Designing a Low- Pass Fir Digital Filter By Using Rectangular Window and Bartlett Window Technique

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ABSTRACT

The aim of our paper is to design FIR filter using Rectangular and Bartlett Window Techniques of order 10. In this paper we represent the significance of filter in our daily life. 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 low pass FIR filter.

Keywords— DSP, FIR, Low-Pass Fir Digital Filter, Rectangular Window and Bartlett Window.

1. INTRODUCTION

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 [1].

Digital Signal Processing (DSP) is an important (imp) 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. With MP3 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. [2]

Applications of DSP in area wise are as following:-

1).Telecommunication- Echo cancellation in telephone networks, equalization, telephone dialing 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.

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. [3]

2. WINDOW TECHINIQUES

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

$$\mathbf{H_d(e^{j\omega})}= \sum_{n=-\infty}^{\infty} h_d(n)e^{-j\omega} \quad (1)$$

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

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.

2.1 BARTLETT WINDOW FUNCTION

The window function of a non-causal Bartlett window is expressed by

$$w_{Bart}(n)=\begin{Bmatrix} 1+n, -\frac{M-1}{2} < n < 1 \\ 1-n, 1 < n < \frac{M-1}{2} \end{Bmatrix} \quad (3)$$

2.2 RECTANGULAR WINDOW FUNCTION

The weighting function for the rectangular window is given by

$$W_R(n) = \begin{Bmatrix} 1 for\ |n| \le \frac{M-1}{2} \\ 0,\ otherwise \end{Bmatrix} \quad (4)$$

3. SIMULATION AND RESULT

PARAMETER	VALUES
Sampling Frequency(F_s)	1950
Cut off Frequency(F_c)	450
Order(N)	10

Table 1: Parameter Specification

Filter coefficient h(n)	Window Techniques	
	Rectangular	Bartlett
h(0)=h(10)	0.0522830	0
h(1) =h(9)	0.0369040	0.0080511
h(2)= h(8)	0.0990004	0.0431966
h(3)= h(7)	0.0380084	0.0248762
h(4)= h(6)	0.3153271	0.2751715
h(5)	0.4605716	0.5023998

Table2: Filter coefficients of Rectangular & Bartlett Window Techniques

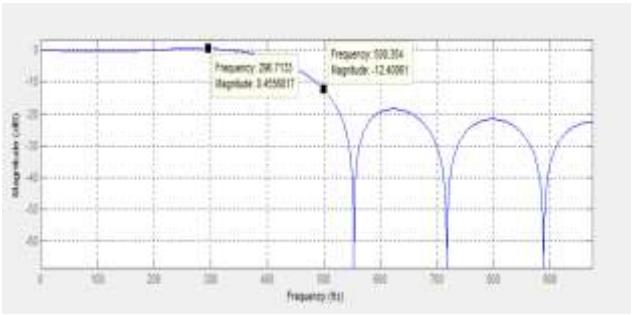

Figure 1: Magnitude response of Rectangular Window Technique

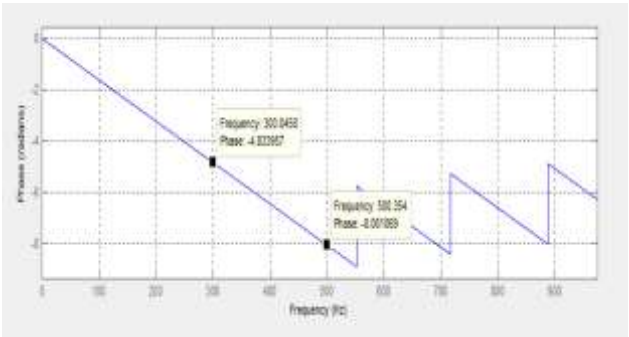

Figure 2: Phase response of Rectangular Window Technique

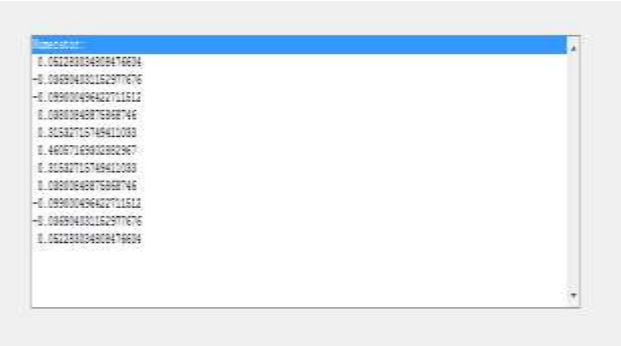

Figure 3: Filter coefficient of Rectangular Window Technique

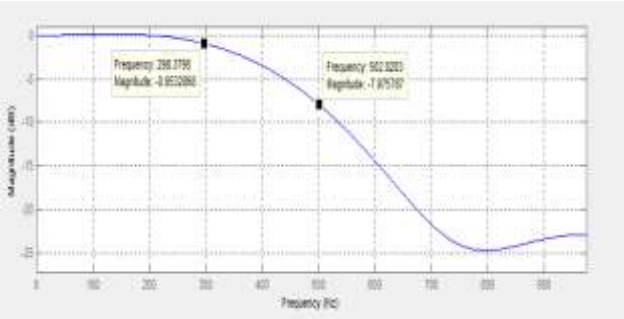

Figure 4: Magnitude response of Bartlett Window Technique

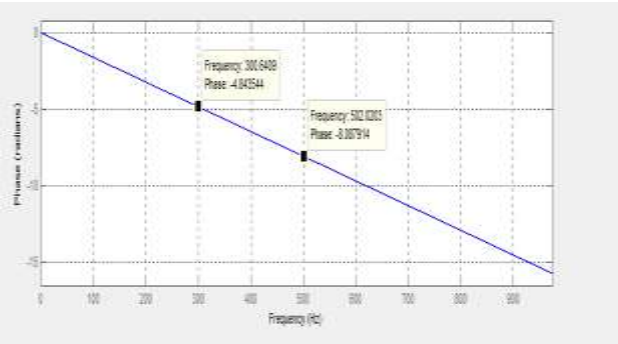

Figure 5: Phase response of Bartlett Window Technique

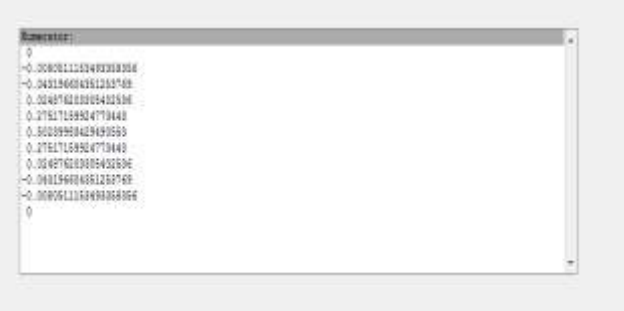

Figure 6: Filter coefficient of Bartlett Window Technique

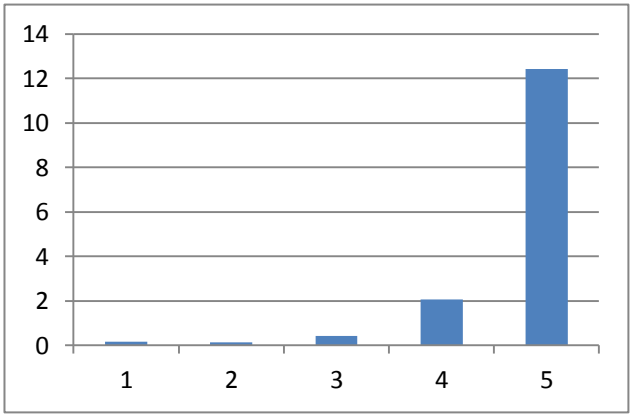

Figure 7: Magnitude and Frequency plot of Rectangular Window Technique.

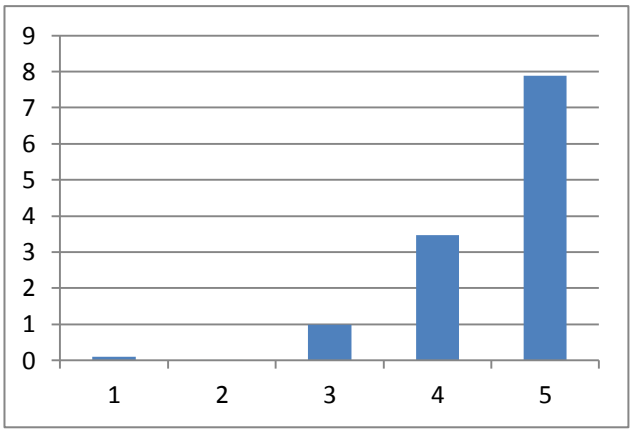

Figure 8: Magnitude and Frequency plot of Bartlett Window Technique

4. CONCLUSIONS

By analysis we conclude that Rectangular Window Technique has better response than Bartlett Window Technique

REFERENCES

[1]. Sanjit K. Mitra "digital signal processing" a computer-based approach second edition McGraw Hill, pp.1.

[2].Michael Weeks, "digital signal processing " Using MATLAB and Wavelets, 2007 edition, Infinity Science Press, pp.1

[3].S Salivahanan, C Gnanapriya, A Vallavaraj, "digital signal processing " Tata McGraw-Hill, second edition pp.3,443-448.

[4].John G. Proakis, Dimitris G. Manolakis, "digital signal processing " PEAESON, fourth edition

[5]. Sen M Kuo, Bob H Lee, "digital signal processing " John Wiley & Sons, 2001 edition

[6].Richars G. lyons, Understandingdigital signal processing "PEARSON, second edition.

[7].A Nagoor Kani, "digital signal processing" McGraw-Hill Education, second edition.

[8].Alan V. Oppenheim, Ronald W. Schafer,"digital signal processing", PEARSON education, 2015 edition.

[9]. Steuen W. Smith, "digital signal processing", Elsevier,2005 edition.

[10]. Emmanuel Ifeachor, Barry W. Jervis,"digital signal processing " PEARSON, second edition

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