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Comparative Performance Analysis of AEC using Adaptive Filter and Linear Prediction Filter

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Abstract— *With the development of technology the whole dimension of communication has changed. In modern day people are more interested in hand-free communication. So, the users use an ordinary loudspeaker and a microphone, at place of telephone receiver. Benefit of wireless communication is that it allows the users to have both hands free and to travel freely in the area. The truth of a huge audio mixture with the loudspeaker and microphone would produce a loud echo that would make conversation difficult. With an echo control or echo cancellation model these problem can be removed because it offers a simple solution to overcome the echo problem. But a major disadvantage is that it chains only half-duplex communication. Half-duplex communication permits only one presenter to talk at a moment. Due to this drawback echo cancellers is invented. Advantage of echo canceller over echo suppressor is that it uses full-duplex communications, which allow both speakers to talk at the same time. Keeping in mind these short coming, various acoustic echo cancellation methods are used. In this paper, two methods of AEC are presented and their simulink model are designed and implemented. Lastly, in order to evaluate the performance of the echo cancellation algorithm the measure of Echo Return Loss Enhancement (ERLE) is used.*

Keywords— *Loudspeaker, Microphone, Acoustic Echo, Adaptive Filter, Linear Prediction Filter.*

I. Introduction

Sound is reflected back to the transmitter due to the difference between the original sound or delayed sound is known Echo. When reflected wave arrive after a very small time, it is measured a reverberation. On the other hand if, the reflected waves make contact with a little ten of milliseconds then it is hear as an echo. Due to the reason of this Echo, communication becomes difficult or impossible.

A. Types of Echo

There are two types of echo in a telephone system which is described as given bellow

1. Acoustic Echo

Acoustic echo is generating due to acoustic coupling between the speaker and the microphone. The acoustic echo, which is also known as a “multipath Echo”, is produced by poor voice coupling between the loudspeaker and the microphone in handsets and hands-free devices.

2. Electrical line Echo or Hybrid Echo

Electrical line echo is due to impedance mismatch at the hybrid circuit connecting a 2-wire subscriber to a 4-wire trunk line in public switched network. Main cause of Hybrid echo is the public-switched telephone network (PSTN). Echoes on a telephone line are due to the reflection of speech signals at the points of impedance mismatch on the connecting circuits.

II. Literature Survey

Siqueira et al. In this paper the stable convergence behavior of LMS-based adaptive algorithms is studies.

*Eneroth et al***.** In this paper a wideband stereophonic acoustic echo canceller is presented.

Park et al. In this paper a new residual echo cancellation and a new double-talk detector appropriate for real-time implementation is proposed.

Marco Liem et al. The algorithm used by a low cost, single chip is studies in this paper. This proposed that the LMS type of algorithms is much more sensitive to noise and often require a broad parameterization specific to the operating environment.

Xiao Hu et al. Here without double-talk detection (DTD) a novel adaptive acoustic echo Cancellation is proposed. So it can work stably under all operating models.

Chankawee et al. Linear predictor is used in this paper to recover the presentation of AFC in the hearing-aid devices.

Spriet et al. This paper proposed adaptive feedback cancellation techniques that can reduce bias in the feedback path.

Huang et al. In this paper an FIR cascade structure for adaptive linear prediction was studied, in which each stage FIR filter was independently adapted using LMS algorithm.

Rafid Ahmed Khalil. In this paper LMS algorithm of adaptive filtering was examined and its application in acoustic echo cancellation system was found. A discrete signal processing was employed in MATLAB for simulation with real acoustic signals.

Amit Munjal et al. In this paper RLS algorithm is used to decrease the unwanted echo so that the communication quality increased.

Schnell et al. In this paper a time-varying linear prediction was proposed for speech analysis and synthesis.

Keisuke et al. In this paper long-term multi-step linear prediction, first estimated the late reverberation and then reduced the late reverberation effect by employing spectral subtraction.

Bekrani et al. A neural network based adaptive filtering approach is proposed in this paper.

Gazziro et al. In this paper the noise can be reduced which is make by the two people when they talk at the same time.

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III. Acoustic Echo Cancellation using Adaptive Filter

Since the original signal is known which goes to the loudspeaker, which can be used to predict and remove the signal picked up by the microphone. The process of doing this is called acoustic echo cancellation. There are various adaptive methods which are used to cancels the acoustic echo listed below:

- Least Mean Square (LMS) Algorithm.
- Normalized Least Mean Square (NLMS).
- Recursive Least Square (RLS) Algorithm.

Out of this here explain the Least Mean Square (LMS) Algorithm.

Acoustic Echo Cancellation using LMS algorithm

Windrow and Hoff develop the Least Mean Square (LMS) algorithm in 1959 during their study on pattern recognition [6]. Since LMS algorithm is becoming one of the mainly used algorithms in adaptive filtering. Fig 1.1 shows the block diagram of conventional AEC [14]. The remote speaker signal, which is always referred as far-end signal and denoted as '$x(n)$', passes through room acoustic filter 'h', producing an acoustic echo termed '$y(n)$'. The micro phone receives near-end speech signal '$v(n)$', together with the echo disregarding of the surrounding noise; the received signal '$d(n)$', thus consists of both $v(n)$ and $y(n)$.

$$d(n) = y(n) + v(n) = f(x,h) + v(n) \qquad (1.1)$$

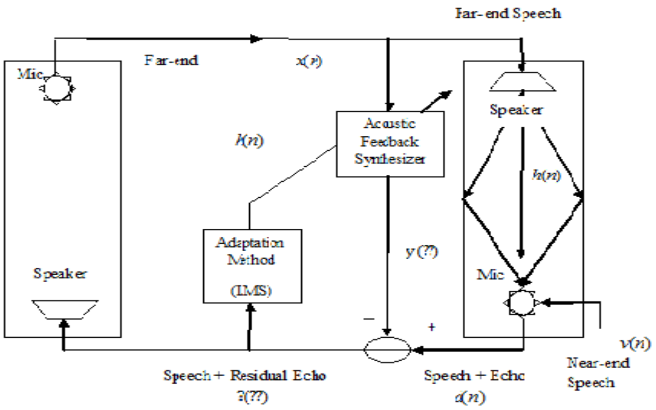

Fig. 1.1 Block diagram of the conventional AEC

The mission of the AEC is to model the room acoustic path with an adaptive filter $\hat{h}x(n)$ as well as possible and remove the echo signal from the measured signal, yielding a residual signal '$e(n)$', which will only consist of the near-end speech. The echo path in room is mainly time-varying which can be changed by the movement of objects or the moving of the loudspeaker or microphone one place to another. However, to capture the difficulty of an acoustic echo path, filter needs the infinity length, but a large filter order will bring a high computational load. So evidently there is a trade-off between the complexity and the performance of AEC.

The residual signal,

$$e(n) = d(n) - \hat{y}(n) = d(n) - \hat{h}x(n) \qquad (1.2)$$

should only consists of the near-end signal, which is the case when the acoustic adaptive filter is close to the echo path, namely $\hat{y}(n) \approx y(n)$, then $e(n) \approx v(n)$.The adaptive filter uses the residual signal $e(n)$ to estimate the error and update new filter coefficients, only if there is no near-end speech. It is main to decide whether the near-end speech is present or not. Hence an AEC normally includes component which is adaptive filter, or double-talk detector to detect if there is a near-end speech exists, and possibly a non linear processor to eliminate the residual echoes.

Model Description of AEC using LMS Adaptive Filtering

The Near-end and Far-end speech are pre-recorded speech. Which are loaded and coupled with the model by the use of "To wave device" blocks. Far-echo is create using "Echo Generator" subsystem. White noise has zero mean and variance unity where Near-end and Far-echo are added and then fed into the LMS filter i.e. "Desired Signal".

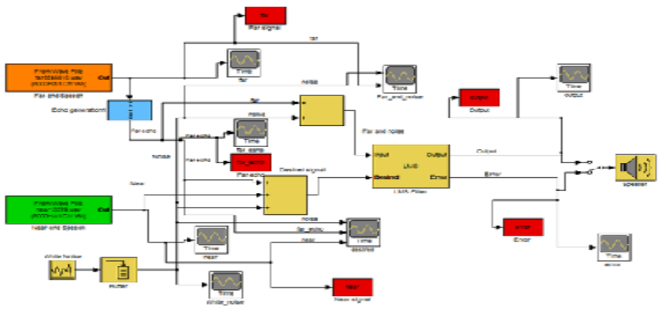

Fig. 1.2 Simulink model of AEC using LMS adaptive filtering

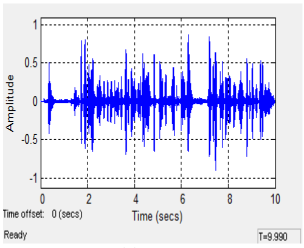

Fig. 1.3(a) Time scope Plot for near-speech

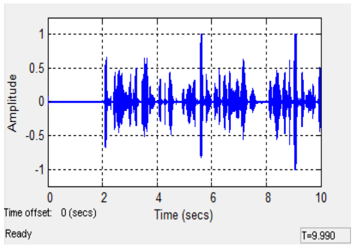

Fig.1.3 (b) Time scope Plot for far-end speech

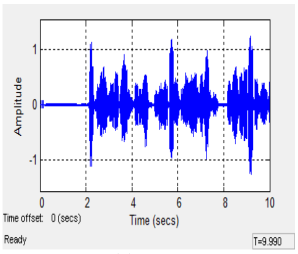

Fig.1.3(c) Time scope For far-echo speech

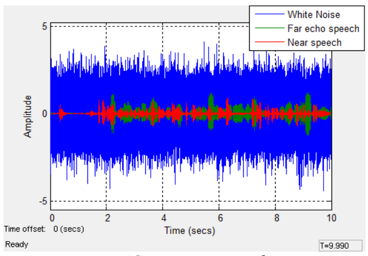

Fig.1.3 (d) Time scope for microphone speech signal

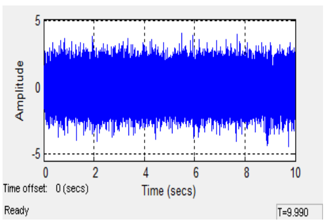

Fig.1.3(e) Time scope plot for output speech

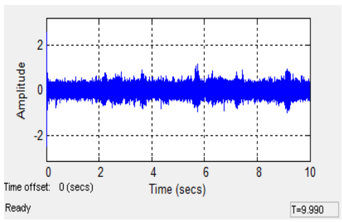

Fig.1.3 (f) Time scope plot for Error signal

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The plots for Near-end Speech and Far-end Speech of the typical LMS adaptive AEC are presented in figures 1.3 (a), (b).The Far-end signal is passed through the echo path to produced the echo signal. Far echo are presented in figures 1.3(c). The desired speech is a combined form of near speech, white noise and delayed echo of far speech which become the desired signal for the LMS adaptive filter to feed as shown in figure 1.3(d). The plots for LMS adaptive filter outputs; Error Speech and Output Speech are shown in figures 1.3(f) and (e). Double-talk did not take place in this simulation for the reason of open simulation environment and quicker convergence of the algorithm. The output of this module is the error signal, which is presented in figure 1.3(f). In the case of an ideal echo canceller the error signal should be perfectely same as that of near-end speech signal. However, due to the presence of residual echo, noise and nonlinearities; the error signal is not a perfect copy of the near-end speech signal. The results are not accurate using LMS adaptive algorithms. Thus the problem arises here. Therefore Linear prediction algorithms are used for better results.

IV. Acoustic Echo Cancellation using Linear Prediction (LP) Filter

As illustrated in figures 1.4 [4] a linear prediction model is an all-pole filter that forecasts the future values of a signal from a linear combination of its past values. LP filter outline the spectrum of the input signal by convert an uncorrelated excitation signal to correlated output signal. Where the inverse LP filter as the name indicate perform the reverse function of the LP filter which convert the correlated signal back to an uncorrelated smooth-band signal. An Inverse LP filter is an all-zero filter. Where the zero positioned at the same place in pole-zero plot like the poles of the all-pole filter and is also known as a spectral whitening filter. Adaptive echo cancellation systems work better (i.e. converge faster) if the input and the situated signals are uncorrelated white noise process. Speech signals can be pre-whitened by first modelling the speech with a linear prediction model and then using an inverse linear predictor for whitening the signals as even when speech signal are highly correlated or not as illustrated in figure 1.5 [4].

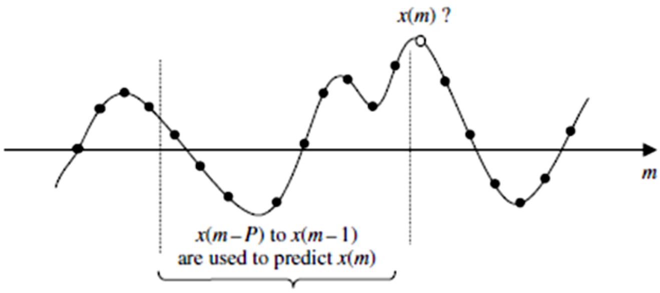

Fig. 1.4. Concept of prediction of the future sample $x(m)$ from its ‘P’ past samples

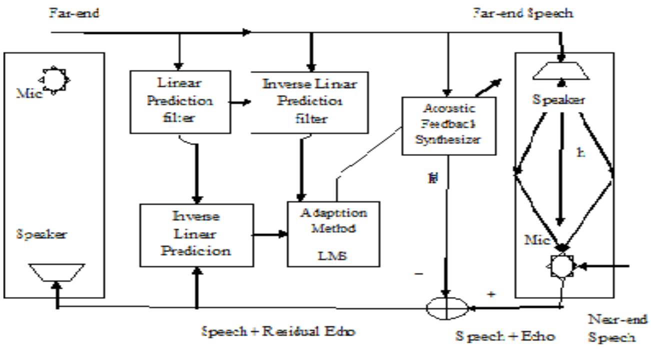

Fig. 1.5. Block diagram of the LMS adaptive AEC with LP filter

The pre-whitened input to adaptive filter, i.e. pre-whitened incoming signal, is given by

$$e(m) = x(m) - \sum_{k=1}^{P} a_k\, x(m-k)[4].$$

Same equation can be used to pre-whiten the adaptive echo canceller’s reference signal as shown in figure 1.5.

Model Description of AEC using Linear Prediction

To improve the performance of echo cancellation systems the process of pre-whitening are used. The model for LMS adaptive AEC with LP and ILP Filters is designed in MATLAB which called the simulink model.

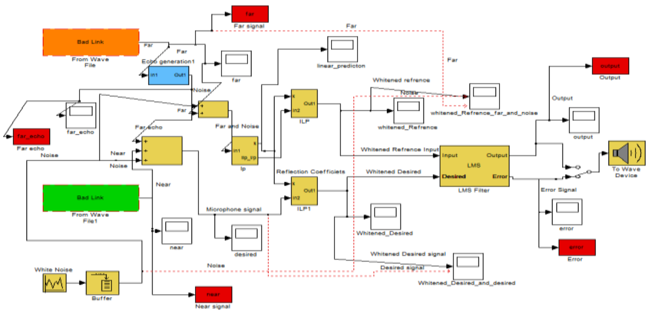

Fig. 1.6. Simulink model of AEC using linear prediction

The Audio files for both near-end and Far-end signal are recorded as a wave files and then stored in PC which is used in this model. LP is a well known all-pole method which calculates approximately the spoken signal.

Results of LMS Adaptive Echo Canceller with Linear Prediction Filter (LPAEC)

This section presents a graphical representation of the results obtained by simulating the LMS Adaptive Echo Canceller with Pre-whitening Linear Prediction Filter in MATLAB. Near-end Speech and Far-end Speech, far echo signal, Microphone signal, Noisy Far signal used in this model are same as used in Conventional LMS Adaptive Echo Canceller and represented in figure 1.3 (a), (b), (c), (d). Noisy Far-end signal is used as input for LP filter shown in figure 1.3(c) and plot for Linear Predicted Prewhitened Speech i.e. output of the LP filter is presented in figure1.7(c).

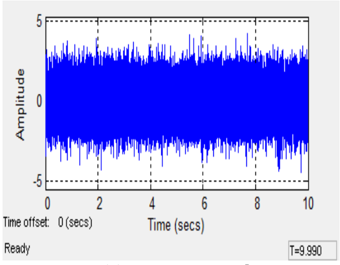

Fig.1.7 (a) Time scope plot For LP filter input

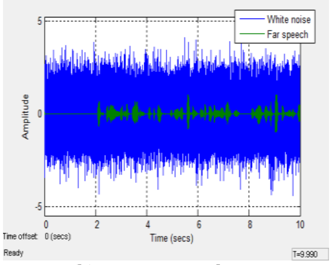

Fig.1.7 (b) Time scope plot far speech and white noise

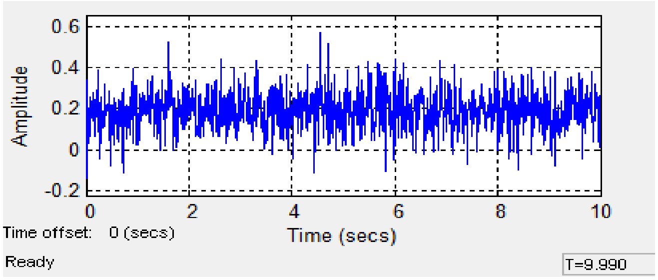

Fig. 1.7(c).Time scope plot for linear predicted prewhitened speech

The plots for LMS adaptive filter outputs; Error Speech and Output Speech are shown in figures 1.7(d) and (e). It is assumed that double-talk did not take place during this simulation. The output of this model is the error signal, which is presented in figure 1.7(d). The error speech of the LPAEC shows the better signal results as compared to Conventional AEC.

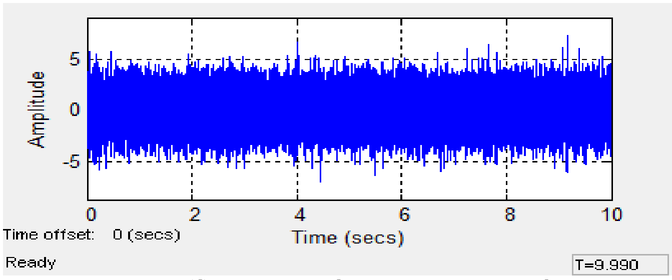

Fig. 1.7(d). Time scope plot LPAEC output speech

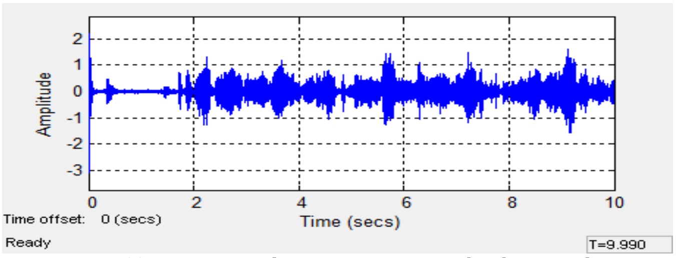

Fig. 1.7(e). Time scope plot LPAEC error speech echo return loss enhancement (ERLE)

In order to evaluate the performance of the echo cancellation algorithm the measure of Echo Return Loss Enhancement (ERLE) is used. ERLE is the ratio of the instantaneous power of the signal, $d(n)$ to the instantaneous power of the residual error signal, $e(n)$, immediately after cancellation [12].

$$ERLE = 10 \log_{10} \frac{E(d^2(n))}{E(e^2(n))} \quad (1.3)$$

It is a smoothed measure of the amount (in dB) that the echo has been attenuated. Referring to the literrature ERLE should be stablize in the interval [-40 dB, 30 dB] for a good performance. Figure 1.8(a) and (b) shows the ERLE of the adaptive AEC and for the linear predction AEC model.

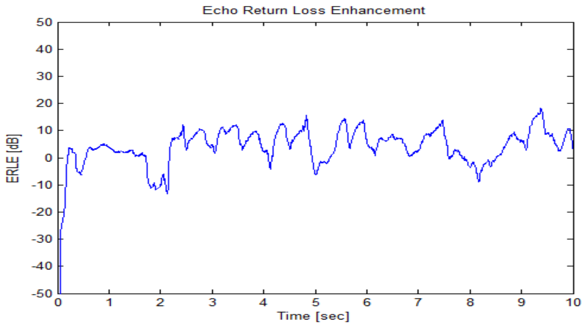

Fig. 1.8 (a). ERLE for LMS adaptive filter

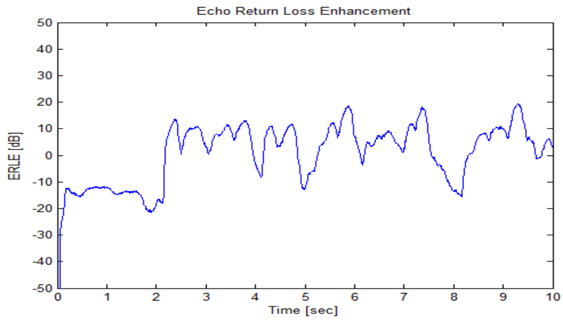

Fig. 1.8 (b). ERLE for linear prediction filter

V. CONCLUSION AND FUTURE WORK

Echo cancellation is used to cancel the problem of echo. This paper shows the comparison between conventional AEC and an integrated form of AEC using Pre-whitening Filter. The Results obtained from the Pre-whitening model is better as compared to the conventional AEC model. In acoustic echo cancellation performance has been improved and better results have been found by using Pre-whitening filter. So the AEC by using Pre-whitening (Linear Prediction) Filter is more useful and beneficial than adaptive AEC.

The performance of AEC can be improved and it is used in the Linear Prediction for different areas which are listed below.

- For Automatic Speech Recognition purpose linear predictor filter can be used.
- Linear Prediction can also be used for Spectral analysis; Signal restoration; and Noise reduction etc.

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