Hearing Aids System for Impaired Peoples

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Abstract: *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. Approximately 10% of the world's population suffers from some type of hearing loss, yet only a small percentage of this statistic use a hearing aid. The stigma associated with wearing a hearing aid, customer dissatisfaction with hearing aid performance, and the cost associated with a high performance solution are all causes of low market penetration. 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 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 Aids system, filtering, noise reduction.*

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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. With the microchips available today, hearing aids have gotten smaller and smaller and have significantly improved quality. Roughly 10% of the world population bears from some hearing loss. However, only a portion uses hearing aid. This is due several factors which include the stigma associated with wearing a hearing aid, customer dissatisfaction with the devices not meeting their expectations, and the cost associated with the new digital versions of hearing aids [1]. Hearing loss is typically measured as the shift in auditory threshold relative to that of a normal ear for detection of a pure tone. This is why there are many types of hearing aids with a wide range of functions and features to address individual needs. Table 1 shows the classification of degrees of Hearing Loss [2].

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.

Classification of Hearing Loss	**Hearing level**
Normal hearing	-10 dB – 26 dB
mild hearing loss	27 dB - 40 dB
moderate hearing loss	40 dB - 70 dB
severe hearing loss	70 dB - 90 dB
profound hearing loss	greater than 90 dB

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 have 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. The aid automatically adjusts the volume and pitch on it's 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 [4]. The user can switches 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.

2.1Noise Addition

Since the input speech signal for this system is a clean signal, some noise is added in order to simulate a real situation. In this system, 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.

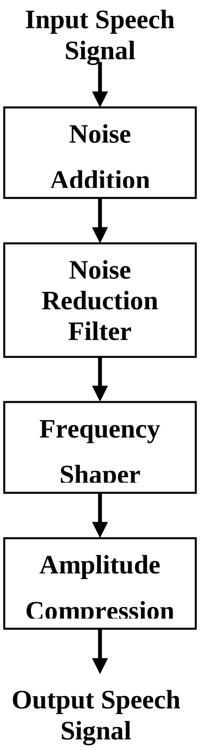

Figure 1: System Block Diagram

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. Hence, to eliminate the noise, a reduction filter function is used in this design. 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 [5]. 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. The typical frequency shaper transfer function is shown in figure 2.

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.

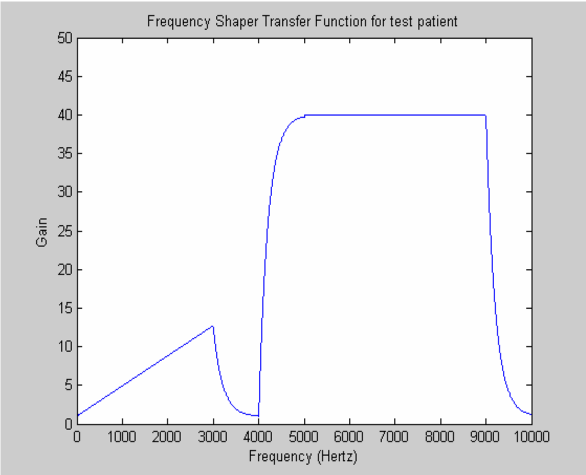

Figure 2: Typical Frequency transfer function

3. 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.
For the analysis purposes, a sample of speech signal is selected. The sample is a male speaker voicing, "I am Mohd Syahril Nizar, a final year student at UIA" This signal is added by Adaptive White Gaussian Noise (AWGN) and random noise.

For simplicity, a Graphic User Interface (GUI) was built to run this Digital Hearing Aid System simulation demo. To run the demo successfully, it is needed to input all the parameters which are maximum gain to be applied, saturation power and four frequency values where the gain changes. Figure 3 below shows GUI of this system.

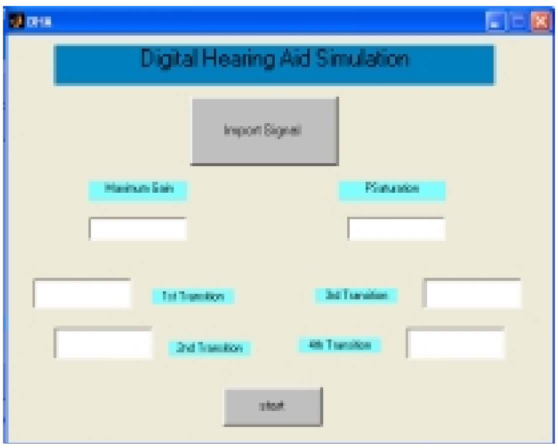

Figure 3: GUI of Digital Hearing Aids

In this simulation, one sample of hearing loss patient is obtained from “Jabatan Audiologi dan Sains Pertuturan, Fakulti sains Kesihatan Bersekutu, Universiti Kebangsaan Malaysia”. 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.

4. Results

Figure 4 below is the original speech signal which is plot on time versus amplitude axis.

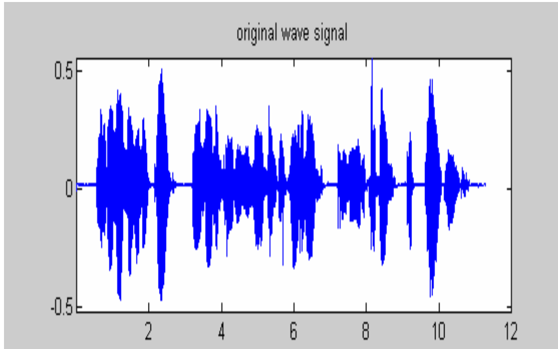

Figure 4: Original Speech Signal

Next, Adaptive White Gaussian Noise is added to the original wave signal. The purpose of this addition just to simulate noises in the real life situation. Figure 5 shows the signal after noise addition.

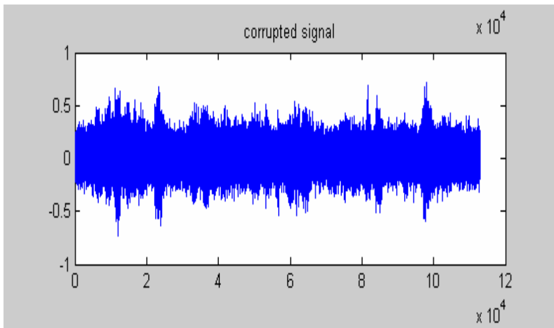

Figure 5: Corrupted Speech Signal

Afterward, the denoising process takes place which removes most of the noise in the signal as shown in figure 6.

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 figure 7.

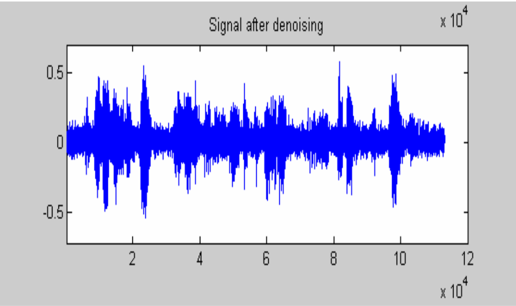

Figure 6: Signal after denoising

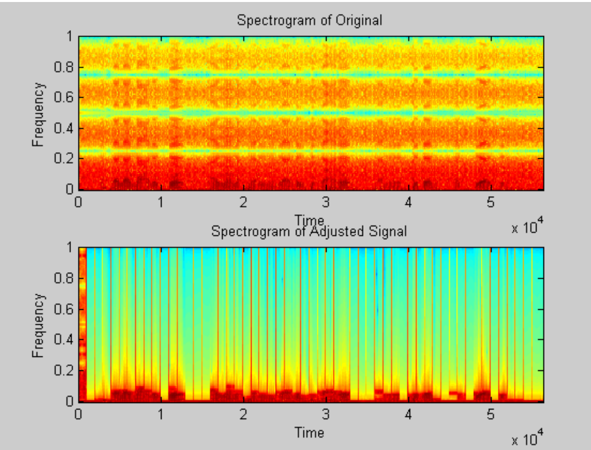

Figure 7: Spectrogram of Original and Adjusted Signal

However the strength of the adjusted signal is not increase as our expectation. Possibly the cause of this error is due to the gain function improperly implied.

5. Conclusion

The newer digital aids offer more ability to fine-tune the sound without distorting the quality and help the listener. 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 its possible to precisely analyze & filter the signals. The signals can be processed 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 environment, better speech intelligibility in group conversations and more flexibility in case of progressive hearing less.

6. References

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