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Modelling of Speech Signal Algorithm for Improving Intelligibility in Hearing Impairment

Kaustubh A. Mahakalkar[1], Mahesh T. Kolte[2]

M.E. student, Department of Electronics& Telecommunication, MIT College of Engineering, Pune, India[1]

Professor, Department of Electronics & Telecommunication, MIT College of Engineering, Pune, India [2]

Abstract: As hearing aids partially overcome auditory deficits and normally employed for the hearing impaired people to compensate loss in hearing.This paper mainly focus to design a simulink based model which have a digital filter bank which can separate the input speech signal into different bands and automatically gain is adjust for individual band. The design model consists of analog to digital (A/D) converter, digital filter bank, gain control block, compressor and digital to analog (D/A) converter. The A/D converter converts the analog speech signal into a digital speech signal. Further the input speech signals gets process by digital filter bank and its gain is adjusted. As we know that for a person with hearing impairment the range of threshold levels between weakest sound that can be heard and the most intense sound that can be tolerated is less than normal listener. So, to compensate this factor, hearing aids amplify weak sounds more than they amplify intense sound. The D/A will convert the output processed signal into analog speech output signal. The speaker converts the analog speech signal into an acoustical output signal that is directed into ear canal of the hearing instrument user.

Keywords: Hearing aids, digital filter bank, gain control, compression.

I. INTRODUCTION

The human being is receiving any information in terms of audio, visual and sensory responses. Also they can communicate by taking the complicated information from outside environment and then interpreting it in a meaning way. Auditory system is one of them. Mainly the human auditory system consists of outer ear, middle ear and inner ear. The outer ear is responsible for gathering sound energy and funnel it to ear drum, the middle ear which acts as a mechanical transformer and the inner ear where the auditory receptors (hair cells) are located which accepts the mechanical signal and convert it into electrical signal transfers it to the brain. So, for person suffering with severe sensory-neural deafness, the auditory nerve fibers using electrical stimulation are design [1]. Over the years several hearing aids were developed which have following features in common: a microphone that picks up the sound and a signal processor that processes the audio signal & converts it into the audible form for hearing impaired person [2]. The designers of hearing aids are faced with challenge of developing signal processing strategies that can extract sufficient amount of spectral information from the speech signal.

There are different types of difficulties occurs during signal processing such as: The compression algorithm is a system dependent characteristics since the core of used hearing aid forces the set of allowed algorithms. Noise Reduction is an important stage in the hearing aid signal processing since hearing-impaired people have to understand speech with background noise. The problem of Feedback Reduction produced when the sound goes from the loudspeaker to the microphone. It always limits the maximum gain and reduces the sound quality. So, these factors demand to develop high performance speech processor for hearing aids.

II. ABOUT DIGITAL HEARING AID

All modern hearing aids have the following architecture and functional blocks. There are different strategies such as Continues Interleaved Scheme (CIS), *n*-of-*m*, spectral peak (SPEAK), advanced combination encoder (ACE) and Hi-Resolution (HiRes) [1]. The *n*-of-*m*, SPEAK and ACE strategies each use a channel-selection scheme. In these techniques for the different channels envelope signals are scanned prior to each frame of stimulation across the intra-cochlear electrodes. But amongst all these, the CIS strategy is best. With present-day hearing aids the Figure 1 shows one of the simpler and most effective approaches for representing speech and other sounds. In this case a bank of band pass filters is used to filter out the speech or other input sounds into bands of frequencies from CIS strategy. Then at corresponding electrodes in the cochlea different bands shows different envelope variations.

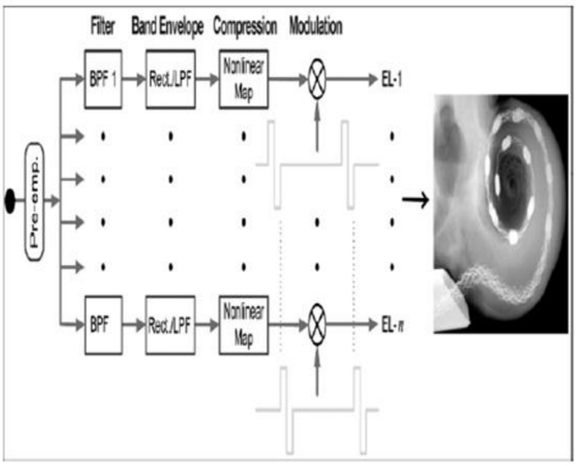

Fig.1Block diagram and signal processing in the CIS strategy.

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Before the modulation takes place, a nonlinear mapping function is used to extract the envelope signals from the band pass filters which get compressed with in order to map the wide dynamic range of sound in the environment (up to about 100 dB) into the narrow dynamic range. The corner or "cut-off" frequency of the low-pass filter is typically set at 200 Hz or higher in each envelope detector, so that the fundamental frequencies (F0) of speech sounds, e.g., 120 Hz for male voices, are represented (exclusively) in the modulation waveforms.Till date CIS(Continuous Interleaved Sampling) scheme uses between 4 to 24 channels for signal processing the speech or other sound signals [4-5].

III.TECHNIQUES USED FOR SIGNAL PROCESSING

As it is seen that there arises different problems while performing signal processing in the hearing aids. Hence, for avoiding those problems there are certain techniques adopted for signal processing.

A. Dynamic Range Compression

For a particular frequency the maximum sound level that can be heard by patient comfortably is called the Uncomfortable loudness level (ULL). So, this difference between the ULL and hearing threshold is called as Dynamic Range [3].

The compression techniques are mainly classified into 3 categories.

1. Low Level Compression

- For signal level amplify factor is reduced below the compression threshold.
- For signals above the compression threshold linear amplification is provided.
- This compression leads for weak to moderate signals.

2. High Level Compression

- For signal level amplify factor is reduced below the compression threshold.
- For signals above the compression threshold value linear amplification is provided.

3. Wide Dynamic range compression

- Compression is applied over a wide range of input sound level.
- There are no sounds levels for which the output levels need to be squashed together closely.

Compression technique is used to reduce the volume of intense sounds or amplifies quietsounds by narrowing or compressing an audio signal's dynamic range.

B. Noise Reduction

The major drawback of present day hearing aids is that they cannot distinguish between speech signals and noise. So the amplifier treats both noise and speech signals in the same manner, and thus it amplifies both speech and noise, As a result no improvement in the signal to noise ratio is done. For speech processing, after separating the input signal into different frequency channels, each channel is analysed individually. Speech has a temporal structure different from most noises. With the help of Digital Signal Processing we can identify this different temporal structure to a certain degree, which is not possible with analog technology. In each of the frequency ranges these results are used to decide whether the signal is more likely speech or noise. This processing technique makes it feasible to suppress background noise without affecting the speech signals [3].

C.Feedback Reduction

Acoustic feedback is another common problem existing with present day hearing aids in the design of hearing aid the receiver and transmitter are place on a small distance from each other so some of the output of hearing aid may get back to the input of the aid. The signal feedback will be processed along with the incoming sound signal. One common method to avoid feedback oscillation is to adjust gain frequency response. In single channel hearing aids this is achieved by reducing the overall gain. In multichannel hearing aids it is possible to reduce the gain only at those frequencies where actually feedback occurs. Also the maximum gain in the channel can be limited to value.

IV.SYSTEM DESIGN MODEL

For implementing the speech signal modelling we have mainly developed a system flow for designing each subsequent block that we required for effective speech processing. The block need to be implemented in matlab Simulink is as given in figure 2 below.

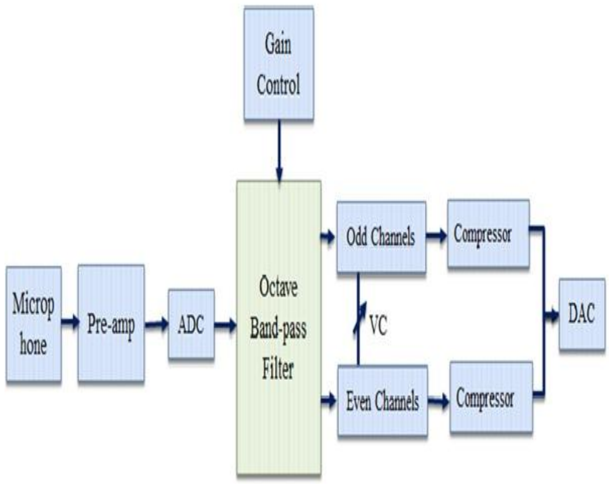

Fig. 2 Block diagram for speech signal processing

V. SOFTWARE IMPLEMENTATION OF DESIGN

For implementing the modelling part we use Math Works product i.e. Simulink. Simulink is a block diagram environment for multi domain simulation and Model-Based Design. It supports system-level design, automatic code generation, simulation and continuous test and also

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verification of embedded systems. Simulink is providing with customized block libraries, graphical editor and solvers for modelling and simulating dynamic systems.Basically it is an integrated part of MATLAB, enabling us to transform MATLAB algorithms into models and export simulation results to MATLAB for further analysis.

VI. RESULTS AND DISCUSSION

Initially the audio signal(.wav format) is added with the noise signal and then given to A/D converter that converts the analog signal to digital and pass to next stage for processing.Then digital filter separate input signal into multiple channels each having particular band of frequencies. After that gain is adjusted internally for required band automatically. Then signal is get compressed and finally D/A converter convert the processed signal into digital form to analog form. The figure 3 below shows complete simulink model designed.

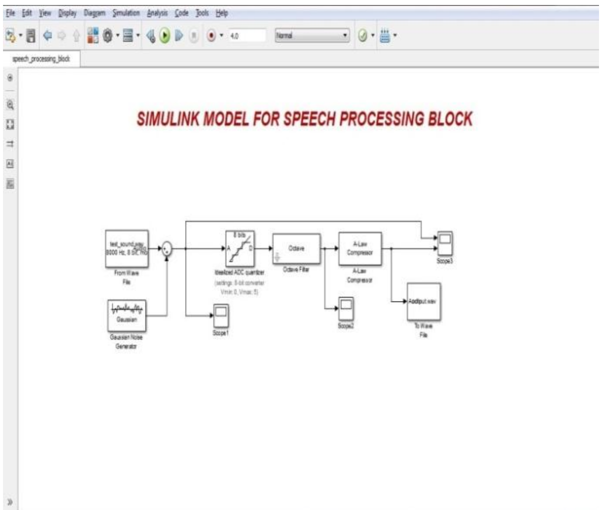

Fig.3 Simulink model for system

The input and the output for system designed can be seen in the figure 4 and 5 as shown below.

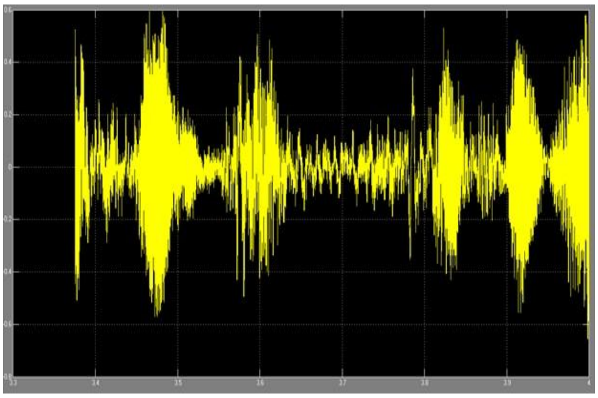

Fig.4 Audio Input to simulink model (.wav)

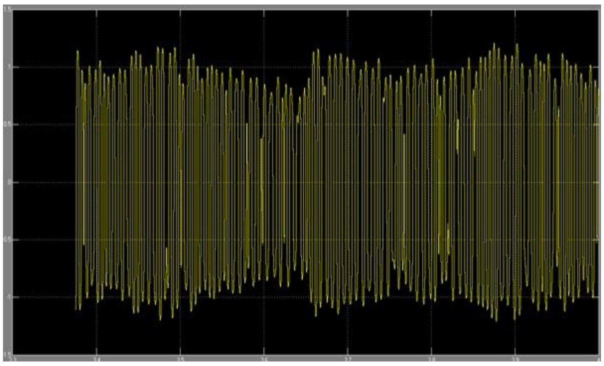

Fig.5 Audio output from Simulink model (.wav)

VII. CONCLUSION

A personsuffering with hearing impairment, the hearing aids are the only device which make them audible to the outside environment.The proposed system gives a new signal processing approach for hearing aids. This presents the systematic and comprehensive design and specificationof digital hearing aid. Simulink based model is developed in MATLAB and further different parameters are assigned to get the desired outcome of it.

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BIOGRAPHY

Kaustubh Mahakalkarhas received his B.E. degree in Electronics Engineering from RTM Nagpur University, India in 2012 and is currently pursuing M.E. degree in VLSI and Embedded systems from Pune University. His domain for project work is embedded signal processing.