Performance Evaluation of Adaptive Filters Structures for Acoustic Echo Cancellation

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Abstract

We have designed and simulated an acoustic echo cancellation system for conferencing. This system is based upon a least-mean-square (LMS) adaptive algorithm and uses multi filter technique. A comparative study of both structure has been carried out and it is found that this new multi-filter converge faster than similar single long adaptive filter.

Keywords: LMS,Multiple Sub Filter ,Echo Cancellation

1. INTRODUCTION

Acoustic echo cancellation (AEC) [1] is used in teleconferencing and its purpose is to provide high quality full-duplex communication. The main part of an AEC is an adaptive filter which estimates the impulse response of the loudspeaker-enclosure-microphone (LEM)[2] system. There are various adaptive algorithms for the AEC filter update, these are the least mean square, normalized least mean square (LMS, NLMS), affine projection (AP) and recursive least squares (RLS) algorithms. As the echo cancellation environment is not stationary therefore echo reduction in rooms with long reverberation time is necessary. Hence, the signal processing methods are in demand in industry. Several partial update methods for computational complexity reduction of various adaptive filtering algorithms have been proposed and analyzed, e.g. [25]-[31] for the LMS/NLMS/AP .The technique used in earlier stages was echo suppression [23][24].Due to some disadvantages of echo suppression echo cancellation came into picture and the process of Acoustic echo cancellation [24][27], [29] is achieved with the help of adaptive filter which models the LEM system. The purpose of an acoustic echo-canceller is to reduce the amount of sound which a far-end teleconference transmits from returning to them. Traditional approaches to acoustic echo cancellation have used filtering algorithms which try to estimate the impulse response of the acoustic path and filter the incoming signal from the far-end [23],[24]. In this paper, we are proposing a multi sub filter approach for echo reduction and comparing their results. This paper is organized in four sections. Section two describes the simulation model of AEC in mat lab using multifactor approach. Further, section three discusses the results. In the end section four concludes the paper.

2. MULTIFILTER APPROACH FOR AEC

First of all before considering the other issues of this multiple sub filter approach the first most desired thing is the modeling of acoustic path through which communication takes place. As shown in figure1 when signal travel it experience the echo's of the signal and some noise also added in the signal. The multiple sub filters (MSF) structure is constituted by using the time delays and filter orders estimated as shown in fig.2.

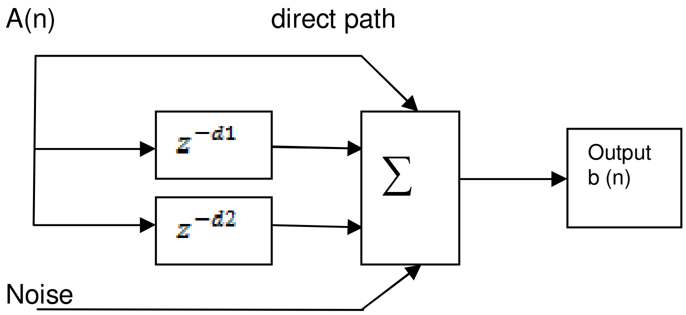

FIGURE 1: Modeling of Acoustic Path

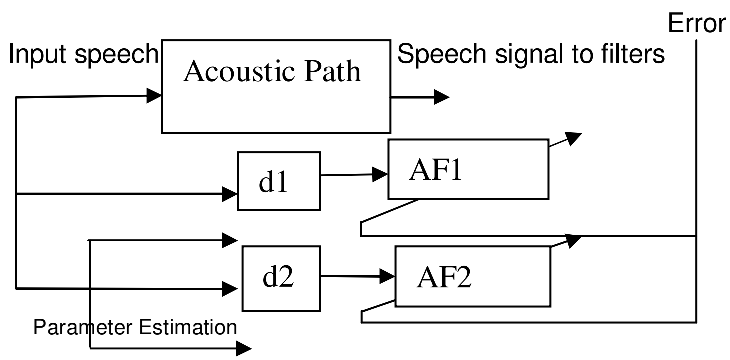

FIGURE 2: Model of Multiple Subfilter

Following equations described the behavior of multiple sub filter model used. $e(n) = b(n) - \sum_{k=0}^{M} W_k^T(n)\hat{X}_k(n);$ Adaptation of each sub filter is given as $W_k(n+1) = W_k(n) + \mu_k \hat{X}_k(n)e(n); \quad k = 0,1,2,...M$ The output of the multiple sub filter is given by $b(n) = \sum_{k=0}^{M} W_k^T(n)\hat{X}_k(n); \quad n = 0,1,2,...., N-1$ Each individual filter is updated with the help of a adaptive algorithm as shown in fig.2 and adaptation step size is separately chosen for each sub filter. the step size of the multiple sub filter structure[33][34] is same as for single long adaptive filter .Let us define the weight error vector Vj(n)=Wj(n)–Hj.The MSE is ζ(n)=E(e^2(n)],

$\zeta(n) = \sigma_\eta^2 + \sum_{\substack{j=0 \\ j\neq l}}^{M}\sum_{k=0}^{M} E[V_j(n)]G_{kj}E[Vj\ (n)] + \sum_{j=0}^{M}\tau\{G_{j,j}E[Vj\ (n)V_j^T(n)]\}$ here $G_{j,j} = E\left[X_j(n)X_j^T(n)\right]$.For the MSE to converge, it is necessary that E(Vj(n)] and $\tau\left\{G_{j,j}E\left[V_j(n)V_j^T(n)\right]\right\}$coverage ∀ j = 0,1,2,…, M . Taking expectation both sides we have $E(V_j(n+1)] = (I - \mu_j G_{j,j})E[V_j(n)] - \mu_j \sum_{\substack{k=0 \\ k\neq i}}^{M} G_{jk}E(V_k(n)]$ The mean of the j^{th} sub filter weight error vector converse, i.e. E(Vj(∞)]=0 If 0 < μ_j ≤ $\frac{2}{(\lambda_{\max})_j} \forall\ j = 0,1,2,...,\ M$ this is adaptation step size expression[33][34]. Stability is improved if the step size is chosen such that

$0<\mu_j \leq \frac{2}{3\tau(G_{j,j})_j} \forall \; j = 0,1,2,..., M$

The next work is to do simulations of conventional approach and multiple filter approach. For simulations comparisons we are taking far end and near end speech signals. Two basic performance criteria mean square error (MSE) and ERLE are the main comparison parameters for these approaches .The comparison results will show the path that which approach is better for AEC. Simulations is carried taking into account the time delay ,gain and stepsize.we are using step size 0.5,delay 351 and 254,gain .85 and .9.The order of low pass filter is 10.Length of adaptive filter chosen is 194 and leakage factor is 1.

3.SIMULATION RESULTS OF MULTIPLE SUB FILTER APPROACH AND SINGLE FILTER

Following are the results of two approaches compared. We are using lms and its different variants for updating the filter coefficients.Fig.3is representing the far end & near end speech signals, fig. 4& 5 are MSE results of multiple sub filter's and conventional approach. Fig. 6, 7,8are results of conventional approach of single long adaptive filter method for acoustic echo cancellation.fig.9, 10,11are ERLE graphs of MSF for LMS and NLMS.

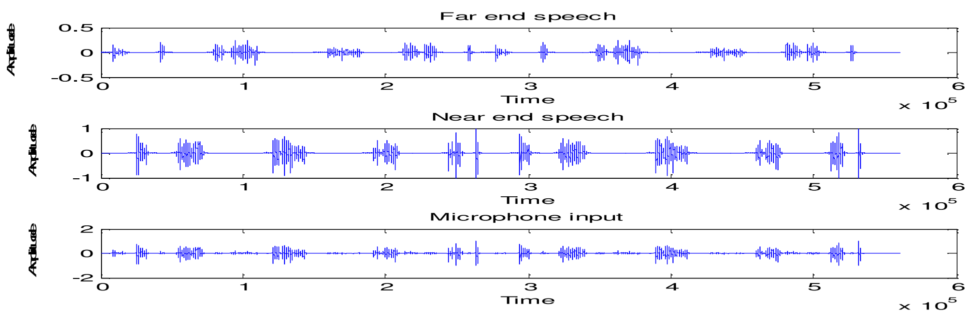

FIGURE 3: Far end and Near end speech signal

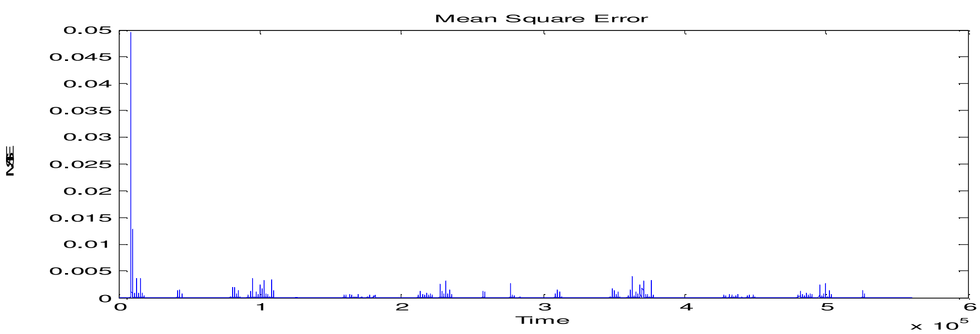

FIGURE 4: MSE of multiple filter approach (LMS)

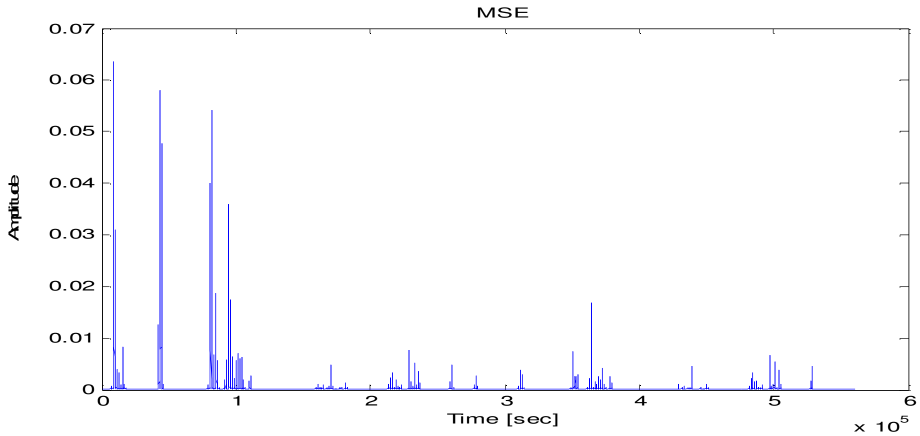

FIGURE 5: MSE of single filter approac(LMS)

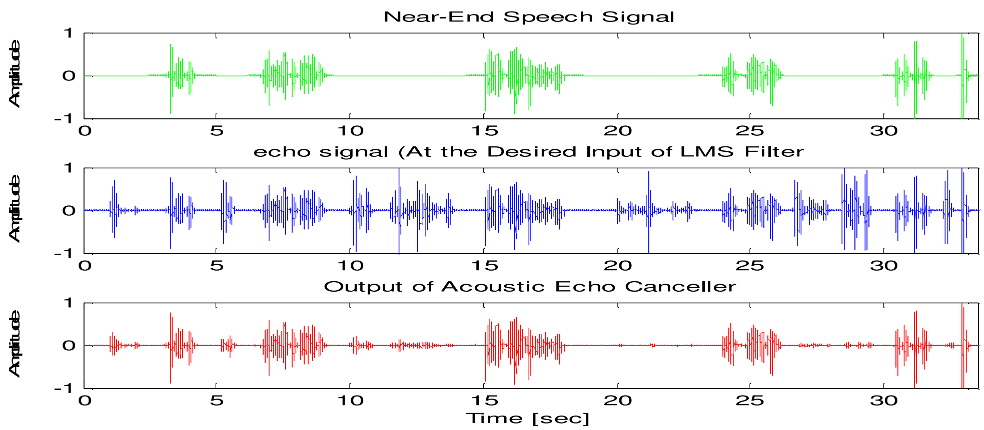

FIGURE 6: Single Adaptive Filter approach output

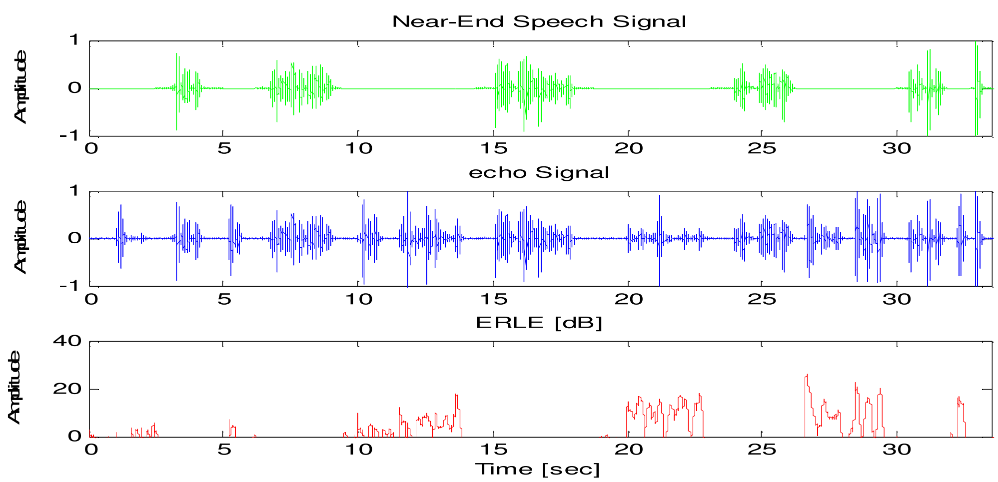

FIGURE 7: ERLE of Single long Adaptive Filter

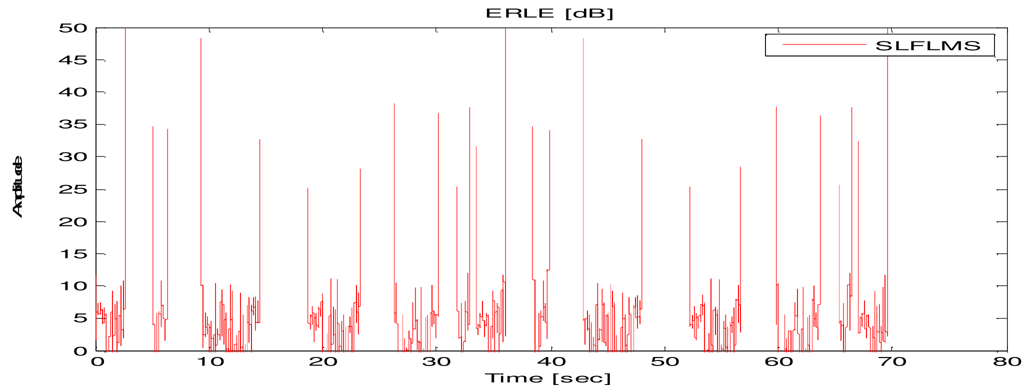

FIGURE 8: ERLE of single long Adaptive Filter(LMS)

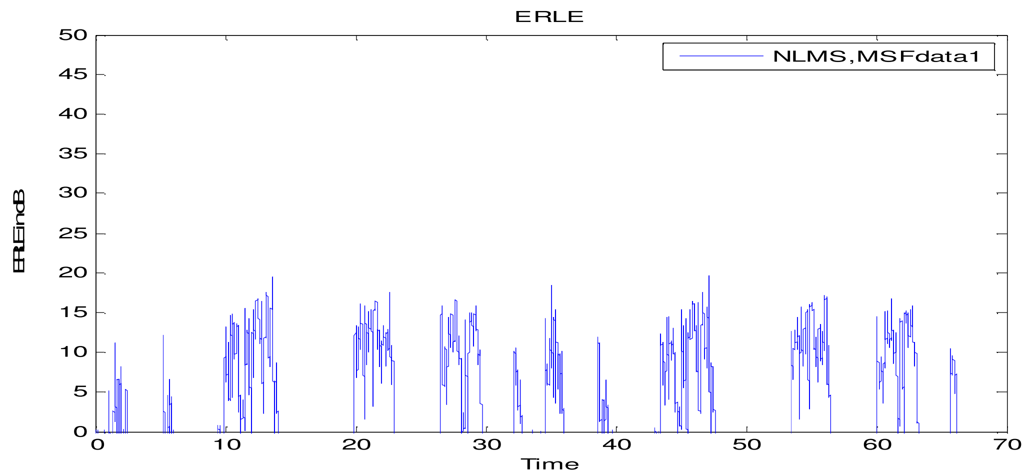

FIGURE 9: ERLE for MSF using NLMS

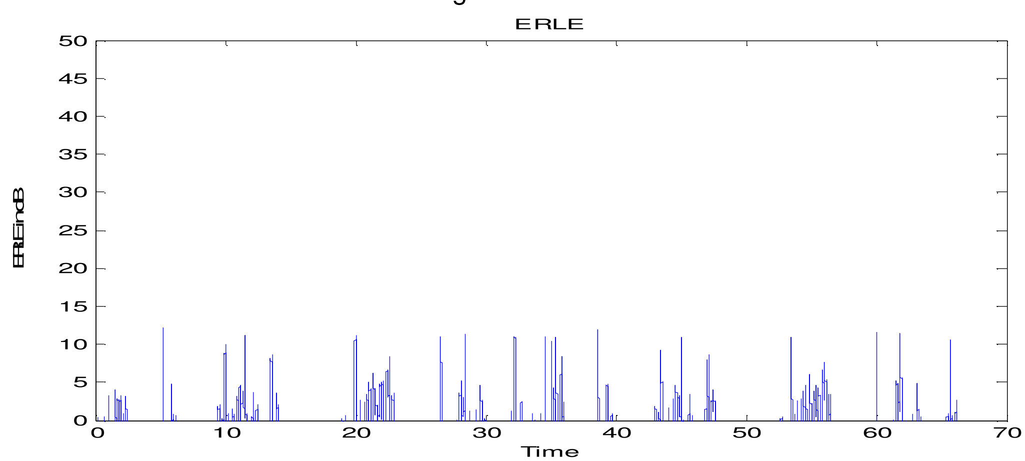

FIGURE 10: ERLE for MSF using SELMS

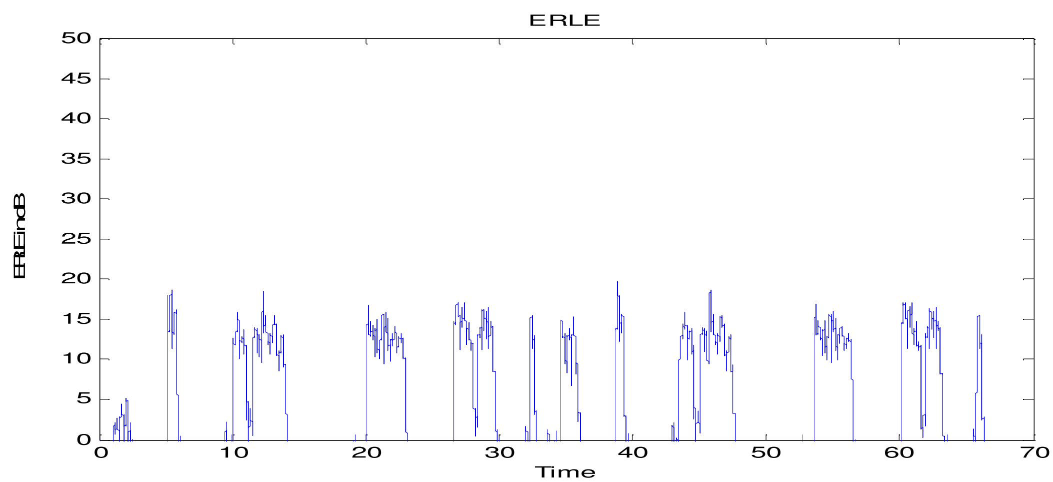

FIGURE 11: ERLE for MSF using LMS

4.CONCLUSION

As seen from different figures we are comparing the performance behaviour of long and multiple sub filters.Fig.3 indicating the Far end , Near end speech signal and input to microphone signals in the case of multiple filter and conventional approach.Fig.4and5 representing the MSE of two different approaches ,MSE in case of multi subfilter approach is less as compared to conventional approach.ERLE(Fig.7,8) in case of conventional approach is more than 40 db where as in case of multiple sub filter(Fig.9,10,11) it is near to 15-20 db. Thus MSE results shows that single long adaptive filter shows poor performance as compared to multiple subfilter structure.Thus our results achieved confirms the idea of using multiple adaptive filters.

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