A Novel Adaptive method for Acoustic Echo Cancellation

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Abstract— Speech is compulsory in audio teleconferenceing system. In present scenareo acoustic echo is a major setback for user and causes a lessening in the quality of the communication.By means of some adaptive filtering methods acoustic echo canbe eliminated and can be reachrd in a desired value. A detail performance assessment is reported, including echo return loss enhancement (ERLE), convergence time and system distance metrics.We have also compared two different signals and how noise can be cancelled out using NLMS algorithm.

Keywords- *ERLE, NLMS, Acoustic echo cancellation.*

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I. INTRODUCTION

In teleconferencing system acoustic echo cancellation (AEC) [13]is used. The main 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. First, we describe the acoustics of the loudspeaker-to-microphone signal path where the speakerphone is located. We can use a long finite impulse response filter to describe these characteristics. The teleconferencing system's user is typically located near the system's microphone. Here is what a male speech sounds like at the microphone. Now we describe the path of the far-end speech signal. A male voice travels out the loudspeaker, bounces around in the room, and then is picked up by the system's microphone. Let's listen to what his speech sounds like if it is picked up at the microphone without the near-end speech present. The signal at the microphone contains both the near-end speech and the far-end speech that has been echoed throughout the room. The goal of the acoustic echo cancellers is to cancel out the far-end speech, such that only the near-end speech is transmitted back to the far-end listener. The algorithm that we will use in this paper is the Frequency-Domain Adaptive Filter (FDAF). This algorithm is very useful when the impulse response of the system to be identified is long. The FDAF uses a fast convolution technique to compute the output signal and filter updates. This computation executes quickly in MATLAB®. It also has improved convergence performance through frequency-bin step size normalization. We'll pick some initial parameters for the filter and see how well the far-end speech is cancelled in the error signal. Since we have access to both the near-end and far-end speech signals, we can compute the echo return loss enhancement (ERLE), which is a smoothed measure of the amount (in dB) that the echo has been attenuated. From the plot, we see that we have achieved about a 35 dB ERLE at the end of the convergence period. To get faster convergence, we can try using a larger step size value. However, this increase causes another effect, that is, the adaptive filter is "mis-adjusted" while the near-end speaker is talking.

II. ACOUSTIC ECHO CANCELLATION

Occurrence of acoustic echo is a common problem in telecommunication system. When an audio signal is reverberated in a real environment, resulting in original signal with the attenuated signal is generated that is called as acoustic echo . The signal interference caused by acoustic echo is off-putting to both the users and causes a reduction in quality of communication. Acoustic echo is reflected from a multitude of different surfaces like walls, ceilings and floors and travel through different paths. Normally it occurs when the input and output operate in full duplex mode. In this situation the received signal has outlet through the loudspeaker, the audio signal is reverberated through the physical environment and picked up by the microphone. This effect causes time delay and attenuates the original speech signal, hence reducing the speech quality [9].In the case of acoustic echo in telecommunications; the optimal output is an echoed signal that accurately emulates the unwanted echo signal.

III. SYSTEM MODEL

The model shown in the figure 1.The terminal receives a down-link (or loudspeaker) signal x(n) from a far-end speaker ,and transmits an uplink (or microphone) signal y(n). In addition to near-end speech s(n) and additive background noise n(n) the uplink signal potentially includes an additional echo component d(n), which is a result of the acoustical coupling between the loudspeaker and the microphone. It is generally modeled with linear convolution d(n=x(n)∗h(n)) where h(n) is the impulse response which characterizes the acoustical coupling AEC may thus be implemented by estimating h(n) with a filter $\hat{h}(n)$ in order to give an estimate of the coupled echo signal $\hat{d}(n) = x(n) * \hat{h}(n)$. The echo is attenuated simply by subtracting $\hat{d}(n)$ from the uplink signal. Since the acoustical coupling is generally time varying $\hat{h}(n)$ is usually an adaptive filter. Near-end speech disturbs the adaptive filter and so $\hat{h}(n)$ is usually updated during echo-only periods, i.e. when s(n) = 0. Noise can also disturb the adaptive filter but, if we also suppose that the noise is negligible, i.e. .n(n)=0 and y(n)=d(n) and thus the resulting error signal, e(n) is simply the difference between the echo signal and its estimate, i.e. $e(n) = d(n) - \hat{d}(n)$.The error e(n) is used to update the filter h(n) whose goal is to drive e(n) to zero. AEC rarely operates under such ideal conditions, however, and thus it is interesting to study the robustness under more realistic conditions. i.e. with near-end speech, non-linear echo and additive background noise. As adaptation is simply paused during intervals of near-end speech, only disturbances from non-linear echo and background noise are considered here.

Each of the approaches to AEC that are considered are described below.

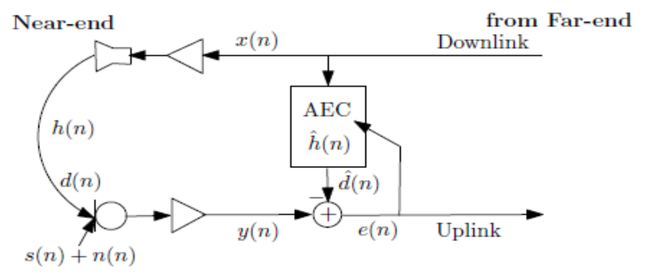

Fig. 1. System/echo model illustrating the acoustical coupling between the loudspeaker and microphone and a general approach to adaptive AEC.

III. DETERMINATION OF THE ECHO CANCELLATION:

A. ECHO RETURN LOSS ENHANCEMENT (ERLE):

It is the ratio of input desired signal power and the power of a residual error signal without delay after echo cancellation. It can be measured in decibel (dB). ERLE [3] depends on the dimension of the adaptive filter and the algorithm design. The better echo cancellation means higher the value of ERLE. It also measures the amount of loss introduced by the adaptive filter. It can be described as

$$ERLE = 10\ \log_{10} \frac{p_d}{p_e} \qquad (1)$$

Where p_d is the desired signal power and p_e is the power of a residual error signal after echo cancellation.

IV.ADAPTIVE ALGORITHM

A. LMS ALGORITHM:

It is based on steepest descent algorithm. LMS algorithm is used extensively for many applications such as channel equalization and echo cancellation. This algorithm is used due to its computational simplicity. Computational complexity for LMS is 2N+1 multiplications and additions. The equation below is LMS algorithm for updating the tap weights of the adaptive filter for each iteration [10].

$$W(n+1) = W(n) + \mu e(n)\, x(n \qquad (2)$$

Where

$$X(n) = [x(n),\ x(n-1),\ x(n-2)\ \ \ldots x(n-N+1)]$$

is the input vector of time delayed input value and
and

$$w(n) = [W_0(n), W_1(n-1), W_2(n-1)\ \ \ldots W_{N-1}(n)]$$

is the weight vector .

One of the main disadvantages of the LMS algorithm is that it has a fixed step size for each iteration. Determining the upper bound step size is a problem for the variable step size algorithm, if the input signal to the adaptive filter is non-stationary. A convenient way to incorporate this step size into the LMS adaptive filter is to use a time varying step size. NLMS is an extension of the LMS algorithm which bypasses the issue by selecting a different step size value μ(n) ,for each iteration of the algorithm. Step size is inversely proportional to the inverse of the total expected energy of the instantaneous values of coefficients of the input vector x(n).

B. APPLICATION OF NLMS ALGORITHM:

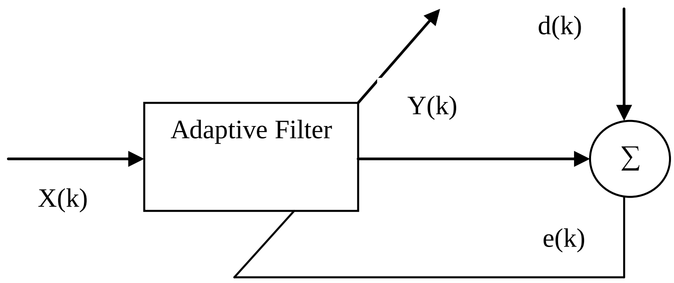

Figure 2: INTERFERENCES CANCELETION MODEL

The adaptive filter output can be calculated as

$$Y(n) = W^T X(n) \qquad (3)$$

An error signal is calculated as the difference between the desired signal and the filter output

$$e(n) = d(n) - y(n) \qquad (4)$$

The filter tap weights are updated for the next iteration

$$(n+1) = W(n) + \mu e(n)\, x(n) \qquad (5)$$

The NLMS algorithm gives greater stability with unknown signals. It's convergence speed and relative computational simplicity make the NLMS algorithm ideal for the real time adaptive echo cancellation system [16].

V. SIMULATION AND DISCUSSION:

- First, we explain the acoustics of the loudspeaker-to-microphone signal path where the speaker phone is located. These characterist can be described by using a long finite impulse response filter.
- Generate the near-end and far-end Speech Signal.
- From loudspeaker male voice comes out , bounces around in the room, and then is selected up by the system's microphone. Let's listen to what his speech sounds like if it is

selected up at the microphone without the near-end speech present.

- The signal echoed through out the room is nothing but both the near-end speech and the far-end speech.
- The objective of the acoustic echo canceller is to cancel out the far-end speech, such that only the near-end speech is transmitted back to the far-end listener.

We can apply adaptive techniques such as fast convolution technique to compute the output signal and filter updates. This

computation executes quickly in MATLAB.

- We'll choose some initial parameters for the filter and see how well the far-end speech is cancelled in the error signal.

- Since we have access to both the near-end and far-end speech signals, we can compute the echo return loss enhancement (ERLE)

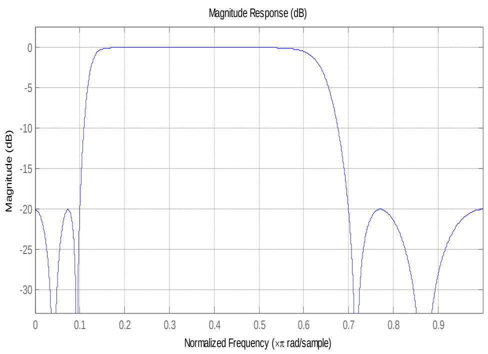

Figure3: Analyze the filter

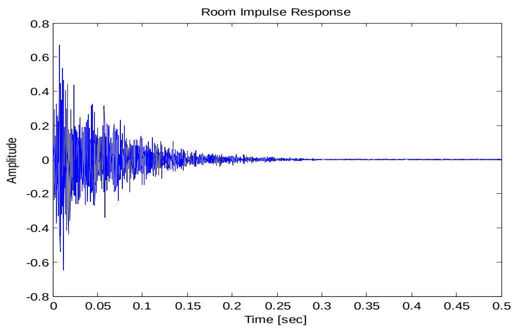

Figure 4: Room Impulse Response

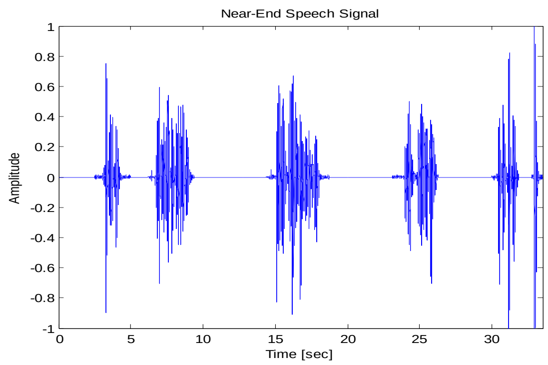

Figure5: Near-End Speech Signal

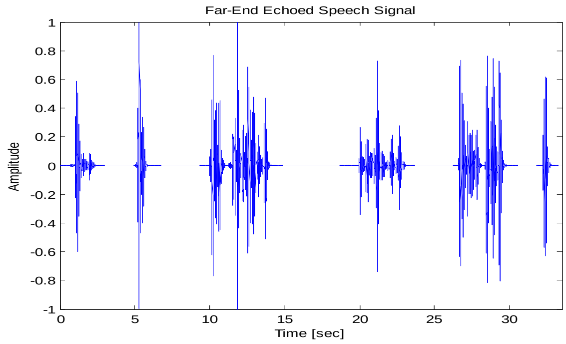

Figure 6: Far-End Speech Signal

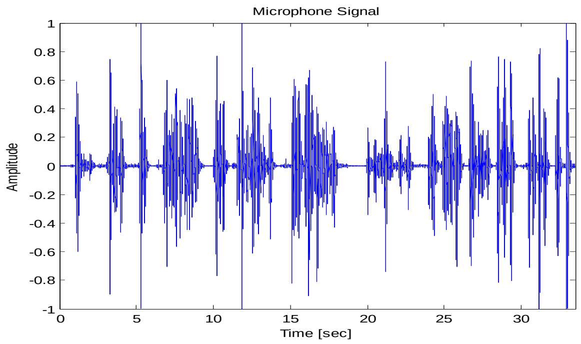

Figure 7: Microphone Signal

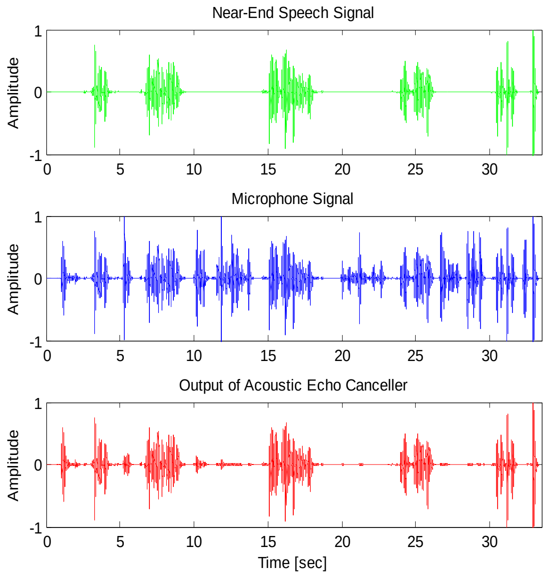

Figure 8: Plot of various signals

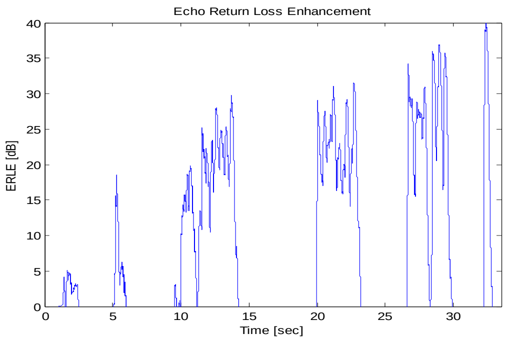

Figure 9: Plot of ERLE

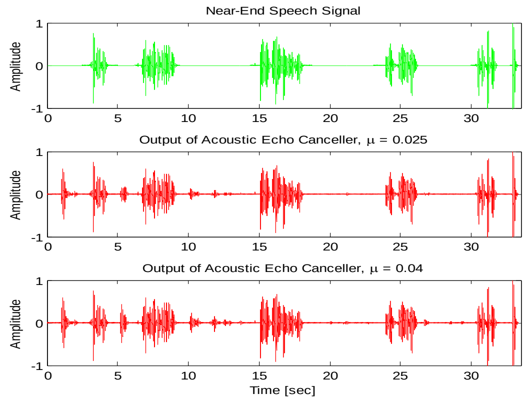

Figure 10: Output of Acoustic Echo Canceller of different values of µ

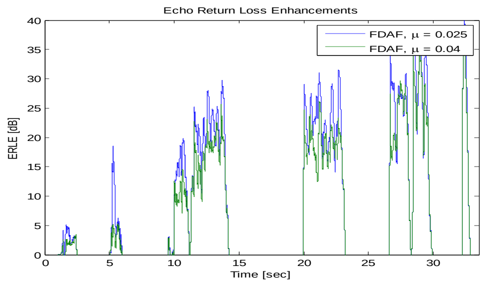

Figure 11:Plot of ERLE of different values of µ

SIMULATION II:

In this part of simulation we take two different signals one is A*sin(2*pi*f*t*Ts)

Where A is amplitude (Here A=1) ,f=1000Hz,Sampling period Ts=125e-6 and We select another random signal and noise can be reduced using NLMS algorithm .plot MSE and ERLE.

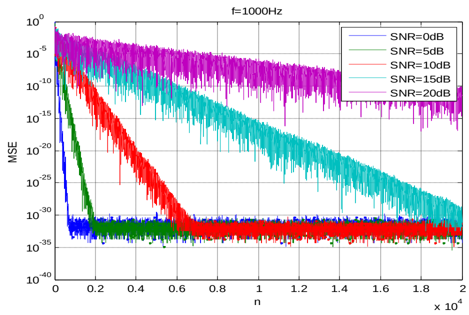

Figure 12 :Plot MSE of different values of SNR(Case-i)

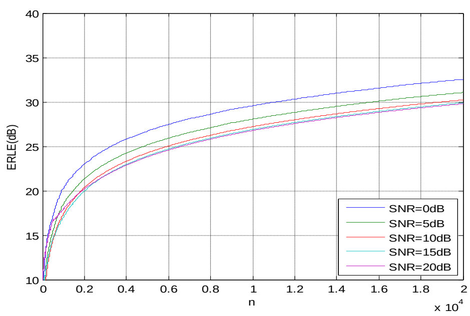

Figure13: Plot ERLE(dB) of different values of SNR(Case-i)

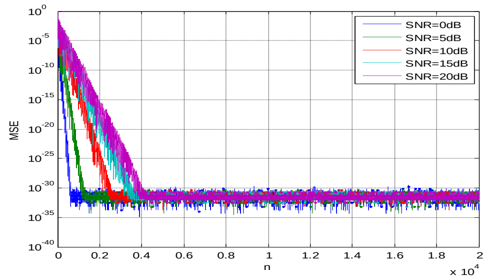

Figure14:Plot MSE of different values of SNR(Case-ii)

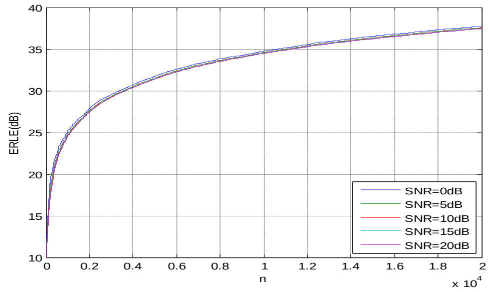

Figure15: Plot ERLE(dB) of different values of SNR (case-ii)

SIMULATION III:

In this part of simulation

We take a signal along with noise (input to the adaptive filter)

Echo percentage of signal is 0.4

Mu=0.01,Adaptive filter order is 20.

Signal frequency is 2000Hz.

Sample rate is 8000(samples/sec)

Echo time delay is 0.05.

Apply algorithm using LMS.

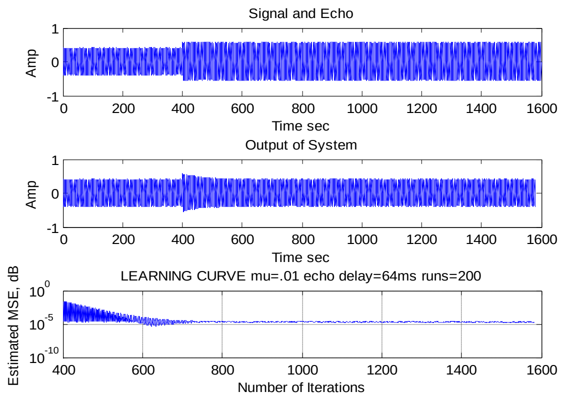

Figure 16:Echo Cancellation simulation

CONCLUSIONS AND FUTURE WORK:

In this work we have seen from different figures i.e the performance behaviour of Far end , Near end speech signal and input to microphone signals. A detail performance assessment is reported, including echo return loss enhancement (ERLE), convergence time and system distance metrics. The algorithm that we have used in this is the Frequency-Domain Adaptive Filter (FDAF).We have compared two different signals and how noise can be cancelled out using NLMS algorithm . Future work can be extended by implementing the experimental setup for the double talk situations, where both the, far-end user and near-end user speak simultaneously.

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