Design of FIR Filter for Efficient Utilization of Speech Signal

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Abstract— Digital filters are very effective in the present generation of communication systems. The digital filter performance is important and hence to design a digital finite impulse response (FIR) filter for reduction of noise from signal and improve the efficiency of the signal using different methods. A zero-valued window function is a mathematical function that is choosen outside some interval. The product is also zero-valued outside the interval, When another function is multiplied by a window function. The performance of different windows like Hamming, Hanning, Bartlett and Blackman window have been mainly compared with their magnitude response, phase response, equivalent noise bandwidth, response in time and frequency domain using MATLAB simulation. Comparing simulation results of different windows, it has been found that Blackman window is better than Hamming, Hanning, and Bartlett windows. These windows have also been encountered with speech signal using MATLAB simulation.

Key words: Introduction, FIR filter, Impulse response, Magnitude response, Equivalent noise bandwidth, Group delay, Periodogram.

I. INTRODUCTION

Digital filtering is one of the most important tools of Digital signal processing (DSP). Digital filters have ability of performing specifications which are extremely difficult, to achieve with an analog implementation. The digital systems eliminates unwanted noise, provide spectral shaping, or perform signal detection or analysis using signal filtering. A digital filter performs mathematical operations on a sampled, discrete-time signal to reduce or to improve certain aspects of that signal. Filtering is the fundamental aspect of digital signal processing. Digital filters are commonly used in digital signal processing applications, like digital signal filtering, noise filtering, signal frequency analysis, speech and audio compression [6]. Depending on impulse response, there are two fundamental types of digital filters: Infinite Impulse Response (IIR) filters, and Finite Impulse Response (FIR) filters. Digital filters with infinite duration impulse response referred to as IIR filters. Digital filters with finite duration impulse response referred to as FIR filters. On the other hand, FIR filters are non-recursive type filters where the present output depends on present input and past inputs. In a typical digital filtering application, software running on a DSP reads samples from an analog to digital converter [3]. Finite Impulse Response digital filter has strictly exact linear phase, relatively easy to design, highly stable, computationally intensive, less sensitive to finite word-length effects, arbitrary amplitude-frequency characteristic and real-time stable signal processing requirements etc. Thus, it is widely used in different digital signal processing applications[5].

II. FIR FILTER:

FIR (Finite impulse response) filters are most preferred filters that are implemented in software. In a typical digital filtering application, software running on a DSP (Digital signal processor) reads samples and performs the mathematical manipulations using analog to digital converter and digital to analog converter [3]. Linear Time Invariant Finite impulse response filters acts as the backbone to the DSP systems are the most widely used digital filter. There are two uses of these filters namely Signal restoration and Signal separation. When the signal has been distorted in some way the signal restoration is used. Signal separation is needed when the signal is contaminated with noise or interference of other signals. A finite impulse response (FIR) filter structure can be used to implement any digital frequency response. A FIR filter is implemented with a series of delays, multipliers, and adders to obtain the filter's output. Figure illustrates the basic block diagram of a FIR filter with N. The h values represents coefficients used for multiplication, so that the output is the summation of all the delayed samples multiplied by the appropriate coefficients at time n.The frequency response of Nth-order causal FIR filter is,

$$H\,(e^{jw}) = h(n)e^{-j\omega} \quad \text{…….(2.1)}$$

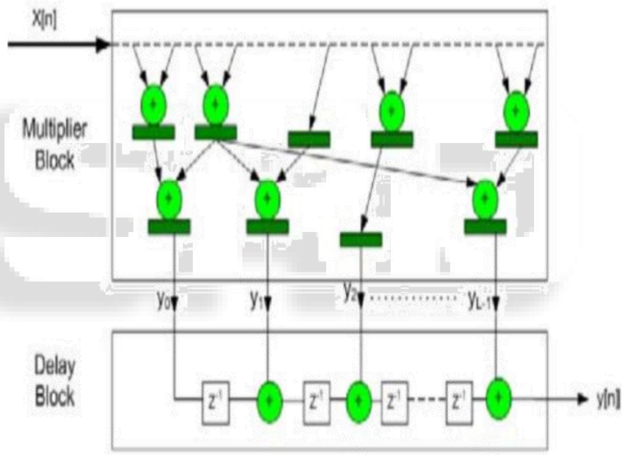


Fig. 1: FIR Filter Structure

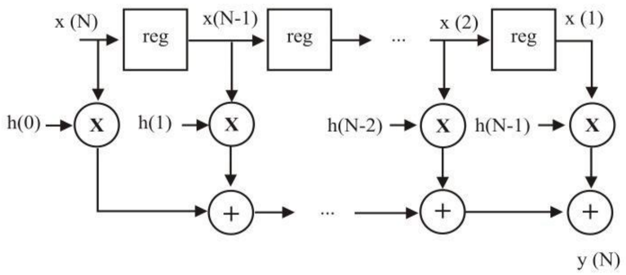


Fig. 2: Direct Form Realization Structure Of An FIR System.

There are four methods to design a FIR filter as Fourier series method, the widow method, the frequency sampling method and the optimal filter design method.

A. Fourier Series Method:

The design includes truncation of ideal symmetric impulse response. A symmetric impulse response gives a linear phase response. Truncation involves the use of window function. The main-lobe width determines transition characteristics .The side lobe level determines rejection characteristics [4].

B. Window Method:

The impulse response of ideal filter was determined by applying inverse Fourier transform to the ideal frequency characteristics of a digital filter. This infinite impulse response is multiplied by a finite length widow function to obtain impulse response of FIR filter. Different widow are used in different DSP applications like Hamming, Hanning , Kaiser, Blackman window.

C. Frequency Sampling Method:

The basic technique is to specify the desired magnitude of the frequency response. The unit sample response is then found by the inverse DFT. The troublesome point with this method is properly setting up the desired magnitude with even symmetry which is needed to ensure the resulting filter is casual, has a constant delay, and has real coefficients. If a zero phase shift was encountered for each then the resulting would be centered at the origin [4][7].

D. Optimal Filter Design Method:

In order to reduce the errors involved in frequency sampling method, this method is employed and is best method for designing of a FIR filter. The primitive idea of this method is to design fir coefficients again and again until a particular error is reduced. Using this method it is very complex to determine the filter coefficients. There are various methods to design FIR coefficients as follows:

(1) Least squared error frequency domain design
(2) Weighted Chebyshev approximation
(3) Nonlinear equation solution for maximal ripple FIR filters
(4) Polynomial interpolation solution for maximal ripple FIR filters [4]

III. FILTER DESIGN USING WINDOW TECHNIQUE:

In this method various window functions have been proposed. Listed below are some of the most commonly used fixed window:

- Rectangular window
- Hanning window
- Hamming window
- Blackman window

And below is adjustable window

- Kaiser window

So as to minimize the oscillations in Fourier series method,the fourier coefficients are modified by multiplying the infinite impulse response by finite weighing sequence w(n)called as a window. The two desirable characteristics of a window function is as follows-.

- Fourier transform of a window function should have main lobe with smaller width.
- Side lobes in the fourier transform of a window function should decrease in energy as or tends to 0.

The basic design principles of window function is to calculate h(n) by the inverse Fourier transform depending on the ideal desired filter frequency response $H_d(e^{j\omega})$ .The formula of h(n) is shows as[5]

$$h_d(n) = \frac{1}{2\pi}\int_{-\pi}^{\pi} H_d(e^{j\omega}) \quad ……(3.1)$$

As h(n) is infinitely long, we have to deal with it by using window function to get to the unit impulse response h (n). Now it is written as

$$h(n) = w(n).\, h(n) ……….(3.2)$$

Where w (n) is the window function.

A. Fixed window:

1) Hanning window:

The Hanning window is a raised cosine window which is used to minimize the side lobes, while preserving a good frequency resolution when compared to the rectangular window. The hanning window is given as

$$w(n) = \begin{cases} 0.5\left(1 - \cos\dfrac{2\pi n}{M-1}\right), & \text{for } n = 0 \text{ to } N-1 \\ 0, & \text{elsewhere} \end{cases} \quad ..(3.3)$$

2) Hamming window:

The hamming window is, similar to the Hanning window, with a raised cosine window. The hamming window posses similar characteristics as the Hanning window but further suppress the first side lobe. The hamming window is given as

$$w(n) = \begin{cases} \left(0.54 - 0.46\cos\left(\dfrac{2\pi n}{N}\right)\right), & \text{for } n = 0 \text{ to } N-1 \\ 0, & \text{elsewhere} \end{cases} \quad ....(3.4)$$

3) Blackman window:

The Blackman window is like both Hanning and Hamming windows. The Blackman window posses an advantage over other windows as it has better stop band attenuation with less pass band ripple. Blackman window is given as

$$w(n) = \begin{cases} \left(0.42 - 0.5\cos\left(\frac{2\pi n}{N}\right)\right) + 0.08\cos\left(\frac{4\pi n}{N}\right), & \text{for } n = 0 \text{ to } N-1 \\ 0, & \text{elsewhere} \end{cases} \quad ...(3.5)$$

There is a 58db difference among the dc magnitude and the largest side lobe. A low pass filter designed with Blackman window provides side lobes that are 74dB below the pass band gain.

B. Adjustable window:

1) Kaiser window :

The Kaiser window with parameter β is given as

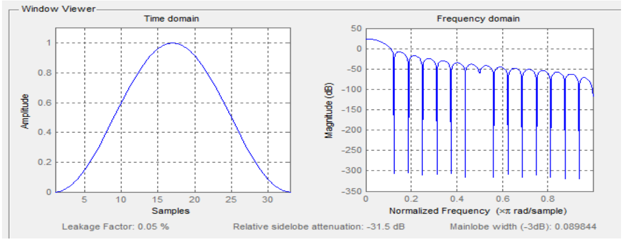
$$w(n) = \begin{cases} \dfrac{I_0\left(\beta\sqrt{1-(2(n+1)/(N+1))^2}\right)}{I_0}, & for\ n - 0 \text{ to } N-1 \\ 0, & elsewhere \end{cases} \quad ..(3.6)$$

The parameter β defines the shape of the window and hence controls the trade-off occuring between main-lobe width and side-lobe amplitude [5]. Kaiser window is widely used in practice. It is Zero-th order modified Bessel function of first kind.

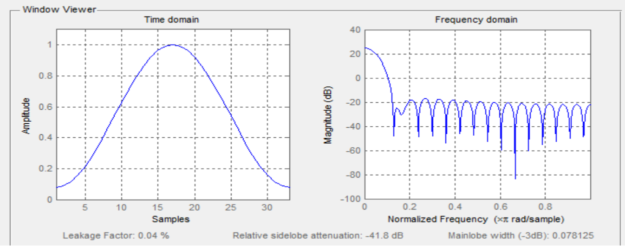
IV. DESIGN SPECIFICATION OF FILTER

Response type: Low Pass, High Pass
Design method: Window Functions
Order of Filter: 33
Sampling frequency: 8000Hz

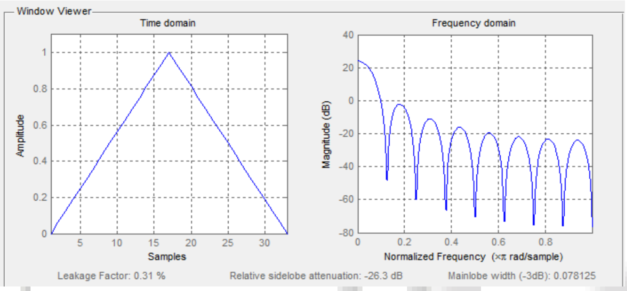
Cut Off frequency: 0.25
Input speech signal: 1000Hz



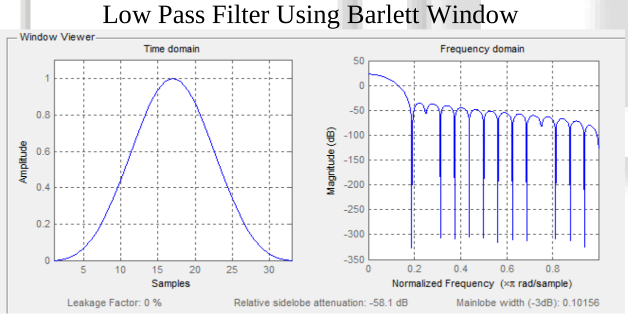
Low Pass Filter Using Hann Window



Low Pass Filter Using Hamming Window

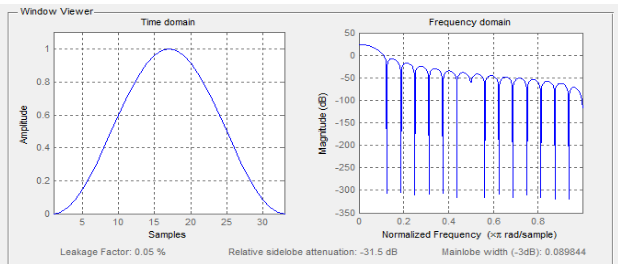


Low Pass Filter Using Barlett Window

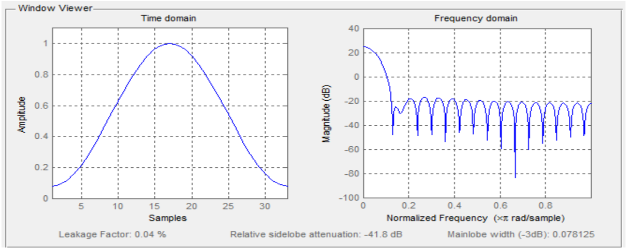


Low Pass Filter Using Blackman Window

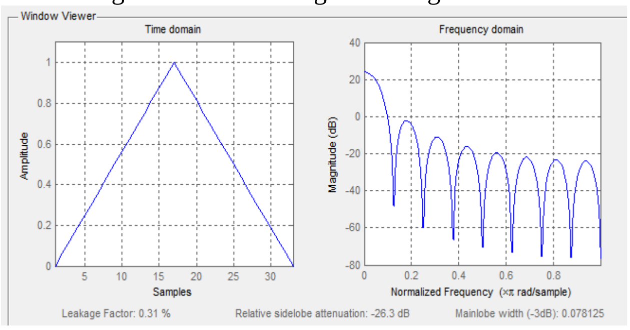
There is an extra cosine term in Blackman function. This extra term in terms reduces the side lobes. In FIR low pass filter design, there are many lobes exists in case of Hamming, Hanning and Bartlett window on the other hand for same specification and filter order, in Blackman window, there exists less side lobes. Reducing side lobes means that the efficiency is increased. This means less power is lost.



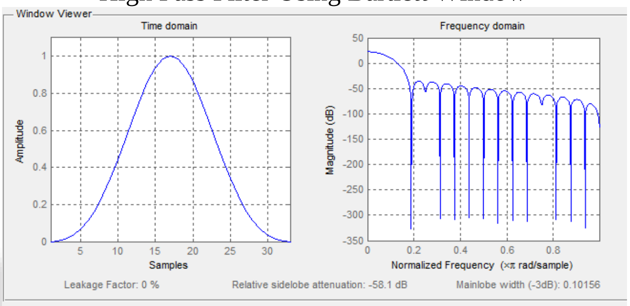
High Pass Filter Using Hann Window



High Pass Filter Using Hamming Window



High Pass Filter Using Bartlett Window



High Pass Filter Using Blackman Window

The comparison of design high-pass FIR filter using Hanning, Hamming, Bartlett and Blackman window of order 33. The normalized cutoff frequency is w_c=0.5. For length N=33 and normalized cutoff frequency w_c=0.5, this efficient window function provides higher side lobe attenuation compare to Hanning, Hamming and Bartlett windows however the main-lobe width is slightly increased.

V. GROUP DELAY:

Group Delay is defined as the negative slope of the phase response vs. frequency. At different frequencies in a system it is a measure of the relative delay from input to the output. Group delay is a measure of the time delay of the amplitude envelops of the various sinusoidal components of the signal. All frequency components of a signal are delayed when it is passed through a device like filters, an amplifier or a medium, like air. Group delay is a useful measure of time distortion and is calculated by differentiating with respect to frequency.

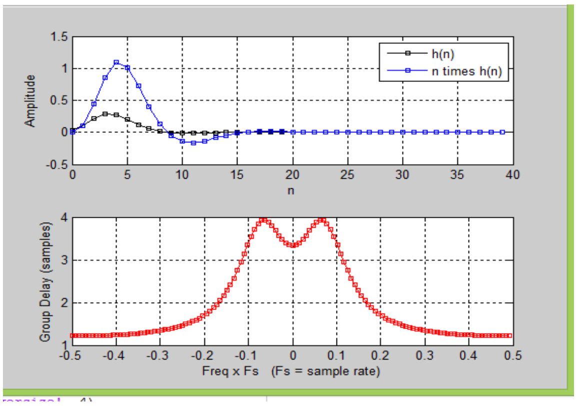


Fig. 3: Group Delay

VI. ANALYSIS OF PARAMETERS FOR EXISTING WINDOW FUNCTION

A. Power Spectral Analysis:

Power spectral analysis is most important application domain in Digital signal processing. There are broadly two types of power spectrum estimation (PSE) techniques: Parametric and Nonparametric. In Parametric or non-classical techiques an analyzed process is replaced by an appropriate model with known spectrum.

Non parametric method do not make any assumption on the data generating process [1]. It starts by computing autocorrelation sequence from a given data. The power spectrum is then estimated from FT of an computed autocorrelation sequence. Window function is mathematical function which has zero value outside some chosen period. When another function or waveform or data sequence is multiplied by a window function, the result is also zero-valued outside the period. It is simple to use and easy to understand. In this technique the frequency of a filter, $H_D(w)$ and the corresponding impulse response, $h_D(n)$,are related by the anti- Fourier transform:

$$h_D(n) = \frac{1}{2\pi}\int_{-\pi}^{\pi} H_D(\omega)e^{j\omega n}\,d\omega \quad \text{....(6.1)}$$

Where $H_D(w)$ is frequency response of a filter and $h_D(n)$ impulse response. The subscript D is employed to make out the difference ideal and practical responses.

VII. RESULTS AND DISCUSSION OF PERIODOGRAM AND WELCH METHOD

A. Periodogram:

A Periodogram is a method of estimating the spectral density of a given signal. A Periodogram manipulates the importance of different frequencies in time-series data to recognize any intrinsic periodic signals. Practically, the Periodogram is often computed by a finite –length digital sequence using the Fast Fourier Transform (FFT). The raw Periodogram is not a better spectral estimation because of spectral bias and with the fact that the variance at a given frequency does not reduce with computed increase in number of samples used. The spectral bias problem arises from a sharp truncation of the sequence, and can be minimized by first multiplying the finite sequence with window function which truncates the sequence gradually rather than abruptly [8].

B. Parametric Welch's Method

Welch's Method was proposed by Welch which is improved estimator of the PSD. This method involves dividing of time series data into segments, calculating a modified periodogram of each segment, and then averaging the PSD estimates. The result is Welch's PSD estimate. In the Signal Processing Toolbox the Welch's method is implemented by the pwelch function. By default, the data is divided into eight segments with 50% of overlapping among them. It is used for calculating the power of a signal at various frequencies. The technique depends upon the concept of using periodogram spectrum estimates, which are the result by converting a signal from the time domain into the frequency domain.

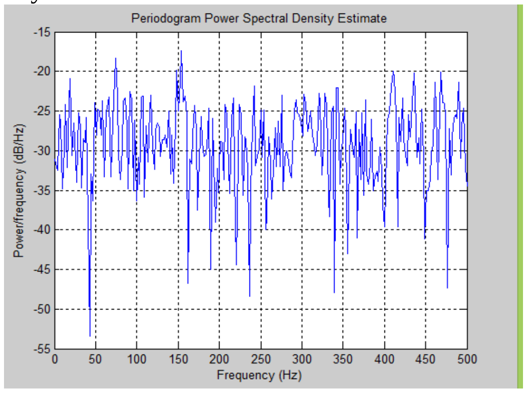


Fig. 4: Periodogram Power Spectral Density Estimation

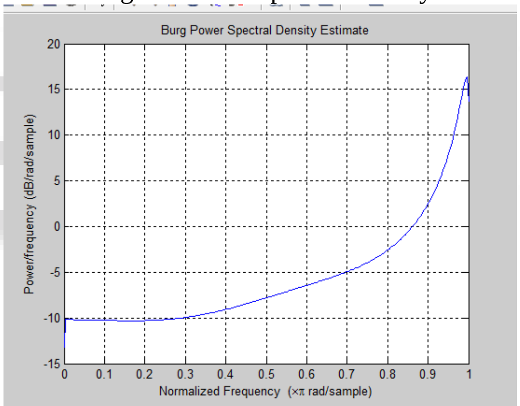


Fig. 5: Burg Power Spectral Density Estimate

It is encountered from the simulated result that Welch's Method provides better performance in terms of pass band power and Resolution when Compared to Periodogram Method.

VIII. CONCLUSION

According to the given specifications various entities like order of filter and the cutoff frequency can also be changed. Along with the mathematical comparison between Hamming, Hanning and Blackman window function, we have also encountered with the design of low pass and high pass FIR filter with a view to comparing their responses for various entities such as magnitude response, phase response, equivalent noise bandwidth and response in time and frequency domain using MATLAB simulation. In all cases, it has been found that Blackman window illustrating superiority in performance and demonstrating best functionality among these three windows.

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