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Acoustic Echo Cancellation Using Adaptive Least Mean Square Algorithm

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***Abstract*— In telephony system, the received signal by the loudspeaker, is reverberated through the environment and picked up by the microphone. It is called an echo signal. This is in the form of time delayed and attenuated image of original speech signal, and causes a reduction in the quality of the communication. Adaptive filters are a class of filters that iteratively alter their parameters in order to minimize a difference between a desired output and their output. In the case of acoustic echo, the optimal output is an echoed signal that accurately emulates the unwanted echo signal. This is then used to negate the echo in the return signal. The better the adaptive filter simulates this echo, the more successful the cancellation will be. This paper examines LMS algorithm of adaptive filtering in acoustic echo cancellation system. A discrete signal processing is employed in MATLAB for simulation with acoustic signals.**

***Keywords*— Acoustic Echo Cancellation, Adaptive Filter, LMS Algorithm, FIR filter, MATLAB**

I. INTRODUCTION

As voice communication becomes an ever-more important and pervasive part of everyday lives, the issue of speech quality becomes more critical. With the rise in mobile communication, it is becoming more frequent to use a communication device in an enclosed noisy environment, such as a subway or in a lobby. In this setting however, the received microphone is severely degraded by the echo from the speaker and background noise. One of the reasons for the undesirable quality degradation is the appearance of audible echoes. This kind of quality degradation is inherently from network equipment and end-user devices. The audio processing necessary to clarify the desired speech can be broken down into two parts, removal of the acoustic echo and removal of the background noise. To increase speech quality and improve listening experience, it is necessary to design effective acoustic echo cancellation systems. Acoustic Echo Cancellation (AEC) is commonly done with an adaptive filter, frequently done with stochastic-gradient adaptive algorithms that use a Least-Mean Square (LMS) approximation. However, background noise and other non-desired artifacts such as voice reverberation; negatively affect the performance of these filters. In general, the adaptive algorithm is used to estimate the acoustic echo and subtracts this estimation from the near-end microphone signal.

II. ACOUSTIC ECHO CANCELLATION

Acoustic echo is formed when the sound emitted by a speakerphone's loudspeaker gets reflected from the walls, ceilings, floor, furniture, people, and etc. back to the speakerphone's microphone. Sound pressure level decreases with each reflection. The reflections, being repeated multiple times, create reverberation effect. High-level acoustic echo becomes very annoying and disturbing, and thus it shall be removed to enable handset-like conditions. Moreover, if both people use speakerphones, the acoustic feedback may (and often does) lead to ringing and howling, thus disabling the conversation entirely.

With nominal gains in the loudspeaker's and microphone's path, the acoustic echo is usually about -8…-3 dB in typical rectangular plain-wall office rooms without much of soft furniture. The resonating cabinet of the speakerphone itself usually drives the total echo up to +5…+10 dB. In some conditions, acoustic echo may be 20 dB higher than the original received signal. The echo cancellation scheme is depicted in Fig. 1.

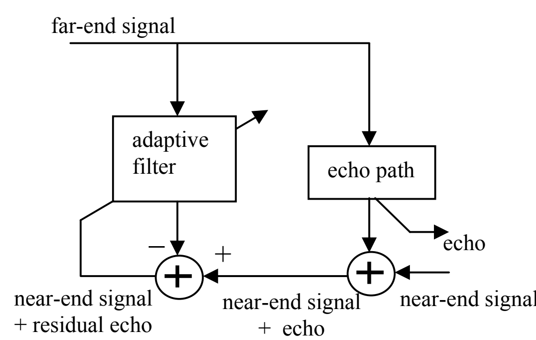

Fig. 1 Echo cancellation scheme

Here the echo path is the 'plant' or 'channel' to be identified. The goal is to subtract a synthesized version of the echo from another signal (for example, picked up by a microphone) so that the resulting signal is 'free of echo' and really contains only the signal of interest. A simple echo cancellation set up is shown in Fig. 2.

In this research, a male voice travels out the loudspeaker, bounces around in the room, and then is picked up by the system's microphone. 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 canceller is to cancel out the far-end speech, such that only the near-end speech is transmitted back to the far-end listener.

This scheme applies to hands-free telephony inside a car, or teleconferencing in a conference room. In teleconferencing system, user is typically located near the system's microphone.

The far end signal is fed into a loudspeaker (mounted on the dashboard, say, in the hands-free telephony application). The microphone picks up the near-end talker signal as well as an echoed version of the loudspeaker output, filtered by the room acoustics. The desired signal (see Fig.1 again) thus consists of the echo ('plant output') as well as the near-end talker signal. It is assumed that the near-end signal is statistically independent of the far-end signal, which results in the adaptive filter trying to model the echo path as if there were no near-end signal.

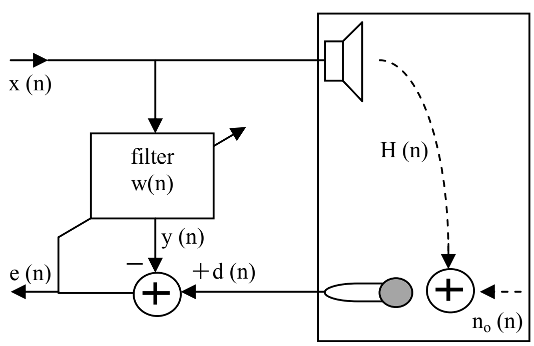

Fig. 2 Acoustic echo cancellation set-up

The filter weights are adjusted principally in those periods, when only the far end party is talking. In these periods, the error signal is truly a residual echo signal, and hence may indeed be fed back to adjust the filter. Recall that the adaptive filter has an adaptation and a filtering process. The filtering process is run continuously, even in the presence of the near-end talker, to remove the echo. It only the adaptation of the filter weights that gets switched off. Such a scheme clearly requires an extra circuit that can detect when the near-end talker is speaking.

The method used to cancel the echo signal is known as adaptive filtering. Adaptive filters are dynamic filters which iteratively alter their characteristics in order to achieve an optimal desired output. An adaptive filter algorithmically alters its parameters in order to minimize a function of the difference between the desired output $d(n)$ and its actual output $y(n)$. This function is known as the objective function of the adaptive algorithm. Fig.2 shows a block diagram of the adaptive echo cancellation model. Where the $x(n)$ is input signal, the filter $H(n)$ represents the impulse response of the acoustic environment, $w(n)$ represents the adaptive filter used to cancel the echo signal. The adaptive filter aims to equate its output $y(n)$ to the desired output $d(n)$ (the signal reverberated within the acoustic environment). The external noise input $n_o(n)$ is neglected here. At each iteration the error signal, $e(n)$,$d(n)$,$y(n)$, is fed back into the filter, where the filter characteristics are altered accordingly [1].

A. The Echo Cancellation Process

(i) The AEC reference signal is initially sampled by an adaptive filter.
(ii) The parameters of the adaptive filter are adjusted during convergence.
(iii) The resulting filtered signal is a model of the acoustic echo paths, 180 degrees out of phase.
(iv) Phase cancellation removes the echo when filtered signal is combined to the AEC input microphone signal.
(v) Non linear processing removes residual echo if any.
(vi) Finally, noise reduction is applied to remove static background noise.

III. Adaptive Filters

One common application is to use adaptive filters to identify an unknown system, such as the response of an unknown communications channel or the frequency response of an auditorium, to pick fairly divergent applications. Other applications include echo cancellation and channel identification.

The adaptive algorithm used to reduce the error between the output signal *y(k)* and the desired signal *d(k)*. When the LMS performance criteria for *e(k)* has achieved its minimum value through the iterations of the adapting algorithm, the adaptive filter is finished and its coefficients have converged to a solution. Now the output from the adaptive filter matches closely the desired signal *d(k)*. When you change the input data characteristics, sometimes called the filter environment, the filter adapts to the new environment by generating a new set of coefficients for the new data. Notice that when *e(k)* goes to zero and remains there you achieve perfect adaptation; the ideal result but not likely in the real world.

In the Fig. 3, the unknown system is placed in parallel with the adaptive filter. Clearly, when *e(k)* is very small, the adaptive filter response is close to the response of the unknown system. In this case the same input feeds both the adaptive filter and the unknown [2].

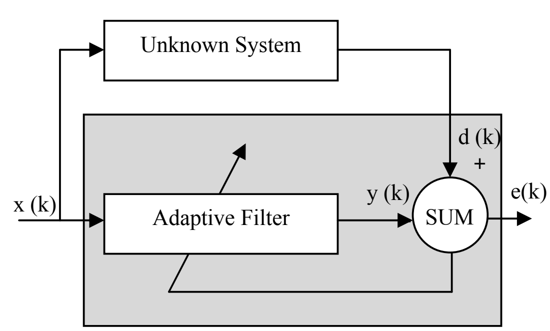

Fig. 3 Using an adaptive filter to identify an unknown system

A. System Identification

There are several ways to reduce the computational complexity of echo cancellation algorithms including block adaptive filters, subband filtering, and frequency domain adaptive filters. Both the subband method and the frequency domain method not only are more computationally efficient but also converge faster than the standard LMS algorithm.

Here *w* represents the coefficients of the adaptive filter tap weight vector, *x* (*n*) is the input vector samples, the tapped delay line D, is needed to make full use of the filter. The input signal enters from the left and passes through *N*-1 delays. The output of the tapped delay line (TDL) is an *N*-dimensional vector, made up of the input signal at the current time, the previous input signal, etc. the *y*(*n*) is the adaptive filter output, *d*(*n*) is the desired echoed signal and *e*(*n*) is the estimation error signal at time n.

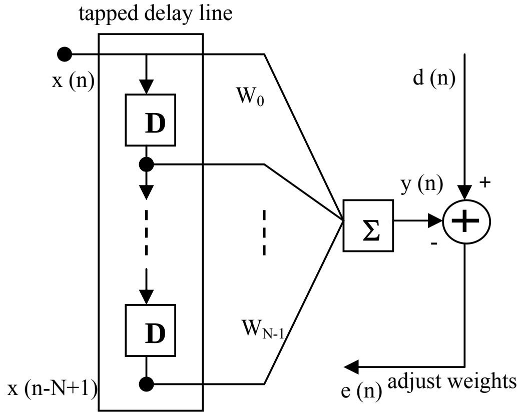

Fig. 4 Adaptive filter block diagram

Each iteration of the LMS algorithm requires three distinct steps:

Step 1: The output of the FIR filter, y(n) is calculated using Equation (1)

$$y(n)=\sum_{i=0}^{N-1} w(n)x(n-i)=w^{-T}(n)\bar{x}(n) \quad (1)$$

Step 2: The value of the error estimation is calculated using Equation (2)

$$e(n)=d(n)-y(n) \quad (2)$$

Step 3: The tap weights of the FIR vector are updated in preparation for the next iteration, by Equation (3)

$$\bar{w}(n+1)=\bar{w}(n)-2\mu e(n)\bar{x}(n) \quad (3)$$

Note that for each iteration the LMS algorithm requires 2N additions and 2N+1 multiplications (N for calculating the output y(n), one for 2μe(n)and an addition N for the scalar by vector multiplication.[1]

B. Frequency Domain Adaptive Filter

FFT domain computations of the linear correlation are as follows.

The adaptation equation is

$$\underline{w}(k+1)=\underline{w}(k)+\mu\sum_{i=0}^{M-1}\underline{u}(kM+i)e_{\underline{u}(k)}(kM+i)$$

$$=w(k)+\underline{\mu}\emptyset \quad (4)$$

Due to linearity of FFT, we can write

$$\mathrm{FFT}\begin{bmatrix}\underline{w}(k+1)\\ \underline{0}\end{bmatrix}=\mathrm{FFT}\begin{bmatrix}\underline{w}(k)\\ \underline{0}\end{bmatrix}+\mu\mathrm{FFT}\begin{bmatrix}\underline{\emptyset}\\ \underline{0}\end{bmatrix} \quad (5)$$

The fast LMS algorithm (Frequency Domain Adaptive Filter (FDAF)) for each block of M data samples does the following:

1. Compute the output of the filter for the block kM,…,kM+M-1

$$\begin{bmatrix}\underline{C}\\ \underline{y}\end{bmatrix}=\mathrm{IFFT}\left(\mathrm{FFT}\left(\begin{bmatrix}\underline{w}(k)\\ 0\end{bmatrix}\right)\times\mathrm{FFT}(|\underline{u}|)\right)$$

2. Compute the correlation vector

$$\begin{bmatrix}\underline{\emptyset}\\ \underline{D}\end{bmatrix}=\mathrm{IFFT}\left(\mathrm{FFT}\left(\begin{bmatrix}\underline{0}\\ \underline{e}\end{bmatrix}\right)\times\overline{\mathrm{FFT}(|\underline{u}|)}\right)$$

3. Update the parameters of the filter

$$\mathrm{FFT}\begin{bmatrix}\underline{w}(k+1)\\ \underline{0}\end{bmatrix}=\mathrm{FFT}\begin{bmatrix}\underline{w}(k)\\ \underline{0}\end{bmatrix}+\mu\mathrm{FFT}\begin{bmatrix}\underline{\emptyset}\\ \underline{0}\end{bmatrix}$$

Frequency-Domain Adaptive Filter (FDAF) 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 [5].

IV. SOFTWARE IMPLEMENTATION

A long finite impulse response filter is used to describe the acoustics of the loudspeaker-to-microphone signal path characteristics. A random impulse response is assumed a system sampling rate of f_s = 8000 Hz. In Filter Design Toolbox, `adaptfilt.fdaf` is used to implement frequency domain adaptive filter. Fig. 5 shows the flow of the software [2].

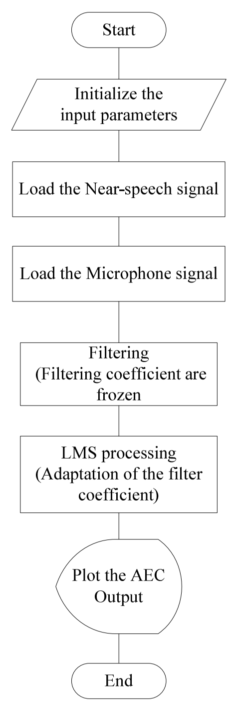

Fig. 5 Flowchart of the AEC system

In this system the FIR adaptive filter that uses frequency domain with bin step size normalization is implemented with filter length of 10 and step size of 0.03 and 0.06.

V. Simulation Results

The application of adaptive filters to acoustic echo cancellation (AEC) is illustrated by using MATLAB. Some initial parameters for the filter are picked and see how well the far-end speech is cancelled in the error signal.

Fig. 6 shows the filter response curve of frequency versus magnitude in dB. The output of acoustic echo canceller with its near end speech signals and microphone signals are simulated in Fig. 7. To get faster convergence, larger step size values are tested. However, this increase causes another effect, that is, the adaptive filter is "mis-adjusted" while the near-end speaker is talking. The output of the AEC with various step sizes is depicted in Fig. 8.

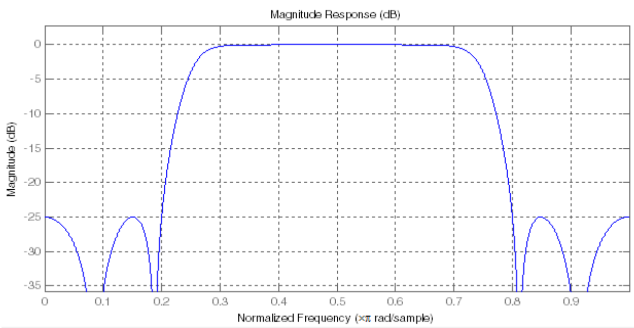

Fig. 6 Filter response curve of frequency versus magnitude (dB)

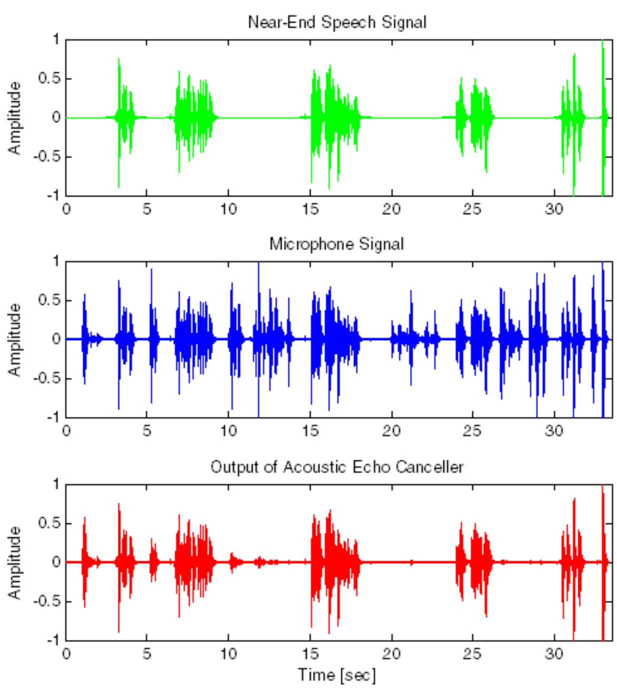

Fig. 7 Output of acoustic echo canceller

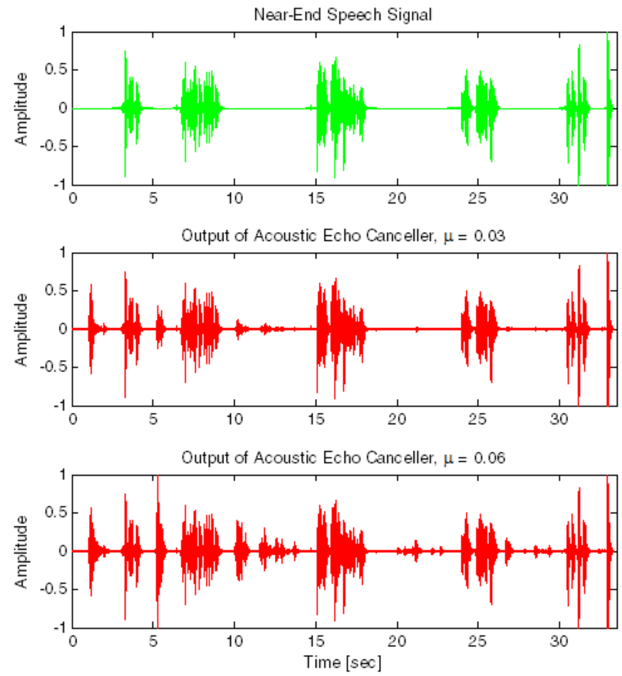

Fig. 8 Output of acoustic echo canceller with various step size

VI. Conclusions

In this paper, an AEC scheme approach is proposed, whereby a 4th order Cheby2 filter with an Adaptive LMS algorithm is employed. The various step sizes of 0.03 and 0.06 were simulated and the proposed approach provides faster convergence in larger step size values and better AEC performance. The performance of the LMS adaptive filtering algorithm is expressed by its simplicity to implement and its stability when the step size parameter is selected appropriately. This made the LMS algorithm the acceptable choice for implementing acoustic echo cancellation system. One of the main reasons is that FDAFs have the ability to reduce complexity, especially when the filter length becomes very large.

Acoustic echo cancellers include a detection scheme to tell when near-end speech is present and lower the step size value over these periods. The performance of the system with the larger step size is not as good as the former. This performance comes at the cost of computational complexity and considering the large FIR order required for echo cancellation, this is not feasible for real time implementation. In future, this echo cancellation with other adaptive algorithms can also be performed.

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