INTERNATIONAL JOURNAL
FOR RESEARCH

IN APPLIED SCIENCE & ENGINEERING TECHNOLOGY

Design of Low Pass Fir Filter Using Parzen And Nuttall Window Technique

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Abstract: Since from the beginning in the field of dsp the research and development has led to the enormous success. Basically the main role of the filter is to improvise the quality of the signal. In this research paper the efficient design of low pass fir filter is to be discussed. According to the nuttall & parzen window technique are used for the analysis of magnitude and phase response. The following result from the nuttall & parzen provide the better result in respect of magnitude and phase response of low pass fir filter of order 30.
Keywords: DSP, LPF, DIGITAL FILTER, FIR FILTER, PARZEN WINDOW, NUTTALL WINDOW.

I. INTRODUCTION

Signals play a major role in our life. In general, a signal can be function of time, distance, position, temperature, pressure, etc., and it represents some variable of interest associated with a system. For example, in an electrical system the associated signals are electric current and voltage. In a mechanical system, the associated signals may be force, speed, torque etc. In addition to these, some examples of signals that we encounter in our daily life are speech, music, picture and video signals.

A signal can be represented in a number of ways. Most of the signals that we come across are generated naturally. However, there are some signals that are generated synthetically. In general, a signal carries information, and the objective of signal processing is to extract this information.

Signal processing is a method of extracting information from the signal which in turn depends on the type of signal and the nature of information it carries. Thus signal processing is concerned with representing signals in mathematical terms and extracting the information by carrying out the algorithmic operation on the signal. Mathematically, a signal can be represented in terms of basic functions in the domain of the original independent variable or it can be represented in terms of basic function in a transformed domain. Similarly, the information contained in the signal can also be extracted either in the original domain or in the transformed domain.

A system may be defined as an integrated unit composed of diverse, interacting structures to perform a desired task. The task may very such as filtering of noise in a communication receiver, detection of range of target in a radar system, or monitoring the steam pressure in a boiler. The function of a system is to process a given input sequence to generate an output sequence.

DSP

Digital signal processing (DSP) is the process of analyzing and modifying a signal to optimize or improve its efficiency or performance. It involves applying various mathematical and computational algorithms to analog and digital signals to produce a signal that's of higher quality than the original signal.

A. Advantages
1) Better control of accuracy in digital system compared to analog system.
2) Digital systems are independent of temperature, ageing and other external parameter.
3) Digital signal processed off line, i.e., these are easily transported.
4) Digital circuits are less sensitive to tolerances of component values.
5) Digital circuits can be reproduced easily in large quantities at comparatively lower cost.
6) Storage of digital data is very easy. A signal can be stored on various stirage media such as magnetic tapes, disks and optical disks without any loss. On the other hand, stored analog signals deteriorate rapidly as time progress and cannot be recovered in their original form.

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B. Disadvantages

1) Higher power consumption and size of a DSP implementation can make it unsuitable for simple very low-power or small size applications.
2) For reconstructing a signal from its sample, the sampling frequency must be at least twice the highest frequency component present in the signal.
3) The active devices are less reliable than passive components.

C. Application

1) Military Radar signal processing, sonar signal processing, navigation, secure communications and missile guidance.
2) Seismology DSP techniques are employed in geophysical exploration for oil and gas, detection of underground nuclear explosion and earthquake monitoring.
3) Telecommunication Echo cancellation in telephone networks, modems, line repeaters, channel multiplexing, data communication, data encryption, video conferencing, cellular phone and FAX.
4) Consumer electronics Digital Audio/TV, electronic music synthesizer, educational toys, FM stereo application and sound recording application.
5) It is used for analysis and control of industrial processes, medical imaging such as CAT scans and MRI.

D. Digital Filter

A digital filter is a system that performs mathematical operations on a discrete and sampled time signal, so as to enhance or reduce certain aspects of that particular signal as may be necessary. It is largely used in signal processing and differs from an analog filter, which is an electronic circuit working with continuous signals. Digital filters are expensive compared to analog ones, but they can turn many impractical or impossible designs into possibilities. In everyday life, they can be found in devices like cell phones, radios and audio/video receivers.

In Digital Signal Processing (DSP), an essential part is Filter.

For different purposes Different types of filters are used i.e.

Table 1. Type of Filter and Purpose

SN	FILTER	PURPOSE
1.	Low pass filter	Passes low frequency band
2.	High pass filter	Pass high frequency band
3.	Band pass filter	Passes selected range of frequency band
4.	Band stop filter	Stops selected range of Frequency

There are two types of filters i.e.

1) *FIR filter:* Finite impulse response (FIR) filter is the basic elements in a digital signal processing system, and it can assurance a firm linear phase frequency characteristic with any kind of amplitude frequency characteristic. The unit impulse response is finite; so FIR filters are stable system.
2) *IIR filter:* The infinite impulse response (IIR) filter has a feedback loop and is recursive structure, and it. IIR filters are not linear phase and the precision of amplitude frequency characteristic is very high.

II.   FIR FILTER DESIGN METHOD

FIR filter design consists of Approximation problem and Realization problem. The approximation stage gives transfer function. 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 various methods to design FIR filter i.e. Equiripple, least-squares, Window, Least-squares, Complex Equiripple, Maximally flat Least Pth- norm, Constrained Equiripple, Generalized Equiripple, Constr. Band Equiripple and

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Interpolated FIR.

III.   WINDOW TECHNIQUE

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

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

Where,

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

The fourier coefficients of the series h(n) are identical to the impulse response of a digital filter. There are two difficulties with the implementation of eq(1) for designing a digital filter. First, the impulse response is of infinite duration and second, the filter is non-causal and

unrealisable. 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 unrealisable IIR filter.

The windowing method requires minimum amount of computational effort, so window method is simple to implement. For the given window, the maximum amplitude of ripple in the filter response is fixed. Thus the stop band attenuation is fixed in the given window, but there is some drawback also of this method. The design of FIR filter is not flexible. The frequency response of FIR filter show the convolution of spectrum of window function and desired frequency response because of this; the pass band & stop band edge frequency cannot be precisely specified.

 Some of window commonly used are Barlett Window, Barlett-Hanning window, Blackman window, Blackman-Haris window, Bohman window, Chebyshew, flat Top window, Hamming window, Hanning window, Nuttall, Parzen window, Rectangular window and the Kaiser window. In this paper we use Nuttall and Parzen window method to design filter.

A.   Nuttall Window

The Nuttall window has the widest main lobe and lowest maximum side lobe level among the Blackman, exact for the Nuttall window.

$$\omega(n) = a_0 - a_1\cos\left(2\pi\frac{n}{N-1}\right) + a_2\cos\left(4\pi\frac{n}{N-1}\right) - a_3\cos\left(6\pi\frac{n}{N-1}\right) \dots\dots\dots\dots\dots(3)$$

where, n=0,1,2,3…….N-1
The equation for the periodic Nuttall

$$\omega(n) = a_0 - a_1\cos\left(2\pi\frac{n}{N}\right) + a_2\cos\left(4\pi\frac{n}{N}\right) - a_3\cos\left(6\pi\frac{n}{N}\right) \dots\dots\dots\dots\dots(4)$$

where, n=0,1,2,3…….N-1. The periodic window is N periodic. The coefficients for this window are given in table.

Table 2. Coefficient of Nuttall window

a_0	0.3635819
a_1	0.4891775
a_2	0.1365995
a_3	0.0106411

B.   Parzen Window

The following equations defines the N point Parzen window over the interval

$$\frac{(N-1)}{2} \le n \le \frac{(N-1)}{2}$$

$$\omega(n) = \begin{cases} 1 - 6\left(\frac{|n|}{\frac{N}{2}}\right)^2 + 6\left(\frac{|n|}{\frac{N}{2}}\right)^3 & 0 \le |n| \le (N-1)/4 \\ 2\left(1 - \frac{|n|}{N/2}\right)^3 & (N-1)/4 \le |n| \le (N-1)/2 \end{cases}$$

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……….…….…(4)

In above equation we can calculate Parzen window technique with magnitude and phase response with other math work in matlab[5].

IV. DESIGN SIMULATION

To design the low pass FIR filter using MATLAB Nuttall Window and Parzen Window technique the parameter specifications are given in table 3. As

Table 3. Parameter specification

PARAMETER	VALUES
Sampling Frequency(F_s)	36000Hz
Cut off Frequency(F_c)	8200Hz
Order(N)	10
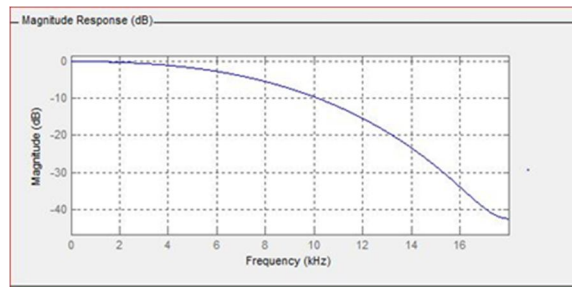
Fig.1 Magnitude Response of Nuttall Window Technique
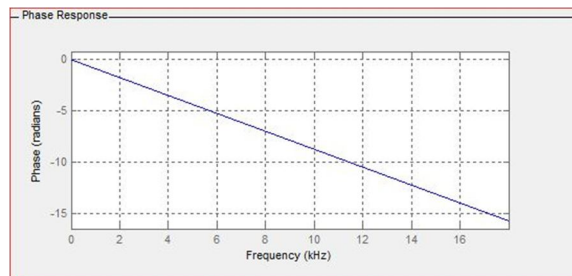
Fig.2 Phase Response of Nuttall Window Technique
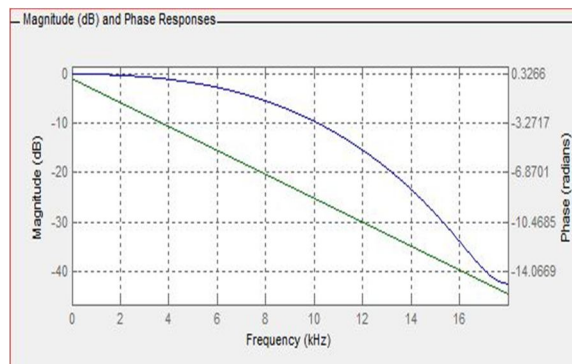
Fig.3 Magnitude [dB] and Phase [rad]of Nuttall Window Technique

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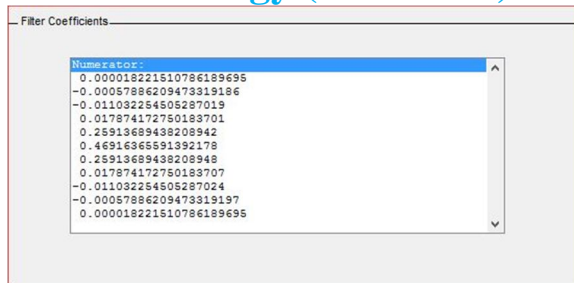

Fig. 4 Filter coefficient on Nuttall Window Technique

Fig. 1 shows the magnitude response of Nuttall window and Fig. 2 shows phase response of Nuttall window. The magnitude response contains ripples in stop band and good pass band response.
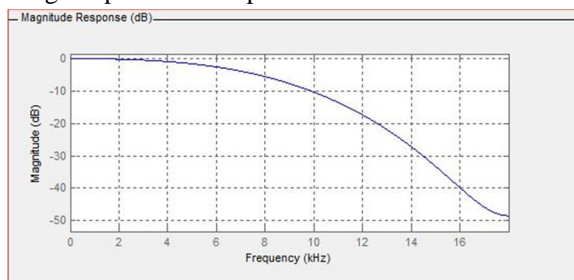

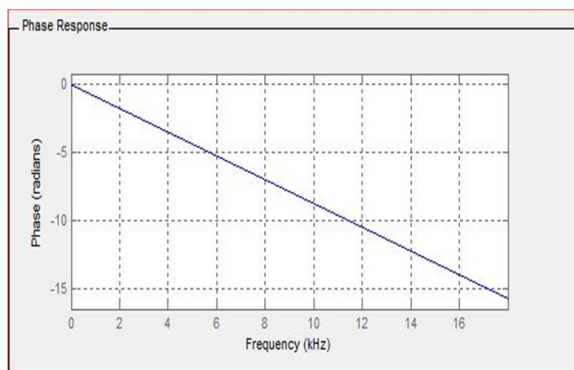Fig.5 Magnitude Response of Parzen Window Technique

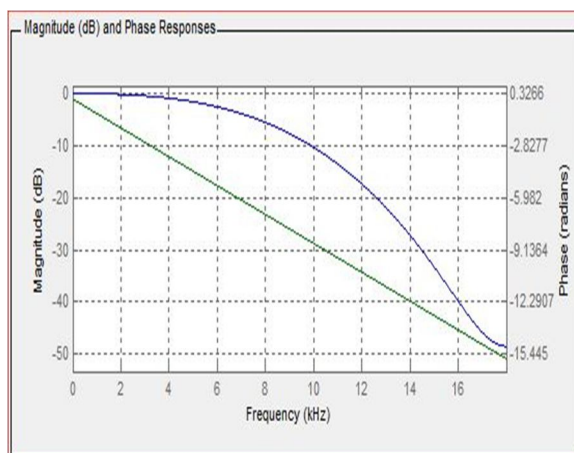Fig.6 Phase Response of Parzen Window Technique

Fig.7 Magnitude [dB] and Phase [rad] of Parzen Window Technique

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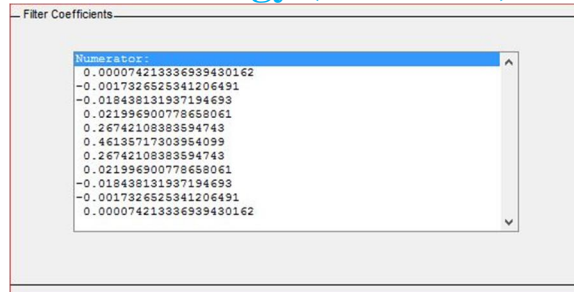

Fig.8 Filter coefficients of Parzen Window Technique

Magnitude and phase response of Parzen window is shown in Fig. 5 and 6. In this method magnitude response contains negligible ripples in stop band and it is also having good pass band response.
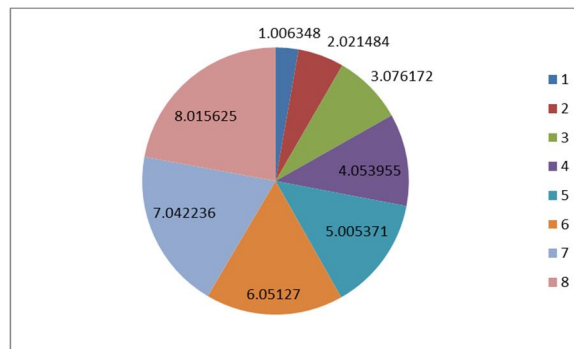
V.   COMPARATIVE ANALYSIS:-

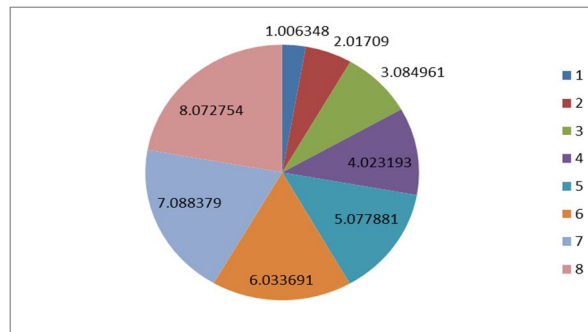
Fig.9 Magnitude and Frequency plot of Nuttall Window

Fig.10 Magnitude and Frequency plot of Parzen Window

VI.   RESULT

Window Technique	Mainlobe width (-3dB)	Relative Sidelobe attenuation
Nuttall	0.125	-88.3dB
Parzen	0.11719	-53dB

Table:(4)- Simulation Result in Matlab

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Frequency (in kHz)	Magnitude (in dB)	
	Nuttall window	Parzen window
1	-0.87204	-0.87820
2	-1.76407	-1.76024
3	-2.68446	-2.69213
4	-3.53774	-3.51089
5	-4.36801	-4.43127
6	-5.28072	-5.26538
7	-6.14551	-6.18577
8	-6.99495	-7.04480

Table:(5)-Magnitude & Frequency Result in Nuttall & Parzen Window

VII.   CONCLUSION

From this research paper the low pass FIR filter has been designed like Nuttall and Parzen window techniques. The differential analysis of magnitude of filter has been performed the same frequency.

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