CHAPTER 5

APPLICATION OF THE IMPROVED ADAPTIVE FILTER BASED NOISE CANCELLATION TECHNIQUES FOR SPEECH SIGNALS USING VSSNDLMS ALGORITHM

5.1 INTRODUCTION

This chapter discusses about the application of the improved adaptive filter based noise cancellation for speech signals. STT Model is proposed for Hearing Impaired persons, Noiseless Speech Signal is used for STT model for better performance. Here Dragon Naturally Speaking Software is used only for Speech to Text Conversion.

Hearing impaired or deafness is when hearing is affected by a condition or injury. Some people are born with a hearing loss while others may develop it as they get older. Most commonly hearing loss happens with age or is caused by loud noises. Sound waves enter into the ear and cause eardrum to vibrate (Kumbharanana 2007). These vibrations are passed to the three small bones (popsicles) inside the middle ear. The popsicles amplify the vibrations and pass them on to the inner ear where tiny hair cells inside the cochlea move in response to the vibrations and send a signal through the auditory nerve to the brain.

Students who are deaf or hard of hearing face unique challenges when reading, particularly those youngsters who have been deaf since birth. Difficulties continued in the consultation. Understanding the characteristics of students who are deaf or hard of hearing, as well as the communities in which
they live, is an important step toward developing effective instruction and appropriate assessment for these students. Speech Recognition (SR) is the translation of spoken words into text. It is also known as Automatic Speech Recognition (ASR). Speech-To-Text (STT) converts spoken words to text.

Here, the speech signal is stored as Wav file and this speech is analyzed using improved adaptive filters based noise cancellation algorithm. By this adaptive algorithm noise present in the speech signal is removed. Now the speech signal output is given to STT transformation software to convert STT. These signals are processed and at the location of the needy, they are converted to text using necessary algorithms (here dragon naturally speaking). Initially noise is removed as effectively as possible using filters and then the text conversion involves capturing and digitizing the sound waves, converting them into basic language units or phonemes, constructing words from phonemes, and contextually analyzing the words to ensure correct spelling for words that sound like.

5.2 INTRODUCTION TO SPEECH SIGNAL

The speech signal (Kumbharana 2007) is very dependent on the physical characteristics of the speaker. The STT software requires noiseless speech signal for efficient transformation from speech signal to text information. When a person first encounters a speech recognizer he will be in an unfamiliar situation and so will speak to it in a formal manner. However, it is on this occasion that the machine will be trained to recognize his voice. So training the voice is the first step of the STT software.

Speech enhancement in the past decades has focused on the suppression of additive background noise. From a signal processing point of
view additive noise is easier to deal with than convolute noise or nonlinear disturbances. Moreover, due to the busty nature of speech, it is possible to observe the noise by itself during speech pauses, which can be of great value. Thus the goal of speech enhancement is to find an optimal estimate, given a noisy measurement. Here the proposed improved Adaptive filter Based Noise cancellation techniques for speech signals is used to remove or suppresses the noise from speech signals.

Digital Signal Processing (DSP) techniques were at the heart of progress in speech processing during the last 25 years. Simultaneously, speech processing has been an important catalyst for the development of DSP theory and practice. Speech processing involves speech coding, speech analysis, speech enhancement. The advent of desktop computing in the 1980s and 1990s brought affordable speech synthesis and recognition within the reach of the average computer user.

5.3 SPEECH TO TEXT MODEL

Speech recognition, or STT, involves capturing and digitizing the sound waves, converting them into basic language units or phonemes, constructing words from phonemes, and contextually analyzing the words to ensure correct spelling for words that sound like (Kumbharana 2007).
Figure 5.1 Speech Processing Mechanism

Figure 5.1 shows the speech processing mechanism, which shows five blocks user, microphone, sound file, filtering and transformation. The user speaks into the microphone, the microphone accepts the speech sound, sound is stored in *.wav file format, Improve adaptive filter based Noise cancellation is used to remove Noise from Speech signal, STT transformation in Speech Applications.

5.3.1 User

User is a person using a generic system or a person or software using an information system or a party using a telecommunications system. It is an entity that has authority to use an application, equipment, facility, process, or system, or one who consumes or employs a good or service to obtain a benefit or to solve a problem, and who may or may not be the actual purchaser of the item. Here user defines the person who gives the input to the microphone. Here input specifies the speech signal which is given to the microphone.
5.3.2 Microphones

Microphones are devices which convert pressure fluctuations into electrical signals. A microphone is an example of a transducer, a device that changes information from one form to another. Sound information exists as patterns of air pressure; the microphone changes this information into patterns of electric current. The most common microphones in use are dynamic, ribbon, or condenser microphones. Besides the variety of basic mechanisms, microphones can be designed with different directional patterns and different impedances.

5.3.3 Sound File

Waveform Audio File Format (WAVE, or more commonly known as WAV due to its filename extension (*.wav)) is a Microsoft and IBM audio file format standard for storing an audio bit stream on PCs. The usual bit stream encoding is the linear pulse-code modulation format. It stores the sound signal in .Wav format. Windows Waveform format: The WAV format is developed by Microsoft. Many players including QuickTime can play WAV files.

5.3.4 Filter

Adaptive filter is used as a filter, the proposed improved adaptive filter based noise cancellation for speech signal here. The input speech signal is analyzed by the proposed algorithm and noise free signal is obtained. Input speech signal input is given as *.wav file, wave file is analyzed using MATLAB software by improving adaptive filter based noise cancellation algorithm. The output wave file is being generated, the output signal will be a noise free signal.
5.3.5 **Speech Applications**

In speech to text model, the speech signal is converted into text format. The noise free signal is converted to text format using Dragon naturally Speaking software. This transformation of STT is used for hearing impaired persons. Filtered noise free audio is given as input to STT. Thus it converts the speech into text efficiently.

5.4 **WORKING OF SPEECH TO TEXT TRANSFORMATION**

In chapter 4, the results were compared in different environments, performance analysis of different voice samples are discussed. The output wave file is stored and it is given as an input to Dragon Naturally Speaking software and the speech is converted as text.

5.4.1 **Flow Chart**

Figure 5.1 shows the flow chart of the STT model. The user feeds the speech input using the microphone. Microphone being a transducer converts the speech signal into an electrical signal. These electrical sound signals are stored in a file using .Wav extension. The noise present in this signal is removed using filtering process. The adaptive filter is used as a filter. Improve ANC for speech signal is used as a filter to remove noise present in the speech signal. Finally after the filtering is completed perfectly it is converted into text format using STT Software.
Figure 5.2 Flowchart of STT model
5.4.2 Working of Dragon Naturally Speaking Software

Dragon Naturally Speaking Software is installed in personal Computer.

Figure 5.3 Dragon Naturally Speaking Software Installation

When the input signal is given to a computer in the form of a waveform through .Wav file format, the regenerated message will be obtained in a text format in a pop-up window with the aid of dragon naturally speaking software. The Figure 5.3 shows the output of a certain wave pattern to be “Welcome to general training”. After installing the Naturally Speaking Software, the voice is trained. The voice is trained with different modulation and different phases. After training the software it is ready for Transformation.
Figure 5.4 Train Dragon Naturally Speaking for New user

The above given dialogue box gives instruction to the reader for train the Dragon naturally speaking software. The reader carefully reads the information which is interpreted by the computer. Figure 5.4 shows the training stage of the software, voice of the new user is trained. Once the voice is trained the Dragon Naturally Speaking Software is ready for STT.
Figure 5.5  Dragon Naturally Speaking Software testing with different voice

In this window, the Figure 5.5 is shown how the software responds to different voices. In case of different voices, the software checks with the user as to which is the appropriate word. For example, in the above Figure 5.5, it is shown that the word “teaming” corresponds to different voices. So the software checks all other alternatives and confirms it with the user.
Figure 5.6 Dragon Naturally Speaking Software STT Transformation

It can be easily inferred that with the help of dragon naturally speaking software, the speech signals can be converted into texts within no time. Hence, this software makes the process of dictation very much simpler and also doesn’t consume much time. Figure 5.6 shows the software STT Transformation. In word document the speech is transformed into text format. Text “Using Dragon to dictate is much faster than typing on a keyboard” is displayed in word document.
5.5 RESULTS AND DISCUSSION

The Performance of the various Adaptive Algorithms is analyzed in the previous chapter. The analysis proves that proposed improved Adaptive filter based noise cancellation Techniques for speech signals using VSSNDLMS Algorithm is effective and perform well. The outputs of the Noise Suppressed speech signals are used for Speech to Text Transformation. In this section the results of STT software are discussed. Three speech signals and one drum sound signal are considered for analysis. Speech samples are taken from a bus stand, Seminar Hall and Classroom.

Here speech outputs of proposed improved adaptive filter for noise cancellation speech signals and text output of Dragon Naturally Speaking software are shown.

Sample 1: Speech signal observed in the bus stand.

Sample 2: Drums Music Signal recorded in the Studio

Sample 3: Speech Signal observed in Seminar Hall

Sample 4: Speech Signal Observed in Class Room

Sample 1:

When a speech signal observed at a bus stand is fed as input to the STT software, dramatic results were found. The software could transform a speech signal into a Text format. The Figure 5.7 shows the output waveform of the speech signal observed in bus stand without noise.
Figure 5.7  Output waveform of Speech signal observed in bus stand without noise

The output of the improved adaptive filter Based Noise Cancellation technique suppress the noise from the speech signal. This output is fed to STT software for speech to text transformation. Figure 5.8 shows the STT output observed in Bus stand. Text output “the birds can use slide on smooth plank “ is the word observed in the bus stand.

Figure 5.8 STT Output observed in the bus stand
Sample 2:

Figure 5.9 Drums musical signal without noise

Figure 5.10 STT Output observed in the Studio
As an alternative approach, a musical signal is given as input to the system. It can be seen that even though the signal is processed correctly, there is no text equivalent interpretation of the signal thereby making it clear that the input is not a speech signal. So, Figure 5.10 shows only a blank window opens up.

Sample 3:

![Image](image)

**Figure 5.11  Output waveform of Speech signal observed in the seminar hall without noise**

The Figure 5.11 shows the output signal taken from a seminar hall. A noiseless signal is considered. Now this signal gets processed by the software and the texted format of this is shown in the window shown below. The output of the improved adaptive filter Based Noise Cancellation technique suppresses the noise from the speech signal. This output is fed to STT software for speech to text transformation. Figure 5.12 shows the STT output observed in the Seminar Hall.
Figure 5.12 STT output observed in seminar Hall

Sample 4:

Figure 5.13 Output waveform of Speech signal observed in the classroom without noise
The Figure 5.13 shows the output signal taken from a classroom. A noiseless signal is considered. Now this signal gets processed by the software and the texted format of this is shown in the window shown below. The output of the improved adaptive filter Based Noise Cancellation technique suppresses the noise from the speech signal. This output is fed to STT software for speech to text transformation. Figure 5.14 shows the STT output observed in the classroom.

![Image of STT output in class room]

**Figure 5.14 STT output observed in Class Room**

The exact text format of the above given speech signal is obtained with the help of dragon naturally speaking software. Thus four different signals were discussed and output obtained from STT software. First sample Speech signal observed in the bus stand. Second Sample Drums Music Signal recorded in the Studio, Third Sample Speech Signal observed in Seminar
Hall, Fourth Sample Speech Signal Observed in Class Room. Hence this Text output is very much useful for Hearing Impaired persons to know the information.

5.6 SUMMARY

This chapter analyzed the STT model and working on Dragon Naturally Speaking Software. Thus the proposed STT model tested with different signals, this model is very useful for Hearing impaired persons. The results of the speech signals are tested and discussed with STT Model. The speech signals observed in the bus stand, classroom and in seminar hall is analyzed. By suppressing the noise from the speech signal using an “improved Adaptive filter based Noise cancellation techniques for speech signals”, the noise free signals are used in the STT model for speech signal to text transformation. Thus this proposed Algorithm removes noise from the speech signal effectively and use for STT model. This model will be very much useful for the hearing impaired and for person who lose their hearing after learning to speak and hear.