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A contingent adaptive echo cancellation method by actualizing different algorithms

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Abstract— An Echo can be defined as the repetition of a waveform due to the reflection from points through which it propagates. In the process of Communication, it degrades the quality of service. Echo that result from a feedback path set up between the speaker and microphone in a mobile phone are known as Acoustic Echo. Echo that result from an impedance mismatch at telephone exchange hybrids between 2-line & 4-wire line are known as Hybrid Echo. This paper is an attempt at the cancellation of echo and comparison with various types of LMS Algorithms, which has shown positive outcome towards the process of Echo Cancellation using Adaptive Filters.

Index Terms— Adaptive Filters, Discrete Time Signals, Echo Cancellation, LMS Algorithm, NLMS Algorithm, Transversal FIR Filters, VSLMS Algorithm, VSNLMS Algorithm.

1. Introduction

In today's Telecommunication systems, Adaptive echo cancellation has a very common occurrence in the establishment of a communication link. It basically occurs when an audio source and sink has a full duplex operation mode, an example for which is a hands-free loudspeaker telephone. The signal interference caused by adaptive echo is distracting to both the users which are communicating and causes a mass reduction in the quality of the communication.

This paper focuses on the use of various techniques and algorithms of adaptive filtering, employing discrete signal processing in MATLAB, to reduce this unwanted echo and thus increase communication quality.

Adaptive filters are a class of filters that iteratively alter their parameters in order to minimize a function of the difference between a desired audio signal and their system output signal. In the case of adaptive echo in telecommunications, 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.

An example of a hands-free telephone system:

In this scenario the system has both an active loudspeaker and microphone input operating simultaneously. When a signal is received by the system, it is output through the loudspeaker into an acoustic environment. This signal is reverberated within the environment and returned to the system via the microphone input. These reverberated signals contain time delayed images of the original signal, which are then returned to the original sender. The occurrence of adaptive echo in speech transmission causes signal interference and reduced quality of communication.

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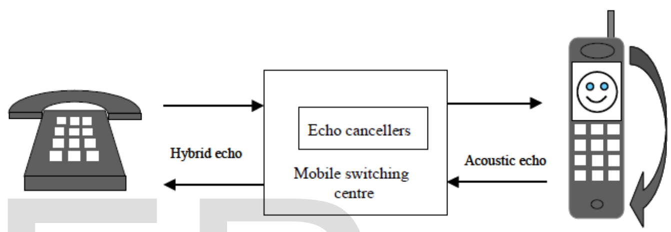

Fig.1 Illustration of echo in a mobile to landline system

The problem stated above is faced by each and every person who uses a communication system in his/her daily life.

So, in order to reduce this problem to optimum minimal and generating an Efficient Generalized Algorithm for the echo cancellation motivated us for cancellation of Echo with the help of basic Digital Signal Processing. In general, echoes with appreciable amplitude and a delay of more than 1 ms are noticeable. However, echoes become increasingly annoying and objectionable with the increasing round-trip delay and amplitude in particular for delays of more than 20 ms. Hence echo cancellation is an important aspect of the design of modern telecommunication systems such as conventional wire line telephones, hands-free phones, cellular mobile (wireless) phones, or teleconference systems. There are two types of echo in a telephone system:

(a) Acoustic echo due to acoustic coupling between the speaker and the microphone.

(b) Electrical line echo due to mismatch at the hybrid circuit connecting a 2-wire subscriber line to a 4-wire truck line in the public switched telephone network.

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 cost function of the adaptive algorithm.

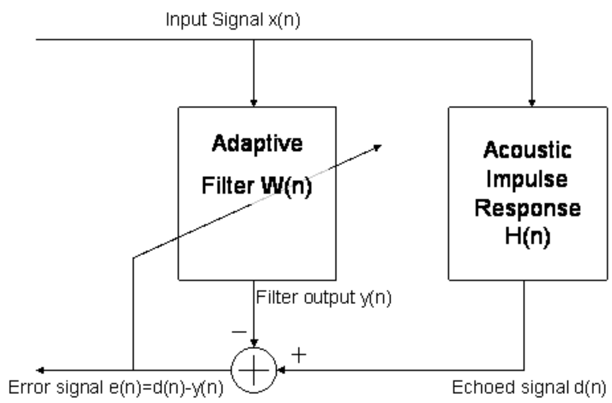

Fig. 2 Block diagram of Adaptive Echo Cancellation System

2. Background Theory

In order to understand the content presented in this paper, it is first necessary to provide some background information regarding digital signal theory. It will start out rather elementary and then progress to more complicated matters. Later chapters will build upon this theoretical basis in the derivation and implementation of the adaptive filtering techniques used in acoustic echo cancellation.

2.1 Discrete Time Signals

Real world signals, such as speech are analog and continuous. In modern day communication systems these signals are represented electronically by discrete numeric sequences. In these sequences, each value represents an instantaneous value of the continuous signal. These values are taken at regular time periods, known as the sampling period, Ts.

For example, consider a continuous waveform given by x(t). In order to process this waveform digitally we first must convert this into a discrete time vector. Each value in the vector represents the instantaneous value of this waveform at integer multiples of the sampling period. The values of the sequence, x(t) corresponding to the value at n times the sampling period is denoted as x(n).

$$x(n) = x(nTs) \quad (1)$$

2.2 Transversal FIR Filters

A filter can be defined as a piece of software or hardware that takes an input signal and processes it so as to extract and output certain desired elements of that signal. This paper shall be contained to adaptive filtering method.

The characteristics of a transversal FIR filter can be expressed as a vector consisting of values known as tap weights. It is these tap weights which determine the performance of the filter. These values are expressed in column vector form as,

$$w(n) = [w_0(n)w_1(n)w_2(n) \ldots \ldots w_{N-1}(n)]\,T \quad (2)$$

The number of elements on this impulse response vector corresponds to the order of the filter, denoted in this paper by the character N.

The output of the FIR filter at time n is determined by the sum of the products between the tap weight vector, w(n) and N time delayed input values. If these time delayed inputs are expressed in vector form by the column vector x(n) = [x(n) x(n-1) x(n-2) .. x(n-N+1)]T, the output of the filter at time n is expressed by equation 2.2. Throughout paper the vector containing the time delayed input values at time n is referred to as the input vector, x(n). In adaptive filtering

the tap weight values are time varying so for each at each time interval a new FIR tap weight vector must be calculated, this is denoted as the column vector

$$w(n) = [w_0(n)w_1(n)w_2(n) \ldots \ldots w_{N-1}(n)]\,T$$

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

2.3 Adaptive Filters

Figure 3 shows the block diagram for the adaptive filter method utilized where w represents the coefficients of the FIR filter tap weight vector, x(n) is the input vector samples, z-1 is a delay of one sample periods, y(n) is the adaptive filter output, d(n) is the desired echoed signal and e(n) is the estimation error at time n.

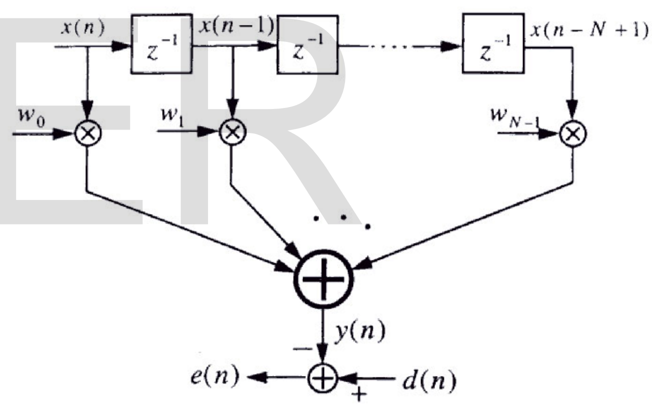

Fig. 3 Adaptive Filter Design

The aim of an adaptive filter is to calculate the difference between the desired signal and the adaptive filter output, e(n). This error signal is fed back into the adaptive filter and its coefficients are changed algorithmically in order to minimize a function of this difference, known as the cost function. In the case of acoustic echo cancellation, the optimal output of the adaptive filter is equal in value to the unwanted echoed signal. When the adaptive filter output is equal to desired signal the error signal goes to zero. In this situation the echoed signal would be completely cancelled and the far user would not hear any of their original speech returned to them.

3. METHODOLOGY

3.1 Least Mean Square (LMS) Algorithm

It is well known algorithm and is widely used due to its computational simplicity. It has a low memory consumption and has an ease of implementation. 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 additional N for the scalar by vector multiplication).

The LMS Algorithm is performed using the following steps:

i) The output of the FIR filter,y(n) is calculated using
$$w(n+1) = w(n) + 2\mu e(n)x(n) \quad (3)$$
ii) The value of the errorestimation is calculated using
$$e(n) = d(n) - y(n) \quad (4)$$
iii) The tap weights of the FIR vector are updated in the preparation for the next iteration using
$$w(n+1) = w(n) + 2\mu e(n)x(n) \quad (5)$$

3.2 Normalised Least Mean Square(NLMS) Algorithm

The normalized least mean square algorithm (NLMS) is an extension of the LMS algorithm which bypasses this issue by selecting a different step size value, μ(n), for each iteration of the algorithm. This step size is proportional to the inverse of the total expected energy of the instantaneous values of the coefficients of the input vector x(n). Each iteration of the NLMS algorithm requires 3N+1 multiplications, this is only N more than the standard LMS algorithm and this is an acceptable increase considering the gains in stability and echo attenuation achieved.

The NLMS Algorithm is performed using the following steps:

i) The output of the FIR filter, y(n) is calculated using
$$y(n) = \sum_{i=1}^{N-1} w(n)x(n-i) = w^T x(n) \quad (6)$$
ii) The value of the error estimation is calculated using
$$e(n) = d(n) - y(n) \quad (7)$$
iii) The step size value for the input vector is calculated using
$$\mu(n) = \frac{1}{x^T x(n)} \quad (8)$$
iv) The tap weights of the FIR vector are updated in preparation for the next iteration using
$$w(n+1) = w(n) + 2\mu e(n)x(n) \quad (9)$$

3.3 Variable Step Size LMS (VSLMS) Algorithm

In the Variable Step Size Least Mean Square (VSLMS) algorithm the step size for each iteration is expressed as a vector, μ(n). Each element of the vector μ(n) is a different step size value corresponding to an element of the filter tap weight vector, w(n). With ρ=1, each iteration of the VSLMS algorithm requires (4N+1) multiplication operations.

The VSLMS Algorithm is performed using the following steps:

i) The output of the FIR filter, y(n) is calculated using
$$y(n) = \sum_{i=1}^{N-1} w(n)x(n-i) = w^T x(n) \quad (10)$$
ii) The value of the error estimation is calculated as the difference between the desired output and the filter output.
$$e(n) = y(n) - d(n) \quad (11)$$
iii) The gradient, step size and filter tap weight vectors are updated using the following equations in preparation for the next iteration,
$$\begin{aligned}
&For\ i = 0,1,\ldots N-1\\
&g_i(n) = e(n)x(n-i)\\
&g(n) = e(n)x(n)\\
&\mu_i(n) = \mu_i(n-1) + \rho g_i(n)g_i(n-1)\\
&if\ \mu_i(n) > \mu_{max}, \mu_i(n) = \mu_{max}\\
&if\ \mu_i(n) < \mu_{min}, \mu_i(n) = \mu_{min}\\
&w_i(n+1) = w_i(n) + 2\mu_i(n)g_i(n) \quad (12)
\end{aligned}$$

3.3 Variable Normalized Step Size LMS (VSNLMS) Algorithm

The VSLMS algorithm still has the same drawback as the standard LMS algorithm in that to guarantee stability of the algorithm, a statistical knowledge of the input signal is required prior to the algorithms commencement. In the VSNLMS algorithm the upper bound available to each element of the step size vector, μ(n), is calculated for each iteration. As with the NLMS algorithm the step size value is inversely proportional to the instantaneous input signal energy.

The VSNLMS Algorithm is performed using the following steps:

i) The output of the FIR filter, y(n) is calculated using
$$y(n) = \sum_{i=1}^{N-1} w(n)x(n-i) = w^T x(n) \quad (13)$$
ii) The value of the error estimation is calculated as the difference between the desired output and the filter output.
$$e(n) = y(n) - d(n) \quad (14)$$
iii) The gradient, step size and filter tap weight vectors are updated using the following equations in preparation for the next iteration,
$$\begin{aligned}
&For\ i = 0,1,\ldots N-1\\
&g_i(n) = e(n)x(n-i)\\
&g(n) = e(n)x(n)\\
&\mu_i(n) = \mu_i(n-1) + \rho g_i(n)g_i(n-1)\\
&if\ \mu_i(n) > \mu_{max}, \mu_i(n) = \mu_{max}\\
&if\ \mu_i(n) < \mu_{min}, \mu_i(n) = \mu_{min}\\
&w_i(n+1) = w_i(n) + 2\mu_i(n)g_i \quad (15)
\end{aligned}$$

With $\rho = 1$, each iteration of the VSNLMS algorithm requires 5N+1 multiplication operations.

4. SOFTWARE SIMULATION

Each of the adaptive filtering algorithms outlined in Section 3

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were implemented using MATLAB. In each simulation the echoed signal was generated by defining an appropriate impulse response then convolving this with a vocal input wav file.

4.1 LMS Algorithm

The figure shows the desired signal, adaptive output signal, estimation error and cost function (MSE) for the LMS algorithm with vocal input, FIR filter order of 1000 and step size of 0.007 (determined empirically). The MSE shows that as the algorithm progresses the average value of the cost function decreases, this corresponds to the LMS filters impulse response converging to the actual impulse response, more accurately emulating the desired signal and thus more effectively cancelling the echoed signal.

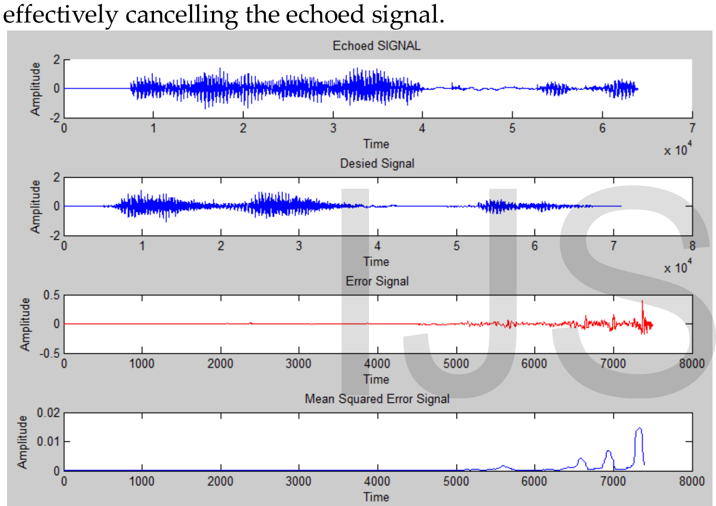

Fig. 4 LMS algorithm outputs for vocal input, N=7500, μ=0.007.

The success of the echo cancellation can be determined by the ratio of the desired signal and the error signal.

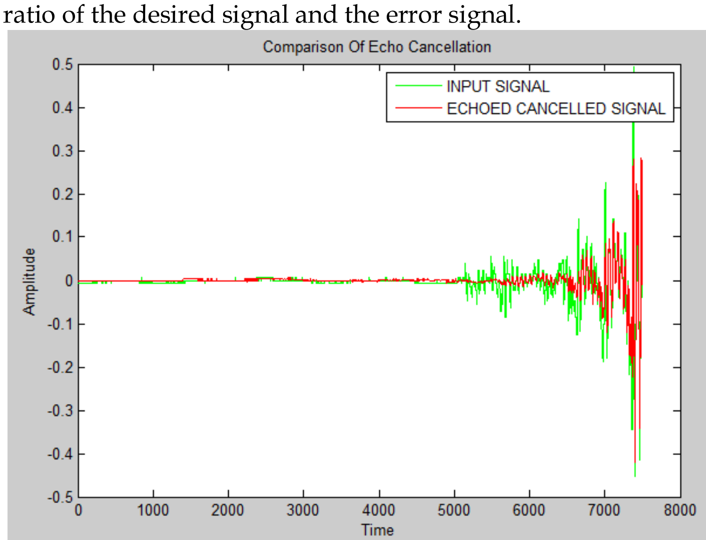

Fig. 4 Echo Cancellation using LMS Algorithm

4.2 NLMS Algorithm

The Normalized LMS algorithm was simulated using MATLAB, it is also the algorithm used in the real time echo cancellation system.

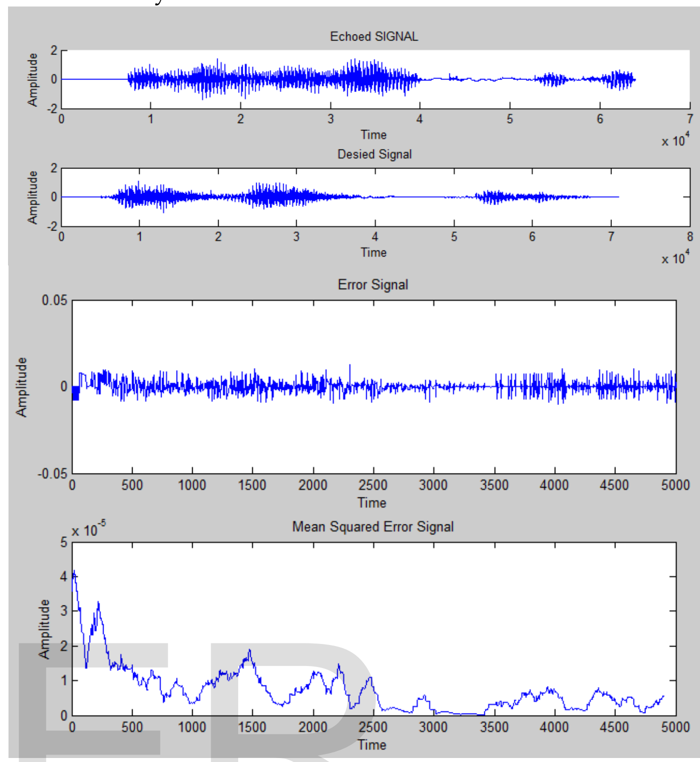

Fig. 5 NLMS algorithm outputs for vocal input, N=5000, μ=0.007.

4.3 VSLMS Algorithm

The VSLMS algorithm was simulated using MATLAB, it is also the algorithm used in the real time echo cancellation system.

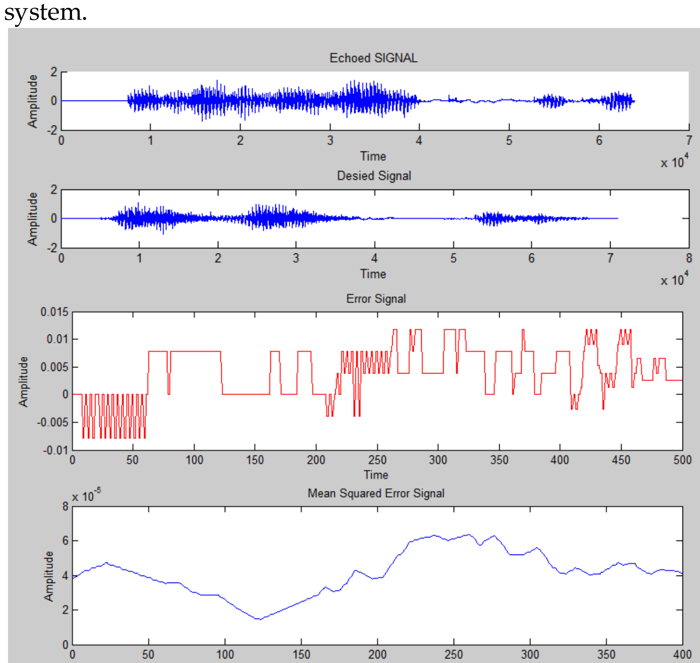

Fig. 6 VSLMS algorithm outputs for vocal input, N=500.

4.4 VSNLMS Algorithm

The VSNLMS algorithm was simulated using MATLAB, it is also the algorithm used in the real time echo cancellation system.

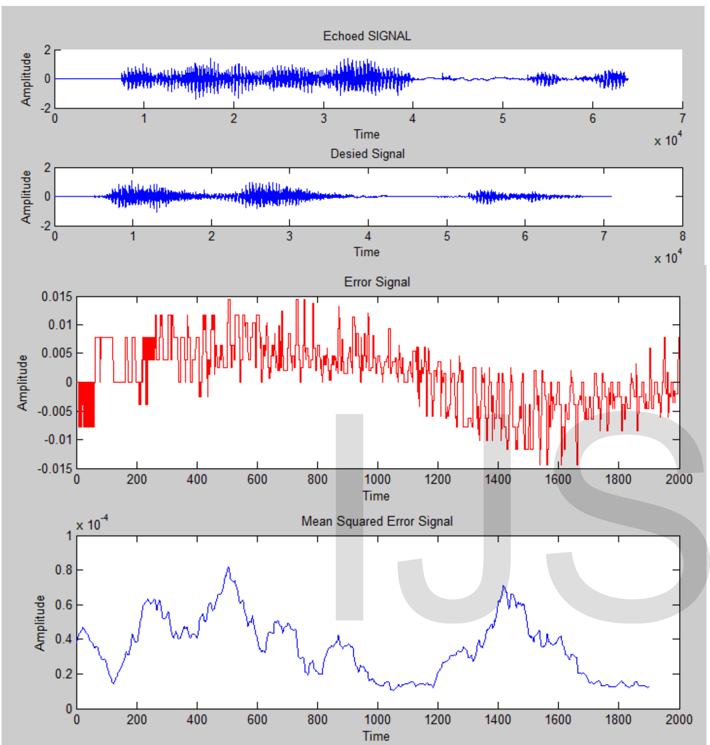

Fig. 7 VSNLMS algorithm outputs for vocal input, N=1500.

5. CONCLUSION

By the results obtained by implementing the Algorithms with the help of MATLAB Software, we draw a conclusion that the best Algorithm for Real Time Implementation is **Normalized Least Mean Squared Algorithm** as it is able to show the maximum echo cancellation with the following most desirable iterations that are ranging up to 10000.

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