Adaptive Algorithms for Acoustic Echo Cancellation: A Review

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Abstract— *One of the major issues involved with communication is acoustic echo, which is actually a delayed version of sound reflected back to the source of sound hampering communication. Cancellation of these involve the use of acoustic echo cancellers involving adaptive filters governed by adaptive algorithms. This paper presents a review of some of the algorithms of acoustic echo cancellation covering their merits and demerits. Various algorithms like LMS, NLMS, FLMS, LLMS, RLS, AFA, LMF have been discussed.*

Keywords— *Adaptive Filter, Acoustic Echo, LMS, NLMS, FX-LMS, AAF, LLMS, RLS.*

I. INTRODUCTION

The prime need of the developing world is effective voice communication. In acoustic applications noise reflected from surrounding environment reduces quality of speech and audio signal and sometimes it becomes impossible to recover the original speech signal which was transmitted. To remove noise from the noise contained signal and to enhance the quality of signal, the technique which gain much attention is Acoustic Noise Cancellation (ANC) [1]. Filters are the basic component in telecommunication system and signal processing and find wide applications in areas of noise reduction, channel equalization and audio processing etc. [2] . An important block of ANC is Adaptive filter which provides reduction of noise without any prior knowledge of signal and noise. As conventional filters distort the quality of desired signal, Adaptive filters are used in situation where speech and noise signals are random in nature. Adaptive filter automatically adjusts its impulse response using an algorithm that responds to an error signal. Thus with the proper algorithm filter can operate with the changing conditions and readjusts it continuously to minimise the error signal. The basic concept was first introduced by Bernard widrow et al. [3].

Due to reflection from points where the characteristics of the medium through which the wave propagates changes, there occurs a repetition of waveform which is known as Echo. Echo is used in sonar and radar for exploration and detection purposes [4]. In telecommunication, echo can severely affect the quality and intelligibility of voice conversation in a telephone system. The perceived effect of an echo depends on its amplitude and time delay. In general, the noticeable echoes have an appreciable amplitude and a delay of more than 1 ms . Echo will give telephone call a sense of "liveliness" provided the round-trip delay is of the order of a few milliseconds. However, with the increasing amplitude and also round trip delay of more than 20 ms echoes become increasingly annoying and objectionable. Hence an important aspect of the design of modern telecommunication systems is echo cancellation [5]. Although there are other ways of cancelling echo such as echo cancellers, echo barriers, echo suppressors but adaptive filters are widely used due to its stability and wide scope of improvements. Adaptive filters are also use for low frequency noise [6].

An adaptive filter minimizes the function of difference between the desired output d(n) & actual output y(n) ,by altering its parameters regularly. This parameter alteration can be done with the help of various adaptive algorithms discussed in the following sections. The block diagram of the adaptive echo cancellation system is shown in figure 1. Here, w(n) represents the adaptive filter used to cancel the echo signal and h(n) represents the impulse response of the acoustic environment which generates the echo. The adaptive filter tries to equate its output y(n) to the desired output d(n) (the echo signal generated within the acoustic environment). After every iteration, the error signal i.e. e(n) = d(n)– y(n) is generated which is given as a feedback to the filter. Adaptive algorithm guides the change in filter parameters. When the adaptive filter output is equal to desired signal, the error signal ideally becomes zero and far user will not experience any kind of disturbance [7].

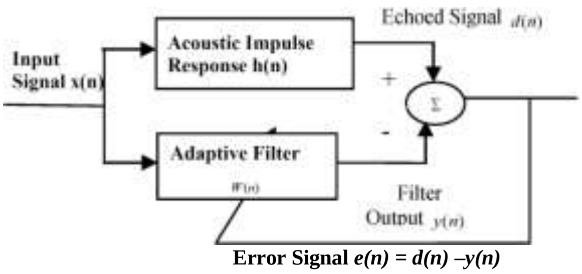

Fig. 1. Block diagram for AEC [7]

II. OVERVIEW OF ALGORITHMS

A. Least Mean Square (LMS) Algorithm

LMS is initially proposed by widrow Hoff in 1959. It is used to determine the minimum mean square error and is based on steepest descent method and gradient search technique. Mostly this algorithm is used because of ease of implementation, simplicity and low computational complexity. If x(n) is the input signal vector and w(n) is the weight vector of the adaptive filter then, output of adaptive filter y(n) is given by

$$y(n) = w(n)^T x(n) \qquad (1)$$

error signal e(n) is given by

$$e(n) = d(n) - y(n) \qquad (2)$$

weight vector update equation is given by

$$w(n+1) = w(n) + \mu e(n)x(n) \qquad (3)$$

where µ is the step size and controls the convergence rate. Small value of µ leads to more convergence time. Large value of µ cause the algorithm to diverge and degrades the performance of adaptive filter. Therefore selecting a step size is very important. One of the primary disadvantages of LMS algorithm is a fixed step size for every iteration. This requires an prior understanding of statistics of input signal which is rarely achievable. Also in LMS algorithm the correction that applied to w(n) is proportional to the input vector x(n). Therefore when x(n) is large, the LMS algorithm experiences a problem with noise gradient amplification [1].

B. Normalized Least Mean Square Algorithm

The problem of large value of µ in LMS was overcome by NLMS. NLMS was developed to overcome this problem. NLMS filter is more stable to unknown signal because step size is depending upon the input vector. This factor of NLMS filter makes an ideal adaptive filter for echo cancellation [8]. The NLMS algorithm can be summarized from following equations.

$$w(n+1) = w(n) + \frac{\mu}{||x(n)||^2} \cdot x(n)\, e(n) \qquad (4)$$

When we compare LMS and NLMS, NLMS has high convergence rate due to variable step size and has 3N+1 multiplication at each iterations which is acceptable because of stability and attenuation of echo.

C. FX-LMS Algorithm

The LMS algorithm did not perform well in the ANC framework because of the assumption made that the signal perceived as the error at the microphone is the output of the filter which is not the case in practice. The presence of the actuators, A/D converters, anti-aliasing filter and D/A converters in the path from the output of the filter to the signal received at the error microphone cause significant change in the signal . There is a need to incorporate the effect of this secondary path function in the algorithm as the change in signal demands. The equation of the FX-LMS algorithm is:

$$w(n+1) = w(n) + \mu(n)e(n)X'(n) \qquad (5)$$

It can be used for simple real-time realization as it has low computational complexity [7].

D. Average Adaptive Filter

LMS and NLMS is not good choice when convergence rate is at high priority. In order to obtain high convergence rate AAF is used. AAF belongs from stochastic gradient algorithm. Tap weight adaptation of AAF is shown in following equation.

$$w(n+1) = 1/n \sum_{m=1}^{N} w(n) + 1/n^{\gamma} \sum_{n=1}^{N} \gamma u(n) e*(n)$$

In terms of convergence rate AAF is the best algorithm. The main drawback is its lesser stability than LMS and NLMS [9].

E. Leaky Least Mean Square Algorithm

Leaky least mean square is form of standard least mean square which differentiates only because of cost function. When implemented in fixed point operation a leaky factor is added to standard LMS equation to avoid filter coefficients divergence. The time varying filter coefficients determined the dynamic range of filter output which is unknown earlier in adaptive filter. The coefficient overflow problem is avoided in LLMS. Following equation defines the LLMS tap weight adaptation [10].

$$w(n+1) = (1-\mu\gamma)w(n) + \mu x(n)e(n) \qquad (6)$$

Where γ is leakage factor where $0 < \gamma \leq 1$.

F. Recursive Least Square Algorithm

Recursive Least Square algorithm attempts to minimize the cost function. Each iteration of the RLS algorithm requires $4N^2$ multiplication operations and $3N^2$ additions which leads to greater computational complexity. It is also sensitive to computer round off error which leads to instability. It is numerically robust and not feasible for real time implementation [11]-[13]. The filter weight equation for RLS algorithm is given by:

$$\overline{W}(n) = \overline{W}(n-1) + k(n)\bar{e}_{n-1}(n)$$

$$\bar{e}_{n-1}(n) = [d(n) - \overline{W}^T(n-1)X(n)] \qquad (7)$$

III. COMPARISON

Table below gives the performance improvement in terms of SNR for LMS, NLMS, LLMS, AFA filters.

TABLE I
SIGNAL TO NOISE RATIO

Algorithms	Signal to Noise ratio (dB)
LMS	14.8857
NLMS	23.8809
LLMS	21.8768
AAF	24.8454

TABLE II
COMPARISON OF VARIOUS PARAMETERS

Algor-ithm	Computati-onal Complexity in (sec)	Speed of Conve-rgence	Stabi-lity	Robustness
LMS	3.3	Slow	Stable	Less
NLMS	3.6	Fast	More than LMS	Less
FX-LMS	2.9	Slower than LMS	Stable	Less
RLS	4.2	Fast	Unstable	More

Table 2 gives comparison of various algorithms in terms of computational complexity, speed of convergence, stability and robustness.

IV. CONCLUSION

Acoustic echo cancellation has its wide range of applications such as in mobile phone, speakerphones, hand free car fits, Bluetooth accessories, hearing heads and multi-channel teleconferencing systems due to advancement in technology. The main aim of adaptive algorithm is to lower the mean square error at the cost of higher convergence rate and lesser computational complexity. So by keeping these things in mind we conclude that FX-LMS is the best algorithm for acoustic echo cancellation because of its lower computational complexity that is 2.9 seconds and higher stability. We also conclude that each algorithm has its own pros and cons, so they should be applied according to the demand of the situation.

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REFERENCES

[1] Ms. Mugdha. M. Dewasthale and Dr. R. D. Kharadkar , *"Acoustic Noise Cancellation using Adaptive Filters: A Survey"*, in proceedings of *2014 International Conference on Electronic Systems*, Signal Processing and Computing Technologies, pp. 12-13 (DOI: 10.1109).

[2] Mr. Mayur B. Kachare , Prof. D. U. Adokar, *"A Distributed Arithmetic (DA) Based Digital FIR Filter Realization"* in proceedings of International Conference on Global Trends in Engineering, Technology and Management (ICGTETM-2016), pp.24.

[3] Bernard widrow, John Glover, *" Adaptive Noise Cancelling : Principles and Applications",* IEEE proceeding, Vol. 63 No 12,1975.

[4] Per Åhgren, *"Acoustic echo cancellation and doubletalk detection using estimated loudspeaker impulse responses"*, Speech and Audio Processing, *IEEE Transactions on* (Volume:13 , Issue: 6) Nov. 2005.

[5] Vladimir M. Matic and Srdan N. Abadzic, *"Acoustic and line echo cancellation using adaptive filters"*, 15th Telecommunications forum TELFOR. pp.322-325, November 2007.

[6] Farhang-Boroujeny, *"Adaptive Filters Theory and Applications"*, John Wiley and Sons, New York, 1999.

[7] Abhishek Deb, Asutosh Kar and Mahesh Chandra, *"A Technical Review on Adaptive Algorithms for Acoustic Echo Cancellation"*, in proceedings of International Conference on Communication and Signal Processing, April 3-5, 2014, India, pp. 041-043.

[8] P. Gil-Cacho, T.V. Waterschoot, M. Moonen and S.H. Jensen *"Multi-Microphone Acoustic Echo Cancellation using Multi-Channel Warped Linear Prediction of Common Acoustical Poles"* in Proc. 18th European Signal Processing Conference (EUSIPCO '10), Aalborg, Denmark, Aug. 2010, pp. 2121–2125.

[9] Sahar Mobeen, Iram Baig and Anum Rafique, *"Comparison Analysis Of Multi-Channel Echo Cancellation Using Adaptive Filters"*, in proceedings of 2015 International Conference on Open Source Systems and Technologies (ICOSST), pp.137-138.

[10] H. Buchner, *"Acoustic echo cancellation for multiple reproduction channels: From first principles to real-time solutions,"* Proc. ITG Conf. on Speech Communication, Aachen, Germany, Oct. 2008. Best Paper Award.

[11] J. Velazquez Lopez, Juan Carlos Sanchez and Hector Perez Meana, *"Adaptive Echo Canceller Using a Modified LMS Algorithm"*, 2nd international Conference on Electrical and Electronics Engineering, pp.93-96, September 2005.

[12] Xinyi Wang, Tingzhi Shen, Weijiang Wang, *"An Approach for Echo Cancellation System Based on Improved NLMS Algorithm"*, School of Information Science and Technology, Beijing Institute of Technology, Beijing, China, pp. 2853-2856, 2007.

[13] Constantin Paleologu, Jacob Benesty, and Silviu Ciochin, *"An Improved Proportionate NLMS Algorithm based on the l_0 Norm"*, ICASSP, pp.309-312, 2010.