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MATLAB BASED ECHO CANCELLATION SYSTEM

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Abstract - Because of call setup information and actual "voice data" is transported on an open network like Internet, there are some main challenges with VoIP security today. This section will give a brief introduction to the technology in use and it also introduces short description of security risk. Voice over Internet Protocol (VoIP) is a technology that enables one to make and receive phone calls through the Internet instead of using the traditional analogy PSTN (Public Switched Telephone Network) lines. it is packetisation and transport of classic public switched telephone system audio over an IP network. It allows 2-way voice transmission over broadband connection. It is also called IP telephony, internet telephony, and voice over broadband, broadband telephony.

Keywords - Voice over Internet Protocol (VoIP), Inverse filtering

I. INTRODUCTION

Voice over IP (VoIP) is a methodology and group of technologies for the delivery of voice communications and multimedia sessions over Internet Protocol (IP) networks, such as the Internet.
VoIP systems employ session control and signaling protocols to control the signaling, set-up, and tear-down of calls. They transport audio streams over IP networks using special media delivery protocols that encode voice, audio, video with audio codes and video codes as digital audio by streaming media.

Echo is your voice coming back to you, as if you were repeating yourself. During a normal, two-person phone conversation, your voice is transmitted from your mouth to the ear of the person at the other end, and their voice is returned from their mouth to your ear. However, in any conversation, a certain amount of your own voice is also part of what you hear, whether you are talking face-to-face with someone who is sitting in your office, or talking to someone on the phone. This experience of hearing your own voice is not echo. Commonly called “side tone,” it’s a normal aspect of talking and listening. Your own voice becomes echo when it comes to your ear with a significant delay from the time you spoke.

In the field of Telecommunication, the concept of Echo cancellation is generally used in telephonic system or in transreceiver systems (a single system which does the working of transmission and reception at a time) to remove the echo signals from it. Echo cancellation is the best way through which the quality of calls can be improved as well as to enhanced the network performance which is to maintain customer reliability.Echo is nothing but repetation of waveform because of reflection of some points from original one’s. Echo degrades the actual signal output and hence the echo cancellation plays very important role in the field of Telecommunication.
The reasons of echo are as follows:

1. Poor room acoustics
2. Marginal microphones for soft terminals
3. Low quality cellular handsets
4. Deficient echo control in the terminal device
5. Bridge-taps
6. Use of lengthy untwisted wire within the subscriber's premises

II. PROPOSED SYSTEM

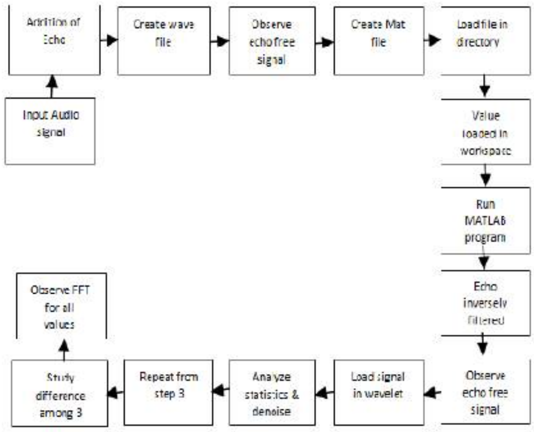

Fig.1. Proposed system of echo cancellation

Inverse filtering (IF) is a widely known method for voice and speech analysis, which mainly works on estimating the source of voiced speech. This method enables to estimate the glottal volume velocity waveform or glottal airflow. The idea behind inverse filtering is to form a computational model for glottal pulse detection by filtering the speech signal.The signal can be denoised using Matlab to remove echo from this signal.

In this method, we are using Acoustic Echo Cancellation system. Generally, in the conferences whenever the far side speech is played at loudspeakers and if that signals are catched by the microphone which is in the room then that are being transmitted back to the far side end, which is an Echo.

These transmitted signals are some different kind of signals which are delayed as compared to original one's. Generally Acoustic Echo Cancellation (AEC) can be used for the applications such as, Double talk, Echo Return Loss (ERL), Echo Return Loss Enhancement (ERLE), Non Linear Processing (NLP), Noise Cancellation (NC), Voice Activity Detection (VAD).

There are so many techniques by which AEC can be done such as FIR frequency domain adaptive filter, FIR frequency domain adaptive filter with LSM algorithm, FIR frequency domain adaptive filter with NLSM algorithm, FIR frequency domain adaptive filter with Sign-Error algorithm.

For the fast convolution output, the Adaptive filter is used in AEC technique which is very beneficial. This is also used because the impulse response of system to be identified is long.

In the time domain applications, the use of Adaptive filter are most advantageous, which works most efficiently.

The frequency domain weight vector is defined as, W(k) = $[W0\ k\ , \dots \dots , WM{-}1\ k\ ]\ T$.

And the input signal matrix is given as,

X(k) = diag{$X0$ (k),……,$XM{-}1(k)$},

Where diag{.} is an operator which generates a diagonal matrix.

III. RESULTS AND DISCUSSION

In th telephonic conference system, the users are generally located near the system's microphone which is known as near end speech signal. As through the loudspeaker the voice travels out in that particular room and bounces around which again picked by the sysetem's microphone, these voice signals are known as far end speech signals. And hence the microphone now carrying both the signals that are near end speech and far end speech, that has been echoed through the room.The main aim of experiment of Adaptive echo cancellation system is to remove the bounced far end speech signals from the microphone or from the system so that only near end speech signals are transmitted back to the far end listener.

For the experiment at first voice signal is acquired by using speech recorder. Then this acquired voice signal is used to create a .wave file using the audio signal. After that Matlab software is opened the data of the signal is imported to create .mat file. This mat file has to be loaded to transfer the value of the speech signal in workspace. To remove echo from the signal, wavelet is used. To inspect the signals by wavelet wave menu is typed in the command window of Matlab; then a new window of wavelet will be displayed. The echoed signal is loaded in wavelet and then the signal is analyze, view the statistics, denoise and analyze the signal. Doing all these signal with echo can be analyzed.

By running the program & hearing the signal it will be clear that echo is removed from the echoed signal and the quality of signal is improved. The signal is clear and each part of speech is present. If the result is observed closely then it can be seen that the wave of main signal is like overlapped in the echoed signals figure. The refined signal is almost look like the echo free signal.

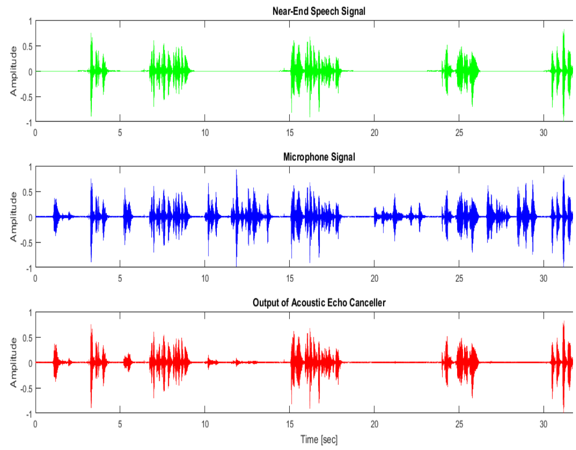

Fig.2. Result of the echo cancellation system.

CONCLUSIONS

This method is easy to use and simpler than the conventional methods of suppressing echo. Power spectrum density has also been used to observe the difference of the received signal. The proposed system has found better performance in finding the echo cancellation than the conventional methods which can be adopted to suppress echo and take the fullest advantage of VOIP telephony.

The goal of an echo canceller is to completely remove any emanating signal from impedance mismatching. Real systems are incapable of perfect cancellation for a variety of reasons. It is important, therefore to be able to quantify actual adaptive echo canceller performance and to understand system and physical limitations that effect cancellation so that an adaptive echo canceller can be optimized for its environment.

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