Comparison of Performance of Stereophonic Acoustic Echo Canceller using LMS and NLMS Adaptive Algorithms

[1]Sandeep Beniwal, [2]Dr. Sanjeev Kumar Dhull
Student, Department of ECE, Guru Jambheshwar University of Science & Technology, Hisar (Haryana), India[1]
Department of ECE, Guru Jambheshwar University of Science & Technology, Hisar (Haryana), India[2]

***Abstract*— Acoustic echo cancellation is a common occurrence in today's telecommunication systems. Acoustic echo occurs when a speech signal is reverberated in real environment. Acoustic echo causes the signal interference which is distracting to users and reduces the quality of the communication. This paper focuses on the use of LMS and NLMS algorithms to reduce this unwanted echo in case of stereophonic acoustic echo cancellation, thus increasing the quality of speech.**

***Keywords*- Acoustic echo cancellation, SAEC, adaptive filter, LMS, NLMS**

I. INTRODUCTION

Communication is the area which has undergone tremendous advancement in recent decade. As more and more people are using personal communication devices, personal computers and wireless mobile phones, therefore there is need of advance secure and noise free communication systems. Audio conference plays a key role in communication system which is cost effective and also aimed for user comforts. The problem often arises during the conversation is the creation of acoustic echo. The acoustic echo is generated at one end (speaker side) and the microphone causes disturbance to the speaker at the other end (listener side) in which listener hear his own voice with delay. This is called acoustic echo. This leads to degradation in the speech quality at the receiving end. The degradation in the received speech signals makes conversation between the users uncomfortable. The main aim of acoustic echo canceller is to provide a noise free voice quality when two or more people communicate from different places[1]. In conference communication system, the microphone pick up both the far end speech and the local speech, where former is produced by loudspeaker and the latter is produced by the local speaker. Besides degrading the recorded speech, it can also result in acoustic feedback. As the microphone signal is amplified and sent to the far end where it's again fed to remote microphone. Thus every time, the generated acoustic echo is fed back to the other end which result in disturbances in conversation. The acoustic feedback is often perceived as a loud tone at other end. In a telephone conferencing set up, the contribution of loudspeaker signal to microphone signal can be eliminated using different methods of echo cancellation. The conventional acoustic echo cancellation (monophonic acoustic echo cancellation)[1] identify the acoustic echo path and simultaneously eliminate it but the main problem with conventional acoustic echo cancellation is that it does not take care about what will happen at the remote transmission room.

II. STEREOPHONIC ACOUSTIC ECHO CANCELATION

Stereophonic acoustic echo cancellers (SAEC) are fundamentally different from conventional echo cancellers (monophonic/single-channel echo cancellers). Here stereophonic implies two audio channels. Stereophonic acoustic echo cancellation can be viewed as a straightforward generalization of the single-channel acoustic echo cancellation principle [2].

A general stereophonic acoustic echo canceller is shown in Figure 1. The transmission room is referred as far-end and receiving room is referred as near-end. Here two loudspeakers and two microphones are used at each end. Thus there are now four acoustic echo paths to identify-two for each microphone [3]. This will not only cause increased calculation complexity, but also a new fundamental problem of solution. The fundamental problem is that the two audio channels may carry linearly related signals which in turn may cause the normal equations to be solved by the adaptive algorithm singular.

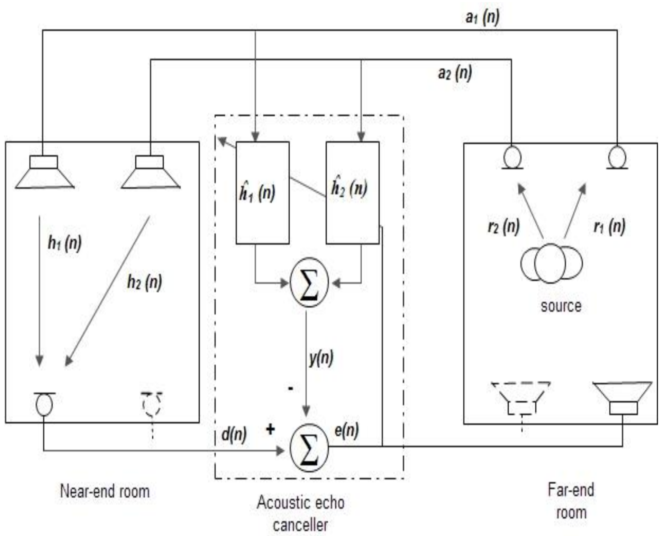

Figure 1: Schematic Diagram of SAEC

Stereophonic acoustic echo cancellation consists of direct identification of a multi-input, unknown linear system, consisting of the parallel combination of two acoustic paths h_1, h_2 extending through the receiving room from the loudspeakers to the microphone. The stereophonic AEC tries to model this unknown system by a pair of adaptive filter $\hat{h}_1, \hat{h}_2$. In a stereophonic acoustic echo cancellation system shown in Figure 1, $s_1(n)$ is the source of $a_1(n)$ and $a_2(n)$ signals in the transmission room. We have $a_i(n) = r_i(n)s(n)$, where $r_i(n)$ is the impulse response between the source and microphone in the transmission room. There is a strong cross-correlation between two input signals (a_1, a_2) which cause the main problems in stereophonic acoustic echo cancellation. The input signal vectors $a_1(n)$ and $a_2(n)$ and filter coefficient vectors $\hat{h}_1(n)$ and $\hat{h}_2(n)$ are combined

$$a(n) = \left[a_1^T(n), a_2^T(n)\right]^T \;\&\; h(n) = \left[\hat{h}_1^T(n), \hat{h}_2^T\right]^T \quad [3].$$

The combined filter coefficient vector $\hat{h}(n)$ is updated by an adaptive algorithm. Early examples of SAEC implementations were mainly based on the use of a single adaptive filter for each return channel. The performance of most adaptive algorithms depends on the condition number of the input signal's covariance matrix[4]. In the SAEC case, the condition number is very high; as a result, algorithms such as least mean squares (LMS) or normalized least mean squares (NLMS), that do not take the cross-correlation between the input signals into account, converge very slowly to the true solution. Straightforward extensions of single channel algorithms may not be the best choice for the SAEC application.

This paper draw attention on the performance of SAEC using two adaptive filters LMS and NLMS. The performance of SAEC is measured using echo return loss enhancement (ERLE) and mean square error (MSE). The echo return loss enhancement (ERLE)[13] is a measure of how good an echo canceller is in terms of steady-state residual echo and convergence time. Let $\sum_{k=1}^{n} y_j^2(k,h)$ be the power of echo signal $y_j(k,h)$ at time k, and $\sum_{k=1}^{n} e_j^2(k)$ be the power of the residual-echo signal. The ERLE is given as $ERLE = 10\log_{10}(\sum_{k=1}^{n} y^2(k) / \sum_{k=1}^{n} e^2(k))$ .The mean square error (MSE) [1] is used as a performance index in stereophonic acoustic echo cancellation system. The MSE energy of residue echo is used in SAEC. The MSE is defined as $MSE = P_e / P_d$.

III. ADAPTIVE FILTERS

Adaptive filters are dynamic filters which updates their parameters in every iteration in order to converge to an optimal desired output. These filters algorithmically change their characteristics in order to minimize a function which is difference between the desired output d(n) and its actual output y(n). This function is known as the cost function of the adaptive algorithm. Figure 2 shows a block diagram of the adaptive echo cancellation system.

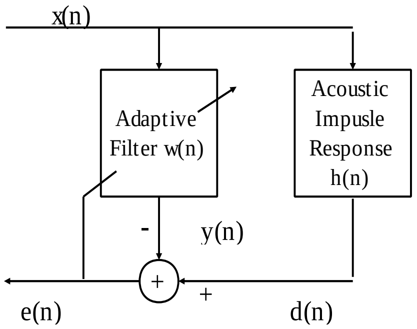

Figure 2: Adaptive Echo Cancellation

Here the filter h(n) represents the impulse response of the acoustic environment, w(n) represents the filter weights. The aim of adaptive filter is to equate its output y(n) to the desired output d(n). At each iteration the error signal, e(n) =d(n)-y(n), is fed back into the filter, where the characteristics of filter are altered accordingly.

A. LEAST MEAN SQUARE (LMS) ALGORITHM

LMS is one of the most widely used algorithms in adaptive filtering technique. The LMS algorithm is stochastic gradient-based algorithms as it utilizes the gradient vector of the filter tap weights to converge on the optimal wiener solution [14]. It is popular in use because of its computational simplicity. It is this simplicity that has made it the benchmark against which all other adaptive filtering algorithms are judged.

The filter tap weights of the adaptive filter are updated in each iteration according to the following equation:

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

Here x(n) is the input vector of time delayed input values, $x(n) = [x(n)\ x(n-1)\ x(n-2)\ ..\ x(n-N+1)]^T$. The vector $w(n) = [w_0(n)\ w_1(n)\ w_2(n)\ ..\ w_{N-1}(n)]^T$ represents the coefficients of the adaptive FIR filter tap weight vector at time n. The parameter μ is known as the step size parameter and is a small positive constant. This step size parameter controls the influence of the updating factor. Selection of a suitable value for μ affects to the performance of the LMS algorithm, if the value is too small the time the adaptive filter takes to converge on the optimal solution will be too long; if μ is too large the adaptive filter becomes unstable and its output may diverges.

B. NORMALIZED LEAST MEAN SQUARE (NLMS) ALGORITHM

One of the main disadvantages of the LMS algorithm is having a fixed step size parameter for every iteration. The normalized least mean square algorithm (NLMS) is an extension of the LMS algorithm which overcome this problem by selecting a different step size value, μ(n), for each iteration of the algorithm [14]. This step size is proportional to the inverse of the total expected energy of the instantaneous values of the coefficients of the input vector x(n). This sum of the expected energies of the input samples is also equivalent to the dot product of the input vector with itself, and the trace of input vectors auto-correlation matrix, R.

$$R = \sum_{i=0}^{N-1} E[x^2(n-l)] \qquad (2)$$

The recursion formula for the NLMS algorithm is stated in equation 3 [14]:

$$w(n+1) = w(n) + \frac{1}{x^T(n)x(n)} e(n)x(n) \qquad (3)$$

IV. SIMULATION RESULTS

For the implementation of stereophonic acoustic echo cancellation (SAEC) MATLAB 7.10.0 software is used. Here for simulation of the stereophonic acoustic echo canceller we assume that both far-end and near-end rooms have the same characteristics (size, acoustic features). The speech signals (including far-end and near-end signals) used in MATLAB software at the sampling rate of 8 kHz. The speech signal is the audio signal.

A. SIMULATION RESULT OF LMS ALGORITHM

Figure 3 shows the near-end signal, far-end signal and the error signal for the LMS adaptive filter. Figure 4 shows the echo return loss enhancement (ERLE) which is a measure of performance of SAEC. Figure 5 shows the graph of mean square error (MSE). The length of LMS adaptive filter is set to 128. The step size was 0.02. The value of ERLE should be high and the value of MSE should be minimum for better performance of SAEC. In the figure shown below we can see that as the algorithm progresses the average value of ERLE increases and the average value of MSE decreases.

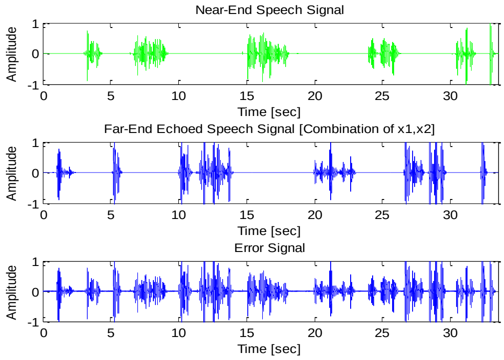

Figure 3: Near-end, Far-end and Error Signal of LMS filter

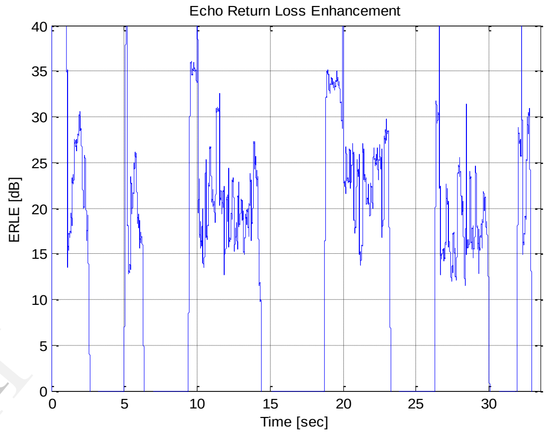

Figure 4: Echo Return Loss Enhancement

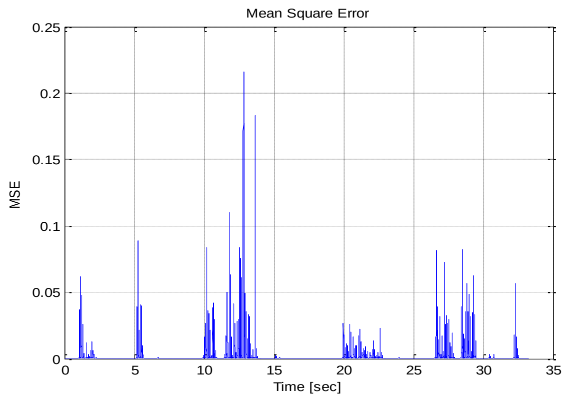

Figure 5: Mean Square Error

B. SIMULATION RESULT OF NLMS ALGORITHM

After the simulation of SAEC using NLMS, Figure 7 shows the echo return loss enhancement (ERLE) and Figure 8 shows the graph of mean square error (MSE) for NLMS algorithm. The length of NLMS adaptive filter is set to 128. The step size was 0.02 and the value of offset was 0.01. The value of ERLE should be high and the value of MSE should be minimum for better performance of SAEC. In the figure shown below we can see that as the algorithm progresses the average value of ERLE increases and the average value of MSE decreases.

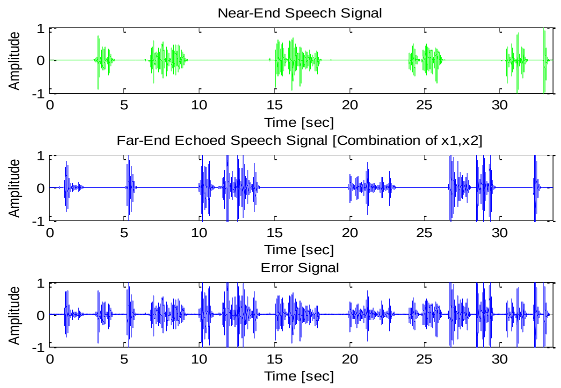

Figure 6: Near-end, Far-end and Error Signal of NLMS filter

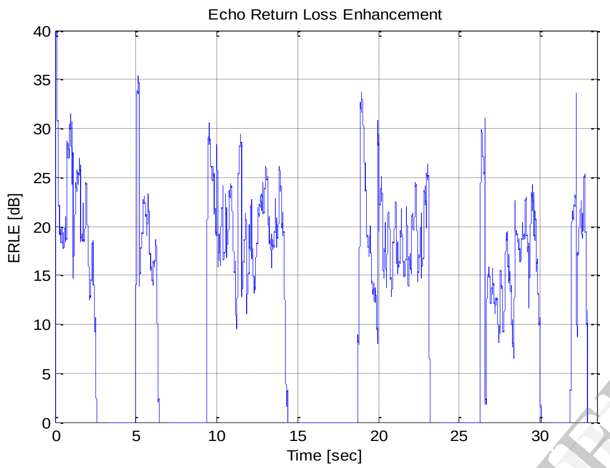

Figure 7: Echo Return Loss Enhancement

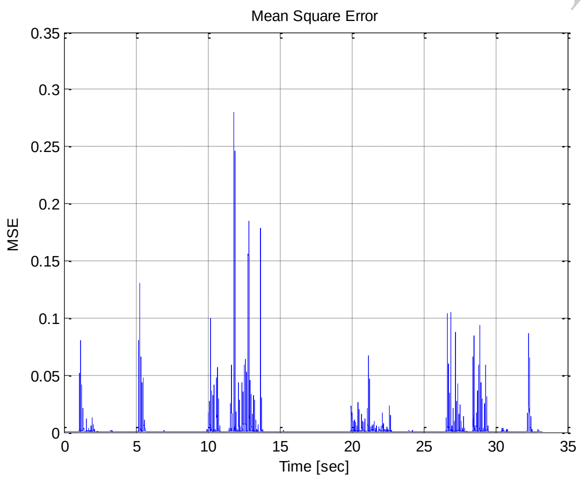

Figure 8: Mean Square Error

IV. CONCLUDING REMARKS

The LMS algorithm is the most popular adaptive algorithm because it is very simple but the LMS algorithm has slow and data dependent convergence behavior. The NLMS algorithm is more robust variant of the LMS algorithm and it exhibits a better balance between simplicity and performance than the LMS algorithm. Due to its good properties the NLMS gives better performance than LMS for stereophonic acoustic echo cancellation.

V. REFERENCES

[1] M. M. Sondhi, D. R. Morgan, and J. L. Hall, *"Stereophonic Acoustic Echo Cancellation– An Overview of the Fundamental Problem"*, IEEE SP Let., vol.2, no.8, pp.148-151, Aug. 1995.

[2] J. Benesty, D. R. Morgan, and M. M. Sondhi., *"A better understanding and an improved solution to the specific problems of stereophonic acoustic echo cancellation"*, IEEE Trans. Speech and Audio Processing, vol. 6, no. 2, pp. 156–165, March 1998.

[3] J. Benesty, F. Amand, A. Gilloire, and Y. Grenier, *"Adaptive filtering algorithms for stereophonic acoustic echo cancellation"*, in Proc. IEEE Int. Conf. Acoustics Speech Signal Processing, pp. 3099–3102, 1995.

[4] Gansler T., Benesty J., *"Stereophonic acoustic echo cancellation and two-channel adaptive filtering: an overview"*,Int. J. Adapt. Control Signal Process, vol.14, pp. 565-586, 2000.

[5] Xiao Hu, Ai-Qun Hu, Qiang Luo and Tian-You Cai,*"A novel adaptive acoustic echo cancellation for teleconferencing system"*, Machine Learning and Cybernetics, 2002. Proceedings 2002 International Conference on Digital Object Identifier, vol.2, pp. 1005 - 1009, 2002.

[6] A. W. H. Khong, P. A. Naylor, *"Stereophonic acoustic echo cancellation employing selective-tap adaptive algorithms"*, IEEE Trans. Audio, Speech, Lang. Process., vol. 14, no. 3, pp. 785–796, May 2006.

[7] H. I. K. Rao, B. Farhang-Boroujeny, *"Fast LMS/Newton algorithms for stereophonic acoustic echo cancellation"*, IEEE Trans. Signal Process., vol. 57, no. 8, pp. 2919–2930, Aug. 2009.

[8] P. Eneroth, S. L. Gay, T. Gänsler and J. Benesty, *"A real-time implementation of a stereophonic acoustic echo canceller"*, IEEE Trans. Speech Audio Process., vol. 9, no. 4, pp. 513–523, Jul. 2001.

[9] Akihiko Sugiyama, Akihiro Hirano, and Kenji Nakayama, *"Acoustic echo cancellation for conference systems"*, NEC Media and Information Research Laboratories

[10] Shoji Makino,*"Review on Stereophonic acoustic echo cancellation: An overview and recent solutions"*, Acoust. Sci. & Tech. 22, (5) Jan. 2001.

[11] T. Gänsler, J. Benesty, *"New insights into the stereophonic acoustic echo cancellation problem and an adaptive nonlinearity solution"*, IEEE Trans. Speech Audio Process., vol. 10, no. 5, pp. 257–267, Jul. 2002.

[12] Harsha I. K. Rao, Behrouz Farhang-Boroujeny, *"Analysis of the Stereophonic LMS/Newton Algorithm and Impact of Signal Nonlinearity on Its Convergence Behavior"*, IEEE Trans. Speech Audio Process., vol. 58, no. 12, Dec. 2010.

[13] T. Gänsler, P. Eneroth, *"Influence of audio coding on stereophonic acoustic echo cancellation"*, in Proc. IEEE Int. Conf. Acoustics Speech Signal Processing, pp. 3649–3652, 1998.

[14] S. Haykin, *"Adaptive Filter Theory, Third Edition"*, Prentice Hall, 1996.

[15] R Chinabonia, D. S. Ramkiran, H. Khan, M. Usha, B. T. P. Madhav, K. P. Srinivas and G. V. Ganesh, *"Adaptive algorithm for acoustic echo cancellation in speech processing'*, IJRRAS, vol. 7, issue 1, pp. 38-42, 2011