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Echo Cancellation using Adaptive Filtering

Anuj Kumar(Student)
anujkumar.ssm@gmail.com

Nutan saini(Assistant Professor)
nutansaini.99@gmail.com

***Abstract*--**In hands-free telephony and in teleconference systems, the main aim is to provide a good free voice quality when two or more people communicate from different places. But while our telephonic conversation the acoustic echo mixes with the original sound. This results in bad sound quality due to which the receiver could not listen the clear sound. Event due to this problem the sound may be modified to some other words or some important information may be lost. This acoustic echo is actually the noise which is created by the reflection of sound waves by the wall of the room or objects available in nearby environment. In this area lots of work has been done by our engineers and scientist and many more work has to be done yet.

So in my work I have tried to solve this problem by using different adaptive filter algorithm and tried to improve some of them to get better result.

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1.1 Introduction

The main problem with the telephonic conversation is the echo generation. Acoustic echo cancellation (AEC) removes the echo captured by a microphone when a sound is simultaneously played through speakers located near the microphone. Many high-noise environments such as noisy conference rooms, war field, Airplanes or lobbies and hands-free telephony in cars require effective echo cancellation for enhanced communication.

The echoes arise due to many reasons, the main reason being an mismatch of the impedance. The mismatch of impedance occurs when the two-wire network meets the four-wire network, this interface is known as the hybrid. This mismatching of impedance causes some of the signal energy to be returned to the source as an echo [1]. See in Figure 1.

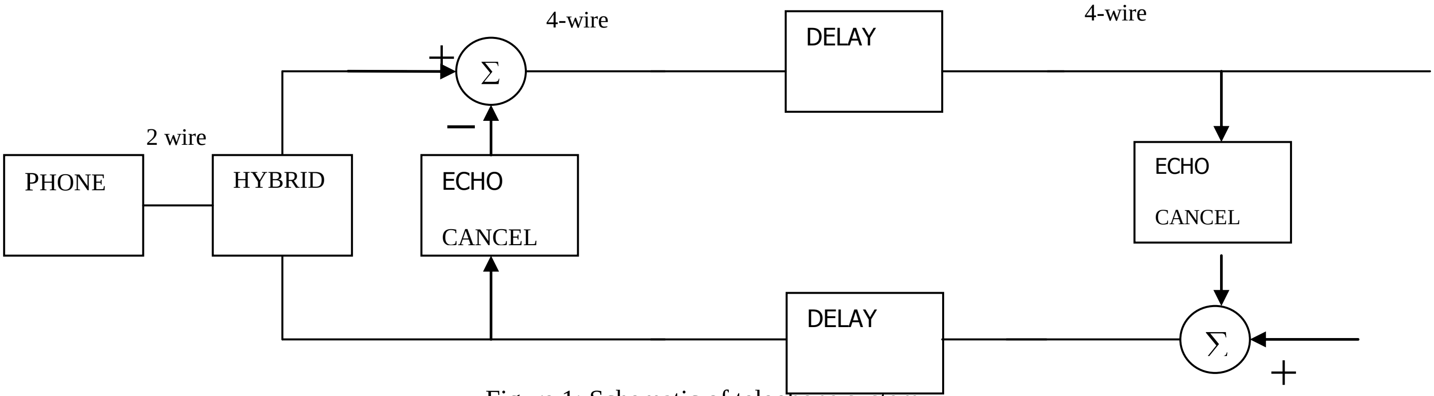

Figure 1: Schematic of telephone system

The eco is directly proportional to the delay in the transmission of the signal from source to destination and the delays between original and echo signals are directly related to the transmission distance. For example, if the signal was sent to a satellite that redirected the signal back to another location on earth, that signal would have a very large time delay compared to a signal sent to a local switching station and back. Very short delays (less than 50 ms) will not affect the quality of the signal as much as longer delays. Delays of this length are not noticed by the receiver and therefore are not considered an annoyance. However, these echoes may have an effect on data being transmitted through transmission line [3].

As a input signal here we are using sinusoid. When we input sinusoid then the DSP board creates an echo of the sinusoid and then the echo is added to the original sinusoidal signal, thus this action leads in creation of distorted version of the input signal. The DSP will then use NLMS adaptive filtering to estimate the echo, and remove the echo from the distorted signal creating a reconstructed signal. The NLMS algorithm seeks to minimize the excess mean-square error (MSE) between the echo signal and the estimated echo. The

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excess MSE refers to the NLMS algorithm fluctuations about the adaptive filter.

1.2 Design Procedure

A sinusoid wave is be used as the test signal in simulation and in experiments.

The NLMS equations that we have used is as follows:-

i. equation of errors finding i.e.noise

$$e(i) = N(i) - w(i)' * N(i); \quad (1)$$

(a) e(i) is the error value.
(b) N(i) is input value
(c) **w**(i) are the present filter coefficients,

ii. equation for co-efficient updation

$$w(i+1) = w(i) + mu * e(i) * N(i) \quad (2)$$

(d) w(i) are the present filter coefficients,
(e) w(i+1) are the future adaptive filter coefficients
(f) N(i) are the input values
(g) e(i) is the error value.
(h) mu is the step size

The mu value in the NLMS algorithm is an important value in determining the performance of the echo cancellation. μ must be chosen between zero and 1.

$$0<mu <1 \quad (4)$$

To update mu we divide mu with power.

$$mu=mu/P.*P \quad (5)$$

μ determines the convergence or divergence speed and precision of filter coefficients in the adaptive filteration. If μ is large, the filter will converge fast, but could diverge if μ is too large. When μ is large, the adaptation is quick, but there will be an increase in the average excess MSE [2]. This excess MSE may be undesirable result. If the μ is small, the filter starts converging slowly, and which is equivalent to the algorithm having "long" memory, and an undesirable quality. In this case Every application will have a different step size which needs to be adjusted. When choosing the value of μ , this needs to be balance between speed convergence and the MSE. The μ values are decided through trial and error so that speed at which the adaptive filter learns and the excess MSE is obtained within application requirements. Due to the inherit differences between systems μ values differ from simulation to real-time.

The system equations used by the adaptive filter to determine the errors and the output of the filter are listed below. The output of adaptive filters are found though convolution of the adaptive filter coefficients w(i) with the input signal x(i).

Output of adaptive filter:

$$yd(i) = sum(w(i)' * N(i)); \quad (3)$$

The error signal e(i) is created from the subtraction of the desired signal from the output of the adaptive filter.

After much iteration, the echo signal resembles the output of the adaptive filter and the reconstructed signal equals the input signal. The input signal is used as an input to the adaptive filter so that the adaptive filter's frequency response approximates the echo frequency response of the system. Then error signal converges to zero(approx.). The error signal is feedback to the adaptive filter and is used as an input with the original input signal to update the NLMS adaptive filter.

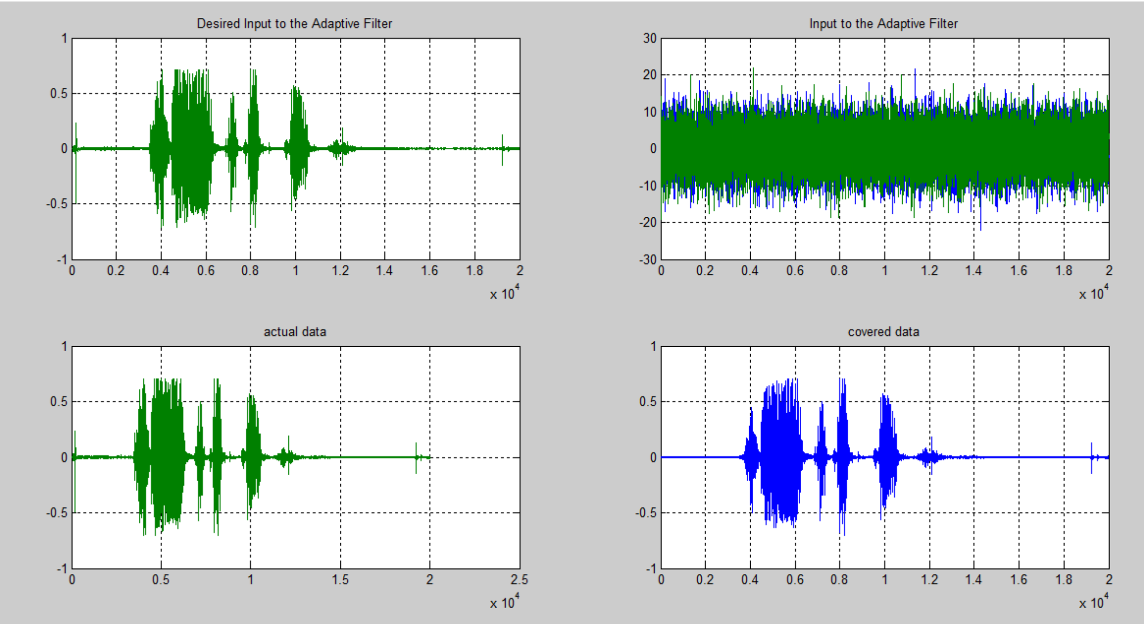

Figure 2. input signal, signal to adaptive filter, actual signal,. Recovered signal

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1.3 Conclusions

The NLMS algorithm here by successfully cancels the echo and then reconstructs the original signal and then returns to us as output. It takes very less iteration to identify the echo system's transfer response. The system worked in simulation and in real-time. The NLMS algorithm is a very powerful and simple tool for echo cancellation. The nonlinear quantization of the DSP most likely to be accounted for these effects. A method, which expands the project, would be to use an actual system with real echoes. A microphone and speaker with a box in between them could be used to create the echoes. Also, longer time delays could be used. Longer time delays would results more filter taps and faster DSPs since the number of calculations increases for the longer filter lengths. The echo cancellation system that was implemented would be useful in the initialization sequence of a phone call.

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