Design And Simulation Of An Acoustic Echo Cancellation System For Hand-Free Telecommunication

Ein Gyin Pwint, Su Su Yi Mon, Hla Myo Tun

Abstract: Acoustic echo cancellation is a common occurrence in today's telecommunication systems. The signal interference caused by acoustic echo is distracting to users and causes a reduction in the quality of the communication. This paper is implementing the overall system of acoustic echo cancellation system using LMS and NLMS algorithms for adaptive filter, normalized cross correlation (NCC) algorithm double talk detector. The result of echo return loss enhancement (ERLE) and mean squared error (MSE) which show that how much the amount of echo signal cancelled and the amount of residual error signal for cancelling acoustic echo cancellation on a PC with the help of the MATLAB software.

Keywords: LMS, NLMS, NCC, MSE, ERLE, MATLAB

I. Introduction

In teleconference system figure 1, the speech signal from far-end generated from loud speaker after directing and reflecting from the wall, floor and other objects inside the room is receipt by microphone of near-end, as the result, this makes the echo that is sent back to the far-end. The acoustic echo problem will disturb the conversation of the people and reduce the quality of system. This is a common problem of the communication networks. [1].

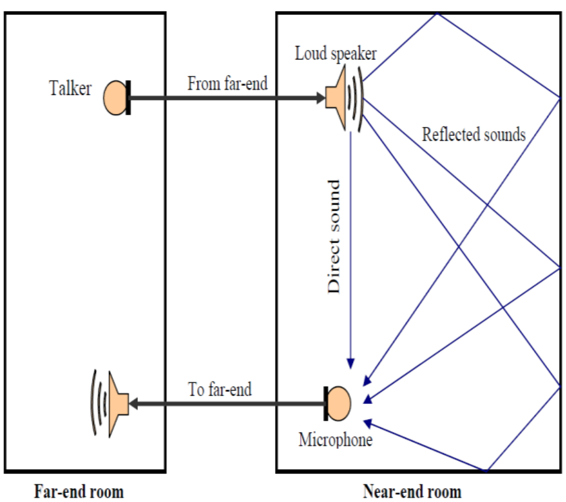

Figure 1. *A teleconference system with echo paths of room[2]*

Using a hands free loudspeaker telephone, the microphone and loudspeaker are enclosed in a single unit. The sound from the loudspeaker will be picked up by the microphone and transmitted back to the sender which is recognized as an echo. Acoustic echo canceller (AEC) suppresses the unwanted echo signal. to improve the conversation between far-end speaker and near-end speaker.

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II. System Overview

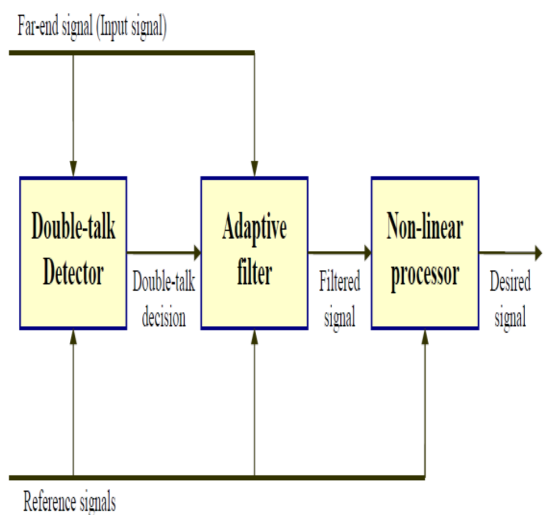


Figure 2. *Block diagram of AEC*

This paper describes a block diagram of an AEC system as in figure 2.This system consists of following three components:

1. Adaptive filter.
2. Doubletalk detector.
3. Nonlinear processor.

A. Adaptive Filter

Adaptive filter is the most important component of acoustic echo canceller and it plays a key role in acoustic echo cancellation. The adaptive filter estimates the echo path, based on which a replica of the echo is created and subtracted from the combination of the actual echo and the near-end speech signal. It requires an adaptive update to adapt to the environmental change.

B. Double-talk Detector

Double talk occurs when both ends are talking. The task of a doubletalk detector is to arrange the correct order of signal to pass adaptive filter.

C. Non-linear Processor
Nonlinear processor (NLP) is used to minimize residual error signal to obtain nearly near end signal by adjusting and training step size values.

III IMPLEMENTATION OF ADAPTIVE FILTER
Adaptive filtering is the process which is required for echo canceling in different applications. Adaptive filter is such type of filter whose characteristics can be changed for achieving optimal desired output. An adaptive filter can change its parameters to minimize the error signal by using adaptive algorithms. The error is the difference between the desired signal and the output signal of the filter. The figure below shows the basic model of adaptive filter used in AEC.

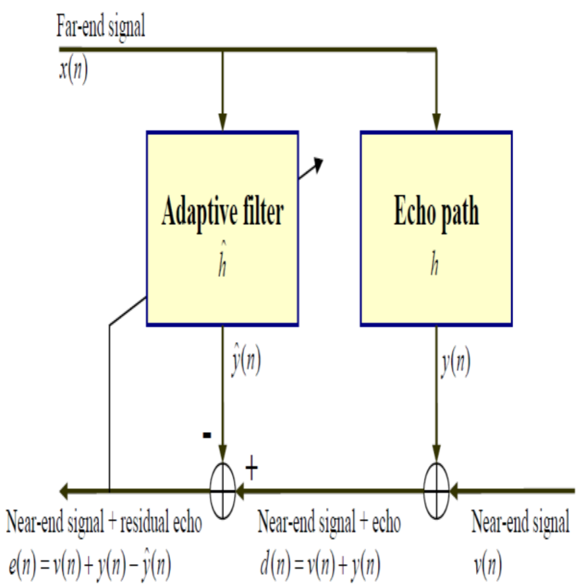


Figure 3. *The basic model of Adaptive Filter*

The notations are used in the figure above and during this thesis in turn are:

- Far-end signal: x(n)
- Near-end signal: v(n)
- The true echo path (room impulse response): h
- Echo signal: y(n)
- Microphone signal: d(n) = v(n) + y(n)
- Estimate echo path: $\hat{h}(n)$
- Estimate echo signal: $\hat{y}(n)$
- Error signal: e(n) = v(n) + y(n) – $\hat{y}(n)$

The echo path h of the room normally variable depends on the room structure and the moving object inside. The estimated echo $\hat{y}(n)$ is calculated from the reference input signal x(n) and the adaptive filter $\hat{h}(n)$. The near-end signal v(n) and background noise are added into echo signal y(n) to create the desired signal d(n) ,

$$d(n)=v(n)+y(n)$$

The signal x(n) and y(n) are correlated. Then, it get the error signal as,

$$e(n)=d(n)-\hat{y}(n)=v(n)+y(n)-\hat{y}(n)$$

The adaptive filter works to minimize the echo (y(n) – $\hat{y}(n)$) to be zero to obtain only near-end signal v(n) in the perfect case.

IV IMPLEMENTATION OF DOUBLE-TALK DETECTOR
In Acoustic Echo Cancellation, the most difficult problem is to handle with the situation of Double-talk presence. Double-talk occurs when far-end and near-end talk at the same time, as a result, the far-end speech signal is corrupted by near-end signal. To solve this problem, one introduces the Double-talk Detector. The task of DTD is freezes the adaptation step during filtering algorithm in case of near-end speech present to avoid the divergence of adaptive algorithm. Without DTD, when the near-end talking would make the system estimation process fail and produce extremely erroneous results. Now we see the Figure (4), when near-end speech is not present (v(n) = 0) then the adaptive algorithm will quickly converse to an estimate echo path. This is the best case of canceling echo. But when near-end speech present (v(n) ≠ 0) then this signal could influence to the adaptation of the filter and cause the divergence. The process of adaptive algorithm will be incorrect and the echo cannot be removed.

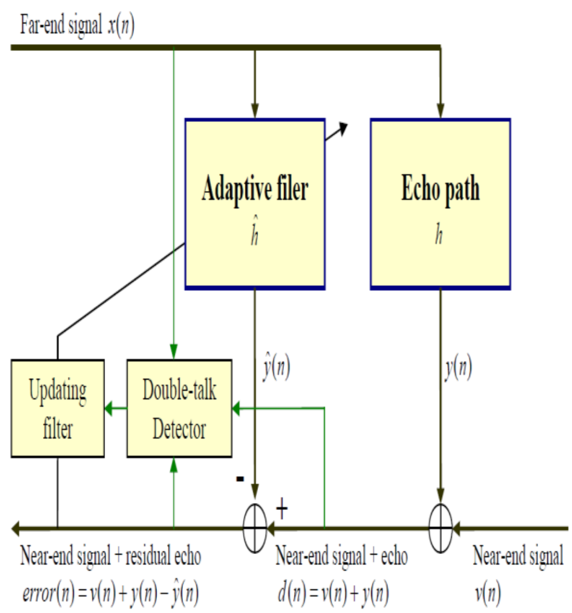


Figure 4. *Double-talk detector with AEC*

V SYSTEM DESIGN FOR ECHO CANCELLATION
The echo cancellation system design flowchart is shown in following figure 5. This flow chart is designed for acoustic echo cancellation system using LMS algorithm. This program must created echo signal and desired signal d(n). From overall flow chart, the process must read far-end signal, x(n) and room impulse response, h. Then, the program created echo signal from near-end speech signal, v(n) and read near-end signal. The program must create desired signal, d(n) to compare the result from the output results. The echo canceller run using LMS and NLMS algorithm to compute error signal. In double detection process, normalize cross correlation (NCC) method can use to check for double case. If double-talk case has happened, the adaptive filter can't update coefficient of weight vector. In this case, the weight vector still worked at previous single talk condition. If double-talk case has not existed, the adaptive filter updated coefficient of weight vector.

Then, the echo cancellation process gets residual echo by subtracting estimate echo from the desired signal. Finally, the nonlinear processor removed the remaining echo from adaptive filter output.

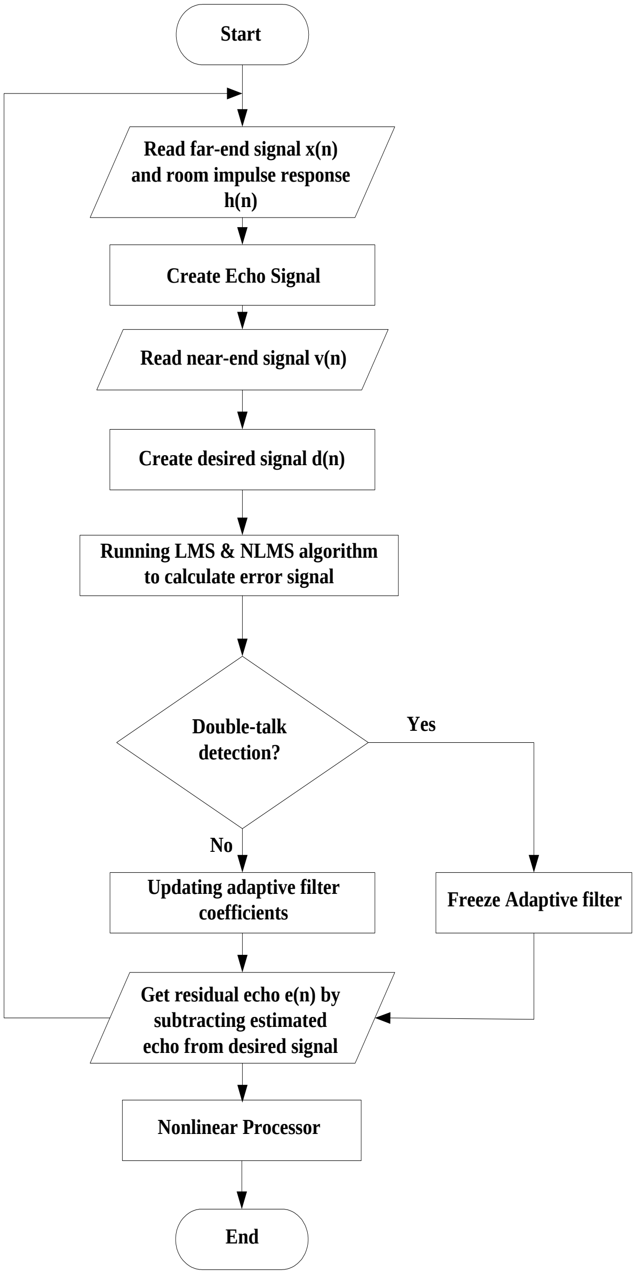


Figure 5. *Flowchart for Echo Cancellation*

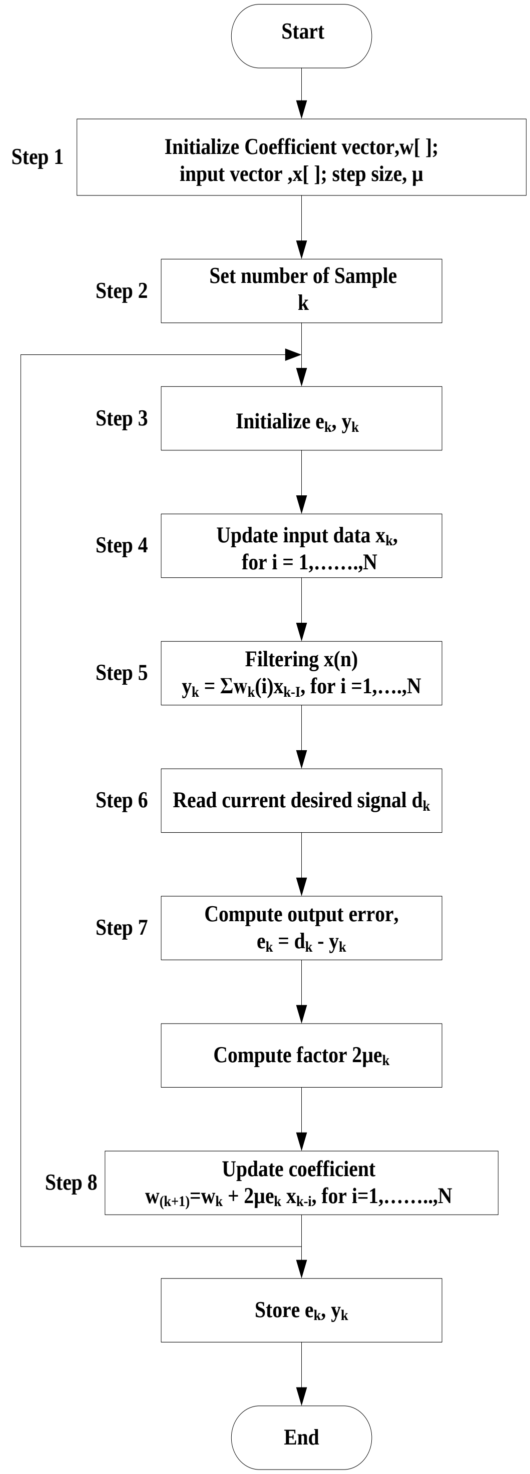


Figure 6. *Flowchart for LMS Algorithm*

A. Implementation of LMS algorithm

The least mean square, (LMS), is a search algorithm that is widely used in various applications of adaptive filtering. The main features that attracted the use of the LMS algorithm are low computational complexity, proof of convergence in stationary environments and stable behavior when implemented with finite precision arithmetic.

B. Implementation of NLMS algorithm

There are a number of algorithms for adaptive filters, which are derived from the conventional LMS algorithm. The objective of the alternative LMS-based algorithms is either to reduce computational complexity or convergence time. The normalized LMS, NLMS algorithm utilizes a variable convergence factor that minimizes the instantaneous error.

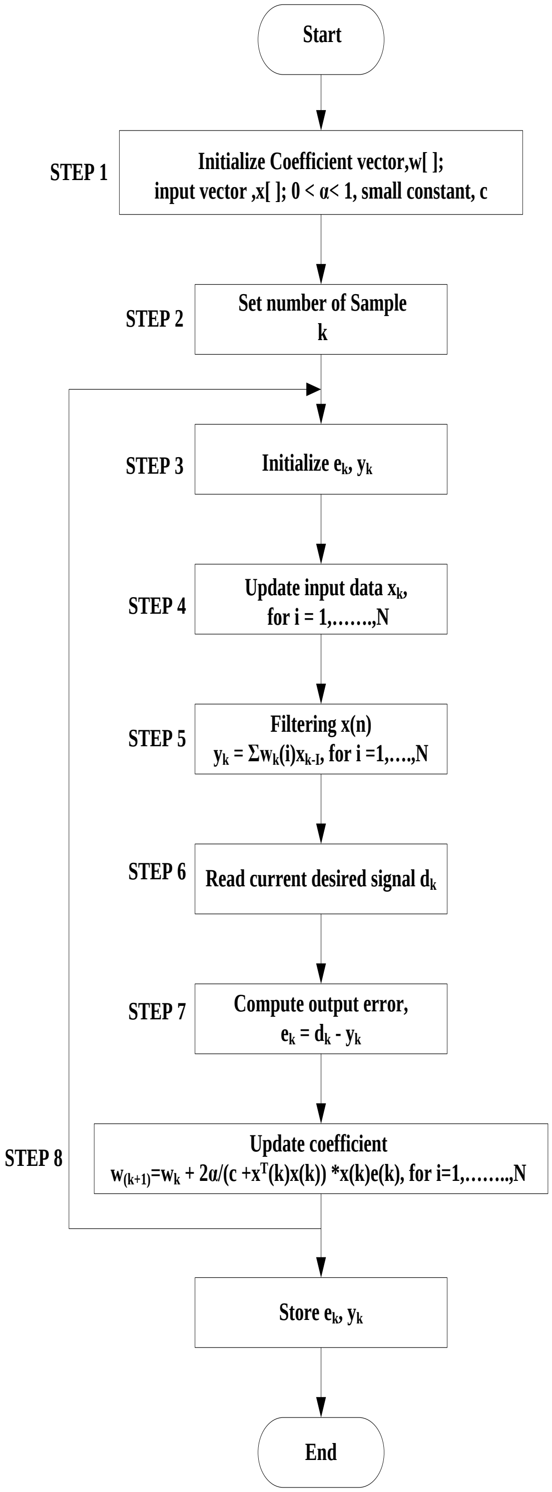
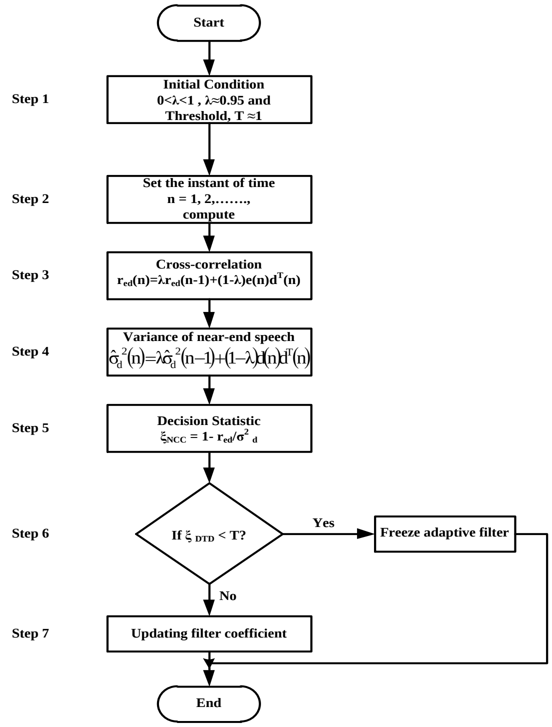


Figure7. *Flowchart for NLMS Algorithm*

C. Implementation Normalized cross-correlation (NCC) algorithm

Acoustic echo cancellation for doubletalk detection uses the Normalized Cross- Correlation algorithm [3]. The NCC algorithm computes the decision statistic depending on the relations of microphone signal and error signal. It can be approached by considering the values of variance of near-end signal and cross-correlation between error signal and microphone signal.



Step 1: Initialize condition for this simulation, Threshold, $T \approx 1$ and λ = 0.95, $0 \langle \lambda \langle 1$

Step 2: Compute the instant of time n = 1,2,…compute

Step 3: The cross-correlation r_{ed} between the error signal e(n) and microphone signal d(n) which is given as, Cross-correlation: $r_{ed}(n) = \lambda r_{ed}(n-1) + (1-\lambda)e(n)d^T(n)$

Step 4: Calculate the near-end speech to decision double-talk condition, $\hat{\sigma}_d^2(n) = \lambda\hat{\sigma}_d^2(n-1) + (1-\lambda)d(n)d^T(n)$

Step 5: Decision double-talk statistic using equation,

$$\xi_{NCC} = 1 - \frac{\hat{r}_{ed}}{\hat{\sigma}_d^2}$$

Step 6 and 7: If $\xi_{DTD} \langle T$, freeze adaptive filter If $\xi_{DTD} \rangle T$, updating adaptive filter coefficients

VI Performance Evaluation

Two types of performance used in this paper are

- Mean square error (MSE)
- Echo return loss enhancement (ERLE)

I. MEAN SQUARED ERROR (MSE)

Mean squared error (MSE) is the sequence of mean squared error. This column vector contains predictions of the mean squared error of adaptive filter at each time instant. The mean squared error is calculated as

$$\text{MSE} = \frac{\sum e^2}{n}$$

Above unknown variables are e is error signal and n is number of samples.

II. ECHO RETURN LOSS ENHANCEMENT (ERLE)

The ERLE is defined as the ratio of microphone signal power (Pd) and the power of a residual error signal immediately after the cancellation (Pe), and it is measured in db. The ERLE measures the amount of loss introduced by the adaptive filter alone. ERLE depends on the size of the adaptive filter and the algorithm design. ERLE is a measure of the echo suppression achieved and is given by

$$ERLE = 10\log_{10}\frac{P_d}{P_e}$$

VII SIMULATION RESULTS

For testing the LMS and NLMS adaptive algorithms in acoustic echo cancellation, the same speech signal has 8kHz sampling frequency and 4kHz frequency with duration of 2.5 seconds and magnitude of the range between -1 and 1 for PC sound card for near-end signal, far-end signal and far-end echo signal which is created from far-end signal.

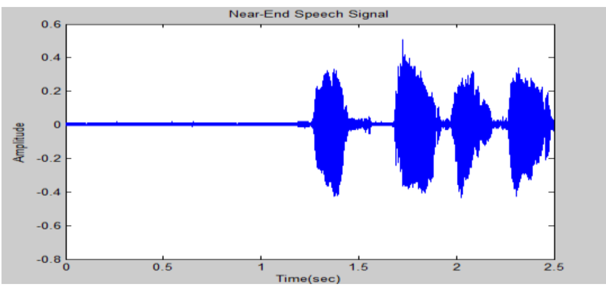


Figure 8. *Near-end Speech Signal for LMS and NLMS*

Figure 8 shows the result of near-end signal without echo which is first recorded.

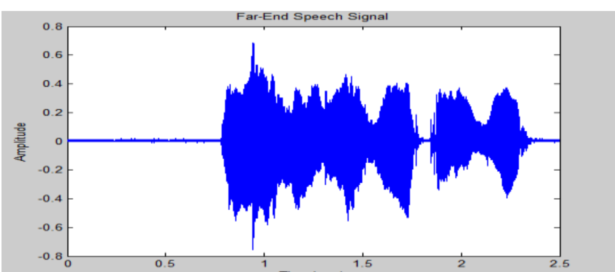


Figure 9. *Far-end Speech Signal for LMS and NLMS*

Figure 9 shows the result of far-end signal without echo which is second recorded.

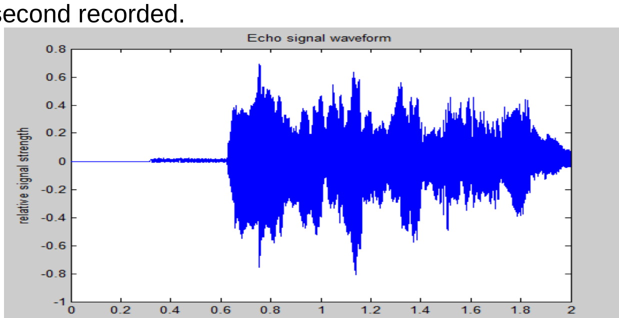


Figure 10. *Echo Signal for LMS and NLMS*

Figure10 shows the result of echo signal which is passed through echo path.

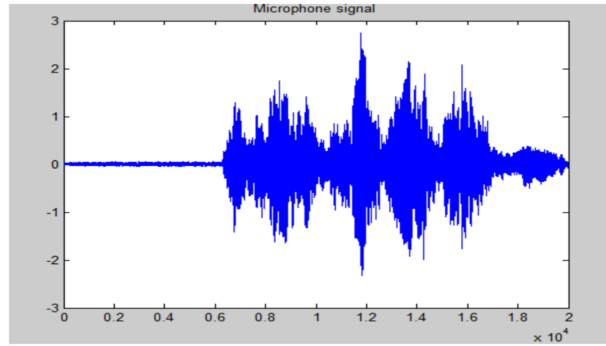


Figure 11.*.Microphonefor LMS and NLMS*

Figure11 shows the result of microphone signal which is combination of near-end signal and echo signal.

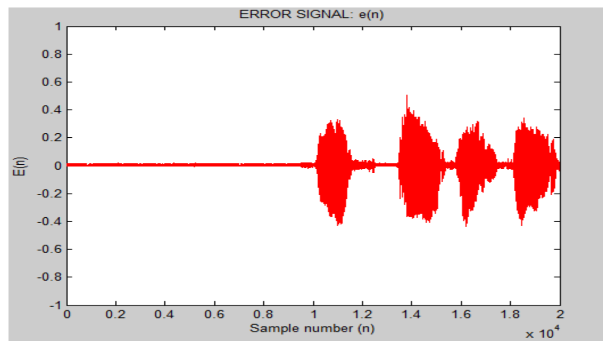


Figure 12. *Error signal for LMS*

Figure12 shows the result of error signal for LMS which is subtraction of echo signal from microphone signal.

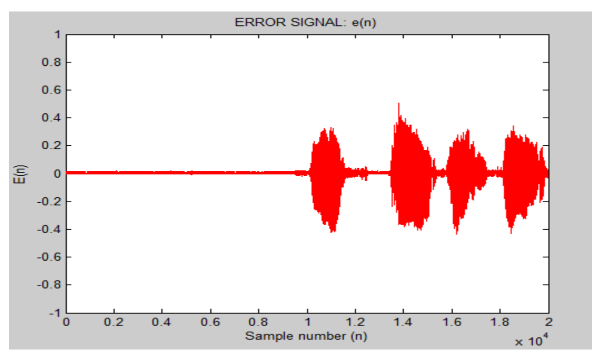


Figure 13. *Error signal for NLMS*

Figure13 shows the result of error signal for NLMS which is subtraction of echo signal from microphone signal.

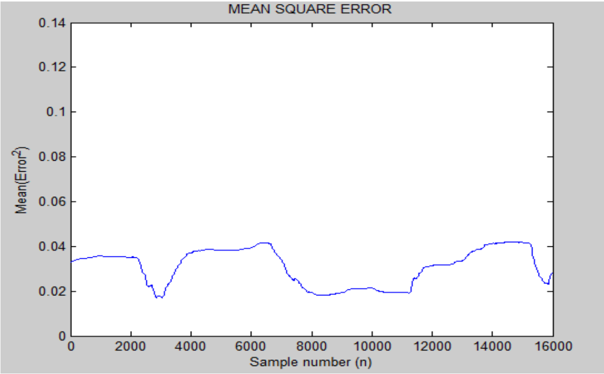


Figure 14. *MSE for LMS*

Figure14 shows the result of MSE for LMS which is root mean square value of error signal to compare MSE of NLMS .

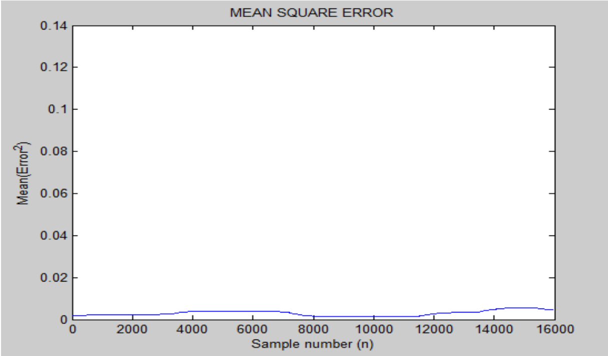


Figure 15.*MSE for NLMS*

Figure15 shows the result of MSE for NLMS which is root mean square value of error signal to compare MSE of LMS.

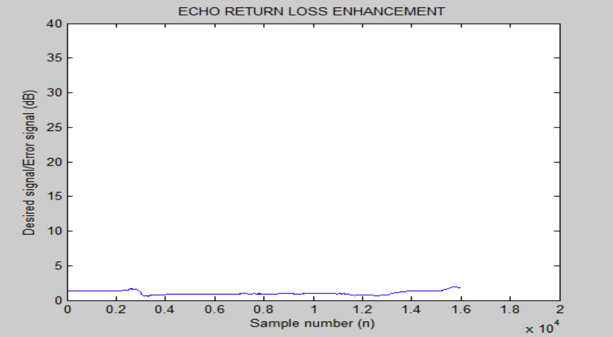


Figure 16. *ERLE for LMS*

Figure16 shows the result of the result of ERLE for LMS after comparing with microphone signal and error signal with dB range and duration of 2.5 seconds.

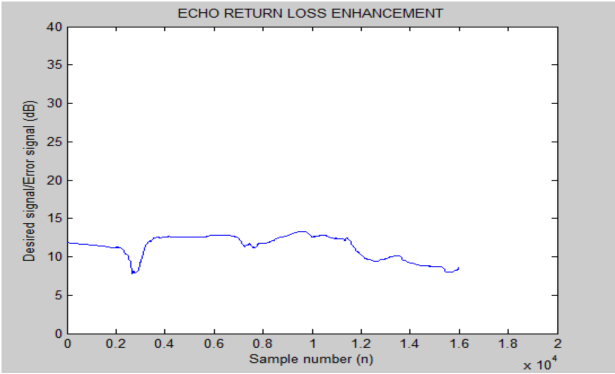


Figure 17. *ERLE for NLMS*

Figure13 shows the result of the result of ERLEfor LMS after comparing with microphone signal and error signal with dB range and duration of 2.5 seconds.

TABLE I
COMPARISON OF MSE AND ERLE

Algorithms	MSE	ERLE(dB)
LMS	0.038	2
NLMS	0.014	12

VIII. CONCLUSION

This paper is implementing the overall system of acoustic echo cancellation system using LMS and NLMS algorithms for adaptive filter, normalized cross correlation (NCC) algorithm double talk detector. The result of echo return loss enhancement (ERLE) and mean squared error (MSE) which show that how much the amount of echo signal cancelled and the amount of residual error signal for cancelling acoustic echo cancellation on a PC with the help of the MATLAB software**.** The higher ERLE in dB has higher rate of convergence with low complexity and the higher rate of MSE has the less amount of error signal. The lower ERLE in dB has higher rate of divergence with high complexity and the lower rate of MSE has the more amount of error signal. The NLMS algorithm has higher the value of ERLE and lower value of MSE than LMS algorithm. Thus, NLMS algorithm is the better the echo canceller and has minimum residual error signal and maximum reduced echo signal to train by using NLP.

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