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Mini Review Paper

Implementation of Digital Hearing and Speech Aid using MATLAB: A Survey

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Abstract

The analog hearing aids can be tuned and adjusted for volume, bass and treble. Hence they are similar to a simple radio. The digital technology filters the signals in such a way that it allows the speech signals while filters out the background noise. Thus, the digital hearing aids offer various advantages over analog hearing aids through the use of digital signal processing. Thus the implementation of digital hearing aid using MATLAB includes noise reduction filter, frequency shaping function and amplitude compression function. The inability to produce the normal speech is referred as speech impairment. There are various types of speech disorders in which dysarthria refers to that type of disorder in which there is difficulty in production of normal speech. The prototype system senses the vocal chord vibrations which are then amplified. Then the spoken word will be displayed in MATLAB programming language. In this, the digital hearing and speech aid implementation is to be done using MATLAB programming language. Thus, the system is a blend of hardware and software. Thus with this system, the basic needs of impaired people can be understood.

Keywords: Digital Aid system, filtering, Speech Aid system.

Introduction

Hearing aids are devices that are mainly used by deaf people to compensate in hearing loss. To fit the dynamic range of speech into the restricted dynamic range of impaired ear is the main objective of a hearing aid. Some of the sounds are inaudible completely whereas some can be detected because their part of spectra is audible but they are not identified correctly mainly those parts at higher frequencies. Thus a hearing aid mainly amplify the weak signals rather than intense sounds. Sensorineural hearing loss is the most common type of hearing loss. People with this type of loss cannot bear loud sound i.e. a slight increase in sound level above the threshold can be dangerous for them and the low intensity sounds are inaudible to them. Thus a hearing aid system makes the sound louder and makes the speech easy to understand to the impaired people. It is designed in such a way that it picks up the sound signals with a microphone , convert weak signals into loud signals and then, finally send to the ear through a speaker.

The most basic and natural form of communication i.e. speech is used by all the human beings. The mechanism behind the speech production is quite complex. Speech is not just a common sound coming out of the mouth and listened through ears. The speech production mechanism mainly consist of three functions i.e. motor control, articulatory motion and generation of sound. Motor control function is performed by human brain which generates what to speak and send the signals through sensory nerves to the speech production organs. Then the speech production organs receive the signals from the control unit and take the shape of the words or the sound that is to be produced. This is referred as “articulatory motion”. Then finally the human speech generation is done by throwing the air through the mouth and the nasal cavity and the sound is produced.

Table-1
Different Degrees of Hearing Loss

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
Profound	Greater than 90 dB

In most existing work of implementation of digital hearing and speech aid various different techniques and procedures has been used. In first paper of my survey paper Mota Gonzalez G. and Cardiel E. presented a hearing aid based on frequency transposing[1]. In this, age related hearing loss is taken into account. Now M.H. Akhtar and M.S. Baharom proposed various techniques for software simulation of digital hearing aid in MATLAB[2]. For removing the problem of adaptive feedback which exists due to the leakage of sound between speaker and microphone, Ashutosh Pandey and V. John Mathews determines the techniques of offending frequency suppression[3]. Aparna P. and Jeba Jaculin B. presented a hardware circuitory for speech aid which was developed using speech ICs[4].

Organization of Paper: Now in next section we describe the summarized methodology from all reference papers which we have referred. In section III, we will discuss the conclusion and future scope of our work.

Methodology

The method for digital and speech aid are as follows:

For Digital Hearing Aid: Below are the block diagrams of MATLAB implementation of digital hearing and speech aid system. The input speech signal takes the form of human voice. This will then pass through several functions i.e. noise addition, applying filter for reducing the noise, shaping the frequency and finally compression of amplitude before producing an adjusted output speech signal which is audible to hearing impaired person.

Noise Addition: The input speech signal which is a clean signal is simulated by adding adaptive white Gaussian noise by using MATLAB function.

Noise Reduction filter: A major anxiety for the people with hearing loss is the capability of hearing aid is to differentiate intended speech signal in a noisy environment. Hence to eliminate this noise, a noise reduction filter is used.

Frequency Shaper: Sometimes the hearing aid amplifies all the signals instead of the significant signal that the user wants to hear. Hence frequency shaping is carried to remove this loss.

Amplitude Compression: For controlling overall gain of the system, this is to be carried out.

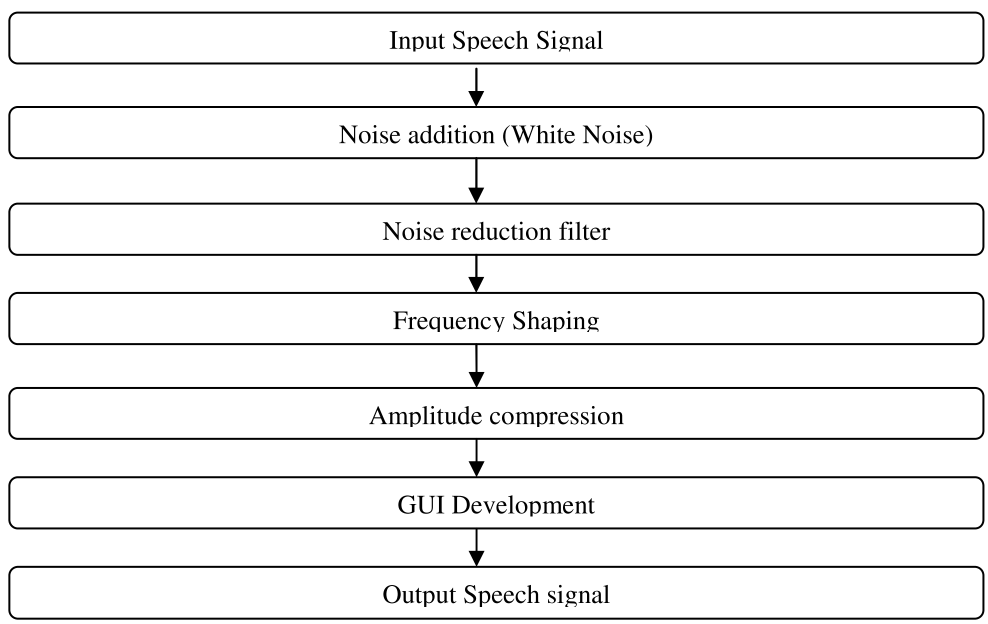

Figure-1
Flowchart for Hearing aid

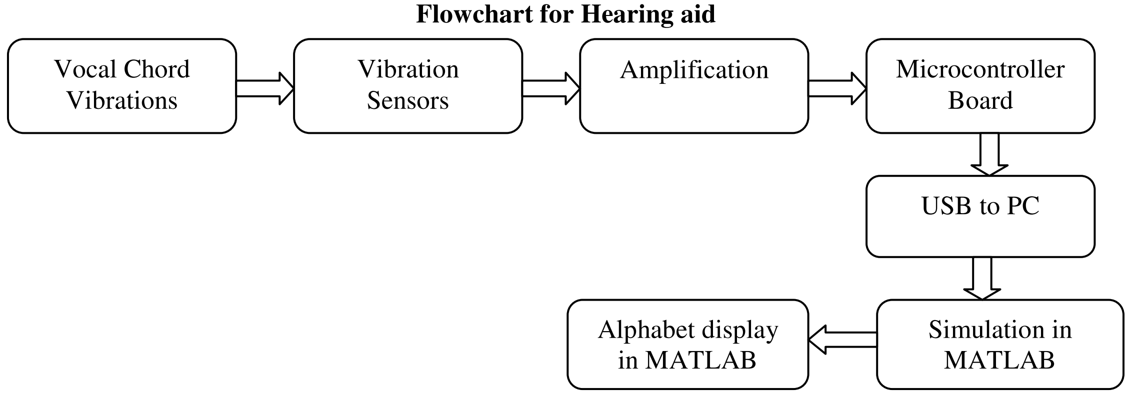

Figure-2
Block Diagram for Speech Aid

In this prototype design, at first the vibrations at the vocal chord will be sensed by the piezoelectric vibration sensors which results in generation of emf of very low magnitude. This emf is then fed to amplification circuit which amplifies the low magnitude signals. After amplification the signals are send to microcontroller board which is then connected to computer via USB. The signals are then realized using MATLAB programming language and the corresponding letters will be displayed in MATLAB screen using coding.

Conclusion

Thus the newer digital aids offers fine tuned sound without distorting the quality of the sound and in this way helps the listener. Thus the output speech signal is a noise free signal from the digital hearing aid. And the speech aid designed using MATLAB could determine the word the impaired person wants to speak with high efficiency.

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