DEVELOPMENT OF HEARING AID SYSTEM

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ABSTRACT

Hearing aid device is a small electronic gadget that is fit in or behind the ear to improve one’s hearing consequently communication ability. About 10% of world’s population suffers from some type of hearing loss. Traditional analog hearing aids are similar to a simple radio. They can be tuned and adjusted for volume, bass and treble. But hearing loss is not just a technical loss of volume. Rather, hearing deficiency can increase sensitivity and reduce tolerance to certain sounds while diminishing sensitivity to others. For instance, digital technology can tell the difference between speech and background noise, allowing one in while filtering out the other. Through the use of digital signal processing, digital hearing aid now offers what the analog hearing aid cannot offer. It proposes the possibility of performing signal-to-noise enhancement, flexible gain-processing, digital feedback reduction, etc. In this paper, the simulation of simple digital hearing aid (DHA) was developed using MATLAB programming language. The implementation of this configurable digital hearing aid (DHA) system includes the noise reduction filter, frequency shaper function, and amplitude compression function. This digital hearing aid system is design to adapt for mild and moderate hearing loss patient since different gain can be set to map different levels of hearing loss.

Keywords: -Digital hearing aids, filtering, noise reduction, AWGN

[1] INTRODUCTION

Hearing Aids systems are one of the most important issues for human being. They are a small electronic instrument which makes sound louder and makes speech easier to hear and understand. The hearing aid is designed to pick up sound waves with a tiny microphone, change weaker sounds into louder sounds and send them to the ear through a tiny speaker. The aim of the hearing aid is to amplify sound signals in such a way that they become audible for the hearing-impaired person.

Giovanni Batista Porta was the first to actually describe one of those early hearing aids. Porta wrote a book entitled “Natural magic in which the wooden aids shape animal ears” in 1627. These hearing aid devices were probably not manufactured in the way we know it today. In 1899 Miller Reese and J. Wilson established the evaluation company in Alabama. They held the pattern for the first practical hearing aid which employed a carbon microphone or transmitter, a battery and pair of earphones.

Roughly 10% of the world population bears from some hearing loss. Some people are born with hearing problem some others develop it as they grow. This problem can occur as a result of disease, aging, injury from noise or intake of certain medicines. It is also due to several factors which include
the stigma associated with wearing a hearing aid, customer dissatisfaction with the devices not meeting their expectations. Hearing problems could be that of complete deafness or partially impaired type. Hearing problem could occur after a person learned to talk. Hearing loss is typically measured as the shift in auditory threshold relative to that of a normal ear for detection of a pure tone.

The digital hearing aid system is designed to adapt for Mild and Moderate hearing loss patient since different gain can be set to map different levels of hearing loss\[1\]. Table 1 shows the classification of degrees of Hearing Loss. A hearing aid is an electronic device that makes sounds louder and can help to offset hearing loss. The aim of the hearing aid is to amplify sound signals in such a way that they become audible for the hearing-impaired person.

<table>
<thead>
<tr>
<th>Classification of Hearing loss</th>
<th>Hearing Level</th>
</tr>
</thead>
<tbody>
<tr>
<td>Normal Hearing Loss</td>
<td>-10dB – 26dB</td>
</tr>
<tr>
<td>Mild Hearing Loss</td>
<td>27dB – 40dB</td>
</tr>
<tr>
<td>Moderate Hearing Loss</td>
<td>40dB – 70dB</td>
</tr>
<tr>
<td>Severe Hearing Loss</td>
<td>70dB – 90dB</td>
</tr>
<tr>
<td>Profound Hearing Loss</td>
<td>Greater than 90dB</td>
</tr>
</tbody>
</table>

Table 1: Different degree of Hearing Loss

Basically, all hearing aids were using the analogue technology for the treatment of sound. Improvements have been made by using the development of digital sound treatment for the efficiency of hearing aids. Nowadays, the digital hearing aids are small, which can be hidden inside the ear and have an almost perfect sound reproduction. The research of Digital hearing aids has been growth and now a small programmable computer that are capable in amplifying millions of different sound signals had been constructed in the devices, thus improving the hearing ability of hearing-impaired people. The first digital hearing aids were launched in the mid 80’s, but these early models were slightly unpractical. After ten years later, the digital hearing aids really became successful, with small digital devices placed either inside or discreetly behind the ear\[3\]. Today, digital technology is very much a part of daily life. Most households have a variety of digital products, such as telephones, video recorders and personal computers. Hearing aids also was benefited from the emergence of digital technology. Among the advantages of digital Signal Processing that allows hands free operation.

Though the use of digital signal processing, digital hearing aid now offer what the analog hearing aid cannot offer. It proposes the possibility of signal to noise enhancement, flexible gain processing, digital feedback reduction, etc. In this, the simulation of simple digital hearing aid was developed using MATLAB programming language. The implementation of configurable digital hearing aid (DHA) system includes the noise reduction filter, frequency shaper function and amplitude compression function.

The aid automatically adjusts the volume and pitch on its own. It performs thousands of adjustments per second which results in reduced background noise, improved listening in noisy situations, sound
quality and Multiple program settings. The user can switch between varieties of programs for different listening situations.

[2] METHODOLOGY

Below is a block diagram for the MATLAB implementation of Digital Hearing Aid System. The input speech signal takes the form of human voice. The input speech signal will pass through several functions i.e. noise addition, noise reduction filter, frequency shaper and amplitude compression before producing an adjusted output speech signal which is audible to the hearing-impaired person.

2.1 NOISE ADDITION
Since the input speech signal for this system is a clean signal, some noise is added in order to simulate a real situation. The Adaptive White Gaussian Noise (AWGN) and random noise are added to the input speech signal by using MATLAB function. Noise (AWGN) has a continuous and uniform frequency spectrum over a specified frequency band and has equal power per Hertz of this band. It consists of all frequencies at equal intensity and has a normal (Gaussian) probability density function.

![Block diagram of MATLAB implementation of Digital Hearing Aid System](image-url)
2.2 NOISE REDUCTION FILTER

A major anxiety for the people with hearing loss is the capability of hearing aid to differentiate intended speech signal in a noisy environment. So, to eliminate noise, a reduction filter is used in this system to suppress the noise in the signal, the wavelet filter function is used.

2.3 FREQUENCY SHAPER

One major complaint of hearing aid users is that the hearing aid amplifies all signals rather than the significant signal that they desire to hear. Most hearing impaired has difficulties to hear high frequency signal. Therefore, the frequency shaper is designed to correct for loss of hearing at certain frequencies. It applies high gain for higher frequencies and vice versa.

2.4 AMPLITUDE COMPRESSION

Fundamentally, amplitude compression function is the task of controlling the overall gain of a speech amplification system. Amplitude compression will ensure that the amplified signal will not exceed saturation power. Saturation power is where the sound signal begins to become uncomfortable.

[3] HEARING AND HEARING LOSS

Hearing occur when sound waves reaches the structures inside your ear, where the sound wave vibration are converted into nerve signals that your brain recognised as sound. Our ear consists of three major area, Outer, Middle and Inner ear [4].

Sound wave passes through the outer ear and cause vibration at the eardrum. The eardrum and three bones of middle ear amplify the vibration as they travel to the inner ear. There the vibration pass through fluid in a snail shaped structure in the inner ear(Cochlea).

Cochlea attached to the nerve cell and there are thousands of tiny hairs that help translate sound vibration into nerve signal that are transmitted to your brain. The vibration of different sound affect these tiny hairs in different way causing the nerve cell to send different signal to the brain.
Ageing and exposure to loud sound may cause wear and tear on the hair or nerve cell in the Cochlea that send sound signal to the brain. When these nerve cell are damaged or missing, nerve signal are not transmitted as efficiently and hearing loss occur.

There is one more cause that is build up of earwax. In this case, it blocks the ear canal and protect conduction of sound waves. These can be restored by removal of earwax. And also, due to Glue ear- is a common childhood condition where the middle ear become filled with fluid. Its medical term is Otitis media with effusion (OME) [5].

[4] IMPLEMENTATION & SIMULATION

The code, written in MATLAB, loads the input wave signal, takes the sampling frequency and the number of bits of that signal. Then, Adaptive White Gaussian Noise (AWGN) and random noise are added to the signal before they are processed by various MATLAB function to get an output which is audible to the hearing-impaired person. This signal is added by Adaptive White Gaussian Noise (AWGN) and random noise.

In this simulation, one sample of hearing loss patient is taken from “Hill view Ujjain, Widhan, singrauli, MP”. This patient suffers moderate hearing loss which characterized by:
- Threshold of hearing at 40 dB
- Threshold of pain at 90 dB
- Have difficulties to hear high frequency.

[5] OUTPUT

Fig below of original speech signal which is plotted between time versus amplitude axis.

Fig 2: - Original Speech Signal

Now, Adaptive White Gaussian Noise is added to the original wave signal. The main aim of addition is just to simulate in real life situation.

Fig 3: - Speech signal along with Noise
After that filtering of noise i.e. reduction of noise takes place which removes most of the noise signal present in the speech signal. Reduction of noise from corrupted signal is shown in fig below:

![Signal after denoising](image1)

**Fig 4:** Signal after denoising

Comparing the spectrograms of the original signal and the filtered signal, we can see that the amplitude of the noise in the signal was noticeably reduced as shown in fig. below:

![Spectrogram speech signal of MATLAB](image2)

**Fig 5:** Spectrogram speech signal of MATLAB

The strength of the adjusted signal is increase as our expectation. If the output is not up to our expectation then the cause of this error is due to the gain function improperly implied.

[6] CONCLUSION

The aim of this paper was to design a system that pre-amplify an acoustic signal picked up by a condenser microphone. The pre-amplified signal is then further amplified before being converted to sound by another transducer (speaker). The designed and constructed circuit was tested on different set of people with different degree of hearing problem. In this digital hearing aids system implementation using MATLAB, sound processing is digitalized. Thus, it is possible to refine the sound signal, for instance by reducing noise and improving speech signals. In addition, by using digital technology, the amplification can be done only at the frequencies that the user needs to amplify. This will eliminate the problem with conventional amplifier which amplified the whole
signal including noise. In general, digital hearing aids, when the incoming signals are converted to
digital signals. This digitalization makes it possible to precisely analyse & filter the signals. The
signals can be preprocessed in one or more frequency channels. At the end, the digital signal is again
converted to its analog form. The benefits of using digital aids could improve quality of life by
improving sound quality, higher listening comfort, better communication in noisy environments, better
speech intelligibility in group conversations and more flexibility in case of progressive hearing loss.

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