IJIREEICE

ISSN (Online) 2321 – 2004
ISSN (Print) 2321 – 5526

International Journal of Innovative Research in
Electrical, Electronics, Instrumentation and Control Engineering
ISO 3297:2007 Certified
Vol. 4, Issue 9, September 2016

Adaptive Filtering Algorithm for Acoustic Echo Cancellation in Hands Free Communication System

Rashi Baraliya[1], Prof. Mugdha Dewasthale[2]

MKSSS's Cummins College of Engineering, Pune, India[1, 2]

Abstract: Acoustic Echo Cancellation (AEC) is a common occurrence in today's telecommunication systems. It occurs when an audio source and sink operate in full duplex mode. In this situation the received signal is output through the telephone loudspeaker (audio source), this audio signal then gets echoed through the physical environment and picked up by the systems microphone (audio sink). The effect is the return to the distant user of time delayed and attenuated images of their original speech signal. The signal interference caused by acoustic echo is distracting to both users and causes a reduction in the quality of the communication. Here the focus is on the use of echo cancellation algorithm by making use of adaptive filtering techniques to reduce this unwanted echo, thus increasing communication quality. Adaptive Filters are a class of filters that iteratively alter their parameters in order to minimize a function of the difference between a desired target output and their output. In the case of acoustic echo in telecommunications, the optimal output is an echoed signal that accurately emulates the unwanted echo signal. This is then used to negate the echo in the return signal. The better the adaptive filter emulates this echo, the more successful the cancellation will be. Performance parameter of pro-posed algorithm will be analyzed using Echo Return Loss Enhancement (ERLE), Mean square Error (MSE), and Convergence Time.

Keywords: Acoustic echo cancellation (AEC), Echo Return Loss Enhancement (ERLE), Mean square Error (MSE), Convergence Time.

I. INTRODUCTION

In real life, echoes often occur among conversations. The echoes of speech waves can be heard as they are reflected from the floor, wall and other neighboring objects. In such a case when the reflected wave arrives a few tens of milliseconds delay after the direct sound, it can be heard as an obvious echo. These echoes are bothering and may unexpectedly interrupt a conversation. Advancement in technology in recent decades has changed whole dimension of communication. Today's world is more interested in hands free communication. In such a case the use of regular loudspeaker and microphone, in place of telephone receiver is more applicable.

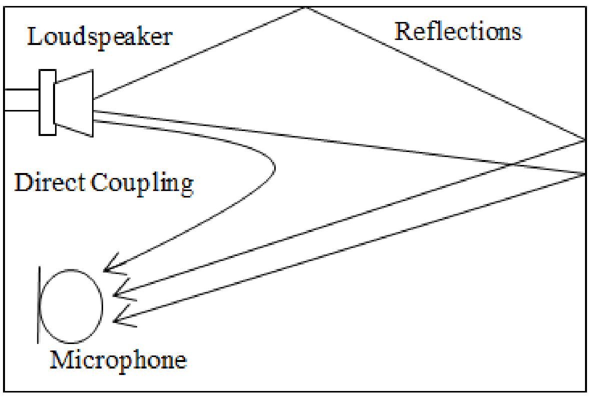


Figure 1: Origin of Acoustic Echoes

The advantage is that it would allow the person to have both hands free and to move freely in the room. However the presence of acoustic coupling between the loudspeaker and microphone would produce an echo that would create conversation difficult. This type of Echo is known as acoustic echo [1]. Generation of Acoustic echoes is shown in Figure 1 below

II. LITERATURE REVIEW

Hung Ngoc Nguyen, et al. presented the study to cancel acoustic echo by AEC. One of the major problems in a telecommunication application over a telephone system is echo. The Echo cancellation algorithm presented in this thesis successfully attempted to find a software solution for the problem of echoes in the telecommunications environment. AEC is the conventional method for solving the acoustic echo problem. Under ideal conditions AEC can achieve perfect echo cancellation, because it estimates both the phase and amplitude of the echo signal. The algorithm was capable of running in any PC with MATLAB software installed. In addition, the results obtained were convincing [1].

Yuksel Ozbay, et al. aimed to enhance the intelligibility of speech by cancelling out the echo noise. For this purpose, the data transfer software, which was necessary for real time processing of voice signals and the adaptive filtering

IJIREEICE
ISSN (Online) 2321 – 2004
ISSN (Print) 2321 – 5526
International Journal of Innovative Research in
Electrical, Electronics, Instrumentation and Control Engineering
ISO 3297:2007 Certified
Vol. 4, Issue 9, September 2016

algorithm software for the application of acoustic echo cancellation, had been developed. An algorithm has been proposed for the determination of optimum adaptation rate for the least mean-square (LMS) adaptation algorithm that is used in the adaptive filter. The effectiveness of optimum value determination algorithm was demonstrated on a single direction voice conference application with one speaker [5].

Ranbeer Tyagi, et al. presented an acoustic echo cancellation model using Least Mean Square (LMS) algorithm. The results showed that the steady state error increases with increase in step size parameter and the optimality of the LMS algorithm is no longer hold. The results also reveal that choosing smallest value of step size parameter guarantees the smallest mis-adjustment [8].

Lu Lu presented a good echo removal algorithm which is capable of providing convincing results for PC application. The basic components of an echo canceller are an adaptive filter and a double-talk detector. The task of a doubletalk detector is to sense the doubletalk, so that to stop the adaptive filter in order to avoid divergence. Since there has been a revolution in the field of personal computers in recent years, he implemented the acoustic echo canceller algorithm on a PC with the help of the MATLAB software [10].

III. PRINCIPLE OF ACOUSTIC ECHO CANCELLATION

In teleconferencing environment, the end speaker speech is picked up by the microphone placed at near end and is again being sent back to far end as an echo. Adaptive Filters are popularly used for the cancellation of acoustic echoes. Adaptive Filters are dynamic filters which iteratively alter their characteristic in order to achieve an optimal desired output [2]. Adaptive Filters uses algorithmic procedures which aims to identify the acoustic path between loudspeaker and microphone and tries to generate the replica of echo path that is to be removed from output of microphone.

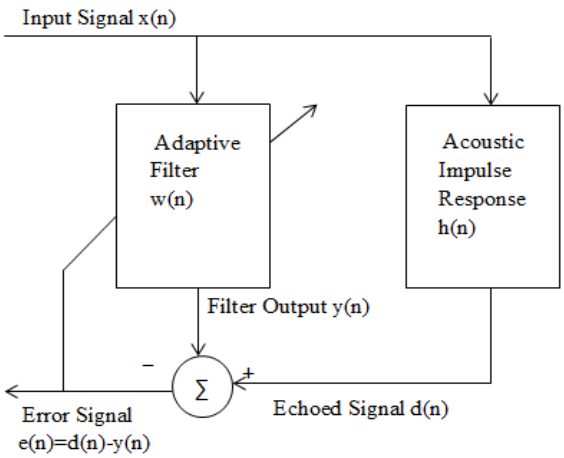


Figure 2: Adaptive Echo Canceller

Figure 2 shows the model for Acoustic Echo Cancellation (AEC). Here x(n) represents the input signal, h(n) acoustic impulse response, w(n) impulse response of adaptive filter which gets iteratively altered to match to actual impulse response, d(n) is the echoed signal which gets created when input signal passes through acoustic environment and error signal is given by :

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

From Figure 2 it can also be noted that past values of error signal e(n)is given back to adaptive filter, to iteratively alter its characteristic up to optimum required output.

IV. ADAPTIVE ALGORITHMS

The adaptive algorithms used for cancellation of echoes are Least Mean Square (LMS), Normalized Least Mean Square (NLMS) and Recursive Least Square (RLS) [3],[4].

4.1 LMS ALGORITHM: Due to the computational simplicity, the LMS algorithm is most commonly used in the design and implementation of integrated adaptive filters. Following are the basic steps for implementation of LMS:

1. Echo path output $d(n)$ is calculated as

$$d(n) = \sum_{i=0}^{N-1} h(n)x(n-i) = h^T(n)x(n)$$

2. Estimated Error is calculated as

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

3. The Weight vector update equation is given by

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

$w(n)$: Weight vector at time n; Where μ is known as step size lying between $0 \langle \mu \langle \frac{2}{\lambda_{max}}$

λ_{max} maximum eigenvalue of autocorrelation matrix.

4.2 NLMS ALGORITHM: In LMS algorithm the correction that is applied to $w(n)$ is directly proportional to input signal $x(n)$. So when $x(n)$ increases the LMS algorithm experiences the problem of gradient noise amplification. By normalizing LMS step size by $\|x(n)\|^2$ in NLMS algorithm helps reducing the noise amplification.

Every iteration of the NLMS algorithm gets accomplished in following 4 steps:

1. Echo path $d(n)$ is calculated as

IJIREEICE

ISSN (Online) 2321 – 2004
ISSN (Print) 2321 – 5526

International Journal of Innovative Research in
Electrical, Electronics, Instrumentation and Control Engineering

ISO 3297:2007 Certified

Vol. 4, Issue 9, September 2016

$$d(n) = \sum_{i=0}^{N-1} h(n)x(n-i) = h^T(n)x(n)$$

2. Error estimation is calculated using the equation

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

3. Step Size calculation

$$\mu(n) = \frac{\beta}{x(n)^T x(n)} = \frac{\beta}{\|x(n)\|^2}$$

4. The tap weight of the FIR vector are updated, by the equation below

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

β known as normalized step size with $0 \langle \beta \langle 2$. By replacing μ in the LMS weight vector update equation with $\mu(n)$ leads to NLMS algorithm.

4.3 RLS ALGORITHM: The other class of adaptive filtering techniques studied in this thesis is known as Recursive Least Squares (RLS) algorithms. In contrast to the LMS algorithm, the RLS algorithm uses information from all past input samples (and not only from current tap input samples). Following are the basic steps for implementation of RLS:

1.Filter Output

$$d(n) = h^T(n-1)x(n)$$

2.Intermediate gain vector

$$k(n) = \frac{P(n-1)x^*(n)}{\lambda + x^*(n)P(n-1)x^T(n)}$$

λ is forgetting factor having values less than or equal to 1

3.Error signal

$$e_{(n-1)}n = d(n) - y_{n-1}(n)$$

4.Tap weight update

$$w(n) = w(n-1) + k(n)e_{n-1}(n)$$

5.Inverse of weighted auto correlation matrix update

$$p(n) = \frac{1}{\lambda}[P(n-1) - k(n)x^T(n)P(n-1)]$$

V. SIMULATION AND RESULTS

The LMS, NLMS and RLS algorithm were simulated employing MATLAB software for acoustic echo cancellation application. The length of the acoustic echo response in a typical teleconferencing room is in the region of 100 to 400 ms and hence adaptive filters employing 1000 taps or 1024 taps or more are typically required in order to achieve adequate levels of echo cancellation so here the algorithms are tested for 1000 tap lengths. Figure 3 shows the i/p speech signal and Figure 4 represents the desired echo signal obtained from i/p signal.

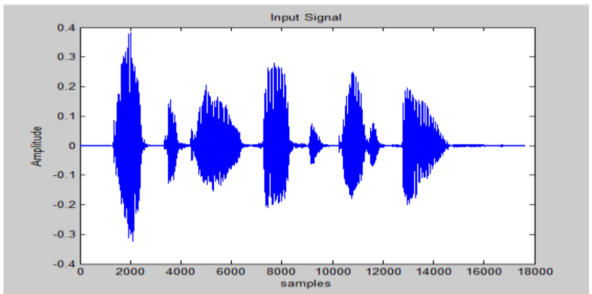


Figure 3: input Speech Signals

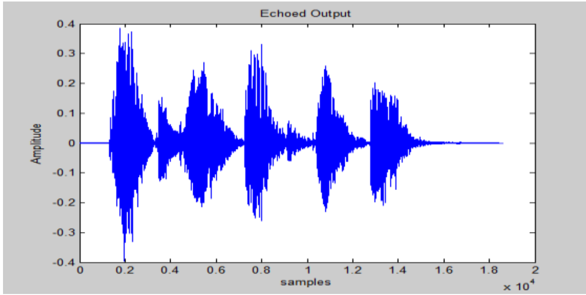
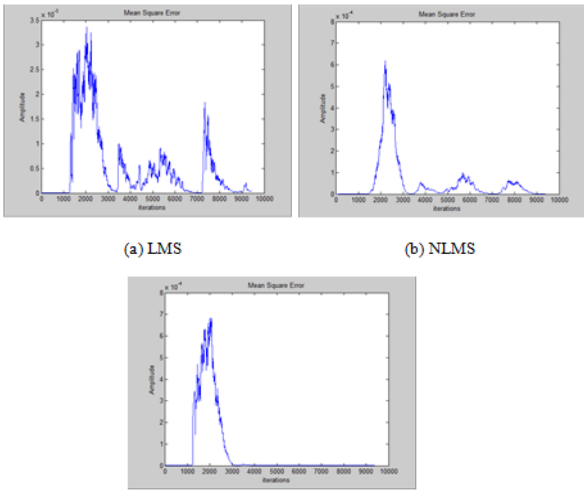


Figure 4: Echoed Speech Signals

Figure 5 shows the MSE comparison for algorithms. It can be concluded that as the algorithm progresses the average value of the cost function decreases, from Figure 5 (a), (b), (c) it can be seen MSE converges faster for NLMS as compared to LMS and RLS converges faster of all three algorithms.



(a) LMS (b) NLMS

(c) RLS

Figure 5: MSE Comparison

Different characteristics to identify effective echo cancellation are:

A. Echo Return Loss Enhancement (ERLE): It can also be defined as the measure of how much echo is suppressed in decibel (dB).

$$ERLE_{dB} = 10 \log_{10} \frac{E(d^2(n))}{E(e^2(n))}$$

IJIREEICE
ISSN (Online) 2321 – 2004
ISSN (Print) 2321 – 5526
International Journal of Innovative Research in
Electrical, Electronics, Instrumentation and Control Engineering
ISO 3297:2007 Certified
Vol. 4, Issue 9, September 2016

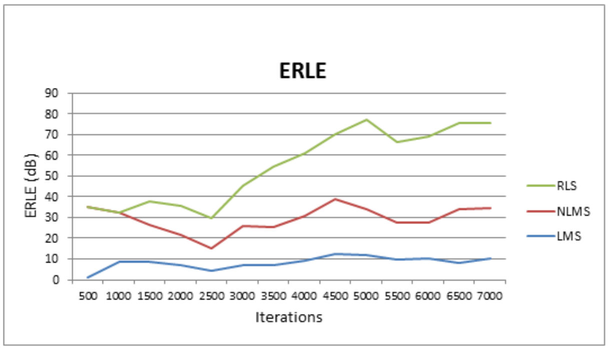


Figure 6: ERLE Comparison

From Figure 6 it can be seen that RLS has higher ERLE than LMS and NLMS and RLS has higher of all three algorithms.

B. Attenuation: It is reduction in signal strength represented in decibels (dB).

$$Attenuation = -10 \log_{10} \frac{(P_d(n))}{(P_e(n))}$$

Figure 7: Attenuation Comparison

From Figure 7 it can be seen that RLS has higher attenuation than LMS and NLMS

Table 1: Numerical analysis of Comparison parameters for LMS, NLMS and RLS

Filte-r	Filter Order	Specifi-cation	Attenua-tion (dB)	Conver gen-ce (Iteratio ns)	ERLE (dB)
Speech Signal: "Hedge apples may stain your hands green" (Male voice)					
LMS	1000	Step Size μ=0.06	-4.25	8518	9.69
NL MS	1000	β=1	-9.87	8494	21.85
RLS	1000	λ=0.99	-17.47	3848	28.32

From Table 1 it can be observed that all the characteristics shows better performance for RLS as compared to LMS and NLMS algorithm.

VI. CONCLUSION

One of the major problems in a telecommunication application over a telephone system is echo. The Echo cancellation algorithms were successfully implemented to find a software solution for the problem of echoes in the telecommunications environment. Considering the analysis of the plots for various parameters, average values of ERLE obtained is maximum for RLS algorithm whereas the estimation error and the mean square error are several orders smaller for RLS algorithm. The convergence rate of NLMS algorithm is greater than the LMS algorithm and the RLS algorithm has a far greater convergence rate compared to the LMS algorithm. Though the RLS algorithm gives much better results compared to other algorithms, still it is not used, as every iteration requires $4N^2$ multiplications and $3N^2$ additions. For echo cancellation systems the FIR filter order is usually in the thousands. Thus the number of multiplication required are very large because of which the RLS algorithm is too costly to implement.
From the results obtained it can be concluded that the NLMS algorithm, an equally simple, but more robust variant of the LMS algorithm, exhibits a better balance between simplicity and performance than the LMS algorithm. Due to its good properties the NLMS has been largely used in real-time applications.

REFERENCES

[1]. Hung Ngoc ,Nguyen Majid Dowlatnia, Azhar Sarfraz, " Implementation of the LMS and NLMS algorithms for Acoustic Echo Cancellation in teleconference system using MATLAB", December2009,Report 09087,Vaxjo University ISSN 1650-2647.
[2]. J. Gerardo Avalos, Juan C. Sanchez and Jose Velazquez, "Applications of Adaptive Filter", National Polytechnic Institute, Mexico.
[3]. Bhanudurga Sri Satya Simha Edamakanti, Vishnu Chaitanya Gundu, "Subband Adaptive Filtering Technique employing APA For Stereo Echo Cancellation over Audio Signals", Technical Report, Dept. of Applied Signal Processing, May 2014.
[4]. Monson H. Hayes, "Statistical Digital Signal Processing and Modeling", Wiley India Edition.
[5]. Yuksel Ozbay, Ahmet Resit Kavsaoglu, "An optimum algorithm for adaptive filtering on acoustic echo cancellation using TMS320C6713 DSP", Digital Signal Processing 20(2010)133-148, 2010.
[6]. Radhika Chinaboina, D. S. Ramkiran, Habibulla Khan, M.Usha, B. T. P. Madhav, K. Phani Srinivas, G. V. Ganesh, "Adaptive Algorithms For Acoustic Echo Cancellation", In Speech Processing, IJRRAS-7-1-05.pdf, Vol.7, Issue 1, April 2011.
[7]. Per Ahgren, "Acoustic Echo Cancellation and Doubletalk Detection Using Estimated Loudspeaker Impulse Responses", IEEE Transactions On Speech And Audio Processing, Vol. 13, No. 6, November 2005.
[8]. Ranbeer Tyagi,Dheeraj Agrawal, "Analysis the Results of Acoustic Echo Cancellation for Speech Processing using LMS Adaptive Filtering Algorithm", International Journal of Computer Applications (0975 8887), Volume 56, No.15,October 2012.
[9]. Abhishek Deb, Asutosh Kar, Mahesh Chandra, "A Technical Review on Adaptive Algorithms for Acoustic Echo Cancellation", International Conference on Communication and Signal Processing, India, April 3-5, 2014.
[10]. Lu Lu, "Implementation of Acoustic Echo Cancellation For PC Applications Using MATLAB", Master Thesis, IMIT/LECS/2007-63, 2007.