Audiogram Equalizer using Fast Fourier Transform

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Abstract:

The audiogram is a graph, to indicate the hearing capability of a person. An audiogram presents the quietest sounds a person can just hear. The audiogram is used to describe the hearing capability of a person for the various frequencies tested. The hearing loss can be cured by Hearing aids. The very important part of the algorithm to be implemented is to identify the frequency of the audio signal for selective amplification. During the fitting process of a hearing aid, the audiogram of the person is considered as a prescription formula. There are different methodologies used by the researches across the globe to identify the frequency and classify the signals, the most used is the filter bank approach and multi rate filtering. The present work uses Fast Fourier Transform [FFT] to develop the algorithm. The entire system is developed in LabVIEW.

Keywords: Audiogram, Hearing Aid, FFT

I. Introduction:

Many people in the world are suffering from hearing losses. The increasing average age, sound pollution and the growing population are the main reasons. The hearing loss can be cured by Hearing aids. However, study made by Kochkin suggests that only about 50-60% are satisfied with hearing aid instruments. Even though the first electrical hearing instrument was introduced almost 100 years ago, hearing technology even today is a prominent research area. The introduction of the digital hearing aids opened the door for a new world for signal-processing possibilities. Advanced algorithms like noise reduction, feedback suppression, directionality and listening environment classification could now be implemented in the hearing instrument. Digital-Signal-Processors (DSP) will counteract the hardware limitations in the future. The new algorithms have however not resulted in a very good improvement in terms of speech intelligibility. Research on software development for optimizing audio signal processing will be very important for future hearing instruments.

The audiogram is a graph, to indicate the hearing capability of a person. An audiogram presents the quietest sounds a person can just hear. The audiogram is used to describe the hearing capability of a person for the various frequencies tested. It's a measure of hearing handicap a person has. It may be used as a tool to determine the cause for hearing loss. Basically the goal of the audiogram is to determine the lowest signal level a person can hear. The most basic

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audiogram is a screening test. A series of tones at fixed levels are presented to the listener and he indicates which he can hear. The levels are set so that those who hear them should have hearing in normal limits and who don't are referred for a threshold audiogram. The threshold audiogram produces a picture of how a person hears air-conducted signals, such as pure tones. A person is made to sit in a quiet environment, listens for pure-tone pulses and indicates when they are heard. The test consists of presenting these beeps at different intensities at different test frequencies, recording at each frequency the lowest intensity at which there are responses from the listener.

The output from the prescription is the gain-frequency response to be implemented in the hearing aid instrument.

Figure 1 shows an example of an audiogram. The notation used for representing the threshold levels is a Red circles for right ear and blue crosses to represent left ear. On the X – axis, there is a measure of frequency from the lower frequency sounds on the left going to higher frequency sounds on the right. Each red circle and blue cross represents the individual frequencies of sound that have been presented. These frequencies are measured in Hertz.

The Y-axis is a measure of loudness. At the top of the graph are the very quiet sounds, going down to moderate, and then very loud sounds. The red circles and blue crosses are marked on the graph to represent the quietest sound which can be just heard. This loudness is measured in decibels (dB).

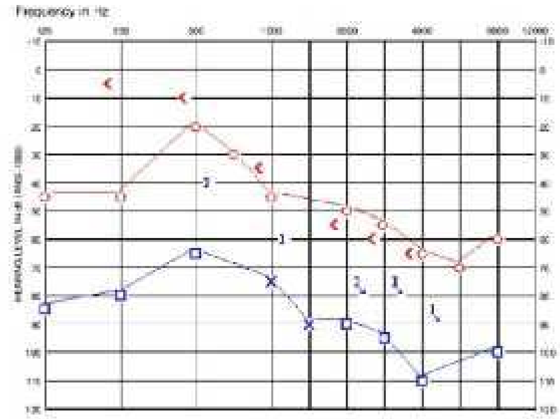

Figure 1 : Audiogram representing the hearing loss

Any points that are heard at 20dB or less are considered to be within the normal range. Lower down the graph the points are plotted, worse the hearing. If an individual's thresholds are all between 40 and 60 dB we say he has a moderate hearing loss. The most common way to increase the hearing ability is to fit hearing aids. However the worse is the hearing loss is, the more difficult it is to fit hearing aids.

During the fitting process of a hearing aid, the audiogram of the person is considered as a prescription formula.

II. Hearing Aid Prosthesis :

A hearing aid is a device designed to improve hearing. Figure 2 presents the three important blocks of digital hearing aid, the Analog to Digital converter gets the analog audio signals from the microphone and converts them into digital format, the Digital Signal Processor (DSP) does the processing on these signals depending on the algorithm implemented on it and finally the Digital to Analog converter converts the processed digital signals to analog, which are then fed to speaker.

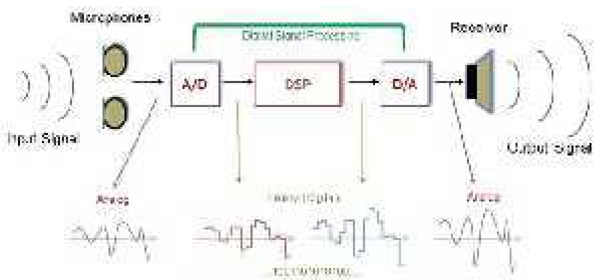

Figure 2 : Block diagram of Hearing aid.

The algorithm to be implemented on the DSP is to amplify the digitized audio signals and compensate the losses of the ear, in consideration with the audiogram of the patient. The amplification is based on the frequency of the signal at that instant of time.

The very important part of the algorithm to be implemented is to identify the frequency of the audio signal for selective amplification.

There are different methodologies used by the researches across the globe to identifythe frequency and classify the signals, the most used is the filter bank approach and multi rate filtering.

This work implements Fast Fourier Transformation (FFT) to identify and then classify the signals based on the frequency.

III. Fast Fourier Transform

The DFT is very important in the area of analysis of frequency because it takes a discrete signal in the time domain and transforms the signal into discrete frequency domain representation. The FFT is a faster version of the Discrete Fourier Transform (DFT). The FFT utilizes some thealgorithms to do the same as the DFT, but in much less time. Equation 1 presents the DFT equation.

$$Yk = \sum_{n=0}^{N-1} x_n e^{-j2\pi \quad /N} \text{…. Equation 1}$$

Where xn is the input sequence, N is the number of elements of x_n and Yk is the transform result. The frequency resolution or the frequency spacing between the componentsof Yk is given by,

$$\Delta f = fs / N \quad \text{….. Equation 2}$$

The table 1 illustrates the pattern of the elements of FFT (x):

Array Element	Corresponding Frequency
Y_0	DC component
Y_1	Δf
Y_2	$2\Delta f$
Y_3	$3\Delta f$
⋮	⋮
Y_{k-2}	$(k-2)\Delta f$
Y_{k-1}	$(k-1)\Delta f$
Y_k	Nyquist Frequency
Y_{k+1}	$-(k-1)\Delta f$
Y_{k+2}	$-(k-2)\Delta f$
⋮	⋮
Y_{n-3}	$-3\Delta f$
Y_{n-2}	$2\Delta f$
Y_{n-1}	Δf

Table 1 : Illustrates the pattern of elements of FFT (x)

With the help of the table we can identify the frequency of the signal after computing the FFT and considering its bin values.

IV. Implementation and Results

As explained in the last section the concept of FFT is used to identify the frequency of the audio signal with known N and Fs the sampling frequency. Once the frequency of the incoming audio signal is known the remaining duty is to apply amplification to the signal in consideration with audiogram of the patient. Hence amplifier is programmed depending on the patient requirements.

The software used to implement the complete algorithm is LabVIEW (Laboratory Virtual Instrument Engineering Workbench) from National Instruments, is a system design platform and development environment for a visual programming language.

Figure 3 is the Virtual instrument (VI) written for calculating the frequency and figure 4 is the VI to implement amplifier to fit the audiogram.

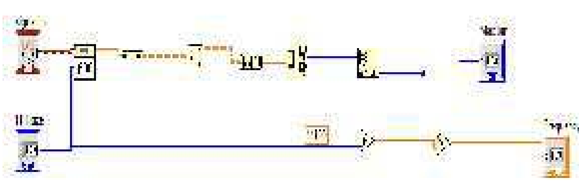

Figure 3 : The VI to implement FFT and Calculate the frequency

Figure 5 represents the audio speech signal which is considered to be the input signal to the patient.

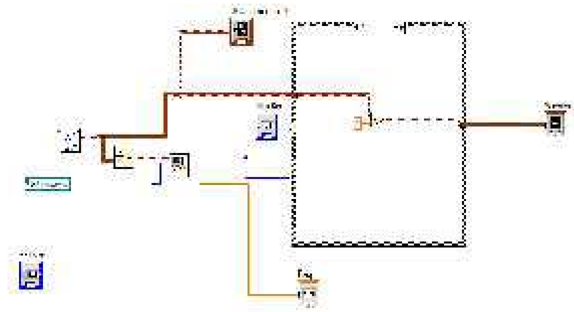

Figure 4 : The VI for amplification

Figure 5: Input speech signal

Figure 6 is the waveform to represent the final output signal after amplification based on the requirement of audiogram.

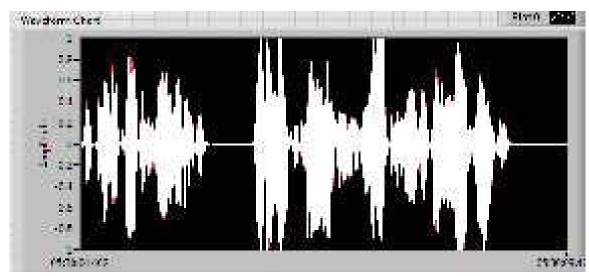

Figure 6 : The output signal after Frequency selective amplification

VI. Conclusion :

The audio equalizer is satisfactorily implemented with the novel approach through finding the frequency of the audio signal and then fitting the signal in consideration with the audiogram. The goal is met with the prescription given by the audiologist to maximize speech audibility and speech comfort for the person with disability.

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