PERFORMANCE EVALUATION COMPARISON OF NLMS AND RLS ALGORITHM for AEC in SPEECH PROCESSING

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Abstract: acoustic echo cancellation is one of the main issues in telecommunication which is truly a late version of sound reflected back to the source of sound obstructing communication. An acoustic echo canceller another problem like deliberate convergence rate (mainly speech signal) & computational complexity in echo that require filter with more than thousands taps, unbounded speech signal, slowly time varying system. The cancelation use the acoustic echo canceller involving adaptive filter governed by adaptive technique involving adaptive filters governed by adaptive technique. This technique received signal quickly disturbed by noise so both signal and noise changes constantly, and then raises the need of adaptive filter. In this paper, should be used two algorithms to eliminate the noise on speech signal and data signal like normalized least mean square algorithm (NLMS) and recursive least square algorithm (RLS) with implemented in MATLAB. In adaptive filter technique the evaluation of the algorithm are established on for getting factor and tap weights. The main purpose of this algorithm to implement and provided real performance with less computational complexity.

Keywords: Adaptive Filtering, adaptive algorithm, Normalized Least Mean Square (NLMS), Recursive Least Squares (RLS), convergence rate, MSE and ERLE and SNR.

(1)Introduction:

Teleconferencing systems are likely to look forward to supply a great sound quality. Speech by the far end speaker is received by the close to finish microphone and being sent back to him as echo. In recent time adaptive filtering system commonly used in communication, signals control and many other applications. The adaptive filter technique adapting the filter parameter varies with the applying object, among these adaptive filtering technique normalized least mean square algorithm and recursive mean square algorithm become the most standard adaptive flittering algorithm as a consequence of their simplicity and robustness [3,4].adaptive filtering arrangement become the popular because of their simplicity and robustness. The elementary object of an adaptive filter is to adapt its parameter according to sure criteria to attenuate a minimize selected objective perform like MSE, noise variance etc. and maximize a selected objective function like SNR, quantitative relation, gain, likelihood output power etc. [2].

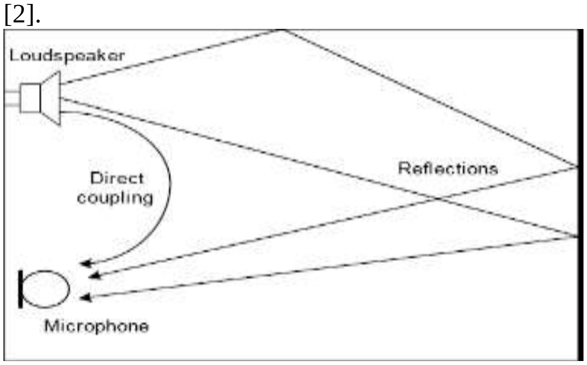


Fig 1Generation of acoustic echo across direct coupling and reverberations

The signal from the far point speaker to the near point speaker is sent through the loudspeaker and other surrounding and then received by the microphone at near point. The signal/speech transmitted back to the speaker at far end by providing echo sound which can be explained with figure 1

Excess mean squared error is the main disadvantage in NLMS algorithm [6]. There is a miss adjustment in such algorithm which increases linearly with the desired signal power. According to the performance basis, it is well known that RLS algorithm offers better convergence rate and in the existence of any sound, it has good error performance []

2) The methodology of acoustic echo cancellation:

Speech by near end speaker in telecommunication scenario often capture by far end microphone and response being send back to him as echo. The aim is to cancel the echo error signal using acoustic echo cancelation. The starting speech transmitted to the far point is adaptively filter to track down the echoed speech be transmitted from the far point the dissimilarity of the two signal is transmitted to the near end. From figure x(k) be the input signal transferable towards the close point speaker along the loud speaker. End d(k) is the signal receive by microphone. Here the adaptive filter is employ to modal the transfer function of the surrounding area, where the loud speaker and microphones are used to create a copy of the echoed signal y(k),the estimated echo is eliminated from the given input signal.

$$E(k) = d(k) - y(k) \tag{1}$$

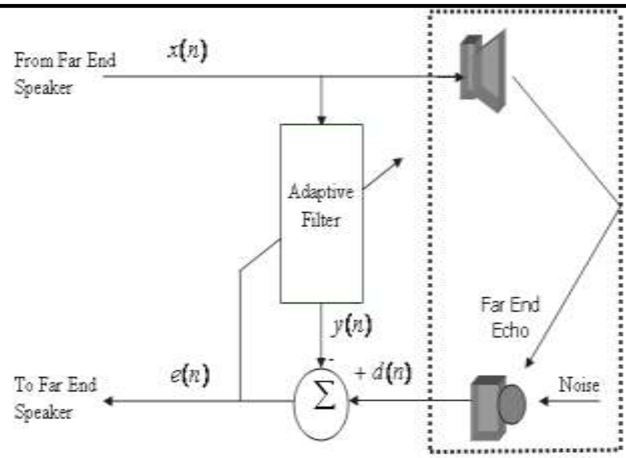


Figure 2 Acoustic echo cancelation methodologies

a) Discrete time signal:

Discrete time signals in communication are entitled by electronically discrete numeric sequences. In these sequences instantaneously value of the continuous signal is represented by each value in the sequence these values are noted at regular period which is called as sampling period.

$$\mathbf{X(k) = x(kT_s)} \qquad \textbf{(2)}$$

b) Speech signal:

There classes of sound makes speech signal that are voiced fricative and plosive sounds. Excitation of the vocal tract with quasi periodic pulses of air flow makes voice sound. Fricative sounds are formed by constructing the vocal tract and passing air through it causing noise like sound. Plosive noises are treated by shouting of the vocal tract are back side and then frequently realize it.

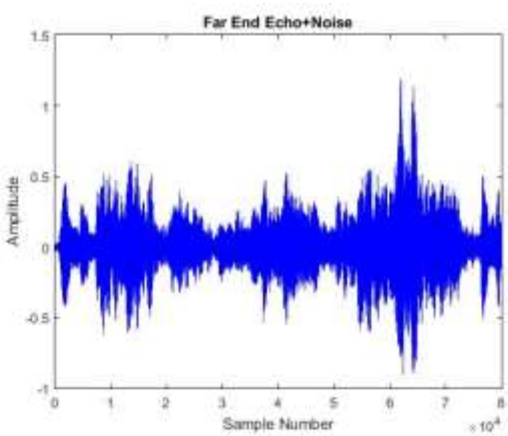


Figure 3 Speech signal representation

3) Adaptive filter algorithm:

In order to adapt the changing input signal environment, adaptive filter [12] has various filters whose parameters changed with time, and output signal characteristics [6]. The basic block diagram of the adaptive filter is shown in Fig 1. Where x(k) is the input signal of the variable filter, y(k) is the filtered output signal, d(k) is the desired response and e(k) is the error signal that represent the difference between d(k) and y(k).The adaptive algorithm regulates the coefficients of the filter to have the error signal at a low level (in the mean square sense). Some examples of adaptive filter are LMS, NLMS, RLS, AAF, APA, FAPA etc. Echo cancellations are implemented by using NLMS and RLS algorithms.

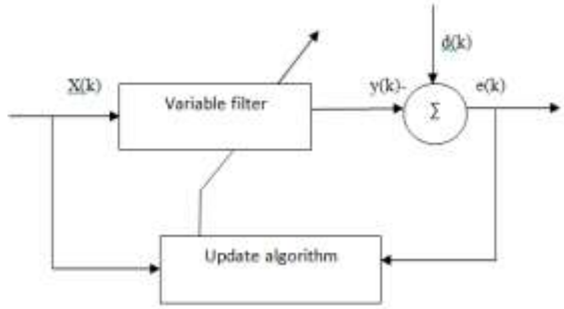


Figure 4 Adaptive filter

A. Normalized mean square algorithm:

The issue of maximum value of step size can be solved by using NLMS system. The drive of this technique between two side by side filter coefficient changes. Similarly the convergence rate has also influence [4]. In adaptive noise cancellation, the noise free signal is obtained by estimating the noise and interference so adaptively filtering them out from the received signal. During this technique, in adaptive noise canceller recovers the specified signal corrupted by noise using NLMS (Normalized Least Mean Square) algorithmic rule. In this iteration the algorithm consist steps as follows-

a) the result of adaptive filter is shown below -

$$\boldsymbol{y(k) = \sum_{i=0}^{K-1} w(k)x(k-i) = w(k)^T x(k)} \qquad \textbf{(3)}$$

b) A signal which is an error is given below –

$$\boldsymbol{e(k) = d(k) - y(k)} \qquad \textbf{(4)}$$

c) The step size value for input vector is shows that-

$$\mu_k = \frac{\mathbf{1}}{\boldsymbol{x(k)^T \times x(k)}} \qquad \textbf{(5)}$$

Where, μ shows and adjust the convergence speed of the design parameter and the convergence rate of the algorithm is controlled & also the situation 0<μ<2 is fulfilled & the smallest positive number α is used to explain the problem that may occur when x(k) is very small.

d) The equation of the weight vector update is shown as bellow -

$$\boldsymbol{w(k+1)=w(k)+\mu\, e(k)x(k)} \qquad \textbf{(6)}$$

B. Algorithm of recursive least square:

Recursive least mean square process is very prominent adaptive filter technique. The problem of large value of step size of step size in NLMS can be solved using RLS algorithm. Where as in RLS algorithm the step size dependent on the input vector hence it is unstable the main characteristic of RLS is its high rate of convergence. It has better convergence rate for highly interrelated input signal when compared with NLMS [5]. The only problem in such algorithm is its calculation complications which is much complex as compared to NLMS. In this method the condition and information is early defined which helps to renew the previous calculated information of new data. The implementation of RLS algorithm contains the resulting steps:

a) The adaptive filter output y(n) is as given

$$\boldsymbol{y(k)} = \sum_{i=0}^{k-1} \boldsymbol{w(k)x(k-i)} = \boldsymbol{w(k)^T\, x(k)} \qquad (7)$$

b) The error signal is as given

$$e(k)= d(k) - y(k) \qquad (8)$$

c) The recursive weights are as given

$$r(k)=d(k)*x(k) \qquad (9)$$

d) The gain vector is as given

$$\boldsymbol{v(k)} = \frac{\boldsymbol{1}}{\boldsymbol{\lambda + d(k) \times r(k)}} \qquad (\mathbf{10})$$

e) The update weight factor equation is given as

$$w(k+1)=w(k)+\mu e(k)x(k) \qquad (11)$$

4) Performance parameters in adaptive filter

MSE, SNR and ERLE area unit are the performance parameter that calculated in beyond algorithm and given as below-

(a) MSE:

The mean squared error of a procedure for an unobserved quantity measure the average square dissimilarity between the predictable values and real values.

$$\boldsymbol{MSE} = \frac{\boldsymbol{1}}{\boldsymbol{K}} \sum_{\boldsymbol{i=0}}^{\boldsymbol{K}} \boldsymbol{e(k)^2} \qquad (11)$$

(b) ERLE:

The ERLE stands for echo return loss enhancement which helps to analysis the performance of echo canceller. It is the quantity of extra signal loss introduced by the echo canceller.

$$ERLE=10\log_{10}\frac{d(k)^2}{e(k)^2}DB \qquad (12)$$

(c) SNR:

To determine the strength of a signal to noise ratio is necessary to calculate. To extract the useful information from the row signal would be easier if the SNR ratio is higher. It is the ratio between the signal and noise in decibel (dB) unit.

$$SNR=10\log_{10}\frac{x(k)^2}{d(k)^2}DB \qquad (13)$$

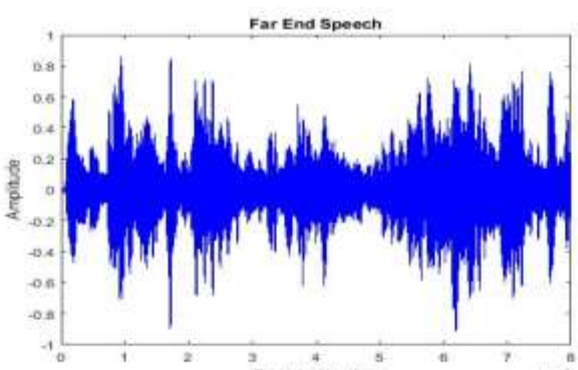


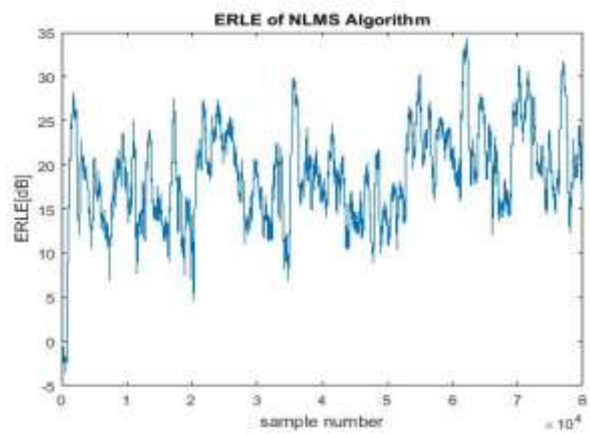
Figure 5 Performance of far end speech

5) SIMULATION RESULTS AND DISCUSSION:

A) NLMS Algorithm:

This algorithm verified to be surpassed than LMS algorithms. The NLMS algorithm was simulated by using MATLAB[7] and Simulations including real input speech signal contained of 80,000 samples and the impulse response of echo path was supposed to have known, 400 points long of h(k). 256 taps was the length which was taken to the length of filter taps. For NLMS technique the value of μ and noise variance coefficient α should be 0.8 and 0.0001 & the close point speaker should be noisy.

It shows the Microphone signal in figure 6. It observes the residual echo is extremely tiny. The performance of Mean square error shown in figure 7. The average of the MSE quickly decays nearly to 0. ERLE for NLMS shown in figure8. It shows the ERLE for NLMS algorithms have higher peaks so merging is quicker and additional echo suppression is achieved compared.



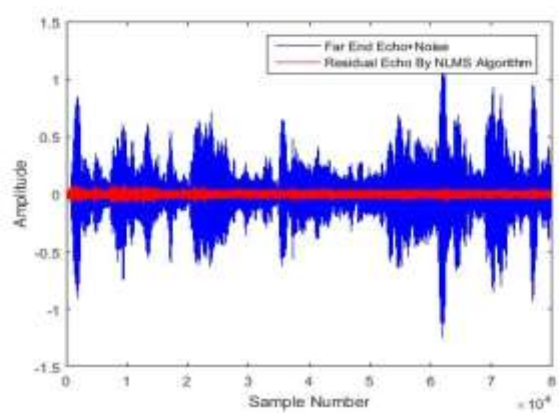


Figure 6 Evaluation of remaining residual echo in NLMS algorithm in Microphone Signal.

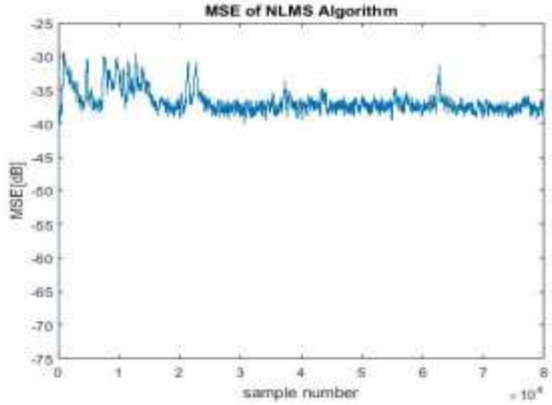


Figure 7 performance of MSE by using NLMS algorithms

Figure 8 performance of ERLE by using NLMS algorithm

The simulation result shows that giant step size of ERLE (in red) greater peaks than the two lesser rate of step size (in blue & magenta). The conjunction performance of NLMS technique totally based on the value of step size parameter. So, design of given curve of MSE and ERLE for different values of μ (.4, .7 & .10) are illustrated in figure 9 & figure 10 separately.

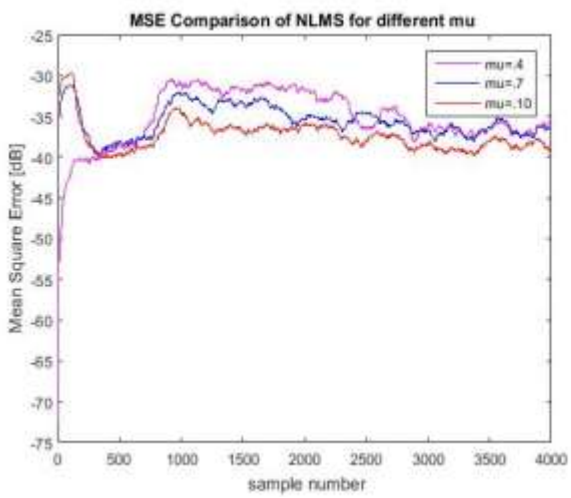


Figure 9 Performance of NLMS algorithms for different step size in MSE.

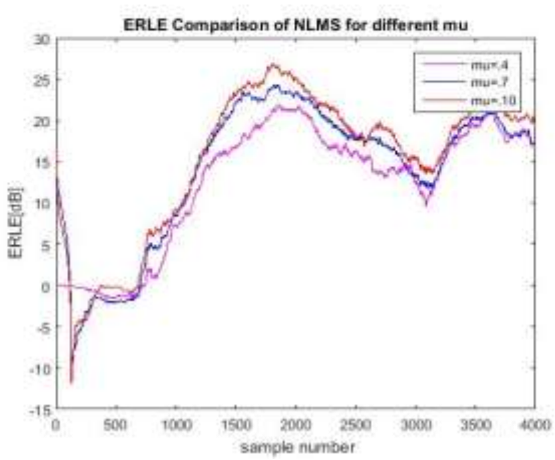


Figure 10 Performance of NLMS processes for changed step size in ERLE.

B) RLS Algorithm:

The RLS algorithmic program was simulated in MATLAB[7]. This procedure shows to be actual effective and outperformed all actual speech input consists of 80,000 sample points & also the echo path was expected to known impulse response, h(k) of 400 points long. The parameter of RLS algorithmic program λ and δ was set to be 0.99 and 0.004 and also the close point speaker was expected to be strident. At 0.0001 is the noise variance to be set.
Figure 11 shows the acoustic echo path impulse from wherever output of speaker is crossed. Residual echo of RLS filter and its matched with microphone signal in fig. 12. It is seen that the left over echo is very less. Mean square error performance is shown in figure 13. The average of the MSE quickly decays nearly to 0. ERLE for RLS algorithm is shown in figure 14. it is seen that the ERLE for RLS process has peak point than NLMS algorithm therefore convergence is faster and a lot of echo suppression is achieved for the RLS algorithm

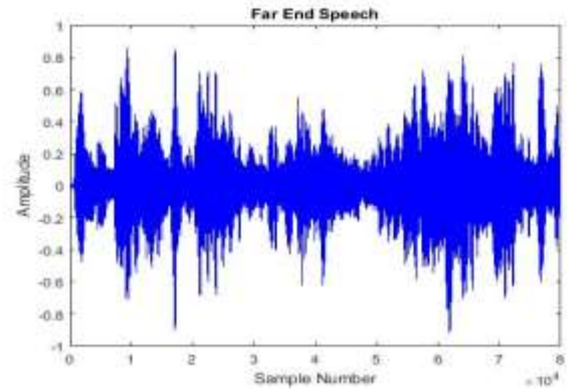


Figure 11 Performance of far end speech

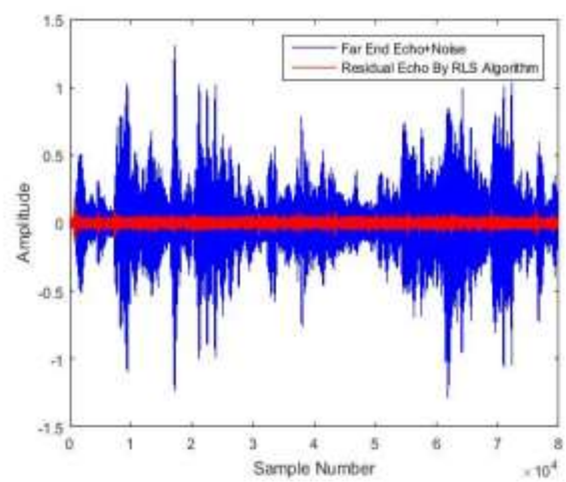


Figure 12 Evaluation of remaining residual echo in NLMS algorithm in Microphone Signal.

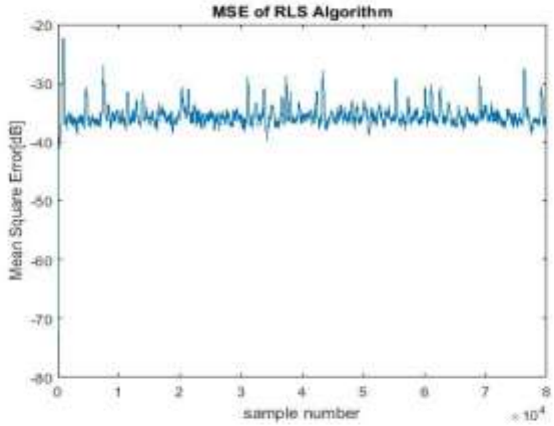


Figure 13 MSE Performance of RLS Algorithm.

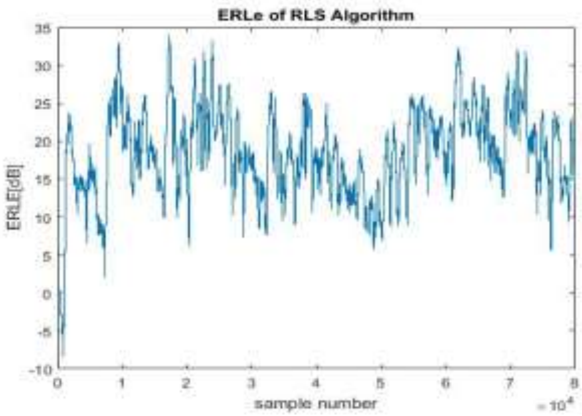


Figure 14 ERLE Performance of RLS Algorithm

The forgetting element λ is ordinarily chosen to be $1-\frac{1}{M}$ wherever M is that the filter length. This value is chosen such for the simplest tracking behavior. However, for stability reason, the selection of λ is ordinarily chosen to be $1-\frac{1}{(3M)}$ and $1-\frac{1}{10M}$. The convergence behavior of the RLS algorithmic rule is depends on the λ. as subordinate degree illustration, the training curves of MSE & ERLE for 3 totally dissimilar values of the forgetting issue are depicted in fig. 15 and fig 16 correspondingly. Increasing the value of λ leads to quickly adaptation, the optimum performance results when λ= .993.

From figure 15 and figure 16, its determined that there is minor mis-adjustment error for large value of forgetting factor (λ= .993) and learning curves converges all the way down to a suitable steady state situation. On the other side, the rate of convergence will increase however provides large steady state mis- adjustment error.

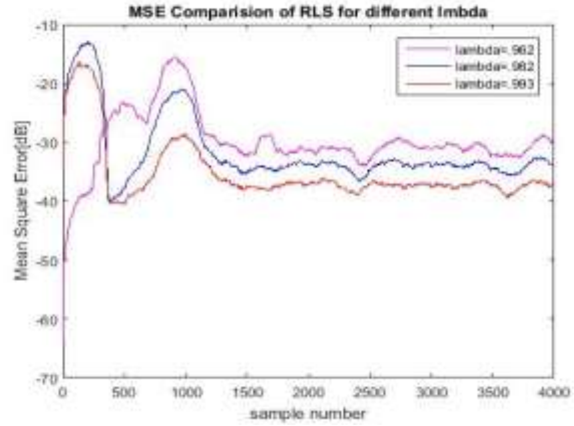


Figure: 15 Performance of RLS algorithms for different step size in MSE

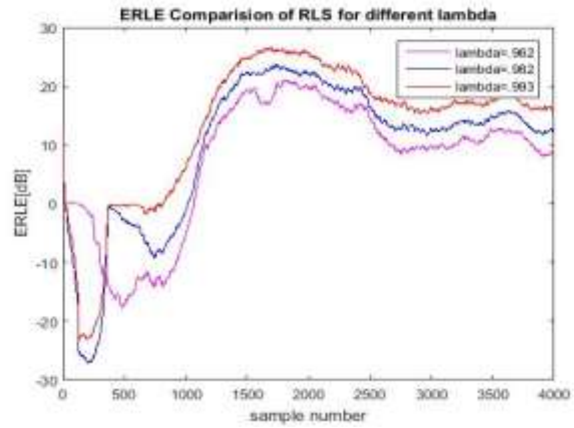


Figure: 16 Performance of NLMS algorithms for different step size in ERLE.

5)Conclusion

In this paper, system of ACE was realized by using adaptive filter process method, NLMS process & RLS process to calculate the best performance between them. It is observed that when we increase the step size (μ) then error also increase. In order to reduce the error, minimum value of step size should be taken. The main reason of this method is to higher the convergence rate, lower mean square error and also lowers computational complexity. Thus marking these points, it conclude that in NLMS algorithm computational complexity is less so speed of convergence is slow, stable and less robustness. It also conclude that every method has its merit & demerits so they should useful according to the need of the condition.

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