iJRASET
International Journal For Research in Applied Science and Engineering Technology



INTERNATIONAL JOURNAL FOR RESEARCH

IN APPLIED SCIENCE & ENGINEERING TECHNOLOGY

Volume: 5 **Issue:** II **Month of publication:** February 2017

DOI: http://doi.org/10.22214/ijraset.2017.2042

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Volume 5 Issue II, February 2017
ISSN: 2321-9653
International Journal for Research in Applied Science & Engineering Technology (IJRASET)

Optimal Design of Low Pass Filter By Parzen, Taylor and Hamming Technique

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Abstract: ***Digital filtering plays a significant role in the world of technology. Therefore, processing of input signal has to be done to get useful signal. This paper deals with the design of finite impulse response digital filter using window techniques. Here various windows are compared and demonstration of the best window is done, which is the one with minimum side lobes. In the processing of Digital Signal Finite Impulse Response (FIR) filter plays an important role. Using MATLAB the FIR filter is designed and simulated. Through this paper the intricacies of the window methods are explained in a simple and a subtle manner.***

Keywords— FIR Filter, DSP, FFT, Hamming Window, MATLAB, Taylor Window, Parzen Window, frequency sampling, amplitude-frequency.

I. INTRODUCTION

Digital signal processing is the technique employed for the mathematical manipulation of an information signal so as to modify or improve it. For this purpose filters are mainly used. Digital filters are used in numerous applications e.g. control system, system for audio and video processing and communication system. Digital filters can be applied to very low frequency signals, such as those occurring in biomedical and seismic applications very efficiently.

These are mainly two types of filter algorithms. They are finite impulse response filter (FIR) and infinite impulse response filter (IIR). In case of a FIR filter, the response due to an impulse input will decay with in a finite time. But for IIR filter, the impulse response never dies out. FIR filters are commonly known as non-recursive filters and IIR filters are known as recursive filters [1]. The advantage of FIR filter over IIR filters are, FIR filter has linear phase and easily to control where as IIR filter has no particular phase and difficult to control. FIR filter is stable and depends only on input. FIR filters consists of only zeroes and IIR filters consists of both poles and zeroes.

A. Types of digital filters

Based on the type of impulse response, a digital filter can be broadly classified into two categories namely FIR and IIR filters. In signal processing, the impulse response, or impulse response function (IRF), of a dynamic system is its output when presented with a brief input signal, called an impulse. More generally, an impulse response refers to the reaction of any dynamic system in response to some external change. A finite impulse response (FIR) filter is a type of digital filter whose impulse response is of finite duration, since the impulse response settles to zero within a finite amount of time. Whereas an IIR filter has impulse response function that is non-zero over an infinite length of time [2]. The impulse response never dies out. FIR filter has a number of useful properties, which gains a lot of preferences over the IIR filter.

B. Fir filters

Finite impulse response (FIR) filter 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. A finite impulse response (FIR) filter is a filter structure that can be used to implement almost any sort of frequency response digitally. An FIR filter is usually implemented by using a series of delays, multipliers and adders to create the filter's output. FIR filters also known as feed forward or non recursive, or transversal filters. The unit impulse response is finite; so FIR filters are stable system [3].

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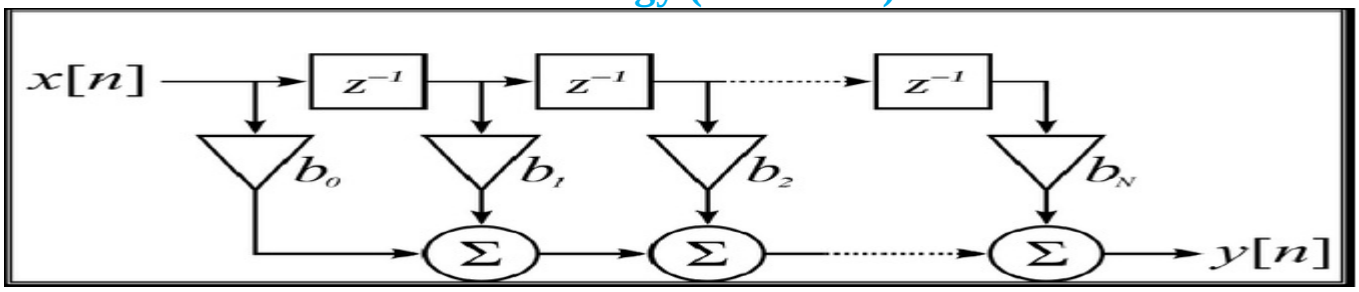


Figure I.II Block Diagram of a FIR Filter.

C. R FILTERS

IIR filters may be implemented as either analog or digital filters. In digital IIR filters, the output feedback is immediately apparent in the equations defining the output. Note that unlike FIR filters, in designing IIR filters it is necessary to carefully consider the "time zero" case in which the outputs of the filter have not yet been clearly defined.

Design of digital IIR filters is heavily dependent on that of their analog counterparts because there are plenty of resources, works and straightforward design methods concerning analog feedback filter design while there are hardly any for digital IIR filters. As a result, usually, when a digital IIR filter is going to be implemented, an analog filter (e.g. Chebyshev filter, Butterworth filter, Elliptic filter) is first designed and then is converted to a digital filter by applying discretization techniques such as Bilinear transform or Impulse invariance [4].

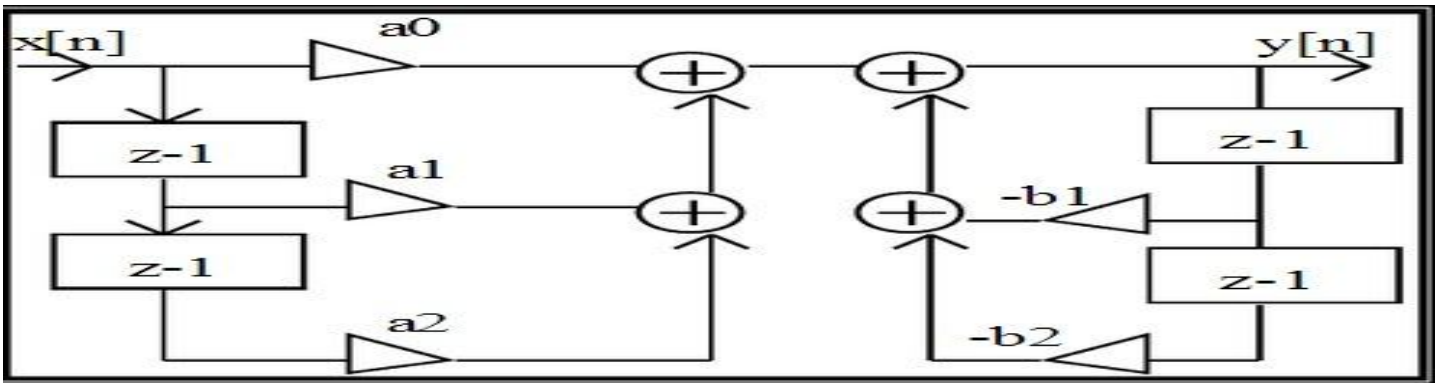


Figure I.III Block Diagram of a IIR Filter.

II. WINDOW FUNCTION

In this method, a certain bandwidth is generated using a truncated ideal low-pass filter, and then we use a selected window to get certain stop-band attenuation. In signal processing, a window function is a mathematical function that is zero-valued outside of some chosen interval. For instance, a function that is constant inside the interval and zero elsewhere is called a rectangular window, which describes the shape of its graphical representation. When another function or waveform/data-sequence is multiplied by a window function, the product is also zero-valued outside the interval: all that is left is the part where they overlap, the “view through the window” [5].

The FIR filter design process using window functions can be enumerated as.

Define filter specifications.

a window function according to the filter specifications.

Compute the filter order required for a given set of specifications.

Compute the window function coefficients.

Compute the ideal filter coefficients according to the filter order.

Compute FIR filter coefficients according to the obtained window function and ideal filter coefficients.

If the resulting filter has a very wide or a very narrow transition region, it is mandatory to change the filter order by decreasing or increasing it according to needs, and after this process the steps 4, 5 and 6 are iterated as many times as needed.

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A. Parzen lowpass fir filter

Parzen windows estimation is one of the classical non- parametric methods in the field of machine learning and pattern classification, and usually uses Gaussian density function as the kernel. Although the relation between the kernel density estimation (KDE) and low-pass filtering is well known, it is very difficult to setting the parameters of the other kinds of density functions. This paper proposes a novel method to deal with the parameters of Laplace kernel through measuring the degree of exchanged information among interpolating points [6]. Experimental results showed that the proposed method can improve the performance of Parzen windows significantly.

z

The parzen window is described as the following equation defines the N–point Parzen window over the interval. $-\frac{(N-1)}{2} \leq n \frac{(N-1)}{2}$

$$\mathbf{W(n)} = \begin{cases} \mathbf{1 - 6(\frac{|n|}{\frac{N}{2}})^2 + 6(\frac{|n|}{\frac{N}{2}})^3 \; 0 \leq |n| \leq \frac{N-1}{4}} \\ \mathbf{2(1 - \frac{|n|}{\frac{N}{2}})^3 \; (N-1)/4 \leq |n| \leq (N-1)/2} \end{cases} \qquad \mathbf{1}$$

B. Taylor lowpass fir filter

Taylor windows are similar to Chebyshev windows. Whereas a Chebyshev window has the narrowest possible mainlobe for a specified sidelobe level, a Taylor window allows you to make tradeoffs between the mainlobe width and the sidelobe level. The Taylor distribution avoids edge discontinuities, so Taylor window sidelobes decrease monotonically [7]. Taylor function are obtained from a weighted – cosine series added to a constant, or a pedestal. A similar form of these function can be obtained by dropping from of some of the higher – order terms. Taylor window coefficients are not normalized. Taylor windows are typically used in radar applications, such as weighting synthetic aperture radar images and antenna design. In Taylor lowpass FIR filter is used for design the FIR filter in windowing technique in matlab [8]. It is same as a parzen lowpass FIR filter but it phase response it is used for analysis of filters and other math works.

Due to the equiripple condition, the time-domain window has discontinuities at the edges. An approximation that avoids them, by allowing the equiripples to drop off at the edges, is a Taylor window. This window has a greater amplitude level. Taylor windows are widely used in radar applications, such as antenna designing [9].

C. Hamming lowpass fir filter

Hamming window has a bell-like shape. Its first and last samples are not zero. The window is optimized to minimize the maximum side lobe. To eliminate the some pass band and stop band ripples, Hamming window technique is used. The coefficients of a Hamming window are computed from the equation.

$$w[n] = 0.54 - 0.46\cos(2\pi)\, x \frac{n}{N}, \; 0 \leq n \leq N \qquad 2$$

Richard W. Hamming observed that the sidelobes of the rectangular and Hanning windows are phase reversed relative to each other, so a linear combination of the two would tend to cause them to cancel each other. He searched for the linear combination that minimized the maximum sidelobe amplitude and came up with the following formulation, which represents [10]. a raised cosine on a rectangular pedestal.

III. DESIGN SIMULATION

To design the low pass FIR filter using MATLAB Parzon, Taylor and Hamming Window the parameter specifications are given in table 1. As

Table (1). Parameter specification.

PARAMETR	VALUE
Sampling frequency(fs)	48000 Hz
Cutt off frequency(fc)	10800 Hz
Order (N)	20

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Volume 5 Issue II, February 2017
ISSN: 2321-9653
International Journal for Research in Applied Science & Engineering Technology (IJRASET)

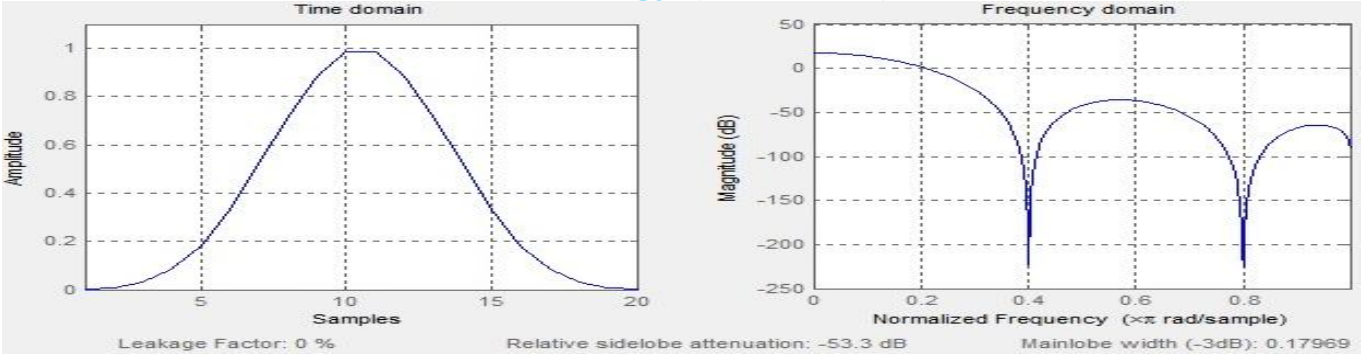


Figure (1): Time domain and Frequency domain of Parzon window.

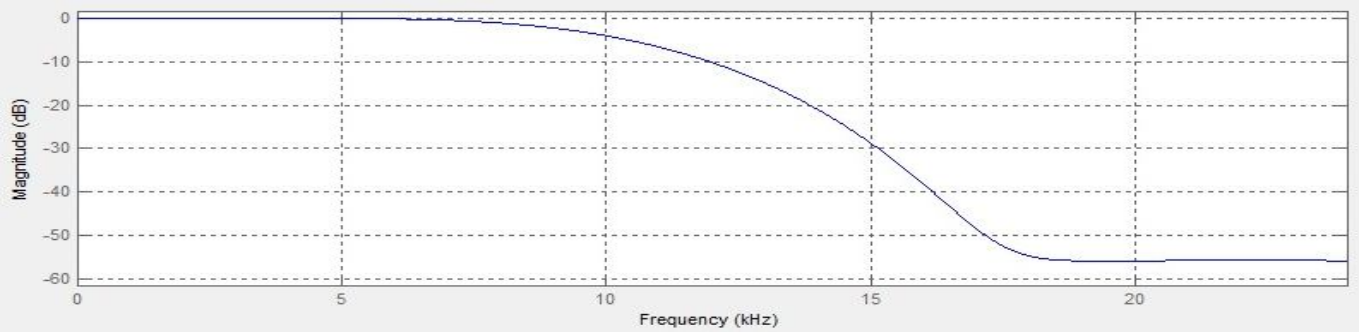


Figure (2): Magnitude response of Parzen window technique.

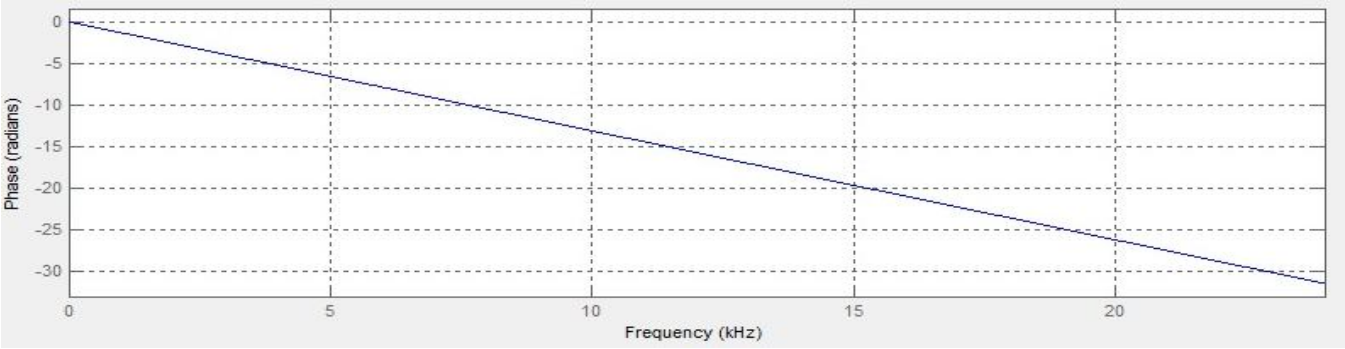


Figure (3): Phase response of Parzen window technique.

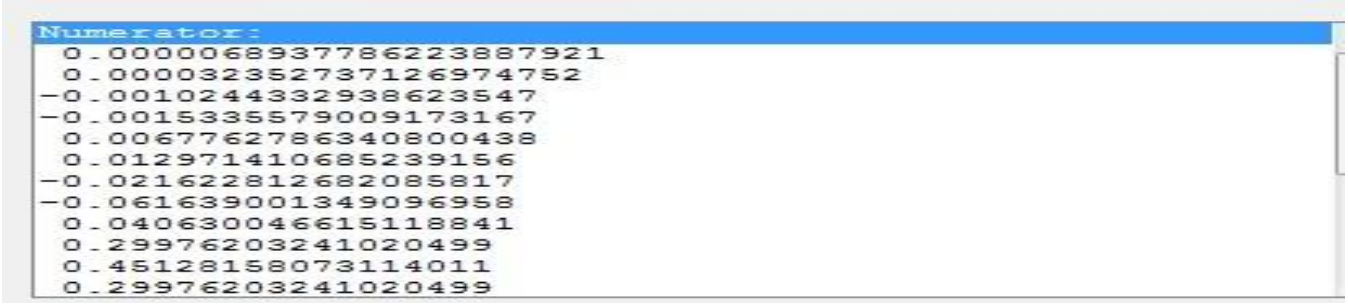


Figure (4): Filter coefficient of Parzen window technique.

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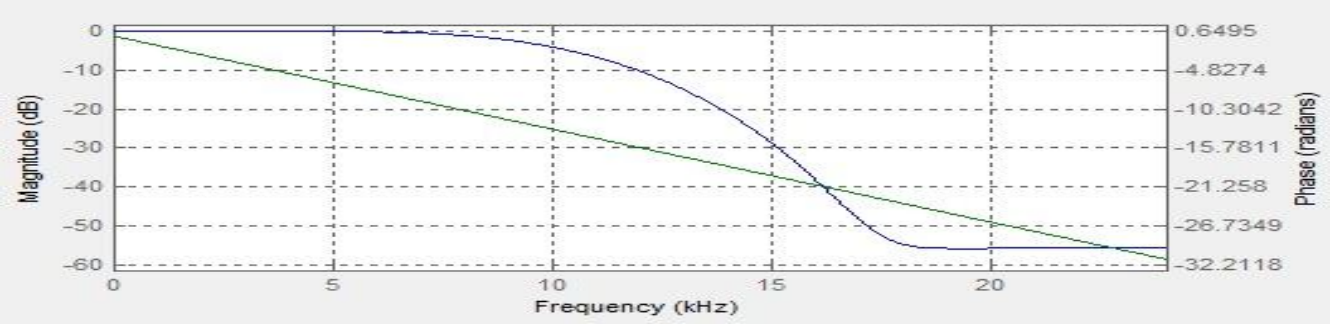


Figure (5): Magnitude and Phase response of Parzen window technique.

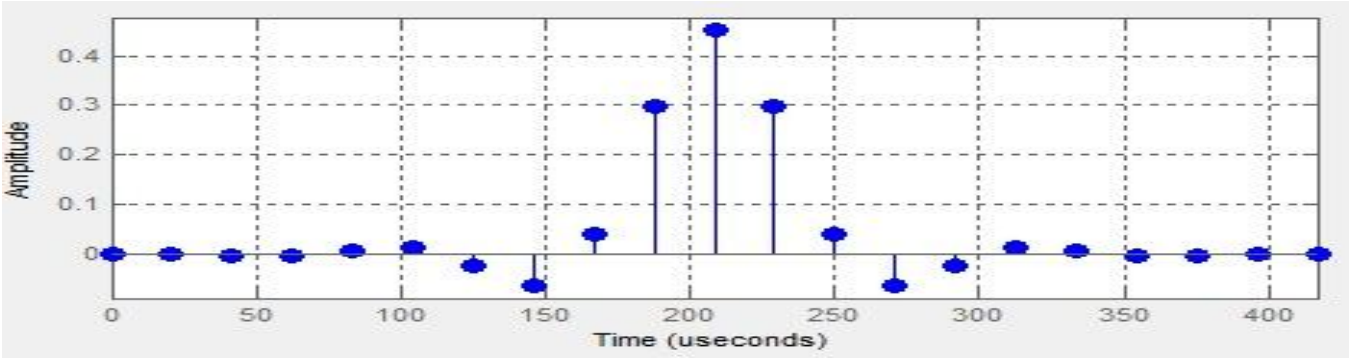


Figure (6): Impulse Response of Parzen window technique

Figure (1): Time domain and Frequency domain of Parzon window. In which the relative sidelobe attenuation is -53.3 dB and main lobe with (-3dB) is 0.17969. Figure (2): Magnitude response of Parzen window technique. Figure (3): Phase response of Parzen window technique. Figure (4): Filter coefficient of Parzen window technique. Figure (5): Magnitude and Phase response of Parzen window technique. Figure (6): Impulse Response of Parzen window technique

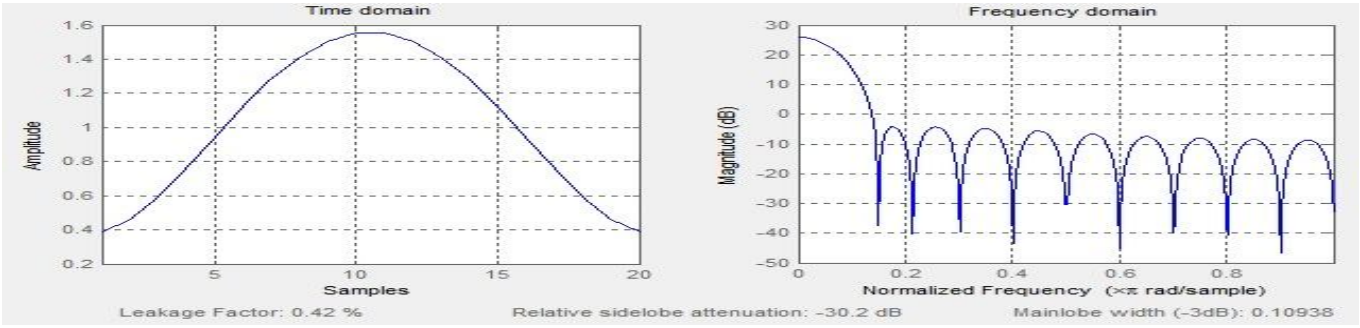


Figure (7): Time domain and Frequency domain of Taylor window.

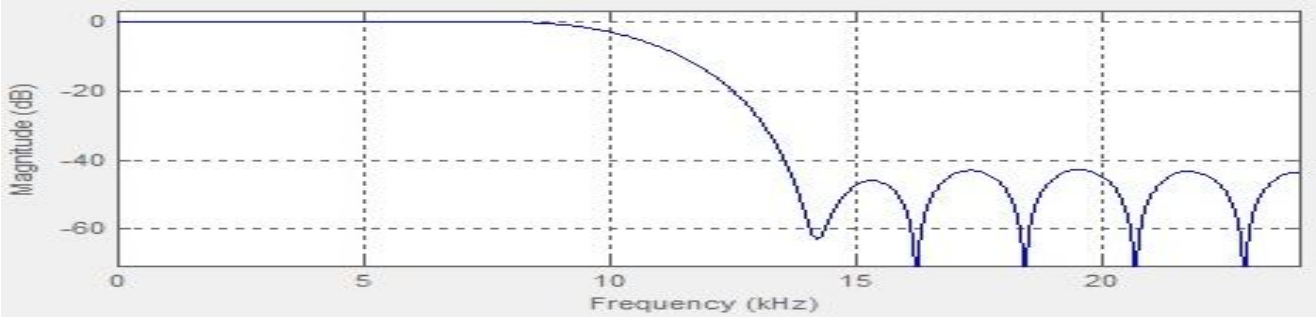


Figure (8): Magnitude response of Taylor window technique.

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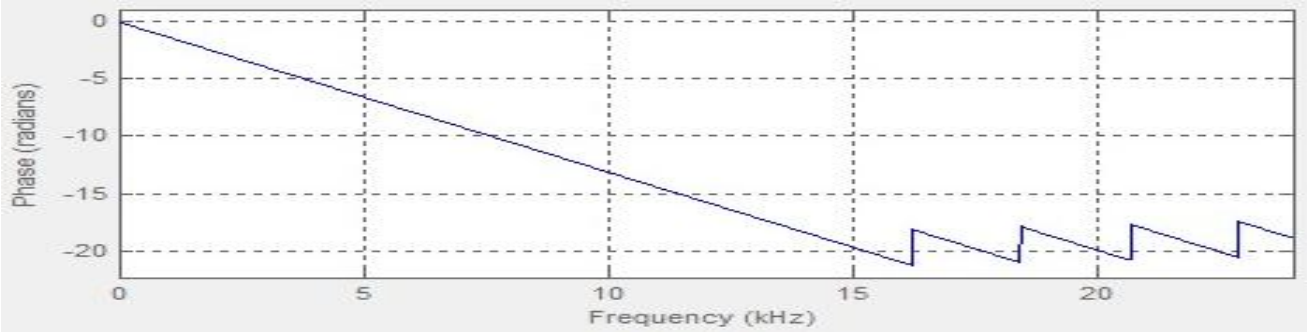


Figure (9): Phase response of Taylor window technique.

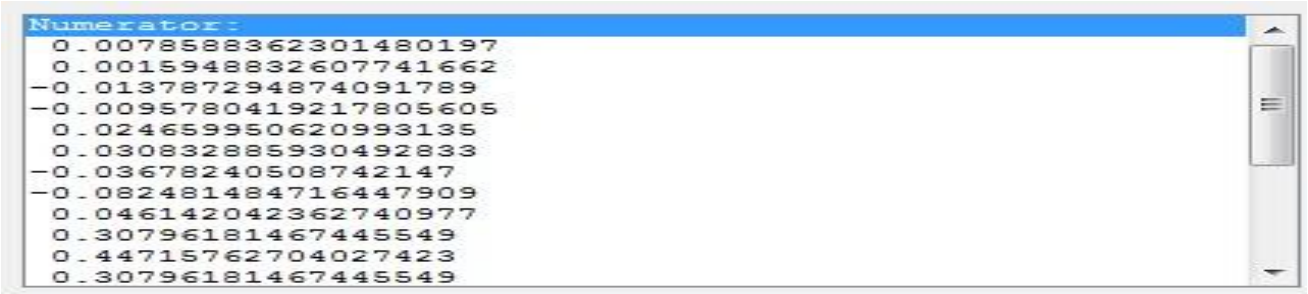


Figure (10): Filter coefficient of Taylor window technique.

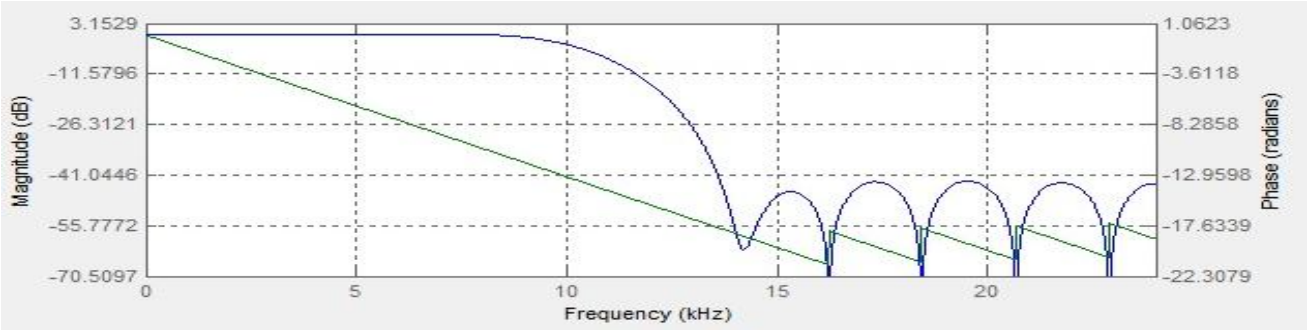


Figure (11): Magnitude and Phase response of Taylor window technique.

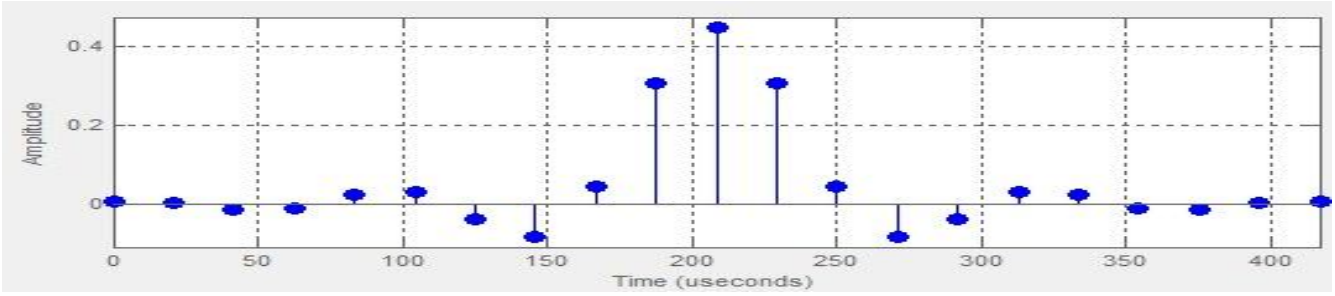


Figure (12): Impuls Response of Taylor window technique.

Figure (7): Time domain and Frequency domain of Taylor window. In which the relative sidelobe attenuation is -30.2 dB and main lobe with (-3dB) is 0.10938. Figure (8): Magnitude response of Taylor window technique. Figure (9): Phase response of Taylor window technique. Figure (10): Filter coefficient of Taylor window technique. Figure (11): Magnitude and Phase response of Taylor window technique. Figure (12): Impulse Response of Taylor window technique.

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ISSN: 2321-9653

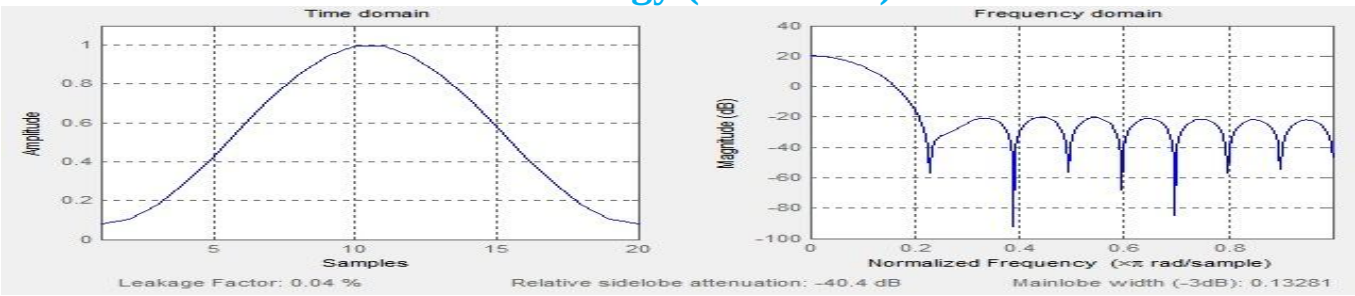


Figure (13): Time domain and Frequency domain of Hamming window.

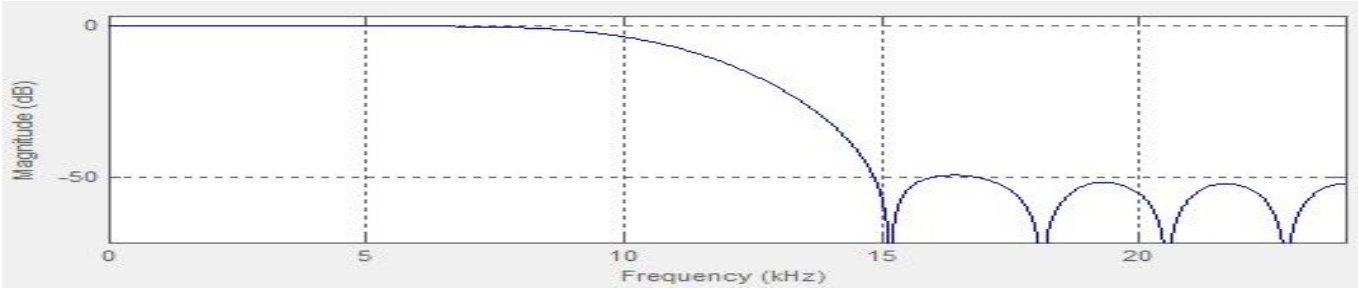


Figure (14): Magnitude response of Hamming window technique.

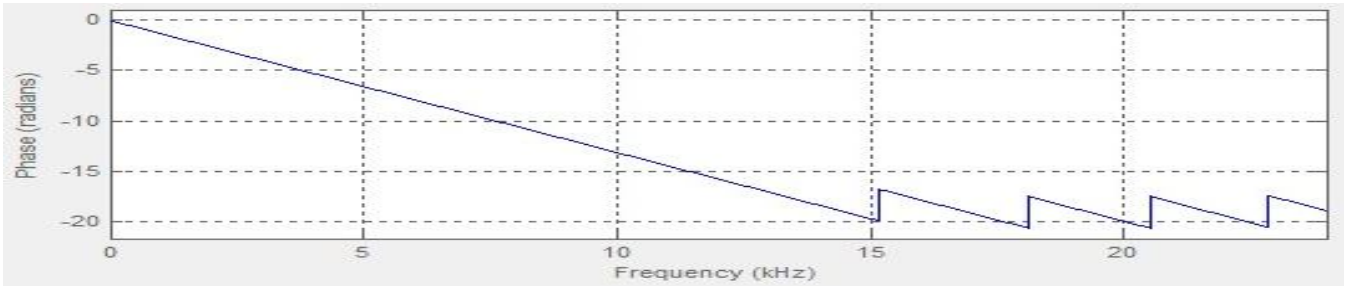


Figure (15): Phase respnse of Hamming window technique.



Figure (16): Filter coefficient of Hamming window technique

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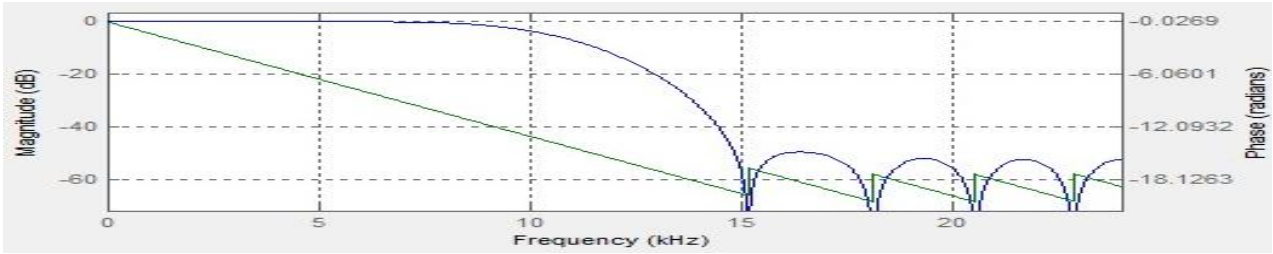


Figure (17): Magnitude and Phase response of Hamming window technique.

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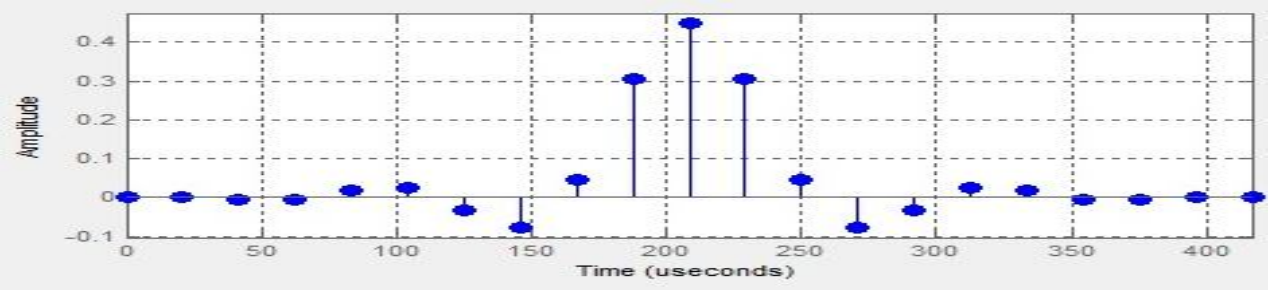


Figure (18): Impulse Response of Hamming window technique.

Figure (13): Time domain and Frequency domain of Hamming window. In which the relative sidelobe attenuation is -40.4 dB and main lobe with (-3dB) is 0.13281. Figure (14): Magnitude response of Hamming window technique. Figure (15): Phase response of Hamming window technique. Figure (16): Filter coefficient of Hamming window technique. Figure (17): Magnitude and Phase response of Hammingwindow technique. Figure (18): Impulse Response of Hamming window technique.

A. Comparative analysis

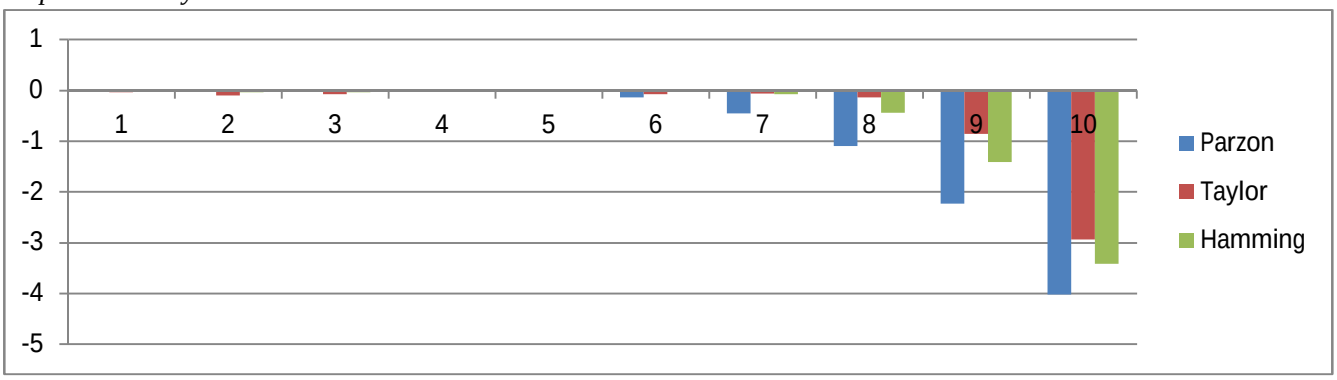


Figure (19): Magnitude and Frequency plot of Parzon, Taylor and Hamming window technique.

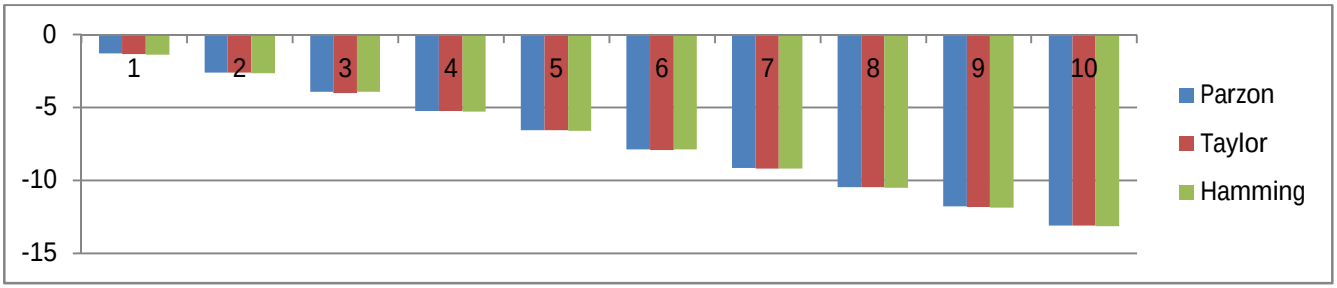


Figure (20): Phase and Frequency plot of Parzon, Taylor and Hamming window technique.

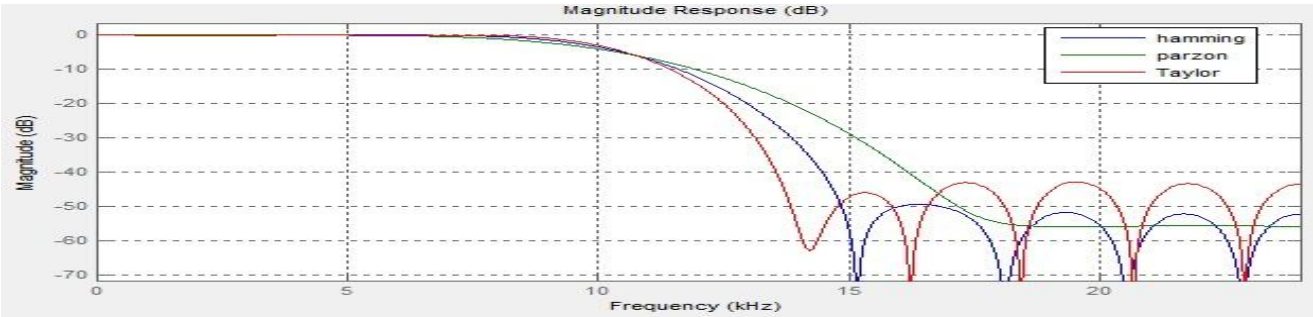


Figure (21): Magnitude comparison of Hamming, Parzon and Taylor window.

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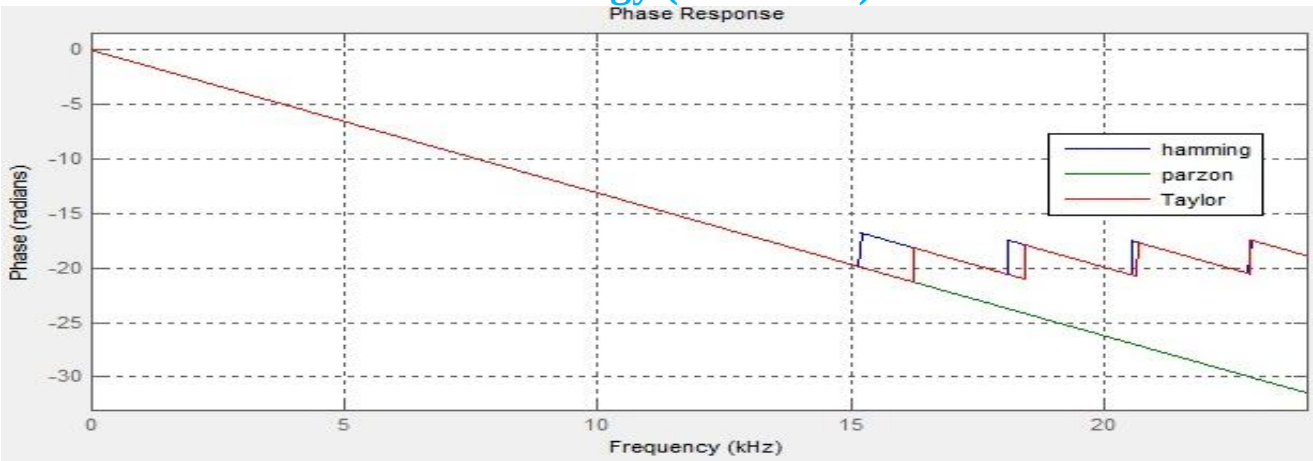


Figure (22): Phase comparison of Hamming, Parzon and Taylor window.

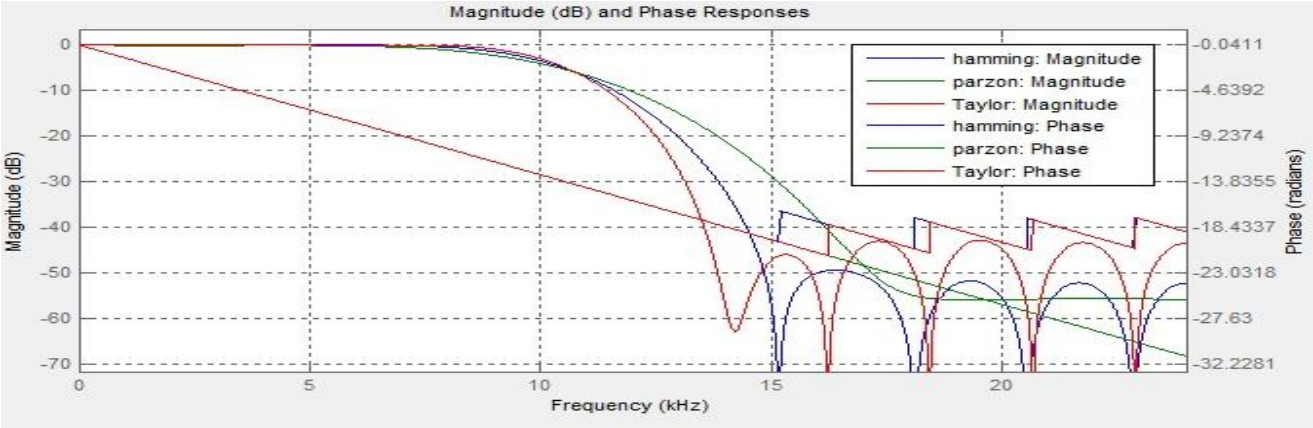


Figure (23): Magnitude and Phase comparison of Hamming, Parzon and Taylor window.

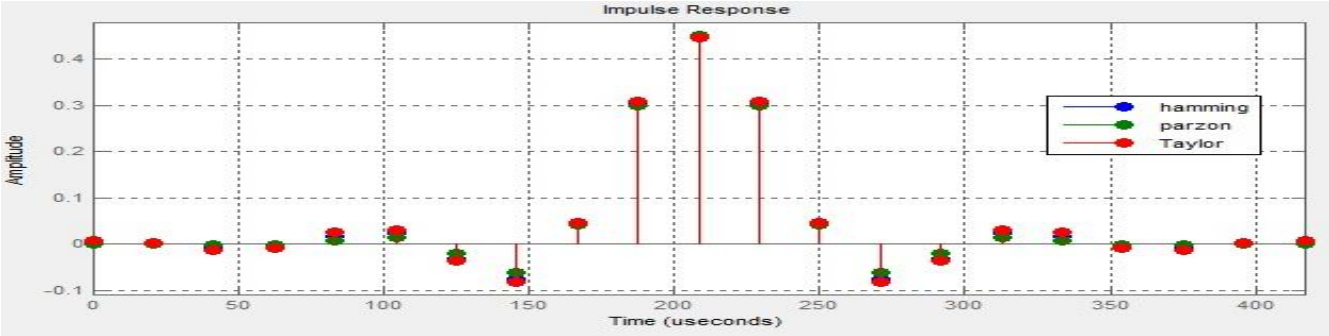


Figure (24): Impulse Resoponse comparison of Hamming, Parzon and Taylor window.

By using MATLAB Parzon Taylor and Hamming windows techniques are design, analysis, Figure (19): Magnitude and Frequency plot of Parzon, Taylor and Hamming window technique. Figure (20): Phase and Frequency plot of Parzon, Taylor and Hamming window technique. Figure (21): Magnitude comparison of Hamming, Parzon and Taylor window. Figure (22): Phase comparison of Hamming, Parzon and Taylor window. Figure (23): Magnitude and Phase comparison of Hamming, Parzon and Taylor window. Figure (24): Impulse Resoponse comparison of Hamming, Parzon and Taylor window. Hamming, Parzon and Taylor Windows techniques are used for design analysis and compared these three windows. Table 2 shows the comparison of these three windows in terms of leakage factor, relative side lobe attenuation and main lobe width. The leakage factor is minimum and main lobe width is maximum. When beta is increase then main lobe width is increase but leakage factor is decrease.

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Volume 5 Issue II, February 2017
ISSN: 2321-9653
International Journal for Research in Applied Science & Engineering Technology (IJRASET)

IV. RESULT

Table (2). Simulation Result in MATLAB

Window Technique	Leakage Factor	Relative sidelobe attenuation	Mainlobe width (-3dB)
Parzen	0	-53.3 dB	017969
Taylor	0.42	-30.2 dB	0.10938
Hamming	0.04	-40.4 dB	0.13281

Table (3). Magnitude, Phase and Frequency Result of Parzen, Taylor and Hamming window technique.

Frequency	Windows Technique					
kHz	Parzen Magnitude	Parzon Phase	Taylor Magnitude	Taylor Phase	Hamming Magnitude	Hamming Phase
1	0.001108	-1.311554	-0.048755	-1.342233	-0.016868	-1.395923
2	0.003765	-2.626942	-0.110196	-2.626942	-0.041288	-2.676796
3	0.006552	-3.938496	-0.077552	-4.030535	-0.036787	-3.953833
4	0.003396	-5.238544	-0.008619	-5.269224	-0.009788	-5.311408
5	-0.028772	-6.553933	-0.023066	-6.553933	0.008615	-6.592282
6	-0.149103	-7.873156	-0.083954	-7.911506	0.001193	-7.865486
7	-0.463997	-9.165535	-0.069438	-9.196215	-0.084805	-9.180875
8	-1.101295	-10.48092	-0.140856	-10.48859	-0.442282	-10.5116
9	-2.23464	-11.80398	-0.862351	-11.8549	-1.419779	-11.86151
10	-4.02001	-13.09636	-2.937813	-13.10787	-3.421454	-13.15772

Interpretation of above Table*:* Table 1 show parameter specification of windows designing of low pass filter using hamming, Parzon and Taylor Window. Sampling frequency is 48000 Hz and cut off frequency is 10800 Hz.Filter order is 20. Table (2). Simulation Result in MATLAB. Table (3). Magnitude, Phase and Frequency Result of Parzen, Taylor and Hamming window technique.
In Taylor Window beta value is increase then main lobe width is increases but leakage factor is decreases. When leakage factor is 0.42% in Taylor Window then wider main lobe width (0.10938).

V. DISCUSSIONS

The low pass FIR filter design with sampling frequency =48000 Hz, cut off frequency =10800 Hz, based on Taylor window design and specify order is 50. The design of FIR low pass filter based on adjustable window design method implementing by using filter design and analysis tool (FDATool) from MATLAB (R2011a) programs.
Leakage factor is equal to ratio of power in side lobes to total window power. Side lobe attenuation is equal to difference in height from main lobe peak to the highest side lobe peak. Main lobe width (-3 dB) is equal to the width of main lobe at 3 dB below the main lobe peak.

VI. CONCLUSIONS

In this paper FIR low pass filter has been designed and simulated using Parzon, Taylor and hamming Window technique. It has been compared leakage factor, main lobe width and relative side lobe attenuation of the three windows from the simulated result.Taylor Window main lobe width is increase and decreases the amplitude of side lobes that is increases the attenuation. Taylor window main lobe width (-3dB) is 0.10938 at sampling frequency 48000 Hz, cut off frequency 10800 Hz and order 20. Taqylor window has greater main lobe width and less leakage factor in comparison of Hamming window and Parzon window.

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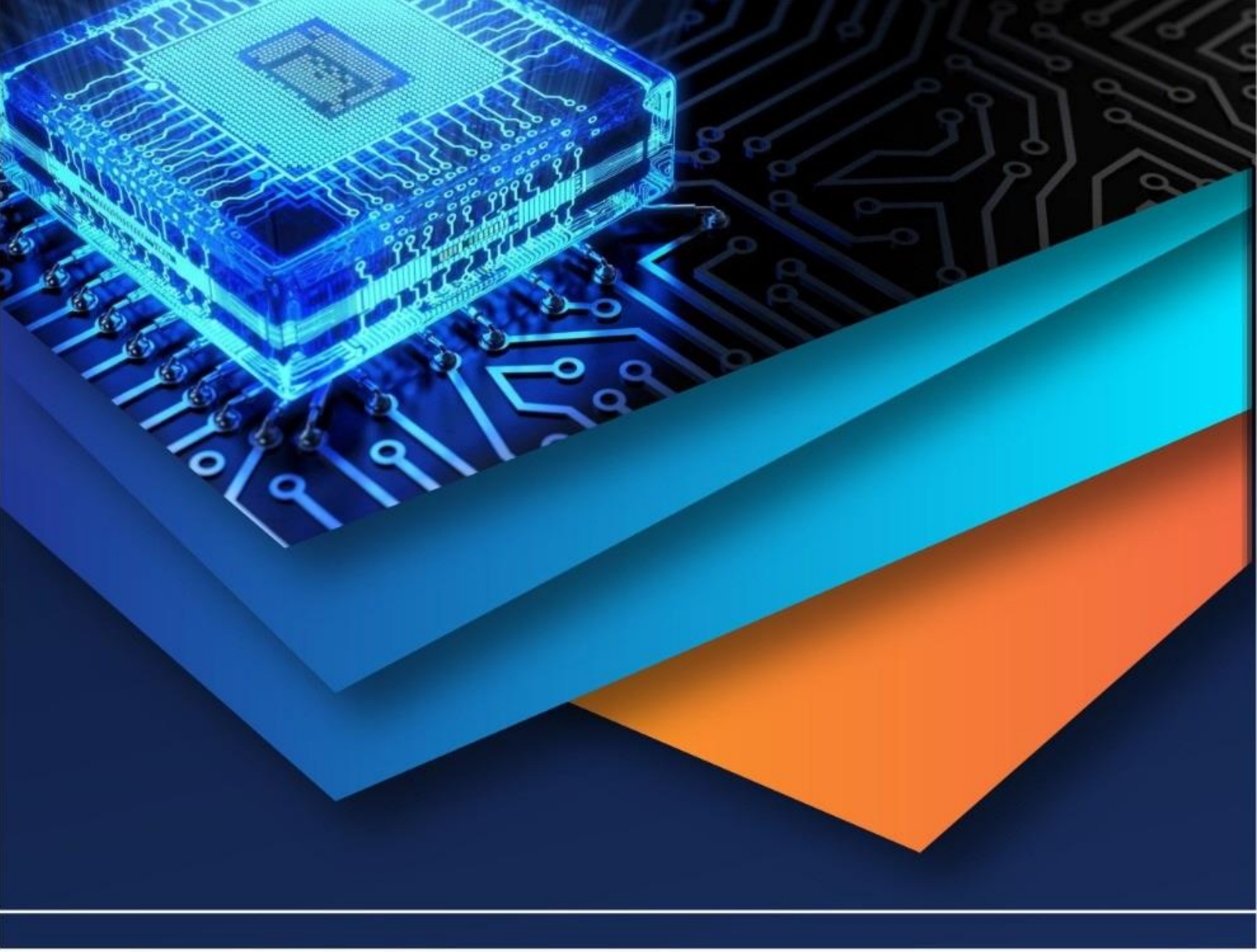


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