Comparison Study of Active Noise Cancellation

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Abstract— ***In this paper, the performance study of "Active Noise Cancellation (ANC) system" in the presence of AWGN has been performed using several algorithms such as Least Mean Square (LMS), Normalized Least Mean Square (NLMS) and Filtered x Least means square algorithms (FxLMS). For speech data, ANC system performance is verified by calculating MSE against distinct variable parameters like number of iterations, filter order (M), step size (mu), and Signal to Noise Ratio (SNR). It is noticed that to achieve minimum MSE, FxLMS algorithm outperforms in comparison with other algorithms. A comparison table is prepared based on simulation results for three algorithms using MATLAB tool.***

Keywords: ***ANC, LMS, NLMS, FxLMS, Adaptive filter.***

1. Introduction

In signal processing pull out the speech signal from the noise-interference is an critical process in communication systems. In which the problems are encountered with the contamination of the useful signals by undesired signals or unwanted signals. To reduce the unwanted signal or noise, using some processing techniques to filter out from the corrupted signal by unwanted signal, so that the original speech signal can be retrieved. But in many situations the characteristics of noise is varying with time and unknown. Sometimes the noise power may exceed the power of speech signal. In such situations ANC system using adaptive filters can give better performance in removing background noise as compared with conventional adaptive filters.

Basically the system [1]–[5] involves an electromechanical or electro audible system that eliminates the unwanted signal with the concept of superposition; In which produced signal has same amplitude and inverse phase is combined with the corrupted signal, thus gives eliminates the noise. The "ANC system" mostly used for contract the small-frequency noise applications. In that cancellation of noise come to be an adaptive process i.e. the system gets modified itself to the environmental changes. So that the ANC process removing background of undesired signal from the corrupted signal using an adaptive method, and enhanced SNR at the receiving end [4]. Using algorithms to amend the filter weights to minimizing the error signal in ANC [10]. The adaptive filter can be formulated by finite impulse response (FIR), infinite impulse response (IIR), and other transform domain filters. In that "FIR filters" are frequently preferred in the form of adaptive filter. In many applications, to enhance the system performance the different adaptive algorithms are introduced. "In this paper", the comparative study of active noise control is made using basic LMS and its variants, and to validate system performance, the parameters "Mean Square Error (MSE)" is calculated using number of iterations, varying SNR, step size and filter order.

1.1 Background of Noise

The voice communication system frequently affected by AWGN. A random signal is as "white noise", it have equal spectrum over a frequency range [13]. Theoretically it has infinite bandwidth but practically has limited bandwidth.

2. Methods of controlling Noise

There are two types of noise controlling techniques to minimize undesired noise.

A. Passive noise control B. Active noise control

Passive Noise Control

This is the traditional method, here the undesired sound waves are controlled by silencers, mufflers, enclosures and barriers. It is very cost, bulky in size and inefficient at low frequencies, and effective only for global control frequencies. To overcome the drawbacks in Passive noise control, the researchers were introduced "Active Noise Control (ANC)".

Active Noise Control [1-5]

It is an electro audible or electromechanical system works on the concept of destructive interference [6]. ANC technique is a good area of research because; with the help of adaptive algorithms it can reduce noise at low frequency. ANC can be modelled by producing a secondary noise signal with same amplitude and opposite phase of primary noise signal, by adding both the noise signals are cancels to each other The secondary noise signal (anti noise signal) can be produced at the loudspeakers, and the noise signal can be produced by noise source or sensed by known frequency, and is input to an adaptive filter, which produces a required anti-noise signal. In the acoustic environment, the both signal are add, and produce a residual noise. It is slowly minimized by adjusting filter weights using adaptive algorithms.

The ANC diagram shown in Fig. 1. In that, the sensor signal $S(n)$ of sensor disturbed by noise signal $N_o(n)$, results a corrupted noise signal $d(n)$. From the sensor $N_i(n)$, passes noise signal, it is not correlated with speech signal, but correlated with noise signal $N_o(n)$, Here the adaptive filter is excited by noise signal, and produces a signal $y(n)$, which is subtracted from the noise corrupted signal and generates an error signal $e(n)$, and minimized by adjusting the "adaptive filter weights" with adaptive algorithm.

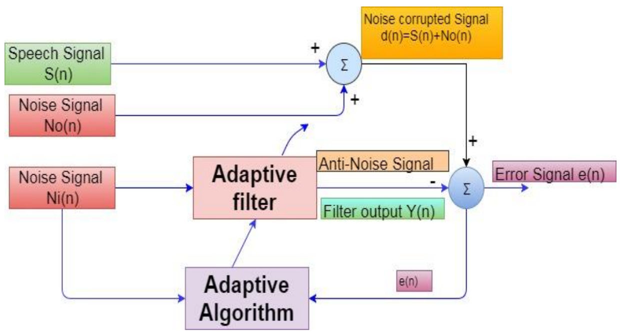


Figure 1 ANC Block diagram

3. Adaptive Algorithms

3.1 Least Mean Square (LMS) Algorithm

In 1960 widrow and Hoff was first introduced LMS algorithm [6]. In this algorithm using gradient descent stochastic method the filter is adapted. The" LMS algorithm" [7-8] followed by iterative procedure in which each iteration, In that the filter weights are updated by following expression:

$$w(n+1) = w(n) + 2\mu e(n)x(n)$$

Where, *x(n)* is input, *w(n) is* the adaptive weights at time *n* and μ is the step size.

Based on the step-size value the "LMS algorithm" is performed, for small values of step-size, the convergence of adaptive system will takes long time, and large values of step-size, it becomes unstable and outputs are divergence.

LMS algorithm

Input signal $x(n)$
Desired signal: $\mathrm{d(n)}$
Filter order: M
Step-size: μ
Steps:
Step 1: Adaptive response
$\mathrm{y(n) = x(n)w^T(n)}$
Step 2: Residual signal
$\mathrm{e(n) = d(n) - y(n)}$
Step 3: Updated weights
$\mathrm{w(n+1) = w(n) + \mu e(n)x(n)}$
Repeat step 1 to step 3, for minimum residual signal.

3.2 Normalized LMS Algorithm

The "LMS algorithm" is sensitive to the scaling of with inputs; due to the system stability becomes poor. To predict this drawback "Normalized LMS algorithm" was proposed [6-7][9]. Using this algorithm, to normalize the input power to resolves the problem of variation of LMS [11].

NLMS algorithm:

Input signal: $\mathrm{x(n)}$
Desired signal: $\mathrm{d(n)}$
Filter order: M
Step-size: μ
Steps:
Step 1: Adaptive output
$\mathrm{y(n) = x(n)s^T(n)}$
Step 2: Residual signal
$\mathrm{e(n) = d(n) - y(n)}$
Step 3: Normalization of step-size
$\mu(\mathrm{n}) = \frac{\mu}{\mathrm{X^T(n)X(n)}+\psi}$
Step 4: Update the adaptive weights
$\mathrm{s(n+1) = s(n) + \mu(n)e(n)x(n)}$
Repeat step 1 to step 4, to get minimum error signal.
In this algorithm, the optimum convergence rate μ is lies between 0<μ<2, and $\boldsymbol{\psi}$ is static term for normalization, which choose always less than one.

3.3 Filtered x LMS algorithm (FxLMS)

In real situations, the employing of "LMS algorithm" faces inconvient by the anti-noise generated by the adaptive filter, when the secondary signal moves from the output of adaptive filter to the error microphone, which introduces a non-negligible phase delay and frequency distortions at this transition of secondary path S(z). Due to these problems the "LMS algorithm" performance has been degraded. So that it gives low convergence speed due to increased residual noise power, then it comes to be unstable. To resolve this problem with help of "FxLMS" [2] using stochastic methods [5].The "FxLMS algorithm" diagram "shown in Fig" 2. In that using offline and online [14] modelling methods [12] to determine coefficients of the "secondary path" S(z) and estimated of "secondary path"S^(z).

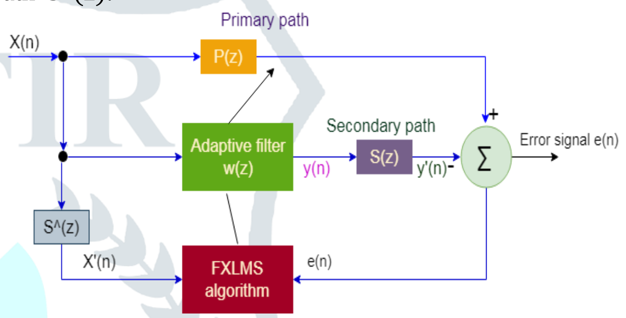


Figure 2 FXLMS algorithm diagram

FXLMS algorithm

Input signal :$x(n)$
Desired signal : $\mathrm{d(n)}$
filter order: M
Step-size: μ
Steps:
Step 1: Adaptive response
$\mathrm{y(n) = x(n)h^T(n)}$
secondary path response
$\mathrm{y'(n) = s(n) * y(n)}$
where $\mathrm{s(n)}$ is the impulse response of "secondary path"S(z)
Step 2: Error signal
$\mathrm{e(n) = d(n) - y'(n)}$
The filtered reference signal
$\mathrm{x'(n) = s^{\wedge}(n) * x(n)}$
Where $\mathrm{s^{\wedge}(n)}$ is the impulse response of estimated "secondary path" S^(z)
Step 3: Update the filter weights:
$\mathrm{h(n+1) = h(n) + 2\mu e(n)x'(n)}$
Repeat the steps from 1 to 3 to get minimum error signal.

4. Simulation Results and Analysis

In this, the "ANC system" using "adaptive algorithms" performance will be reported. Using AWGN, the contrast performance of "ANC system" with LMS, NLMS and FxLMS algorithms evaluated by effect of number iterations. Besides this, the parameters such as variation of the filter order, step-size and SNR has been studied. In all effects "the ANC system" can be computed in terms of "Mean Square Error (MSE)". The above all investigations are done by using computer simulations implemented by MATLAB. Using a "recorded speech signal" to performed ANC system with algorithms. The "recorded speech signal" "shown in Fig" 3.

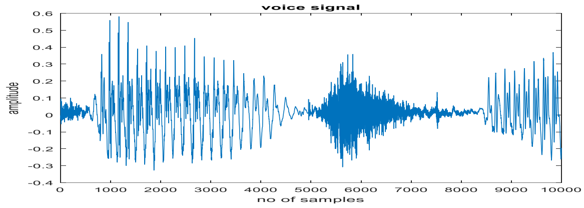


Figure 3: The recorded Speech Signal

4.1 Noise Cancellation with "LMS Algorithm"

In this simulation the original speech signal, "white noise signal" and combined to produce a desired signal "shown in Fig" 4, and the "Mean square error" signal "shown in Fig" 5(a). The "original speech signal" and the adaptive output signal (recovered speech signal) "shown in fig" 5(b). From this figure investigate very slight difference having between the adaptive signal and "original speech signal" during the convergence period.

The following specifications are used in simulations results: M=32, μ= 0.005 and, iterations= 1000.

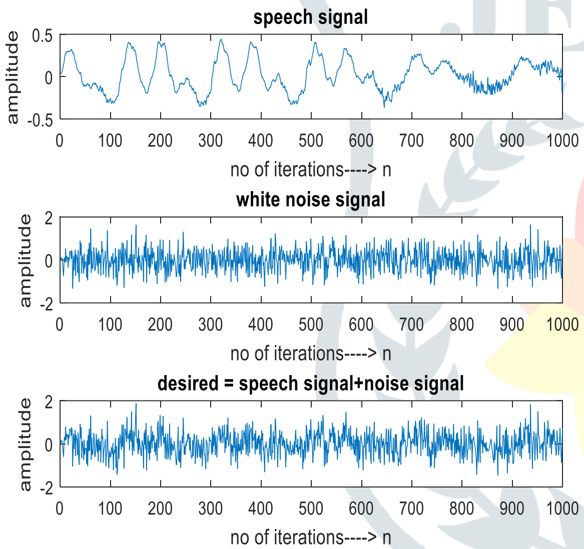


Figure 4: (a) Speech signal, (b) white noise signal, (c) Desired signal

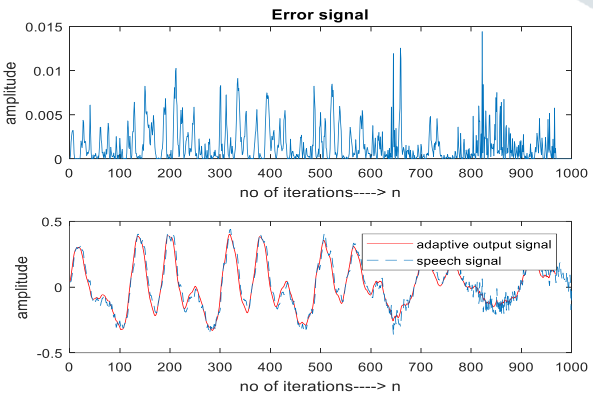


Figure 5: (a) MSE signal, (b) Comparison of adaptive output signal and speech signal

4.2 Noise Cancellation with NLMS Algorithm

In this simulation results the speech signal, "white noise signal" and combined a desired signal shown in Fig 6, and the MSE signal "shown in Fig" 7(a). The comparison between the original signal and the adaptive output signal (recovered speech signal) "shown in Fig" 7(b). From this figure investigates very slight deviation having between the recovered signal and original signal during the convergence period.

"The following specifications are used in simulations results": M=32, μ= 0.005 and, iterations= 1000.

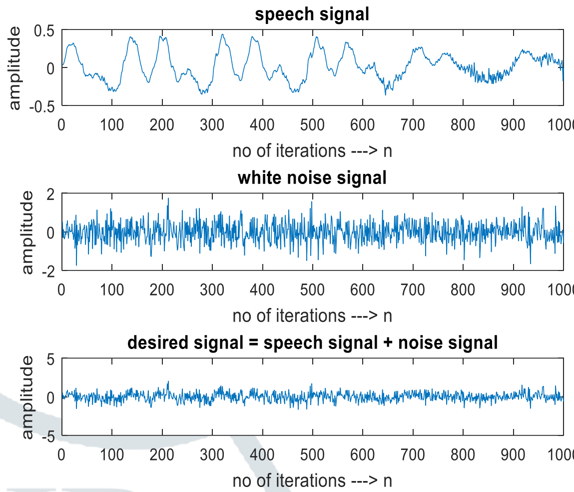


Figure 6: (a) Speech signal, (b) white noise signal, (c) Desired signal

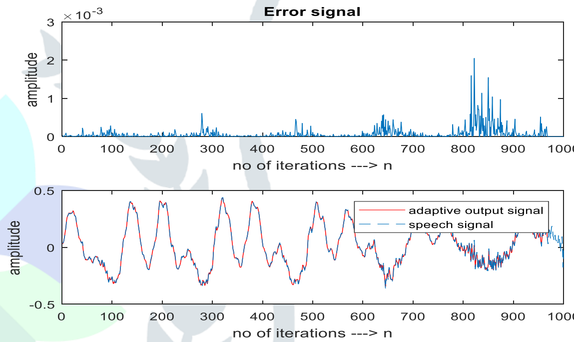


Figure 7: (a) MSE signal, (b) Comparison of recovered signal and speech signal

4.3 Noise Cancellation with FxLMS Algorithm

In simulation the "secondary path" and "estimated secondary path" coefficients shown in Fig 8. The speech signal, "white noise signal" and combined desired signal shown in Fig 9, and the MSE signal shown in Fig 10(a), and the comparison between the propagated signal and the adaptive output signal (control signal) "shown in fig" 10(b). From this figure investigates a small difference having between the adaptive signal and propagated signal during the convergence period.

"The following specifications are used in simulations results": M=32, μ= 0.0001 and, iterations = 1000

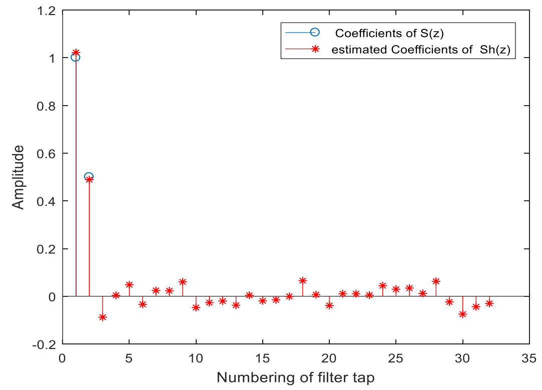


Figure 8: Secondary path and estimated Secondary path coefficients

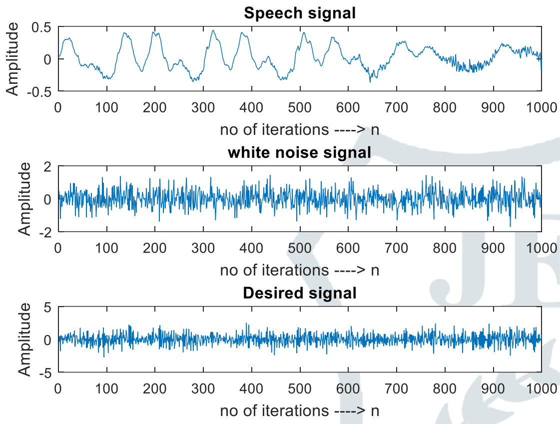
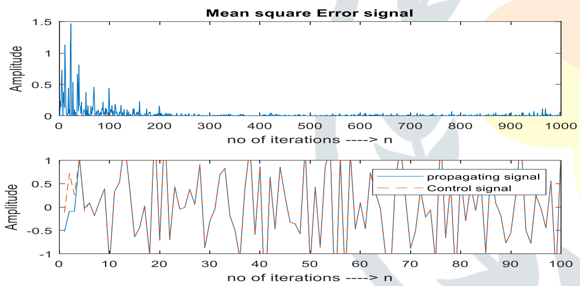


Figure 9: (a) Speech signal, (b) white noise signal, (c) desired signal



10: (a) MSE signal, (b) Comparison propagated signal and Control signal

4.4 Effect of Step-size (μ) on MSE

The "ANC system" with the "effect of Step-size on MSE" "shown in fig" 11. It shows that how the (MSE) changes, with different Step-sizes (μ). To achieve minimum MSE, compared to" LMS", "NLMS" and FxLMS with varying of Step-size. From the comparison, concludes the "FxLMS algorithm" has given minimum MSE. Compare to LMS achieved minimum MSE in NLMS. From the analysis simulation results "FxLMS algorithm" has given good results compare to "NLMS" and "LMS algorithms".

For simulation "the following specifications are used": M=32, μ= 0.005:0.005:1 and, iterations = 1000.

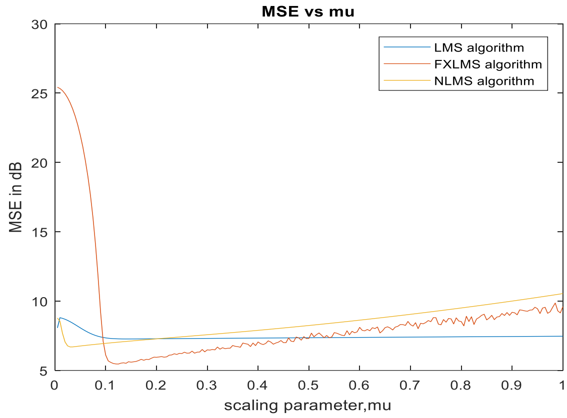


Figure 11: Performance of Step-size (μ) on MSE

Table-I

ANC with different step-size values

S no	Step-size (mu)	MSE in dB		
		LMS	NLMS	FxLMS
1	0.01	9.2dB	8.6	15
2	0.05	8.5	7.5	7
3	0.1	7.4	6.95	5
4	0.15	7.3	7.2	5.2
5	0.2	7.32	7.3	5.4
6	0.25	7.4	7.37	6
7	0.3	7.42	7.41	6.3
8	0.35	7.45	7.62	6.5
9	0.4	7.7	7.9	7
10	0.45	7.95	8	7.4
11	0.5	8.7	8.5	7.92

From the Table-I, we observe the "FxLMS algorithm" has Low MSE and good stability compared to other algorithms with various values Step size (mu).

4.5 Effect of SNR on MSE

The "ANC system" with the "effect of SNR on MSE" "shown in fig" 12. It shows how the MSE changes, with various values of SNR. To achieve minimum MSE, compared LMS, NLMS and FxLMS with various values of SNR. From the comparison, concludes the "FxLMS algorithm" has given minimum MSE. And NLMS has obtained minimum MSE compare to "LMS algorithm". From the analysis of simulation results "FxLMS algorithm" has given good results compare to "NLMS" and "LMS" algorithms.

For simulation "the following specifications are used": M=32, mu=0.005, iterations=1000 and SNR :-100dB to 100dB.

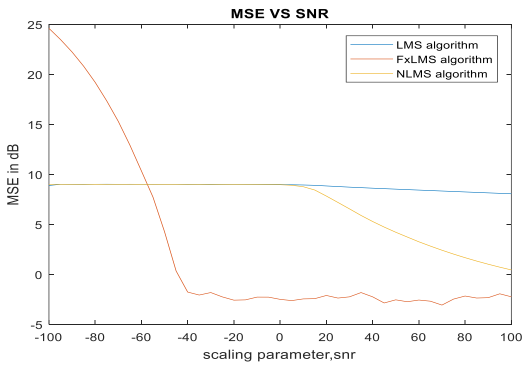


Figure 12: Performance of SNR on MSE

Table-II

ANC with different SNR values

S no	SNR	MSE in dB		
		LMS	NLMS	FxLMS

1	-50	9.8	9.6	1.2
2	-40	9.2	9	-0.56
3	-30	8.9	9	-0.8
4	-20	8.85	8.65	-1.1
5	-10	8.82	8.4	-1.4
6	0	8.8	8.2	-1.6
7	-10	8.75	8	-2.0
8	-20	8.7	7.5	-2.6
9	-30	8.65	6.5	-3.1
10	-40	8.6	5.2	-3.24
11	-50	8.2	4.1	-3.4

From the Table-II, we observe the “FxLMS algorithm” has least MSE and good stability compared to other algorithms with various values SNR.

4.6 Effect of Filter order on MSE

The “ANC system” performance using filter order (M) “shown in fig” 13. It shows how the MSE changes, various values of “filter order” (M). From the comparison analysis of “LMS and NLMS algorithms, concludes the NLMS algorithm has given minimum MSE and good performance.

“The following specifications are used” in simulation: filter order (M)=0 to 60, Step-size μ= 0.08 and iterations = 1000.

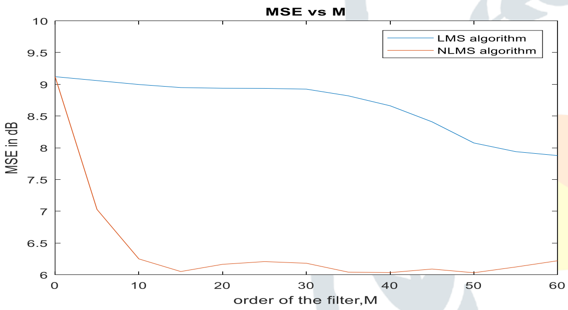


Figure 13: Performance of filter order on MSE

5. Conclusions and Discussions

In this paper, we performed and compared the “ANC system” has been tested with three algorithms are reported in Table [1-2], and to test MSE considering with different parameters like Step-size, SNR, filter order and No of iterations. From the simulation analysis, we perceived that the “FxLMS algorithm” attain least MSE and good ANC system performance compare to “LMS” and “NLMS” algorithms.

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