# New Technique for Recursive Least Square Adaptive Algorithm for Acoustic Echo Cancellation of Speech signal in an auditorium

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**Abstract:** ***In today's technological society, human computer interactions are ever increasing. In many new systems, voice recognition platforms are implemented to give users more convenient ways of operating equipment and systems. To improve the audibility of the speech, the noise and acoustic echo must be removed from the speech signal. In this paper, we presented a new adaptive algorithm in the frequency domain for acoustic echo cancellation of speech signal in an auditorium. The RLS algorithm, the forgetting factor remains constant, which is utilized for the stability of the adaptive algorithm. However, the constant value of the forgetting factor will not support for the sensitive system. The value of the forgetting factor depends on the echo and reverberation. In an auditorium speech, the echo and reverberation signals are not in a stable manner since the constant value of forgetting factor is not a perfect solution for the removing the echo and reverberation. In order to solve this problem we presented average recursive least square adaptive algorithm, which produces the flexible forgetting factor in a min-max manner. The estimated echo values are constructed with the aid of combined feature of the min-max manner, which leads to increase the quality of the speech signal. Finally, our proposed algorithm is implemented using MATLAB and the experimental results showed that the proposed ARLS algorithm outperformed than the existing RLS algorithm.***

***Keywords:*** *frequency domain for acoustic echo cancellation, adaptive filter, recursive least square, average recursive least square, reverberation.*

## 1. INTRODUCTION

The acoustic echo, which is well-known as a "multipath echo", is formed by poor voice coupling between the earpiece and microphone in handsets and hands-free gadgets. Additional voice degradation is caused as voice-compressing and encoding/decoding devices process the voice paths within the handsets and in wireless networks. This gives returned echo signals with highly variable properties. At the point when compounded with inherent digital transmission delays, call quality is incredibly reduced for the wireline caller. Acoustic coupling is because of the reflection of the loudspeaker's sound waves from walls, door, ceiling, windows and other different objects back to the microphone. The aftereffect of the reflections is the formation of a multipath echo and multiple harmonics of echoes, which are transmitted back to the far-end and are heard by the talker as an echo unless wiped out. Adaptive cancellation of such acoustic echoes has turned out to be critical in hands-free communication systems such as teleconference or video conference systems [1-11].

Echo signal is the delayed type of original speaker signal. That implies, echo signal can be expected as a noise in speaker signal. The eliminating of noise from the speaker signal cannot be executed by classical filters, which suppress certain frequency parts and pass the others. This is the reason that, filter design used to eliminate echo is the subject of optimal filter design. The essential reason for the optimal filter design is to minimize the dissimilarity between desired response and actual response of the filter. Filter response does not just rely on the statistical information; because physical signal's statistical information has usually a changing nature. Consequently, a filter structure, which adjusted its response, according to the change of the error signal, is essential to adapt filter coefficients in a manner to minimize error signal [8]. Adaptive filter is the answer to this issue. Adaptive filter is a filter with coefficients, which are adjusted periodically keeping in mind the end goal to attempt meeting some performance criterion, which is normally in the form of some error or cost function minimization [9, 11]. An adaptive filter is a digital filter that can alter its coefficients to give the best match to a given desired signal. At the point when an adaptive filter works in a changeable environment, the filter coefficients can adapt in response to changes in the applied input signals. The main task of the adaptive filter is to estimate the characteristics of the echo path, creating the echo and compensate for it. To do this the echo path is viewed as an unknown system with some impulse response and the adaptive filter must mimic this response. Adaptive filters have been utilized as a part of different parts of signal processing in recent years. Among the possible applications is the Acoustic Echo Cancellation [11,12].

Adaptive Filters are usually actualized in the time domain, which functions admirably in many scenarios on the other hand; in numerous applications, the impulse response turns out to be too long, increasing the complexity of the filter beyond a level where it can no longer be implemented efficiently in the time domain. Then again, there exists an alternate solution and that is to actualize the filters in the frequency domain. The Discrete Fourier Transform or more precisely the Fast Fourier Transform (FFT) permits the conversion of signals from the time domain to the frequency domain in an efficient manner [12,13].

## 2. RELATED WORKS:

Yüksel Özbay *et al.* [11] have presented an algorithm for the determination of optimal adaptation rate (μ) for the least-mean-square (LMS) adaptation algorithm that has been utilized in the adaptive filter. The efficiency of their optimal μ value determination algorithm has been demonstrated on a single direction voice conference application with one speaker. A DSP card (TMS320C6713), a Laptop computer, an amplifier, a loudspeaker and two microphones in the two applications has been utilized. In the first application, two microphones had placed close to the loudspeaker, while in the other application, one microphone had placed close to loudspeaker and speech trial had been implemented in the far-end microphone. Output of the adaptive filter has been observed for μ values of 0, 0.1, 100 and optimal (a value between 0.01 and 100). The best outcomes in the adaptive filter had been achieved from optimal μ value.

Sarmad Malik and Gerald Enzner [14] have discussed about the adaptive acoustic echo cancellation in the vicinity of an unknown memory less nonlinearity preceding the echo path. Through absorbed the coefficients of the nonlinear expansion into the unknown echo path, the cascade observation model had been altered into an equal multichannel structure, which further increased with a multichannel first-order Markov model. For the subsequent multichannel state-space model, a recursive Bayesian estimator that takes the form of an adaptive Kalman algorithm in the discrete Fourier transform (DFT) domain, has been derived. The paper has also demonstrated that such a recursive estimator acknowledged by means of a stable and structurally proficient multichannel state-space frequency-domain adaptive filter. The paper has additionally shown the proposed algorithm, which comes from a contained structure, gave successful nonlinear echo cancellation in the vicinity of continuous double-talk, fluctuating degree of nonlinear distortion, and changes in the echo path.

Luis A *et al.* [15] have introduced a new method for nonlinear acoustic echo cancellation based on adaptive Volterra Filters with linear and quadratic kernels, that mechanically choosed those diagonals contributing most to the output of the quadratic kernel with the objective of minimizing the overall mean-square error. In the echo cancellation scenarios, not all coefficients were similarly relevant for the modeling of the nonlinear echo, but coefficients close to the main diagonal of the second-order kernel depict the majority of the nonlinear echo distortions, such that not all diagonals need to be executed. Then again, that was hard to choose the most suitable number of diagonals apriori, since there have numerous elements that effect the decision, for example, the energy of the nonlinear echo, the shape of the room impulse response, or the step size utilized for the adjustment of kernel coefficients. The proposed method includes adaptive scaling components that control the impact of every group of adjacent diagonals contributing to the quadratic kernel output. Zoran M. Šari´c *et al.* [16] have proposed a computationally proficient form of the partitioned block frequency domain adaptive filter with many iterations on current data block. The algorithm executed as a cascade of two adaptive filters. The first filter minimized the Least Square (LS) criteria leading to unbiased estimate of a room response. The second filter accelerates the convergence rate utilizing many iterations to minimize adjusted LS criterion. Coefficients upgrades computed in a single step substitute for several iterations and cut computational costs. The difficulty of the algorithm is o(log2(R)), where R had a number of iterations. The proposed algorithm has been tested in a simulated room and a real reverberant room. Luis A. Azpicueta-Ruiz *et al.* [17] have presented an AEC based on combination of filters in discrete Fourier transform domain. Considering that both the input signal and the cancellation scenario make the performance of adaptive filters was frequency dependent, the proposed method have exploited the combination capabilities employing different mixing parameters to separately combine

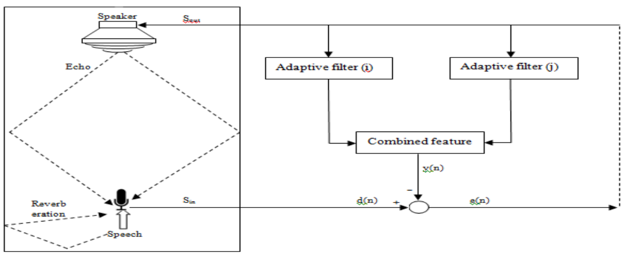



Figure 3.1: Block Diagram for acoustic echo canceller

independent spectral regions of two frequency-domain adaptive filters with different step sizes. Thusly, the proposed method outclassed recent algorithms where only a single combining parameter mixes the overall outputs of two frequency-domain adaptive filters. These advantages were shown by means of realistic experiments.

## 3. PROPOSED METHODOLOGY:

### 3.1 Acoustic echo cancelation in auditorium

An echo is a reflection of sound, arriving at the listener sometime after the direct sound. Echo is the reflected replica of the voice heard eventually later and deferred version of the original. Echo cancellation is the procedure that eliminates unwanted echoes from the original signal. It incorporates first recognizing the originally transmitted signal that re-shows up, with some deferral, in the speech signal. When the echo is accepted, it can be removed by 'subtracting' it from the speech signal. Numerous reflections in acoustic enclosures and transmission delay affect the sound quality, which on account of a teleconferencing system lead to a poor understanding of the conversation.

### 3.2 Acoustic problems in auditorium

The assembly room, as a spot for listening created from the classical open-air theaters. The outline of different sorts of auditoriums has turn into a mind boggling issue, because in addition to its different, sometimes conflicting, aesthetics, functional, technical, artistic and economical requirements, an auditorium regularly needs to suit a remarkably large audience. In a few ways, even the largest hall is same as the smaller rooms, the essential acoustic criteria are the same. On the other hand, the primary defects of the auditorium conferencing are reverberation and echo. Keeping in mind the end goal to take care of this issue, in this paper, we proposed effective acoustic echo cancellation methodology for an auditorium.

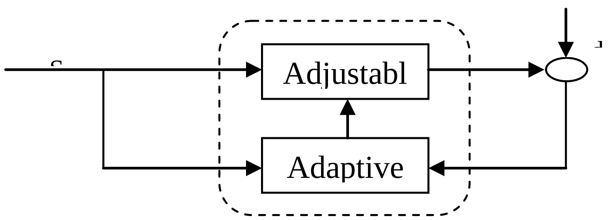
The figure 3.1 is represented the overall block diagram of the proposed acoustic echo canceller. In this figure 1, the acoustic echo cancellation in an auditorium is illustrated. The speech signal with the reverberation of voice and auditorium noise is collected by the microphone and collected speech signal is passed to the speaker. The problem in this audio setup is that the passed voice signal is played

through loudspeaker and its reflections of the room boundaries will also collected by the microphones and passed to the speaker. This makes listener hear the repeated voice with delayed reflections of the auditorium walls. The presence of acoustic echo in the auditorium makes the listeners feel that they are being interrupted with the repeated voice, forcing them to stop speaking until the echo faded away and the process is repeated over and over again. This acoustic echo and reverberation degrades the quality of the communication considerably.

### 3.3 Adaptive filters for acoustic echo cancellers in frequency domain adaptive filter

The fundamental function of the AEC is to estimate the acoustic transfer function from the speakers to the microphone including the reflections way. Filtering the incoming voice signal through the evaluated acoustic transmission function delivers an estimate of the echo signal y(n). Subtracting this evaluated echo from the microphone signal results in the echo free signal e(n)= d(n)-y(n) which is send to speaker rather than the microphone signal d(n). Acoustic echo cancellers typically utilize adaptive finite impulse response filters to assess the acoustic echo path. The FIR coefficients are adjusted utilizing an adaptive algorithm to minimize the error signal. The figure 3.2 represents the block diagram of the adaptive filter. The adaptive filter is indicated in the dotted box of the figure 2, which contains the two portions specifically filter part and update part. The function of the filter part is to compute the convolution of the input signal $S_{out}$ and the filter coefficients resulting in the filter output y (n). The set of filter coefficients are constantly adjusted by the update part. The update part is additionally called as adaptive algorithm, which is responsible for updating the filter coefficients so that the filter output y (n) turns out to be as close as possible to the desired signal d (n). In most cases update part changes the filter coefficients in small steps to minimize a certain function of the error signal e(n). The error signal e(n) represents the difference between the desired signal d(n) and the filter output i.e. e(n) = d(n)-y(n).

Figure3.2: Block Diagram of Average RLS Filter




Frequency-domain adaptive filtering is an attractive solution to deal with this difficult problem. There are two principal advantages to frequency- domain implementations of adaptive filters. First the amount of computation can be greatly reduced by repla- cing time-domain convolution and/or correlation by fast transform domain block-convolution and/or block cor- relation based either on the fast Fourier transform (wr). The second advantage comes from the decorrelating property of the discrete Fourier transform and the possibility of using different step sizes for each transform domain adaptive weight, which results in a quasi-optimal convergence rate, even in the presence of large variations in the input power spectrum (a situation where time-domain LMS-type algorithms perform very poorly).

In frequency domain adapative filter, both filtering and coefficient update can be performed sample-per-sample or n blocks of sample. A block of L sample are collected in a buffer and the adaptive filter function is called to process the whole buffer resluting in L output samples and updating all the filter coefficient every buffer-full samples. In block processing case,it is possible to perform the filtering and coefficient update functions entirely in frequency-domain. This is achieved by first applying the fourier transformation on the data buffer and performing the filtering and update by complex element ise multipplication in the frequency domain. The result is then converted back to time domain using the inverse fourier transform. This procedure results in a very efficient implementation of large adaptive filters, suchas those commonly used in acoustic echo canccellers.

## 4. PROPOSED AVERAGE RECURSIVE LEAST SQUARES METHOD

### 4.1 Recursive least squares adaptive filter

The Recursive least squares (RLS) is an adaptive filter which recursively finds the coefficients that minimize a weighted linear least squares cost function relating to the input signals. This is rather than different calculations, for example, the least mean squares (LMS) that goal is to decrease the mean square error. In the derivation of the RLS, the input signals are considered deterministic, while for the LMS and similar algorithm they are viewed as stochastic. Contrasted with most of its competitors, the RLS exhibits extremely fast convergence. As specified the previously the memory of the RLS algorithm is restricted to a limited number of values, relating to the order of the filter tap weight vector. Firstly, two factors of the RLS implementation ought to be noted: the first is that in spite of the fact that matrix inversion is crucial to the derivation of the RLS algorithm, no matrix inversion calculations are needed for the execution, hence significantly reducing the amount of computational complexity of the algorithm. Secondly, unlike the LMS based algorithms, current variables are updated within the iteration they are to be utilized, utilizing values from the previous iteration. To implement the RLS algorithm, the following steps are executed in the following order.

1. The filter output is calculated using the filter tap weights from the previous iteration and the current input vector.
$$\bar{y}_{n-1}(n) = \bar{w}^{t}(n-1)x(n) \ldots \qquad (1)$$
2. The intermediate gain vector is calculated using eq. (2).
$$u(n) = \bar{\psi}_{\lambda}^{-1}(n-1)X(n)$$
$$k(n) = \frac{1}{\lambda + X^{T}(n)u(n)}u(n) \ldots \qquad (2)$$
3. The estimation error value is calculated using eq. (3).
$$\bar{e}_{n-1}(n) = d(n) - \bar{y}_{n-1}(n) \ldots \qquad (3)$$
4. The filter tap weight vector is updated using eq. (4) and the gain vector is calculated in eq. (2).
$$w(n) = \bar{w}^{T}(n-1) + k(n)\bar{e}_{n-1}(n) \ldots \qquad (4)$$
5. The inverse matrix is calculated using eq. (5).
$$\psi_{\lambda}^{-1}(n) = \lambda^{-1}(\psi_{\lambda}^{-1}(n-1) - k(n)[X^{T}(n)\psi_{\lambda}^{-1}(n-1)]) \qquad (5)$$

From the above description of the RLS algorithm, where the forgetting factor remains constant value which is laid between zero and one. Selecting the value of the forgetting factor is a based on the following condition. The smaller value of the forgetting factor is, the smaller contribution of previous samples. This makes the filter more sensitive to recent samples, which means more fluctuations in the filter co-efficients. The case is referred to as the growing window RLS algorithm. In practice, is usually chosen between 0.98 and 1. The constant value of the forgetting factor is used for stability of the adaptive algorithm. However, the constant value of the forgetting factor will not support for the sensitive system. The value of the forgetting factor is depends on the echo and reverberation. In an auditorium speech, the echo and reverberation signals are not in a stable manner since the constant value of forgetting factor is not suitable for this application. This problem motivated us to design a RLS algorithm with flexible forgetting factor.

### 4.2 Average RLS estimation

In our proposed methodology, we introduce the average recursive least square adaptive algorithm in the frequency domain for effective acoustic echo cancellation. In a standard RLS algorithm, the value of forgetting factor placed remains constant. In the case of error of the signal is larger sensitivity of the adaptive algorithm needs to be increase. The sensitivity of the RLS algorithm depends on the forgetting factor. By decreasing the value of the forgetting factor, the sensitivity of the RLS adaptive algorithm is increased. In our research, we discuss the problem acoustic echo cancellation in an auditorium. In the auditorium, the speech signal is affected by the both echo and reverberation signal since the error value become larger. In order to remove the larger error in this paper, we designed novel average recursive lease square (ARLS) adaptive algorithm.

From the above figure 1, represents the proposed method of acoustic echo cancellation, from which the adaptive filer (i) and adaptive filter (j) are presented where 'i' and 'j' has the minimum and maximum value of forgetting factor values respectively. Here, we used the average recursive least square as adaptive filter. According to that, the adaptive filters process the input signal mixed with the echo and reverberation and produces the estimated echo. Our proposed algorithm selects the combined estimated echo of the both adaptive filer (i) and adaptive filter (j) which, leads to reduce the error value.

Let's consider two system in parallel, based on forgetting factor

$$\psi_{\lambda_1}^{-1}(n) = \lambda_1^{-1}(\psi_{\lambda_1}^{-1}(n-1) - k(n)[X^T(n)\psi_{\lambda_1}^{-1}(n-1)]) \qquad (6)$$

$$\psi_{\lambda_2}^{-1}(n) = \lambda_2^{-1}(\psi_{\lambda_2}^{-1}(n-1) - k(n)[X^T(n)\psi_{\lambda_2}^{-1}(n-1)]) \qquad (7)$$

Calculate Average filter value for RLS filter,

$$\psi_{\lambda Avg}^{-1}(n) = Avg\,(\psi_{\lambda_1}^{-1}(n), \psi_{\lambda_2}^{-1}(n)); \qquad (8)$$

### 5. SIMULATION AND RESULTS

This paper presents a details sketch of an Acoustic Echo canceller, (AEC). The software simulation and the results of simulation of the ARLS-AEC algorithm, which was performed in MATLAB, are discussed. The proposed Average recursive least square adaptive algorithm in frequency domain is implemented in MATLAB Version 8.1.0.604 (R2013a). The system on which the technique was simulated was having 4 GB RAM with 64 bit operating systems having i5 Processor. For assessment of the proposed method, randomly generated signals has been used.

In order to evaluate the quality of the echo cancellation algorithm the measure of ERLE was used. ERLE, measured in dB is defined as the ratio of the instantaneous power of the signal, d(n), and the instantaneous power of the residual error signal, e(n), immediately after cancellation. ERLE measures the amount of loss introduced by the adaptive filter alone. Mathematically it can be stated as

ERLE = 10log Pd (n) /Pe (n) = 10log E[d (n)]^2/ E[e (n)]^2

For a good echo canceller circuit, an ERLE in the range of 30 dB – 40dB is considered to be ideal. The Table 1 shows comparison between existing and proposed methods ERLE values in the range 30-40 dB

| ERLE | | | |
|---|---|---|---|
| Signals | Existing method with forgetting factor 0.98 | Existing method with forgetting factor 0.90 | Proposed method with average forgetting factor |
| 1 | 0.5638 | 13.0075 | 0.0884 |
| 2 | 0.5612 | 12.7446 | 0.0861 |

Table: 5.1 ERLE comparison

***a.Comparison between original signal and estimated signals in RLS Algorithm in max forgetting factor***

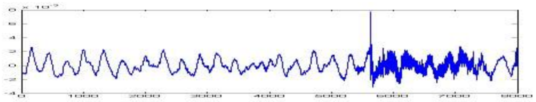


Figure 5.1: Input near end and far end signal for echo cancelation Fs = 8000

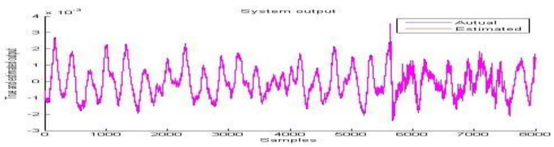



Figure 5.2: Actual and Estimated output for input signal

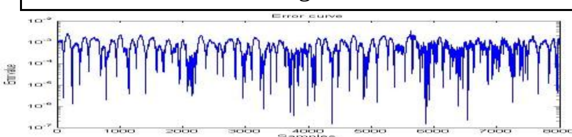



Figure 5.3: Estimated error

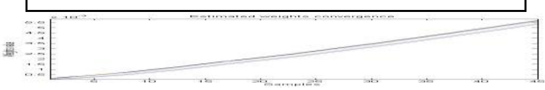



Figure 5.4: Weightage Curve for existing system for forgetting factor 0.98

**Comparisons graphs from figure 5.1-5.4:**The Estimated curves are obtained by varying forgetting factor lamda. The Echo and reverberation both are maintain in stable condition. The comparative curves have plotted between actual and estimated signal. Figure 5.2 gives estimated curves for stable echo and reverberation in standard time limit. Figure 5.3 gives estimated error in employed auditorium for sample N = 8000. From analysing the results, we can infer that all the cases gave good results. Among the Weightage curves, the distance between the curves are high in stable situation it represents this forgetting factor lamda gives only dilute weightage values.

***a. Comparison between original signal and estimated signals in RLS Algorithm in min forgetting factor***

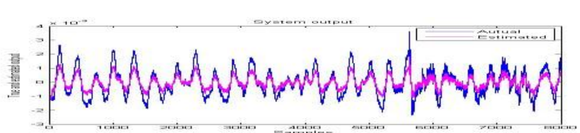



Figure 5.5: Actual and Estimated output for input

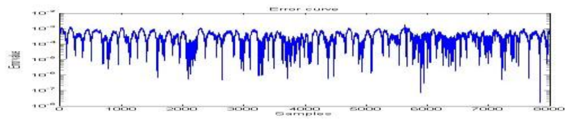



Figure 5.6: Estimated error

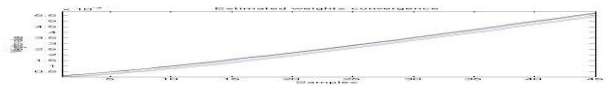
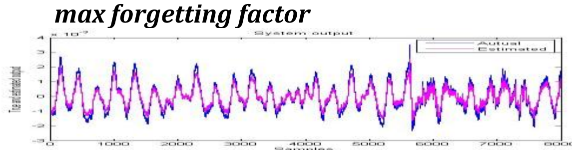



Figure 5.7: Weightage Curve for existing system for forgetting factor 0.90

**Comparisons graphs from figure 5.5- 5.7:** The Estimated curves are obtained by varying forgetting factor lamda into minimum (0.90). The Echo and reverberation both are maintain in stable condition. The comparative curves have plotted between actual and estimated signal. Figure 5.6 gives estimated curves for stable echo and reverberation in standard time limit and minimum lamda = 0.90. Figure 5.7 gives estimated error in employed auditorium for sample N = 8000 and lamda = 0.90. From analysing the results, we can infer that all the cases gave worst results. Among the Weightage curves, the distance between the curves are high in stable situation it represents this forgetting factor lamda gives only dilute weightage values. Moreover the estimated and actual signals are not matched in any condition.

***b. Comparison between RLS Algorithm in min-max forgetting factor***



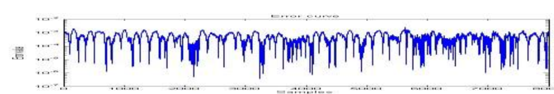




Figure 5.9: Estimated error

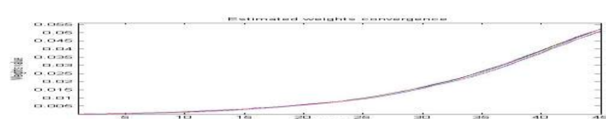



Figure 5.10: Weightage Curve for proposed system forgetting factor as a average

## 6. CONCLUSION

A new algorithm was proposed for an acoustic echo canceller with average RLS algorithm. Its performance was studied in comparison with conventional algorithms in a simulation. Good performance was confirmed with the proposed algorithm. Furthermore, a parallel echo cancelling architecture suitable for hardware implementation by frequency domain transfer processing. Near end signal, Far end signal echo and reverberation in auditorium was gradually optimized using average RLS filters by changing forgetting factor. The proposed system is stable, when echo and reverberation is high. Finally, the relationship between the echo cancellation algorithm the measure ERLE and signal and Weightage of error signal characteristics was clarified.

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