SSRG International Journal of Electronics and Communication Engineering Volume 6 Issue 5, 35-44, May 2019
ISSN: 2348 – 8549 /doi:10.14445/23488549/IJECE-V6I5P106

Comparison Study of Hamming and Kaiser Window over Band-pass Filter Banks in Cochlear Implant System

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Abstract

The cochlear implant is the successful neural prosthesis ever developed. This prosthesis is the most effective way to restore hearing purpose. The device is surgically implanted inside the cochlea (brain) to stimulate current pulses similar to neuron stimulation. The audio signal sensed through a microphone is amplified and digitized before the audio/speech signal processing. Continuous Interleaved Sampling (CIS) is one of the most important speech processing strategies used in the speech processors of Cochlear Implant. The speech processing ensures the splitting of the input signal into various frequency bands and provides the filtered signal to appropriate electrodes. The electrodes activate the auditory nerve fibers to provide hearing sensation. The focus of the paper is to design a bank of Band-pass Filters, which is used in the CIS algorithm for separating the frequencies for processing the signal and finally feeding it to different electrodes. Filters are targeted for audio frequencies from 200 to 7.5 kHz. MATLAB based Filter banks using Kaiser Window and Hamming window are considered for design. The center frequencies of the respective band-pass filters are taken into account to test the eight filters. In Xilinx ISE 14.7 Verilog based synthesis for bank of band-pass filters will be implemented.

Keywords: *CIS, FFT, FIR, IIR, LPC, RF.*

I. INTRODUCTION

A cochlear implant is an inside ear implant that can restore deeply deaf people to normal hearing [1]. Fig 1 shows a typical structure, consisting of an external part and an implant, of existing commercial cochlear implant devices. The functioning principle of the microphone is to collect environmental sounds in the outside part of the body and to transform the analog signals into digital audio with an A / D converter. The digital audio is processed by a specific algorithm that collects the data needed from the body's inside. A transmitter is used for data modulation and transmission of data and energy to the inner part of the body. In the interior part of the body, the receiver recovers the data and energy of RF signals and then decodes the data so that the stimulator can generate electric stimuli. The audio nerves are excited and transmit the excitement into the user's brain to generate a hearing sensation.

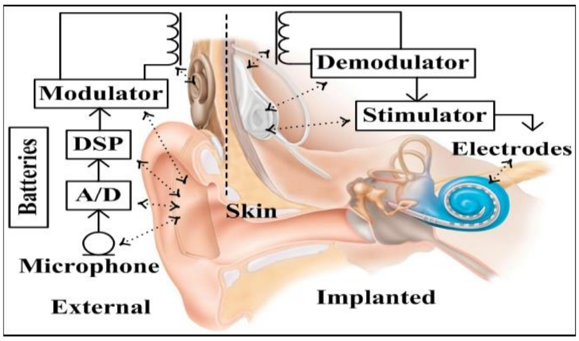


Fig 1: Typical architecture of cochlear implants

A channel vocals strategy was initially developed by the cochlear implant device inner aid produced by Symbion, Inc., Utah. The signal is first compressed by automatic gain control and then filtered in the central frequencies at 0.5, 1, 2, and 3.4 kHz [9] in four adjacent frequency bands. The filtered waveforms were simultaneously supplied in analog form to four electrodes. The interaction between channels caused by the summation of electric fields from individual electrodes is a major concern associated with simultaneous stimulation. Stimulations from other electrodes may distort neural responses to stimuli from one electrode significantly. These interactions can distort speech and thus degrade the understanding of speech. This speech processing strategy is called a Compressed Algorithm.

Researchers of the Research Triangle Institute (RTI), using non-simultaneous, interlaced pulses [10], proposed a simple solution to this problem of channel interaction. They proposed modulating the filtered waveforms by trains of biphasic pulses delivered to the electrodes in a non-overlapping (non-simultaneous) fashion, that is, in such a way that at one time, only one electrode was stimulated. The amplitudes of the pulses were derived by extracting the envelopes of the waveforms of the band-passed signals. The strategy that resulted was called the Continuous Interleaved Sampling (CIS) strategy.

II. PREVIOUS WORKS

Cochlear implant development can be traced back at least 200 years back by the Italian scientist Alessandro Volta who was the battery inventor. He used the battery as a research tool to show that electrical stimulation could directly evoke sensations of auditory, visual, and touch in humans (Volta, 1800). When placing in each of his ears one of the two ends of a 50 Volt battery, he observes that at the moment the circuit is over, a shock has been received in the head and, a few moments after they start to hear a sound, it was a sort of crackling with shocks. The uneasy feeling, which could be harmful because of the brain shock, prevented him from repeating the experiment. Safe and systematic studies have not been published for another 150 years on the effect of electric stimulation on the hearing until modern electronic technology has developed. Stevens and colleagues identify at least three mechanisms for "electrophonic perception," equipped with vacuum tube-based oscillators and amplifiers (Stevens 1937; Stevens and Jones 1939; Jones et al. 1940).

The first mechanism was the "electromechanical effect," whereby the electric stimulation causes the hair cells in the cochlea to vibrate, leading to an acoustically stimulated perception of the signal frequency's tonal pitch.

The second mechanism was related to transforming the electric signal into an acoustic signal from the tympanic membrane, resulting in the perception of a tonal pitch, but the frequency of the signal doubled. Stevens and colleagues isolated the second mechanism because they found that in patients who lacked the tympanic membrane, only the initial indicator pitch was sensed with electrical stimulation.

The third mechanism was the direct electrical activation of auditory nerves, as a response to sinusoidal electric stimulation, a much steeper increase in loudness with electro current, and occasional activating facial nerves; some patients reported a noise-like sensation.

In 1939, people watched the first talking machine at the World Fair in New York City with intense curiosity. The machine spoke with the help of a human operator sitting before a console, like a piano keyboard, with 10 keys, a pedal, and a brace. Inside the machine, analog circuits were connected to a loudspeaker by band-pass filters, switches, and amplifiers.

The talking machine has the first artificial speech synt hesis system hardware. This speech synthesis system developed by Homer Dudley of Bell Laboratories was the channel vocoder (voice maker)[2]. Dudley's idea of vocoding has had a profound effect not only on telephony and speech transmission applications [3, 4] but also on the development of cochlear implant processors. Cochlear implants are the latest and most successful signal processing strategies based on vocoding analysis principles. A modified version of the vocoder analysis algorithm is provided for all cochlear implant equipment.

III. CONCEPTUAL BACKGROUND

A. Normal ear Vs. Deafened ears

When sound travels to the ear, it moves through a sequence of transformations as it passes through the outer ear, middle ear, inner ear, auditory nerve, and brain. Sound is sound pressure caused by abrupt changes in the air pressure. A series of small bones in the inner ear then responds to the incoming sound by vibrating itself. The middle ear converts sound into mechanical vibrations. The cochlea, a snail-shaped cavity filled with fluid in the inner ear, then transforms these mechanical vibrations into vibrations in the fluid. Pressure differences within the fluids of the cochlea lead to the fluid displacement of the basilar membrane. These displacements contain significant information about the frequency of the sound signal. Hair cells that attach the basilar membrane are bent according to the change of position in the basilar membrane. The bending of the hair cells causes neurons to fire by releasing an electrochemical substance. These neurons transmit information about the sound signal to the brain. The brain detects the excitation in the auditory nerve and then interprets the sound.

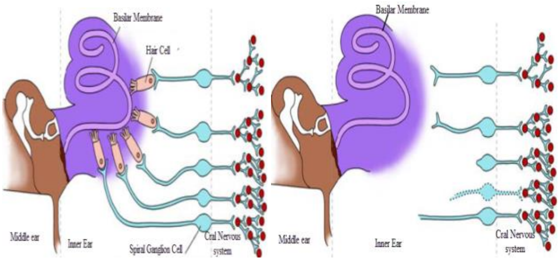


Fig 2: Structure of normal and deafened ears

Fig 2 shows the structure of normal and defended ears. At present, two types of deafness exist, one is conductive, and the other is sensorineural. Conductive deafness results from middle ear damage, which reduces the transmission efficiency of the mechanical sound vibration. The use of acoustic amplification by audio aids can usually rectify this. In sensorineural deafness, the hair cells are typically damaged (or are dead), and the activation of the auditory nerve fiber is reduced (or non-existent).

The bending movements of the hair cells in the basilar membrane's vicinity are responsible for converting mechanical vibrations into electrical signals. If the hair cells are damaged, the auditory system cannot properly convert sound into electrical impulses in the nerve. However, the wave cannot reach the auditory nerve and the brain because of the damaged hair cells. This problem has led to the development of cochlear implants.

B. Function of Cochlea

The cochlea is the principal functional unit of the inner ear. A cochlear implant can benefit people with severe hearing loss. The cochlea's basilar membrane divides the input signal into different frequencies, shown in Fig 3. The location of internal hair cells along the basil membrane determines the hair cells' optimum response to different frequencies. When the sound signal is sent in the form of a traveling wave in the cochlea, the hair cells on the apex react to low frequencies, while the hair cells at the bottom respond to high frequencies.

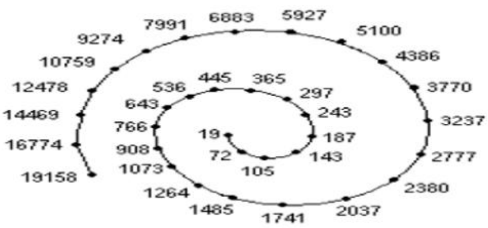


Fig 3: The Basilar membrane is structured according to the speech frequencies in the apex and base [2].

C. Speech Processing Strategies

The main difference between implant devices is the strategy of signal processing for transforming speech into electrical stimuli. The speech processor is a cochlear implant brain. It extracts specific acoustic attributes, codes them through RF transmission, and controls electric stimulation parameters. The cochlea encodes low frequencies, while the basal code is high frequencies. Thus, all these implants have implemented a filter bank to divide speech into different frequency bands, although their processing strategies to extract, encode, and deliver the right features differ significantly. Usually, two different types of channel implants are being used. Single-channel implants provide electrical stimulation at one cochlea location through a single electrode. The mechanism of place code used by a normal cochlea for encoding frequencies is not used by single-channel stimulation since only one single site is stimulated in the cochlea. Multichannel implants provide electric stimulation at various cochlea sites using an array of electrodes. An electrode array is used to stimulate various auditory nerve fibers for frequency coding.

D. Requirements for Designing of a filter bank

Digital filtering has specific characteristics that one must need to consider while designs. The analog input signal must meet certain requirements. Besides, additional signal processing is needed to achieve the appropriate result when converting a digital signal output to analog form.

The functional cochlea mechanism was seen in the previous section. The cochlear implants speech processor replicates the human ear cochlea. Usually, spectral analysis of speech signals is performed in a filter bank model or a linear predictive coding (LPC) model. The speech signal is allowed through a band-pass filter bank with coverage extending from 100-5500Hz. Individual filters that generally overlap in frequency can be implemented as Finite Impulse Response Filter (FIR) or Infinite Impulse Response Filter (IIR). The number of filters in the filter bank is generally dependent on the number of channels, depending on the number of stimulating electrodes as specified in[3]. Therefore, the filter bank consists of eight band-pass FIR filter channels. Although the character of the cochlear filter is non-linear, the linear phase FIR filter is used here to overcome the disadvantage of the IIR filter due to the interaction of speech signals from different channels. FIR has several benefits over IIR filters. FIR filters have no poles and are unconditionally stable. FIR does not accumulate errors because they depend only on a finite number of past input samples. The definition of an FIR filter is,

$$Y[n] = \sum_{K=0}^{M} h(n) * x[n-1] \qquad (1)$$

Where y (n) is the output signal, k is the order of the filters, and h(n) is the filter coefficients.

The desired frequency response of a filter is expanded as shown in equation 2 and known that coefficients having an infinite length. FIR filter is obtained by truncating the infinite length of the filter at K= ± ((M-1)/2), where M is the length of the desired sequence.

Abrupt truncation of the Fourier series results in oscillations in pass-band and stop-band regions of the filter. An optimal way to reduce these oscillations is to use an appropriate finite-length window w (n), which controls the overall filter to yield a smooth frequency response. Therefore, a window should possess some of the following spectral characteristics.

a. The main lobe width (Wm) of the frequency response of the window should be narrow.
b. The ripple ratio (R) should be small.
c. The side-lobes should decreases in energy rapidly.
d. To adequately block the stop-band frequencies, it is necessary to have good stop-band attenuation.

Standard windows such as Rectangular, Hamming, and Hanning have only one independent parameter: the 'window length', N, which controls the 'transition width' of the filter.

This paper aims to design the Filter-Bank in the speech processor used for Cochlear Implant System applications. The Speech Processor used is Continuous Interleaved Sampling.

E. Continuous Interleaved Sampling(CIS)

CIS pre-processing is similar to the compressed analog processor. The block diagram of the CIS strategy is shown in Fig 4. The signal is first pre-emphasized and then applied to a bank of band-pass filters. The envelopes of the outputs of these band-pass filters are then full-wave rectified and low-pass filtered (typically with 200 or 400 Hz cut-off frequency). The envelopes of the outputs of the band-pass filters are finally compressed and used to modulate biphasic pulses. A non-linear compression function (e.g., logarithmic, power-law) is used to ensure that the envelope outputs fit the patient's dynamic range of electrically evoked hearing. Trains of balanced biphasic pulses, with amplitudes proportional to the envelopes, are delivered to the electrodes at a constant rate in a non-overlapping fashion.

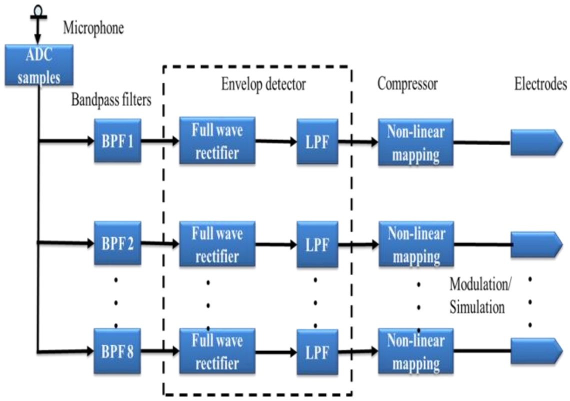


Fig 4: Continuous Interleaved Sampling Strategy

Many variations of the CIS strategy have emerged and are currently being used by the three implant manufacturers. For example, some devices use the Fast Fourier Transform (FFT) for spectral analysis, and some use the Hilbert Transform to extract the envelope instead of full-wave rectification and low-pass filtering. Although all three manufacturers employ the CIS strategy, it is based on different implementations.

III. PROPOSED FILTER BANK DESIGN FOR AUDIO PROCESSING

In general, a filter bank model is used for spectral analysis of the speech signals is shown in Fig 5. The speech signals are passed through a bank of band-pass filters; its coverage extends from 100-7500Hz to the signal's frequency.

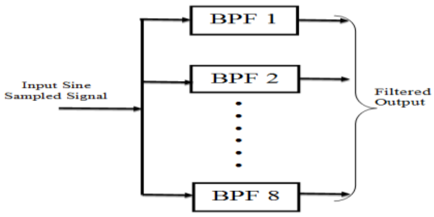


Fig 5: Speech Processor Filter bank

A. Generating Filter Coefficients

The filter coefficients are generated for the following specifications are shown in Table I.

TABLE I
Filter Bank Specifications

Channel No	Pass Band Frequencies (Hz)	Channel No	Pass Band Frequencies (Hz)
1	200-315	5	1225-1927
2	315-495	6	1927-3031
3	495-779	7	3031-4768
4	779-1225	8	4768-7500

The coefficients are obtained from MATLAB coding, which is fixed to 16-bit precision. The COE write command writes a XILINX Distributed Arithmetic FIR filter coefficient. .COE file, this file can be loaded into the XILINX CORE Generator. The coefficients are extracted from the fixed-point dfilt object 'hd'. The fixed-point filter must be a direct form FIR structure dfilt object with one section whose Arithmetic property is set to fixed-point. The coefficients are fixed to 16-bit precision. The coefficients are read into in XILINX, which are used to implement the filter in Verilog.

B. Implementation of Filter Bank in Verilog

The filter-bank is implemented in Verilog using the filter coefficients generated in MATLAB is shown in Fig 6. The generated coefficients were used in the FIR filter's HDL design, and its functional verification was done on Xilinx ISE 14.7 environment. The operation of the filter is further verified by applying the test bench stimulus of sine signal sampled at 16000 Hz frequency to the design, and the results are analyzed in MATLAB to get the analog format of the signal. The results of the simulation show that the FIR filter digitized using XILINX is correct and efficient.

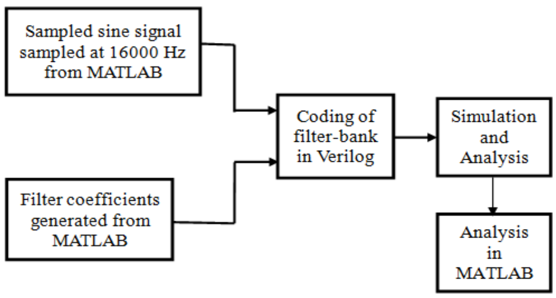


Fig 6: Implementation flow of filer bank

C. Testing a Filter

Here the observed simulation test results are plotted in MATLAB. To know the correctness of the sine samples, the output samples are tested using FFT. The input samples are generated considering the concept of coherent sampling.

a). Coherent Sampling

The primary application of coherent sampling is sine-wave testing of A/D converters. If the proper ratios between Fin and Fs are observed, the need for windowing is eliminated. This greatly increases the spectral resolution of an FFT and creates an ideal environment for critically evaluating the spectral response of the A/D converter. Care must be taken, however, to ensure the spectral purity and stability of Fin and Fs in the testing environment.

b). Definition of Coherence

A coherent sampling of a periodic waveform occurs when an integer number of cycles exist in the sample window. In other words, coherent sampling occurs when the relationship of Equation 2 is rational.

$$\frac{F_{in}}{F_s} = \frac{M}{N} \qquad (2)$$

Where: Fin: The input frequency

Fs: The sampling frequency

N: The integer, factor of 2, number of samples in the in record

M: The integer number of cycles in the data record

IV. RESULTS AND INFERENCES

MATLAB based Filter banks using Kaiser Window and Hamming window designs are shown below. Verilog code written for implementation of the filter bank is verified with sinusoidal and speech signal test vectors under the Isim environment.

A. Magnitude Response of Filter-Bank

The magnitude response of 8 band-pass filters is shown in Fig 7The response is designed from MATLAB from the specifications mentioned in section 3.2.5. The first 4 are 128 order filters, and the last 4 are or 200 order filters. The stop-band attenuation is minimum 30 dB and maximum 60 dB.

B. Simulation Result of Filter-Bank

Filter-bank is simulated in Verilog, the output data from variable "filter_out" is read into a text file, and the output samples are plotted. Eight filters are tested by giving the sine samples with a center frequency of 8 band-pass filters and a speech signal."filter_coeff_reg" is a 16-bit register used to read the filter coefficients from a text file. An input signal is given through the Verilog code test bench, "filter_in" is a 16-bit register used to read the text file of the input signal generated in MATLAB. The inputs and filter coefficients are 2's complemented to perform the 2's complement multiplication. "mult_reg" is the 32-bit register to save multiplied values. "filter_out" is the final output of the filter. Different flags and several intermediate variables are used to drive the required signals.

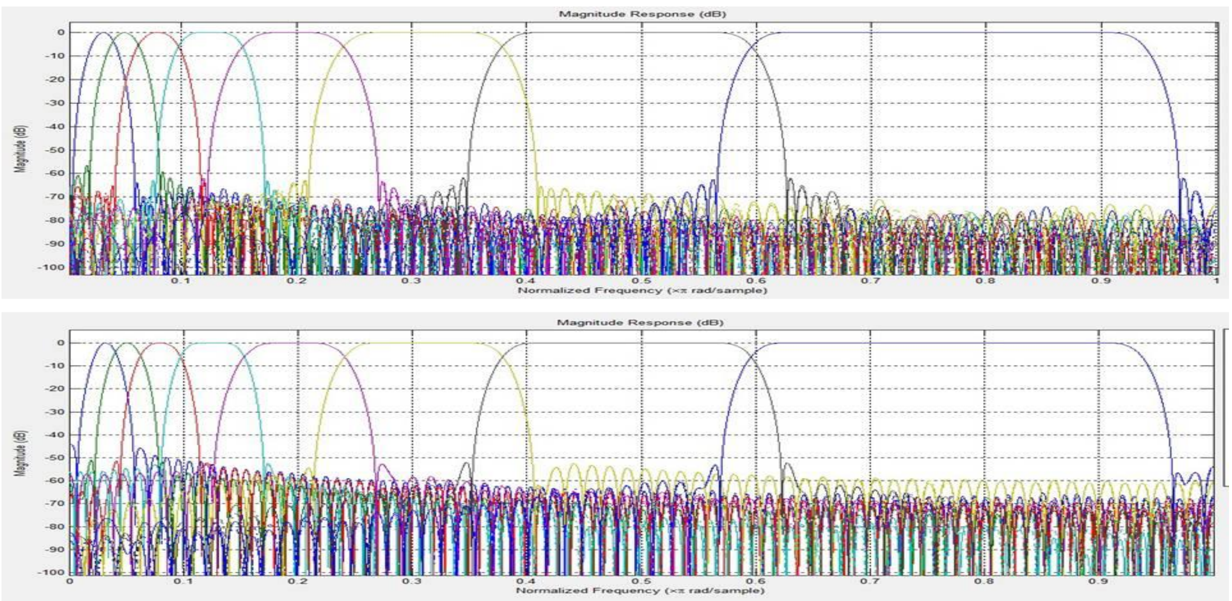


Fig 7: Magnitude response of filter bank

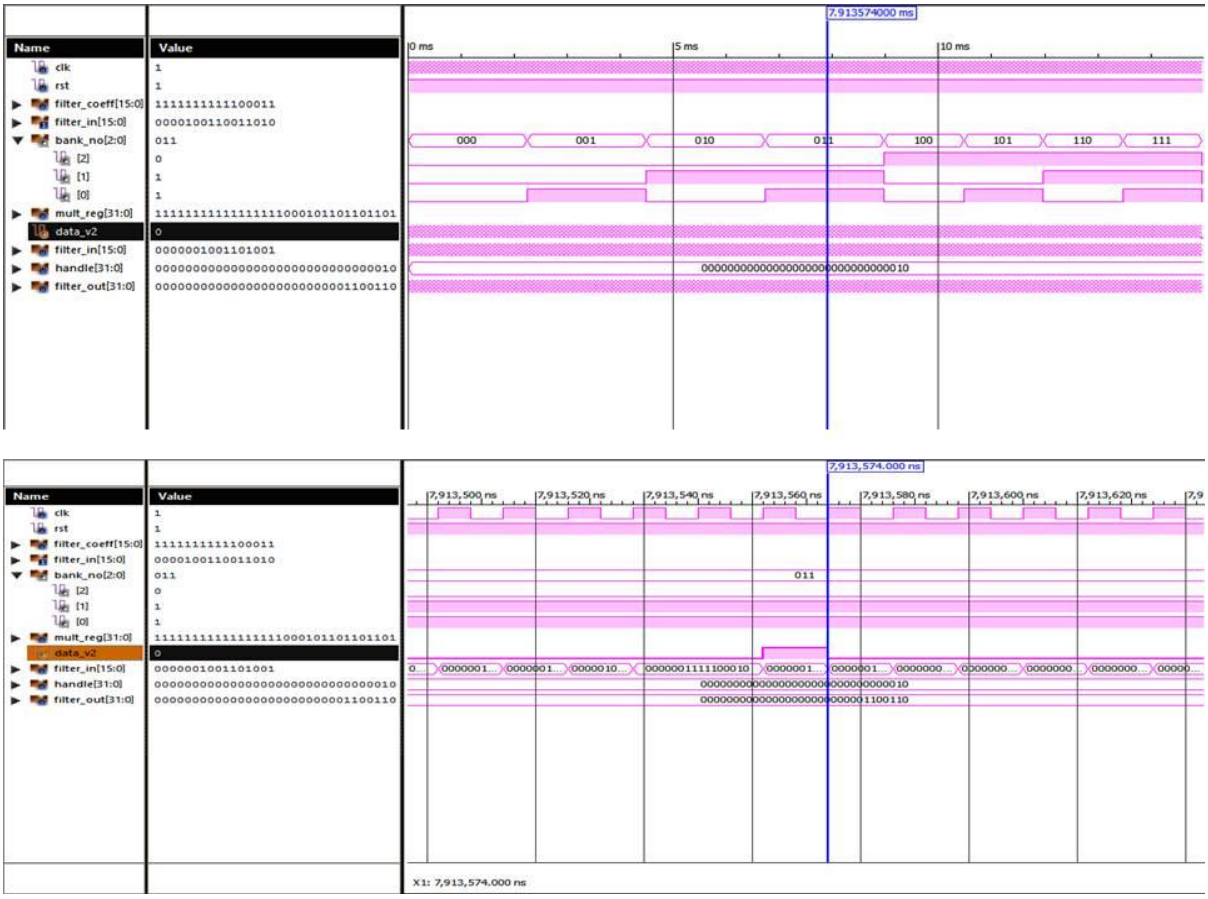


Fig 8: Simulation result of Filter-Bank and Simulation result of sixth band-pass filter in detail

Fig8 is the simulation result of the filter bank for a single center frequency input sinusoidal signal. These kinds of simulation results can be seen for eight input signals or a window technique. So here, in total, sixteen simulation results can be observed. Next, the responses of band-pass filters with window techniques are shown in fig.

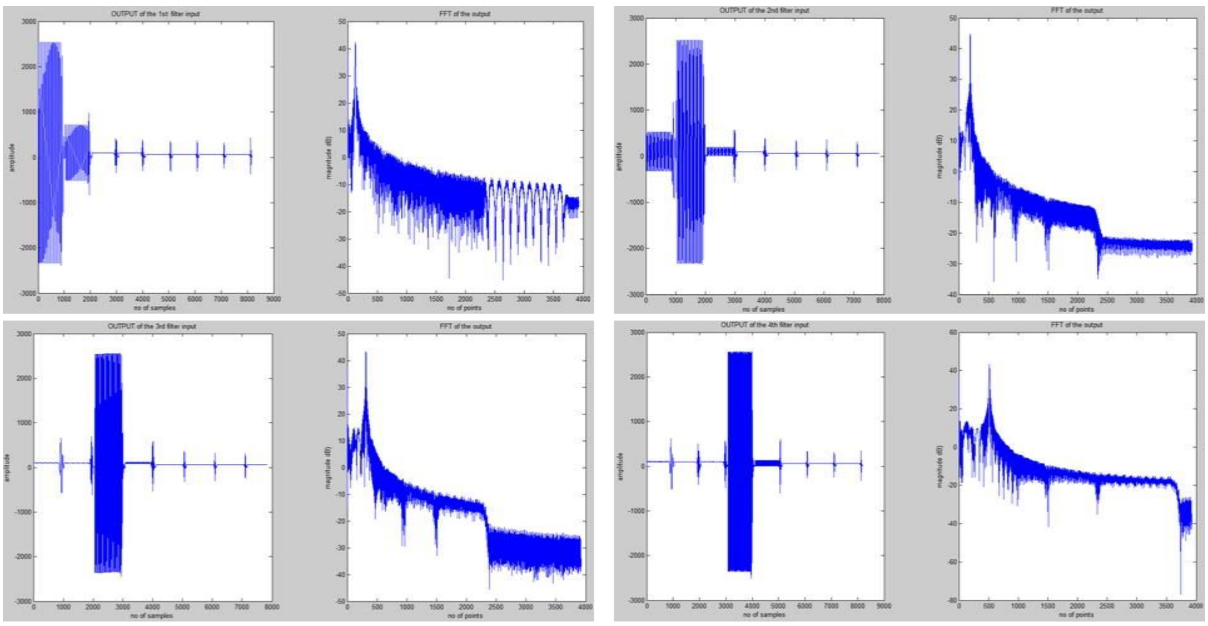


Fig9: Responses of first four Band-pass Filters for Kaiser Window

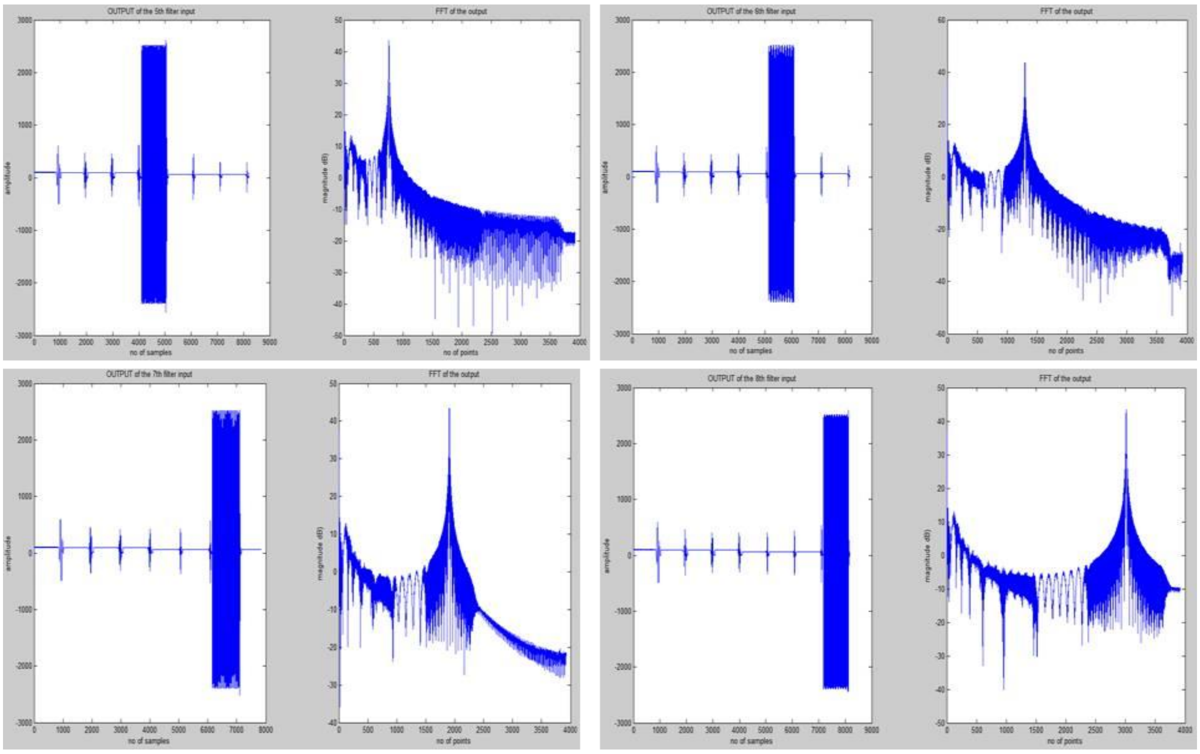


Fig 10: Responses of last four Band-pass Filters for Kaiser Window

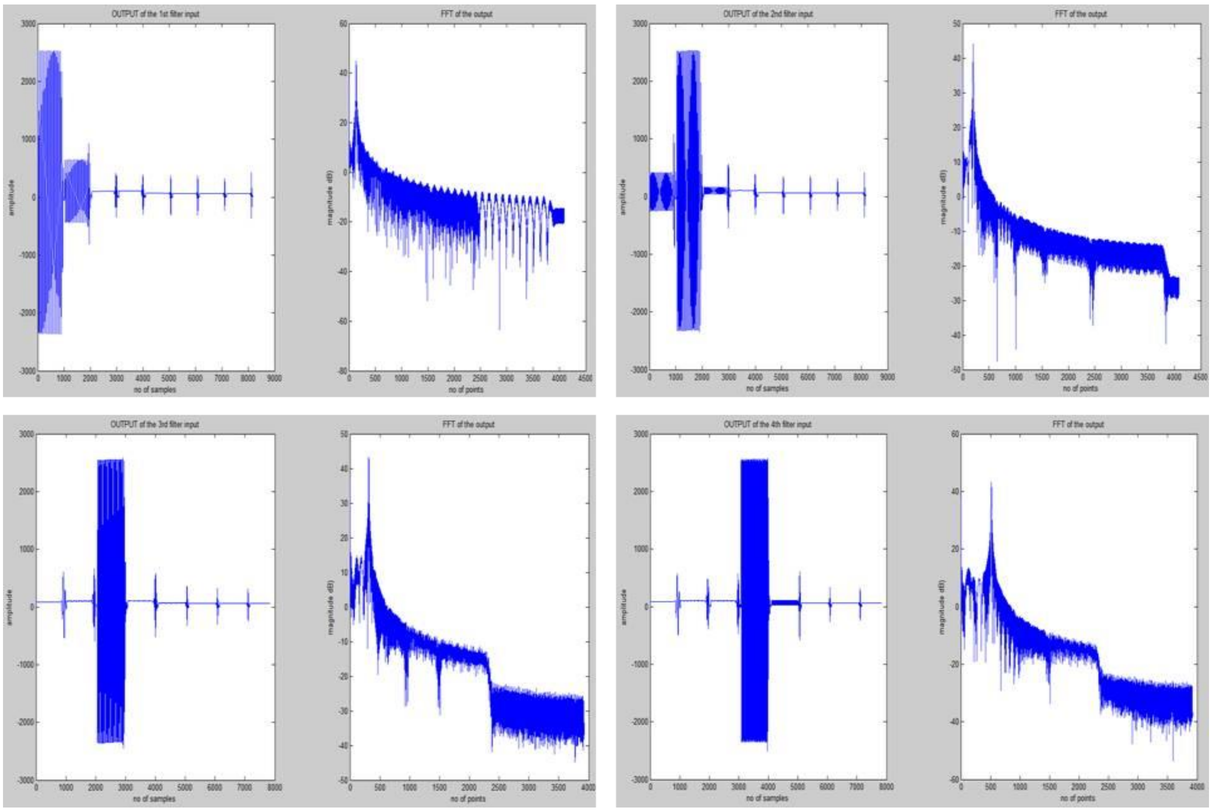


Fig 11: Responses of first four Band-pass Filters for Hamming window

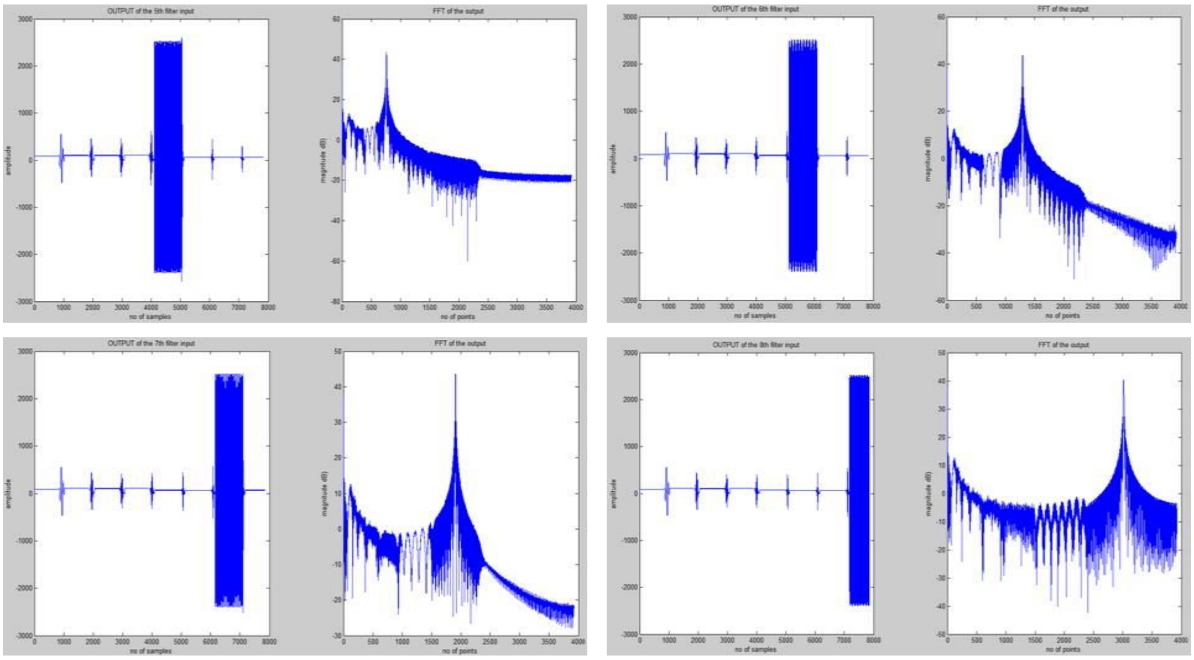
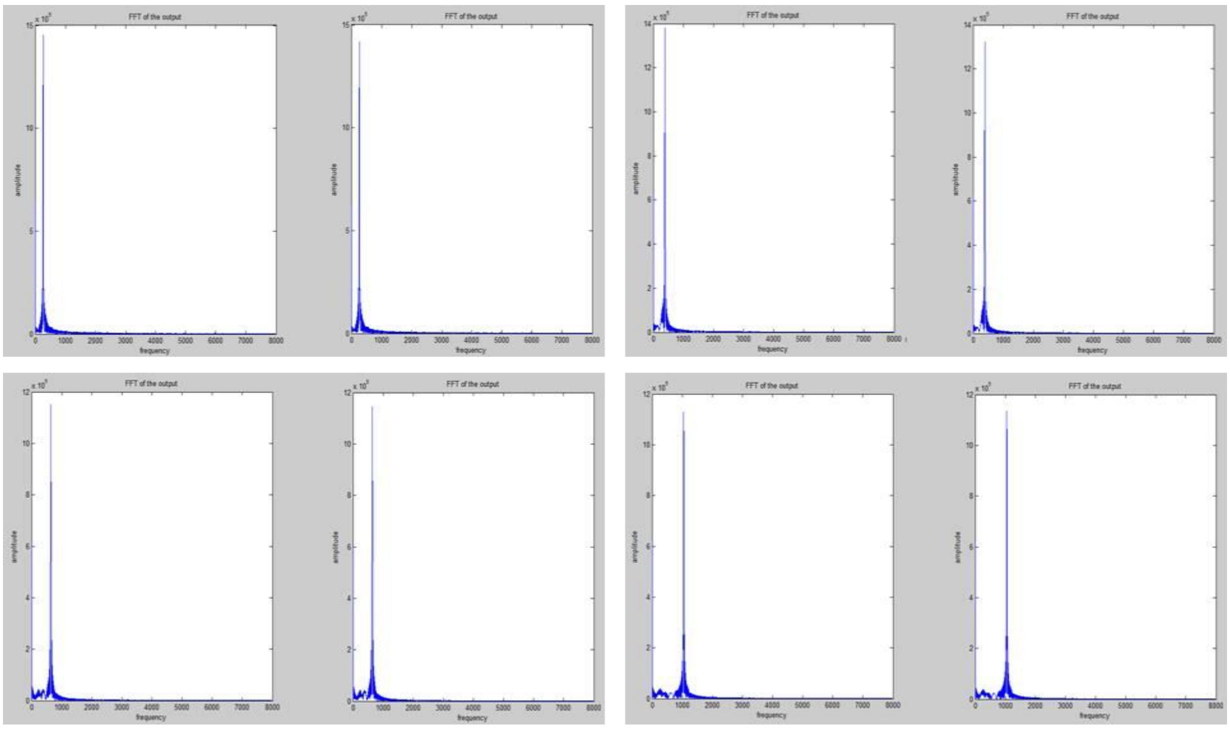


Fig12: Responses of last four Band-pass Filters for Hamming window

Fig 9, Fig 10, Fig 11, and Fig 12 shows the output read from the text file and plotted in MATLAB. Input frequencies are 265.625 Hz, 390.625 Hz, 640.625 Hz, 1046.875 Hz, 1546.875 Hz, 2640.625 Hz, 3890.625Hz, 6140.625 Hz for eight Band-pass Filters, which are the center frequencies of corresponding band-pass filter is passed through 8 Band-pass filters. Input is the sine signal sampled at 16000 Hz. The figures show the input signal passing through Kaiser and hamming window of corresponding band-pass filters and attenuating in all other filters. A small signal is also passed through the second band-pass filter, but it is mostly attenuated. This is because of the small bandwidth of the filter.

The amplitudes of the time domain signal of band-pass filter using Hamming Window in the adjacent BPF is 83-85 units less than Kaiser Window. So the attenuation is slightly better in band-pass filter using hamming window. And the center frequencies of all band-pass filter responses are shown below



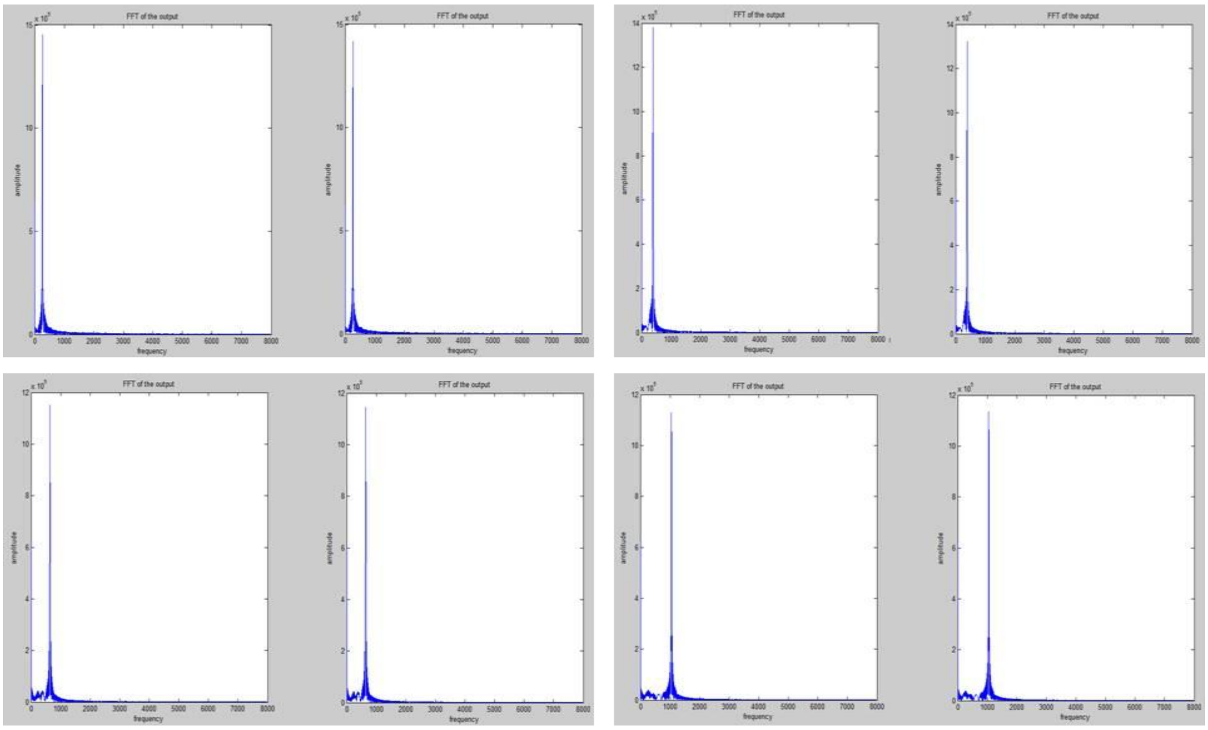


Fig 13: Frequency response of filter bank output using Kaiser Window and Hamming Window with the centre frequency of all BPF.

C. Input and output responses for speech signal

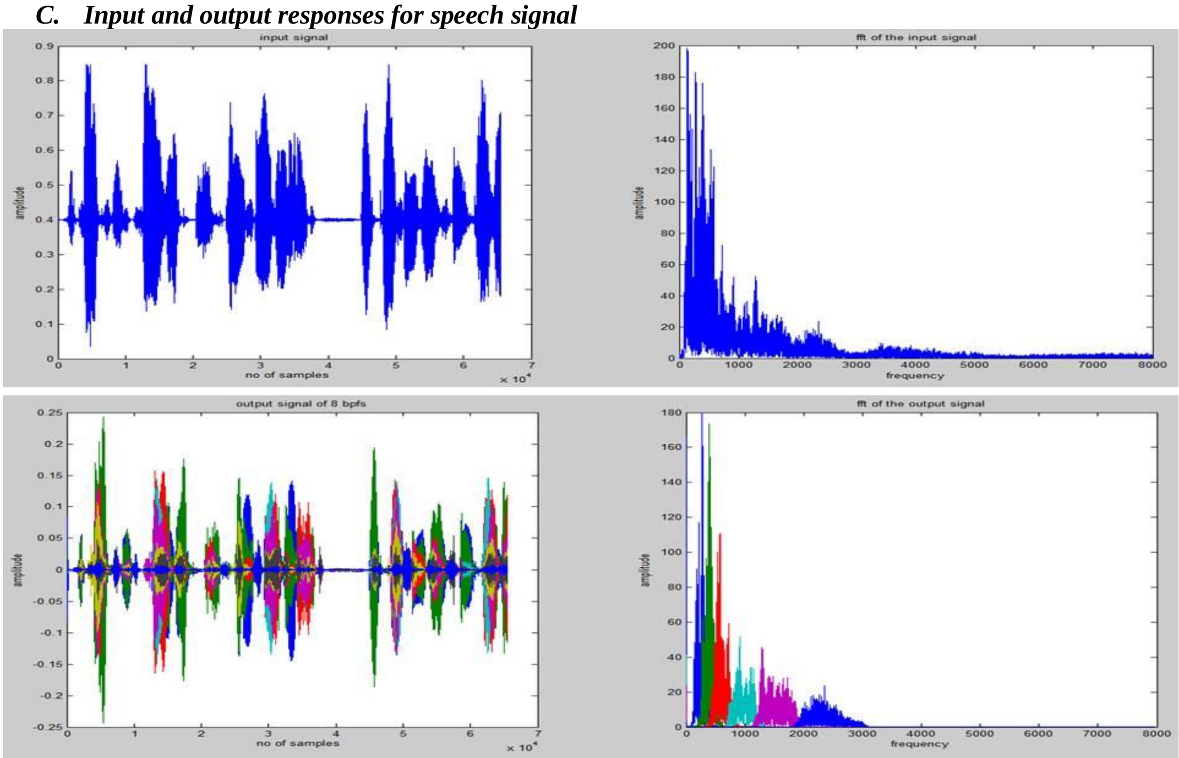


Fig 14: Input and output of speech signal for bank of band-pass filters of hamming window

Here Fig 14 shows a speech signal containing a small number of external sounds in a time domain and its FFT to observe the speech signal frequencies. The below figure shows the output of eight band-pass filters in the time domain using the hamming win dow and its FFT, i.e., in the frequency domain. The output of eight band-pass filters, each colored differently, indicates the output from each band-pass filter.

V. CONCLUSIONS AND FUTURE SCOPE

This paper describes the design and simulation of the filter bank, which is used in the CIS algorithm for the cochlear implant system. Firstly, the

FIR filter bank using Kaiser Window and Hamming Window is developed to generate eight energy levels of the considered input signal. Filter coefficients are generated from MATLAB coding and used in Verilog code for implementing Filter bank. The center frequencies of the respective band-pass filters are taken into account to test the eight filters. It has been observed from the Verilog based synthesis that the filter bank designed with hamming window is giving a slightly better response when compared to Kaiser Window for the first two band-pass filter's center frequency. A small signal is also passed through adjacent band-pass filters, which are mostly attenuated in Hamming window.

In this study, a suitable bank of band-pass filters is designed for a Cochlear Implant System. Choosing a higher-order Band-pass filter will result in a better filter response but requires more hardware. To improve the hearing quality, the eight band-pass filters can be increased, typically to 16 band-pass filters. The filter-bank can be implemented in ASIC flow. The module implemented for filter-bank can be used in other filter applications or speech processing applications.

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