International Journal of
Emerging Research in Management &Technology
ISSN: 2278-9359 (Volume-3, Issue-2)

Research Article **February 2014**

Design of FIR Filter Using Hamming Window

Md. Saiful Islam[1], Md. Shariful Islam[2], Syed Khalid Rahman[3], Neelanjana Subin Ferdous[4]
Department of Electrical & Electronic Engineering
World University of Bangladesh

Jakeya Sultana Jyoti[5]
Department of Electronics & Telecommunication Engineering
Rajshahi University of Engineering & Technology

Abstract - Digital filters are pervasive in the present era of communication systems. As a result good digital filter performance is important and hence to design a digital finite impulse response (FIR) filter satisfying all the required condition is a demanding one. This report deals with the design of FIR digital filter using hamming window technique. This window is optimized to minimize the maximum (nearest) side lobe, giving it a height of about one-fifth that of the hanning window. Hence this type of filter plays very important role in spectral analysis of different types of signal. In spectral analysis applications, a small main lobe width of the window function in frequency domain is required for increasing the ability to distinguish two closely spaced frequency components.

Keywords- Filter; FIR; Hamming; Window; MATLAB.

I. INTRODUCTION

The digital signal processing has become an extremely important subject. A fundamental aspect of digital signal processing is filtering. A digital filter is a system that performs mathematical operations on a sampled, discrete-time signal to reduce or enhance certain aspects of that signal. There are two types of digital filters on the basis of the impulse response of the filter:

1. Infinite Impulse Response (IIR) filters, and
2. Finite Impulse Response (FIR) filters.

Digital filters with infinite duration impulse response referred to as IIR filters. IIR filters are recursive type filters where by the present output depends on the present input, past input and past outputs. Digital filters with finite duration impulse response referred to as FIR filters. FIR filters are non-recursive type filters where by the present output depends on present input and past inputs. FIR filters are widely used than IIR filters, because FIR digital filters have an exactly linear phase, always stable, non-recursive structure and arbitrary amplitude-frequency characteristic etc. [1], [3]. In view of the design and simulation analysis, the design of digital filter is quickly and efficiently achieved by using powerful computing capabilities of MATLAB [2].

FIR filter is described by the difference equation.

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

Where x(n) is the input signal and h(n) is the impulse response of FIR system.

To design the FIR filters the simple and effective way is window method. In this method infinite impulse response of the ideal prescribed filter is truncated by using a window function. The main advantage of this design technique is that the impulse response coefficient can be obtained in closed form and can be determined very quickly. The window method is simple in operation, easy to understand and very convenient method for designing digital FIR filter [4].

The most popular and widely used window functions are; Rectangular window, hanning window, hamming window and Kaiser Window. The Rectangular window response provides side lobes which gives rise to ripples in pass band and stop band. The amplitude of the ripples is determined by the amplitude of the side lobes. For the rectangular window, the amplitude of the side lobes is unaffected by the length of the window. But the main lode width of rectangular window is narrower and higher. For fixed length, the hanning window has significantly lower side-lobe amplitude but the main lobe width is wider compared to Rectangular window. The Hamming window also has the same main lobe width of hanning window but it generates lesser oscillations in the side lobes than hanning window. Hence Hamming window is generally preferred rather than hanning window. The Kaiser window is a kind of adjustable window function which provides independent control of the main lobe width and ripple ratio but the Kaiser window has the disadvantage of higher computational complexity due to the use of Bessel functions.

With regard to these window methods a Hamming window technique is developed here. FIR filter design using Hamming window function provides smaller main lobe width and sharp transition band. This type of filter is very useful in spectral analysis of different types of signals.

II. DESIGN METHODS OF FIR FILTERS

There are different methods for the design of FIR digital filter. The most common methods are

1. Fourier series method.
2. The window method.

3. Frequency sampling method.
4. Optimal filter design method.

1.Fourier series method:

In this method, the desired frequency response specification $H_d(w)$ and corresponding unit sample response $h_d(n)$ is determined using the following relation

$h_d(n)=\frac{1}{2\pi}\int_{-\pi}^{\pi} Hd(e^{jw})exp(jwn)dw$................(2)

Where, $H_d(e^{jw}) = \sum_{n=-\infty}^{n=+\infty}(hd(n)\,e^{-jnw})$...........(3)

In general, unit sample response $h_d(n)$ obtained from the above relation is infinite in duration, so it must be truncated at some point say $n=\pm\frac{(N-1)}{2}$ to yield an FIR filter of length N. The amplitude response of the filter exhibits oscillations in the pass band as well as in the stopband, which are known as Gibb's oscillations. They are caused by the truncations of the Fourier series. As the length of the filter increased, the frequency of the oscillations increases but the amplitude stays constant. In other words, we do not seem to be able to reduce the pass band and stop band errors below a certain limit by increasing the filter length.Therefore, the filters that can be designed using the Fourier series method are of little practical usefulness.

2. Window method:

Several window functions have been proposed .Listed below are some of the most common:

- Rectangular window
- Hanning window (also referred to as Hann)
- Hamming window
- Blackman window
- Kaiser window

To reduce the oscillations in Fourier series method, the Fourier coefficients are modified by multiplying the infinite impulse response by a finite weighing sequence *w* (n) called a window. Windows are characterize by the main lobe width which is the bandwidth between first negative and first positive zero crossing, and by their ripple ratio. The main lobe width and the ripple ratio should be as low as possible; i,e the spectral energy of the window should be concentrated as far as possible in the main lobe and the energy in the side lobes should be as low as possible.

Two desirable characteristics of a window function are [2];

(1). Fourier transform of the window function should have a small width of the main lobe.

(2).Fourier transform of the window function should have side lobes that decrease in energy rapidly as / tends to 0.

Windows can be categorized as fixed or adjustable window function. Fixed windows such has Rectangular, Hanning, Hamming and Blackman window have only one independent parameter window length which controls the main-lobe width. Adjustable windows have two or more independent parameters such as window length and one or more additional parameters that can control the other window's characteristics [5, 7]. The Kaiser window is a kind of two parameter window function. In a Kaiser window width of main lobe can be controlled by adjusting the length of the filter and side lobe level can be controlled by varying the other parameter α. But the Kaiser window has the disadvantage of higher computational complexity due to the use of Bessel functions in the calculation of the window coefficients.

An adjustable window function equation based on hamming window function is given by FREDRIC J. HARRIS [6]. In this window function the width of main lobe can be varied by changing the value of α for a fixed length of the filter. α is selected according to different applications. This generalized window is referred to as the Hamming window for α = 0.54 and Hanning window for α= 0.5. They are both commonly used in speech processing and other digital signal processing applications. But in some application such as spectral analysis of a specified frequency spectrum, if the frequency of interest contains two or more signals very near to each other, then frequency resolution is very important. In such cases a small main lobe width of the window function in frequency domain is required. For an efficient value of α, this window function provides a lesser main lobe width compares to Hanning (α = 0.5) and Hamming window (α= 0.54), however the amplitude of side lobe and ripples in pass band is also increased. The function is

$W(n) = \{ \alpha-(1-\alpha)\cos(\frac{2\pi n}{M-1}),\ 0\le n\le M-1;$...................(4)

Otherwise the value of W(n) is zero.

3.Frequency sampling method:

In this method the given frequency response is sampled at a set of equally spaced frequencies to obtain *N* samples. Thus, sampling the continuous frequency response $H_d(w)$ at *N* points essentially gives us the *N*-point DFT of $H_d(2pnk/N)$. Thus by using the IDFT formula, the filter coefficients can be calculated using the following formula

$h(n)=\frac{1}{N}\sum_{n=0}^{N-1}(H(k)e^{(\frac{j2\pi n}{N})k})$.......................(5)

The main attraction in this method is that it allows for a recursive realization of FIR filters, which can be computationally very efficient.However, it also lacks flexibility in specifying or controlling filter parameters.

4.Optimal filter design method:
With efficient and easy to use programs it is now widely used in industry. To reduce the error in frequency sampling method this method is used and this method is best method to design the FIR filter. The basic idea of this method is to design the filter coefficients again and again until a particular error is minimized. But to determine the filter coefficients by using optimal filter design method is very complex.
A complete procedure for the design of FIR filters are as follows:

- Specification of the filter.
- Choosing an appropriate linear phase filter type.
- Choosing the method of design such as window.
- Calculation of the filter coefficients (impulse response).
- Analysis of the finite word length effects.
- Implementation of the filter in hardware and software.

III. SIMULATION RESULTS

The design of FIR filter using hamming window function for different values of ripple and frequency are shown in the figure below. we considered a pass band ripple of 0.05 and a stop band ripple of 0.03,a pass band frequency of 1300 rad/s and a stop band frequency of 1600rad/s with a normalized frequency of 7400rad/s. Considering these values the order of the filter becomes 26.After simulation it is observed that this window function provides lesser main lobe width than the other window techniques, however the amplitude of the side lobe and ripples in pass band is also increased.

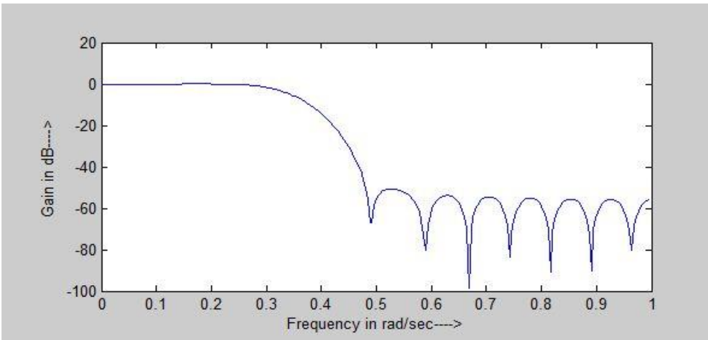

Fig 1. Low pass FIR filter

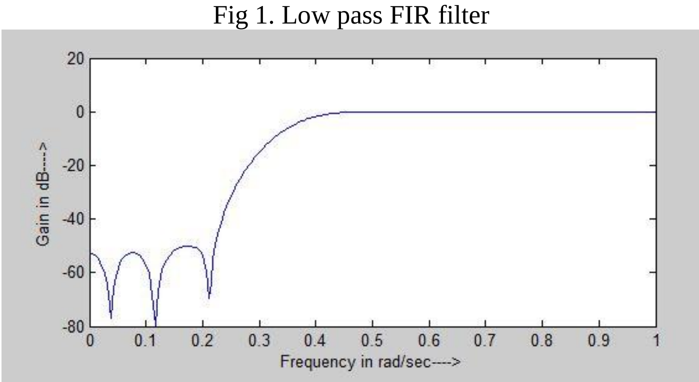

Fig 2. High pass FIR filter

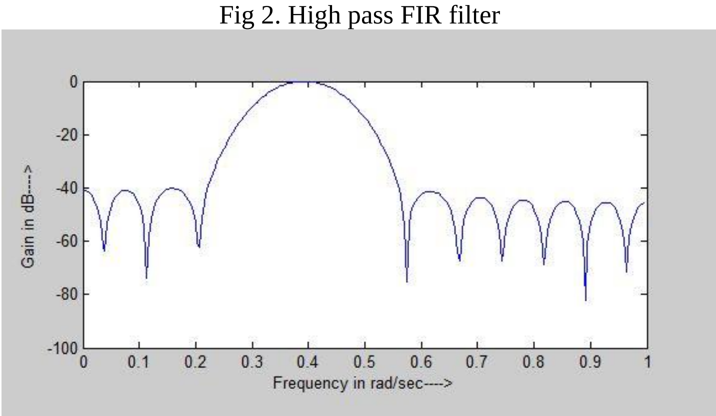

Fig 3. Band pass FIR filter

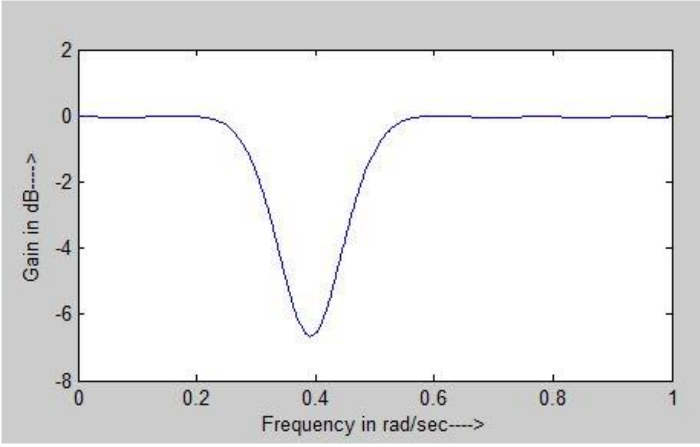

Fig 4. Band stop FIR filter

IV. APPLICATIONS

From the above simulation result it is found that the designed FIR filter has lesser main lobe width so it can be used in speech processing applications such as speech filtering, noise reduction, frequency boosting and digital audio equalizing etc. In speech filtering; filter are used to modify the frequency response of a speech signal according to require applications of speech processing. Designed low-pass FIR filter is used to eliminate the high-frequency spectrum of the speech signal also as the designed filter has lesser main lobe width so it is very useful for spectral analysis of a signal.

V. CONCLUSION

In this paper, an FIR filter has been designed using hamming window function. This type of window function is simple in operation and provides greater flexibility in digital signal processing applications. In frequency resolution problems a small main lobe width of window function in frequency domain is required. In the range $0 \leq \alpha \leq 1$, for large values of α, as the value of α increase the main lobe width is continuously decreasing; however the amplitude of side lobe is also increased**.** In signal processing applications digital filters are more preferable than analog filters. The digital filters are easily designed and also easy to use in several of signal filtering applications. The choice of technique to design the filter depends heavily on the decision of designer whether to compromise accuracy of approximation .FIR filter design by using hamming window is stable as compare to rectangular and hanning window techniques. Ripples in pass band are less in hamming as compared to rectangular and hanning and also hamming has a linear phase than rectangular and hanning.

REFERENCES

[1] J.G. Proakis and D. G. Manolakis, *Digital Signal Processing Principles, Algorithms and Applications* 3rd Edition Prentice- Hall, 2002.

[2] Sanjit K. Mitra, *Digital Signal Processing: A Computer-base approach*, Tata McGraw-Hill, 2nd Edition 2001.

[3] P. Ramesh Babu, *Digital Signal Processing* 4th Edition.

[4] Sonika Gupta, Aman Panghal Performance *Performance analysis of FIR Filter Design by using Rectangular, Hanning, and Hamming window methods* International Journal of Advanced Research in Computer Science and Software Engineering Volume 2, Issue 6, June 2012.

[5] T. Saramaki, *Finite impulse response filter design* in *Handbook for Digital Signal Processing*, Edited by S. K. Mitra and J. F. Kaiser, IBSN 0-471-61995-7 John Wiley &Sons Inc, 1993

[6] FREDRIC J.HARRIS *On the Use of Windows for Harmonic Analysis with the Discrete Fourier Transform processing of IEEE* vol.66 no.1 January 1978.

[7] Oppenheim, R. Schafer, and J. Buck, *Discrete-Time Signal Processing* second edition, Prentice-Hall,1999.