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Design of Low Pass Filter Using Rectangular and Hamming Window Techniques

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Abstract: this paper deals with some of the techniques used to design fir low pass filter using rectangular and hamming window technique. The magnitude and phase responses are demonstrated for window different design techniques at particular cutoff frequency and filter order. There are lot of windows with completely different properties in time and frequency domain. This paper presents the appraisal of the rectangular and hamming window techniques for designing low pass fir filter using matlab simulations.
keywords:-dsp ,digital filter ,low pass fir filter, rectangular and hamming window techniques.

I. INTRODUCTION

Digital signal processing (DSP) refers to a range of techniques for improving the accurateness and reliability of digital infrastructure. DSP refers to manipulating analog signals such as sound or photograph that has to be converted into digital form. Signal processing is concerned with representing signals in mathematical terms and extracting the information by carrying out the algorithmic operations on the signal [1][2].

Digital Signal Processing (DSP) is an important field of study that has come about due to advances in communication theory, digital computer technology and consumer devices. Digital representations of the real world, which requires discrete sampling since the values of the real - world signal are sampled at predefined, discrete intervals, and further more can only take on predefined, discrete values[2].

The following are a few of the advantages of Digital Signal Processing:-

A. High Accurateness

The accuracy of the analog filter is affected by the tolerance of the circuit components used for design the filter, but DSP has superior control of accuracy.

B. Cheaper

The digital realization is much cheaper than the analog realization in many applications.

C. Flexibility in design

For reconfiguring an analog system, we can only do it by redesign of system hardware; where as a DSP System can be easily reconfigured only by changing the program.

D. Simplicity of Data Storage

On magnetic media, without the loss of reliability the digital signals can be stored and can be processed off-line in a remote laboratory.

E. Time allocation

The cost of the processing signal can be reduced in DSP by the sharing of a given processor among a number of signals[4][5].

The following are a few of the boundaries of Digital Signal Processing are as follows:-

1) Processing of signals involves more power utilization.

2) Processing of signals further than higher frequencies (beyond GHz) and below lower frequencies (a few Hz) involves limitations.

3) Information is vanished because we only take samples of the signal at intervals [4].

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II. DIGITAL FILTERS

Digital filters are capable of performing that specifications which are enormously difficult, to achieve with an analog implementation. In addition, the uniqueness of a digital filter can be easily changed under software control. Many digital systems use signal filtering to eliminate unwanted noise in a communication receiver, to give spectral shaping, or to execute signal detection. Digital signal processing is used in various areas such as telecommunication ,military radar signal processing, image analysis and recognition ,speech processing, biomedical monitoring and signal filtering[2][3].There are four types of filters which are as follows:-

A. Low Pass Filter

A low pass filter is a filter that passes signals with a frequency lower than a definite cutoff frequency and attenuates signals with frequencies higher than the cutoff frequency. The exact frequency response of the filter depends on the filter design.

B. High Pass Filter

A high-pass filter is an electronic filter that passes signals with a frequency higher than a certain cutoff frequency and attenuates signals with frequencies lower than the cutoff frequency. The amount of attenuation for each frequency depends on the filter design.

C. Band Pass Filte

A band-pass filter is a device that passes frequencies within a certain range and rejects (attenuates) frequencies outside that range.

D. Band Stop Filte

In signal processing, a band-stop filter or band-rejection filter is a filter that passes most frequencies unaltered, but attenuates those in a specific range to a very low levels.It is the reverse of band pass band-pass filter [1][2][3][6].

A discrete time filter produces a discrete time output sequence y(n) for the discrete time input sequence x(n).A filter may be required to have a given frequency response ,or a specific response to an impulse, step or ramp function. [1][2].Digital filters are classified as two type depending on the form of the unit pulse response of the system which are as follows:-(i).Finite impulse response (FIR) filters Infinite impulse response (IIR) filters

1) Advantages of FIR filter over IIR filter are as follows

a) FIR filters can attain linear phase response and pass a signal without phase distortion

b) They are easier to realize than IIR filters

c) The filter start-up transients have predetermined duration[3][4].

2) There are different methods to find the coefficients of digital filter from frequency specifications:

a) Fourier series metho

b) The window metho

c) Frequency sampling meth

d) Optimal filter design method

FIR filters are also known as non-recursive filters, convolution filters, or moving-average filters because the output values of an FIR filter are described as a finite convolution. A good digital filter performance is obligatory and hence to design a digital finite impulse response (FIR) filter satisfying all the required conditions is necessary. One of the task is to use proper time window. There are numerous windows with completely different properties in time and frequency domain .Some of these windows are Triangular , Rectangular , Hanning , Hamming and Blackman having different side lobes ,attenuation and transition width[2][3].

In signal processing, a finite impulse response (FIR) is a filter whose impulse response (or response to any finite length input) is of finite duration, because it settles to zero in finite time.FIR filter has linear phase and easily control[1][2].An FIR filter of Length M with input x(n) and output y(n) is described by the difference equation:

$$y(n) = \sum_{K=0}^{M-1} b_k x(n-k)$$

where b_k is a set of filter coefficients.

III. WINDOW TECHNIQUES

In signal processing, a window function (also known as an apodization function or tapering function a mathematical function that is zero-valued outside of some chosen interval[4].

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To decrease the oscillation in Fourier series method, the fourier coefficient are modified by multiplying the infinite impulse response by a finite weighing sequence (w_n) called a window. Windows are characterized by the main lobe width which is the bandwidth between first negative and first positive zero crossing, and by their ripple ratio[5][6][7]. Two desirable characteristic of a window function are:

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

Fourier transform of the window function should have side lobe that decrease in energy rapidly tends to zero.

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

A. *Rectangular Window*

The rectangular window (sometimes known as the boxcar or Dirichlet window) is the simplest window, equivalent to replacing all but *N* values of a data sequence by zeros, making it appear as though the waveform suddenly turns on and off. [1][7]The desired function for the rectangular window is given by:-

$w_R(n)$ =1, for [n] ≤ $\frac{M-1}{2}$

0 otherwise

The spectrum of $w_R(n)$ can be obtained by taking fourier transform of the above equation:

$$w_R(e^{j\omega t}) = \sum_{n=\frac{-(M-1)}{2}}^{\frac{(M-1)}{2}} e^{-j\omega nT}$$

Substituting n =m - $\frac{(M-1)}{2}$ and replacing m by n, we get[2]:-

$$w_R(e^{j\omega t}) = \frac{\sin(\frac{\omega MT}{2})}{\sin(\frac{\omega T}{2})}$$

B. *Hamming Window*

The window with these particular coefficients was proposed by Richard W. Hamming. He observed that the sidelobes of the Rectangular and Hamming windows are phase reversed relative to each other, so a linear combination of the two would tend to cause them to stop each other .The Hamming window is optimized to minimize the maximum (nearest) side lobe, giving it a height of about one-fifth that of the Hann window[2][3][7].The coefficients of a Hamming window are computed from the following equation:

$$w_H(n) = 0.54 - 0.46\cos\frac{2\pi n}{M-1}, \quad 0 \le n < M-1$$

0 otherwise

The spectrum of Hamming window can then be obtained as[2]:-

$$\mathrm{w_H}(e^{j\omega t}) = 0.54\frac{\sin(\frac{\omega mT}{2})}{\sin(\frac{\omega T}{2})} - 0.46\frac{\sin(\frac{\omega MT}{2} - \frac{M\pi}{(M-1)})}{\sin(\frac{\omega T}{2} - \frac{\pi}{(M-1)})} - 0.46\frac{\sin(\frac{\omega MT}{2} + \frac{M\pi}{(M-1)})}{\sin(\frac{\omega T}{2} + \frac{\pi}{(M-1)})}$$

Table-1 PARAMETER SPECIFICATION

Parameter	Values
Sampling frequency(Fs)	48000Hz
Cut off Frequency (Fc)	10800Hz
Order(N)	15

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Table-2 FREQUENCY AND MAGNITUDE

FREQUENCY	WINDOW TECHNIQUE	
	RECTANGULAR	HAMMING
0.1π	13.21598	13.87298
0.2π	10.23176	1.30298
0.3π	3.98485	-29.22645
0.4π	-36.1124	-24.07243
0.5π	0	-30.62168
0.6π	1.84435	-21.6182
0.7π	-2.6534	-23.42138
0.8π	-34.26805	-40.27193
0.9π	-2.98422	-25.41986
π	-0.00915	-21.94702

IV. RESULTS AND SIMULATION

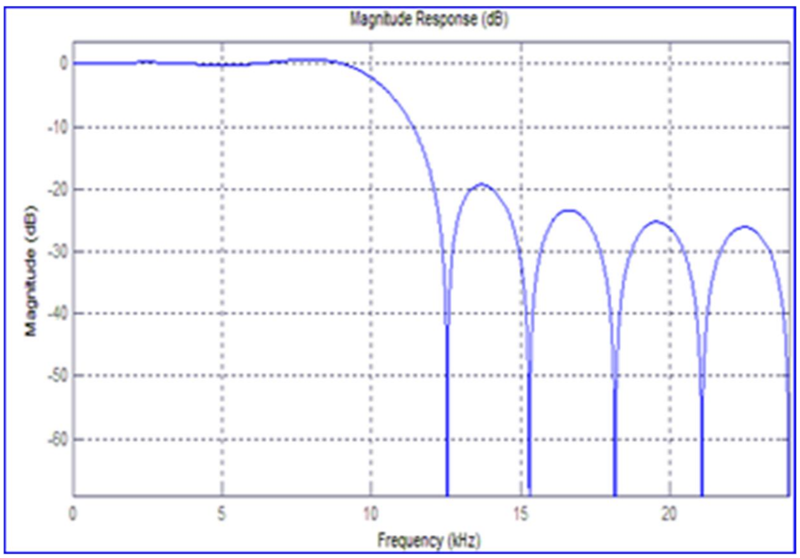

Fig.1 Magnitude response of Rectangular Window

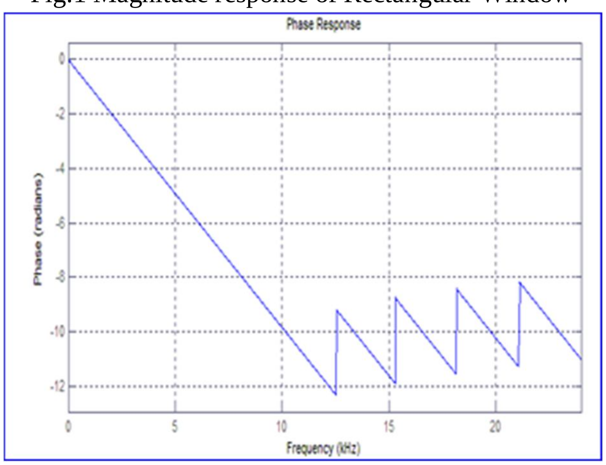

Fig.2 Phase response of Rectangular Window

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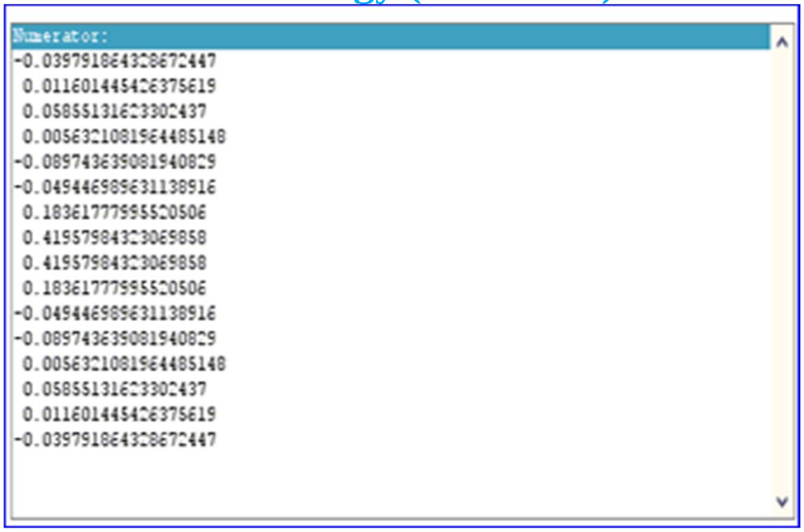
Numerator:
-0.039791864328672447
0.011601445426375619
0.058551316233302437
0.0056321081964485148
-0.089743639081940829
-0.049446989631138916
0.183617779955520506
0.41957984323069858
0.41957984323069858
0.183617779955520506
-0.049446989631138916
-0.089743639081940829
0.0056321081964485148
0.058551316233302437
0.011601445426375619
-0.039791864328672447

Fig.3 Filter coefficients of Recangular window

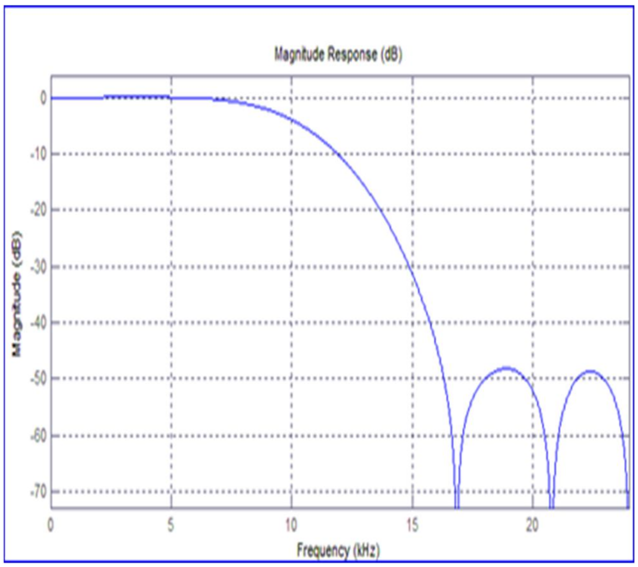

Fig.4 Magnitude response of Hamming Window

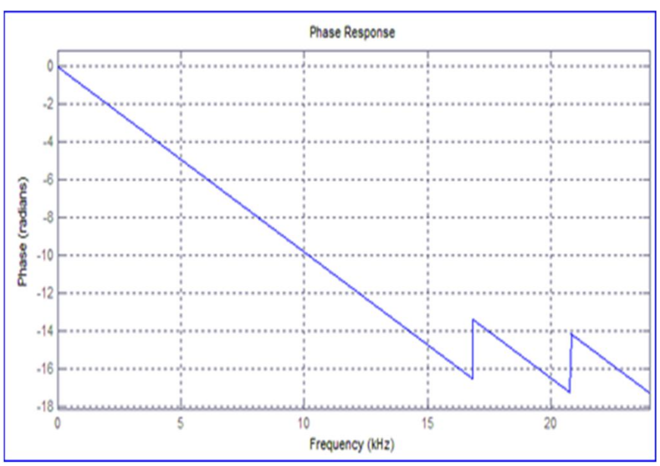

Fig.5 Phase response of Hamming Window

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Numerator:
-0.003145351137666098 4
 0.001372908870445281 8
 0.013433327161586862
 0.002213999894538762
-0.052146748214445551
-0.037619710002305448
 0.16548735595730943
 0.41040421747053674
 0.41040421747053674
 0.16548735595730943
-0.037619710002305448
-0.052146748214445551
 0.002213999894538762
 0.013433327161586862
 0.001372908870445281 8
-0.003145351137666098 4
```

Fig.6: Filter coefficient of Hamming Window

V. CONCLUSION

In this paper, we have encountered with designing low pass FIR filter with a view to comparing their responses for different parameters like magnitude response, phase response, equivalent noise bandwidth, side lobe transition width and response in time and frequency domain all done using MATLAB simulation. In our paper, FIR low pass filter has been designed using Hamming window function and Rectangular window. In frequency resolution problems a small main lobe width of window function in frequency domain is required. FIR filter design by using Hamming window is stable as compare to Rectangular and Rectangular window techniques ripples in pass band are less in hamming as compared to Rectangular window.

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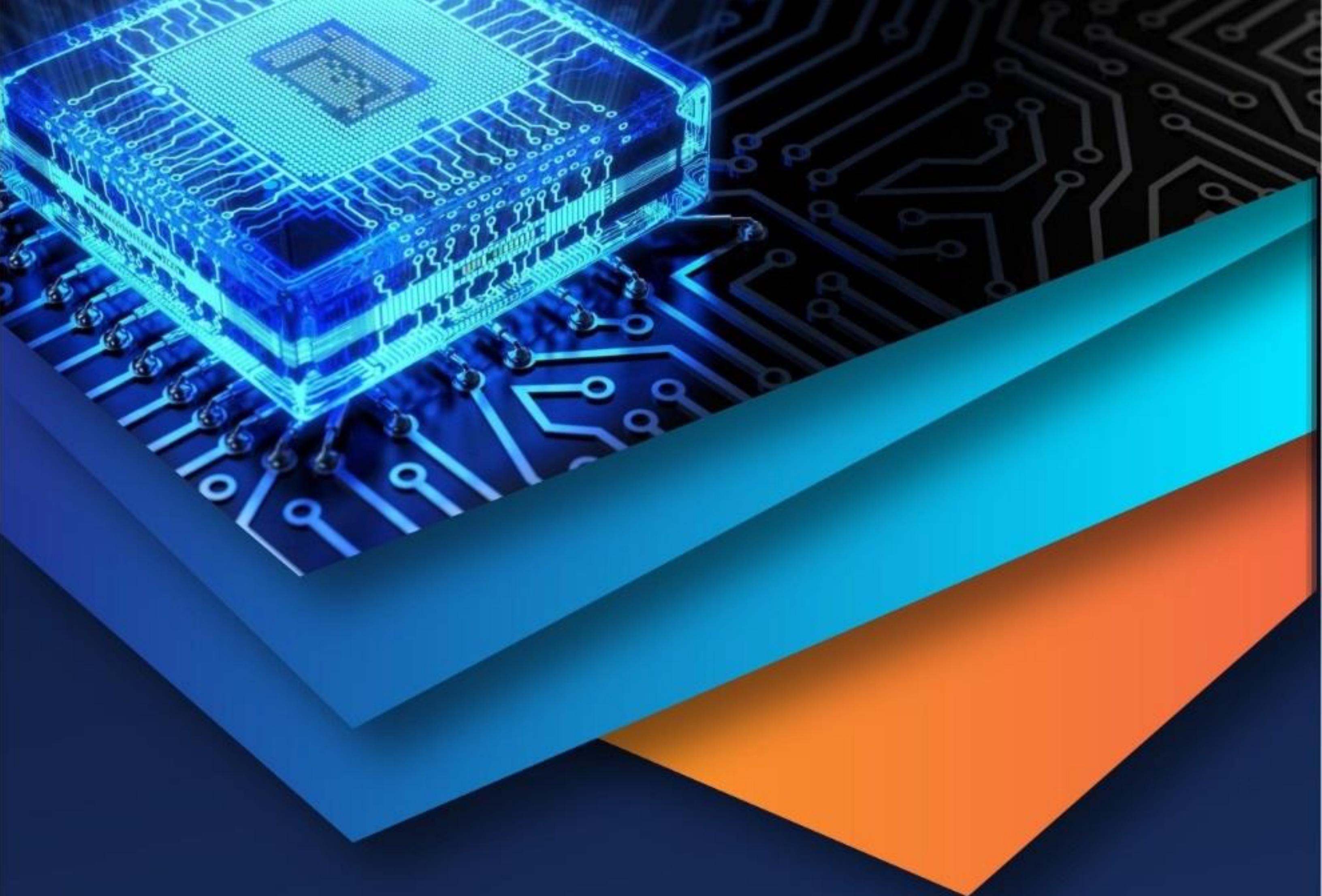

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