



Performance Analysis of FIR Filter Design by Using Rectangular, Hanning and Hamming Windows Methods

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ABSTRACT: Digital filter are widely used in the world of communication and computation. On the other hand to design a digital finite impulse response (FIR) filter that satisfying all the required conditions is a challenging one. In this paper, design techniques of low pass FIR filters using Blackman window method, Optimal Parks McClellan method and Genetic Algorithm method are presented. The stability, number of components required and filter coefficients are demonstrated for different design techniques. It is shown that filter design by using GA is best because the numbers of components required are less and stability is more as compare with other techniques. Design comparisons are presented to show the effectiveness of GA optimization method

KEYWORDS – Rectangular, Hanning and Hamming windows, FIR Filter.

I. INTRODUCTION

Digital filtering is one of the most powerful tools of DSP. Digital filters are capable of performing that specifications which are extremely difficult, to achieve with an analog implementation. In addition, the characteristics of a digital filter can be easily changed under software control. Many digital systems use signal filtering to remove unwanted noise, to provide spectral shaping, or to perform signal detection or analysis. Two types of filters provide these functions are finite impulse response (FIR) filters and infinite impulse response (IIR) filters. Typical filter applications include signal preconditioning, band selection, and low pass filtering.

A. FIR has following advantage over IIR Filter

1. FIR filter is Finite IR filter and IIR filter is Infinite IR filter.
2. FIR filters are non-recursive. That is, there is no feedback involved. Where as an IIR filter is recursive. There is feedback involved
3. The impulse response of an FIR filter will eventually reach zero. The impulse response of an IIR filter may very well keep "ringing" ad-infinitum.
4. IIR filters may be designed to accurately simulate "classical" analog filter responses where as FIR filters, in general, cannot do this.
5. FIR filter has linear phase and easily control where as IIR filter has no particular phase and difficult to control
6. FIR filter is stable and IIR filter is unstable
7. FIR filter depend only on I/P where as IIR filter depend upon both I/P and O/p
8. FIR filter consist of only zeroes and IIR filter consist of both poles and zeroes.

FIR filters are filters having a transfer function of a polynomial in z^{-1} and is an all-zero filter in the sense that the zeroes in the

z -plane determine the frequency response magnitude characteristic. The z transform of a N -point FIR filter is given by

$$H(z) = \sum_{n=0}^{N-1} h(n)z^{-n}$$

FIR filters are particularly useful for applications where exact linear phase response is required.

The FIR filter is generally implemented in a non-recursive way which guarantees a stable filter.

FIR filter design essentially consists of two parts

- (i) Approximation problem
- (ii) Realization problem

The approximation stage takes the specification and gives a transfer function through four steps. They are as follows:

- (i) A desired or ideal response is chosen, usually in the frequency domain.
- (ii) An allowed class of filters is chosen (e.g. the length N for a FIR filters).
- (iii) A measure of the quality of approximation is chosen.
- (iv) A method or algorithm is selected to find the best filter transfer function.

The realization part deals with choosing the structure to implement the transfer function which may be in the form of circuit diagram or in the form of a program.

There are essentially three well-known methods for FIR filter design namely:

- (1) The window method

- (2) The frequency sampling technique
- (3) Optimal Filter Design Method

B. Applications

Traditionally, most digital filter applications have been limited to audio and high-end image processing. With advances in process technologies and digital signal processing methodologies, digital filters are now cost-effective in the IF range and in almost all video markets. Digital filters are commonly used for audio frequencies for two reasons. First, digital filters for audio are superior in price and performance to the analog alternative. Second, audio Analog-to-Digital Converters (A/Ds) and Digital-to-Analog Converters (DACs) can be manufactured with high accuracy and are available at low cost. Thus, the combined cost of filtering and conversion (if necessary) is low. The cost trades are much more difficult in the 1MHz to 100MHz signal range, such as the IF ranges of many radio receivers.

2. The Window Method

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

$$h_d(n) = 1/2\pi \int_{-\pi}^{\pi} H_d(w) e^{jwn} dw$$

$$H_d(w) = \sum_{n=-\infty}^{\infty} h_d(n) e^{-jwn}$$

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 $n = M-1$ to yield an FIR filter of length M (i.e. 0 to $M-1$). This truncation of $h_d(n)$ to length $M-1$ is same as multiplying $h_d(n)$ by the rectangular window defined as

$$W(n) = \begin{cases} a - b \cos(2\pi(n+1)/(N+1)) + c \cos(4\pi(n+1)/(N+1)) & n = 0, 1, \dots, N-1 \\ 0 & \text{otherwise} \end{cases}$$

Thus the unit sample response of the FIR filter becomes

$$h(n) = \begin{cases} h_d(n) w(n) & 0 \leq n \leq M-1 \\ 0 & \text{otherwise} \end{cases}$$

Now, the multiplication of the window function $w(n)$ with $h_d(n)$ is equivalent to convolution of $H_d(w)$ with $W(w)$, where $W(w)$ is the frequency domain representation of the window function

$$W(w) = \sum_{n=0}^{M-1} w(n) e^{-jwn}$$

Thus the convolution of $H_d(w)$ with $W(w)$ yields the frequency response of the truncated FIR filter

$$H(w) = 1/2\pi \int_{-\pi}^{\pi} H_d(v) W(w-v) dv$$

The frequency response can also be obtained using the following relation

$$H(w) = \sum_{n=0}^{M-1} h(n) e^{-jwn}$$

Direct truncation of $h_d(n)$ to M terms to obtain $h(n)$ leads to the **Gibbs phenomenon** effect which manifests itself as a fixed percentage overshoot and ripple before and after an approximated discontinuity in the frequency response due to the non-uniform convergence of the fourier series at a discontinuity. Thus the frequency response obtained by using above equation contains ripples in the frequency domain. In order to reduce the ripples, instead of multiplying $h_d(n)$ with a rectangular window $w(n)$, $h_d(n)$ is multiplied with a window function that contains a taper and decays toward zero

gradually, instead of abruptly as it occurs in a rectangular window. As multiplication of sequences $h_d(n)$ and $w(n)$ in time domain is equivalent to convolution of $H_d(w)$ and $W(w)$ in the frequency domain, it has the effect of smoothing $H_d(w)$. The several effects of windowing the Fourier coefficients of the filter on the result of the frequency response of the filter are as follows:

- (i) A major effect is that discontinuities in $H(w)$ become transition bands between values on either side of the discontinuity.
- (ii) The width of the transition bands depends on the width of the main lobe of the frequency response of the window function, $w(n)$ i.e. $W(w)$.
- (iii) Since the filter frequency response is obtained via a convolution relation, it is clear that the resulting filters are never optimal in any sense.
- (iv) As M (the length of the window function) increases, the main lobe width of $W(w)$ is reduced which reduces the width of the transition band, but this also introduces more ripple in the frequency response.
- (v) The window function eliminates the ringing effects at the band edge and does result in lower side lobes at the expense of an increase in the width of the transition band of the filter.

The FIR filter design process via window functions can be split into several steps:

1. Defining filter specifications;
2. Specifying a window function according to the filter specifications;
3. Computing the filter order required for a given set of specifications;
4. Computing the window function coefficients;
5. Computing the ideal filter coefficients according to the filter order;
6. Computing FIR filter coefficients according to the obtained window function and ideal filter coefficients;
7. If the resulting filter has too wide or too narrow transition region, it is necessary to change the filter order by increasing or decreasing it according to needs, and after that steps 4, 5 and 6 are iterated as many times as needed.

The final objective of defining filter specifications is to find the desired normalized frequencies (ω_c , ω_{c1} , ω_{c2}), transition width and stopband attenuation. The window function and filter order are both specified according to these parameters.

Some of the windows commonly used are as follows:

1. Bartlett triangular window:

$$W(n) = \begin{cases} \frac{2(n+1)}{N+1} & n=0, 1, 2, \dots, (N-1)/2 \\ 2 - \frac{2(n+1)}{N+1} & n=(N-1)/2 \dots N-1 \\ 0, & \text{otherwise} \end{cases}$$

2-5. Generalized cosine windows (Rectangular, Hanning, Hamming and Blackman)

$$W(n) = \begin{cases} a - b \cos(2\pi(n+1)/(N+1)) + c \cos(4\pi(n+1)/(N+1)) & n=0, 1, \dots, N-1 \\ 0 & \text{otherwise} \end{cases}$$

6. Kaiser window with parameter β

$$W(n) = \begin{cases} 10^{\frac{\beta \left(\left(1 - \left(\frac{2(n+1)}{N+1} \right)^2 \right)^{1/2} \right)}{10(\beta)}} & n=0,1,\dots,N-1 \\ 0 & \text{otherwise} \end{cases}$$

The Bartlett window reduces the overshoot in the designed filter but spreads the transition region considerably. The Hanning, Hamming and Blackman windows use progressively more complicated cosine functions to provide a smooth truncation of the ideal impulse response and a frequency response that looks better. The best window results probably come from using the Kaiser window, which has a parameter β that allows adjustment of the compromise between the overshoot reduction and transition region width spreading.

Special attention should be paid to the fact that minimum attenuation of window function and that of the filter designed using that function are different in most cases. The difference, i.e. additional attenuation occurs under the process of designing a filter using window functions. This affects the stopband attenuation to become additionally higher, which is very desirable. However, a drawback of this method is that the minimum stopband attenuation is fixed for each function.

II. Simulation and Results:-

In simulation number of sample point (N)=65, Filter Order=61, $W_s=.4580, W_p=.341$

3.1 Ideal Magnitude Response

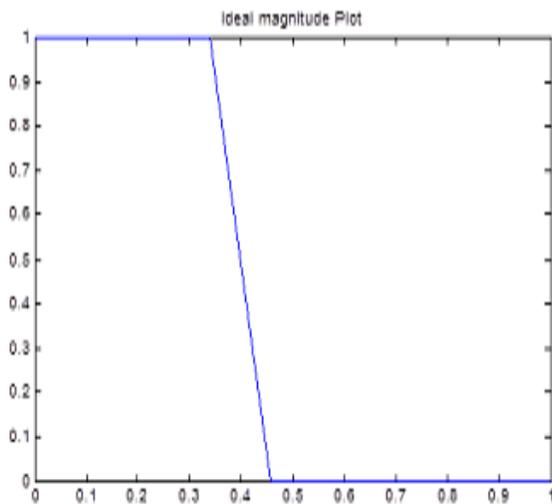


Fig:-1

3.2 Magnitude response of rectangular window

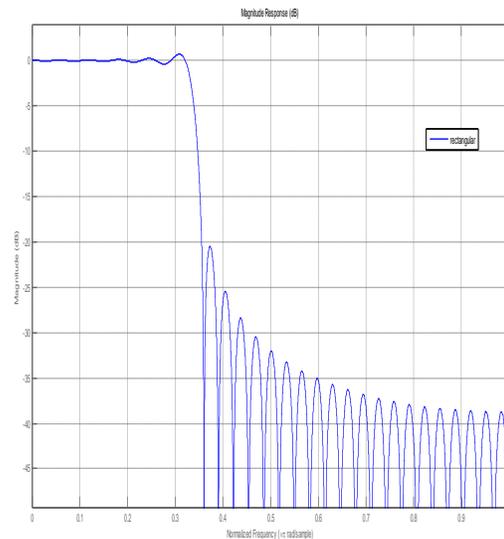


Fig:-2

3.3 Magnitude response of hanning window

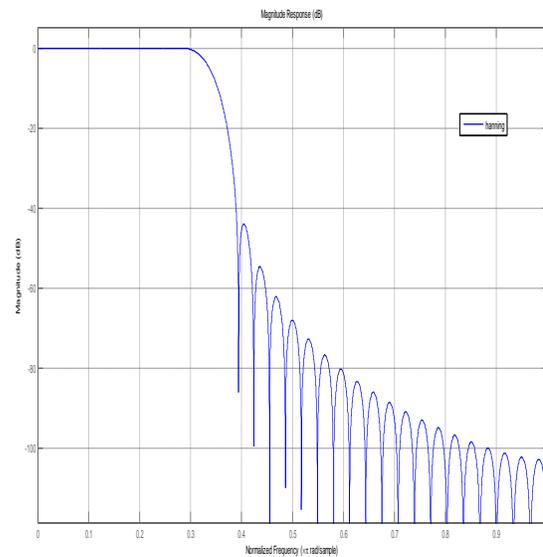


Fig:-3

3.4 Magnitude response of hamming window

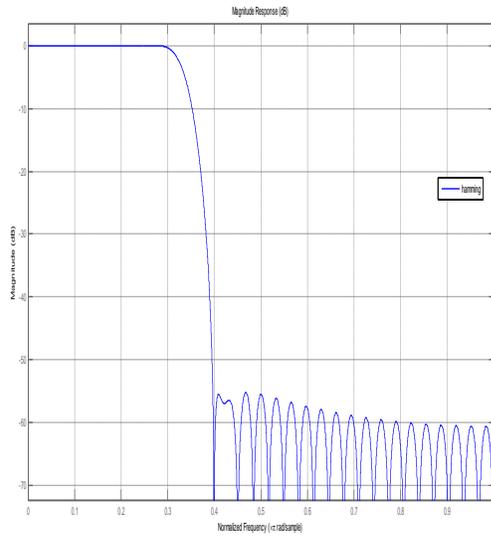


Fig:-4

3.6 Phase Response of Hanning window

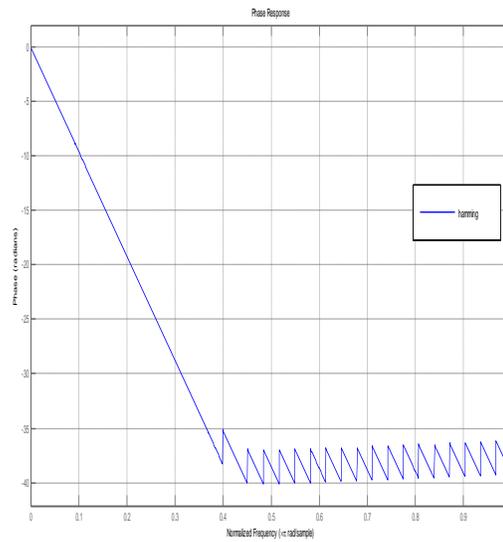


Fig:-6

3.5 Phase Response of Rectangular window

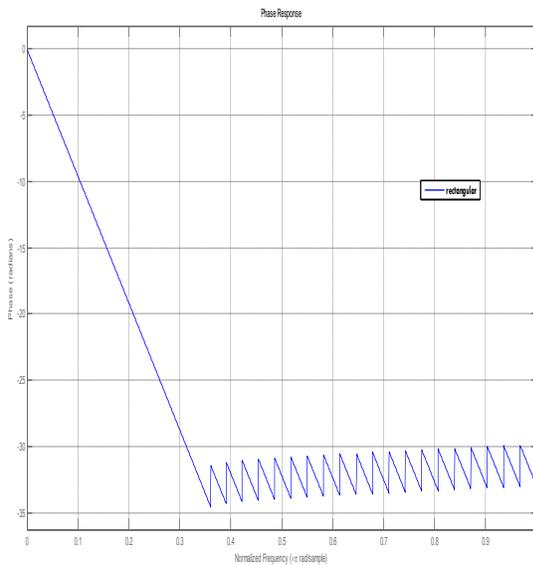


Fig:-5

3.7 Phase Response of Hainmmg window

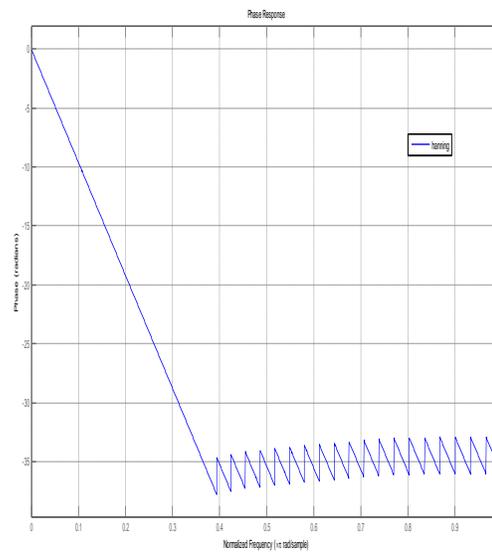


Fig:-7

4. Conclusion

FIR filter design by using hamming is stable as compare to rectangular and hanning windows techniques. Ripples in pass band are less in hamming as compare to other two techniques (as shown in fig 2, 3, 4).

Hamming has linear phase as compare to rectangular and hanning windows.

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