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Simulation Of Echo Cancellation Of Speech Signal In Noisy Environment

R.Jegan **R.Manikandan** **W.S.Nimi** **W.S Kiran**

***Abstract*— Acoustic echo cancellation is important for audio teleconferencing when simultaneous communication of speech is necessary. In acoustic echo cancellation, a measured microphone signal contains two signals such as the near-end speech signal and the far-end echoed speech signal. The goal is to remove the far-end echoed speech signal from the microphone signal so that only the near-end speech signal is transmitted. In speech signal processing slow convergence and high computational burden are the main problems. The proposed method is based on using adaptive filter and uses the least mean square (LMS) algorithm to control a filter coefficients to reduce the echo in the input speech. Here the algorithm is developed and simulated with help of LabVIEW.**

***Index Terms*—Adaptive filter, LMS algorithm, Echo cancellation, LabVIEW**

I.INTRODUCTION

In global communications, wireless phones are essential for communications and have a direct impact on people's personal and business communications. As new network infrastructures are implemented and competition between wireless carriers increases, digital wireless subscribers are becoming ever more critical of the service and voice quality they receive from network providers. Speech is a very basic way for humans to share the information to each one with a bandwidth of only 4 kHz. Interference noise masks the speech signal and reduces its intelligibility. Interference noise can come from acoustical sources such as bio medical equipment, traffic, and commonly, reverberation and echoes.

Acoustic echo occurs when an audio signal is reverberated in a real environment, resulting in the original intended signal plus attenuated, time delayed images of this signal. This paper will focus on the occurrence of acoustic echo in communication systems and improve the quality of speech signal.

Ultimately, the search for improved speech quality has led to important research in signal processing applications. Such research is conducted with the aim of providing solutions that can reduce acoustic echoes. By employing echo cancellation technology, the quality of speech can be improved significantly.The occurrence of acoustic echo in speech transmission causes signal interference and reduced quality of communication. The method used to cancel the echo signal is known as adaptive filtering.

1Assistant Professor, Electronics and Instrumentation Engineering, Karunya University, Coimbatore, TamilNadu-641114.
2Assistant Profesor, Electronics and Communication Engineering, SUN College of Engg and Technology,Nagercoil, TamilNadu-629004.
3Assistant Professor, Electronics and Communication Engineering, Ranganathan Engineering College, Coimbatore, TamilNadu
4 PG Scholar, Electronics and Communication Engineering, Karpagam College of Engineering, Coimbatore, TamilNadu

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. Echoes of our speech are heard as they are reflected back from the floor, walls and other neighboring objects.

If a reflected wave arrives after a very short time of direct sound, it is considered as a spectral distortion or reverberation. However, when the leading edge of the reflected wave arrives a few tens of milliseconds after the direct sound, it is heard as a distinct echo. An echo canceller is basically a device that detects and removes the echo of the signal from the far end after it has echoed on the local end's equipment.

The significance of this paper is to obtain better output signal by cancelling the echo of speech signal. This is because the contamination of a signal of interest by echo is a problem often encountered in many applications. For example, in the noisy environment the original signal is affected by the noise and echo, then the quality of the voice is reduced. In order to obtain better result we need to cancel the noise and echo. So that we used Least Mean Square algorithm, Least Mean Square (LMS) algorithm is the most successful adaptive algorithm. The LMS algorithm adjusts the filter coefficients from sample to sample in such a way to minimize Mean Square Error.

II.MATERIALS AND METHODS

When the telephone connection is between hands-free telephones or between two conference rooms, then an acoustic echo problem emerges that is due to the reflection of the loudspeaker's sound waves from the surfaces and other objects back to the microphone[1].

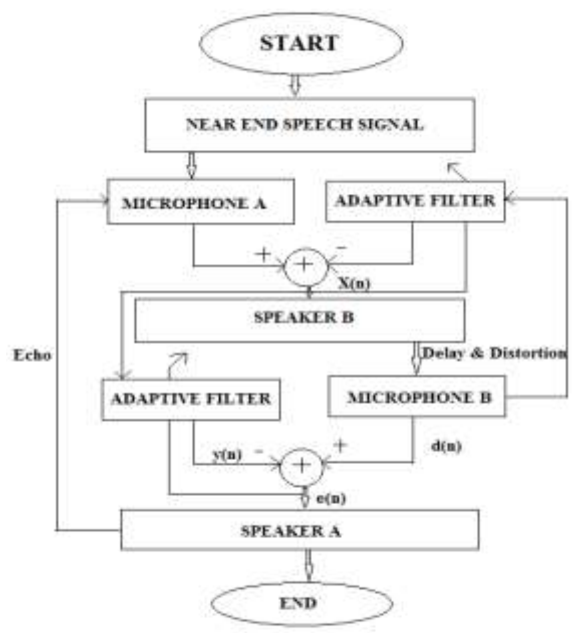

Fig 2.1: Flow Structure of Echo cancellation

The figure 2.1 shows that the basic flow structure of echo cancellation. When place a speaker near a microphone in a full-duplex communication environment, acoustic echo occurs. As Microphone A acquires the voice from the near-end person and transfers the voice to Speaker at the far end. Microphone B then picks the voice from Speaker at the far end and transfers the voice back to Speaker at the near end. When the near-end person talks to Microphone A, this person hears his or her own distorted and delayed voice in Speaker. The distorted and delayed voice is acoustic echo, which affects the communication negatively when the delay is large enough.

A. Adaptive algorithm objectives

As previously demonstrated, the best solution for reducing the echo is to use adaptive algorithm. Basically filtering is a signal processing technique whose objective is to process a signal in order to manipulate the information contained in the signal. An adaptive filter is necessary when either the fixed specifications are unknown or time-invariant filters cannot satisfy the specifications. Additionally, adaptive filters are time varying since their parameters are continually changing in order to meet a performance requirement.

Most adaptive algorithms have one objective, that is "to minimize the average power (or mean square value) of error signal, e(n), and thus to maximize the output signal-to-noise ratio (SNR)".

The estimated desired signal is given by

$$e(n) = d(n) - y(n) = S(n) + N(n) - y(n)$$

Squaring this equation:

$$e^2(n) = [S(n) + (N(n) - y(n))]^2$$

$$= S^2(n) + 2\,S(n)[N(n) - y(n)] + [N(n) - y(n)]^2$$

To find average power of error signal, take expectations, E[ ], of both sides

$$E[e^2(n)]=E[S^2(n)]+2E[S(n)\{N(n)-y(n)\}]+E[\{N(n)-y(n)\}^2]$$

Since the desired signal, S(n), is uncorrelated with N(n) or with y(n),

$$E[S(n)\{N(n) - y(n)\}] = 0$$

Hence,

$$E[e^2(n)]=E[S^2(n)]+E[\{N(n)-y(n)\}^2]$$

If the estimate of noise is exactly replica of noise, that is the value y(n)=N(n), the average output power will contain only the average signal power, that is E[e2(n)] = E[S2(n)]. However, it is impossible to achieve y(n)=N(n). Practically, when $y(n)\approx N(n)$, the average remnant noise power will be minimized. This also means that the average output power will be minimized:

$$\min E[e^2(n)]=E[S^2(n)]+\min E[\{N(n)-y(n)\}^2]$$

B. Adaptive filter

Adaptive echo cancellation uses adaptive filters at both ends to estimate the transfer function between each speaker and its corresponding microphone. When Microphone A transfers the speech signal to Speaker B, the same signal also feeds into an adaptive filter at the far end. The adaptive filter then adjusts the filter coefficients iteratively to estimate the distorted and delayed speech. If the estimated signal is close to the signal from Microphone B, no acoustic echoes occur at Speaker A. An adaptive filter is a filter that has time varying coefficients and frequency response. It is intelligent and flexible because it tracks the input signal characteristics, and automatically self-adjusts the filter coefficients to improve its performance.

C. Frequency Domain Adaptive Filter

The algorithm that we will use in this demonstration is the Frequency-Domain Adaptive Filter. This 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.

D. Least Mean Square Algorithm

The least mean square, (LMS), is a research algorithm that is widely used in various applications of adaptive filtering. The main features that attracted the use of the LMS algorithm are low computational complexity, proof of convergence in stationary environments and stable behavior when implemented with finite precision arithmetic. Least Mean Square (LMS) algorithm is the most successful adaptive algorithm. The LMS algorithm adjusts the filter coefficients from sample to sample in such a way to minimize Mean Square Error (MSE).

The LMS algorithm is used to determine the error value, filter output and update filter coefficients[8].The update coefficients are expressed as,

$$W(k,n+1)=w(k,n)+\mu e(n)x(n)$$

This is the results of the mathematical expression of adaptive algorithm such as least mean square algorithm, where μ is the step size and e(n) is the error signal and w(k,n) is the initial filter coefficients and y(n) is the filter output and x(n) is the input samples and w(k,n+1) is the update filter coefficient. It is simple and does not require squaring, averaging or differentiating.

According to G. Schmidt, "Applications of acoustic echo control: An overview," in Proc. Eur. Signal Process. Conf. (EUSIPCO'04), Vienna, Austria, 2004, pp. 9–16. They developed the basic technique for echo cancellation. For this the problem of acoustic echo cancellation is usually solved by using an adaptive filter in parallel to the acoustic echo path. An estimate of the near end speech signal is then obtained by subtracting the estimated acoustic echo signal, i.e., the output of the adaptive filter, from the microphone signal.

With respect to V. Turbin, A. Gilloire, and P. Scalart,[2] They developed a method with combine noise and echo cancellation of speech signal using LMS algorithm. In some applications, like hands-free terminal devices, noise reduction becomes necessary due to the relatively large distance between the microphone and the speaker. The first attempts to develop a combined echo and noise reduction system. Both employed more than one microphone. Based on B. Yegnanarayana and P. Murthy[4] proposed another single microphone dereverberation technique in which a time-varying weighting function was applied to the linear prediction (LP) residual signal. The weighing function depends on the signal-to-reverberation ratio (SRR) of the reverberant speech signal and was calculated using the characteristics of the reverberant speech in different SRR regions. Unfortunately, these techniques are not accurate enough in a practical situation and do not fit in the framework of the post filter which is commonly formulated in the frequency domain. Recently, practically feasible single microphone speech dereverberation techniques have emerged. Lebart proposed a single microphone dereverberation method based on spectral subtraction of the spectral variance of the late reverberant signal. The late reverberant spectral variance is estimated using a statistical model of the AIR.

According to Emmanuel and Barrie "Noise cancellation of speech signal using LMS algorithm". They designed and developed the system with basic LMS algorithm. They were explaining the objective of adaptive algorithms which is "to minimize the average power. If the estimate of noise is exactly replica of noise, then the average output power will contain only the average signal power, that is $E[e^2(n)] = E[S^2(n)]$. Least Mean Square (LMS) algorithm is the most successful adaptive algorithm developed by Widrow and his coworkers (Emmanuel and Barrie, 1993). Michael Hutson, Acoustic echo cancellation using digital signal processing, November 2003 deals with the technique of echo cancellation of speech signal. Here they were explaining that acoustic echo cancellation in today's telecommunication systems. It occurs when an audio source and sink operate in full duplex mode; an example of this is a hands-free loudspeaker telephone. In this situation the received signal is output through the telephone loudspeaker (audio source), this audio signal is then reverberated through the physical environment and picked up by the systems microphone. The signal interference caused by acoustic echo is distracting to both users and causes a reduction in the quality of the communication. This paper mainly focusing the use of adaptive filtering techniques to reduce this unwanted echo, thus increasing communication quality.

III SIMULATION RESULT

The Least-mean squares algorithm (LMS) can be implemented by using the graphical language LabVIEW.

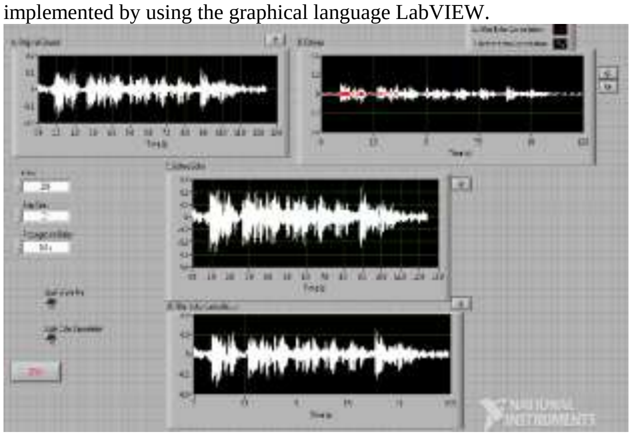

Fig 3.1: Simulation results of Echo Cancellation

IV CONCLUSION

One of the major problems in a telecommunication application over a telephone system is echo. The Echo cancellation algorithm presented in this paper successfully attempted to find a software solution for the problem of echoes in the telecommunications environment. The proposed algorithm was completely a software approach without utilizing any microcontroller and DSP hardware components. The algorithm was capable of running in any PC with LabVIEW software. The audio of the output speech signals were highly satisfactory and validated the goals of this research.

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[1]R.Jegan, Assistant Professor from Karunya University, completed Bachelor of Engineering in Electronics and Communication in Sun college of engineering and Technology, Nagercoil and Master of Engineering in Karunya University, Coimbatore. Currently pursuing Ph.D in Wireless technology in Anna University. He was awarded First Rank in P.G Degree Course in Embedded systems and currently guiding many Embedded projects. His area of Interests are Embedded Control system and signal processing. He has published 3 international journals and 8 international conferences.

[2]R.Manikandan, Assistant Professor from SUN College of Engineering and Technology, completed Bachelor of Engineering in Electrical and Electronics Engineering in Sun college of engineering and Technology, Nagercoil and Master of Engineering in Sathyabama University, Chennai. He was awarded First Rank in P.G Degree Course in VLSI Design. His area of Interests are Control systems and signal processing.

[3]W.S.Nimi, Assistant Professor from Ranganathan Engineering College, completed Bachelor of Engineering in Electronics and Communication in Sun college of engineering and Technology, Nagercoil and Master of Engineering in St.Joseph Engineering College, Chennai. Her area of Interests are Electronics circuits Design and signal processing.

[4]W.S.Kiran, PG Scholar completed Bachelor of Engineering in Electronics and Communication in VINS college of engineering and Technology, Nagercoil and doing Master of Engineering in Karpagam College of Engineering, Coimbatore. His area of Interests are Electronics circuits Design and signal processing.