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Comparative Analysis of FIR filter

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Abstract — *Filters are designed to pass a specific range of frequencies and eliminate the remaining to extract meaningful data from the signal. Filters are extensively used in various field of signal processing such as radar, EMG, EMG, ECG, noise reduction etc. Depending upon the components (either active or passive) used in a filter, they are classified as low pass filter(LPF), band pass filter(BPF) ,high pass filter(HPF), and band stop filter(BSF), however the type of input signal used made a further classification as analog and digital filter, the digital filter is further classified as IIR (infinite impulse response) and FIR(finite impulse response) filters. This paper describes an approach to compare basic window functions used to design digital FIR filter for LPF, HPF, BPF and BSF with filter order 50 and sampling frequency 48 kHz.*

Keywords- *LPF,HPF, BPF, BSP, FIR Filter, Blackman window, Kaiser window, Chebyshev window, Hamming window, Rectangular window.*

I. INTRODUCTION

In digital signal processing, filter is used to remove unwanted components from a signal. It is designed to pass a specific range of given frequencies and completely reject the others. Different types of filters are used for distinct purposes depending upon the filtering action, which are basically referred as low-pass filter, high-pass filter, band-pass filter and band-stop filter. Filters can be classified into two categories and they are analog filter and digital filter. Analog filters work on analog signals, electrical passive components are used while implementing the analog filters. Unlike analog filters digital filters operates on digital samples of the signal. Digital filters are better than analog filters because they have linear phase response, low pass band ripples, very high stop band attenuation. Mainly digital filters are two types, those are Infinite impulse response(IIR) and finite impulse response(FIR) . If the impulse response of the filter falls to zero for a finite time that means it has a finite number of non- zero terms. It is called finite impulse response. The infinite impulse response filters have infinite number of non-zero terms, therefore it is called infinite impulse response filter (IIR). While in the implementation of IIR filter needs feedback, on the other hand FIR filters do not need any feedback. In comparison FIR filter has advantages over IIR filter. FIR filter is always stable, linear in phase and easily implemented on hardware on the other hand IIR filter is unstable and non linear phase [4] [5] [6].

II. FIR FILTER

FIR filters are realized generally in a non recursive method that provides stability. FIR filter's transfer function consists of only zeros. Frequency response and magnitude response are determined by all zeros present in the z-plane. FIR filters are particularly designed where accurate and precise linear phase response is required. For a FIR filter the output is a weighted sum of current input and the finite number of antecedent output. The z transform for N point FIR filter is given below:

$$y(n) = \sum_{k=0}^{N-1} h(n)x(n-k) = \sum_{k=0}^{N-1} b_k x(n-k) \qquad (1)$$

Where, y(n) is the output signal , h(n) is the filter coefficient , k is the order of the filter and x(n) is the input signal. Output signal y(n) in terms of frequency response

$$y(n)= x(n)* h(n) \qquad (2)$$

$$y(n)= x(0)*h(n)+ x(1)*h(n-1)+x(2)*h(n-2)+.....+x(n)*h(0) \qquad (3)$$

The system function can be expressed as,

$$H(z) = \sum_{k=0}^{N-1} b_K z^{-k} \qquad (4)$$

FIR filter design involves the two stages as follows:

(i) The approximation stage
(ii) The realization stage

The approximation stage takes the specification and transfer function is designed accordingly:

- In the frequency response, a desired or ideal response.
- Appropriate class of filters.
- A measure of quality of approximation
- A method/ design that is designed to find the best filter transfer function.

According to the realization stage it works with choosing the specific structure or circuit design to implement the transfer function. There are given three well known methods to realize FIR filter [4] [5] [6].

- Using window method
- Using the frequency sampling technique
- Using the Optimal filter design method

III. WINDOW METHOD

Window technique is the most important technique in the designing of FIR filter [1].
The basic design principle of window function is to calculate $h(n)$ using desire frequency response $H_d(e^{j\omega})$. The formula of $h_d(n)$ is given as:

$$h_d(n) = \frac{1}{2\pi}\int_{-\pi}^{\pi} H_d(\omega)e^{j\omega n}\, d\omega \tag{5}$$

Where,

$$H_d(\omega) = \sum_{n=-\infty}^{\infty} h_d(n)e^{-j\omega n} \tag{6}$$

One possible way to design FIR filter that approximate the desired frequency that must be truncated at n=N-1. The N point rectangular window function defined as

$$w(n) = \begin{cases} 1 & 0 \le n \le N-1 \\ 0 & otherwise \end{cases} \tag{7}$$

Where w (n) is the window function.
The unit impulse response of the FIR filter becomes

$$\begin{aligned} h(n) &= h_d(n)w(n) \\ &= h_d(n) \qquad 0 \le n \le N-1 \\ &= 0 \qquad \text{otherwise} \end{aligned} \tag{8}$$

There are two categories of window function i.e. fixed window function and adjustable window function. Fixed window functions are mainly used by Blackman, Hanning, Hamming and rectangular window. Kaiser window is adjustable window function and chebyshev window is optimal window function [3].

3.1 Hamming Window

It is also a cosine window function like hanning window. It has same characteristics as hanning window but further suppress the side lobes. It is defined as

$$w(n) = \begin{cases} \left(0.54 - 0.46cos\left(\frac{2\pi n}{N}\right)\right) & 0 \le n \le N-1 \\ 0 & elsewhere \end{cases} \tag{9}$$

3.2 Blackman window

This window is similar to hamming and hanning window. Blackman window has an advantage over the other windows that it has better stop band attenuation and less pass band ripple. Window function is given as:

$$w(n) = \begin{cases} \left(0.42 - 0.5cos\left(\frac{2\pi n}{N}\right) + 0.08\cos\left(\frac{4\pi n}{N}\right)\right) & 0 \le n \le N-1 \\ 0 & elsewhere \end{cases} \tag{10}$$

3.3 Kaiser window

The Kaiser window is also known as optimal window. It is a function that has maximum attenuation according to the width of main lobe. Its coefficient with β parameter is defined as:

$$w(n) = \begin{cases} \frac{I_0(\beta\sqrt{1-(\frac{2(n+1)}{N+1}^2)})}{I_0} & 0 \le n \le N-1 \\ 0 & elsewhere \end{cases} \quad (11)$$

Where β determines the shape of the window.

3.4 Chebyshev window

The chebyshev window is also known as Dolph-Chebyshev window. Earlier, it was used to solve the problem of optimal direction characteristics in radio antenna. It has optimal characteristics firstly, it has a minimum main-lobe width for a given side-lobe attenuation. Secondly, that it is equi-ripple (the side-lobe height is the same at all frequencies) [2].
The zero-phase chebyshev window function $w_0(n)$ can be described as

$$w_0(n) = \frac{1}{N}\sum_{k=0}^{N-1} W_0(k)\, e^{\frac{j2\pi kn}{N}} \quad (12)$$

Where, $W_0(k)$ is Fourier coefficient.
$W_0(k)$ can be represented as

$$W_0(k) = \frac{cos\{Ncos^{-1}[\beta cos(\pi k/N)]\}}{cosh[\cosh^{-1}(\beta)]} \quad (13)$$

Where, β is fixed value parameter for window function and expressed as

$$\beta = cosh\left[\frac{1}{N}\cosh^{-1}(10^{\alpha})\right] \quad (14)$$

Where, parameter α sets the value of sidelobes to -20 α dB and presented as

$$\alpha = cosh\left[\frac{1}{N}\cosh^{-1}(10^{A/-20})\right] \quad (15)$$

Where, A represents the required sidelobe attenuation.

3.5 Rectangular window

Rectangular window is the simplest window function. When we shift the window to make the function causal a phase term arises always while transformation of window is real in the zero phase. The rectangular window function is explained as

$$w(n) = \begin{cases} 1 & 0 \le n \le N-1 \\ 0 & otherwise \end{cases} \quad (16)$$

Where , N is the window length in samples.
As the parameter N increases the main lobe gets narrow in size. There is no effect on the side lobes.

IV SIMULATION RESULTS

The MATLAB codes for all the above windows were successfully simulated in MATLAB R2014a . The simulation results were obtained for filter length 50 and sampling frequency 48 kHz. The simulation result for basic low pass filter having length 50 and sampling frequency 48 kHz is shown in fig.1.

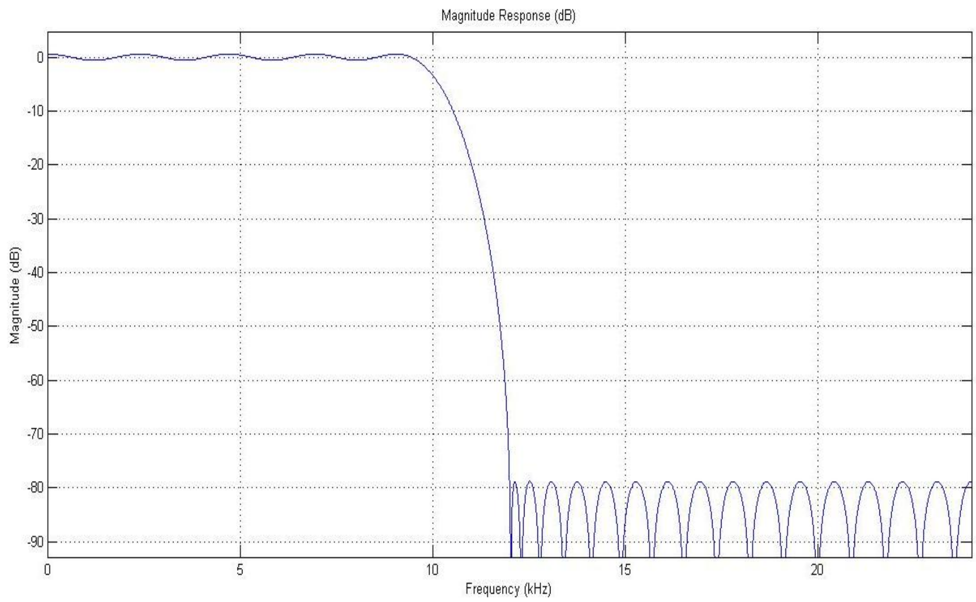

Fig.1. Low Pass Filter magnitude response

Comparative analysis for Low Pass Filter

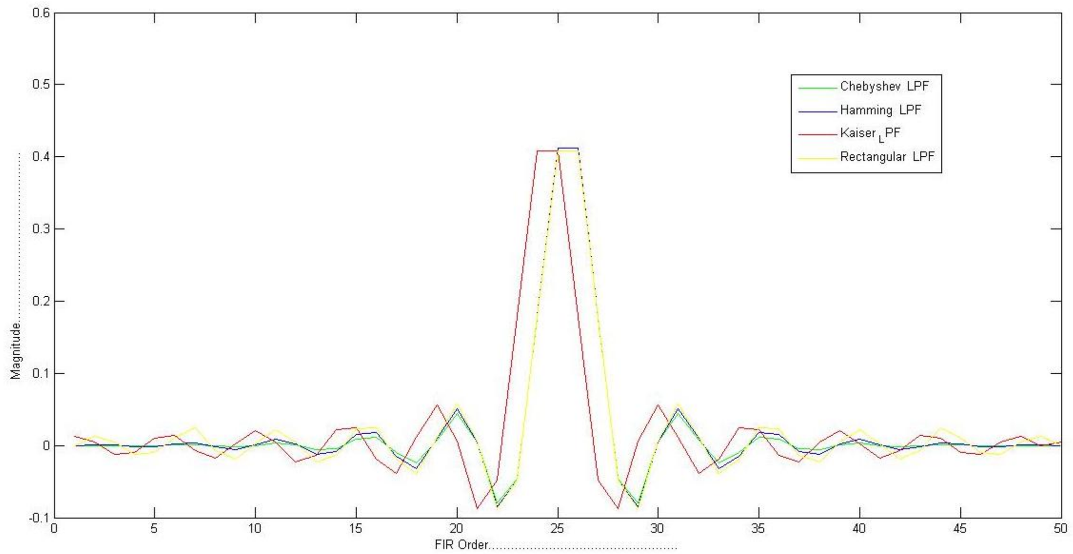

Fig.2 Output response of basic window functions for LPF with sampling frequency 48 kHz.

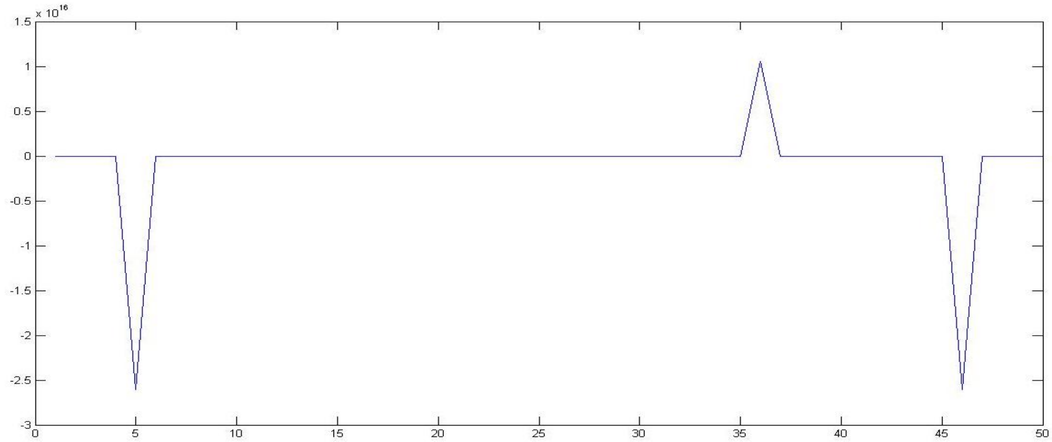

Fig.3 Response of Blackman window functions for LPF with sampling frequency 48 kHz.

The comparative analysis for LPF has performed at sampling frequency of 48kHz (f_s) and cutoff frequency of 10.8kHz is shown in fig.2. , from fig.2 it is concluded that impulse response of Chebyshev, Hamming ,Kaiser and Rectangular window is a sinc function except Blackman window (fig.3).

Comparative analysis for High Pass Filter

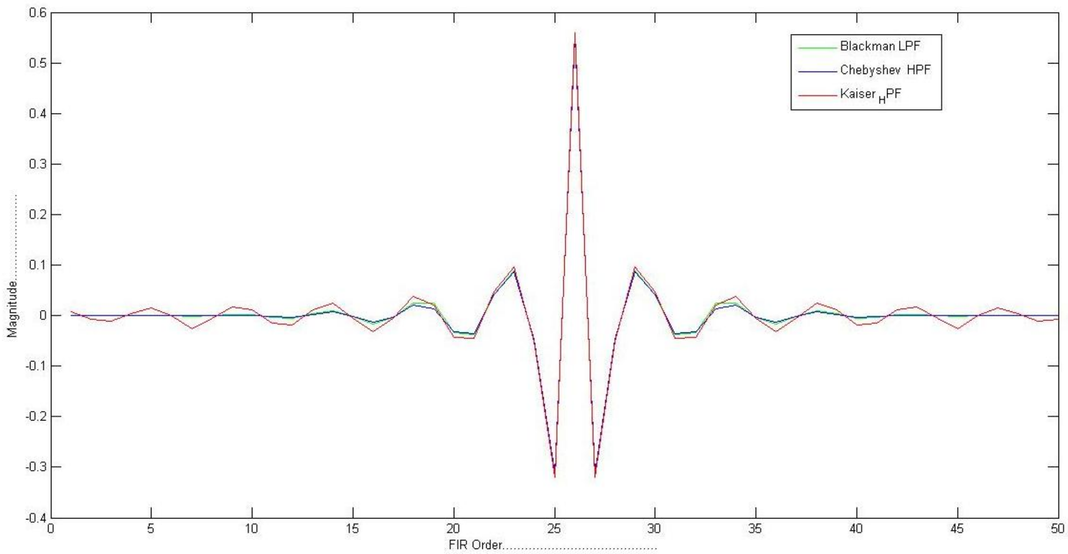

Fig.4 Output response of basic window functions for HPF with sampling frequency 48 kHz.

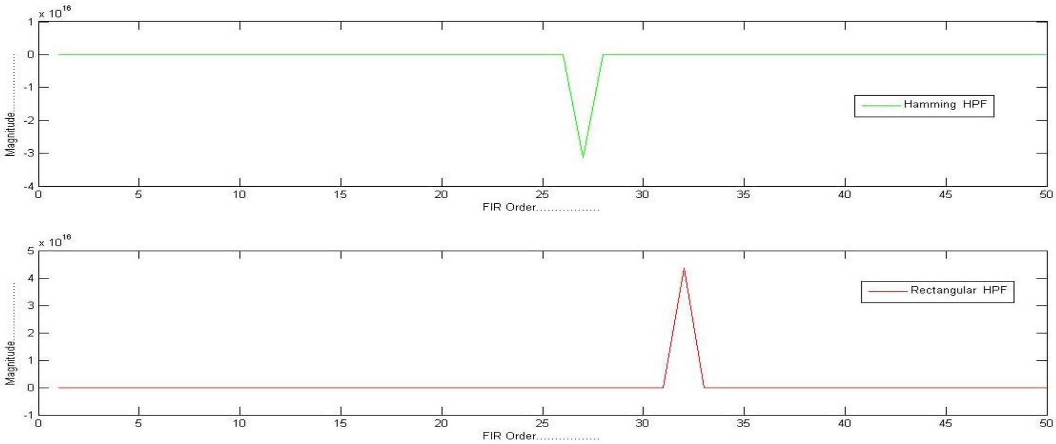

Fig.5 Response of Hamming & Rectangular window functions for LPF with sampling frequency 48 kHz.

The comparative analysis for HPF has performed at sampling frequency of 48kHz (f_s) and cutoff frequency of 10.8kHz is shown in fig.4, from fig.4 it is concluded that impulse response of Chebyshev ,Blackman and Kaiser window is a sinc function except Hamming & Rectangular window (fig.5).

Comparative analysis for Band Pass Filter

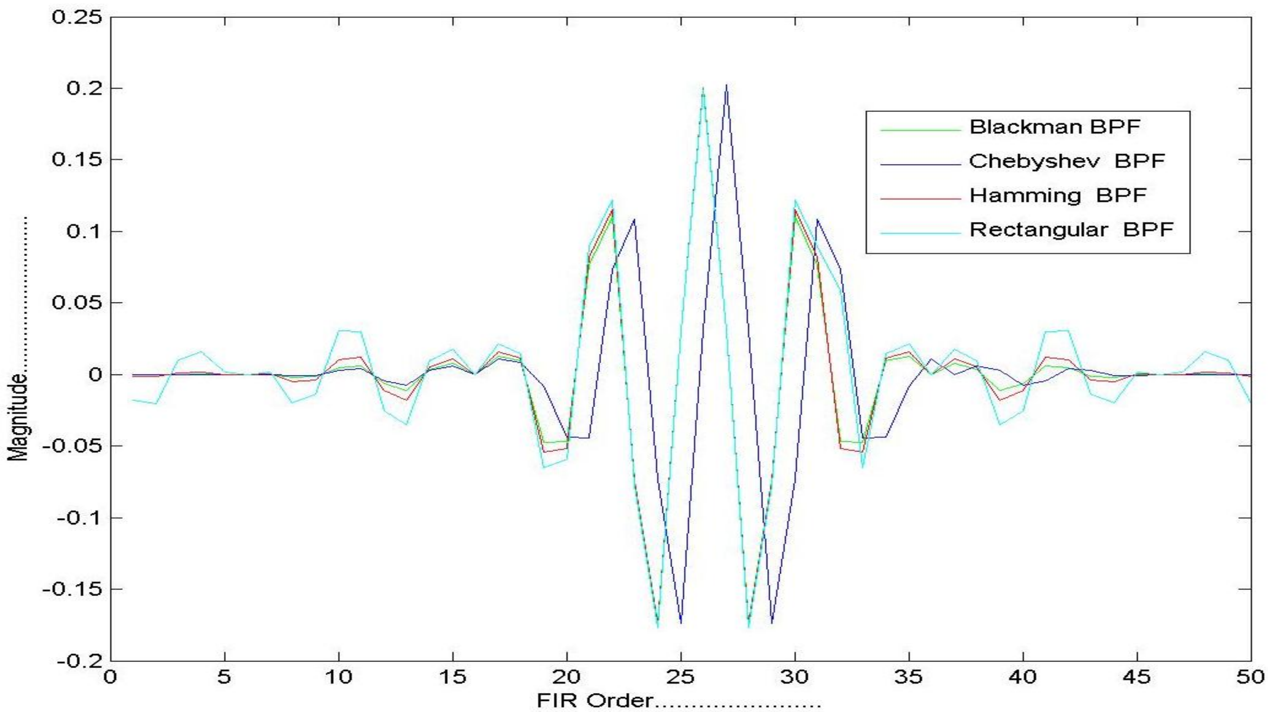

Fig.6 Output response of basic window functions for BPF with sampling frequency 48 kHz.

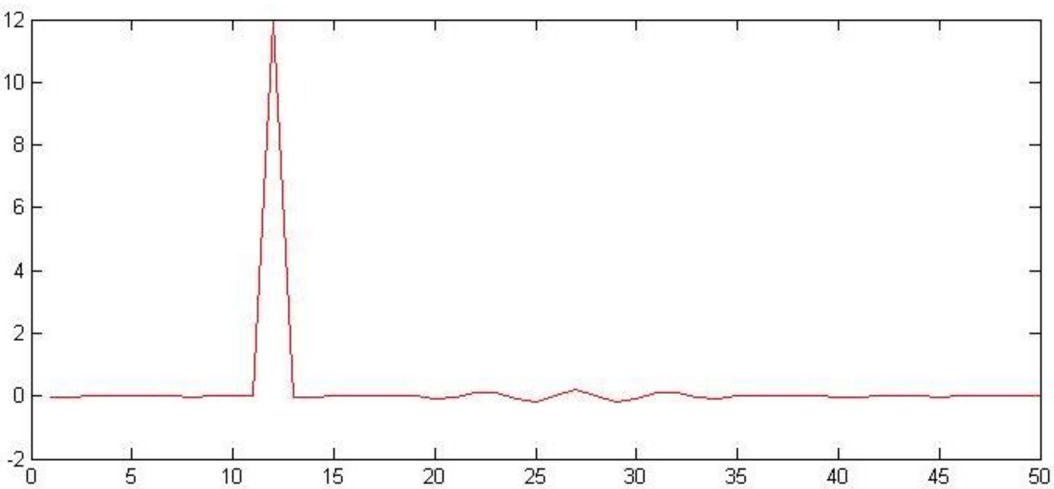

Fig.7 Response of Kaiser window functions for BPF with sampling frequency 48 kHz.

The comparative analysis for BPF has performed at sampling frequency of 48kHz (f_s) and first cutoff frequency of 8.4kHz and second cutoff frequency of 13.2kHz is shown in fig.6,from fig. 6 it is concluded that impulse response of Blackman, Chebyshev, Hamming and Rectangular window is approximately sinc function except Kaiser window (Fig.7).

Comparative analysis for Band Stop Filter

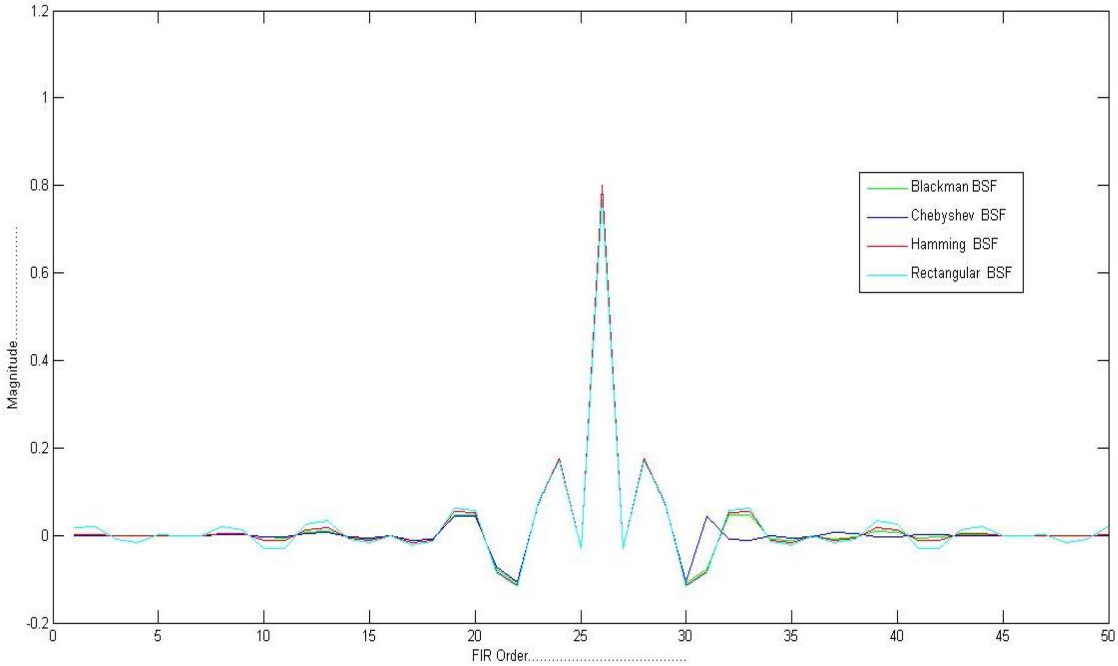

Fig.8 Output response of basic window functions for BSF with sampling frequency 48 kHz.

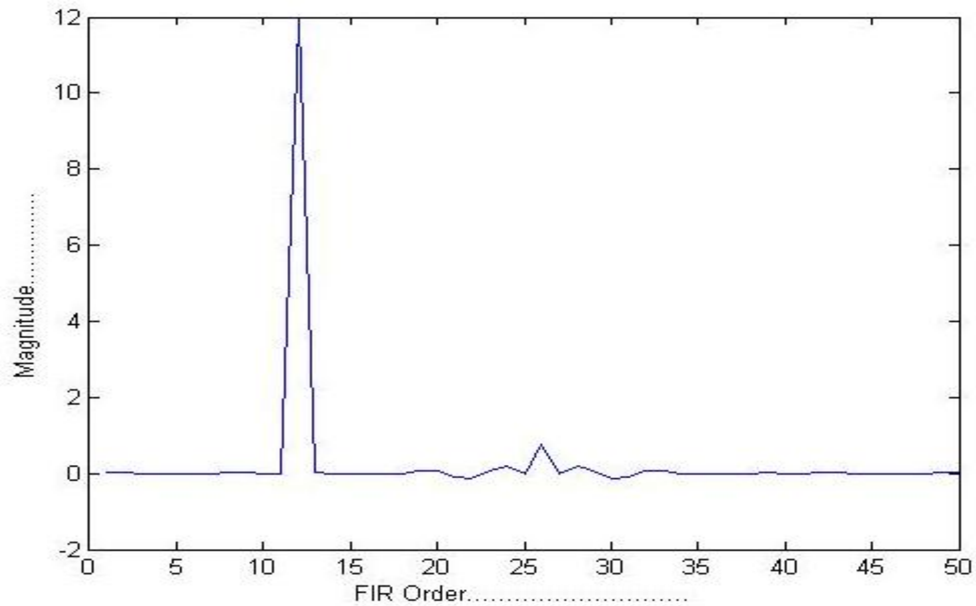

Fig.9 Response of Kaiser window functions for BSF with sampling frequency 48 kHz.

The comparative analysis for BSF has performed at sampling frequency of 48kHz (f_s) and first cutoff frequency of 8.4kHz and second cutoff frequency of 13.2kHz is shown in fig.8,from fig. 8 it is concluded that impulse response of Blackman, Chebyshev, Hamming and Rectangular window is approximately sinc function except Kaiser window (Fig.9). For all the above analysis Direct –Form Type 1 FIR Structure has been considered.

V FUTURE WORK

By utilizing the above approach, analysis may be performed for higher order FIR filters and finding of such may be used in specific application for more appropriate response. This research work further can be extended for real time implementation.

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