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Implementation of Digital Hearing AID for Sensory Neural Impairment

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ABSTRACT

Hearing impairment is a chronicle disability affecting on people in world. The hearing aid is to amplify sound to overcome a hearing loss or impairment. The hearing aid picks up the sound signal with the microphone and amplifies all frequency sound signals but the sensory neural impairment person cannot hear particular frequency of sound in a noisy environment, since the auditory nerve is damaged. In this paper we are using MATLAB to design an adaptive filter for noise removal and filter banks for amplifies the particular frequency that a person with hearing loss can listen.

KEYWORDS: *Hearing Aid, Sensory Neural Impairment, MATLAB, Adaptive Filter, Filter bank*

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I. INTRODUCTION

Hearing loss or impairment is a partial or total inability to hear the sounds. There are three different types hearing impairment.

1. **Conductive loss**: It typically the result of obstructions in the outer or middle ear, which prevents sound from entering the middle ear.
2. **Sensory neural loss**: It occurs when inner ear nerves are damaged or do not transmit sound signal.
3. **Mixed loss**: It is combination of sensory neural and conductive hearing loss.

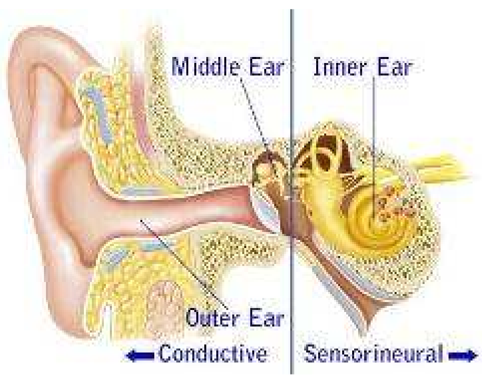

Figure.1. Ear diagram

Sensory neural hearing loss is the most common type of hearing loss. Sensory neural hearing loss happens when there is damage to the inner ear or to the nerves that travel from the ear to the brain. Sensory neural loss causes by aging, noise exposure, genetic conditions, drugs and medication. People with sensory neural hearing 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. This type of hearing loss can rarely be reversed by medical or surgical procedures. Sensory neural hearing loss often helped with hearing aids. Hearing aids are devices that are mainly used by deaf people to compensate in hearing loss. The digital hearing aids have more advantages then analog hearing aids because it has low power consumption, small in size and low noise. But normal hearing aid system makes the sound louder and makes the speech easy to understand to the impaired people. For this reason we design an adaptive filter and 5 filter banks in the MATLAB.

The rest of the paper is organized as follows: Proposed block diagram are explained in section II. Simulation results are presented in section III. Concluding remarks are given in section IV.

The following method has been adopted in this project to remove the noise and amplify the particular frequency signal. The basic block diagram is shown in Figure.2.

II. METHODOLOGY

The following method has been adopted in this project to remove the noise and amplify the particular frequency signal. The basic block diagram is shown in Figure.2.

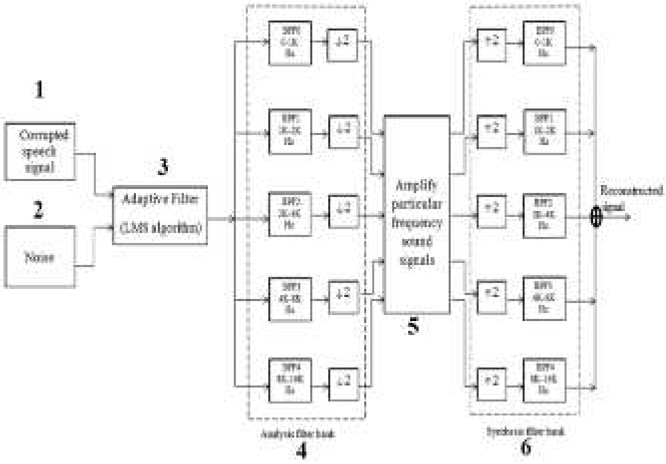

Figure.2. Proposed model

1. Corrupted speech signal
2. Noise signal
3. Adaptive filter
4. Analysis filter bank
5. Amplitude particular frequency
6. Synthesis filter bank

The block diagram contains (Figure.2.) inputs - one is corrupted speech signal and another one is noise signal which is correlated with noise in the corrupted speech signal. The corrupted speech signal is primary input to the adaptive filter and noise signal is secondary input to the adaptive filter.

Primary Input: Adaptive filter

Filter is used to process a signal in such a way that signal-to-noise ratio is enhanced. When the signal and noise are stationary and their characteristic are known or can be assumed, fixed filters are used. The design of fixed filters is based on prior knowledge of signal and noise characteristic. Adaptive filter can be defined as a filter that can automatically adjust their own parameters, based on the incoming signal and noise characteristics. Adaptive filter are used whenever there is a requirement to process the signal whose statistics are not known. Primary and at least one reference input are required to set up an adaptive filter based on the principle proposed by Widrow and Hoff.

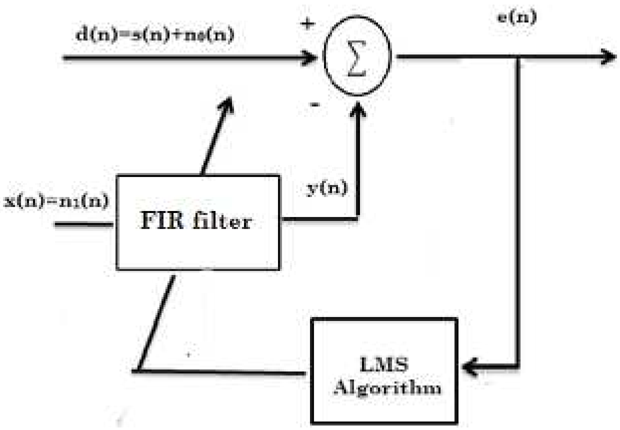

Figure.3. Structure of Adaptive filtering system

Adaptive filter configuration is illustrated in Figure.3. Primary input d (n) consists of the sum of the desired signal s(n) and noise $n_0(n)$. The reference input x(n) is the another noise $n_1(n)$ which is correlated with $n_0(n)$ and uncorrelated with s(n).y(n) be the estimate $n_0(n)$ obtain by filtering $n_1(n)$. The error signal e(n) is used to update filter coefficient $h_n(n)$ through an adaptive algorithm that minimize the error e(n).the output of adaptive filter is given by

$$e(n)=d(n) - y(n) = s(n)+n_0(n) - y(n) \;\ldots\ldots\; (1)$$

$$y(n)=h_n(n) * x(n) = \sum_{k=0}^{N-1} h_n(k)\, x(n-k) \;\ldots\ldots\; (2)$$

Weight adjustment algorithm

$$h_{n+1}(k)=h_n(k)+2\mu\, e(n)\, x(n-k) \;\ldots\ldots (3)$$

Filter bank design

Filter bank is an array of band pass filters that separates input signal in to multiple components, each one carrying a single frequency sub-band of the original signal. The process of decomposition performed by the filter bank is called analysis. The output of analysis is referred to as a sub-band signal. There are 5 filter banks in the analysis and synthesis part of 0-1k, 1k-2k, 2k-4k, 4k-8k, 8-16 kHz frequencies band pass filters are designed for the speech signal. Each band pass filter has decimation to down sample the audio signal. Down sampling is done to decrease the bit rate to transmit over a limited bandwidth. Then amplify the required frequency by multiplied by2.The reconstruction process is called as synthesis. In synthesis part has interpolation to up sample the audio signal for the good quality of sound. Then each band pass filter outputs are added to reconstruct the output signal.

III. RESULT AND DISCUSSIONS

The simulation audio data is selected from the **Microsoft/MS-SNSD and NOIZEUS websites.** MATLAB R2013a software platform s use to perform the simulation. The data is in .wav format.

The first step is to add the speech signal with noise signal. Both the inputs (corrupted speech signal and noise) signal has the same sampling rate; the sampling rate is 16 kHz. The 31[th] ordered adaptive filter removes the noise in the speech signal as shown in Figure.4.

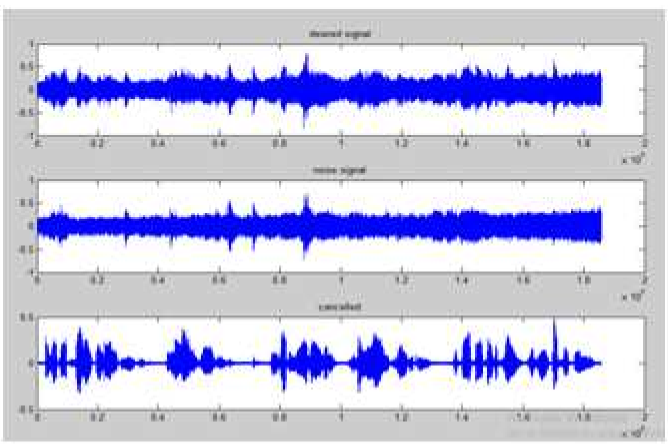

Figure.4. Noise cancellation using adaptive filtering

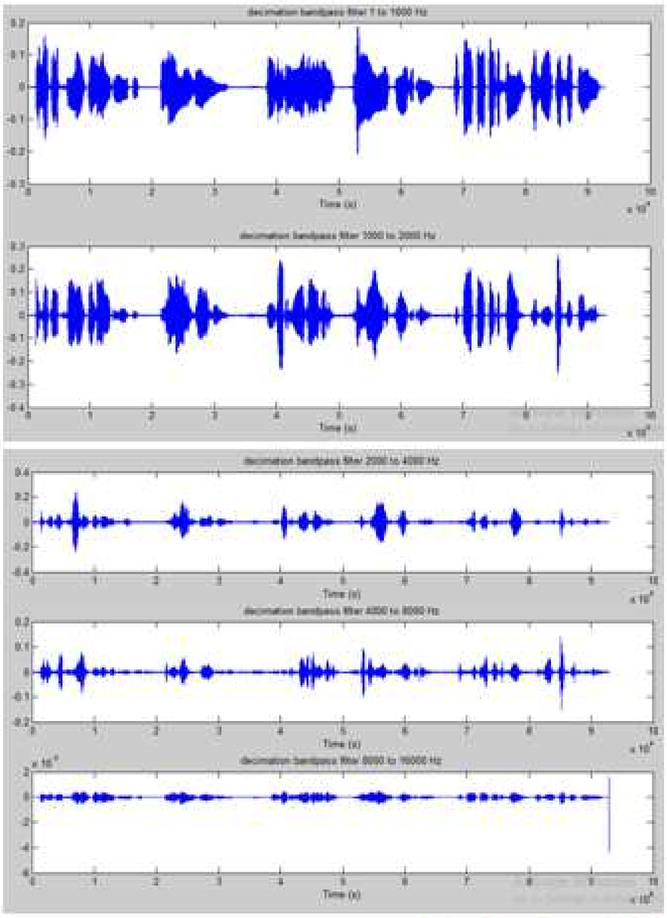

Figure.5. Output of analysis filter bank

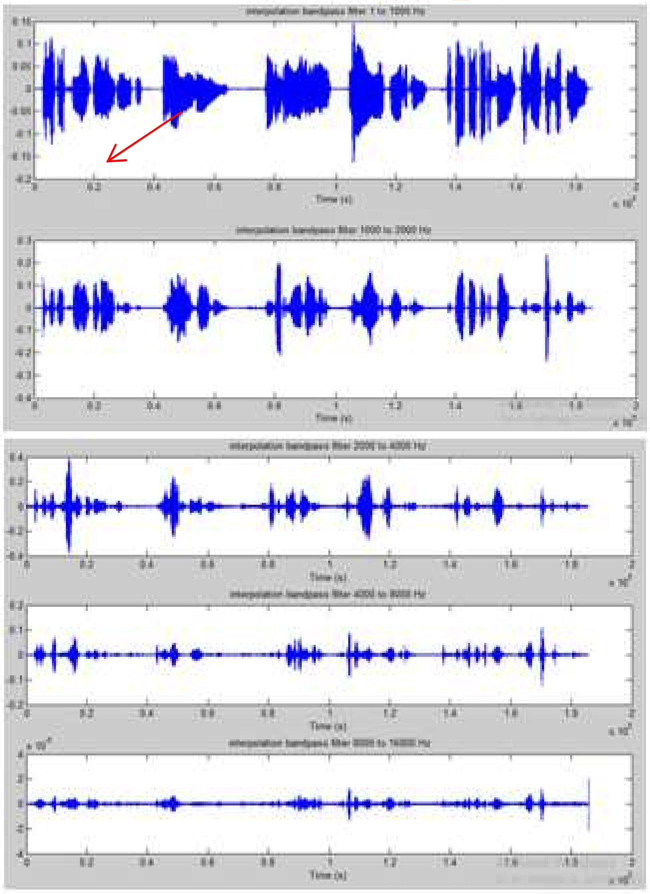

Figure.6. Output of synthesis filter bank

The FIR band pass filters are designed in both Analyses and synthesis filter bank. The filter order is 51. The figure 5 shows the output of analysis filter bank, it consist 5 band pass filter with different bandwidth and decimation to down sample the audio signal. The Figure.6 shows the output of synthesis filter bank after amplifying the particular frequency signal, synthesis consist of 5 interpolation and band pass filters. The arrow mark shows the increasing amplitude of 2k-4k band pass filter output.

Let's take an example a person cannot hear sound frequency between 2 to 4kHz so that filter bank separates the frequency and amplify that frequency signal, this makes that rang sound signal louder. The Figure.7 shows the reconstructed signal after adding different frequency band pass filter outputs.

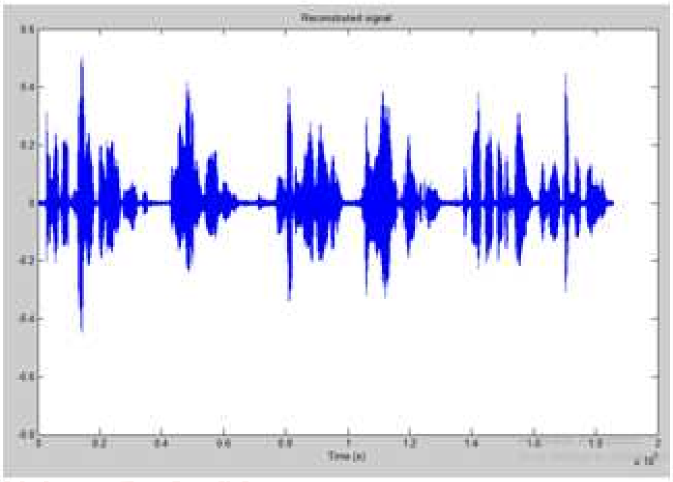

Figure.7. Reconstructed speech signal

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