. Comparing the noise cancellation for different adaptive filters like LMS, NLMS, FxLMS and FBLMS for both stationary and non-stationary signals. When we examine the results we could see that LMS algorithm has better echo cancellation for both stationary and non-stationary signal.

2. Analyzing the results of the amplitude vs. frequency by varying different noises which are indicated with different plots for different adaptive algorithms .From the plots we could see that for stationary case the seminar room with dimension of 7x7 gives better SNR value. For non-stationary case SNR is not improved.

3. Observing SNR for different adaptive filters for both stationary and non-stationary signal in this we observe that the SNR value is maximum for LMS algorithm for all the noises excluding one noise. That is open door noise for

International Research Journal of Engineering and Technology (IRJET) e-ISSN: 2395-0056
Volume: 08 Issue: 09 | Sep 2021 www.irjet.net p-ISSN: 2395-0072

open door noise FBLMS algorithm gives better result as compared to other algorithm

4. Even calculating the SNR parameter for all algorithms we observe that SNR value becomes almost improve for LMS algorithm.

5. Analyzing the different parameters of the different adaptive algorithms with respect to SNR.

6. To calculate the Signal to Noise ratio for different adaptive algorithms for different noises and results are plotted.

7. Even plot amplitude vs. sample plot for different noises and different adaptive algorithms.

RESULT:

Sr no	Noises		Input Parameter			Algorithm Output SNR in DB			
			Sampling frequency	Step size	Input SNR (DB)	LMS	NLMS	FXLMS	FBLMS
1	Cough	man	44100	0.1	-32.6842	-8.1351	-1.2303	-21.8229	-27.8633
		Sick man	44100	0.1	-4.4719	20.4594	-0.0052	-0.9034	-0.4251
		Sick women	44100	0.1	-30.9333	-5.9173	-0.0281	-20.4495	-29.6527
2	Sneeze	man	44100	0.1	-29.0626	-6.5841	-0.0315	-21.0743	-27.1240
		adult	44100	0.1	-27.2611	-7.7628	-4.7152	-21.6984	-10.7465
		women	44100	0.1	-37.7675	-7.5236	-0.0093	-34.7026	-27.4169
		Young girl	44100	0.1	-31.6684	-6.3415	-3.0247e-05	-20.6930	-24.4248
3	Horn	car	44100	0.1	-33.7592	-11.9652	-0.0202	-25.2819	-30.8865
		bike	44100	0.1	-10.9719	18.0379	-3.7313e-05	-1.6201	-10.5785
		truck	44100	0.1	-13.4632	8.4180	-7.0677e-06	-0.3842	-0.2357
		bus	44100	0.1	-5.6612	16.2864	-0.0082	-0.2883	-0.7142
4	Opening Door	1	44100	0.1	-24.5326	-8.8576	-0.0173	-22.1829	-0.7039
		2	44100	0.1	-20.3351	-6.7232	-5.8526e-05	-19.1216	-0.0276
		3	44100	0.1	10.9528	20.0320	-5.2145e-05	-0.6833	-0.5183
		4	44100	0.1	9.7058	21.4206	-0.0184	-0.5869	-0.6336
5	Footstep		44100	0.1	-19.7812	-4.4844	-0.0118	-16.8506	-16.8537
6	Fan		44100	0.1	-1.4993	14.4878	-0.0082	-0.0971	-0.1722
7	Ac		44100	0.1	3.9921	18.7699	-0.0580	0	0
8	Closing Door		44100	0.1	-3.8324	19.0230	-7.6757e-08	-1.0417	-2.0552

Table 1 . Result for step size=0.1

Sr no	Noises		Input Parameter			Algorithm Output SNR in DB			
			Sampling frequency	Step size	Input SNR (DB)	LMS	NLMS	FXLMS	FBLMS
1	Cough	man	44100	0.2	-32.6842	13.6263	-1.9687	-21.8229	-27.3095
		Sick man	44100	0.2	-4.4719	23.3488	-8.7211e-07	-0.7899	-0.2652
		Sick women	44100	0.2	-30.9333	-2.9444	-2.2556e-05	-18.0123	-29.1317
2	Sneeze	man	44100	0.2	-29.0626	-3.5601	-7.6093	-18.3511	-26.9301
		adult	44100	0.2	-27.2611	-4.7793	-4.7152	-19.3399	-10.7441
		women	44100	0.2	-37.7675	-4.7142	-2.8377e-06	-34.7026	-34.7026
		Young girl	44100	0.2	-31.6684	-3.4012	-3.6438e-07	-18.2936	-23.7246
3	Horn	car	44100	0.2	-33.7592	14.7716	-6.5554	-25.2819	-30.5120
		bike	44100	0.2	-10.9719	20.9163	-1.0058e-06	-0.9531	-10.1435
		truck	44100	0.2	-13.4632	6.8423	-9.4624e-08	-0.2111	-0.1292
		bus	44100	0.2	-5.6612	13.8499	-2.2971e-06	-0.1539	-0.3877
4	Opening Door	1	44100	0.2	-24.5326	-6.0766	-7.6093	-22.1829	-0.7039
		2	44100	0.2	-20.3351	-4.4341	4.2230e-07	-17.1833	-0.0185
		3	44100	0.2	10.9528	22.4130	-5.2196e-07	-0.4547	-0.4780
		4	44100	0.2	9.7058	23.7578	-9.7069e-06	-0.3608	-0.5773
5	Footstep		44100	0.2	-19.7812	-1.8590	-4.3330	-16.8506	-15.7322
6	Fan		44100	0.2	-1.4993	14.6351	-3.4352	-0.0971	-0.1232
7	Ac		44100	0.2	3.9921	19.3442	-0.7930	0	0
8	Closing Door		44100	0.2	-3.8324	21.3546	-5.3433	-1.0417	-1.7833

Table 2 . Result for step size=0.2

Sr no	Noises		Input Parameter			Algorithm Output SNR in DB			
			Sampling frequency	Step size	Input SNR (DB)	LMS	NLMS	FXLMS	FBLMS
1	Cough	man	44100	0.3	-32.6842	-3.6467	-0.0251	-18.9716	-27.2768
		Sick man	44100	0.3	-4.4719	25.0129	4.1970e-08	-3.0114	-3.0828
		Sick women	44100	0.3	-30.9333	-1.2339	1.9462e-08	-20.1725	-28.9120
2	Sneeze	man	44100	0.3	-29.0626	-1.7960	5.4738e-08	-16.7429	-26.8384
		adult	44100	0.3	-27.2611	-3.0471	-4.7152	-18.0816	-10.7418
		women	44100	0.3	-37.7675	-3.1436	5.4738e-08	-34.7026	-34.7026
		Young girl	44100	0.3	-31.6684	-1.7251	-8.3187e-09	-17.3010	-23.3092
3	Horn	car	44100	0.3	-33.7592	14.7716	1.6352e-06	-21.9654	-30.9326
		bike	44100	0.3	-10.9719	22.5147	-3.7859e-08	-0.7031	-9.9227
		truck	44100	0.3	-13.4632	6.1040	-2.8315e-09	-0.2988	-0.0933
		bus	44100	0.3	-5.6612	12.4595	5.1272e-08	-0.1070	-0.2687
4	Opening Door	1	44100	0.3	-24.5326	-4.3615	-2.4859e-07	-18.8552	-0.7039
		2	44100	0.3	-20.3351	-3.0792	7.5393e-09	-15.9429	-0.0139
		3	44100	0.3	10.9528	23.7751	-7.9682e-09	-0.3466	-0.4512
		4	44100	0.3	9.7058	25.1906	4.1562e-08	-0.2641	-0.5427
5	Footstep		44100	0.3	-19.7812	-0.4278	-6.0274e-08	-13.8804	-15.1656
6	Fan		44100	0.3	-1.4993	10.3820	2.3625e-08	-0.0405	-0.1196
7	Ac		44100	0.3	3.9921	14.6221	-2.6636e-05	0	0
8	Closing Door		44100	0.3	-3.8324	22.7806	5.9163	-0.5339	-1.6863

Table 3 . Result for step size=0.3

Sr no	Noises		Input Parameter			Algorithm Output SNR in DB			
			Sampling frequency	Step size	Input SNR (DB)	LMS	NLMS	FXLMS	FBLMS
1	Cough	man	44100	0.4	-32.6842	-2.5559	-0.0021	-19.8247	-27.6432
		Sick man	44100	0.4	-4.4719	26.1630	3.5994e-10	-0.3198	-3.0828
		Sick women	44100	0.4	-30.9333	-0.0510	2.6878e-10	-19.1443	-28.8262
2	Sneeze	man	44100	0.4	-29.0626	-0.5519	4.7422e-10	-15.6079	-26.7944
		adult	44100	0.4	-27.2611	-1.8302	-4.7152	-18.0960	-10.7395
		women	44100	0.4	-37.7675	-2.0776	4.7422e-10	-34.7026	-34.7026
		Young girl	44100	0.4	-31.6684	-0.5694	-3.2568e-10	-19.6199	-22.9979
3	Horn	car	44100	0.4	-33.7592	10.6954	1.3782e-08	-21.9654	-31.9409
		bike	44100	0.4	-10.9719	23.5835	-9.3358e-10	-0.5781	-9.8086
		truck	44100	0.4	-13.4632	5.6565	-5.9615e-11	-65.4913	-0.0769
		bus	44100	0.4	-5.6612	11.4932	4.4438e-10	-0.0831	-0.2071
4	Opening Door	1	44100	0.4	-24.5326	-3.1487	-2.0659e-09	-18.3794	-6.7487
		2	44100	0.4	-20.3351	-2.0735	6.3734e-11	-15.0284	-0.0111
		3	44100	0.4	10.9528	24.7571	-6.7201e-11	-0.2818	-0.4308
		4	44100	0.4	9.7058	26.2693	3.7541e-10	-0.2100	-0.5179
5	Footstep		44100	0.4	-19.7812	0.5371	-4.8951e-10	-13.1847	-14.8047
6	Fan		44100	0.4	-1.4993	9.4366	2.1082e-10	-0.0338	-0.2737
7	Ac		44100	0.4	3.9921	13.4806	1.3928e-07	0	0
8	Closing Door		44100	0.4	-3.8324	23.8022	4.9674	-0.5231	-1.6528

Table 4 . Result for step size=0.4

Quantitative Quality Testing:

SR. No	AGE	Gender	LMS	NLMS	FXLMS	FBLMS
1	30-40	Female	5	2	4	3
		Male	4	2	3	3

1-VERY POOR 2-POOR 3-GOOD

4-VERY GOOD 5-EXCELLENT

Conclusion:

To conclude, this paper demonstrated the ANC system using LMS based algorithm and testing using different noises. LMS based algorithm shows that different step size for different noises for all the algorithm has different SNR dB level, which makes it easy to catch on which noise is the difficult to reduce its dB level can be used as a point of concentration.

- Results shows LMS based algorithm performance was at the best when using 0. 4 as its step size for all kind of different noises.
- NLMS algorithm steady-state performance was at the best when using 0.1 as it step size for all noises. NLMS algorithm has better SNR for sneeze, car horn, truck horn, footstep noises for 0.1 step size as compared to LMS algorithm for 0.1 step size. But NLMS algorithm gives poor SNR for AC noise.
- FxLMS algorithm steady-state performance was at the best when using 0.4 as it step size for all noises.FxLMS has poor performance for truck horn, open door noise and AC noises
- FbLMS algorithm steady-state performance was at the best when using 0.4 as it step size for all noises. But at 0.2 step sizes sick man cough noise gives better performance as compared to NLMS and FxLMS algorithm.
- But in real time LMS, NLMS, FxLMS, FbLMS algorithm gives better result for 0.1, 0.2, 0.2 0.3 step size respectively.

REFERENCES:

1) Vidya Mansur, R.B.Shettar, “Noise Cancellation Using LMS Algorithm: An Interval Arithmetic Approach With Intlab”, 2016 IEEE

2) Arya Chowdhury Mugdha, Ferdousi Sabera Rawnaque, Mosabber Uddin Ahmed-“A study of Recursive Least Squares (RLS) adaptive filter algorithm in noise removal from ECG signals”.

3) A. Parvathy and Gayathri Narayanan,” Comparative Study of Adaptive Algorithms Using Matlab and Verilog, c 2018 IEEE

4) Desai, Nirav A. "Active Noise Cancellation by the modified Filtered X-LMS algorithm with online secondary path modeling." Recent Trends in Electrical, Electronics and Communications Engineering Conference, ITM Universe. 2014

5) Plata-Chaves, Jorge, Alexander Bertrand, and Marc Moonen. "Incremental multiple errors filtered-x LMS for node-specific active noise control over wireless acoustic sensor networks." Sensor Array and Multichannel Signal Processing Workshop (SAM), 2016 IEEE. IEEE, 2016.

6) W. S. Gan and S. M. Kuo, “An integrated audio and active noise control headset,”IEEE Trans. Consum. Electron, vol. 48, no. 2, pp. 242–247, May 2002.

7) Peretti, Paolo, et al. "Adaptive feedback active noise control for yacht environments." IEEE transactions

on control systems technology 22.2 (2014): 737-744.

8) P. Lueg, “Process of silencing sound oscillations,” U.S. Patent 2 043 416, June 9, 1936.

9) A.Ganguly, S. H. K. Vemuri and I. Panahi, "Real-time remote cancellation of multi-tones in an extended acoustic cavity using directional ultrasonic loudspeaker," IECON 2014 - 40th Annual Conference of the IEEE Industrial Electronics Society, Dallas, TX, 2014,pp. 2445-2451

10) A S. Kuo and D. Morgan, "Active noise control: a tutorial review", Proceedings of the IEEE, vol. 87, no. 6, pp. 943-975, 1999.

11) Rachana Nagal, Pradeep Kumar, and Poonam Bansal “An Approach to Implement VSS-LMS Algorithm in Frequency Domain for Adaptive Noise Cancellation” ©2015 IEEE

12) Toshifumi Kosaka, Stephen J. Elliott and Christopher C. Bowher “A Novel Frequency Domain Filtered-X Lms Algorithm For Active Noise Reduction”

13) Rachana Nagal Pradeep Kumar Poonam Bansal, “An Approach to Implement LMS and NLMS Adaptive Noise Cancellation Algorithm in Frequency domain “