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RESEARCH ARTICLE

OPTIMAL DESIGN OF HIGH-PASS FIR FILTER BY BLACKMAN, RECTANGULAR, TRINGULAR AND TAYLOR WINDOW TECHNIQUES

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ABSTRACT

This paper deals with the design of FIR filter using Blackman window, Rectangular window, Tringular window, Taylor window Techniques of order 10. In this paper. The analysis of magnitude and phase response of proposed FIR Low-pass filter are performed using MATLAB simulation. The result window technique provides better result in term of magnitude and phase response of High-pass FIR filter.

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INTRODUCTION

We are living in an age of information technology. Most of this technology is based on theory of digital signal processing (DSP) and implementation of the theory by devices embedded in what are known as digital signal processors (DSPs). Of course, the theory of DSP and its applications is supported by other disciplines such as computer science and engineering, and advances in technologies such as the design and manufacturing of very large scale integration (VLSI) chips (Shenoi, 2006). A signal carries information, and the objective of signal processing is to extract useful information carried by the signal. The method of information extraction depends on the type of signal and the nature of the information being carried by the signal (Sanjit K. Mitra, 2001). 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. There is always a driving need to make thing better and DSP provides many techniques for doing this. For example, people enjoy music and to download new songs. However, with slow Internet connection speeds (typically 56 kilobits per second for a dial-up modem), downloading a song could take hours. WithMP3 compression software, though, the size of the song is reduced by as much as 90%, and can be downloaded in a matter of minutes. The MP3 version of the song is not the same as the original, but is a "good enough" approximation that most users cannot distinguish from the original. (Michael Weeks, 2007)

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Advantages of DSP

Digital signal processing has following advantages:

1) Digital signal processing operations can be changed by changing the program in digital programmable system, *i.e.*, these is flexible systems.
2) Better control of accuracy in digital systems compared to analog systems.
3) Digital signals are easily stored on magnetic media such as magnetic tape without loss of quality of reproduction of signal.
4) Digital signals can be processed off line, *i.e.*, these are easily transported.
5) Sophisticated signal processing algorithms can be implemented by DSP method.
6) Digital circuits are less sensitive to tolerances of component values.
7) Digital systems are independent of temperature, ageing and other external parameters.
8) Digital circuits can be reproduced easily in large quantities at comparatively lower cost.
9) Cost of processing per signal in DSP is reduced by time-sharing of given processor among a number of signals.

10) Processor characteristics during processing, as in adaptive filters can be easily adjusted in digital implementation.
11) Digital system can be cascaded without any loading problems.

Disadvantages of DSP

1) Limited to signal with relatively LOW BANDWITH.
2) Flexible -> Easily Modified (Software Base).
3) Handle more complex processing that impossible with analog circuitry.

Applications of DSP in area-wise are as following

1) Telecommunication- Echo cancellation in telephone networks, equalization, telephone dialling application, modems, line repeaters, channel multiplexing, data encryption, video conferencing, cellular phone and FAX.
2) Military- Radar signal processing, sonar signal processing, navigation, secure communications and missile guidance.
3) Consumer electronics- Digital Audio/TV, electronic music synthesizer, educational toys, FM stereo application and sound recording applications.
4) Image processing- Image representation, image compression, image enhancement, image restoration and image analysis compression, image enhancement, image restoration and image analysis.
5) Speech processing- Speech analysis methods are used in automatic speech recognition, speaker verification and speaker identification.
6) Medicine-Medical diagnostic instrumentation such as computerized tomography (CT), X-ray scanning, Patient monitoring and X-ray storage/enhancement.
7) Signal filtering-Removing of unwanted background noise, removal of interference, separation of frequency bands and shaping of the signal spectrum. (Salivahanan *et al*., 2001)

FIR filter

FIR filters are filters having a transfer function of a polynomial in z- 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

$$\mathbf{H}\,(\mathbf{z}) = \sum_{n=0}^{N-1} h(n) Z^{-n} \qquad \text{................ (1)}$$

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

(1) approximation problem
(2) realization problem

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

(1) A desired or ideal response is chosen, usually in the frequency domain.
(2) An allowed class of filters is chosen (e.g. the length N for a FIR filters).
(3) A measure of the quality of approximation is chosen.
(4) 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 methods (Salivahanan *et al*., 2001).

Window techniques

The desired frequency response of any digital filter is periodic in frequency and can be expanded in a fourier series, i.e

$$\mathbf{H_d(e^{j\omega}) = \sum_{n=-\infty}^{\infty} h_d\,(n) e^{-j\omega}} \qquad \text{…………….(2)}$$

$$where,\ h(n) = \frac{1}{2\pi}\int_0^{2\pi} H(e^{j\omega}) e^{j\omega} d\omega$$

The Fourier coefficient of the series h(n) are identical to the impulse response of a digital filter. There are two difficulties with the implementation of above equation for designing a digital filter. First, the impulse response is of infinite duration and second, the filter is non-causal and unrealizable. No finite amount of delay can make the impulse response realizable Hence the filter resulting from a Fourier series representation. of H($e^{j\omega}$) is an unrealizable IIR filter (Pranay rahi *et al*., 2016).

Blackman window function

The Blackman window is defined as

$$\mathrm{Wb(n)} = \begin{cases} 0.42 + 0.5\cos\left(\frac{2\pi n}{M}\right) + 0.08\cos\left(\frac{4\pi n}{M}\right), & \mathrm{for}|n| \le \frac{M}{2} \\ 0, & \mathrm{for}|n| > \frac{M}{2} \end{cases} \qquad \text{………………..(3)}$$

Rectangular window function

The weighting function for the rectangular window is given by

$$\mathrm{WR(n)} = \begin{cases} 1 for|n| \le \frac{m-1}{2} \\ 0, \mathrm{otherwise} \end{cases} \qquad \text{…………….(4)}$$

Taylor window function

A Taylor window (or a set of Taylor weights) are very similar to Chebychev weights. But while Chebychev weights provide the tigh test beam width for a given sidelobe level, Taylor weights provide the least taper loss for a given sidelobe level. The Taylor distribution avoids the edge discontinuties that can occur with Chebychev weights and the resulting sidelobes decay monotonically rather than hold constant like the Chebychev result. For these reasons they are commonly used in antenna and phased array synthesis (Mohd. Schariq Mahboob and Rajesh Mehra, 2014).

Tringular window function

The standard triangular window is defined as

$$w_t(n) = \begin{cases} \frac{-2[n]}{M+2}, for\lfloor n\rfloor \leq \frac{M}{2} \\ 0, \quad for|n| > \frac{M}{2} \end{cases} \quad \ldots\ldots\ldots\ldots\ldots\ldots (5)$$

Simulation

Table 1. Parameter Specification

Parameter	Values
Sampling Frequency(Fs)	48000 Hz
Cut off Frequency(Fc)	10800Hz
Order(N)	10

Table 2. Filter coefficients of Rectangular, Blackman and tringular Window Technique

Filter Coefficient h(n)	Window Techniques			
	Rectangular	Blackman	Tringular	Taylor
h(0)=h(10)	-0.04187	0	-0.00781	-0.01166
h(1) =h(9)	0.04351	0.00188	0.01624	0.01861
h(2)= h(8)	0.08794	0.01898	0.04923	0.05681
h(3)= h(7)	-0.04575	0.02508	-0.03415	-0.03924
h(4)= h(6)	-0.29247	0.26708	-0.27289	-0.29447
h(5)	0.51166	0.55019	0.57288	0.54282

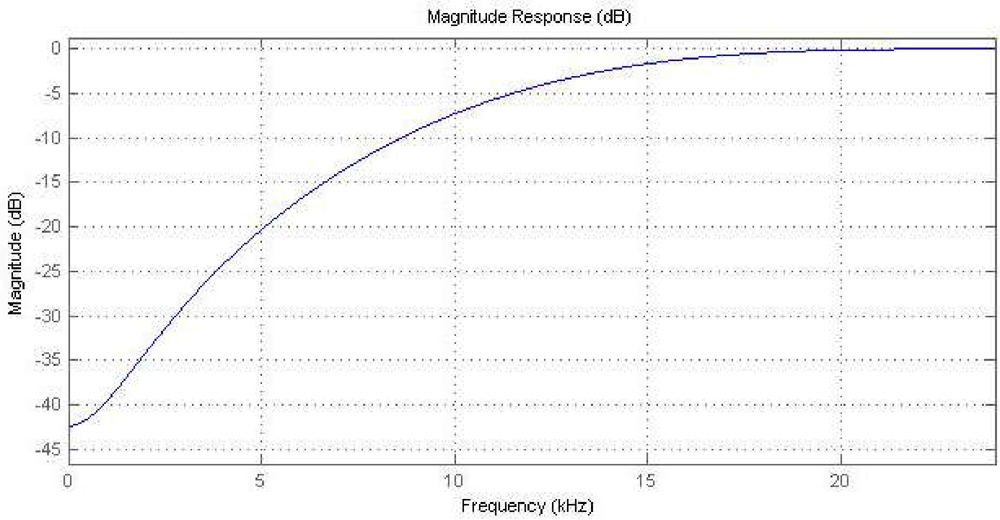


Figure 1. Magnitute Blackman window technique

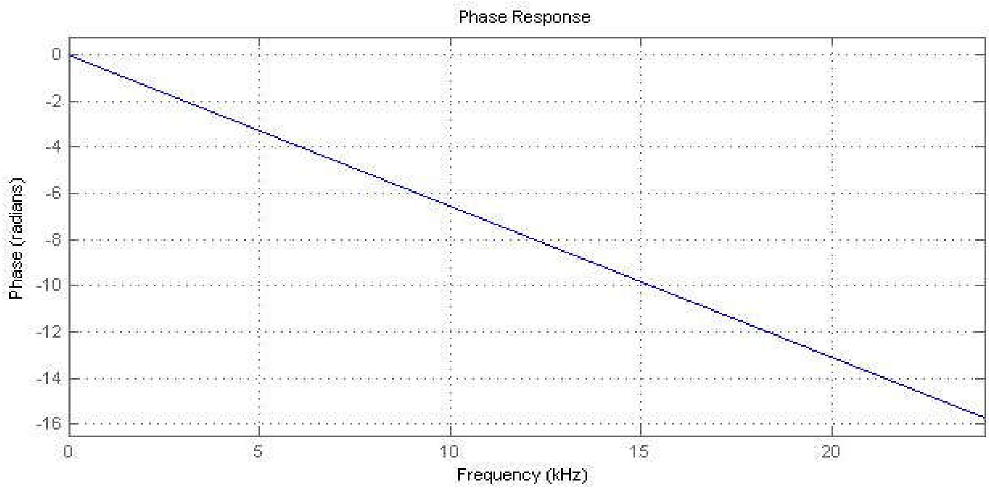


Figure 2. Phase response of Blackman window technique

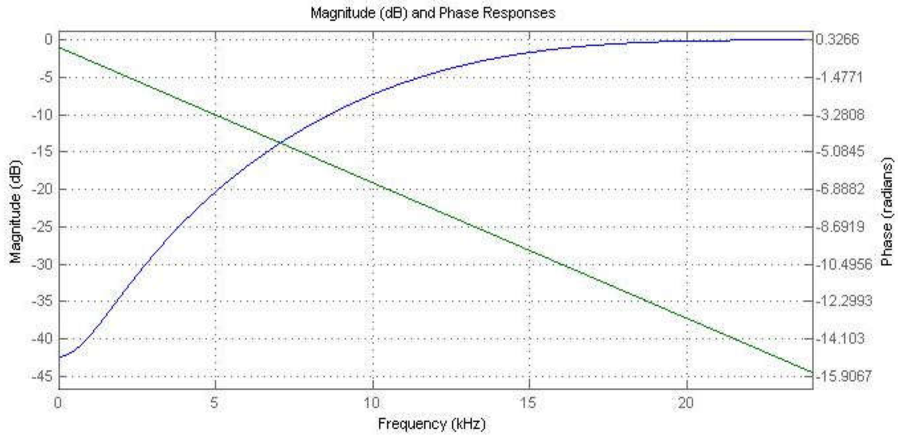


Figure 3. Magnitute and Phase response Blackman window technique

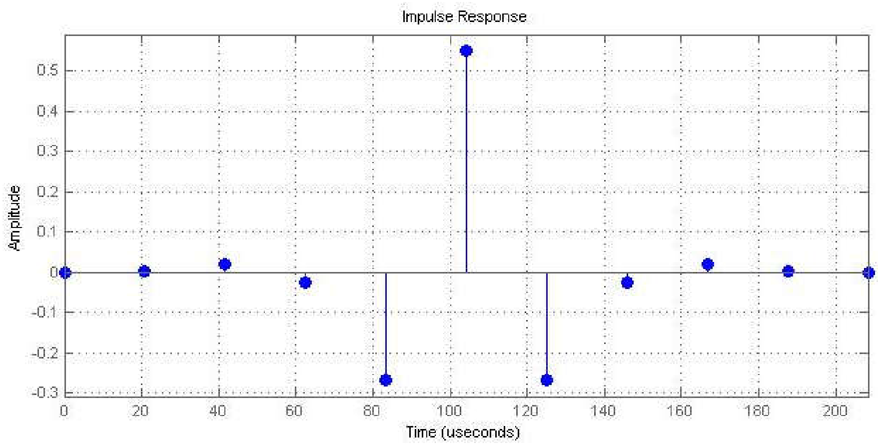


Figure 4. Impulse response of Blackman Window technique

```
Numerator:
 0
 0.0018816183597687257
 0.018987448935354003
-0.025081245705800861
-0.26708717967065182
 0.55019979322146861
-0.26708717967065182
-0.025081245705800861
 0.018987448935354003
 0.0018816183597687257
 0
```

Figure 5. Filter coefficient of Blackman window technique

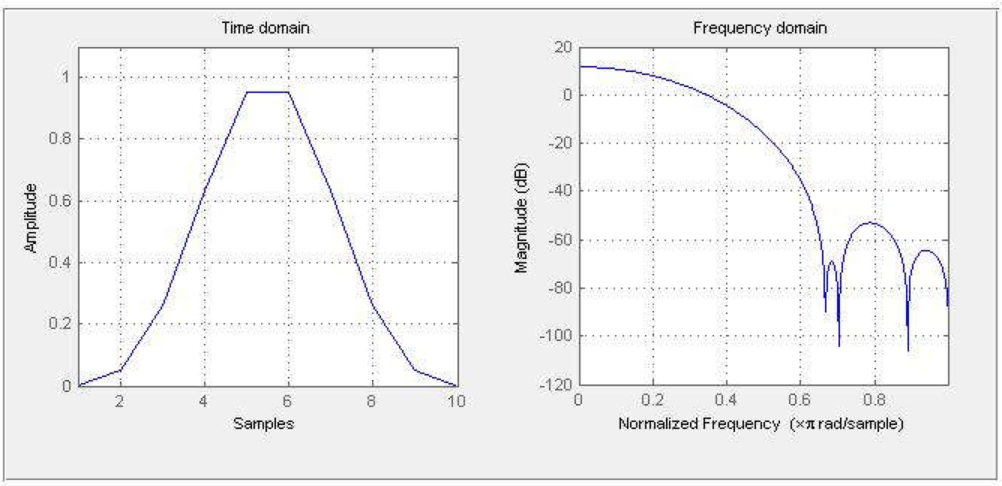


Figure 6. Time doman and Frequency doman of Blackman window technique

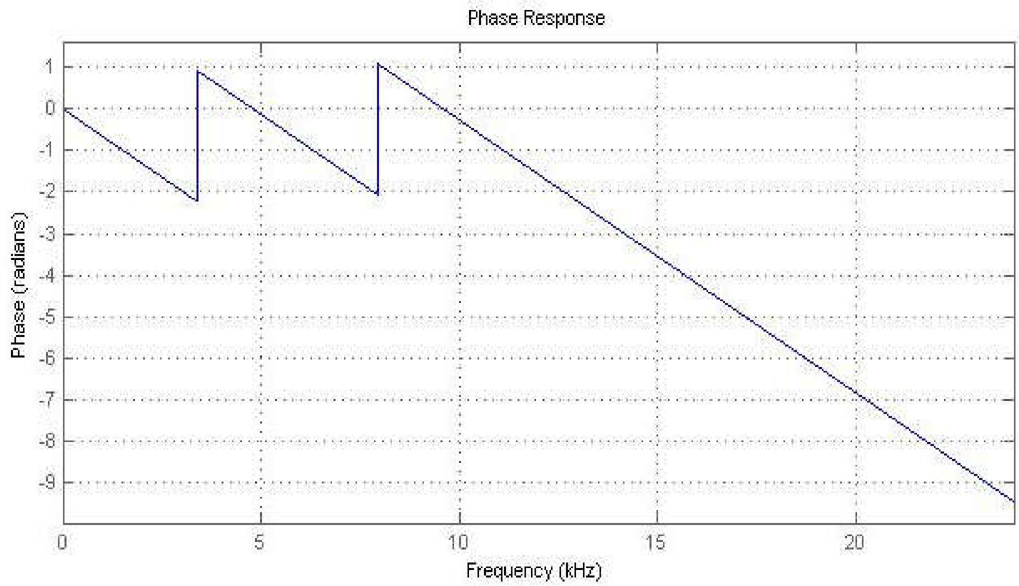


Figure 7. Phase response of Rectangular window Technique

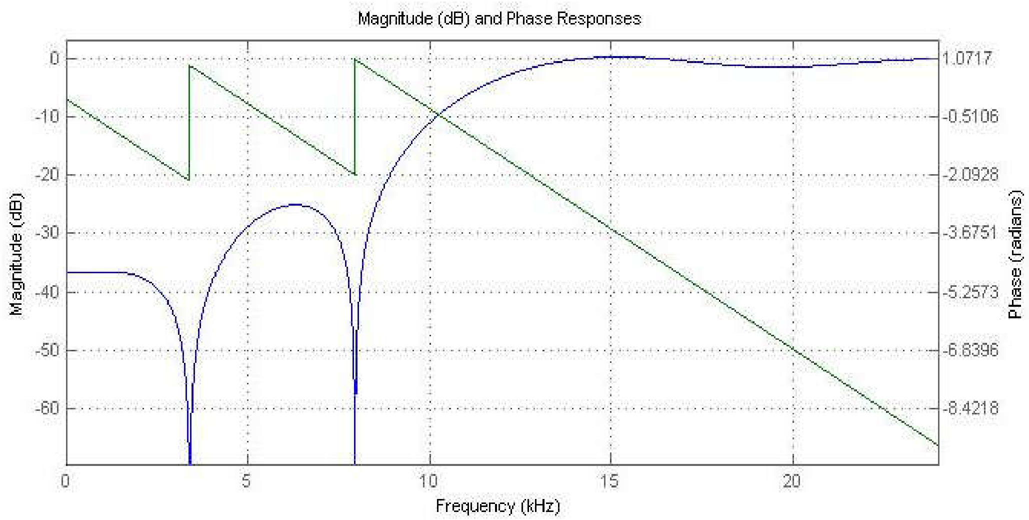


Figure 8. Magnitude and Phase response of Rectangular window Technique

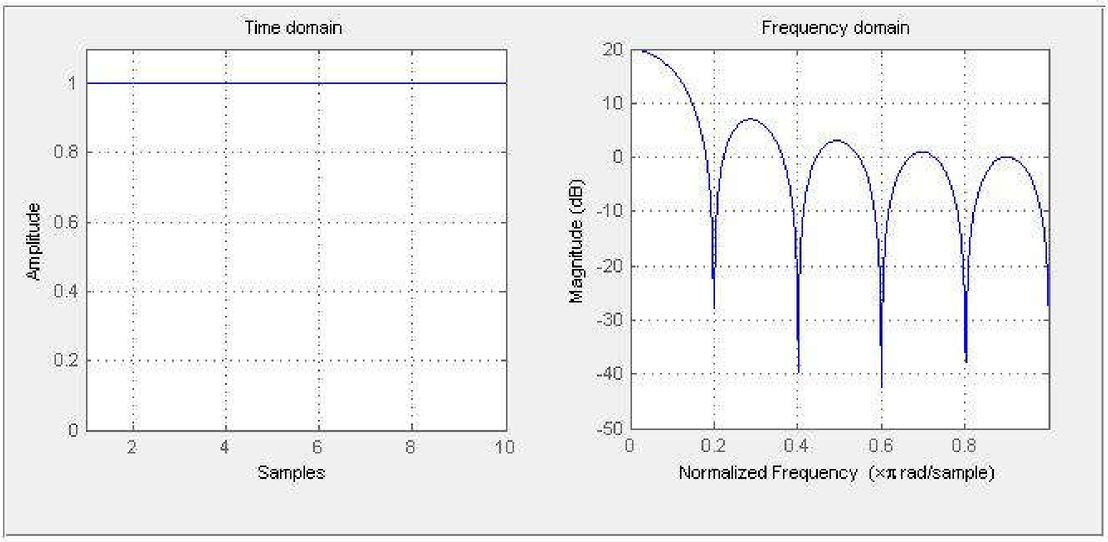


Figure 9. Time doman and Frequency doman of Rectangular window Technique

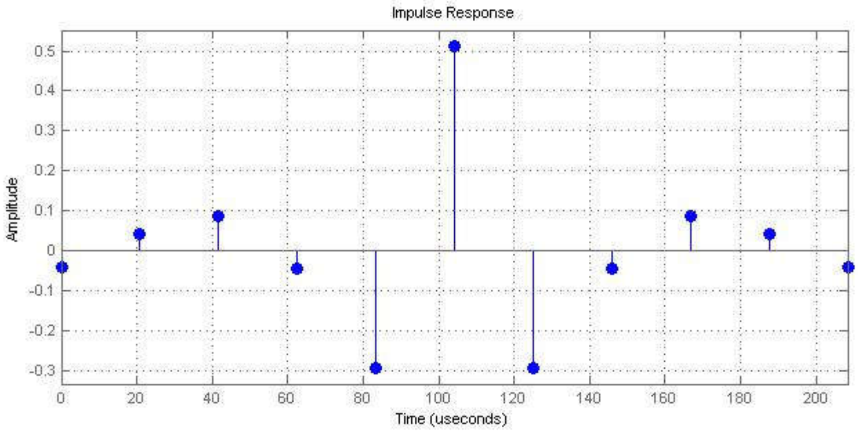


Figure 10. Impulse response of Rectangular window Technique

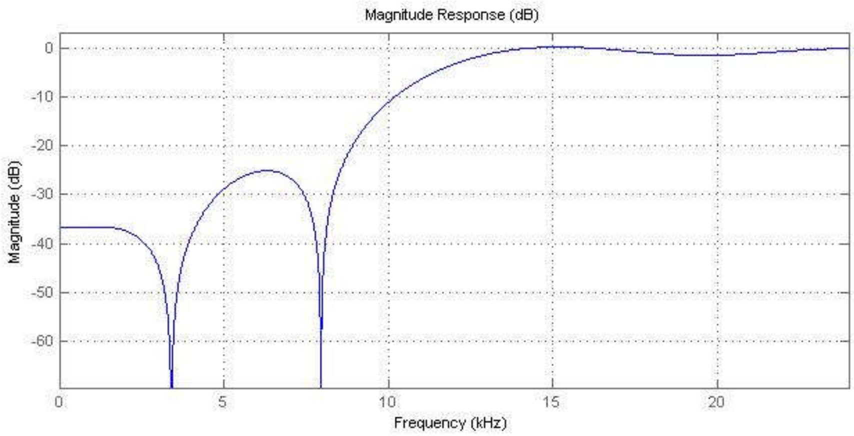
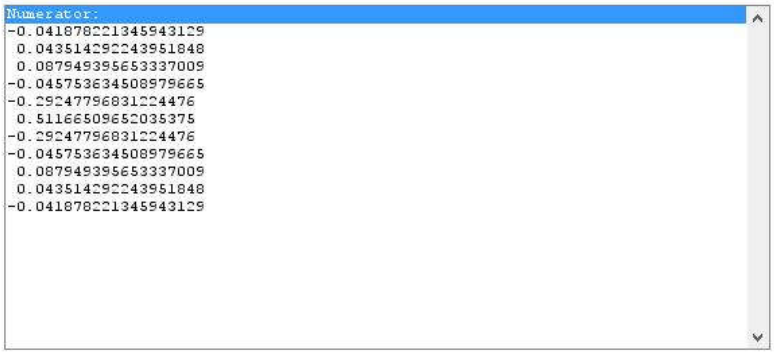


Figure 11. Magnitude response of Rectangular window Technique



Numerator:
-0.041878221345943129
0.043514292243951848
0.087949395653337009
-0.045753634508979665
-0.29247796831224476
0.51166509652035375
-0.29247796831224476
-0.045753634508979665
0.087949395653337009
0.043514292243951848
-0.041878221345943129

Figure 12. Filter coefficient of Rectangular window technique

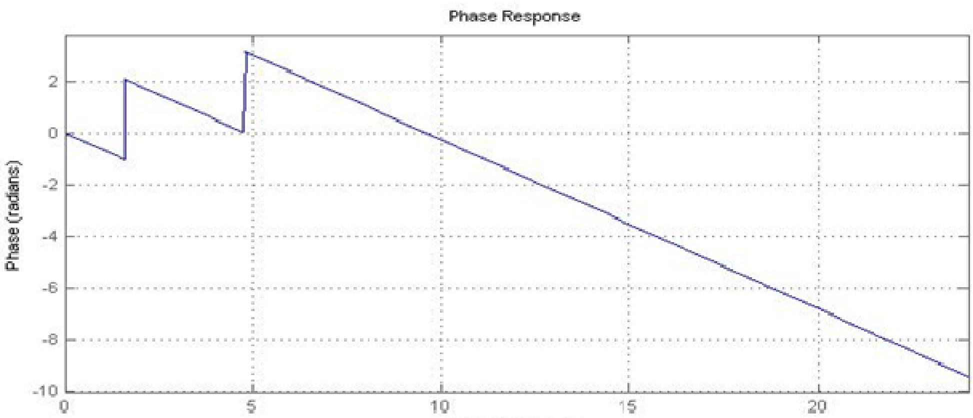


Figure 13. Phase response of taylor window Technique

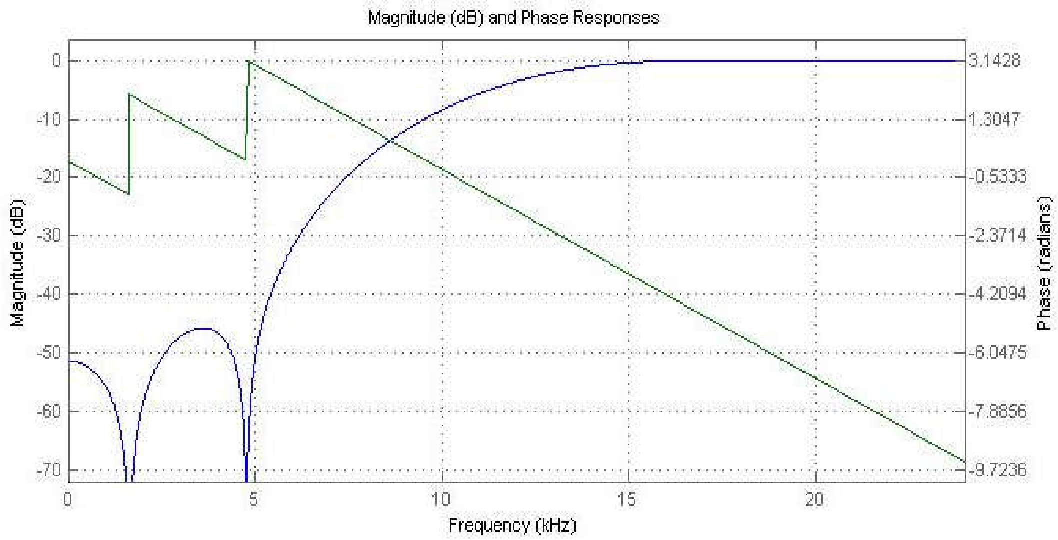


Figure 14. Magnitude response and Phase response of Taylar window Technique

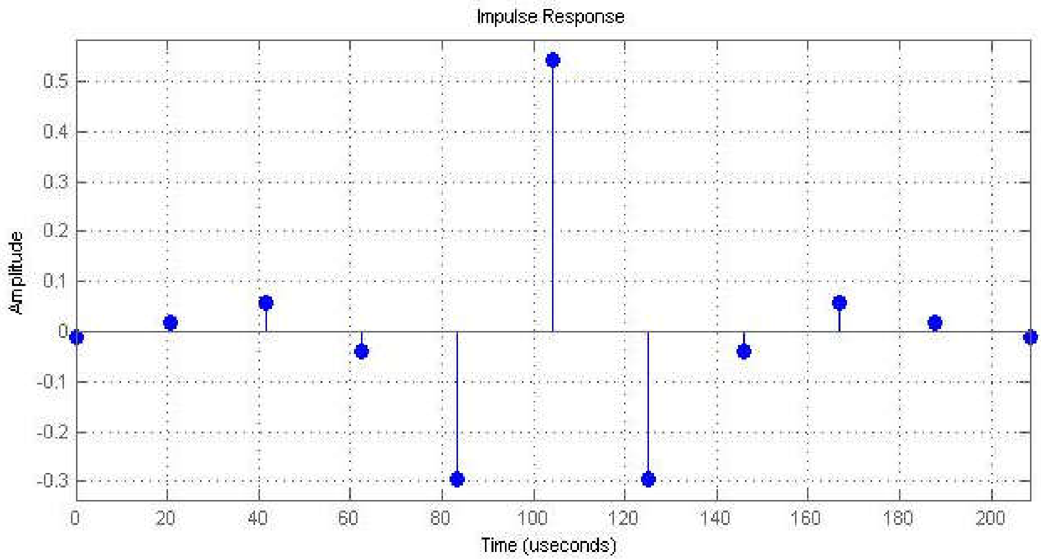


Figure 15. Impulse response of Taylor window Technique

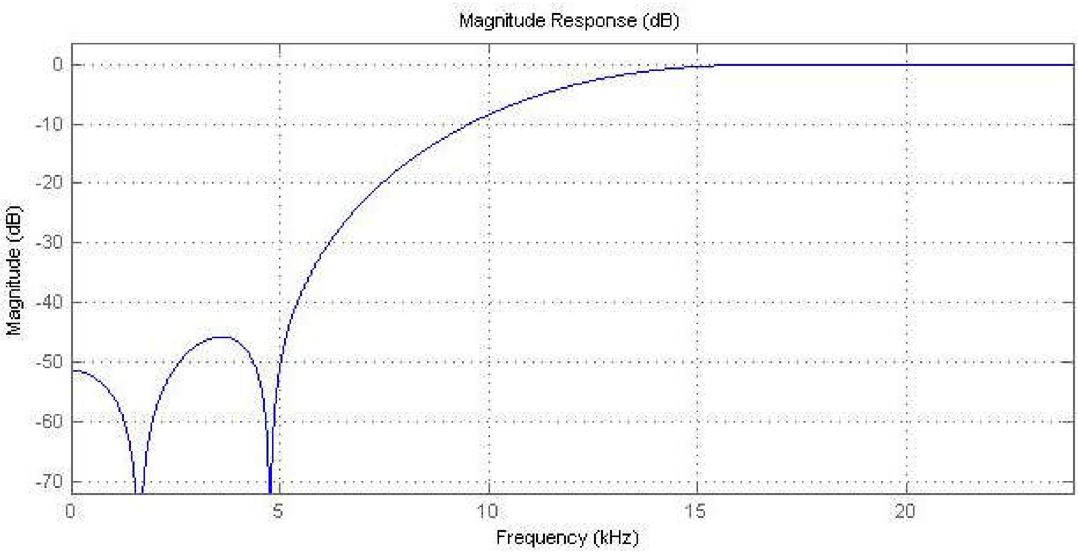


Figure 16. Magnitude response of Taylor window Technique

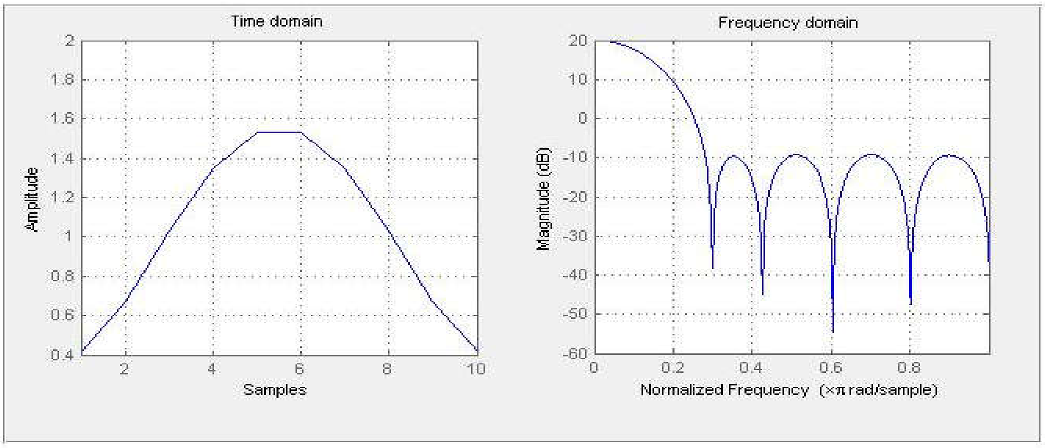


Figure 17. Time doman and Frequency doman of Taylor window Technique

```
Numerator:
-0.011662035509774692
 0.018513861418611423
 0.056818264927696378
-0.039241850189740975
-0.29447015248525649
 0.54282813140758945
-0.29447015248525649
-0.039241850189740975
 0.056818264927696378
 0.018513861418611423
-0.011662035509774692
```

Figure 18. Filter coefficient of Taylor window Technique

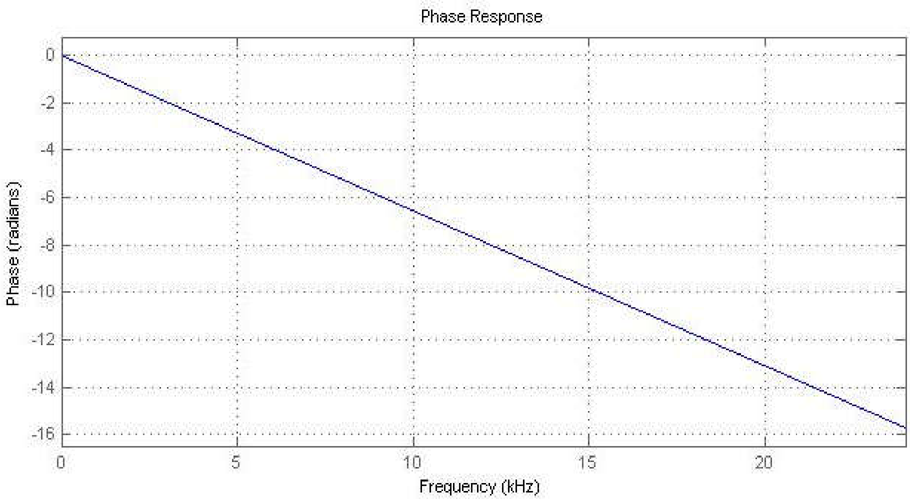


Figure 19. Phase response of Tringular window Technique

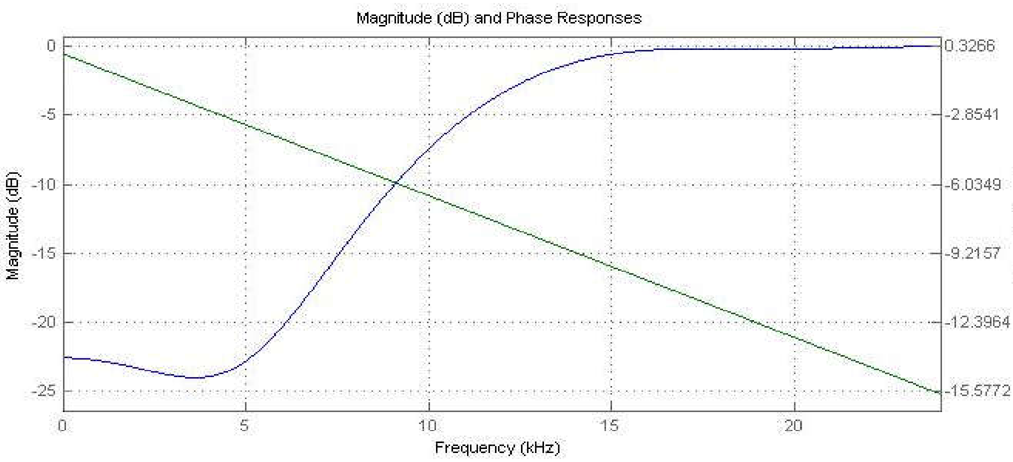


Figure 20. Magnitude and Phase response of Tringular window Technique

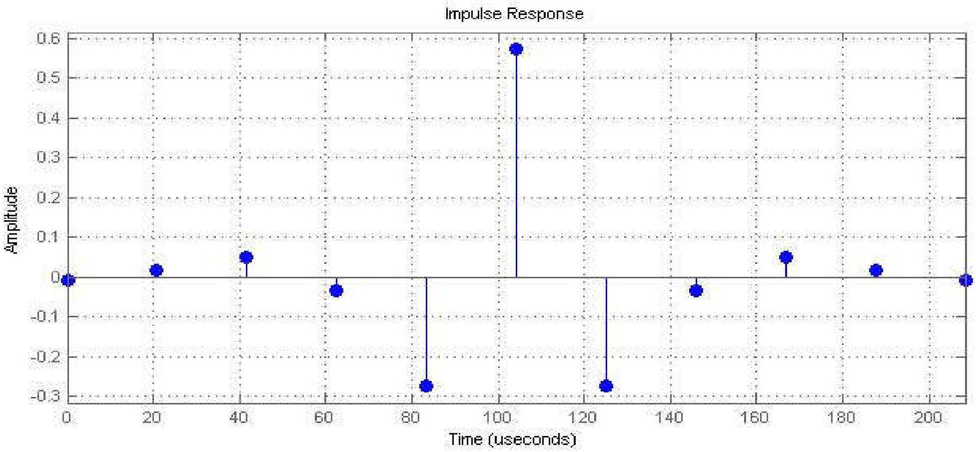


Figure 21. Impulse response of Tringular window Technique

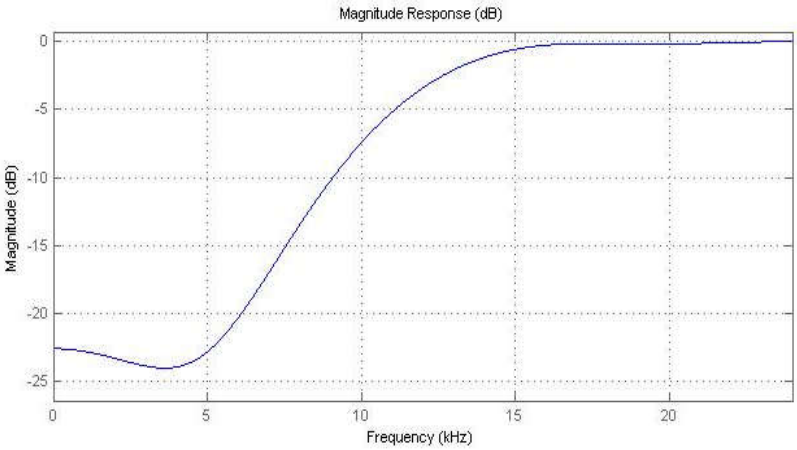


Figure 22. Magnitude response of Tringular window Technique

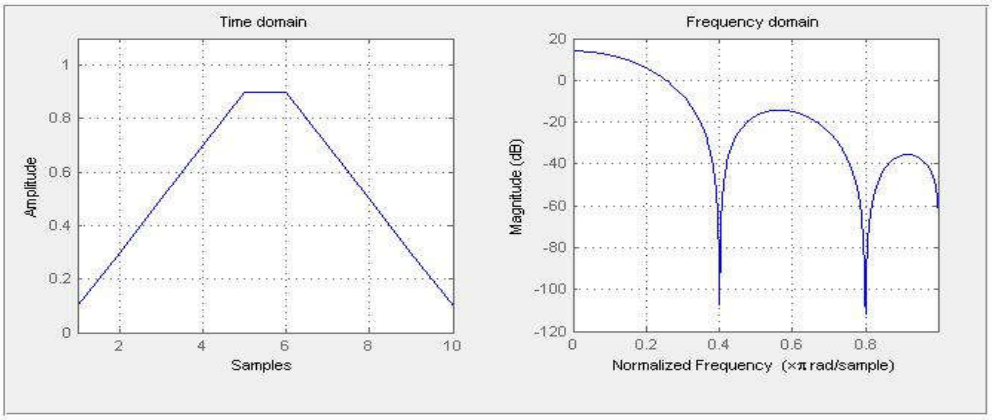


Figure 23. Time doman and Frequency doman of Tringular window Technique

Numerator:
-0.0078147653908919798
0.016240135044328301
0.049235894305822384
-0.034151777031278578
-0.27289204637941838
0.57288144904492455
-0.27289204637941838
-0.034151777031278578
0.049235894305822384
0.016240135044328301
-0.0078147653908919798

Figure 24. Filter coefficient of Tringular window Technique

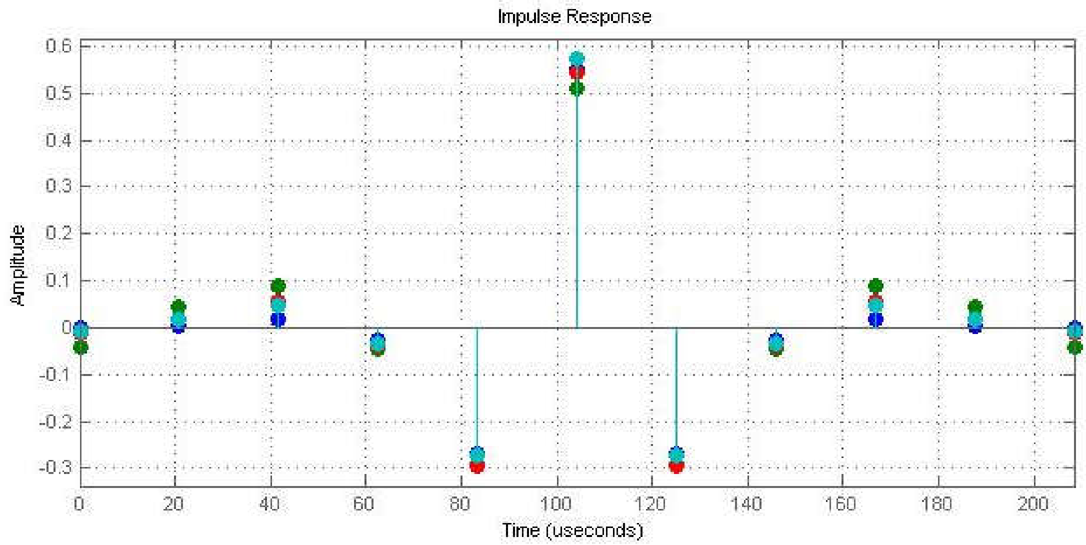


Figure 25. Impulse response of Tringular, Rectangur, Blackman and Taylor window Technique

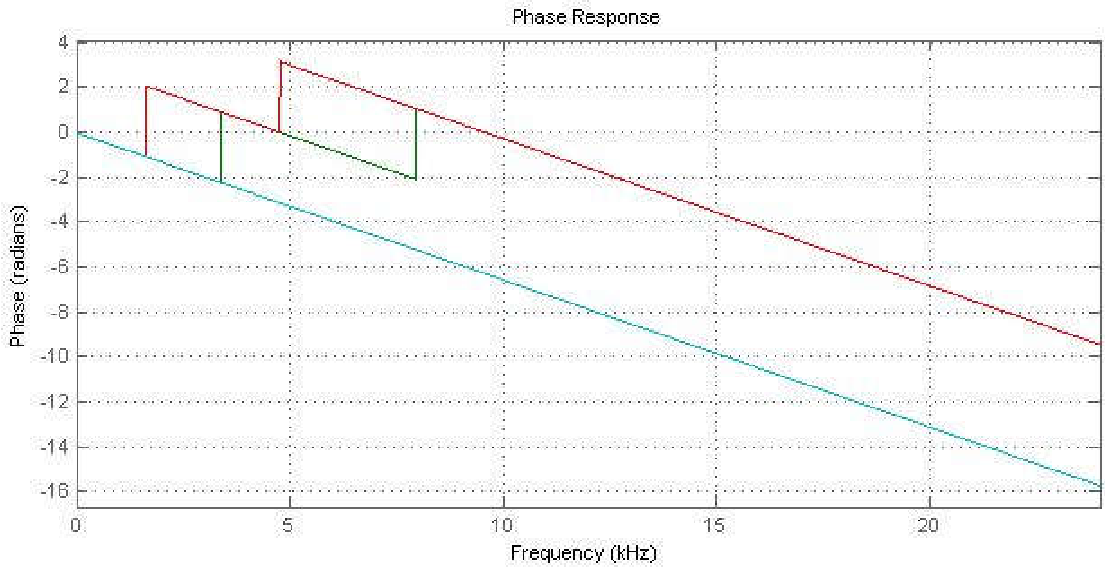


Figure 26. Phase response of Tringular, Rectangular, Blackman and Taylor window Technique

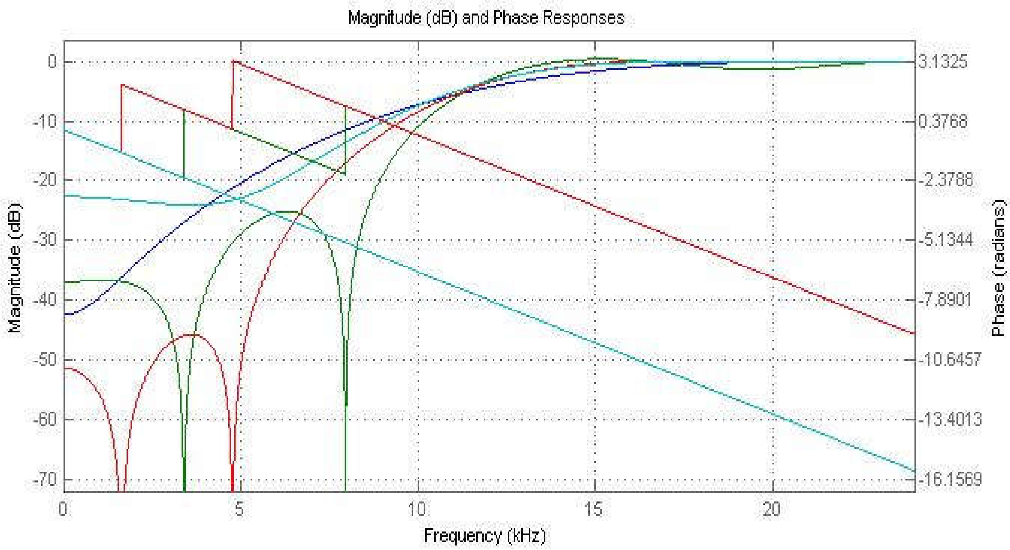


Figure 27. Magnitude and Phase response of Tringular, Rectangular, Blackman and Taylor window Technique

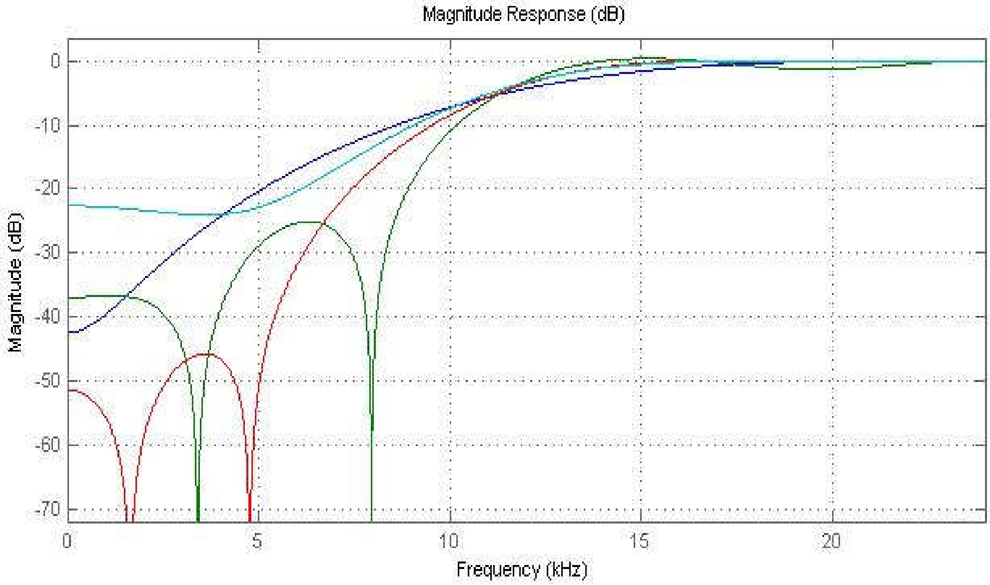


Figure 28. Magnitude response of Tringular, Rectangular, Blackman and Taylor window Technique

RESULTS:- Blackman, Rectangular, Tringular and Taylor Window Technique

Table 3. Simulation Result in MATLAB

Window technique	Leakage Factor	Relative sidelobe atteanuation	Mainlobe width (dB)
Blackman	0	-64.6dB	0.35938
Rectangular	9.2	-13dB	0.17188
Tringular	0.17	-28.2dB	0.25
Taylor	0.34	-29.2dB	0.21875

Conclusion

In this paper FIR high pass filter has been designed and simulated using Blackman, Tringular, Taylor and Rectangular Window technique. It has been compared leakage factor, mainlobe width and relative sidelobe attenuation of the three window from the simulated result. Rectangular window mainlobe width is increase and decreases the amplitude of sidelobes that is increases the attenuation. Rectangular Window main lobe width (-3dB) is 0.17188 at sampling frequency 48000 Hz, cut off frequency 10800 Hz and order 10 Rectangular window has greater mainlobe width and less leakage factor in comparison of Blackman window Taylor and Tringular window.

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