

Acoustic Echo Cancellation Using Modified Normalized Least Mean Square Adaptive Filters

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Abstract

Adaptive filtering techniques are used extensively in various digital signal processing applications such as acoustic echo cancellation (AEC), adaptive noise cancellation, channel identification and adaptive channel equalization. This paper focuses on developing an efficient acoustic echo removal algorithm, capable of providing effective results by implementation of adaptive filtering techniques to reduce undesirable echo in communication systems. The developed echo canceller comprises of a normalized least mean square (NLMS) adaptive filter and a double-talk detector (DTD) which plays an essential role in AEC. The AEC algorithms employed in this work are implemented using MATLAB application and results of simulations are presented.

Keywords: Adaptive filters, Echo, NLMS, AEC, DTD.

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for enhanced voice quality and services over wireless net

1. INTRODUCTION

Multiple access communication systems frequently depend on the orthogonality of the different signals by separating them in time or frequency to prevent interference. In a full-duplex transmission, time-compression multiplexing (TCM) and frequency-division multiplexing (FDM) can be employed. Echo cancellation is a more efficient technique that enables concurrent transmission in two directions using the same frequency band. This reduces the bandwidth requirements approximately in half relative to TCM and FDM. Acoustic echo arises when an audio signal is reverberated in a real environment, resulting in the original intended signal plus time-delayed and distorted version of an original signal [1]. Existence of acoustic echo in speech communication systems results in signal interference and hinders communication [2].

In data communication, echo can lead to high transmission error. Furthermore, in applications such as wireless telephoning and teleconferencing systems, echo in extreme condition can make communication difficult and sometimes impossible. To experience excellent communication, subscribers demand

works. This leads to the need for elimination of reverberation effect in the system and give rise to a new and key technology termed echo cancellation, which can provide near wire line voice quality across a wireless network [3]. This paper focuses on the occurrence of acoustic echo in telecommunication systems.

1.1. ACOUSTIC ECHO

In audio-conferencing and video teleconferencing applications, speakerphones and hands-free cellular phones which allows full-duplex communication without having to hold the phone have been used extensively. As shown in Fig1, the speech signal from the far-end caller is broadcast by the speakerphone or the hands-free cellular phone of the near-end. The waves from the speech signal are fed back to the remote user (far-end caller) through the near-end microphone after reflection from the wall, floor and other objects inside the room. The signal at the microphone consists of the original intended signal, time-delayed and distorted version of an original signal. The reverberation of the speech signal is called an echo [1, 3]. The echo then creates a feedback loop and the far-end caller hears an echo of his or her own voice. The acoustic echo disturbs the conversation and reduces the quality of the system [4]. The problem can be addressed by the acoustic echo cancellation (AEC) to stop the feedback and allow full-duplex communication [1-4].

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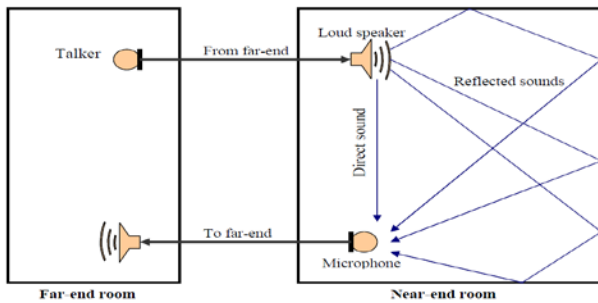


Fig. 1: A teleconference system with echo paths of room

2. IMPLEMENTATION OF ACOUSTIC ECHO CANCELLATION (AEC)

In a full-duplex transmission, time-compression multiplexing (TCM) and frequency-division multiplexing (FDM) can be employed to handle the acoustic echo problem in teleconference systems. Furthermore, voice switches and directional microphones can also be used but these methods placed physical restriction on the speaker. Echo cancellation is a more efficient technique to remove the echo in the system. AEC enhances greatly the quality of the audio signal of the hands-free communication system.

Some echo cancellation algorithms are used for this purpose and these algorithms process the signals following the basic steps as follow:

Step 1: Estimate the characteristics of echo path $h(n)$ of the room: $\hat{h}(n)$

Step 2: Create a replica of the echo signal: $\hat{y}(n)$

Step 3: Echo is then subtracted from microphone signal (includes near-end and echo signals) to obtain the desired signal: $Desired\ Signal = d(n) - \hat{y}(n)$.

Adaptive filter is a good supplement to achieve a good replica because the echo path is usually unknown and time-varying. The Fig. 2 illustrates required steps by which AEC can be implemented using adaptive filter.

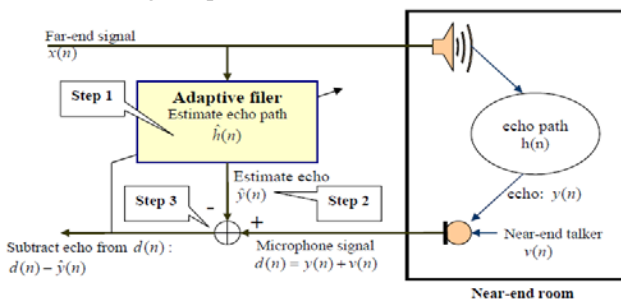


Fig. 2: Implementation of Acoustic Echo Canceller using adaptive filter

In the modern digital communication system such as: Public Switched Telephone Network (PSTN), Voice over IP (VoIP), Voice over Packet (VoP) and cell phone networks; the application of AEC is highly essential because it gives better quality of

service (QoS).

2.1 COMPONENTS OF AN ACOUSTIC ECHO CANCELLER (AEC)

This paper presents an implementation of a basic acoustic echo canceller consisting of the blocks of double talk detection, normalized least mean square (NLMS) and non-linear processor. Fig. 3 depicts the block diagram of the fundamental components of AEC.

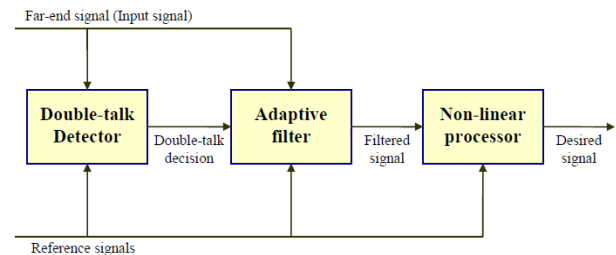


Fig. 3: Block diagram of AEC

2.1.1. ADAPTIVE FILTERS

In Acoustic Echo Cancellation (AEC), the adaptive filter plays the main role to adapt the filter tap weight in order to overcome the echo problem. There are different types of algorithms that can be used for this purpose, namely: Least Mean Square (LMS), Normalized Least Mean Square (NLMS), Recursive Least Square (RLS) and Affine Projection Algorithm (APA) [5]. The LMS is a widely used algorithm for adaptive application such as channel equalization and echo cancellation. This algorithm is the simplest when compared to NLMS and RLS algorithm. It has been observed that when considering the attenuation values and the computational complexity for each algorithm, the NLMS algorithm is the choice for the real time acoustic echo cancellation system. Furthermore, it does not require a prior knowledge of the signal values to ensure stability [3, 6, 7].

2.1.2. DOUBLE-TALK DETECTOR (DTD)

In Acoustic Echo Cancellation, the most difficult problem to handle is the presence of Double-talk. Double-talk occurs when far-end talker and near-end talker speak at the same time; as a result, the far-end speech signal is corrupted by near-end signal. To solve this problem, the Double-talk Detector is introduced. The task of the DTD is to freeze the adaptation step during filtering algorithm in case of the presence of near-end speech to avoid the divergence of adaptive algorithm. Without the DTD, when the near-end talking would make the system estimation process fail and produce extremely erroneous results [2, 4]. This is depicted in Fig. 4.

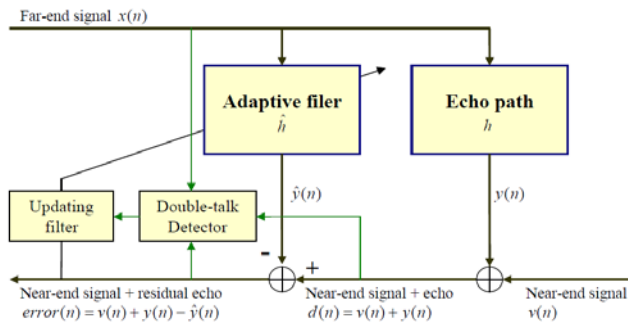


Fig 4: Double-talk detector with AEC

When the near-end speech is not present, $v(n) = 0$ then the adaptive algorithm will quickly converge to an estimate echo path. This is the best case of canceling echo. Furthermore, when near-end speech is present, $v(n) \neq 0$ then the signal could influence the adaptation of the filter and leads to divergence. Therefore, the process of adaptive algorithm will be incorrect and the echo cannot be completely removed.

By implementing the DTD and updating filter blocks, the DTD will estimate the statistic decision which may depend on far-end speech, near-end speech and error signal. After that, it will compare to the threshold to make the DTD decision to control the updating filter (freeze the adaptation or not). The updating filter block permits the update of weight vector or not. There are different DTD algorithms such as the Geigel algorithm, the Cross-Correlation algorithm and the Normalized Cross-Correlation (NCC) algorithm. In this paper, NCC algorithm is employed used.

2.1.3. Non-linear processor (NLP)

The nonlinear processor, (NLP), is a signal processing circuit or algorithm placed in the speech path after the NLMS adaptive filter used for echo cancellation to further suppress or eliminate residual echo signals that cannot be removed completely by an echo canceller [2]. Dual NLP configurations are employed in this work. This dual NLP structure has significant advantages compared to single NLP in a situation of the wrong double-talk detection.

3. METHODOLOGY

A modified NLMS algorithm is employed in this AEC system. The coefficients $w(n)$ are updated as given in [4, 6]. By using the normalized step-size parameter in Least Mean Square algorithm, the resulting algorithm is known as Normalized Least Mean Square (NLMS) algorithm. The step-size for computing the update weight vector is,

$$\mu(n) = \frac{\beta}{c + x^T(n) x(n)} \quad (1)$$

where $\mu(n)$ is step-size parameter at sample n , β is normalized step-size usually $\frac{1}{2}$ and c is safety factor (small positive constant) introduced to prevent division by zero if $x^T(n) x(n)$ is very small..

By replacing μ with $\mu(n)$ in the equation for updating the weight vector in LMS algorithm given as

$$w(n+1) = w(n) + \mu e(n) x(n) \quad (2)$$

A new algorithm is achieved known as the Normalized Least Mean Square (NLMS). The weight vector update now is:

$$w(n+1) = w(n) + \mu(n) e(n) x(n) \quad (3)$$

Substituting (1) into (3)

$$w(n+1) = w(n) + \frac{\beta}{c + x^T(n) x(n)} e(n) x(n) \quad (4)$$

where $w(n)$ is the weight vector, $x(n)$ is the input signal. Equation (4) is the weight vector updating equation for NLMS algorithm.

The flow chart of the AEC algorithm is a summary of all the steps needed in the simulation process. This is shown in Figure 5.

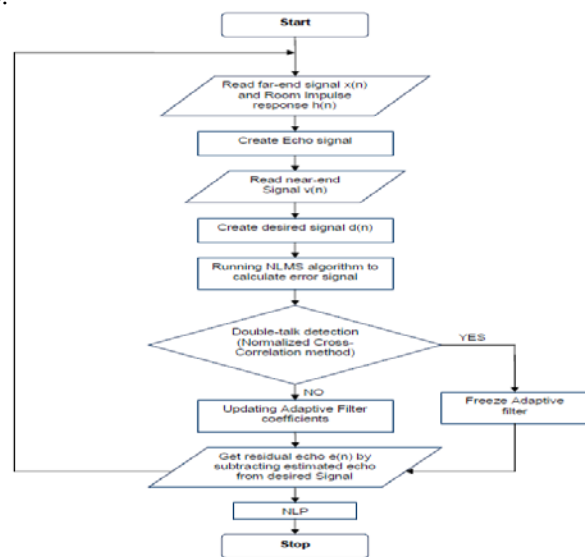


Fig 5: Flowchart of acoustic echo cancellation algorithm

4. SIMULATION RESULTS

The NLMS algorithm is applied in echo canceller to cancel

effect of echo in the system. The algorithm employed has high convergence rate and its computational complexity is less than the conventional algorithm. The measure of the echo canceller efficiency is expressed by Echo Return Loss Enhancement (ERLE) factor which is defined as:

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

$$= \frac{E[d^2(n)]}{E[e^2(n)]} \quad (5)$$

where $ERLE$ is measured in dB, $d^2(n)$ is the microphone signal's power and $e^2(n)$ is the residual error signal's power.

ERLE depends on the algorithm used for the adaptive filter. It considers the convergence time and near-end attenuation. Algorithms are simulated in MATLAB, with input speech signal sampled at a frequency of 8000Hz. In order to examine the efficiency of the acoustic echo canceller, it was necessary to use a room acoustic impulse response which was recorded using a white noise excitation signal.

The far-end signal is shown in Fig. 6. It is scaled in order to produce the echo signal, $r(n)$, which is depicted in Fig. 7. The echo signal was produced when the far-end signal, $x(n)$, passed through the echo path, h .

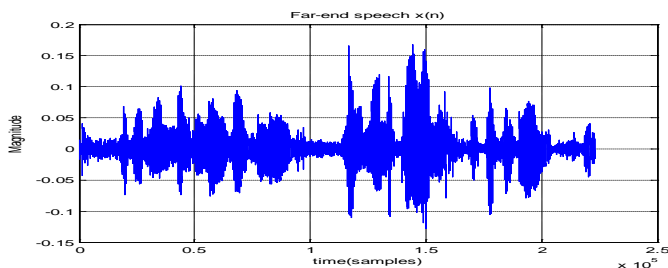


Fig.6: Far-end Signal

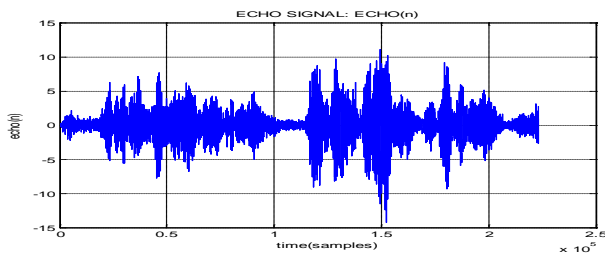


Fig. 7: Echo Signal

The echo signal was added to the near-end signal, $v(n)$ and white noise which was the excitation signal, in order to produce the desired signal, $d(n)$, which became the input for the adaptive filter. The plot of the near-end signal, $v(n)$, is illus-

trated in Fig. 8, the plot of the white noise is shown in Fig. 9 and the plot of the desired signal, $d(n)$ is depicted in Figure 10.

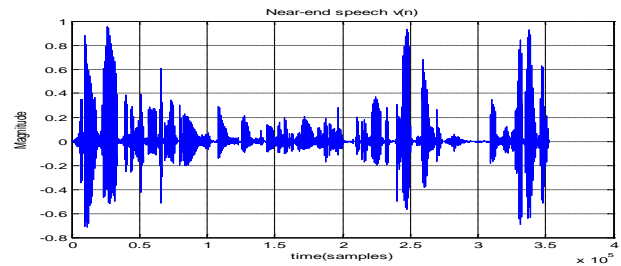


Fig. 8: Near-end Signal

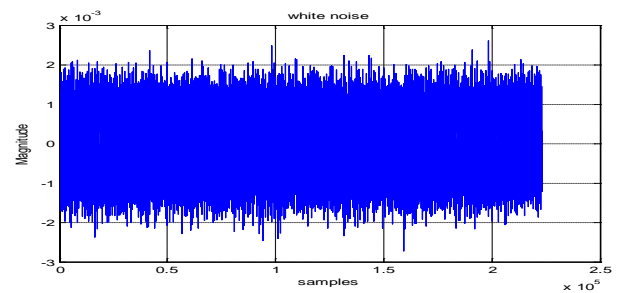


Fig. 9: White Noise Signal

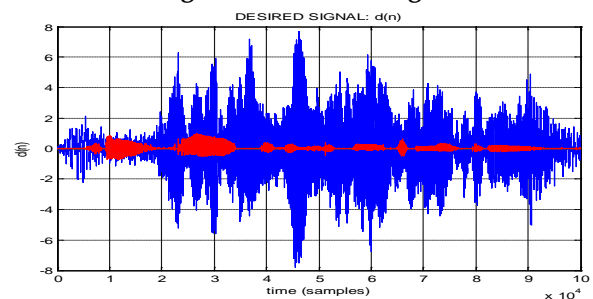


Fig. 10: Desired Signal

The desired signal, $d(n)$ is passed through the adaptive filter and the double talk detector. For the purpose of adaptive filtering the NLMS algorithm was used during the simulation. The algorithm used the normalized cross correlation algorithm for double talk detection. The double talk detection is depicted in Fig. 11.

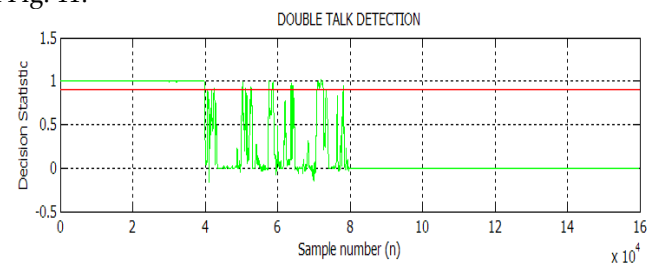


Fig. 11: Double –talk detection NLMS algorithm

In Fig. 11, the decision statistic $\xi_{NCC}(n)$ [green line] is compared to the threshold T [red line]. If $\xi_{NCC}(n) > T$, then only far-end talk is present and if $\xi_{NCC}(n) < T$, double-talk or only near-end talks or both side are silent. The mean square error (MSE) curve is illustrated in Fig. 12. This measures how much the algorithm minimizes the echo.

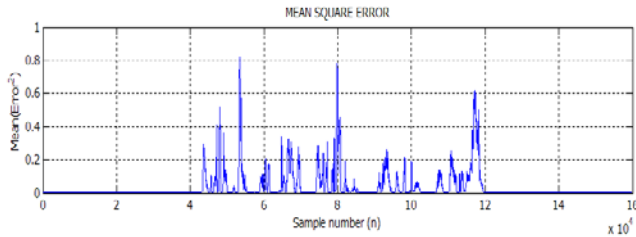


Fig. 11: Mean square error

5 CONCLUSION

In this paper, a modified NLMS algorithm is presented as software solution for the problem of echoes in the teleconference. Also, a robust DTD is employed for performance improvement of acoustic echo canceller. The simulation results showed that AEC performance meets the standard requirement for convergence time and the ERLE. The modified NLMS algorithm is simple and efficient than the conventional NLMS algorithm. Furthermore, results confirmed that an acoustic echo canceller using the modified NLMS algorithm can reliably suppress echoing and enable smooth conversation in teleconference.

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