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# Effective Filter Design Using FDA Tool for Improving Speech Intelligibility of Hearing Impaired

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## Abstract

The most common types of hearing impairments are conductive loss, mixed loss and Sensorineural loss. Among these Sensorineural Loss is severe. Since, the treatment of chronic sensorineural deafness is very difficult .Sensorineural Loss results in degradation in frequency selectivity due to increase in spectral masking. Improvement in speech intelligibility is perceived when speech signal is splitted into two complementary parts on the basis of critical bands. These obtained signals are given to Sensorineural hearing impaired subject.

Project involves designing of finite impulse response filters based on critical bands using frequency sampling technique.Effective filter design is done using the FDA tool provided in MATLAB.The filter coefficients are calculated using the tool and are exported to code composer studio environment to be used in the source code. The design of filter varies based on the audiogram of subject and requires calculation of filter coefficients separately.This is done efficiently using the FDA tool.The speech signal is splitted into two complementary parts. These filtered signals are presented to the subject using sound multimedia card of personal computers.Presentation of processed speech signals is carried out in silent environment. Improvement in processed signal is observed as compared to unprocessed one.

**Keywords**: Audiogram, Code Composer Studio ,FDA tool, Spectral Masking.

# 1. Introduction

The human ear is divided into three main parts the outer ear, the middle ear and the inner ear. The most common types of hearing impairments are conductive loss, mixed loss and Sensorineural loss. Among these Sensorineural Loss is severe. Since, the treatment of chronic sensorineural deafness is very difficult. Sensorineural Loss results in degradation in frequency selectivity due to increase in spectral masking.

Hearing loss is caused by damage to some or all of the nerves in the inner ear. Sensorineural losses cannot be reduced or eliminated by surgery. Sensorineural hearing loss involves both distortion and loudness. Sensorineural loss may affect some or all of the hair cells or nerves in the inner ear responsible for sensing sounds of different pitches. Sensorineural loss can cause temporal masking, which results in degradation in speech perception. Spectral masking results in suppression of two adjacent frequency components by each other. Due to spectral masking adjacent frequency bands overlap with each other. Spectral masking results in perceptual confusion in consonantal segments in which noise bands are not separated. Spectral masking reduces frequency selectivity which is ability to detect a tone in presence of complex stimuli. The sensorineural impairments are characterized by high frequency hearing loss, increase in threshold of hearing, compression in dynamic range, severity of temporal masking, and loss of spectral resolution due to spread of masking.

There is possibility that splitting speech into two complementary parts on the basis of frequency and presenting these dichotically might increase speech intelligibility. The hearing aid based on this principle can be helpful to bilateral sensorineural hearing-impaired people with some residual hearing. The objective of project is to split the speech into two complementary spectra for binaural dichotic presentation as a possible solution to problem of spectral masking. The study involves processing digitized speech and conducting listening tests sensorineural hearing impaired subjects. The objective is to study possible solution to the problem of spectral masking in case of sensorineural hearing-impaired subjects. In view of this, acquired speech is split based on critical multi-band filtering. Critical bandwidths are chosen as per the auditory filter bandwidths (Zwicker, 1961). Zwicker has described the relation between frequency f and critical band-rate z. Auditory frequency selectivity can be described in terms of an equivalent rectangular bandwidth as a function of center frequency.

Designing of finite impulse response filters based on critical bands using frequency sampling technique.Effective filter design is achieved using the Filter Design and Analysis tool provided in MATLAB.The calculation of filter coefficients is done using the tool and are presented to code composer studio environment to be used in the source code. The design of filter varies based on the audiogram of subject and filter coefficients are required to be calculated separately.This is done efficiently using the FDA tool.The speech signal is split into two complementary parts. These filtered signals are presented to the subject using sound multimedia card of personal computer.Presentation of processed speech signal is carried out in silent environment for better results.Significant Improvement is observed in perception of processed signal as compared to unprocessed signal.

## 2. Sections

### 2.1 Filter Design for improving speech perception

The project is based on improvement in speech intelligibility for sensorineural hearing impaired. Spectral Masking in sensorineural loss is a phenomenon where presence of one signal component results in audibility of the neighboring signal component. The information received from both the ears gets integrated. Hence, splitting of information in speech signal for presenting signals to the two ears in some sort of a complimentary fashion helps in reducing the effect of masking. Increased temporal masking results in increased forward and backward masking of weak acoustic segments by strong ones. This degrades the reception of important cues for identification of consonants and results in degradation of speech perception. Splitting the speech temporally into a number of segments and presenting the alternate segments to the two ears may help in reducing the effect of increased temporal masking. The conventional hearing aids provide the amplification of speech. Only the gain of signal is increased and is directly fed to subject. These aids do not improve the speech presentation for person suffering from sensorineural impairment. This loss arises due to loss of hair cells in the inner ear and auditory nerve.

The speech is filtered and divided into two parts in such a way that frequency components lying within a critical band are in one part, components lying in next non-overlapping critical band are in second part, components of third non-overlapping critical band are in first part. The critical bands are spread over a frequency range from 70Hz to 5 kHz. The critical bands are discriminated into two banks based on even or odd frequency components.The odd numbered filter outputs are fed to one ear and even numbered filter outputs are fed to the other ear.

*2.1.1 FDA tool* for filter Design

Designing of filter is done using FDA tool in matlab which helps to design a variety of filters for variety of subjects differing in the audiogram.The filter coefficients are generated and exported to the code composer studio environment.The relevant changes in the source code is done using the obtained filter coefficients.Thus with some modification in source code it can be used for variety of subjects differing in their audiograms.

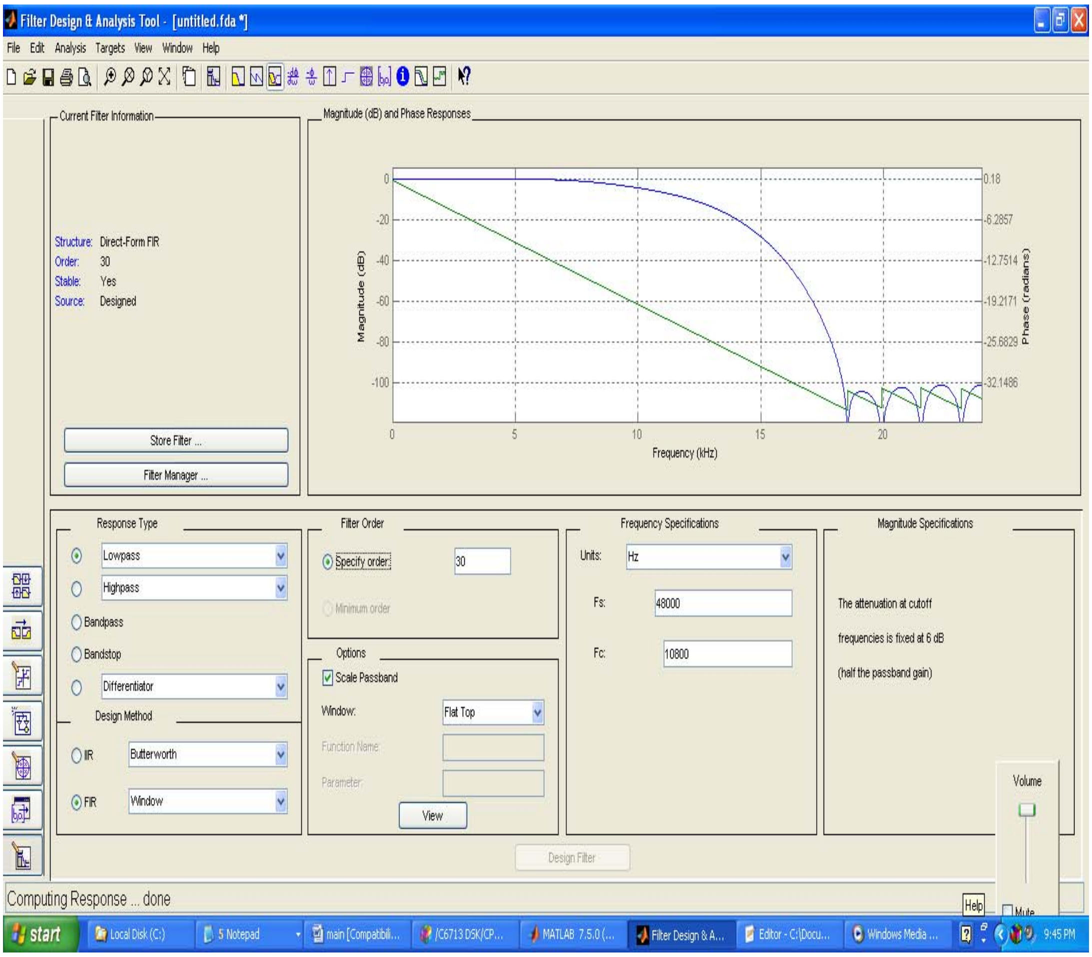


**Fig. 1:** Filter design using *FDA tool in MATLAB.*

## 3. Conclusion

The vowel consonant vowel syllables are processed using finite impulse response filters. The design of filters is done effectively using the FDA tool since varieties of filter coefficients are required for variety of subjects. The unprocessed and processed syllables are presented binaurally to sensorineural hearing impaired subjects in low sound level surrounding. Seven subjects were tested. It is observed that there is significant improvement in recognized score using adjustable gain filters. Thus, filtering scheme is a remedy for reducing the effect of spectral masking and improving speech perception.

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