A Novel Hybrid Technique for Acoustic Echo Cancellation and Noise reduction Using LMS Filter and ANFIS Based Nonlinear Filter

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Abstract - *The design of an efficient and robust echo and noise canceller is now very much required by the growth of mobile radio and teleconference communications. The use of Frequency-Domain Adaptive Filters in the context of acoustic echo cancellation has been extensively studied in the literature. So many adaptive filters have been introduced time to time by the researchers all over the word to address the problems related to speech signal transmission and reproduction. Moreover, usually the techniques available in the literatures are stick to either echo cancelation or noise reduction. However some literatures are available but small in ratio for addressing both the problems echo cancelation and noise removal by a single or combined technique. Hence this paper proposed a novel hybrid technique for echo cancellation and noise reduction for speech signals. The proposed technique utilizes Least Mean Square (LMS) technique for echo cancelation and Adaptive Neuro Fuzzy Inference System based non-linear filtering for noise reduction from speech signals. The performance of proposed Hybrid technique is evaluated using the speech quality parameters Peak Signal to Noise Ratio (PSNR) and Mean Square Error (MSE). The proposed hybrid technique shows highly improve response both in terms of PSNR and MSE as compare to LMS algorithm.*

Key Words: Echo cancellation, Noise Reduction, Speech Signal Processing, PSNR, MSE.

1. INTRODUCTION

The effect of acoustic echo and noise in the application like audio teleconferencing and bridging is a serious problem. Similar issues need to be taken care in speech based access to Digital Television (DTV) in closed rooms. The main objective of any echo and noise cancellation method is to estimate the near end speech signal from a set of mixed observations. In this work, the mixed observation signal includes near end speech signal, far end echo, and noise.
Echo and noise cancellation problems have been dealt with separately in literature. It is well known that two of most frequently applied algorithms for noise cancellation [1] are least mean squares (LMS) [2]-[5] and recursive least squares (RLS) [6]-[10] algorithms. Considering these two algorithms, it is obvious that LMS algorithm has the advantage of low computational complexity. On the contrary, the high computational complexity is the weakest point of RLS algorithm but it provides a fast adaptation rate. Thus, it is clear that the choice of the adaptive algorithm to be applied is always a tradeoff between computational complexity and fast convergence. The convergence property of the FAP and FEDS algorithms is superior to that of the usual LMS, NLMS, and affine projection (AP) algorithms and comparable to that of the RLS algorithm [11]-[14]. In these algorithms one of the filter coefficients is updated one or more at each time instant, in order to fulfill a suitable tradeoff between convergences rate and computational complexity [15]. The performance of the proposed algorithms is fully studied through the energy conservation [16], [17] analysis used in adaptive filters and the general expressions for the steady-state mean square error and transient performance analysis were derived in [15], [18].

On the other side for echo cancelation in speech signals, again the least mean squares (LMS), normalized least mean squares (NLMS) and recursive least squares (RLS) are popular algorithms for estimating the RIR and subsequently canceling the echo [19]. When signals from both near-end and far- end speakers co-exist, the AEC give erroneous results. In such scenarios, a Double Talk Detector (DTD) is used to selectively update the adaptive filter only during single-talk periods. No adaptation is done during double-talk periods to avoid unwanted divergence. A study of Kalman Filter for echo cancellation was carried out in [20] and [21]. However, both noise and echo with the near end speech signal need to be considered in practical echo cancellation systems. Therefore, novel hybrid technique for acoustic echo and noise cancellation in a using LMS and ANFIS based nonlinear filtering framework is proposed in this paper.

2. Acoustic Echo cancelation and Noise Reduction in Speech Signals using proposed Hybrid Technique.

This section presents the basic conceptual formulation of the proposed hybrid technique for the echo cancelation and noise reduction from speech signals. As already mentioned

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that the proposed technique is basically a combination of well-known filtering technique LMS and ANFIS based nonlinear filtering. The complete methodology of the proposed work for the echo cancellation and noise reduction is shown in figure (1) with the help of flow chart representation.

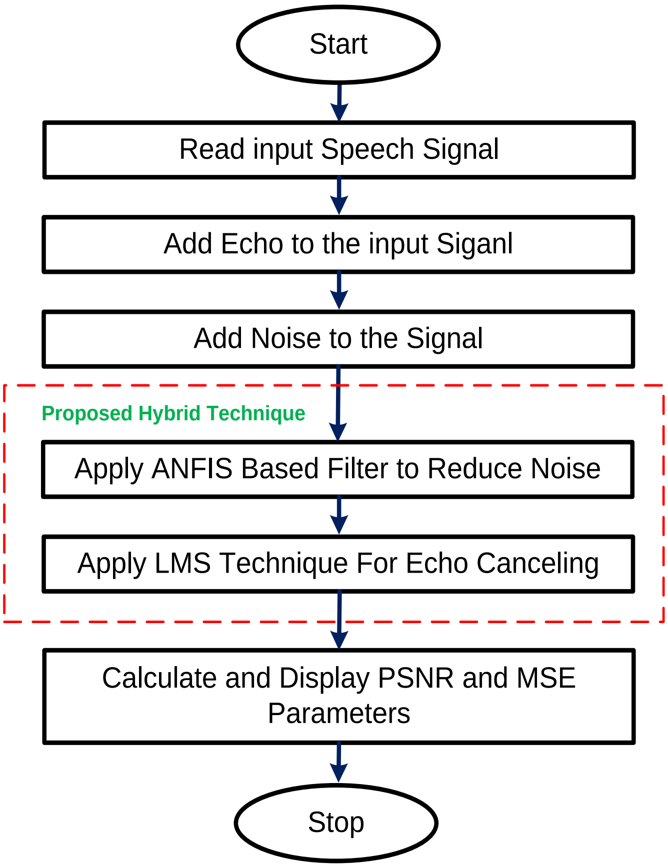

Fig-1: Flow Chart Representation of Proposed Work.

The detail description of the individual parts from the proposed hybrid technique is given in the following sub-sections.

2.1. Least Mean Square (LMS) Technique for Echo Cancelation

An acoustic echo belongs to one of the acoustic modeling problems. It happens when the direct signal follows multipath propagation as shown in Fig-2.

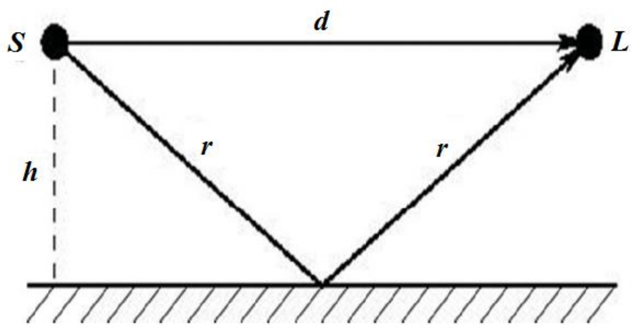

Fig-2: Multipath Propagation

As depicted in Fig-2, direct signal d from the source S at height h reaches the listener L which is followed by the reflected signals r having the magnitude almost same as direct signal. It is referred to as Echo signal. It is formed when the direct signal hits the obstacles in the room and gets reflected. Such an echo signal needs to be eliminated or suppressed for better signal perception [2].

For echo cancellation, adaptive filters driven by an error signal are used. Adaptive filters have adjustable filter parameters to minimize the undesired signal by using an adaptive algorithm. There are numerous adaptive algorithms used in an adaptive filter, out of which LMS (Least Mean Square) Algorithm has been used in this paper.

LMS is a stochastic gradient-based algorithm introduced by Bernard Widrow and Ted Hoff which uses gradient vector of the filter tap weights in order to converge on the optimal Wiener solution. In each iteration of the algorithm, the filter taps weights are updated as per equation (3) where $w(n)$ represents the adaptive filter weight vector at time n , $x(n)$ represents time delayed input signal samples, $e(n)$ represents error signal to be minimized and represents step size or convergence factor.

Output, $y(n) = w^h x(n)$ ... (1)

Error, $e(n) = d(n) - y(n)$ ... (2)

Weight, $w(n+1) = w(n) + \mu\, x(n)\, e(n)$ ... (3)

If μ is chosen to be very small then the algorithm converges very slowly. A large value of may lead to a faster convergence but the adaptive filter becomes less stable around the minimum value and its output diverges.

Now finally the adaptive LMS technique based echo cancelation structure utilized in this work is shown in fig-3.

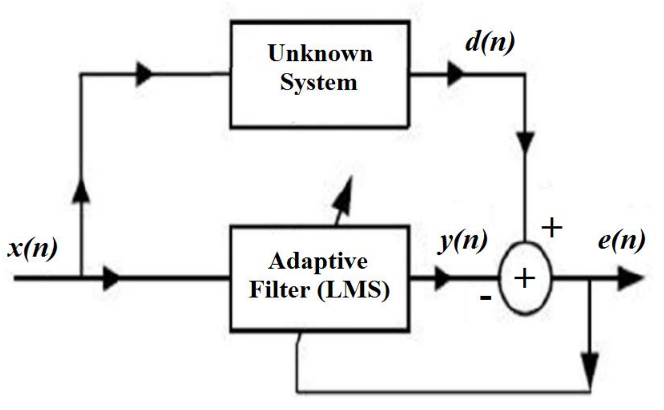

Fig-3: Echo cancelation using LMS technique.

2.2. ANFIS Based Non-linear Filtering for Noise Reduction

The proposed Adaptive Neuro Fuzzy Inference System (ANFIS) based noise reduction technique is basically a nonlinear prediction of noise content in the noisy signal. For instance, let the noise content of the noisy signal $N(n)$ is n_2, which is a random variation. This noise content can be considered as a nonlinear function of a randomly generated variation n_1. Hence we can write,

$$n_2 = f(n_1) \qquad ...(4)$$

The noisy signal $N(n)$, in terms of original signal $x(n)$ and the noise content n_2 is then written as,

$$N(n) = x(n) + n_2 \qquad ...(5)$$

However we don't have the post priory knowledge of function f of equation (4), but this relationship can be predicted with the help of nonlinear regression. Let the predicted or estimated value of n_2 is est_n_2 then the filtered signal $Y(n)$ is given as,

$$Y(n) = N(n) - est\_n_2 \qquad ...(6)$$

In the present paper, ANFIS has been utilized for the prediction or estimation of nonlinear function f given in equation (4).

For the training of the proposed ANFIS structure two inputs have been used in which the first input is the delayed version of a randomly generated signal n_2 while the second input is the signal n_1 itself. The training target taken is the available noisy signal $N(n)$. The final structure of the developed ANFIS is shown in fig-4.

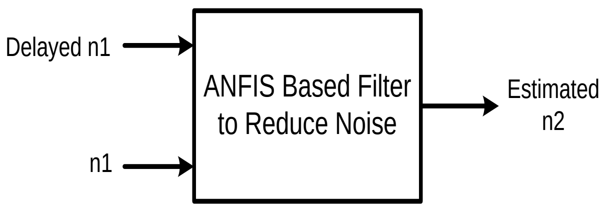

Fig-4: Structure of the developed ANFIS for Noise Reduction.

3. Results and Discussion

This section presents the extensive testing results obtained after echo cancelation and noise reduction from input speech signal using the proposed hybrid technique and LMS technique alone. To properly analyze the efficiency, both the techniques were applied over several input speech signals. Out of several analyzed results here we are reporting the one result input speech signals. The input signal used for testing is shown in fig- (5). Fig- (6) and fig- (7) shows the same input after addition of echo and noise respectively.

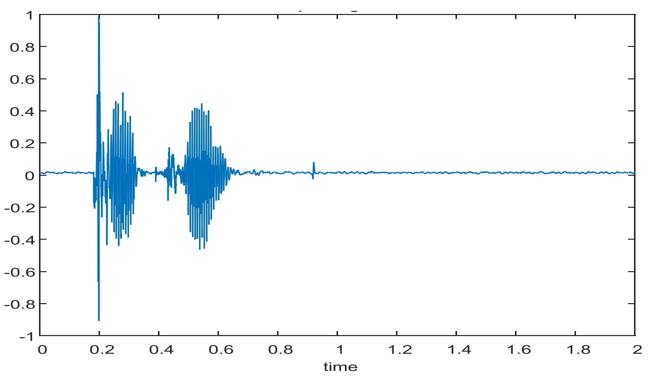

Fig-5: Input speech signal.

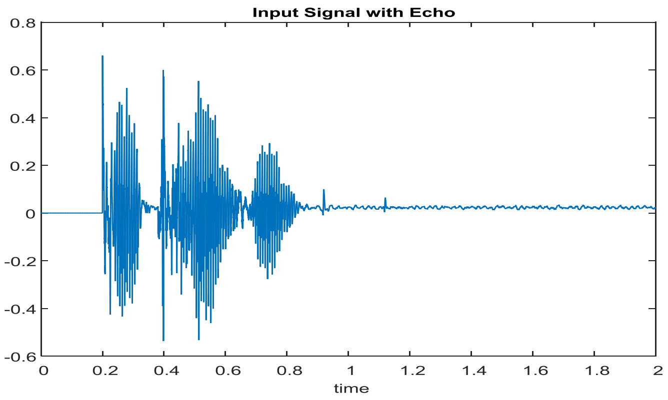

Fig-6: Input speech signal with Echo (Delay = 0.2 sec, Echo Strength = 0.6).

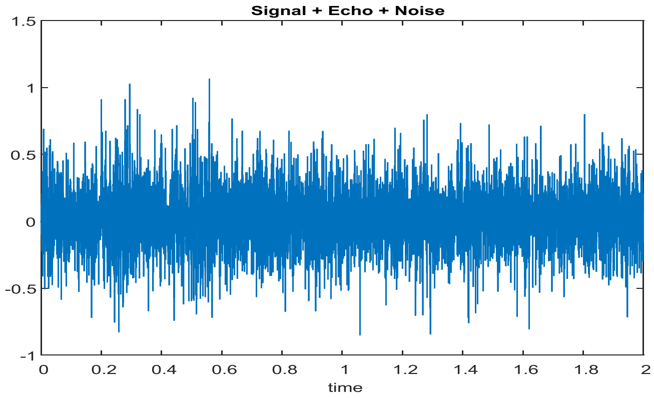

Fig-7: Input speech signal with added Echo and Noise (40% Random Noise).

Now the results obtained after echo cancelation and noise reduction from the signal shown in fig-(7), using the proposed hybrid technique is shown in fig- (8) and fig- (9). Whereas the resultant signal obtained after using the LMS filter alone for echo cancelation and noise reduction is shown in fig-(10).

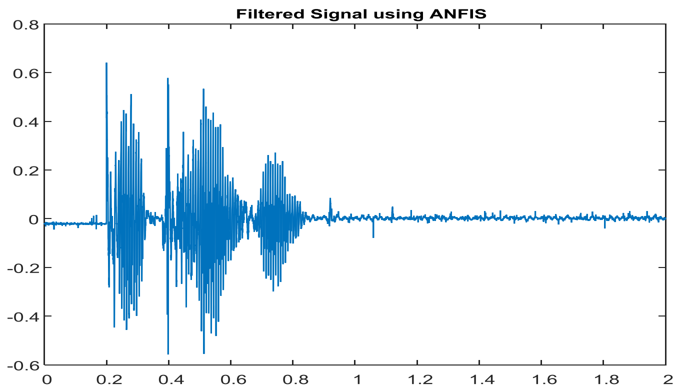

Fig-8: Filtered Signal using Proposed Hybrid Technique (ANFIS Based Filtering)

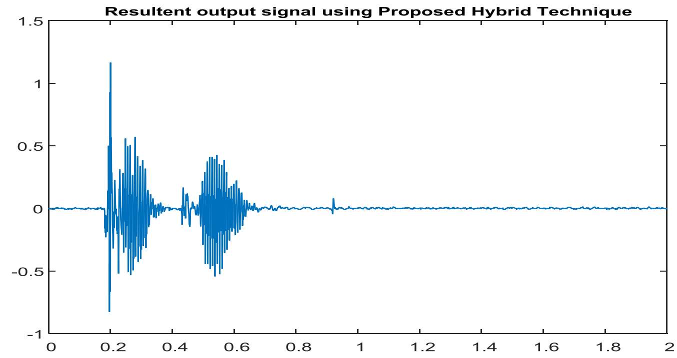

Fig-9: Echo Removed Signal using Proposed Hybrid Technique (Final Output)

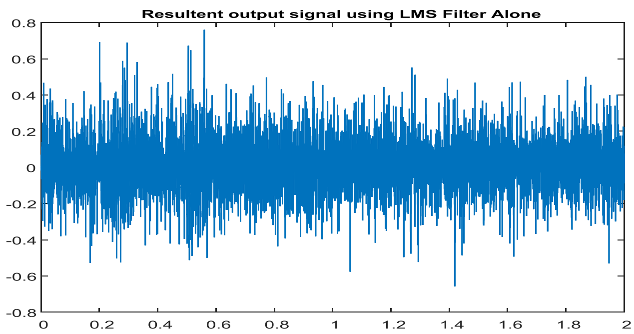

Fig- 10: Echo Removed and Noise Filtered Signal using LMS Technique (Final Output)

From the resultant signals obtained for the technique, it is clearly shown that the proposed hybrid technique provides very high efficiency of echo cancelation and noise reduction as compare to the conventional adaptive LMS technique. In addition to this the same comment is justified with the help of speech signal quality parameter obtained after echo and noise reduction using both the techniques. The PSNR and MSE values obtained after testing are tabulated in table-1.

Table-1 Speech quality Parameters obtained after echo and noise reduction.

S. No.	Parameter	LMS Technique	Proposed Hybrid Technique
1	PSNR	65.7203	73.0963
2	MSE	0.0174202	0.0031875

From table-1 it is clearly observable that the proposed hybrid technique provides very high gain in PSNR as compare to the LMS technique, while on the other side it provides the higher reduction in MSE and hence efficient to provide high quality restoration of speech signals.

4. Conclusion.

In this paper we have a novel hybrid technique for echo cancellation and noise reduction for speech signals. The proposed technique utilizes Least Mean Square (LMS) technique for echo cancelation and Adaptive Neuro Fuzzy Inference System based non-linear filtering for noise reduction from speech signals. The performance of proposed hybrid technique is evaluated using the speech quality parameters Peak Signal to Noise Ratio (PSNR) and Mean Square Error (MSE). Several tests have been performed to test the ability of the proposed technique as compared to conventional LMS technique. It has been found that the proposed hybrid technique provides very high gain in PSNR of about 8 dB as compare to the conventional LMS technique, while on the other side it provides the higher reduction in MSE and hence efficient to provide high quality restoration of speech signals.

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