Digital Hearing Aid for Ski-Slope Hearing Loss

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ABSTRACT
Hearing loss can be defined as total or limited loss of hearing capacity. Noise is one of the cause of hearing loss. Hearing aid is defined as an electro acoustic device which amplifies sound. 5.3 percent of earth's population suffer from hearing loss. DSP hearing aids are desirable compared to analog hearing aids because of their technicality and functionality. The digital hearing aid can be programmed conveniently using digital signal processing and this accounts to self adjustable nature of digital hearing aid. Adaptive filtering is used which effectively cancels out the additive noise. The algorithms used here will help in changing the filter coefficients to adapt to the audio signal. Gain is selectively added at higher frequency according to the person's hearing loss. This makes digital hearing aid flexible. Finally the processed is made to fit user's hearing range. Spectrogram is plotted to show the change in frequency of the signal with time. The MATLAB provides plenty of the toolboxes to carryout engineering calculations for various applications. Thus MATLAB is used for simulation.

General Terms
Signal processing, Ski-Slope hearing loss, Hearing aid, Audiogram.

Keywords
Adaptive filter, LMS algorithm, Fast Fourier Transfer, NLMS algorithm.

1. INTRODUCTION
The first means of assessing hearing aids was the human ear. The hearing aids were designed, tested and modified depending on either the developer's or the user's judgment. The human ear is sensitive to sound waves in the frequency range from about 20-20kHz although the upper limit of the range decreases with advancing age. Hearing loss may result from genetic causes, certain infections disease, exposure to excessive noise and aging. The audiogram is a graph which gives a detailed description of the persons hearing ability and which can be described as a picture of the persons sense of hearing. The audiogram is a measure of hearing threshold for a set number of frequencies. The sound intensity levels are measured in dB HL(hearing level), i.e. the varying sensitivity to different frequencies has been considered. These are the frequencies used to measure the audiogram :125, 250, 500, 750, 1k, 1.5k, 2k, 3k, 4k, 6k and 8k Hz. One must keep in mind that the audiogram only shows what the user cannot hear. When a person has problems hearing high frequency sounds the hearing curve looks like a ski slope in an audiogram and is a special kind of sensorineural hearing loss[1]. It can be difficult to hear children's voices or high pitched female voices. With the increasing capability and decrease in size of digital signal processing, there have been significant advances in hearing aid design. The major advantage of using signal processing is that it can be more exactly matched to acoustic properties of the auditory system than analog signal processing approach. The degree of hearing losses are mild hearing loss, moderate hearing loss, sever hearing loss, profound hearing loss[2]. These losses in an audiogram are shown in Figure 1.

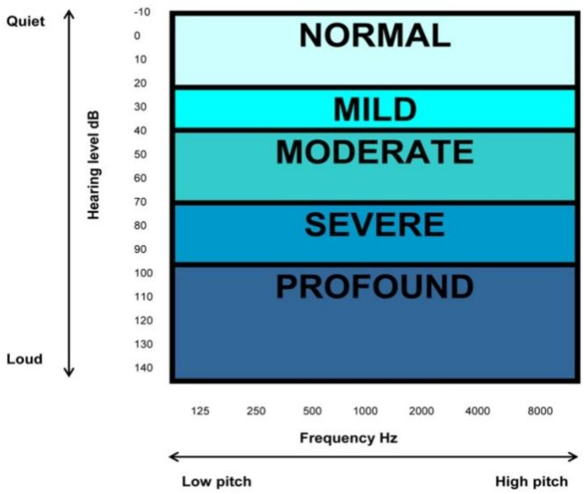

Figure 1: Degree of hearing loss

2. METHODOLOGY
The audio signal is read using MATLAB and then it is sampled. The flowchart in Figure2 shows that the additive noise is removed using adaptive filtering technique and the gain added at particular frequency depending on the need. Further signal amplitude above the threshold of pain is removed. Spectrogram of noisy signal and processed output signal is obtained for comparison. Also the learning curve gives the better understanding of the performance of adaptive filter is plotted. The step size used in the LMS algorithm for noise reduction is varied to get better learning curve resulting in better adaptation.

2.1 Adding Noise
White Gaussian Noise (WGN) is added in order to mimic the effect of many random processes that occur in nature. White Gaussian Noise (WGN) was generated using the MATLAB function wgn. The fan noise was added in order to check for real time condition [3].

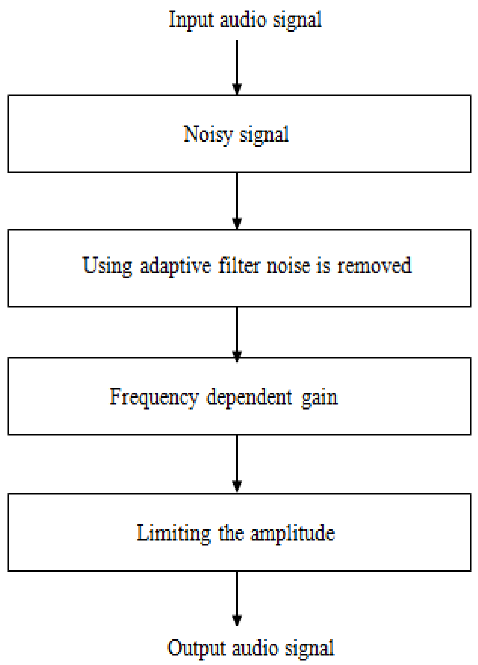

Figure 2: Flow chart for hearing aid system

2.2 Adaptive Filtering

The adaptive filter algorithm used to update the parameter values of the system can take on a myriad of forms and is often derived as a form of optimization procedure that minimizes an error criteria that is useful for the task. One of the most popular algorithm for adjusting the coefficients of the adaptive filter is least mean square algorithm(LMS).LMS was developed by Widrow and Hoff in 1960.since it is robust and low computation complexity it has been used in wide spectrum application[4].

LMS algorithm

$w(n+1)=w(n)+2\mu x(n)[d(n)-x^T(n)w(n)]$

$=w(n)+2\mu x(n)[d(n)-w^T(n)x(n)]$

$=w(n)+ 2\mu e(n)x(n)$

Where, $y(n)=w^T(n)x(n)$ is the filter output

$e(n)=d(n)-y(n)$ is the error

$w(n)=[w_0(n) \; w_1(n) \; w_2(n) \; . \; . \; . \; . \; . \; . \; . \; . w_{M-1}(n)]^T$ is filter taps at time n

$x(n)=[x(n) \; x(n-1) \; x(n-2) \; . \; . \; . \; . \; . \; . \; . \; . \; .x(n-M-1)]^T$ is the input data

The algorithm at each iteration requires that $\mathbf{x}(n)$, $d(n)$, and $\mathbf{w}(n)$ are known. This results in varying the pointing direction of the coefficient vector during the iteration.

NLMS algorithm is as stated below,

$$w(n+1)= w(n)+ \frac{\alpha}{\|x(n)\|^2+\beta} e(n)x(n)$$

Where, $y(n)=w^T(n)x(n)$ is the filter output

$e(n)=d(n)-y(n)$ is the error

$w(n)=[w_0(n) \; w_1(n) \; w_2(n) \; . \; . \; . \; . \; . \; w_{M-1}(n)]^T$ is filter taps at time n

$x(n)=[x(n) \; x(n-1) \; x(n-2). \; . \; . \; . \; . \; . \; x(n-M-1)]^T$ is the input data

$\|x(n)\|^2 = x(n)\times x(n)^T$

β is small positive number to reduce instability problem in division. α value lies between 0 to 2

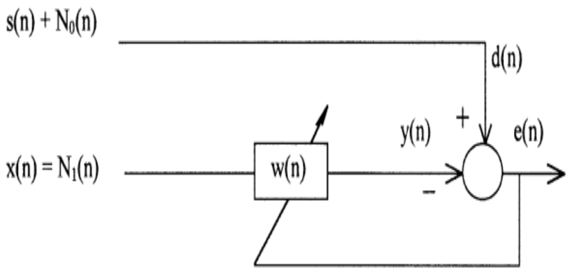

Figure 3: Adaptive noise cancellation

2.3 Frequency Dependent Gain Addition

The audio signal in time domain is transformed to frequency domain for easier computation using Fast Fourier Transform (FFT).In human ear the cochlea separates incoming signal into high and low frequency. After addition of gain in frequency domain at selected frequencies, signal is brought back to time domain by Inverse Fast Fourier Transform(IFFT)[5].

The equation for FFT is given by,

$$X(k) = \sum_{n=0}^{n=N-1} x(n) exp \frac{-j2\pi kn}{N} \qquad \begin{matrix} 0 < k < N\text{-}1 \\ 0 < n < N\text{-}1 \end{matrix}$$

The equation for IFFT is given by,

$$x(n) = \frac{1}{N} \sum_{n=0}^{n=N-1} X(k) exp \frac{j2\pi kn}{N} \qquad \begin{matrix} 0 < k < N\text{-}1 \\ 0 < n < N\text{-}1 \end{matrix}$$

2.4 Limit Amplitude

The amplitude limiting does the following function. The low intensity signal are amplified to be in the user's range and high intensity signal above the users comfort range are reduced. Spectrogram which gives a visual representation of the spectrum of frequencies in audio signal is plotted for comparison.

3. SIMULATION AND RESULTS

The simulations are done using MATLAB. Figure 4 shows the audio signal to which the noise is added and then the noisy signal is filtered using wavelet filtering.

Figure 5 shows the noisy signal recovered using adaptive lms and nlms algorithm. Comparing Figure 4 and Figure 5 it can be seen that adaptive filtering achieves better noise cancellation.

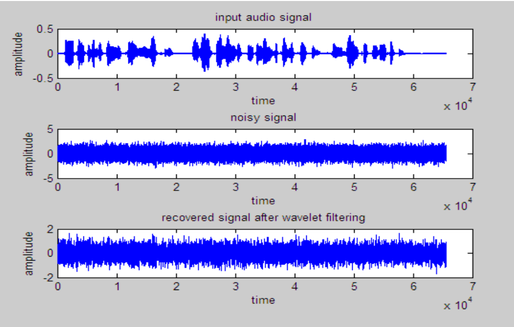

Figure 4: Noise cancellation using wavelet filtering

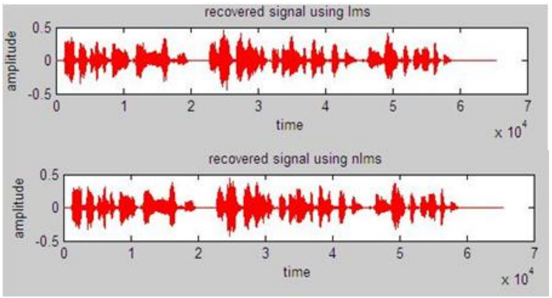

Figure 5: Noise cancelation using lms and nlms algorithm

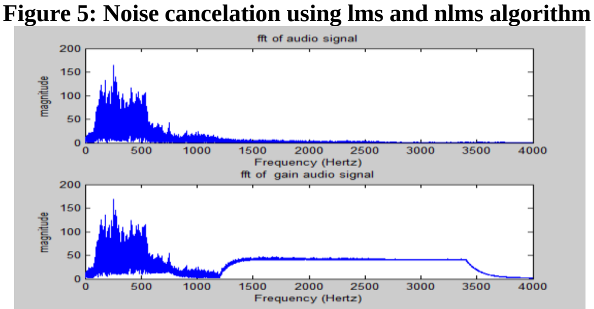

Figure 6: Gain addition in frequency domain

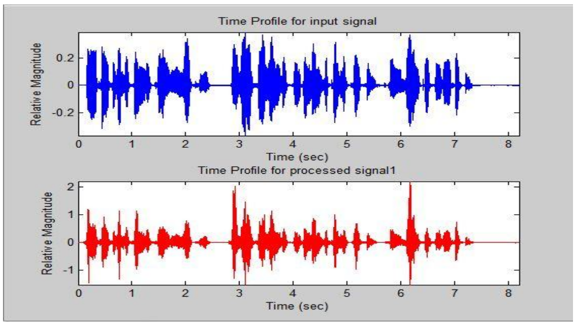

Figure 7: Representation of input and processed audio

Figure 7 represents the signal back in time domain after limiting the amplitude higher than threshold of pain. In Figure 8 the spectrogram of input and processed output is shown.

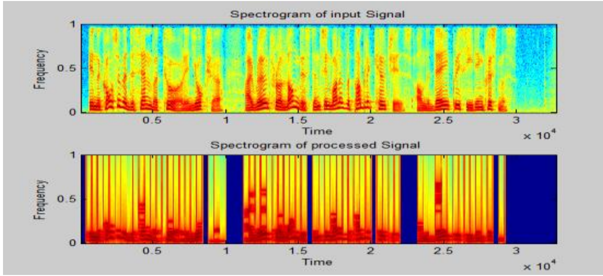

Figure 8: Spectrogram

4. CONCLUSION

In this paper using adaptive filtering technique noise was removed effectively. Also each frequency band can be given different amount of amplification. Hence the system becomes flexible. With DSP hearing aids there are unlimited ways in which digitized signals can be processed.

5. REFERENCES

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