

Design of Bandpass Finite Impulse Response Filter Using Various Window Method

Er. Mukesh Kumar^{*}, Rohit Patel^{**}, Er. Rohini Saxena^{*}, Saurabh Kumar^{**}

^{**}PG Student, ECE, SSET, SHIATS, Allahabad, India

^{*}Assistant Professor, ECE, SSET, SHIATS Allahabad, India

Abstract

In the digital signal processing, digital filtering occupies an extremely important position. This paper introduces the definition and basic principles of FIR digital filters, and the design methods based on MATLAB. After the description of the process of design and simulation of a FIR band-pass filter by means of window function method, the results prove that various performance of the designed FIR filter reach the appointed requirement, the designed method is simple.

Keywords - FIR filter, LTI, bandpass filter, window function, MATLAB

I. INTRODUCTION

In different areas digital filter design techniques are widely used. Finite Impulse Response (FIR) filters are one of the primary types of digital filters used in various Digital Signal Processing (DSP) applications such as audio signal processing, video convolution functions and telecommunications by virtue of stability and easy implementation. Digital filters are highly flexible and portable. Digital filter has minimum or negligible interference noise and other effects. In storage and maintenance digital filters are easier and reduces the failure time. Digital filters are divided in two parts as finite impulse response (FIR) and infinite impulse response (IIR). FIR filters have greater flexibility to control the shape of their magnitude response in comparison to IIR filter. The digital filter can be divided in - lowpass, highpass, bandpass, and bandstop in frequency domain. IIR filter realization is recursive but in FIR it is non-recursive.

In digital signal processing applications digital filter plays an important role. Bandpass filter is to pass a specified frequency band and reject the other band. The filtered signal can be further used for the signal feature extraction. In noise reduction, frequency boosting, digital audio equalizing, and digital crossover, among others, digital filtering is also used.

II. FIR DIGITAL FILTER

2.1 Basic Concept of FIR filter

The basic structure of FIR filter consists of multipliers, delay elements and adders to create the filter's output. The difference equation of M order of the recursive digital filters (FIR) can be represented as:

$$B(n) = \sum_{k=0}^{M-1} h(n-k)A(n-k) = \sum_{k=0}^{M-1} b_k A(n-k)$$

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

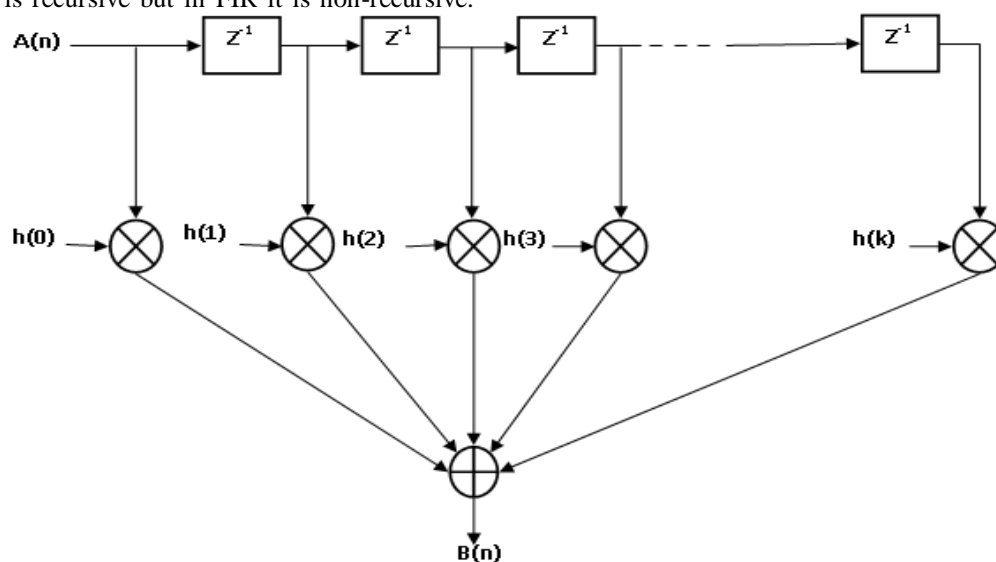


Figure.1 M-order FIR digital filter block diagram

We can express the output signal in frequency domain by convolution of the input signal A(n) and the impulse response h(n).

$$B(n) = A(n) * h(n)$$

The output signal is determined as,

$$B(n) = A(0) * h(n) + A(1) * h(n-1) + A(2) * h(n-2) + \dots + A(n) * h(0)$$

In differential equation, the coefficient b_k equals to the successive value $h(n)$ of unit-sample response. The system function H(z) can be expressed as:

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

H(z) is polynomial of z^{-1} . This means that all poles are only plotted at the origin of the Z-plane.

FIR filters can be designed in different ways, for example window method, frequency sampling method, weighted least squares method, minimax method and equiripple method. Out of these methods, the window technique is most conventional method for designing FIR filters.

2.2 Window function method of FIR filter design

Design method of window function is to change infinite noncasuality sequence into finite impulse response sequence. The basic principle of window function is to use window function of certain width to intercept infinite impulse response sequence for obtaining finite impulse response sequence. The basic design principles of window function are to calculate $h_d(n)$ by the anti-Fourier transform based on the ideal demanded filter frequency response $H_d(e^{j\omega})$. The formula of $h_d(n)$ is shows as

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

Because $h_d(n)$ is infinitely long, we have to deal with it by window function to get to the unit impulse response h(n). Now it is written as

$$h(n) = w(n) \cdot h_d(n)$$

Where w(n) is the window function. Fixed window and adjustable window are the two categories of window function. Blackman window, Hanning window, Hamming window and rectangular window are mostly used fixed window function. Kaiser window is a type of adjustable window function. The window functions are shown by the table 1.

Table1 Different Window Function

Blackman window	$w(n) = \begin{cases} (0.42 - 0.5\cos(\frac{2\pi n}{N})) + 0.08\cos(\frac{4\pi n}{N}), & \text{for } n = 0 \text{ to } N-1 \\ 0, & \text{elsewhere} \end{cases}$
Hamming window	$w(n) = \begin{cases} (0.54 - 0.46\cos(\frac{2\pi n}{N})), & \text{for } n = 0 \text{ to } N-1 \\ 0, & \text{elsewhere} \end{cases}$
Hanning window	$w(n) = \begin{cases} 0.5 \left(1 - \cos\frac{2\pi n}{M-1}\right), & \text{for } n = 0 \text{ to } N-1 \\ 0, & \text{elsewhere} \end{cases}$
Kaiser window	$w(n) = \begin{cases} I_0(\beta\sqrt{1 - (2(n+1)/(N+1))^2}) / I_0(\beta), & \text{for } n = 0 \text{ to } N-1 \\ 0, & \text{elsewhere} \end{cases}$

The Hanning window and hamming window are raised cosine window and can be used to reduce the side lobes while preserving a good frequency resolution compared to the rectangular window. Blackman window has an advantage over other windows is that it has better stop band attenuation and with less pass band ripple. The Kaiser window has parameter β . The parameter β determines the shape of the window and thus controls the trade-off between main-lobe width and side-lobe amplitude.

III. SIMULATION

The Filter Design and Analysis (FDA) tool works with MATLAB and the signal processing toolbox to provide a complete environment for start to finish filter design. The FDA tool supports many advanced techniques not available in SP tool. FDA tool is used to – design filters, quantize filter, analyze filter, modify existing filter designs, realize simulink models of quantized direct form FIR filters.

3.1 Filter Specifications

The filter specification are :

Table.1 Filter specification

Parameters	Values
Filter Type	Bandpass
Design method	FIR window ($\beta=3.4$ for Kaiser window only)
Filter order	38
Sampling frequency	100 Hz
Lower cut-off frequency	10 Hz
Upper cut – off frequency	20 Hz

Putting the above values in Simulink Model shown by figure 2.. In the simulink environment, when the signal $S(t) = \sin(10\pi t) + \sin(30\pi t) + \sin(60\pi t)$

is given to the simulation diagram then the filtering effect is shown by figures.

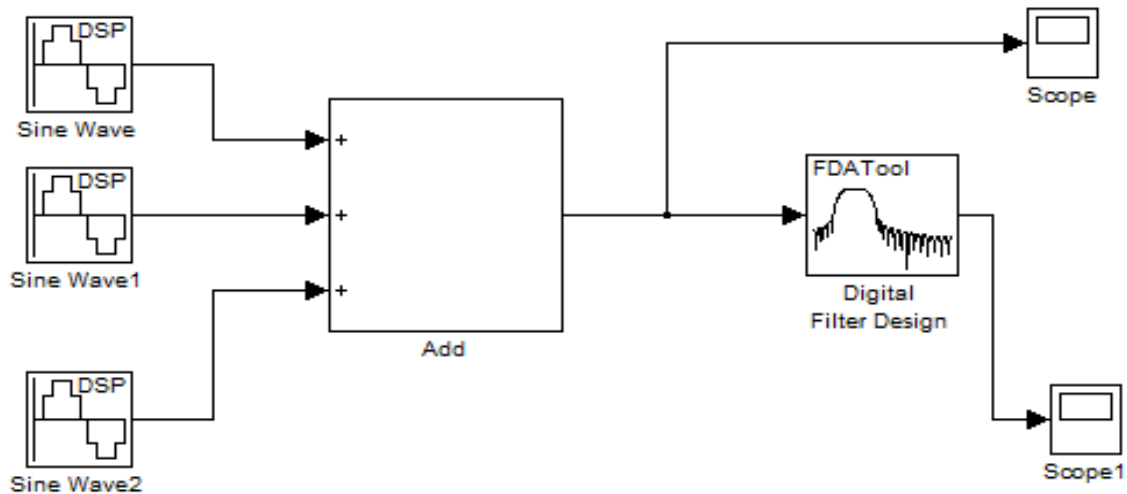


Figure 2 Simulink simulation diagram

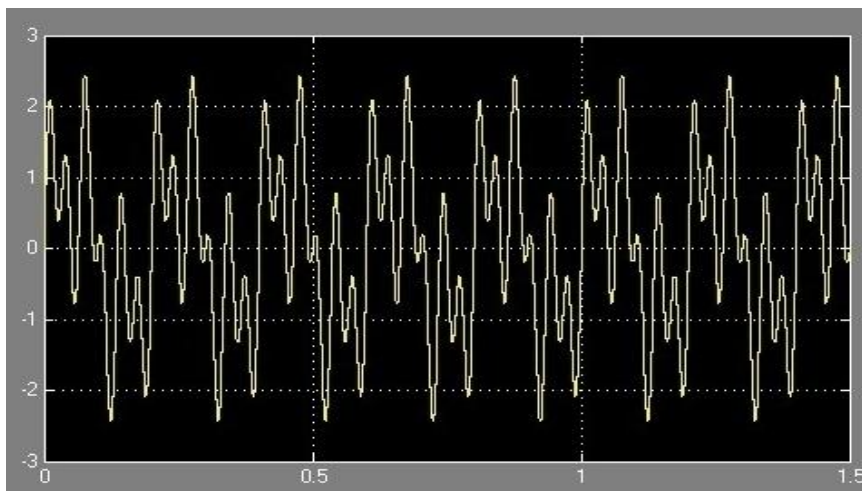


Figure 3 Original discrete waves (scope)

Figure 7 shows the original discrete waves of the simulation setup before filtering on scope. The result is

same for blackman, hanning, hamming and kaiser before filtering.

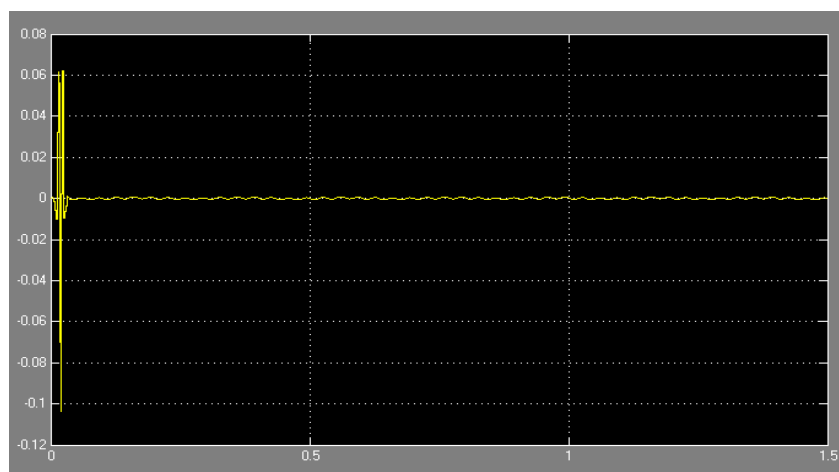


Figure 4 The discrete wave after filtering for blackman Window (scope 1)

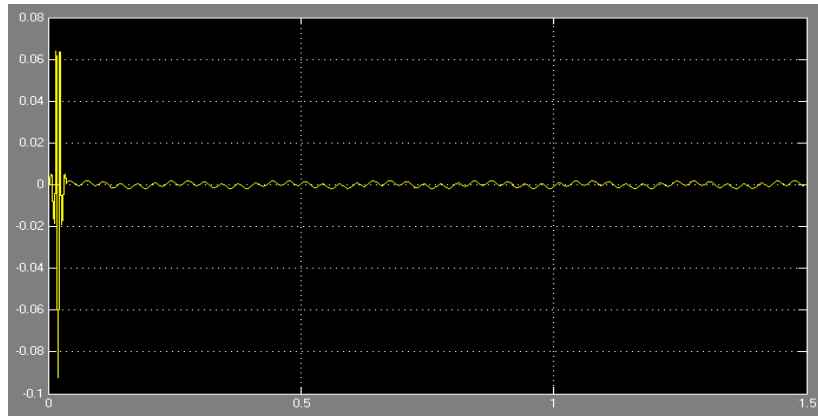


Figure 5 The discrete wave after filtering for Hamming Window (scope 1)

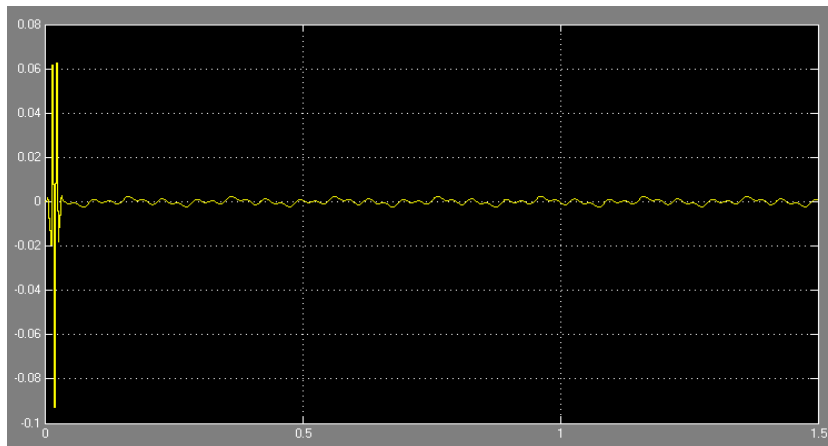


Figure 6 The discrete wave after filtering for Hanning Window (scope 1)

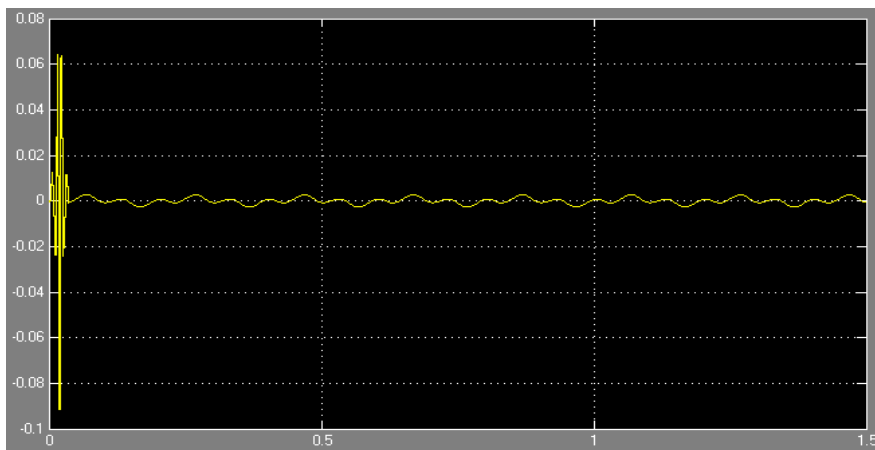


Figure 7 The discrete wave after filtering for Kaiser Window (scope 1)

Figures 4, 5, 6 and 7 show the discrete wave after filtering for kaiser window. The results show that after simulation some part of signal can pass and other part of signal is greatly damped for kaiser window than blackman, hamming and hanning window. So kaiser window is better window than other window functions. From figure 8, 9, 10 and 11, 15Hz frequency sine wave signal can pass, but the 5Hz and 30Hz frequency sine wave will be greatly damped. These figures are also shown that the Kaiser window gives the better result than other windows.

IV. CONCLUSION

FIR filter designed using MATLAB in the digital communications systems and signal processing of the computer field, have broad application prospects. FIR digital filter through the example of the design and analysis, in MATLAB environment, based on MATLAB Signal Processing Toolbox design of digital filters can be convenient, fast and correctly designed to meet the strict linear phase The FIR filter, saving a lot of programming time, improving the efficiency of programming, and parameter changes is

also very convenient. This paper shows that the Kaiser window gives the better result than other windows.

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