HEARING AID FOR THE IMPAIRED USING MATLAB

[1]DEEKSHA HS, [2]ANURAG RANJAN, [3]P LEELA PRAVALLIKA, [4] MOHAMMED HASSAN, [5]RAJESWARI P

[1]Student, [2]Student, [3]Student, [4]Student, [5]Assistant Professor
[1]electronics and telecommunication department
[1]Dayananda Sagar college of Engineering,BENGALURU,INDIA

Abstract:
A great proportion of the human population suffers from hearing loss. Hearing loss is not just a technical loss of volume. Hearing deficiency increases sensitivity and reduces tolerance to certain sounds. Hearing loss is a measure of shift in the auditory system compared to that of a normal ear for detection of a pure tone. But with the availability of modern-day technologies and the recent developments, artificial hearing aid systems can be designed that relax the job of damaged auditory systems to a great extent and make much of the sound available to the hearing impaired. In this project, the simulation of the simple digital hearing aid is developed using MATLAB programming language. The implementation of this configurable digital hearing aid (DHA) system includes noise reduction filter, frequency-dependent amplification and amplitude compression. Through this project there is- reduction of white Gaussian noise, increase in the gain for frequencies which were difficult to hear, and shape the amplitude to prevent any of the frequencies from becoming too loud. This digital hearing aid system is designed to adapt for mild and moderate hearing loss patients since different gain can be set to map different levels of hearing loss. Finally, future trends and expected innovations in the hearing aid industry are discussed.

KEYWORDS: Hearing deficiency, MATLAB, Digital hearing aid, White Gaussian noise, artificial hearing aid systems

I. 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. 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 begun its growth and now a small programmable computer that is capable of amplifying millions of different sound signals has been fitted in the devices, thus improving the hearing ability of hearingimpaired people. There are many types of hearing aids with a wide range of functions and features to address individual needs. The aim of the hearing aid is to amplify sound signals in such a way that they become audible for the hearing-impaired person. 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 were also benefited by the emergence of digital technology. Among the advantages of Digital Signal Processing is that it allows hands free operation. The hearing aid automatically self-adjusts the volume and pitch. 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.

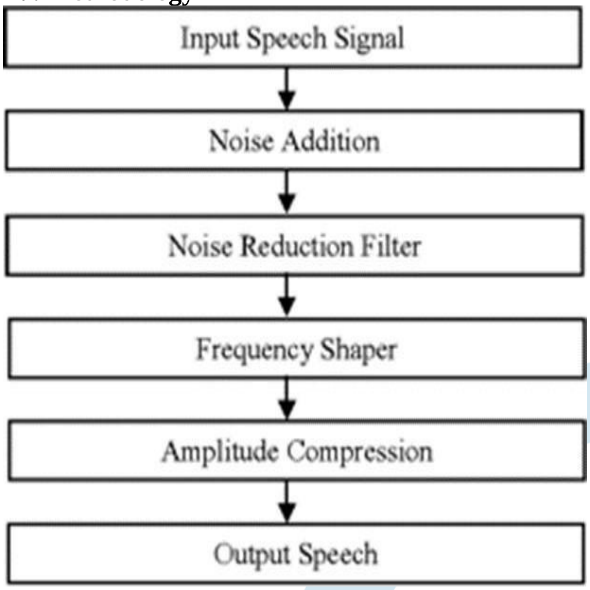
II. Objective of the study

I. To study and understand the working of MATLAB and spectrogram.
II. An application that helps to fit the dynamic range of speech into the restricted dynamic range of impaired ear is the main objective of a hearing aid.
III. The aim of the hearing aid is to amplify sound signals in such a way that they become audible for the hearing-impaired person.
IV. Ability to hear better in noisy situations: with the introduction of dual microphones, we can cancel out much interfering background noise.
V. Improved ease in listening environments (watching television, one-on one conversation), sounds like clocks ticking, refrigerators, computer noise, and footsteps.

III. Scope of the study

I. Can be improved such that it can intelligently detect the required inputs, according to the user's need.
II. To improve the quality of the output signal being produced
III. To be able to remove different types of noise and make it easier for the person, suffering from the hearing loss, to hear the voice.

IV. Methodology



Above 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.

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. In this system, the Additive White Gaussian Noise (AWGN) and random noise are added to the input speech signal by using MATLAB function. The 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.

WHITE GAUSSIAN NOISE: White Gaussian noise (WGN) 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. For example, a hiss or the sound of many people talking can be modelled as WGN. Because white Gaussian noise is random, it can be generated it in MATLAB using the random number generator function, random.

NOISE REDUCTION: 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.

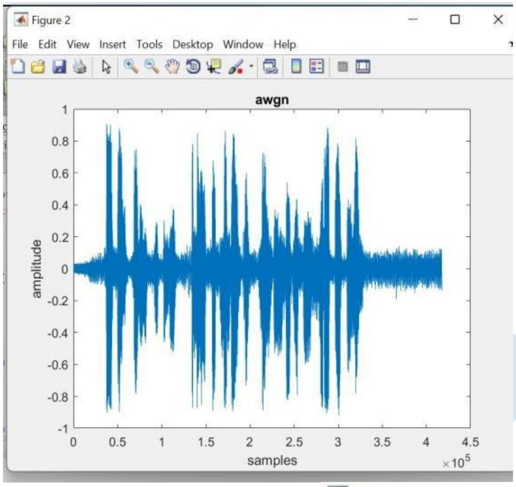
FREQUENCY SHAPER: One major complaint of hearing aid users is that the hearing aid amplifies all signals rather than the significant signal that they desire to hear. Most hearing impaired have difficulty to hear high frequency signals. Therefore, the frequency shaper is designed to correct for the loss of hearing at certain frequencies. It applies high gain for higher frequencies and vice versa. The frequency shaper is designed to correct for loss of hearing at certain frequencies. The filter applies a gain greater than one to the frequencies that the user has difficulty hearing. As one of its parameters, the filter takes in a vector of frequencies, determined by an audiologist that defines the users hearing characteristics. For each range, the frequency shaper applies a certain gain based on the user's specific hearing loss. Thus, our frequency shaper is completely configurable to any user. Implementation of frequency shaper was done using MATLAB.

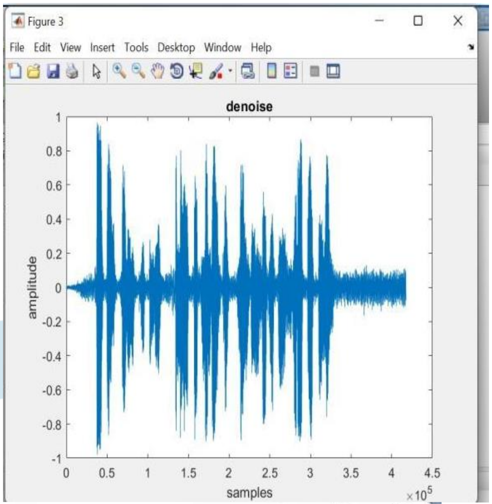
AMPLITUDE COMPRESSION: Fundamentally, amplitude compression function is the task of controlling the overall gain of a speech amplification system. Amplitude compression will ensure that the amplified signal will not exceed saturation power. To implement this algorithm, an amplitude shaper was created using MATLAB.

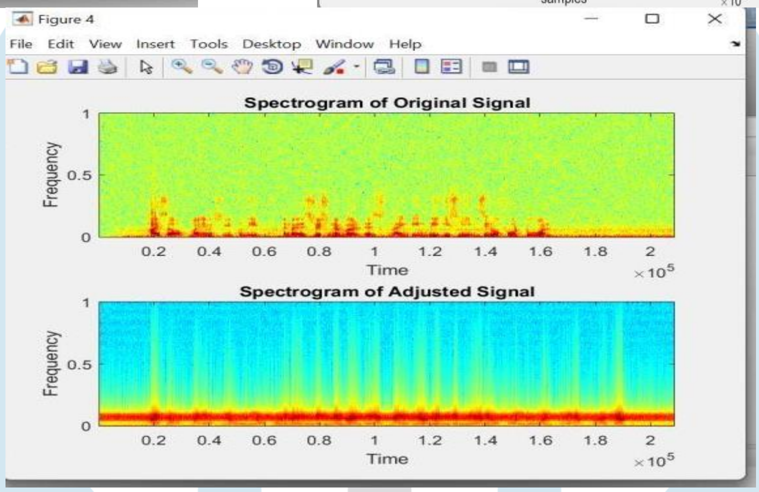
V. Techniques used

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. In this system, the Additive White Gaussian Noise AWGN) and random noise are added to the input speech signal by using MATLAB function.
Denoise: To eliminate the noise, a reduction filter function is used in this design.
ddencmp: It gives default values for all the general procedures related to de-noising.
wdencmp: It returns a denoised or compressed version of the input data X obtained by wavelet coefficients thresholding using the global positive threshold THR.

VI. Simulation Result







VII. Conclusion

The newer digital aid is more capable of fine-tuning the sound without distorting the quality. 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 problems with conventional amplifiers which amplified the whole signal including the noise. In general, digital hearing aid converts the incoming signals to digital signals. This digitalization makes it possible to precisely analyse & 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 can improve quality of life by improving sound quality.

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