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DIGITAL HEARING AID FOR IMPAIRED PEOPLE (SOFTWARE BASED)

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ABSTRACT

Traditional hearing aids function in a similar way to a simple radio. All the aspects of sound reception. This hearing impairment is more than reduction in magnitude of volume. It leads to reduction in reaction capacities to some whereas increasing this intensity to other particular sounds. Our technology shows the difference between speech and background noise, it allows one sound in and filters out the other. Today, 11.78% of the world's population is suffering from some or the other type of loss in hearing, yet only few of this percentage use a hearing aid. Wearing a hearing aid makes people conscious and uncomfortable, customer are often dissatisfied with its performance, and the cost associated with a high performance solution are the major causes of low market penetration and less people opting for use of a hearing aid. Through this use of digital signal processing, digital hearing aid will offer much more than what the normal hearing aid offers. Through this, we propose the possibility of performing signal-to-noise enhancement, flexible gain-processing and digital feedback reduction among various other applications. In this project, the simulation and analysis of simple digital hearing aid was developed using MATLAB programming language. The implementation of this configurable Digital Hearing Aid system includes functions like noise reduction filter, frequency shaper function and amplitude compression. The target audience for this digital hearing aid system is people with mild and moderate hearing loss since different gain can be set to map different levels of hearing loss.

Keywords : Digital Signal Processing; Digital Hearing Aid; Frequency Shapers; Amplitude Shapers; Noise Reduction Filter.

I. INTRODUCTION

Hearing aids are devices partially designed to overcome auditory deficits and are employed to overcome the hearing-loss in hearing-impaired people. The main aim of a hearing aid is to fit in the dynamic range of speech and common everyday sounds into the limited dynamic range of the affected ear. There are sounds which are totally inaudible and others can be detected because part of their spectra is audible, but may not be identified completely because other parts of their spectra generally high frequencies parts remain inaudible. The range of levels between the weakest sound that is audible and the most intense sound that can be tolerated is much less for a person with hearing damage than a normal listener without this hearing impairment. To compensate for this damage, hearing aids amplify weak sounds more than they amplify the sounds which are more intense.

Neural-sensory hearing loss is the most common type of hearing loss. In this, the root cause lies in the nerve of vestibule-cochlear, the inner ear or central processing centers of the brain. People with this type of hearing loss usually experience abnormal perception of loudness in the form of a slight increase in intensity of sound above

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the limits of hearing and it can be unbearably loud for the people suffering with this aid but the low intensity sounds are inaudible. It diminishes the ability of a person to identify and analyze energy at one frequency in the presence of energy at other. Similarly, a person with hearing loss has decreased ability to hear a signal that is rapidly followed by a different signal.

This low frequency and temporal resolution makes it normal for a hearing-impaired person for noise to mask the speech. In order to avoid such problems, various types of compression hearing aids are suggested. The compression algorithm is a system dependent characteristic because the core of the hearing aid allows algorithms. Other than compression, the main specification to program is the Noise Reduction techniques. Reduction in noise is an important stage in the hearing aid signal processing because hearing-impaired people have to understand speech with noise in the background.

Classification of Hearing Loss	Range of Hearing Loss
Normal	-10 dB to 26 dB
Mild	27 dB to 40 dB
Moderate	40 dB to 70 dB
Severe	70 dB to 90 dB

Table1.1DifferentDegreesofHearingLoss

II. METHODOLOGY

A hearing aid amplifies sound vibrations entering the ear. Surviving hair cells detect the larger vibrations and convert them into neural signals that are passed along to the brain. The intensity of sounds picked up by the receiver of aid are carefully segregated and a pitch is inserted considering the function of the aid.

Noise Reduction Filter

People with hearing loss are often anxious about the capability of hearing aid to separate the useful speech signal in in noisy surroundings. Hence to eliminate this noise, a filter with the ability to reduce noise is used.

A. Frequency Shaper

Sometimes the hearing aid amplifies all the signals instead of the significant signal that the user wants to hear. Losses like these are compensated by frequency shaping and the amplitude of the sound is compressed to control the overall gain.

B. Amplitude Shaper

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Fundamentally, amplitude compression function is the task of controlling the overall gain of the system which amplifies the speech. This ensures that the amplified signal will not exceed threshold power, where the sound signal begins to become disturbing

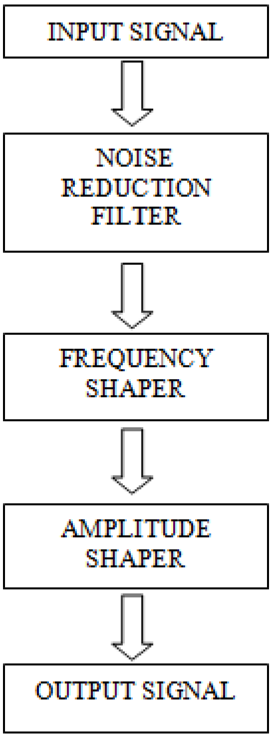

Figure 1. Block Diagram of DHA

III. SIMULATION AND TESTING

This MATLAB code takes the input signal calculate its sampling frequency .Since this analog input consists of noise we assume that noise to be AWGN. For its testing, a sample voice speech is consider which is taken through a microphone that converts the audio speech input to electric signal.

A. Abbreviation and Acronyms

DHA- digital hearing aid, AWGN- adaptive white Gaussian noise.

IV.RESULTS

Fig. 2 below shows the analog input speech signal which is loaded through the microphone

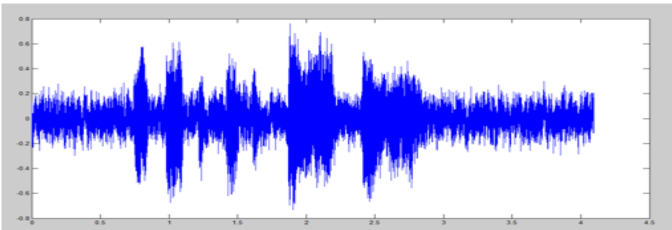

Figure. 2. Input Signal

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AWGN is added to the signal which is shown in fig. 3. This noise is added due to the real life environmental scenarios.

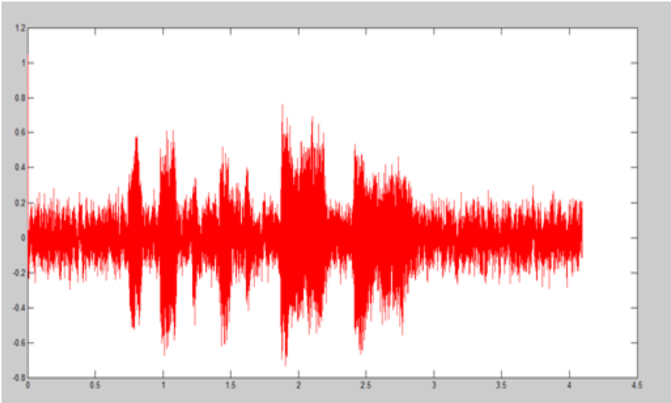

Figure 3. Additive Noise Signal.

The comparative input speech signal to the output signal audible is shown in fig 4.

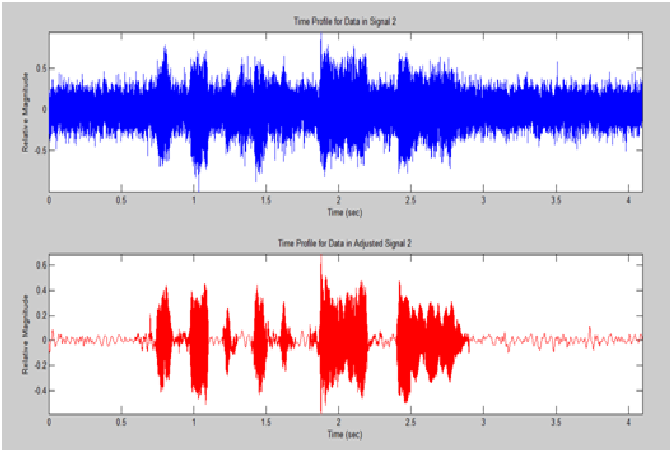

Figure 4.Simulation of Digital Hearing Aid

Further the signal is passed through a spectrogram and the comparative results are shown in fig. 5

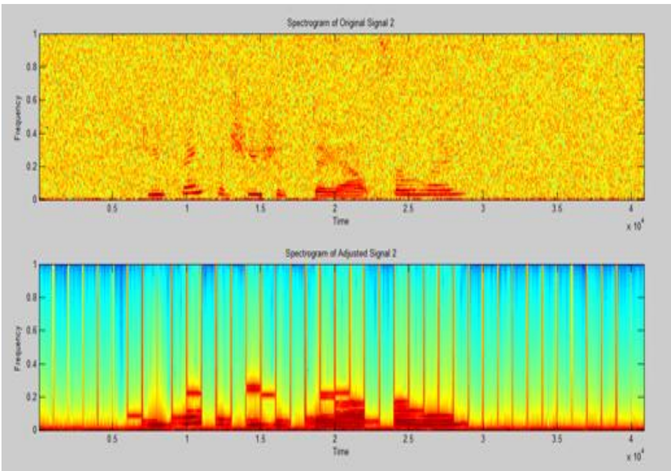

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Figure 5. Spectrogram Results

V. CONCLUSION

In this digital hearing aid system implementation using MATLAB programming language, digitalization of sound processing system takes place. This makes it possible for us to filter the sound signal by noise reduction and improving the speech signals. One of many advantages of using this digital technology is that the amplification can be done only at frequencies that the user needs to amplify. This eliminates the problem with conventional amplifier which amplified the whole signal including noise in the background. In general, DHA, when the incoming signals are converted to digital signals, this digitalization makes it possible to analyze and filter the signals. The signal scan be processed in more than one frequency channels. Finally, the digital signal is again converted to its analog form. The role of the DHA is to maximise the potential of people with hearing impairments by making them comfortable with one-one and group conversations while having flexible utilities throughout

VI. ACKNOWLEDGMENT

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REFERENCES

[1] Harvey Dillon, Hearing Aids. Thieme Medical Publishers, 1st edition, May 2001.

[2] R. Gao, S. Basseas, D.T. Bargiotas, L.H. Tsoukalas, Next-generation hearing prosthet-ics, IEEE Robotics and Automation Magazine, March 2003.

[3] Compression Techniques For Digital Hearing Aids Garini Nikoleta September, 2009.

[4] James M. Kates and Kathryn Hoberg Arehart, Multichannel Dynamic-Range Compression Using Digital Frequency Warping, EURASIP Journal on Applied Signal Processing,vol. 2005, no. 18, pp. 3003-3014, 2005. doi:10.1155/ASP.2005.3003