



IJRASET

International Journal For Research in
Applied Science and Engineering Technology



INTERNATIONAL JOURNAL FOR RESEARCH

IN APPLIED SCIENCE & ENGINEERING TECHNOLOGY

Volume: 5 Issue: VI Month of publication: June 2017

DOI:

www.ijraset.com

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Design of Low pass Fir Filter Using Hanning and Hamming Window Techniques

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Abstract: In this paper we represent the importance of filter in our daily life. low pass filter is designed with sampling frequency and cut off frequency. a filter is essentially a network that selectively change the wave shape of a signal in a desired manner.

I. INTRODUCTION

A filter is a linear time immutable system that removes unwanted noise or disturbances with the desired signal, it can be used in spectral shapes like signal identification of channels, radar, sonar, biomedical, etc.[1]
It is a network that changes the amplitude, frequency, phase frequency characteristics of the signal in the desired manner. A digital filter plays a very important role in DSP Digital filters have features such as linear phase feedback. If we compare digital filters with Analog filters, they prefer in many applications such as speech processing, image processing and data compression. To get the desired change sequence, filter the process of change of the input sequence to cancel. The filter designing process can be described as an optimization problem, where the requirements for each error need to be reduced [2].In Digital Signal Processing (DSP), an essential part is Filter. For different purposes Different types of filters are used *i.e*

S.NO.	FILTER	PURPOSE
1	Low pass filter	Passes low frequency band
2	High pass filter	Passes high frequency band
3	Bandpass filter	Passes selected range of frequency band
4	Bandstop filter	Stops selected range of frequency

Table-1. Type of Filter and Purpose

A. There are two types of filters

- 1) Finite Impulse Response (FIR) filter and
- 2) Infinite Impulse Response (IIR) filter

B. FIR filters have the following advantages over IIR filters

- 1) They can have an exact linear phase.
- 2) They are always stable.
- 3) The design methods are generally linear.
- 4) They can be realized efficiently in hardware.
- 5) The filter start-up transients have finite duration.[3]

C. Advantages of DSP

- 1) Digital Signal Processing Operations can be changed by changing programs in digital program processing systems, meaning they are flexible systems
- 2) Better control of accuracy in digital systems than Analog systems
- 3) Digital signals on magnetic media such as magnetic tape are easily stored, without the loss of the quality of reproduction of the signal.

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- 4) Digital signals can be processed from the line, which means they are easily transported.
- 5) Sophisticated signal processing algorithms can be implemented by DSP method.
- 6) Digital circuit components are less sensitive to the tolerance of value
- 7) s. Digital systems are independent of temperature, aging and other external parameters[4]

D. Disadvantages of DSP

- 1) DSP techniques are limited to signals with relatively low bandwidths.
- 2) The need for an ADC and DAC makes DSP uneconomical for simple applications(e.g. simple filters) [5]

E. Application

The world of science and engineering is full of signals: Volts generated by remote space investigations by painting, heart and brain, radar and sonar echoes, seismic vibrations and countless other applications. Digital Signal Processing is a science to use computers to understand these types of data. Various types of goals are included: filtering, speech recognition, image enhancement, data compression, neural networks, and much more, DSP is one of the most powerful technologies that will shape science and engineering in the twenty-first century. Suppose we give analog-to-digital converter to computers, and then use it to gain a part of the real world data.

II. WINDOW TECHNIQUE

Window function can be divided into two parts: Fixed Window function and adjustable window function Hanning, hamming, blackman and rectangular window The type of fixed window function is Kaiser, The window is a type of adjustable window function.

Window technique destroys a function called window The ceremony It is also called tapering function. It tells you that if Some intervals are chosen, it gives the finite non-zero value Zero value within that interval and outside the zero interval. A The main effect of windowing is that their discontent Frequency response transitions are converted into band Between prices on both sides of dissection

A. Hamming Window

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.

Here Equation of Hamming window sequence can be defined by:

$$W(n) = 0.54 - 0.46 \cos \frac{2\pi n}{N-1} \quad ; \quad 0 < n < N-1$$

B. Hanning Window

The window function of a causal Hanning window is given by

$$W_{\text{Hann}}(n) = \begin{cases} 0.5 - 0.5 \cos \frac{2\pi n}{N-1}, & 0 \leq n \leq N-1 \\ 0, & \text{otherwise} \end{cases}$$

$$W_H(n) = W_R(n) \left[0.54 + 0.46 \cos \frac{2\pi n}{N-1} \right]$$

III. SIMULATION/ RESULT

PARAMETERS	VALUES
Sampling Frequency(Fs)	48000
Cut off Frequency (Fc)	10800
Order(N)	15

Table 1: Parameter Specification

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Frequency	Window Techniques	
	Hanning	Hamming
0.1π	14.0882	14.02139
0.2π	4.7577	1.6632
0.3π	-19.99978	-29.22645
0.4π	-23.52369	-24.07243
0.5π	-25.19936	-30.62168
0.6π	-35.23522	-21.6182
0.7π	-45.13442	-23.24138
0.8π	-41.05892	-40.27193
0.9π	-49.28731	-25.41986
π	-109.2907	-21.94702π

Table 2. phase of haaning and hamming window technique

IV. RESULT

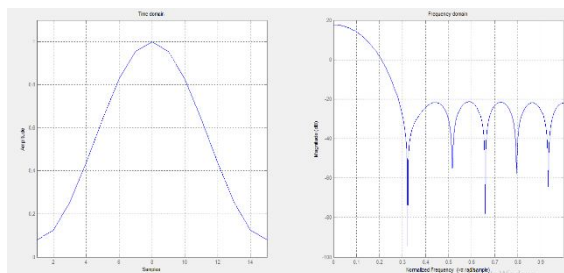


Figure 1. Time domain and frequency domain of hamming window

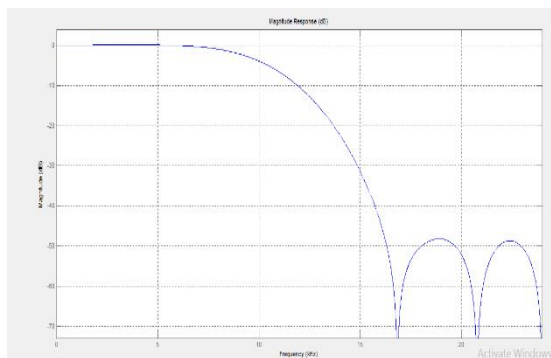


Figure 2. Magnitude response of hamming window

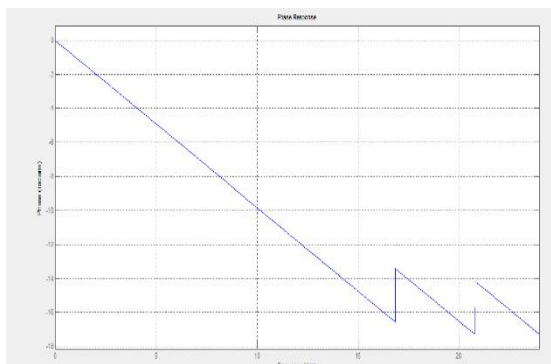


Figure 3. phase response of hamming window

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    -0.0031458511376660984
    0.0013729088704452818
    0.013483327161586862
    0.002213999894538762
    -0.052146748214445551
    -0.037619710002305448
    0.16548735595730943
    0.41040421747053674
    0.41040421747053674
    0.16548735595730943
    -0.037619710002305448
    -0.052146748214445551
    0.002213999894538762
    0.013483327161586862
    0.0013729088704452818
    -0.0031458511376660984
    
```

Figure 4. filter coefficients of hamming window

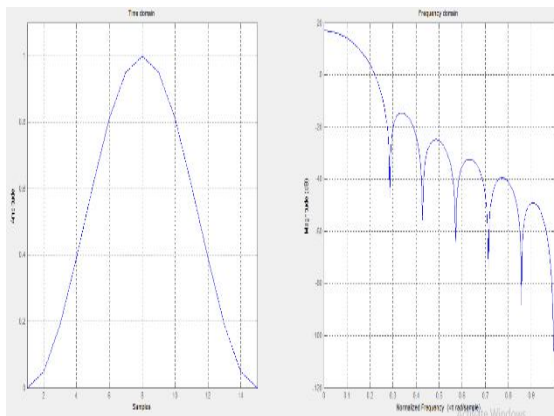


Figure 5. time domain and frequency domain of hanning window

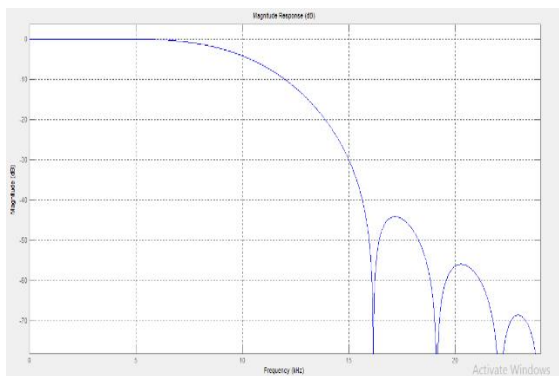


Figure 6. magnitude response of hanning window

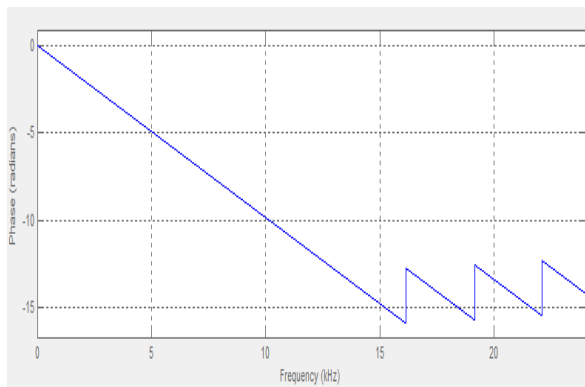


Figure 7. phase response of hanning window

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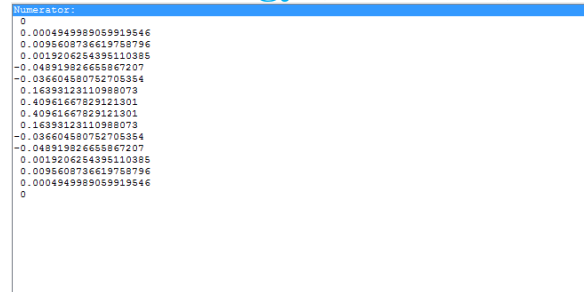


Figure 8. filter coefficient of hanning window

V. CONCLUSION

In this paper, an FIR filter has been designed using hamming window function and hanning window function. The choice of filter depends on the design of the designer, whether to compromise accuracy of approximation or ease of design. In this FIR filter, a hamming and hanning window have been used.

REFERENCES

- [1] Eppili JAYA, "FIR Filter Design using new hybrid window function", International Journal of Science, Engineering and Technology, vol 4, Issue 6 June
- [2] Divya Goyal, "Review of low pass FIR Filter Design Using window method", International Journal of Electronics & Communication Engineering Volume 4, Issue 7, July 2015
- [3] Ayush Gavel, "Design of low pass FIR Filter Using Rectangular and Hamming window Techniques", International Journal of Innovation Science, Engineering & Technology, Vol 3 Issue 8 August 2016.
- [4] Smith, Steven. Digital signal processing: a practical guide for engineers and scientists. Newnes, 2013
- [5] Breining, Christina, et al. "Acoustic echo control. An application of very-high-order adaptive filters." IEEE Signal Processing Magazine 16.4 (1999): 42-69.
- [6] If each or, Emmanuel C., and Barrie W. Jervis. Digital signal processing: a practical approach. Pearson Education, 2002.

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