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DESIGN OF ADAPTIVE FILTER FOR ECHO CANCELLATION

Dr. Krishna Samalla[*1], M. Preethi Sree[*2], K. Shreshta[*3], V. Divya Sree[*4]

[*1]Professor,Dept.Of ECE, Sreenidhi Institute Of Science And Technology, Telangana, India.

[*2,3,4]B.Tech (ECE), Sreenidhi Institute Of Science And Technology, Telangana, India.

ABSTRACT

Now-a-days, all are very much interested in hands free communication. In that case, replacing the telephone receiver with a standard loudspeaker and a high-gain microphone could seem like a good idea. The presence of a large acoustic coupling between the loudspeaker and microphone, which produces a loud echo, makes conversation difficult. Moreover, a loud howling noice might be produced since the system may become unstable. Hence, these problems can be resolved by the elimination of echo using an echo cancellation technique. The signal received by the loudspeaker in a telephony system reverberates through the environment and is caught up by the microphone. An echo signal is what it's called. It is in the form of an attenuated image of the original speech signal that is time delayed, resulting in a degradation in communication quality. Adaptive filters are filters that change their parameters iteratively in order to narrow the gap between their output and a desired result. The ideal output in the case of acoustic echo is an echoed signal that accurately reproduces the unwanted echo signal. It is used to negate the echo in the return signal. The better the adaptive filter models this echo, the more successful the cancellation will be. There are many techniques for echo cancellation. Here we designed a filter by which echo can be cancelled in a easy process.

Keywords: Echo Cancellation, Adaptive Filter.

I. INTRODUCTION

Echo – implies a distinct and delayed version of sound signal . It is the repetition of waveform due to reflection from points where the characteristics of the medium through which the wave propagates changes. Echo is the reflection of sound that arrives at the listener with a delay after the direct sound in audio signal processing and acoustics. The quality of the service can be degraded by echo and an important part of communication system is echo cancellation.

A notable problem in communications is the generation of echoes. The echoes are generated due to number of reasons. The primary reason is an impedance mismatch. The impedance mismatch occurs when the two wire network meets the four-wire network. This is an interface which is called as the Hybrid.

The energy of the signal is returned to the source as an echo because of the impedance mismatch. The delays between primary and echo signals are directly related to the distance of the transmission.

Echo suppression has been employed in conventional telecommunications to lessen the nature of echoes in human communication, when one person speaks while the other listens and vice versa(Half duplex transmission). An echo suppressor recognises the principal direction of speech and permits that channel to progress, while it attenuates or suppresses any signal in the reverse channel, assuming it is an echo. This means that neither channel can speak until the other has finished transmitting speech. Echo cancellation is more effective. Some of the applications are 'Network Echo Cancellers' and 'Terminal Echo Cancelers'. Because speech terminals, such as those used for teleconferencing, need amplifiers to drive their loudspeakers, the microphone and the loudspeaker cannot be separated. The effect of acoustic coupling is reduced by minimising the amount of feedback (reverberation) within the teleconferencing room using adaptive filters used in echo cancellation.

1.1 GENERATION OF ECHO:

In the same manner that a rubber ball bounces off the ground, sound waves can bounce off smooth, hard things. Despite the fact that the sound's direction changes, the echo sounds identical to the original. Echoes can be heard in small spaces with hard walls, such as wells, or in areas with a lot of hard surfaces. Echoes can be heard in a canyon, cave, or mountain range for this reason. However, sounds are frequently reflected. They will be absorbed and will not bounce back if they come into contact with a soft surface, such as a cushion.

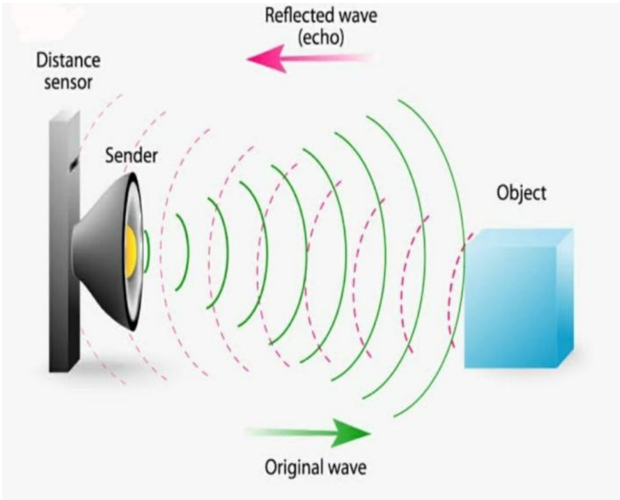


Fig: Generation of echo signals

1.2 NEED FOR ECHO CANCELLATION:

Wireless phones are seen as crucial communications tools in this era of global communications, and they have a direct impact on people's day-to-day personal and commercial communications. Today's subscribers consider speech quality to be a crucial aspect in determining a network's overall quality. It is key to maintain subscriber loyalty, regardless of whether or not the subscribers' opinions are subjective. As a result, the key to maintaining and improving the perceived speech quality of a call is to effectively remove hybrid and acoustic echoes, which are inherent within the telecommunications network infrastructure. Finally, the pursuit of better speech quality led to extensive research into the field of echo cancellation.

II. PROBLEM STATEMENT

Traditional acoustic echo cancellers suffer from issues such as slow convergence (particularly for speech signals) and high computational complexity as the identification of the echo path requires filter with more than thousand taps, non-stationary speech input, slowly time varying systems to be identified. Traditional approaches are unable to meet the demand for fast convergence and low MSE levels, and the use of algorithms for echo cancellation adds to the complexity. We have designed a project in such a way that it removes the above complications.

III. LITERATURE SURVEY

Existing System-1

The traditional acoustic echo canceller has issues with slow convergence (particularly for speech signals) and high computational complexity as the identification of the echo path require filter with more than thousand taps, non-stationary speech input, slowly time-varying systems to be identified. When sound is reflected from the floor, wall, or other nearby objects, it is called an echo. When the round trip delay exceeds 30ms, echos become irritating. When it does, the echo strength will be 30dB higher, which is irritating.

Traditional adaptive filtering methods are unable to meet the demand for fast convergence and low MSE levels. There is a need to be computationally efficient and rapidly converging algorithm.

Existing System-2

Adaptive Algorithms:

1. LMS Algorithm:

Widrow and Hoff created the Least Mean Square (LMS) algorithm in 1959 as part of their pattern recognition research. Since then, it has become one of the most extensively used adaptive filtering algorithms. The LMS algorithm is a form of adaptive filter characterised as a stochastic gradient-based algorithm because it converges on the optimal wiener solution using the gradient vector of the filter tap weights.

2. NLMS Algorithm:

One of the main drawbacks of the LMS algorithm is that each iteration has a set step size parameter. The weight adjustment in the LMS method is proportional to the amplitude of the input vector samples. As a result, the LMS suffers from gradient noise amplification when the input vector[n] is big. To mitigate this issue, the weight vector change done at each iteration is normalised. The normalised least mean square algorithm (NLMS) is an extension of the LMS algorithm that calculates maximum step size to avoid this issue.

3. IPNLMS Algorithm:

As an interesting alternative to the normalised least-mean-square (NLMS) filter, proportionate adaptive filters, such as the improved proportionate normalised least-mean-square (IPNLMS) algorithm, have been presented for echo cancellation. The IPNLMS algorithm combines proportional (PNLMS) and non-proportionate (NLMS) updating methods. The difficulty with the PNLMS algorithm is that it performs better when the impulse response is sparse, but it is slower to convergence than NLMS when the impulse response is not sparse, hence we opt for IPNLMS.

4. RLS Algorithm:

At each iteration, the least square algorithms require all previous samples of the input signal as well as the desired output. Weiner filter's RLS version is a simple adaptive and time update version. This filter detects temporal changes for non-stationary signals, yet it has the same convergence behaviour as a wiener filter for stationary signals.

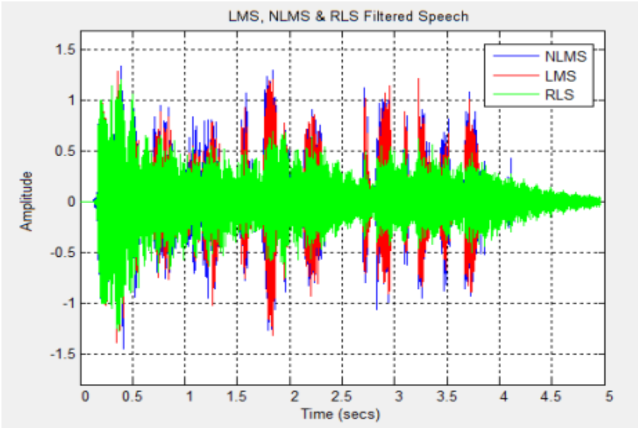
The RLS algorithm is well-known for its superior performance in time-varying situations. These benefits come at the expense of greater computing complexity and perhaps additional stability issues.

A comparison between the LMS, NLMS and the RLS aalgorithm

Algorithms	Convergence rate	Mean Square Error	Computation Complexity	Stability	Signal to Noise Ratio
LMS	More iteration required to converge.	Mediate value of MSE	Simplest	Sensitive to scaling of its input, therefore it's hard to choose step-size to give stability	Lower than NLMS
NLMS	Less iteration to converge, Faster than LMS and RLS	Lowest value of MSE	Simple	Stable with a small value of step-size	High
RLS	Less iteration than LMS to converge	Highest value of MSE	Highly Complex		

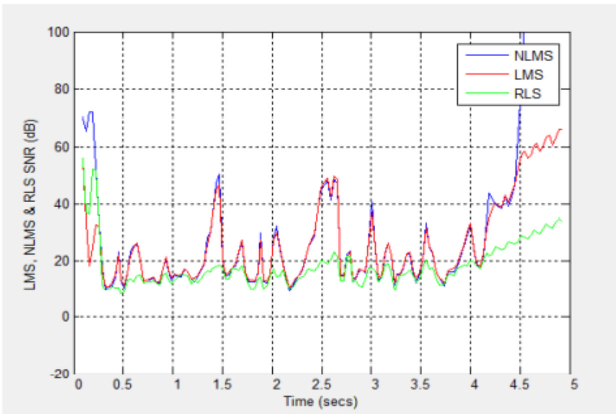
Comparing the LMS, NLMS and RLS algorithms at their best performance:

1. Filtered Speech: As seen in Fig, the LMS and NLMS algorithms function nearly identically, with the exception that the LMS is unstable.



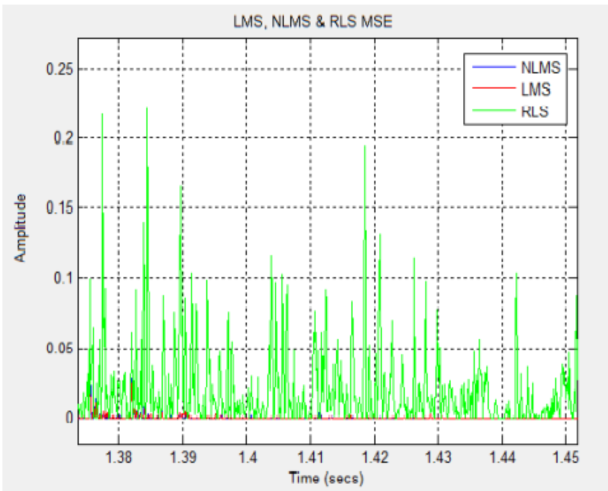
Filtered Speech signals of LMS algorithm at μ = 0.1, NLMS algorithm at μ = 0.1 and RLS algorithm at β = 1.0: N = 10 taps

2. Signal to Noise Ratio (dB): The NLMS performs its best compared to others that is shown in fig



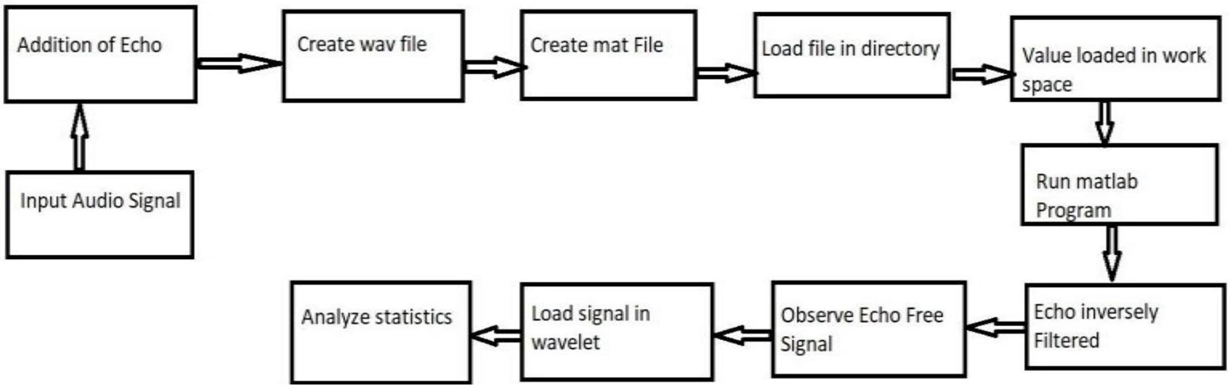
SNR of LMS algorithm at μ = 0.1, NLMS algorithm at μ = 0.1 and RLS algorithm at β = 1.0: N = 10 taps

3. Mean Square Error (MSE): Because of its good convergence rate and stability, the NLMS MSE's signal (blue signal) is not visible; it perfectly attenuates the power of the error, followed by the LMS and then the RLS in this experiment.



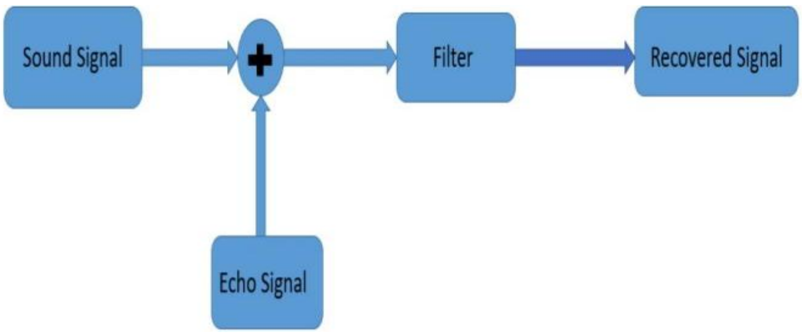
MSE of LMS algorithm at μ = 0.1, NLMS algorithm at μ = 0.1 and RLS algorithm at β = 1.0: N = 10 taps

IV. PROPOSED SYSTEM



Inverse filtering (IF) is a well-known voice and speech analysis technique that focuses on determining the origin of spoken communication. The glottal volume velocity waveform or glottal airflow can be estimated using this method. By filtering the speech stream, inverse filtering creates a computational model for glottal pulse detection. To remove echo from the signal, Matlab can be used to denoise it.

BLOCK DIAGRAM



The above block simply shows the functioning of the project in which various components are connected. Here the microphone generates the sound signal. An echo signal is also created from microphone but with some delay and attenuation. A simple summer is used which is used to add signals. Filters are data processing techniques for smoothing out high-frequency data or removing periodic trends of a given frequency. The output signal is the exact copy of the original sound signal. This is the basic principle behind our project Echo cancellation.

Adaptive Filters

An adaptive filter, as the name implies, is a digital filter that automatically adjusts to changes in its input signals using a combination of adaptive algorithms to modify the filter's coefficients. Echo cancellation, radar signal processing, noise cancellation, biomedical signal improvement, and communication channel equalisation are all applications that use adaptive filters. The application of adaptive filters to echo cancellation is the topic of this report.

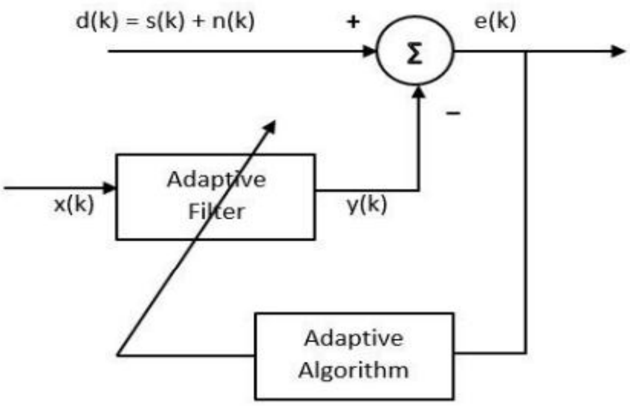


Fig: General setup of adaptive filter

Figure depicts the general setup of an adaptive filtering environment, with k denoting the iteration number, x(k) denoting the input signal, y(k) denoting the adaptive filter output, and d(k) denoting the target signal. The error signal e(k) is calculated by multiplying d(k) by y(k). The error is then utilised to create a performance function or objective function that the adaptive algorithm uses to determine when the filter coefficients should be updated.

In an echo cancellation system, an adaptive filter adjusts approaches that must be employed to obtain correct solutions to the problem of echo. The following steps are included in an adaptive filtering process:

1. A filtering procedure that is designed to produce a desired result in response to input data, and

2. An adaptive process that gives an algorithm for changing the coefficients of a set of filters.

Finite impulse response (FIR) filters and Infinite impulse response (IIR) filters are two forms of adaptive filters that differ in their impulse response. Non-recursive structures are commonly used in FIR filters, whereas recursive realisations are used in IIR filters. In this report, FIR filters will be discussed.

Adaptive FIR Filter :

The tapped delay line, lattice predictor, and Systolic array are the three types of FIR structures. The tapped delay line, also known as a transversal filter, is the most often used adaptive FIR structure. The tapped delay line, lattice predictor, and Systolic array are the three types of FIR structures. The tapped delay line, also known as a transversal filter, is the most often used adaptive FIR structure. A filter coefficient, also known as tap weight, is used to multiply each tap delay input connected to each multiplier. As a result, a multiplier linked to the nth tap input x(k-n) yields the inner product wn(k-n), where wn is the appropriate tap weight and n=0, 1, 2,..., N-1.

The multiplier outputs are added together by each adder in the filter, yielding a total filter output y(k) as a linear combination of the filter coefficients.

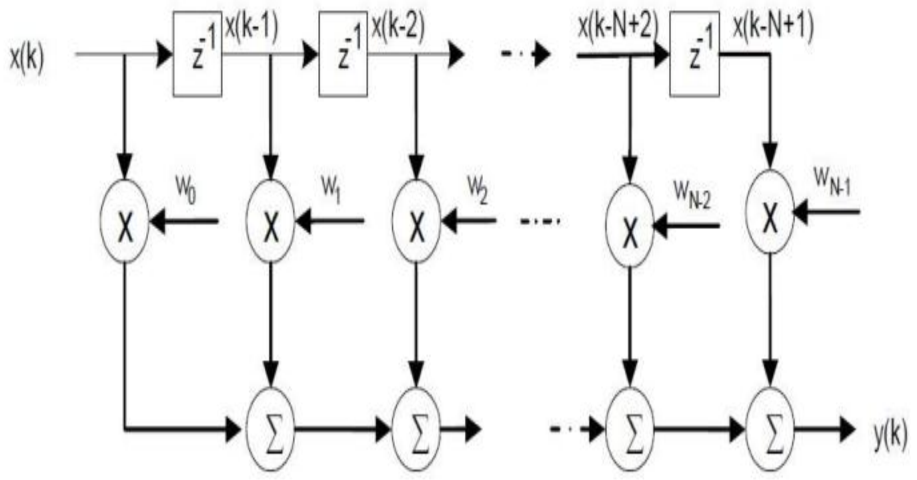


Fig: Transversal Filter

V. SOFTWARE

The software used in this project is MATLAB. MATLAB is a general-purpose, powerful, and efficient mathematical software tool. The graphics and matrix processing capabilities of MATLAB are outstanding. It combines mathematical computing with a sophisticated language to create a secure computing environment for technical applications.

We used MATLAB because the input signals (far-end and near-end talker signals) were voices. These voices are saved as wav files, which can easily be loaded into the code. The intermediate signals (echo signals) and output signals (error signal and signals obtained after echo cancellation) were obtained as wav files. As a result, the audio of the voice signals could be heard, which facilitated enormous judgments regarding the outcomes acquired.

Almost all signal processing applications have built-in functions in the signal processing toolbox. The toolbox improves code efficiency by allowing these functions to be called wherever they are needed rather than having to write separate subroutines.

As MATLAB enables visuals, the outcomes of a simulation could be easily displayed in a graphical format.

Matlab System

There are five main elements in the matlab system.

Development Environments

This is a collection of tools and utilities that make it easier to use MATLAB operations and files. The graphical user interface is employed in many of these tools. It includes the MATLAB desktop and command window, as well as a command history, an editor and debugger, and browsers for looking at help, workspace, reports, and the search route.

MATLAB- Mathematical Function Library

This is a comprehensive collection of computing design, spanning from simple functions such as sum, sine, and cosine, to more advanced features like as the matrix inverse, matrix eigenvalues, Bessel functions, and rapid Fourier transformations.

MATLAB Language

This is a high-level matrix/array programming language that includes control flow statements, functions, data structures, input/output, and object-oriented programming features.

It enables "programming in the tiny" to quickly develop quick and filthy throw-away programmes, as well as "programming in the huge" to quickly create vast and complicated application functionalities.

Graphics

MATLAB includes a lot of features for presenting vectors and matrices as graphs, as well as annotating and printing them. It includes high-level frameworks for data visualisation in two and three dimensions, image processing, animation, and presentation graphics. It also includes low-level components that enable us to freely configure the graphics display and create comprehensive graphical user interfaces in our MATLAB programmes.

MATLAB External Interfaces/API

This is a library that allows us to interface with MATLAB using C and FORTRAN programmes. It includes everything you need to invoke MATLAB routines (dynamic linking), use MATLAB as a computational engine, and read and write MAT-files

VI. WORKING

First a Sound signal is taken. The Sound signal can the Speech signal of a person or a music signal. Then an Echoed signal is generated. The Echo is generated by adding a delay to a signal and adding a attenuation factor. The generated echo signal and the sound signal are added by a summer. Now a Echoed signal is generated. In order to remove echo from the echoed signal we used a filter i.e., 1-D digital filter. It is an Adaptive filter.

A filter is a signal processing system that reduces or enhances specific qualities. Applications are Multimedia, technology, and communication etc.

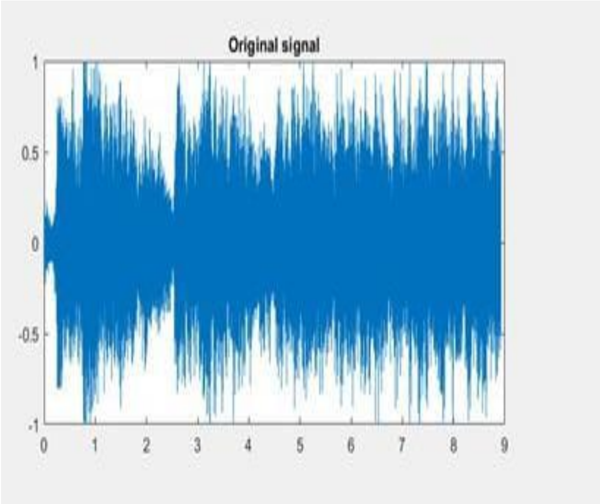
When the echoed signal is applied to the Adaptive filter, it gives the original signal as its output. An adaptive filter is a system, or a linear filter, with a transfer function controlled by variable parameters and the ability to alter those parameters using an optimization technique. The amplitude of the echoed signal will be greater than the original signal. The filter removes the echo and reduces its amplitude. Now, we get a output signal that is an exact replica of original signal. Thus, this project makes easy to remove echo from a echoed signal i.e., echo cancellation.

This project is implemented using matlab. We have written a program in editor window for Echo Cancellation. First and formost enter an audio signal. Following that, an Echo signal is added. To add the audio signal to Matlab, we created a wav file. Save the program with .m extension and a mat file is created. Now load file in the directory. We can see the values loaded in the workspace. Now run the matlab program. Then a graph is plotted for original signal, echoed signal and recovered signal.

VII. SIMULATIONS AND RESULTS

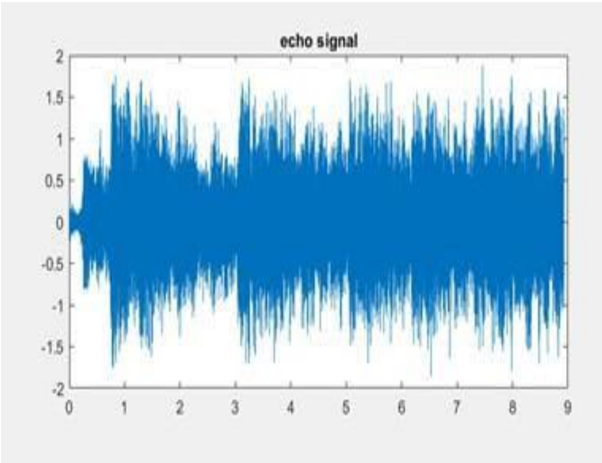
Original Speech Signal:

For this, a live speech was recorded using microphone in a non-stationary setting at 8000 samples per second.



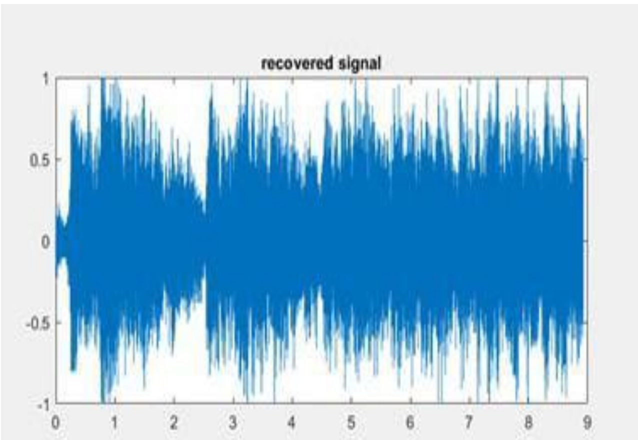
Echo Signal:

The echo signal was created by adding some delay and attenuation factor to the speech signal.



Recovered Signal:

The signal that was recovered is an exact copy of the original signal.



VIII. CONCLUSION

The notion of acoustic echo cancellation is well-known for achieving smooth and flawless speech quality. Acoustic echo cancellation removes acoustic coupling between the microphone and loudspeaker, which causes echo, reverberation, and unwanted noise. When compared to its analog counterpart, which took a long time to create, audio effect generation utilising digital signal processing is a simple and time-consuming procedure.

IX. FUTURE SCOPE

This paper proposes a strategy for dealing with single-channel acoustic echoes. Multichannel sound has, nevertheless, become the norm for telecommunication in most real-world scenarios. For example, multichannel sound abounds when a group of people is in a teleconference scenario and everyone is busy talking, laughing, or simply communicating with one another. Because there is only one microphone, the opposite end will only hear a monophonic sound that is highly nonsensical. The echo cancellation technique created during this research needs be enhanced for the multichannel scenario in order to properly manage such situations.

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