Study Of Digital Hearing Aid Using Frequency Shaping Function

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Abstract

Hearing aids are devices used by hearing impaired people to compensate hearing losses. Digital hearing aids are better in performance over the analog hearing aids, since analog hearing aids work as amplifier. Digital hearing aids are programmable to adjust the gain value, enhance signal to noise ratio, reduction in noise, feedback cancellation etc . In the present work, study of digital hearing aid is proposed to reduce hearing losses by using frequency shaping function. This digital hearing aid provide more flexibility to meet the requirement of hearing impaired people to increase the gain level of audio signal. Noise reduction, frequency shaping and amplitude limiting of speech signal are used in it. In this, signal is manipulated for three different types of hearing impaired people and makes the signal more audible for them.

Keywords- *Digital hearing aid , Frequency shaping , Amplitude shaping and noise reduction.*

1. Introduction.

Hearing Disorders happens normally as the age increases, which causes decrease in normal functionality of ear. Hearing disorders diminished sensitivity to the sounds normally heard [1]. Deafness and speech perception are two categories of hearing losses. People suffering from deafness are unable to understand speech even in the presence of amplification. Another aspect is speech perception, which involves the speech clarity rather than amplitude. Hearing disorders may occur due to birth defects or it may occur due to accident.

The two main categories of hearing losses are conductive and sensor neural. Hearing loss can also be attributed to a combination of both types, a mixed hearing loss [2].

Conductive hearing loss is caused by any obstruction that prevents sound waves from reaching the inner ear. Some of the causes of conductive hearing loss can include: An accumulation of earwax, a collection of fluid in the middle ear or due to Middle ear infections.

Sensor neural hearing loss refers to problems in the cochlea or the auditory nerve. Most are due to deterioration of the tiny inner or outer hair cells. This accounts for 90% of permanent hearing losses and although it may be a natural part of aging other causes can include due to Head injury, certain medical treatments such as chemo- and radiation therapy, Genetic predisposition, Sensor neural hearing losses cannot currently be corrected medically. It is quite possible for a conductive hearing loss to occur together with a sensor neural hearing loss. When this occurs, the hearing loss is referred to as a mixed hearing loss.

Noise-induced hearing loss**,** a common cause of hearing loss is caused through prolonged exposure to harmful sound (noise) or a sudden brief but intense noise like an explosion to the ear. The nature of noise-induced hearing loss is sensor neural, and at present, can only be helped and not cured.

The audible frequency range for human ears is 20 Hz to 20 kHz and human hearing is most sensitive in the range of 1 kHz to 4 kHz [3]. Hearing is measured in decibels. In all frequencies 0 to 20 db is the normal hearing range. To identify the hearing loss pattern, technical researches have been conducted and conclude some plots, like spectrograms and other is audiogram. With the help of Audiogram, hearing can be plot on graph or in other words audiogram may define as**,** a graph that shows the audible threshold for standardized frequencies as measured by an audiometer[4].The horizontal axis of audiogram represent frequency in Hz, vertical axis indicates the amplitude in db as shown in Fig 1. A spectrogram is a representation of how the frequency content of a signal changes with time. Time is displayed along the x-axis, frequency along the y-axis [5].

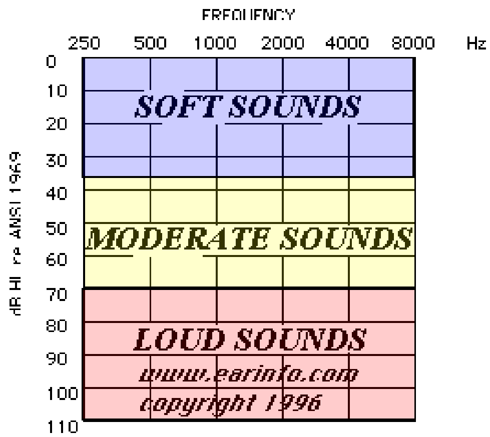

Figure1. Audiogram

At the end of a hearing test , hearing levels decide the degree of hearing loss. Hearing loss is measured in decibels hearing level (dB). If person can hear sounds across a range of frequencies at -10 dB-20dB it a will considered as having normal hearing. The thresholds for the different types of hearing loss are as follows:

Table 1 Degree of hearing losses

Classification Hearing Loss	Hearing Level
Normal Hearing	-10db to 26db
Mild Hearing Loss	27db to 40 db
Moderate Hearing Loss	40 db to 70 db
Severe Hearing Loss	70 db to 90 db
Profound Hearing Loss	Greater than 90 db

To regain the lost hearing pattern there is a requirement of hearing aid. A hearing aid is a device that can amplify sound waves in order to help a deaf or hard-of-hearing person hear sounds more clearly. Hearing losses typically occurs non-uniformly over the audio frequency range. To compensate hearing losses, amplifiers are used in hearing aid which amplify the weak signals from the microphone uniformly over the audio range .The design of filter is another option to compensate losses, it depends upon threshold of hearing. The threshold of hearing is defined as the audio power level at which a person can no longer hear a signal at a given frequency.

Now days digital hearing aid are preferred over analog hearing aid, as these are programmable. Digital hearing aids have greater precision in adjusting electro acoustic parameters with self monitoring capabilities, have also feature of acoustic feedback .Noise in hearing aids can be reduced by advanced signal-processing techniques it also helps in automatic control of signal level[6].

In this proposed work, functions are designed to reduce noise level of signal , increase the gain on specific frequency ranges at which people have difficulty to hear and also do amplitude limiting to maintain the signal level within maximum and minimum limits.

2. Historical developments in digital hearing aids.

As the technology developed there are so many advancement and change in designing process of digital hearing aids. Few of them are discussed below:

- The design of digital hearing aid with digital IIR filter was discussed in paper [7]. The structure of filter was consisting a combination in parallel form of IIR (Infinite Impulse Response) a low-pass, a band-pass and a high-pass filter. This study has shown an advantage of IIR filter can gives a good result in the low complexity digital hearing aids which leads to low hardware resources requirement and low power consumption for VLSI design. The filter coefficients of their IIR filter were obtained from the optimization procedure by genetic algorithm (GA.) The error between desired magnitude response and actual magnitude response was minimized by GA.
- Uniform filter bank design for digital hearing aid application has meet the following requirements such as strong alias suppression, with low delay and sufficient frequency resolution[8] which simplifies sub sampling of the bands and thus helps saving power. Especially with more and more processing inside the frequency bands, sub sampling was

one of the most effective ways to reduce computational complexity by a far amount.

- The focus of this [9] paper was on the algorithms used to build digital compression systems. Of the various approaches that could be used to design a digital hearing aid, this paper considers broadband compression, multi-channel filter banks, a frequency-domain compressor using the FFT, the side-branch design that separates the filtering operation from the frequency analysis, and the frequency-warped version of the side-branch approach that modifies the analysis frequency spacing to more closely match auditory perception.
- In the application report[10], was describe the development of a low power binaural wearable digital hearing aid platform based on the TMS320C5000E fixed point digital signal processor (DSP).This platform was a real-time system capable of processing two input speech channels at a 32 KHz sampling rate for each channel and driving a stereo headphone output. It provided for frequency shaping using multichannel FIR filters, noise suppression, multiband amplitude compression, and frequency dependent intraoral time delay algorithms.
- In [11], a 3-channel VFB approach has been proposed for digital hearing aid applications. As compared with traditional filter-banks with fixed bandwidths, the proposed 3-channel VFB was constructed by parallelizing variable low pass, band pass and high pass digital filters whose both magnitudes (gains) and bandwidths could be independently tuned for matching various hearing loss patterns. To reduce the computational complexity, had theoretically proved the numerator coefficient-symmetries of the variable digital filters such that the symmetries could be exploited for reducing the number of multiplications.

In this proposed work choice of these shaping functions enhance noise reduction and provide flexible gain. Motivation behind this work comes from [12], in proposed work three different frequency shaping functions with different bandwidths are designed for three different types of hearing losses.

3. Methodology

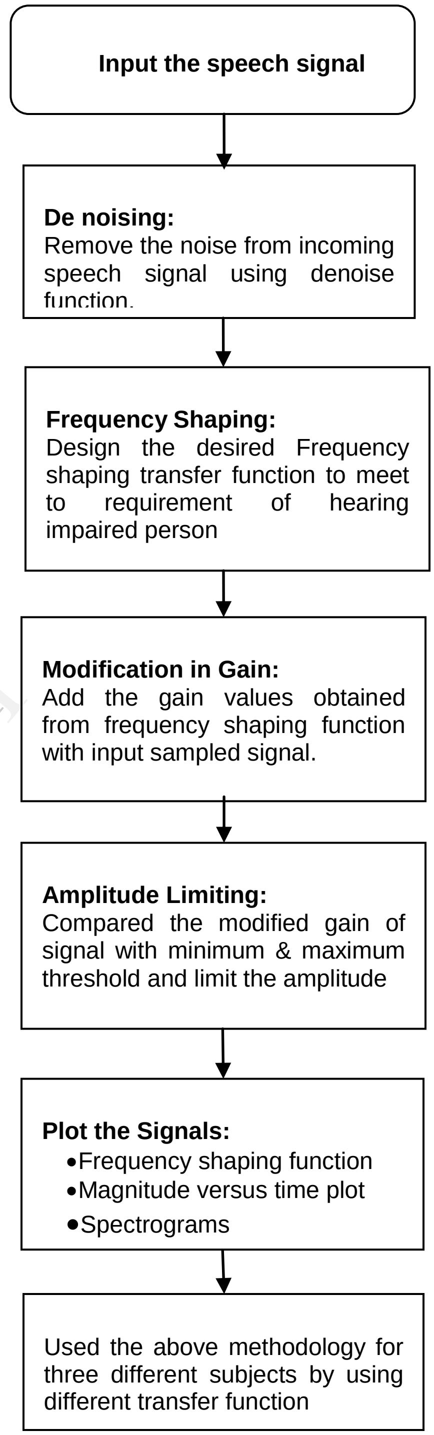

Figure 2.Flowchart for process

As Described in flow chart incoming speech signal is digitized to process by DSP. Signal is Firstly denoised ,then pass through process of frequency shaping and amplitude limiting to keep signal amplitude within limit. The processed signal will audible to hearing impaired people.
The detail description of methodology is as follows:

Step1.The objective to design hearing aid is to reduce moderate hearing loss for subject 1,who has

- Difficulty to hear high frequency
- Threshold of hearing 45db
- Threshold of pain 90db

Step2.The noisy speech signal has been input as wave file.

Step3. Firstly de noise function has been used to remove noise from speech signal.

Step4. Frequency shaping Filter

- Initialized the frequency vector for frequency range 1000 to 10,000 Hz.
- Time domain signal was converted to frequency domain by using FFT function.
- Initialized the gain vector of length N,with zero, to store gain co-efficient.
- The minimum and maximum limit of desired gain was 1 & 45dB respectively.
- Set the gain for first set of frequencies
  - ➢ As the signal was digitized in the form of samples, the total number of samples for incoming signal was divided into different frequency ranges on entire frequency vector of signal by maintaining same sampling frequency.
  - ➢ K was used as pointer for number of samples has been modified.
  - ➢ For getting desired transfer function till 1000Hz, a linear function has been generated which is (K/(N*T)) * gain co-efficient. N gives the total number of samples, whereas T was sampling period.
  - ➢ Hence gain increased with linearity upto some value till 1000 Hz.
- Set the gain for second range of frequencies
  - ➢ The pointer K would be modified.
  - ➢ The modified frequency range was 1000 to 2000 Hz.
  - ➢ The desired transfer function has been plotted as decaying curve for this frequency range, as shown in fig 3.
  - ➢ Hence decaying exponential function was created with time constant of (K/(N*T)-1000)/f.
  - ➢ Whereas f signifies the range of frequency where curve decays.
- Set the gain for third range of frequency.
  - ➢ For frequency range 2000-300Hz, the desired transfer function is rising exponential.
  - ➢ The rising curve has been plotted as shown in fig 3, using rising exponential function with time constant of (K/N*T - 2000).
  - ➢ Final value of curve is G.So there was + G factor with rising exponential function.
- Set the gain for fourth range of frequency.
- Set the gain for the fourth range of frequency 3000 - 6000Hz, gain remains constant with value G.
- Set the gain for fifth range of frequency 6000-7000 Hz, the gain curve is again decaying exponentially with final value 1.
- Set the gain for sixth range of frequency 7000-8000Hz, there was linear rise in gain curve from 1 to some value of gain.
- Set the gain for the seventh range of frequency 9000-10000, curve is again decaying exponential.

Step4.After obtaining frequency shaping function, the gain co-efficient from gain vector has been added to incoming signal to shape the signal according to transfer function.

Step5 .Amplitude Shaping.

- To limit amplitude of signal, the amplitude level of individual sample has been compared with minimum and maximum threshold levels
- If the signal level is more than Psat then reduce the signal level to P_{sat}.
- If signal level is less than P_{min} then reduce the signal level to zero.

Step6.By using above methodology frequency shaping transfer function, spectrogram & magnitude versus time has been plotted.

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Step7.The same methodology is implied for three different types of subjects having different hearing losses, which is being compensated by changing frequency shaping function.

4. Results & Discussion

In this section, performance of the proposed approach has been evaluated through the simulations in MATLAB. The input speech signal is converted to digital signal, then processed by Matlab coding to denoise the signal, generate frequency shaping function to shape frequencies and by amplitude limiting generate the adjusted or modified signal which is audible to patient.

In this paper, three different degree of hearing losses of different patients have been taken as objective to be modified.

For Subject 1: Subject 1 has moderate hearing loss at medium frequency range of 3000 - 6000Hz.The required gain is 45db on this frequency range. So the frequency shaping or modified function is show in Fig.3

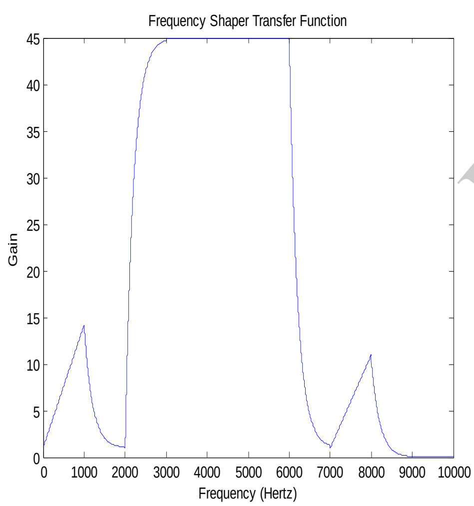

Figure 3. Frequency shaping function 1

By using frequency shaping function show in Fig 3, gain of speech signal has been modified on specific frequency range, and with amplitude limiting function the processed signal has increased the gain within limits, Fig 4 shows the relative magnitude of input speech signal and processed output signal/adjusted signal with no noise.

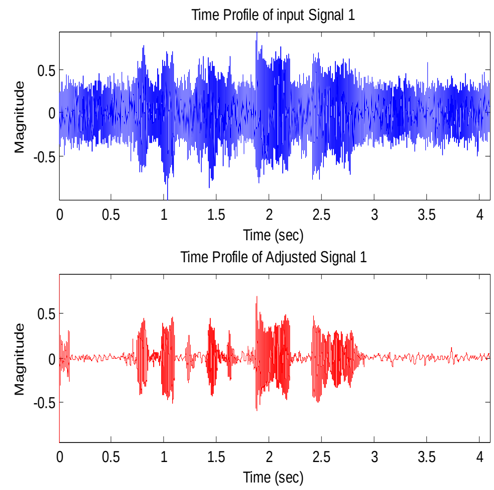

Figure 4.Magnitude versus time plot 1

spectrogram is plot of short time fourier transform of signal or to plot the different frequencies on time axis. Spectrogram for input shows the misaligned frequencies with added noise and

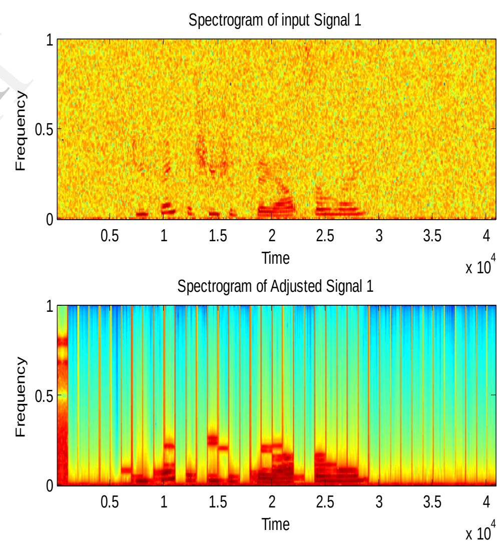

Figure 5. Spectrogram 1

spectrogram of adjusted signal has shown the aligned frequencies as shown in Fig. 5.The brighter color shows more energy.

Subject 2 has severe hearing loss at low frequency range (500-900Hz), so there is need to raise gain up to 60 db at this frequency range.

The gain co-efficient obtained after frequency shaper transfer function, for subject 2 is shown in Fig.6 has been added to input sampled signal.

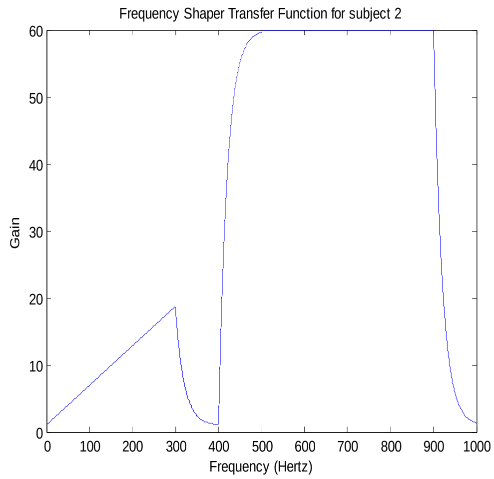

Figure 6.Frequency shaping function 2

In Fig. 7 has plot the relative magnitude response for input as well for adjusted output signal for the hearing impaired people suffering from low frequency losses.

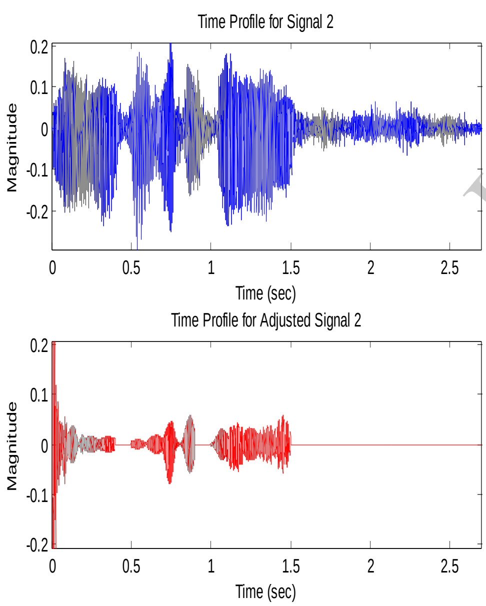

Figure 7. Magnitude versus time plot 2

Spectrogram for subject 2 has shown in Fig. 8,blue color indicates no signal region.

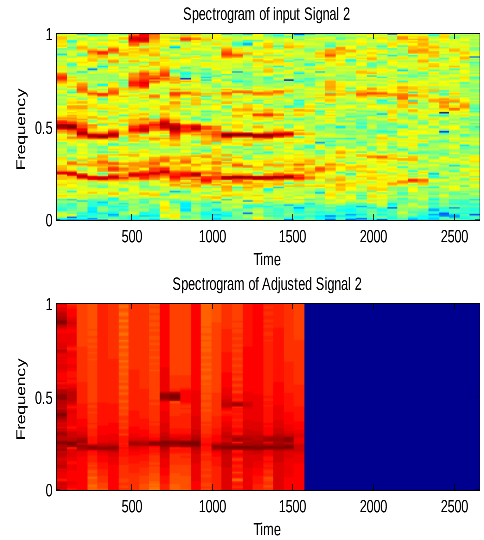

Figure 8 Spectrogram 2

Subject 3 has found mild hearing loss at medium frequency range over 1000 to 5000Hz.So there is required gain of 30 db, on frequency range 1000 to 5000 Hz.

The frequency shaping function for subject 3 is shown in Fig 9 ,which is required to manipulate gain on specific frequencies.

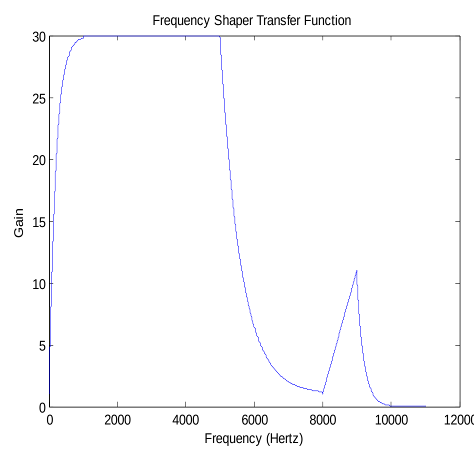

Figure 9. Frequency shaping function 3

Relative magnitude of input speech signal and adjusted signal for subject 3 is shown in Fig. 10.

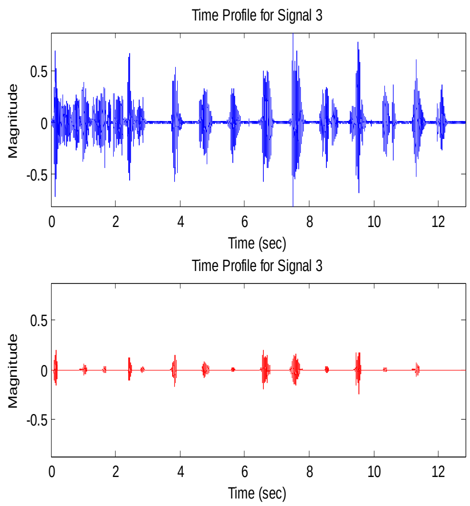

Figure 10. Magnitude versus time plot 3

Spectrogram of adjusted signal for subject 3 in comparison to input signal is shown in Fig.10 the blue region shows no energy.

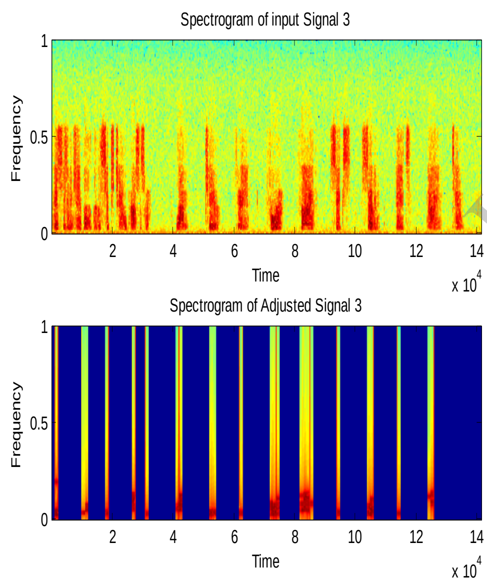

Figure 11. Spectrogram 3

5. Conclusion

It was observed ,from the above investigation that, by using the different functions like denoising, frequency shaping, amplitude limiter on speech signal we can make the signal to be more suitable for hearing impaired people with less complexity and more flexibility. The gain on specific frequencies has been enhanced by adding frequency transfer/shaping function with the input discrete signal. The frequency transfer/shaping function has been designed according to patient's requirement/hearing loss. In this research, three types of subjects have been taken with different types of hearing losses and tried to reduce the losses by providing gain on that particular frequency ranges with other desired functions.

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