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SPEECH ENHANCEMENT SYSTEM USING LABVIEW

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ABSTRACT
Speech enhancement has become one of the most important tools of the modern generation and is widely used in various fields for various purposes. The past decade has seen dramatic progress in speech recognition technology, to the extent that systems and high-performance algorithms have become accessible. Speech enhancement depends on signal processing. Speech enhancement techniques are widely used to enhance the quality and intelligibility of the speech signal in the noisy environment. Conventional noise reduction methods introduce more residual noise and speech distortion. The existing algorithms fail when there are abrupt changes in the noise level. To overcome the shortcomings of the conventional methods, improved noise tracking algorithm is proposed in this paper for speech enhancement. The noise signal is estimated for the existing and the proposed methods. Results are simulated using LabView. This report shows how to recognize and enhance the speech using filters in lab view. Predictable noise reduction methods initiate more enduring noise and speech alteration. The existing algorithms not succeed when there are sudden changes in the noise level. To overcome the shortcomings of the unadventurous methods, enhanced noise tracking algorithm is future in this paper for speech enhancement. The noise signal is estimated for the accessible and the future methods. Calculate the SNR (signal to noise ratio) value of input signal, input signal plus added noise and filtered signal in order to measure the improved SNR value. By using filters we will get the enhanced speech signal with reduced noise. The aim speaker, and the signal-to noise ratio (SNR) specifically to switch definite speakers, noise types and SNRs, are competent of achieving hefty improvement in estimated speech quality (SQ) and speech clearness. A noisy sound of an untrained speech is processed finally; we compare the proposed algorithm with different speech enhancement algorithms. The contribution of all components of the proposed algorithm was analyzed signifying their collective importance.

KEYWORDS: Speech enhancement, speech quality, speech intelligibility, Speech noise estimation, residual noise, SNR and speech distortion, Adaptive filters. .

I. INTRODUCTION
The main purpose of the speech enhancement algorithms is to improve speech quality and signal intelligibility. The objective of speech enhancement is improvement in intelligibility and overall perceptual quality of degraded speech signal using audio signal processing techniques [1]. Speech enhancement is concerned with improving the speech signal which has been degraded by the unwanted signals that is the noise signals which can be in any form that is background noise, traffic noise, exhibition noise etc. [2]. The increased use of speech communication systems over the years goes hand in hand with an increase of the variety of application environments. As a result also the variety of noise sources and noise levels that affect the speech signal increases. Obviously, the clean speech signal is not available in speech enhancement applications, and needs to be estimated from noisy speech data [2]. The need for the enhancement of speech signals originates from a noisy location or is affected by noise over a communication channel [3]. Enhancing of speech degraded by noise or noise reduction is the most important field of speech enhancement and used for many applications such as mobile phones, teleconferencing systems, speech recognition, and hearing aids. The problem of speech signal intelligibility degradation by various types of noise has been widely studied in the past and is still an active field of research. Noise reduction is useful in many applications such as voice communication and automatic speech recognition where efficient noise reduction techniques are required. There are various types of noises which affect the speech signal. The wiener filter, spectral subtraction, log MMSE, MMSE, decision directed approach

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are some of the methods of the noise reduction [3]. The background noise is the most common factor degrading the quality and intelligibility of speech in recordings. The noise reduction module intends to lower the noise level without affecting the speech signal quality. This module is based on the spectral subtraction performed independently in the frequency bands corresponding to the auditory critical bands [3]. The overall objective of this dissertation is to study, implement and compare a number of techniques for enhancement of speech that has been degraded by noise and to check their effect on the speech intelligibility. Three types of noise fields are investigated in multi microphones speech enhancement studies: 1) Incoherent noise caused by the microphone circuitry 2) Coherent noise generated by a single well-defined directional noise source and characterized by high correlation between noise signals; and 3) Diffuse noise, which is characterized by uncorrelated noise signals of equal power propagating in all directions simultaneously [4]. Degradation of speech takes place all around us at any given time, for example, where a person is trying to use their mobile phone in a public place. There is a lot of noise being picked up in the background that needs to be filtered out in order for the user on the other end to hear what the caller is saying to them. Places such as a busy street or a building always have an element of noise that can be picked up when trying to talk to other people. Using a hands-free kit in a car will also pick up interference and noise due to the car or other passengers in the vehicle too [5]. Speech processing is the study of speech signals and the processing methods of these signals. The signals are usually processed in a digital representation, so speech processing can be regarded as a special case of digital signal processing applied to speech signal. Whereas state-of-the-art single-channel noise reduction algorithms for auditory prostheses demonstrate an appreciable suppression of the noise and improved speech quality, they are unable, thus far, to improve the intelligibility of noise-degraded speech signals [5**].** An aspect of speech processing includes the acquisition, manipulation, storage, transfer and output of digital speech signals. The human auditory system is a very robust system; it allows us to be able to talk to people even in a very noisy atmosphere. The addition of high levels of noise can result in a significant reduction of intelligibility of the degraded speech. Therefore, enhancement of speech through noise reduction is often a critical part of speech communication system. This is very true since so many people rely on speech communication to get on with their lives and to do business. Speech enhancement is also sometimes used for “pre-processing” of speech for computer speech recognition systems, since such systems often perform poorly with noisy speech. The problem of speech signal intelligibility degradation by various types of Noise has been widely studied in the past and is still an active field of research [6]. Noise reduction is useful in many applications such as voice communication and automatic speech recognition where efficient noise reduction techniques are required. There are various types of noises which affect the speech signal. The background noise is the most common factor degrading the quality and intelligibility of speech in recordings. The noise reduction method intends to lower noise level without affecting the speech signal quality. There are a wide range of speech enhancement techniques available to filter speech degraded by noise, all with their own way of filtering noisy speech. In this paper a number of the speech enhancement techniques are used. They are very common and popular technique called Spectral Subtraction, wiener filtering, LOG MMSE, MMSE, decision directed approach. These approaches to speech enhancement offer a good ability at filtering the noise from signals that have deteriorated due to additive noise [3]. There are various factors which cause the degradation of the speech signal are as follows [7]. Speech can be corrupted with noise at any stage before it reaches the end listener. The different ways in which the speech can be degraded can be broadly classified as follows. The information that is communicated through speech is intrinsically of a discrete nature i.e. it can be represented by concatenation of elements from a finite set of elements. The symbols from which every sound can be classified are called phonemes’. Each language has its own distinctive set of ‘phonemes’ [8].

When the background noise is suppressed, it is crucial not to harm or garble the speech signal or at least not very badly. Another thing to remember is that quiet natural background noise sounds more comfortable than more quiet unnatural twisted noise. If the speech signal is not intended to be listened by humans, but driven for instance to a speech recognizer, then the comfortless is not the issue. It is crucial then to keep the background noise low. Background noise suppression has many applications. Using telephone in a noisy environment like in streets of in a car is an obvious application. Traditionally, the background noise has been suppressed when sending speech from the cockpit of an airplane to the ground or to the cabin. It is easy to come up with similar examples. Not hearing a grasshopper is a small handicap compared to the situation where the audibility range gets narrower, i.e., powerful sounds become. The main issue of the speech enhancement techniques is concerned with the accurate estimation of the noise statistics, particularly, in real non-stationary environments. The classical estimators are based on voice activity detectors (VAD). Acoustic signals radiated within a scope are linearly imprecise by reactions from walls and other objects. Apart from these reactions, the conditions noise and further interferences are also present. Speech quality of algorithms practiced has been improved and greatly progress has been achieved in the advanced of speech enhancement [9]. In rigorous distinction, little progress has been made in scheming algorithms that can advance speech intelligibility.

II. RELATED WORK

The earliest intelligibility study has been done by Lim [10] in the late 1970s found no intelligibility improvement with the spectral subtraction algorithm for speech corrupted in white noise at −5 to 5 dB SNR. In the intelligibility revise by Hu and Loizou [11], conducted 30 years later, nothing of the eight dissimilar algorithms examined were found to advance speech intelligibility relative to untreated (corrupted) speech. Noise decrease algorithms implemented in wearable hearing aids discovered no considerable intelligibility benefit, but enhanced simplicity of listening and listening comfort [12] for hearing impaired viewers. In short, the ultimate goal of devising an algorithm that would improve speech intelligibility for normal hearing or hearing impaired listeners has been mysterious for nearly three decades. Little is known as to why speech enhancement algorithms, even the most complicated ones, do not improve speech intelligibility. Obviously, one reason is the fact that we often do not have a good approximation of the set noise spectrum, which is considered necessary for the implementation of nearly all algorithms. With the aim of, accurate voice activity recognition algorithms are essential. A lot of progress has been made in the design of voice action finding algorithms and noise estimation algorithms [13] some of which [14] are competent of incessantly tracking, at most, the mean of the noise spectrum. In performing well stationary background noise estimation algorithm are known (car) environments. Substantiation of this was provided by Loizou and Hu [11] where in a small progress (<10 %) in intelligibility was observed with speech processed in car environments, but not in other environments (e.g., gibberish). It has been consider that the small improvement was recognized to the stationary of the car noise, which acceptable for precise noise estimation. This suggests that precise noise estimation can throw in to enhancement in intelligibility, therefore cannot provide significant improvements in intelligibility. since in practice, it is difficult to track perfectly the spectrum of non-stationary noise. For that ground, it is believed that the absence of intelligibility improvement with accessible speech enhancement algorithms is not completely due to the lack of precise estimates of the noise spectrum. In the present paper, we discuss other factors that are liable for the nonappearance of intelligibility improvement with obtainable algorithms. The greater part of these factors center about the fact that nothing of the existing algorithms are designed to improve speech intelligibility, as they employ a cost function that does not inevitably correlate with speech intelligibility. The numerical model based algorithms (e.g. MMSE, Wiener filter), for example, derive the magnitude spectra by minimizing the mean squared error (MSE) among the clean and estimated (magnitude or power) spectra [15]. The MSE metric, however, pays no attention to positive or negative differences between the clean and probable spectra. A positive variation between the clean and estimated spectra would signify shrinking distortion, while a negative spectral difference would signify magnification distortion. The perceptual consequence of these two distortions on speech intelligibility cannot be implicit to be equivalent. The subspace techniques (e.g., [16]) were planned to reduce a mathematically derived speech distortion measure, in order not to differentiate between the two aforementioned distortions. In this paper, we will show analytically that if we can somehow manage or control these two types of distortions, then we should expect to receive large gains in intelligibility. To further maintain our assumption, intelligibility listening tests are conducted with regular hearing. Hearing aid technology is used to boost the speech signal excellence and reduce the hearing loss in such a manner that these hearing impaired group hearing an equal level of the speech signal which is heard by the normal hearing people. In today's technology, speech enhancement methods are broadly used to reduce the noise and to improve speech signal quality with suitable hearing loss [10] generally, the human speech signal is corrupted by the disquieting noise. Noise reduction is almost certainly the most important and most frequently encountered speech enhancement setback. A few examples of the Speech processing applications

- Cell phone (mobile)
- Hands- Free mobile Phones
- In-Car Communication device
- Tele conference Systems
- Hearing Aids
- Voice Coder
- Automatic Speech Recognition systems
- Forensics

In the above mention applications, speech enhancement algorithms are used for removing the noise from corrupted speech.

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III. METHODOLOGY

The block diagram showing the speech enhancement of .wav file using the above described icons in the lab view is shown below in fig. 1. Speech enhancement system developed in LabView. This complete system divided into the two parts. First front panel that consist control signal and second block diagram that consist the back panel control of whole system. Back panel (block diagram) understand by the following way:

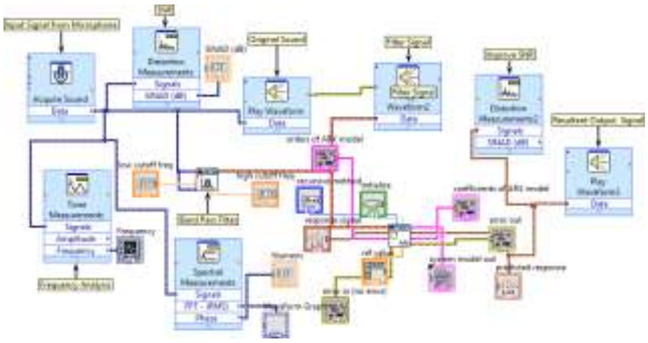

Fig.1. Block Diagram of Speech Enhancement

A. Microphone

The input signal taken from the microphone. The sample signal taken at 22050 sample rate. For the testing purpose we have taken different time response signal at different environments conditions. The output of the microphone signal sends to the speaker to listen original voice. At the same time measurement the speech signal is done and calculate the value of SNR. After that is passes to the band pass filter.

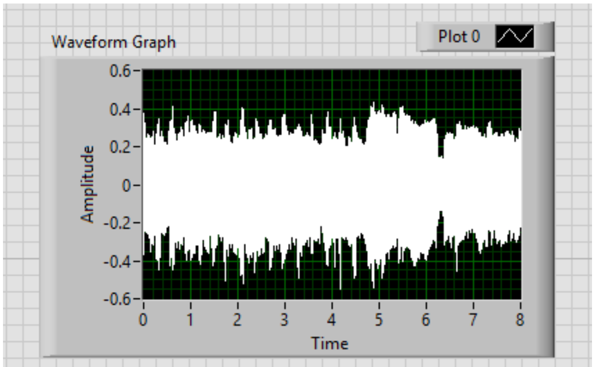

Fig.2. Input Sound Signal

B. Bandpass Filter

Applies a band pass filter to stimulus and response signals. Wire data to the signal in and stimulus signal in inputs to determine the polymorphic instance to use or manually select the instance. The band pass filter is an elliptic infinite impulse response (IIR) filter and has no phase.

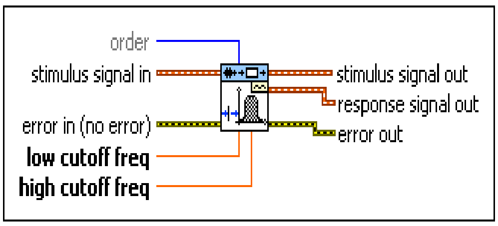

Fig.3. Shows Band Pass Filter

Order specifies the filter order. The default is 6. The value of order must be greater than 0. Increasing the order value generates a sharper transition band from the filter. Low cut off frequency specifies the low cut off frequency of the band pass filter. The cut off frequency must be half of the sampling rate. High cut off frequency specifies the high cut off frequency of the band pass filter. The cut off frequency must be less than the half of the sampling rate and greater than the cut off frequency.

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C. Single Tone Measurements

Finds the single tone with the highest amplitude or searches a specified frequency range to find the single tone with the highest amplitude. You also can find the frequency and phase for a single tone. Searches for a specific frequency in the tone. Contains the following options: Approximate frequency (Hz)—Centre frequency to use in the frequency domain search for the single tone. The default is 10. This option is available only when you place a checkmark in the Search for Specific Frequency checkbox. Search (+/- % of approx. freq.) —Frequency width, as a percentage of the sampling rate, for the frequency domain search for the single tone frequency. The default is 5. This option is available only when you place a checkmark in the search for Specific Frequency checkbox. Results display the measurements you configured this Express VI to perform and the calculated values of those measurements. You can click any measurement listed in the Measurement column, and the corresponding value or plot appears in the result preview graph. Input signal displays the input signal. If you wire data to the Express VI and run it, input signal displays real data. If you close and reopen the Express VI, Input Signal displays sample data until you run the Express VI again. Displays a preview of the measurement. The Result Preview plot indicates the value of the selected measurement with a dotted line. If you wire data to the Express VI and run the VI, Result Preview displays real data. If you close and reopen the Express VI, Result Preview displays sample data until you run the VI again. If the cut off frequency values are invalid, result preview does not display valid data.

The waveform of the .wav file is shown in the fig. 2 that is available in the LabView on which certain parameters including low cutoff frequency, high cutoff frequency, sound format, path and frequency can be specified. By specifying the sampling rate of a wave file, then running the above parameters based model in LabView we get the wave graph. Similarly, we can use other filters to get the SNR value and compare them.

D. Adaptive Filter

The adaptive filter is used after the band pass filter for the signal estimation. The band pass filter passes only the desirable frequency range signal i.e. audio range frequency signal. Then it passes to the adaptive filter which estimated the desired signal. The band pass filter cut the lower frequency signal and high frequency signal and adaptive filter estimate original signal that are mixing of the noise signal. The noisy signal estimated by various adaptive filters and calculate the value of SNR.

At the different conditions different adaptive (recursive) filters are used and decide to remove the noise. Finally we obtain improve SNR value and listen clear voice signal at the speaker.

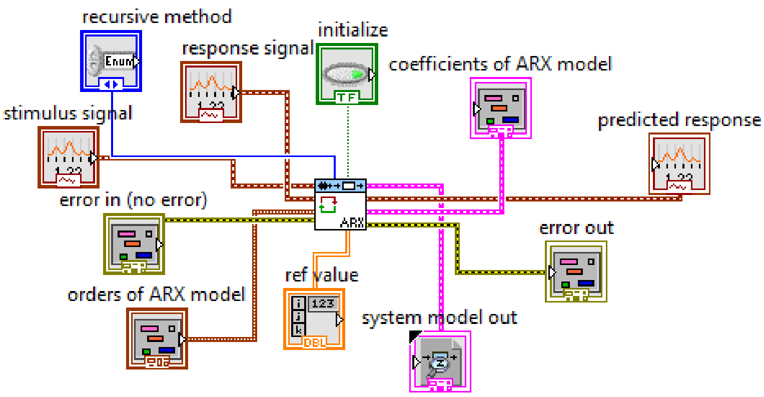

Fig.4. Adaptive (Recursive) Filter

IV. CONCLUSION FUTURE SCOPE

Future work might involve a real time implementation at different environments condition of the system so that the maximum noise is reduced form the audio signals and improve the value of SNR. In the future anybody can extent the order of the different filters and works on higher amplitude signals. They can calculate the efficiency of the filters that they have to implement.

The trend towards good quality signal and data is increasing especially for audio, video and also for medical signals. Clarity in audio gives a comfortable entertainment. Also the need for better quality signals in medical diagnosis helps doctors to diagnose diseases easily and reduces medical errors. Electro Encephalogram (EEG) is a major medical application for adaptive noise cancellers. It also has military applications like over-riding jammers and obtains data accurately. This work can be further extended for medical signals i.e., for both one-dimensional and multi-dimensional

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