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Improving Hearing Capability by Simulation of Speech Signal Algorithm using MATLAB

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Abstract— Hearing impaired people use sound amplifying device to improve their hearing capability. Such devices are called Hearing aids or deaf aid. In this paper emphases have been laid on designing a Simulink based Model. This model mainly contains a digital filter bank which can separate input speech signal into different bands and then automatically adjust its gain. The Simulink model also contain other blocks like converters (A to D and D to A), compressor and gain control block.

Index Terms— Compression, digital filter bank, gain control, hearing aids, modelling, reduction, simulation

1 Introduction

Human beings receives information from outside environment in the form of audio, visual and sensory responses. They also interpret the received information in a meaningful way. Auditory system is one of them. The human ear is a very sensitive device and the faintest sound detected by it has a pressure variation of about 2×10^{-5} N/m^2. The human ear, may be divided into three parts: Outer ear, Middle Ear and Inner ear [1] [2].

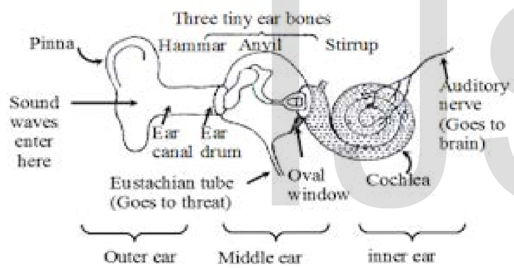

Fig.1. Structure of Human Ear

- The outer ear collects sound waves which are conducted through the auditory canal.
- These waves fall on the ear drum and set it into vibrations.
- The middle ear which is of the size of a small marble and houses osscicles (three bones: hammer, anvil, stirrup) amplifies these oscillations about 60 times.
- The inner ear which contains cochlea and is filled with a fluid converts these pressure variations into electrical signals.
- These electrical signals are conveyed to the brain via auditory nerve for interpretation.

Over the years numerous hearing aids were discovered which processes the input audio signal and changes it into the audible form with the help of signal processor [3].

2 Strategies Used In Digital Hearing Aid

There are varieties of strategies used in digital hearing aid such as

1. n of m
2. Spectral Peak (SPEAK)
3. Advanced Combination Encoder (ACE)
4. Continues Interleaved Sampled (CIS)
5. Hi-Resolution (Hi Res)

The above three strategies uses channel selection scheme. But amongst all these, the CIS strategy is best suited for designing hearing aid [2].

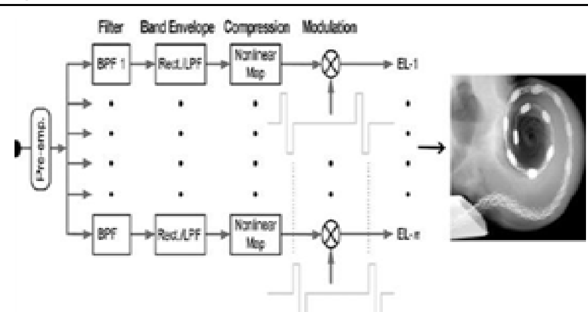

Fig.2. Block Diagram for Signal Processing by Using CIS Strategy

In the figure 2, the speech and other sounds are represented in a simpler manner. In order to represent the fundamental frequencies of speech sounds the cut-off frequency of the LPF is set at greater than or equal to 200 Hz in each envelop detector. As of now, 4 to 24 channels are used in CIS strategy to process speech and other sound signals [5] [6].

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3 Challenges To Signal Processing

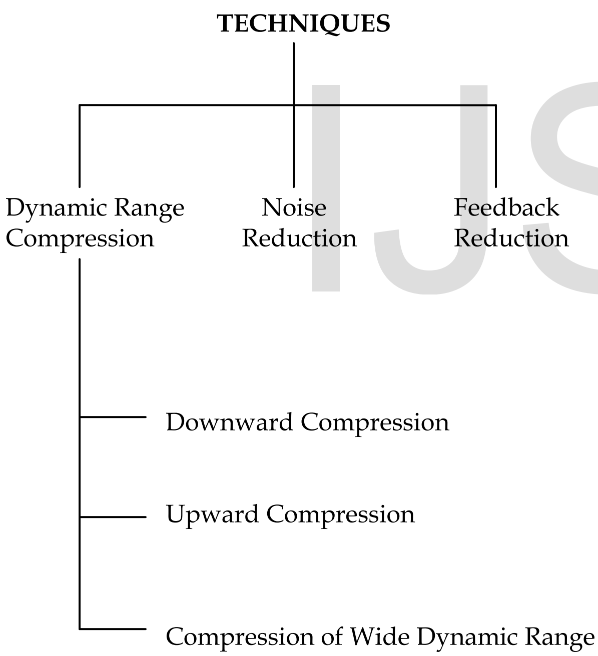
Following are the challenges that are encountered during signal processing of speech and sound signal:

- While using hearing aids, the hearing impaired persons have to face several difficulties like understanding of speech signals along with background noise. This problem can be rectified by using noise reduction technique.
- The maximum gain is limited and the quality of sound is reduced when sound travels from loud speaker to microphone. This problem of acoustic feedback can be rectified by using feedback reduction technique [7].

So, there is a need of high performance processor for the construction of hearing aids.

4 Techniques Used For Signal Processing

The following three techniques are used for signal processing of speech signal to rectify the above mentioned challenges.



4.1 Dynamic Range Compression

The optimum sound level that can be heard comfortably at a specific frequency is defined as Uncomfortable Loudness Level (ULL) and the dynamic range is defined as the difference between hearing threshold and ULL [4].The three types of compression techniques are given below:

4.1.1 Downward Compression

In this type of compression technique the loud sounds are reduced over a certain threshold level of compression whereas there is no effect on quiet sounds. In this technique linear amplification is achieved [8].

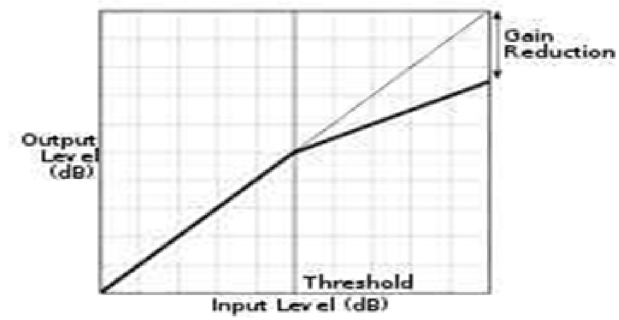


Fig.3. Downward Compression

4.1.2 Upward Compression

In type of compression technique there is an increase in the loudness of sounds below a certain threshold level of compression whereas there is no effect on the louder sounds. In this technique also linear amplification is achieved [8].

Fig.4. Upward Compression

4.1.3 Compression of Wide Dynamic Range

The range of input sound signals to which compression is applied is wide. For no level of sound, the levels of output are condensed.

This compression technique mainly reduces the strong sounds or amplify soft sounds by compressing the dynamic range of audio signal.

4.2 Noise Reduction

The ratio of signal to noise remains same as the amplifier cannot differentiate between speech signal and noise and amplifies speech as well as noise both. In speech processing, the input speech signal consist of various frequency channels and individual channels are analyzed. With the help of digital technology, the unique temporal structure of speech signal can be identified to a certain extent unlike analog technology. Thus, by using this technique the background noise is subdued without disturbing the original speech signal [4].

4.3 Feedback Reduction

The present day hearing aids are facing the problem of acoustic feedback. The drawback of acoustic feedback is that some of the output of hearing aid is fed back to input as there is a very small distance between the transmitter and receiver.

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As a result the feedback as well as the input sound signal are processed simultaneously [7].

This problem can be avoided by adjusting the gain frequency response. In signal channel hearing aid, the overall gain is reduced whereas in multichannel hearing aids the gain can be reduced at particular frequency where feedback occurs.

5 Designing of System Model

The basic block diagram for speech signal processing is given in Fig 5 below.

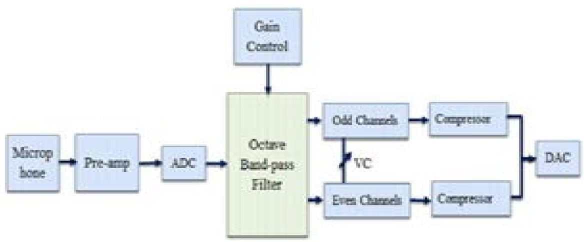

Fig.5. Block Diagram of Speech Signal Processing

6 Software Implementation Of Design

We use MATLAB for implementing the system design model with the help of Simulink. Simulink, created by Math works is a graphical programming condition for demonstrating, modeling and analyzing multi domain dynamic frameworks. Its essential interface consists of a set of libraries and a graphical block diagramming tool. In Simulink, there is a strong combination of the remaining MATLAB environment. It is generally utilized as a part of automatic control and digital signal processing for multi domain simulation and Model Based Design [9] [10]. The figure 6 shows the Simulink model of the above block diagram.

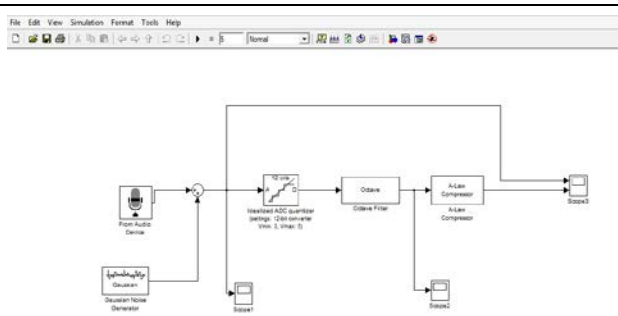

Fig.6. Simulink Model for System

Initially the input audio signal from the audio device is combined with the Gaussian noise which is then fed to A to D converter that converts analog signal to digital one and gives to idealized ADC quantizer. This block converts the smooth signal to a stair step output which is given to an Octave filter. Octave filter is basically a band pass filter which splits the received audible spectrum into even and odd channels or bands by identifying the frequency part of the noise. These bands are called Octave bands.

In this block, automatic gain is adjusted with the help of gain control. Now the output of octave filter is given to A- Law compressor which compresses the input signal by using the A-Law algorithm. Basically, it is 8-bit PCM digital communication system which is used to optimize the dynamic range of an analog signal and finally D to A converter converts the received digital signal into analog form.

7 Results And Discussion

The different output that are obtained from the different scopes of the Simulink Model are described below.

The output from the scope 1 is an amalgamation of signal received from audio device and Gaussian noise generator. It is basically a mixture of original speech signal and background noise.

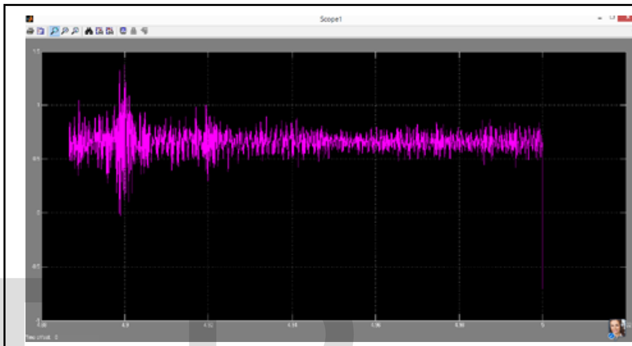

Fig.7. Output of Scope 1

The output of octave filter is obtained across scope 2. In this, the octave filter has converted the input signal into octave bands i.e. speech and noise bands or channels.

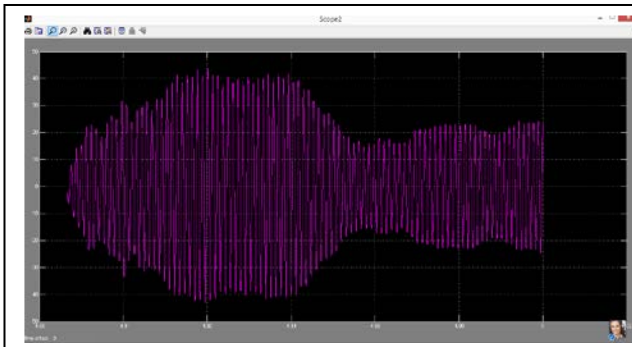

Fig.8. Output of Scope 2

The final output is obtained across scope 3 which is the combination of output of scope 1 and A-Law compressor. In this the A law compressor compresses the noise bands and original speech signal is retrieved with better quality of signal and good signal to noise (S/N) ratio.

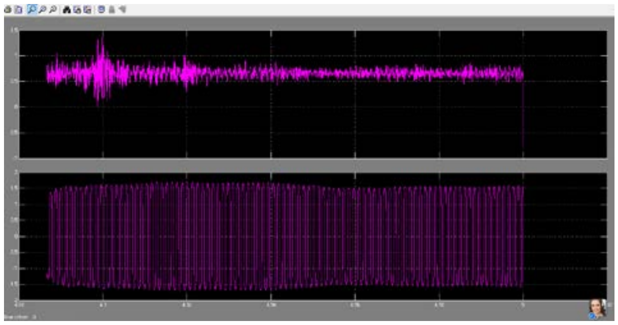

Fig.9. Output of Scope 3

8 Conclusion And Future Scope

A man enduring with hearing disability, the listening devices are the main gadget which makes them capable of hearing the outside condition. The proposed framework gives another signal processing approach for portable hearing assistance. This paper presents the specification of digital aid for hearing in a systematic and comprehensive manner.

As digital hearing aid technology blooms new advancements become harder to create clear designing methodologies have driven applications up to this point, yet future advances will require joint effort crosswise over many fields including psychoacoustics, signal processing and clinical audiology. This work of paper can be extended by using different filters (in the place of octave filter) for different types of noises.

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