Acoustic Echo Cancellation by Adaptive Filtering in Speech Processing

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Abstract - **Adaptive filtering process constitutes one of the core techniques in digital signal processing and finds numerous application areas. A design have been proposed for an echo cancellation system for speech teleconferencing and tested for the performance evolution. This system is based on Adaptive Filtering process and uses Frequency Domain Adaptive Filter (FDAF) technique. Acoustic echo cancellation by Least Mean Square algorithm (LMS) is compared with frequency domain adaptive filter technique and it is found that the result obtained by the proposed method perform better in finding the echo free signal.**

Keywords - Adaptive filters, Adaptive algorithms, Acoustic echo cancellation, Least Mean Square Algorithm, Frequency Domain Adaptive Filter.

I. INTRODUCTION

Acoustic Echo Cancellation (AEC) 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) system. When an adaptive filter works in an adaptive environment, the filter coefficients adapt its response to revolutionize in the applied input signals. The function of the adaptive filter is to estimate the characteristics of the echo path, generating the echo and compensate for it. To perform this, the echo path is viewed as an unknown system with some impulse response

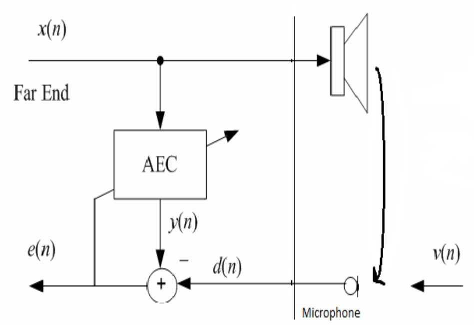

Fig. 1. Echo cancellation scheme

and the adaptive filter must emulate this response.

Acoustic echo cancellation is important for audio teleconferencing when simultaneous communication (or full-duplex transmission) of speech is necessary. Figure 1 shows an acoustic echo cancellation scheme where a measured microphone signal $d(n)$ contains two signals: - the near-end speech signal $v(n)$ and the far-end echoed speech signal $x(n)$.The goal is to remove the far-end echoed speech signal by removing filter output $y(n)$ so that only the near-end speech signal $e(n)$ is transmitted [1][2].

The paper is organized as follows. Section I introduces the acoustic echo cancellation scheme. In section II, the acoustic echo cancellation concept is outlined and various cause of acoustic echo along with working of echo cancellation is discussed. Section III contains the basic idea of previous work done for echo cancellation by Least Mean Square (LMS) algorithm. Section IV contains Frequency Domain Adaptive Filtering algorithms. Further Section V discusses the simulated results for echo cancellation and the last section VI concludes the paper with performance evolution.

II. THE PROCESS OF ECHO CANCELLATION SYSTEM

In telecommunication networks, one source of echo is electrical and the other echo source is acoustic. The electrical echo is due to the impedance mismatch at the hybrid of a Public Switch Telecom Network (PSTN), exchange lines. If a communication is simply between two fixed telephones, then only the electrical echo occurs. However, the development of hands-free teleconferencing systems gave rise to another kind of echo known as acoustic echo. The acoustic echo occurs due to the coupling between the loudspeaker and microphone.

Acoustic echo cancellation is the technique that makes it possible for remotely located groups to interact using audio link as if they are in the same room. AEC is needed whenever an unrestricted, full-duplex conversation is desired between different rooms and the conferencing microphone is able to pick up the audio signal coming from the loudspeaker(s).

An echo canceller is also defined as a device which detects and removes the echo of the signal from the far end after it has echoed on the local end’s equipment. In case of circuit switched long distance networks, echo cancellers reside in the telephone switching offices that connect to the long distance telephone network. These echo cancellers remove electrical echoes made noticeable by delay in the long distance network.

The effectiveness of an echo canceller is measured as Echo Return Loss Enhancement (ERLE) which is a decibel (dB) measurement of the echo canceller. This measures how much loss the echo canceller adds to the transmitted signal to remove echo from the microphone audio.

Echo Return Loss (ERL) is a measurement in dB of acoustic echo loss through decay and absorption as the audio travels from the speaker system to microphone either directly or through reflection. This measurement is affected by the levels at the output of the speakers and the microphone input sensitivity. ERL is affected by the amount of gain at the amplifier as well as by room acoustics [3]-[5].

III. Least Mean Square (LMS) Algorithm

There are many algorithms that could be used in echo cancellation and adaptive filtering process in which the LMS algorithm is the most commonly used algorithm in echo cancellation because it is powerful enough to accomplish the system requirements and is also relatively simple compared to the other algorithms.

The Least Mean Square algorithm was first developed by Widrow and Hoff in 1959 during their studies of pattern recognition. From then it has become one of the most widely used algorithms in adaptive filtering. The step size value in the LMS algorithm is an important value in determining the performance of the echo cancellation. Step size must be chosen between zero and twice the inverse of total input power.

The main function of LMS algorithm is to minimize the Mean Square Error (MSE) between the echo and its output.

If x(n) is the input vector of time delayed input values $x(n)=[x(n),x(n-1)x(n-2)\ldots x(n-N+1)]^T$ and the the vector $w(n)=[w_0(n)w_1(n)w_2(n)w_{N-1}(n)]T$ represent the coefficient of the adaptive FIR filter tap weight vectors. The step size parameter μ controls the influence of the updating factor. With each iteration of the LMS algorithm, the Filter tap weights of adaptive filter are updated according to the following relation [6]-[8].

$$\mathbf{w}(n+1) = \mathbf{w}(n)+\mu e(n)\mathbf{x}(n) \quad (1)$$

where

- **w**(n) are the present filter coefficients,
- **w**(n+1) are the future adaptive filter coefficients
- x(n) are the input values
- e(n) is the error value.
- μ is the step size

The value of μ determines the convergence or divergence speed and precision of the adaptive filter coefficients. In Eq.(1) If μ is large, the filter will converge fast, but could diverge if μ is too large. If μ is small, the filter will converge slowly, which is equivalent to the algorithm having "long" memory, an undesirable quality. Every application will have a different step size that needs to be adjusted. When choosing a μ value, that needs to be balanced between speed convergence and the MSE. The μ value is decided through trial and error so that speed at which the adaptive filter learns and the excess MSE is obtained within application requirements. The μ values differs from simulation to real-time because of the inherited differences between the systems. The system equations used by the adaptive filter to determine the error and output of the filter are listed below. A very important part of the algorithm is to updating the filter coefficients. The step size parameter μ must be chosen accurately to ensure the filter converges. Updating the filter coefficients is important because this is the part of the code that governs how well the filter will converge to the desired response. The steps to update filter coefficient is as fallows [9]-[11].

1. The output y(n) of the adaptive filter is found though convolution of the adaptive filter coefficients **w**(n) with the input signal **x**(n).

Output of adaptive filter:

$$y(n)=\sum_{n=0}^{M-1} w(k)*x(n-k)=w^T(n)x(n) \quad (2)$$

2. The error signal e(n) is created from the subtraction of the desired signal from the output of the adaptive filter, Error Signal :

$$e(n) = d(n) - y(n) \quad (3)$$

3. The tap weights of the FIR vectors are updated in preparation for the next iteration, by equation

4. $\mathbf{w}(n+1) = \mathbf{w}(n)+2\mu e(n)\mathbf{x}(n)$ (4)

β is the delay in samples and α is the attenuation constant [13][14].

IV. Frequency domain Adaptive filtering

The Frequency Domain Adaptive Filter (FDAF) uses a fast convolution technique to compute the output signal and filter updates. It has also improved convergence performance through frequency-bit step size normalization. By conserving some initial parameters for the filter it is observed that how well the far-end speech is cancelled in the error signal. Frequency Domain Adaptive Filtering provides several advantages over its time domain counterpart. Besides being able to perform the filter convolution by a multiplication in frequency domain, also the length of the adaptive filter are effectively decimated by the transformation.

It is often the case that signals are represented in the frequency domain to enable the use of discrete transforms that reduce the processing required in signal processing applications for example convolution. Although there are a few transforms between the two domains, the Fourier Transform is the most widespread. The advantages of the Fourier Transform over other transforms include its high efficiency, adequate representation of data and its distortion less performance when transmitted over linear systems.

4.1 *Frequency Domain Adaptive LMS Filtering*

The implementation of the LMS filter in the frequency domain can be accomplished simply by taking Discrete Fourier Transform (DFT) of both the input data *x(n)* and the desired signal *d(n)*. The advantage of doing this is for the fast processing of the signal using Fast Fourier

Transform algorithm. The procedure in various steps is as follows.

The signals are processed by block-by-block format, that is x(n) and d(n) are sequenced into blocks of length M so that

$$x_i(n) = x(iM+n), \qquad d_i(n) = d(iM+n)$$

$$n = 0,1,......M-1; i = 0,1.....M-1 \quad (5)$$

The values of i^{th} block of signals $x_i(n)$ and $d_i(n)$ are Fourier transform using the DFT to find $X_i(k)$ and $D_i(k)$ where

$$X_i(k) = DFT\{x_i(n)\} = \sum_{n=0}^{M-1} x_i(n) e^{-j\frac{2\pi nk}{M}} \quad (6)$$

$$D_i(k) = DFT\{d_i(n)\} = \sum_{n=0}^{M-1} d_i(n) e^{-j\frac{2\pi nk}{M}},$$

$k=0,1,\ldots,M-1$ (7)

During the i^{th} block processing the output in each frequency bit of the adaptive filter is computed by

$Y_i(k)=W_{i,k}X_i(k)$: $k=0,1,2\ldots\ldots M-1$ (8)

Where $W_{i,k}$ is the k^{th} frequency bit corresponding to the i^{th} update, The error in the frequency domain is

$E_i(K)=D_i(k)-Y_i(k)$; $k=0,1,2\ldots\ldots M-1$ (9)

The system output is given by

$$y_i(n)=y(Im+n)=IDFT\{Y_i(k)\}$$

$$= \frac{1}{M}\sum_{k=0}^{M-1} Y_i(k) e^{j\frac{2\pi nk}{M}}$$

$n=0,1,2\ldots\ldots M-1$ (10)

To update the filter coefficients we have used fallowing LMS recursion:

$$W_{i+1}=W_i+2\mu E_i X_i \quad (11)$$

V. Simulation Results

The Frequency Domain Adaptive Filter Algorithm is simulated using Matlab, Figure 2 shows the room impulse response which describe the acoustics of the loudspeaker-to-microphone signal where the speaker phone is located. We have taken a long finite impulse response filter to describe these characteristics.

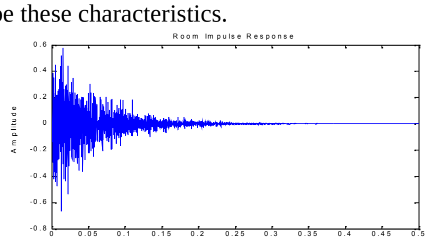

Fig. 2. Room Impulse Response

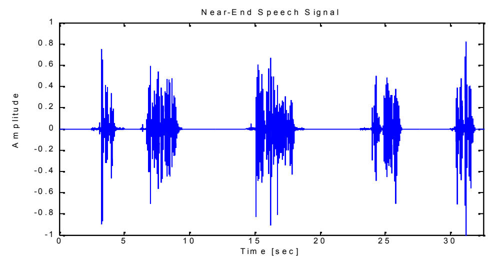

Fig. 3. Near-End Speech Signal

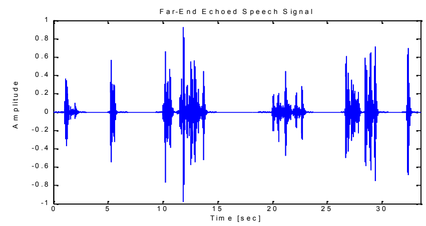

Fig. 4. Far-End Echoed Speech Signal

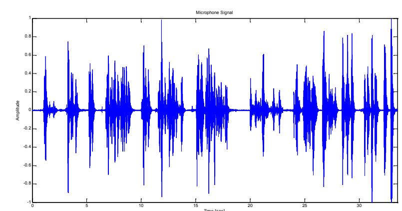

Fig. 5. Microphone Signal

Figure 3 shows the Near-End Speech Signal Located near the system microphone. Figure 4 shows the Far-End Speech Signal and Figure 5 Shows the Microphone Signal containing both the near-end speech and the far-end speech that has been echoed. For performance evaluation of echo canceller and to measure the effectiveness of an echo canceller we have considered Echo Return Loss Enhancement (ERLE), Figure 6 shows the output of acoustic echo canceller for various values of step size. For FDAF with μ=0.025 some echo is still present in the output but if μ is increased to μ = 0.4 then it is observed that we can remove more echo. We have observed the effect of different step size value and found that, faster convergence is achieved by using a larger step size value.

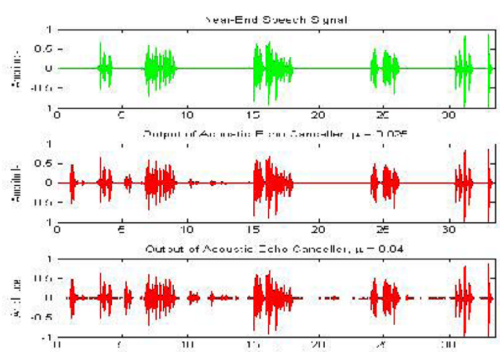

Fig. 6. Effects of different step size value

Comparison between Step Size and ERLE for LMS and FDAF is shown in table 1.

TABLE 1

Step Size in LMS	Max ERLE in dB	Step Size in FDAF	Max ERLE in dB
0.025	14.68	0.025	36.8
0.04	18.1	0.04	38.5

From the above Table1 it is clear that for step size of 0.025, ERLE for LMS filter is 14.68 dB where as for FDAF it is 36.8 dB that is we can attenuate more echo using FDAF but if we increase step size up to 0.04 then with LMS filtering we can attenuate echo up to 18.1 dB. For LMS algorithm 0.04 is the highest step size for which we have achieved the maximum value of ERLE Echo, where as for FDAF 0.04 is the highest step size for which we have achieved the maximum value of ERLE is 38.5.

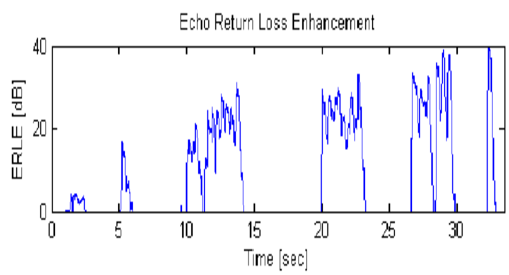

Fig. 7. Echo Return Loss Enhancement Comparison

VI. Conclusion

An acoustic echo cancellation system based on a Frequency Domain Adaptive Filtering algorithm is designed and tested through Matlab and it is found that when the LMS algorithm is adapted in the time domain it requires large memory and results in a significant increase in the computational complexity of the algorithm whereas Frequency Domain Adaptive Algorithm (FDAF) proposed work, provides a better result with Echo Return Loss Enhancement Comparison as shown in figure 7.

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