JUNE, 2013 **Master Degree- Electrical and Electronics Engineering** **DERYA ÇEVİK TAŞDEMİR**

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GRADUATE SCHOOL OF NATURAL AND APPLIED SCIENCE

A NEW WINDOW FUNCTION FOR FIR FILTER DESIGN

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A New Window Function for FIR Filter Design

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M.Sc. Thesis

Supervisor

Prof. Dr. Arif NACAROĞLU

Derya ÇEVİK TAŞDEMİR

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Approval of the Graduate School of Natural and Applied Sciences

Assoc. Prof. Dr. Metin BEDİR
Dean of Institute

I certify that this thesis satisfies all the requirements as a thesis for the degree of M.Sc.

Prof. Dr. Celal KORAŞLI
Head of Department

This is to certify that I have read this thesis and that in my opinion it is fully adequate, in scope and quality, as a thesis for the degree of M.Sc.

A. Nacaroğlu

Prof. Dr. Arif NACAROĞLU
Supervisor

This is to certify that we have read this thesis and that in our opinion it is fully adequate, in scope and quality, as a thesis for the degree of M.Sc.

Examining Committee Members	Signature
Prof. Dr. L. Canan DÜLGER	L. Canan Dülger
Prof. Dr. Arif NACAROĞLU	A. Nacaroğlu
Asist. Prof. Dr. Tolgay KARA	....

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Derya ÇEVİK TAŞDEMİR

Signature

ABSTRACT

A NEW WINDOW FUNCTION FOR FIR FILTER DESIGN

ÇEVİK TAŞDEMİR, Derya
M.Sc. in Electrical and Electronics Engineering
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FIR (Finite Impulse Response) filter is an important research area in digital signal processing and filtering. FIR filters provide easily design at linear phase, which means delay the input signal but is delayed without distorting its phase. In addition to these advantages, FIR filters don't need to feedback. At FIR filters, to design a filter means to select the coefficients such that the system has specific characteristics. Generally, filter specifications refer to the frequency response of the filter. There are different methods to find the coefficients from frequency specifications. One of the important method is the series expansion and windowing. The window method for digital filter design is fast, convenient, and robust.

In this study, a new window function proposed to calculate the FIR filter coefficients obtained by using Fourier Series Method. In basic, proposed window is a modified version of Gaussian window function and Hamming window function. Combination of these windows has resulted with more parameters which means the controlling of window function is more flexible.

Keywords: Window functions, Gaussian window, Hamming window, Modified Gaussian window, Modified Hamming window, Finite Impulse Response (FIR) Filter design.

ÖZET

SONLU DARBE TEPKİLİ (SDT) SÜZGEÇ TASARIMI İÇİN YENİ BİR PENCERELEME FONKSİYONU

ÇEVİK TAŞDEMİR, Derya
Yüksek Lisans Tezi, Elektrik ve Elektronik Mühendisliği
Tez Yöneticisi: Prof. Dr. Arif NACAROĞLU
Haziran 2013, sayfa: 37

Sayısal filtre tiplerinden biri olan SDT Filtre (Sonlu Darbe Tepkili) sayısal veri işleme ve süzme alanında önemli bir araştırma alanıdır. SDT filtre lineer fazda tasarım kolaylığı sunar, şöyle ki veriyi süzerken, sinyali geciktirir fakat fazını değiştirmez . Bu avantajlarına ek olarak, SDT filtrelerin geri beslemesi yoktur. SDT filtrelerde, filtre tasarlamak sisteme has katsayı belirlemek anlamına gelir. Genellikle, filtre özellikleri filtrenin frekans tepkisini işaret eder. Frekans özelliklerinden gelen katsayıları bulmak için farklı yöntemler vardır. Seri açılımı ve pencereleme bu yöntemlerden önemli birtanesidir. Dijital filtre tasarımı için pencere yöntemi, hızlı, rahat ve güvenilirdir.

Bu çalışmada, SDT Sayısal Süzgeç Tasarımında Fourier Seriler kullanılarak elde edilen katsayıların pencerelenmesi için mevcut pencere fonksiyonlarına alternatif yeni bir pencere fonksiyonu önerilmiştir. Önerilen fonksiyon temelde Gauss ve Hamming Pencerelerinde çalışılarak elde edilmiş bir pencere fonksiyonudur. Bu iki fonksiyonunun birleştirilmesi ve etkili değiştirge sayısının arttırılması; ana kulak darlığının ve yan kulak büyüklüğünün daha fazla değişken üzerinden kontrolünü kolaylaştırmıştır.

Anahtar Kelimeler: Pencere fonksiyonları, Gaussian penceresi, Hamming penceresi, Modifiye edilmiş Gauss penceresi, Modifiye edilmiş Hamming penceresi, Sonlu Darbe Tepkili (SDT) Filtre Tasarımı.

To my family, who have created and maintained a wonderful life for me and contributed to my life with their lovely supports

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Lastly, I wish to special thank my parents

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NOMENCLATURE

n	Discrete time index
w	Angular frequency rad/sample
z	Index for z – frequency domain
T	Sampling period in rad/ sample
N	Window or filter length
S	Sidelobe roll-off ratio in dB
R	Ripple ratio in dB
$\vartheta(w)$	Phase spectrum
H(z)	Filter transfer function
H(e^{jwT})	Frequency spectrum of filter
W(e^{jwT})	Frequency spectrum of window
w_s	Sampling frequency
α, β, η	Parameters of Proposed window function
DSP	Digital Signal Processing
FIR	Finite Impulse Response
IIR	Infinite Impulse Response
CR	Contrast Ratio

CHAPTER 1

INTRODUCTION

1.1 Background

In digital signal processing applications, digital filter [1] is the most important and frequently used element, which is classified as finite impulse response (FIR) [1,2] and infinite impulse response (IIR) [1, 3] filters by the duration of their impulse response. Both of FIR and IIR filters have many advantages and disadvantages, due to they are not best for all situations.

FIR filters can be designed as stable and exact linear phase, due to they are commonly used filters. A drawback of nonrecursive filters comparison to recursive filters is their application complexity in case the filter order is large. The application of FIR filters can be done using either recursive or nonrecursive techniques, however a nonrecursive application guarantees a stable filter.

There are many methods to design nonrecursive digital filters such as Fourier series method, discrete Fourier transform, optimization methods and numerical methods. Though optimum design can be reached by using the optimization method, a large computation is needed and this makes method of optimization unappropriate for real time applications. The other side, method of Fourier series [1, 4] with windowing is the most appropriate way to design nonrecursive filters with minimum computation to another methods.

In Fourier series methods, the aim of using window function is to smooth and truncate the infinite duration ideal filter's impulse response.

1.2 Problem Definition

There are many windows proposed in literature better nonrecursive digital filters. They are classified as adjustable or fixed according to having number of independent parameters in their function. Due to their flexibility property, the adjustable windows are preferred for practical applications. There are many windows in literature but the problem is that the filters designed by the windows are suboptimal. For example, the filter order required to satisfy a given prescribed specifications is not the lowest and, a higher degree of filter means more calculations, more components, etc. Hereby, the researches focused on providing new windows to develope the filter characteristics.

1.3 Thesis Objective

The objective of this thesis research is to find a new window function which enables the filter designer to provide a high quality nonrecursive filter. A new window function proposed to calculate the FIR filter coefficients obtained by using Fourier Series Method.

Basically, proposed window is a modified version of Gaussian window function and Hamming window function. This new window function presents better performance side lobes and main lobe.

1.4 Literature Summary

Direct truncation of Fourier series causes the oscillations, and these oscillations were first explained mathematician by Gibbs [5] in 1899 and therefore called Gibbs' Oscillations.

Fejer has dealed with reducing the effects of Gibbs' Oscillations for practical applications in 1900. Then, Lanczos has proposed better approach than Fejer in 1956. These methods considered a function with only one jump discontinuity, but Gibbs' oscillations are characteristics of any truncated Fourier series regardless of the number of discontinuities.

Window functions are used for better smooting operations. The simplest one is rectangular window. Hann, Gaussian, Dolph-Chebysev, Blackman , Hamming, Kaiser, Bartlett window functions have different features.

1.5 Structure of Thesis

The thesis is organized in five chapters, which can be described as follows:

Chapter 1. Introduction – This chapter gives an overview of thesis work, problem definition, research objective, prior works and thesis structure.

Chapter 2. Review of Digital Filters – Explained digital filters and some background information about digital filter types and comparasion of recursive and nonrecursive digital filters.

Chapter 3. Windowing in Finite Impulse Response Filter Design –The use of Fourier series method in the design of nonrecursive digital filters and then nonrecursive digital filters are explained clearly.

Chapter 4. Proposed Window Functions –Thesis study presents a new window function with their application in the design of nonrecursive digital filters.

Chapter 5. Conclusion and Future Works –A brief summary for the results of thesis work is presented.

CHAPTER 2

REVIEW OF DIGITAL FILTERS

2.1 Digital Filters

Filters are circuits or devices such that their output gain and/or phase vary as a function of the frequency of the input. Due to having this frequency sensivity, they are used to pass signals at selected frequencies and reject signals at other frequencies.

In terms of their operations, the filters can mainly be classified as

- Lowpass filters
- Highpass filters
- Bandpass filters
- Bandstop filters

The filters are categorized as analog or digital according to their physical make up and how they work. In an analog filter, analog electronic circuits which may be made up for passive and active components such as resistors, capacitors, operational amplifiers (OPAMP) or operational transconductance amplifiers (OTA) are used in order to produce the required filtering effect [6, 7]. There are well established standart techniques to design an analog filter circuit for a given requirement. The signal being filtered in analog filters is an electrical voltage or current, which is the direct analogue of the physical quantity such as a sound signal or transducer output.

As for a digital filter, it uses a digital processor such as a personal computer or special digital signal processing chip to perform the filtering effect [8]. Since the operation is based on numerical calculations on sampled values of the signal, the analog input signal must first be sampled and digitized using an analog to digital converter. The resulting binary numbers representing the input signal are transferred to the processors, which carry out numerical calculations on them. Note that the signal in a digital filter is represented by a sequence of numbers rather than a voltage or current.

Main advantages of digital filters over analog filters can be listed as

- Digital filters are programmable.
- Digital filters characteristics are extremely stable to the temprature compared to analog filters.
- Digital filters are easily designed, tested and implemented on a processor [9].

2.2 Types of Digital Filters

According to their implementations, the digital filters are classified as finite impulse response filters (FIR) and infinite impulse response filters (IIR) [10].

2.2.1 Finite Impulse Response (FIR) Filter Design Methods

Direct methods are used for design of FIR digital filters. Four well known methods are used. These methods are:

- Fourier series method using the window method – It provides closed form solutions and, as a result, it is easy to apply and involves only a minimal amount of computation. Unfortunately, the designs are suboptimal with respect to filter complexity whereby a filter design is said to be optimal if the filter order is the lowest that can be achieved for the required specifications [11].
- Numerical methods – It uses numerical formulas to design the FIR filters can perform interpolation, differentiation or integration. The most commonly used interpolation formulas are the Gregory-Newton, Bessel, Everret, Stirling and Gauss interpolation formulas.

- Optimization methods – It provides optimal solutions for the design of nonrecursive filters. But it perform this work at the expence of a large amount of computation. The basic idea in the optimization methods is to find the filter coefficients until the particular error is minimized.

- Discrete Fourier transform method - Unlike the Fourier series method with the window method, can be used for any given magnitude response. It is useful for the design of non-prototype filters where the desired magnitude response can take any irregular shape.

2.2.2 Infinite Impulse Response (FIR) Filter Design Methods

The design methods for recursive filters can be classified as indirect or direct. Indirect design approaches are based on deriving the discrete-time transfer function from a continuous-time transfer function, i.e, converting analog filter into a digital filter. However, direct design methods generate the discrete-time transfer function directly in z-domain.

In the indirect approach, first a continuous-time transfer function that satisfies certain specifications is obtained using one of the standard analog filter approximations such as Butterworth and Chebysev. Then a corresponding discrete time transfer function is obtained using one of the following methods.

- Invariant impulse response method
- Modified invariant impulse response method
- Matched z transformation
- Bilinear transformation

2.3 Comparison of FIR and IIR Digital Filters

The main advantages of FIR digital filters over the IIR digital filters can be listed as follows:

- They are simple to design.
- They are guaranteed to be stable.
- They can be guaranteed to have perfect linear phase. This is a desirable property for many applications such as music and video processing.
- They have a low sensivity to filter coefficient quantization errors. This is a desirable property to have when implementing a filter on a DSP processor or on an integrated circuit.

The main disadvantages of FIR filters over IIR filters can be listed as follows:

- They require a higher order to perform the filtering. Higher order means more memory, more power and more processing time [12].

CHAPTER 3

WINDOWING IN FINITE IMPULSE RESPONSE FILTER DESIGN

3.1 Filter Design Using The Fourier Series

The idea of using Fourier series in the design of filters comes from the fact that the frequency response of a FIR filter is a periodic function of w_s with period w_s. Therefore, applying Fourier series for the frequency-domain representation of filters, it can be written as

$$H(e^{jwT})= \sum_{n=-\infty}^{n=\infty} h(nT)\, e^{-jwnT} \tag{3.1a}$$

where

$$h(nT)=\frac{1}{w_s}\int_{-w_s/1}^{w_s/1} H(e^{jwT})e^{jwnT}dw \ \ and\ w_s=\frac{2\pi}{T} \tag{3.1b}$$

By letting e^{jwT}=z in Eq.(3.1a), the transfer function can be obtained as

$$H(\mathrm{z})= \sum_{n=-\infty}^{n=\infty} h(nT)\, e^{-z} \tag{3.2}$$

For a given frequency spectrum, H(e^{jwT}), a corresponding transfer function can be obtained using Eq.(3.1b) and Eq.(3.2). But, the obtained transfer function becomes noncasual and has an infinite order. In order to provide a finite order transfer function, the series in Eq.(3.2) can be truncated as

$$H(\mathrm{z})= \sum_{n=-(N-1)/2}^{n=(N-1)/2} h(nT)\, z^{-n} \tag{3.3}$$

where $h(nT)$=0 is defined for the range $|n| > (N-1)/2$

By expanding Eq.(3.3), it can be written as

$$H(z) = h(0) + \sum_{n=1}^{n=(N-1)/2} [h(-nT)\, z^{n} + h(nT)\, z^{-n}] \tag{3.4}$$

In order to make the transfer function casual, H(z) is multiplied by $z^{-(N-1)/2}$ .This multiplication changes the group delay by an amount of $(N-1)T / 2$, but it doesn't affect the amplitude response of the obtained filter. [13]

3.2 FIR Filter Design Using the Windowing Method

The windowing method using the window functions is applied for reducing undesired oscillations.[14] At this stage, the design of FIR filters using the window functions involves four general steps as follows:

- Firstly, an idealized frequency response is assumed. Then using Eq.(3.1) an idealized infinite-duration noncasual filter is obtained.
- To achieve the desired filter specifications, a suitable window is selected.
- The window function is constructed and applied.
- At last, the resulting finite-duration noncasual filter is converted into casual filter.

3.3 Windowing Functions

The windowing method is used to reduce undesired oscillations resulting from the truncation of a Fourier series by using a class of time-domain functions – known as window functions. Good windows have the following properties:

- The narrow mainlobe
- The small ripple ratio

- The large sidelobe roll-off ratio [15].

3.4 Well-Known Windows in Literature

There are many windows which have been proposed in the literature, they are classified as fixed or adjustable according to having number of independent parameters in their functions. Fixed windows have only one parameter, namely, the window length which controls the mainlobe width. Adjustable windows have two or more independent parameters [16, 17].

3.4.1 Fixed Windows

The most frequently used fixed windows are Rectangular, Triangular, Hanning, Hamming and Blackman. They are used for simple signal processing applications.

3.4.1.1 Rectangular Window

The rectangular window is the simplest window, as the name implies, it has a rectangular shape. It is defined as

$$w_R(n) \triangleq \begin{cases} 1, & |n| \leq \frac{M-1}{2} \\ 0, & \text{otherwise} \end{cases} \tag{3.5}$$

Figure 3.1 shows the spectral graph of rectangular window function. It is seen that, the rectangular window has the narrow mainlobe width, but also it has large ripple ratio.

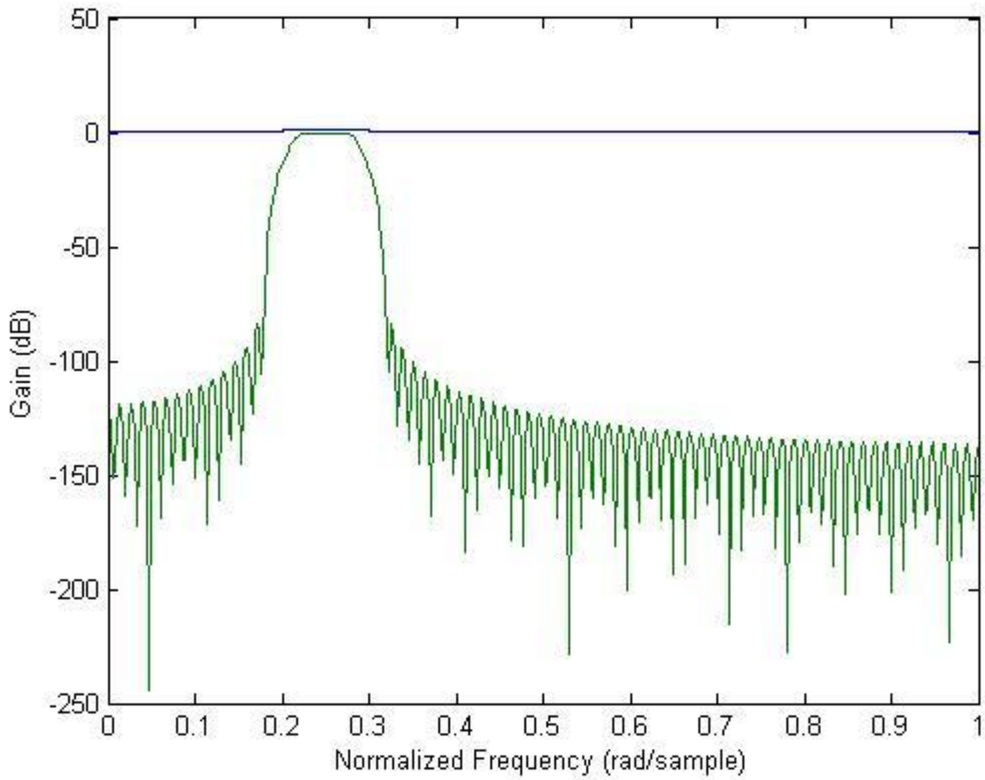

Figure 3.1 Rectangular window function for N=150

3.4.1.2 Hann (Von Hann) Window

Hanning window (or also known as Von Hann) is proposed better ripple ratio than r

Rectangular window, but its mainlobe is nearly two times wider and defined by

$$w_0(n) = 0.5 \left(1 + \cos\left(\frac{2\pi n}{N-1}\right)\right) \quad (3.8)$$

Figure 3.2 shows the spectral graph of Von Hann window function. It is seen that, the Vonn Hann window has better ripple ratio than the Rectangular window function, but its mainlobe width is two times wider.

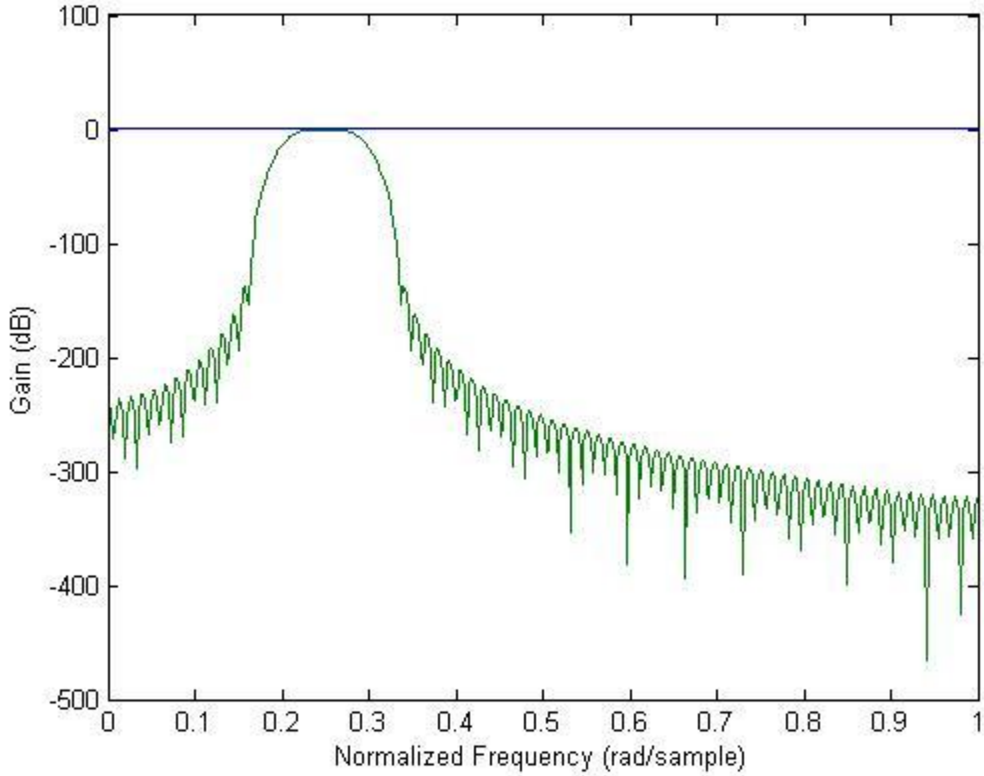

Figure 3.2 Von Hann window function for N=150

3.4.1.3 Hamming Window

The window is optimized to minimize the maximum (nearest) side lobe, it is similar to Hann window.

$$w(n) = \alpha - \beta \ \cos\left(\frac{2\pi n}{N-1}\right), \tag{3.9}$$

with

$$\alpha = 0.54,\ \beta = 1 - \alpha = 0.46,$$

Figure 3.3 shows the spectral graph of Hamming window function. It provides better ripple ratio than the Von Hann window function, and their mainlobe widths are almost the same.

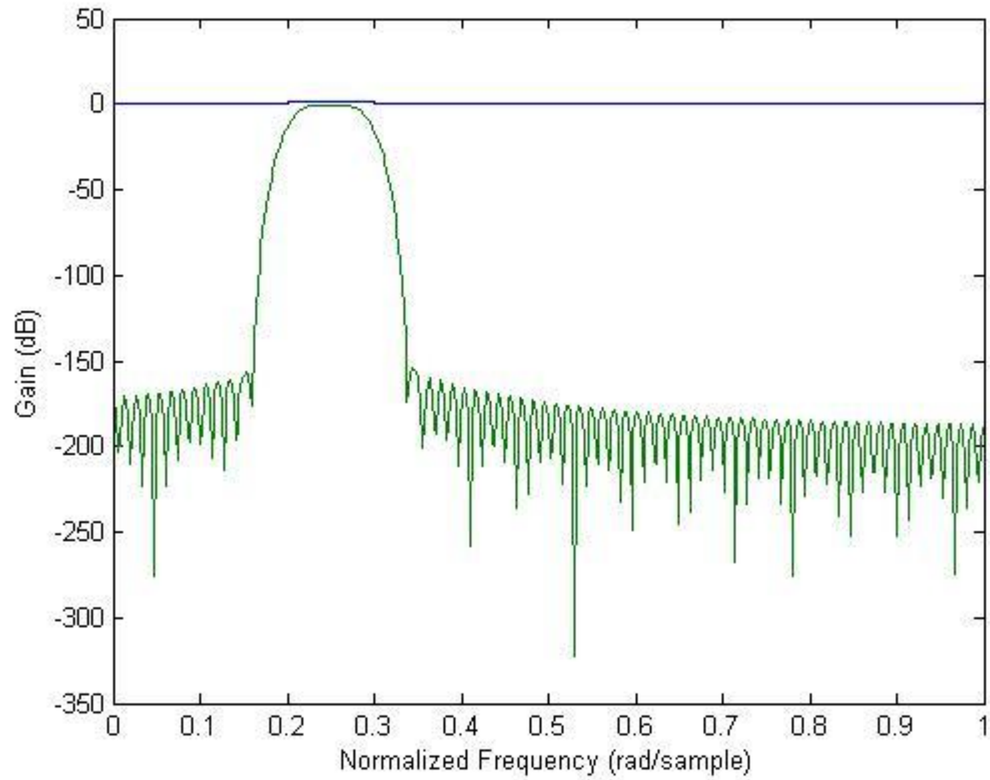

Figure 3.3 Hamming window function for N=150

3.4.1.4 Blackman Window

The Blackman window has an additional cosine term compared to the Von Hann and Hamming windows in order to have a better ripple ratio. It is defined as

$$w(n) = a_0 - a_1 \cos\left(\frac{2\pi n}{N-1}\right) + a_2 \cos\left(\frac{4\pi n}{N-1}\right) \tag{3.10}$$

$$a_0 = \frac{1-\alpha}{2}; \quad a_1 = \frac{1}{2}; \quad a_2 = \frac{\alpha}{2}$$

By common convention, the unqualified term Blackman window refers to $\alpha = 0.16$.

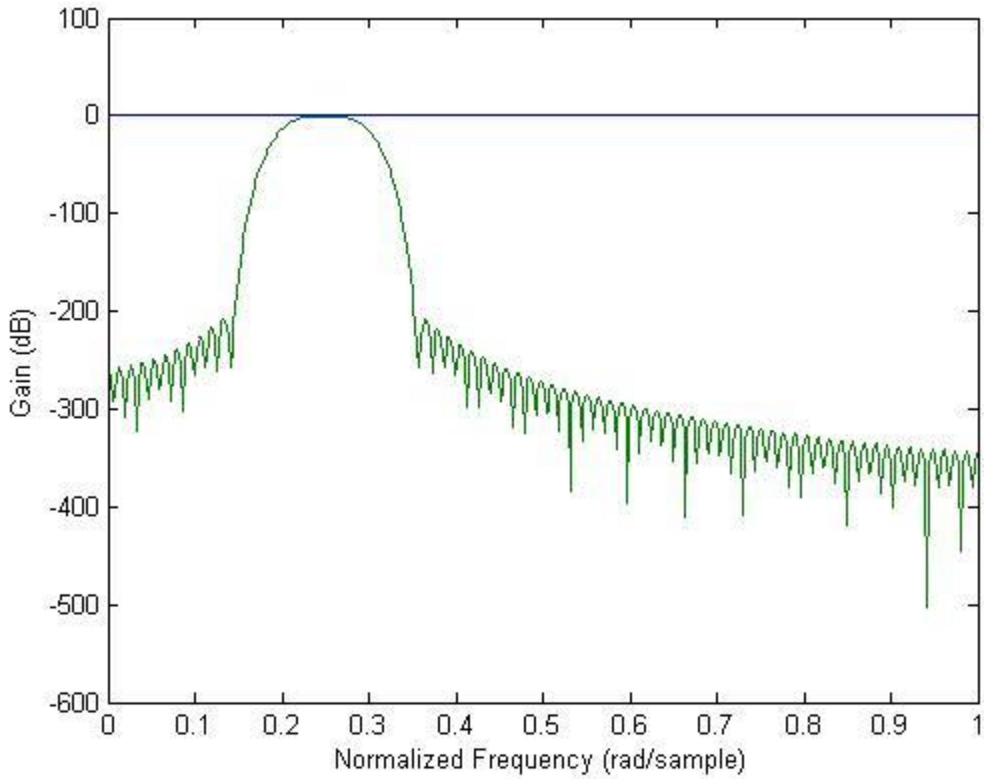

Figure 3.4 Blackman window function for N=150

Figure 3.4 shows the spectral graph of Blackman window function, which has better ripple ratio than the previous fixed window function at the expense of having widest mainlobe width.

3.4.2 Adjustable windows

The most frequently used adjustable windows are Gaussian, Dolph-chebysev, Kaiser. They have two independent parameters

3.4.2.1 Gaussian window

Gaussian's Fourier transform is also Fourier transform's eigen function. Due to the Gaussian function goes to infinity, which must either be truncated at the window's end, or itself windowed with another zero-lasted window [1, 18].

Since Gaussian function's logarithm generates a parabola. In frequency estimation, this could be used for exact quadratic interpolation.

The Gaussian window function is given in Eq.3.11

$$w(n) = e^{-\frac{1}{2}\left(\frac{n-(N-1)/2}{\sigma(N-1)/2}\right)^2} \quad (3.11)$$

where,

$\sigma \leq 0.5$

Figure 3.5 shows the spectral graph of Gaussian window function, which has optium mainlobe and ripple ratio, when criticise in well known literature window function.

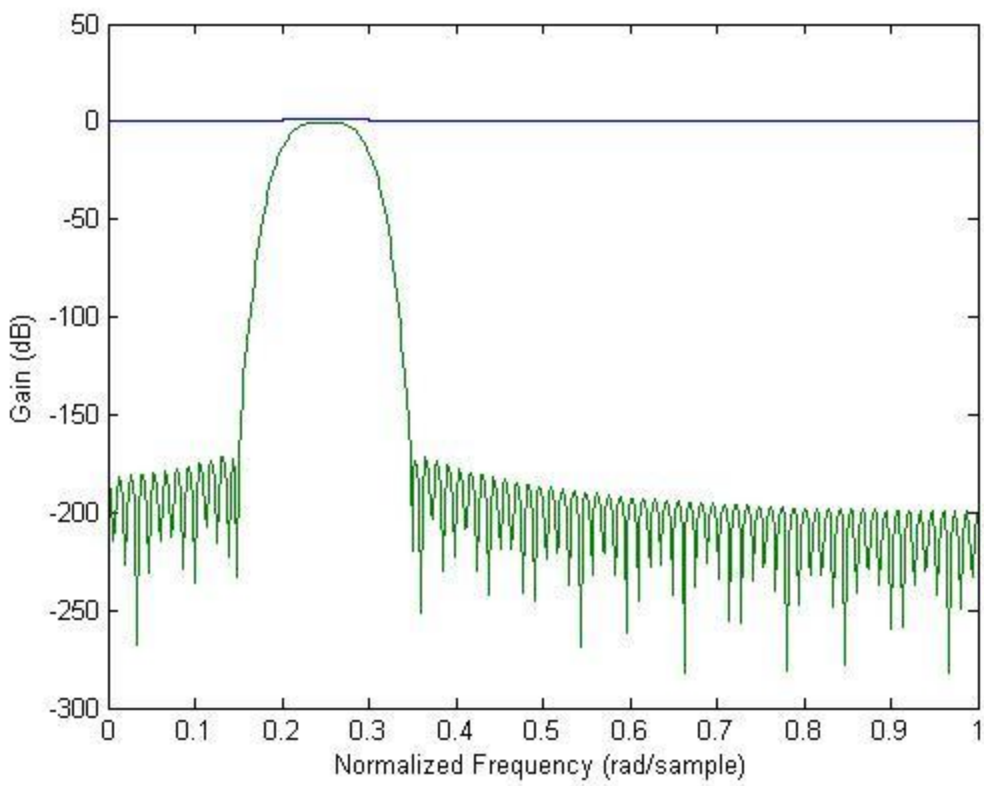

Figure 3.5 Gaussian window function for N=150

3.4.2.2 Dolph–Chebyshev window

The Dolph–Chebyshev window function $w_0(n)$ is illustrated in terms of its real valued discrete Fourier transform, $W_0(k)$:

$$W_0(k) = \frac{\cos\{N\cos^{-1}[\beta\cos(\frac{\pi k}{N})]\}}{\cosh[N\cosh^{-1}(\beta)]}$$
$$\beta = \cosh[\frac{1}{N}\cosh^{-1}(10^{\alpha})] \tag{3.12a}$$

where defined as,

$\alpha = 5$; $B = 1.94$.

The window function can be computed from $W_0(k)$ through discrete inverse Fourier transform:

$$w_0(n) = \frac{1}{N}\sum_{k=0}^{N-1} W_0(k)\cdot e^{i2\pi kn/N} \tag{3.12b}$$

The window's lagged version, with $0 \le n \le N-1$, could be obtained via:

$$w(n) = w_0\left(n - \frac{N-1}{2}\right). \tag{3.12c}$$

Figure 3.6 shows the spectral graph of Dolph- Chebyshev window function.

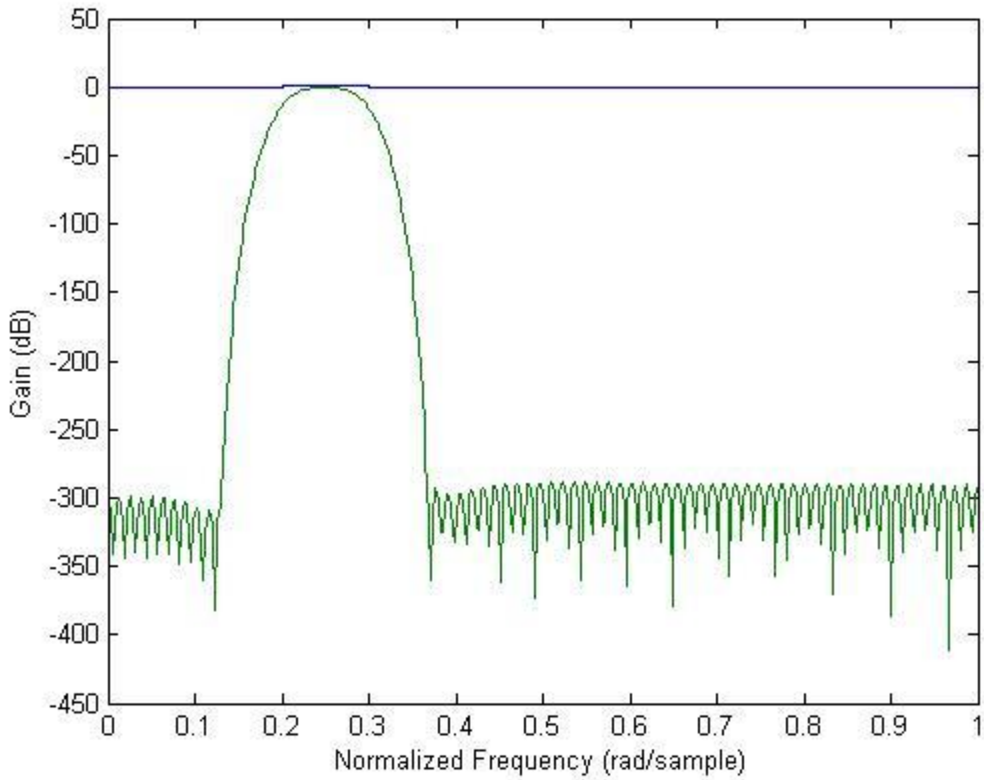

Figure 3.6 Dolph–Chebyshev window function for N=150

3.4.2.3 Kaiser window

The Kaiser window is discovered by Kaiser. It has two independent parameters, namely the window length and the adjustable shape parameter α [19, 20]. The Kaiser window is defined by,

$$w_0(n) = \frac{I_0\left(\pi\alpha\sqrt{1-(\frac{2n}{N-1})^2}\right)}{I_0(\pi\alpha)} \tag{3.13}$$

where α is adjustable parameter, and I_0 is the modified Bessel function of the first kind of zero [21].

Figure 3.7 shows the spectral graph of Kaiser window function.

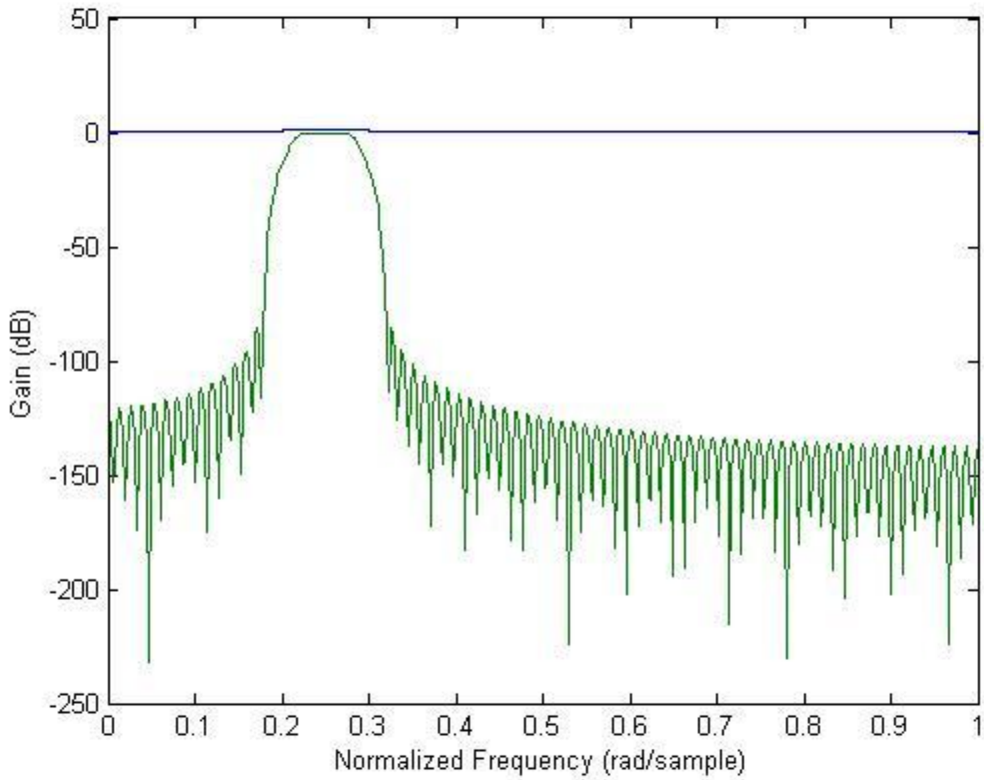

Figure 3.7 Kaiser window function for N=150

It is clearly seen that Gaussian window and Hamming window have good performance on side lobes and main lobe. And, alternatively, the Gaussian window and Hamming window gives reduced passband ripple and increased attenuation relative to other windows for a fixed filter length. [22, 23]

CHAPTER 4

PROPOSED WINDOW FUNCTION

4.1 Introduction

Window functions are commonly used at design of digital system design. In basic, there are two method in digital filter design. Using of Fourier Series Method is a basic approach contrary to method of digital approach filter type, known as Finite Impulse Response (FIR) filter design and Infinite Impulse Response (IIR) filter design [13, 15].

Specially at FIR filters, obtaining maximum gain of lobes, not only designing high degree of filters but also related to design of more suitable window function for same degree (N) [13].

Mainly, probability of approaching an ideal case increase at window functions with using infinite Fourier coefficients as changing coefficient. Specially, spectrums are expected to close impulse function with changing of functions' coefficients. At this thesis shown that, as a result of combination of Gauss and Hamming window functions a new window function proposed and a new window function's spectrum is more similar to impulse function.

A proposed new window function is tested on a lot of numeric sample and with comparing gain lobes, window function's efficiency is discussed.

4.2 Proposed Window Function

At time domain, mainly combination of Gauss window function and Hamming window function, which is developed version of rectangular window, a proposed window function is obtained. Hamming window function is a type of fixed window, explained in section 3.4.1.4. Another window is Gaussian window function, which is a type of adjustable window, explained in 3.4.2.1.

When a proposed window function is discovered, set out from windows' spectrum structure. Mainly, using in suitable form of Gaussian and Hamming window function, denoted in Equation 3.9 (Hamming window) and Equation 3.11 (Gaussian window), it is aimed to reach better result, due to proposed window function reached a good result [13].

$$w(n) = \cos(\pi n / N)^{\eta} * \alpha^{\beta (n/N)^2} \tag{4.2.1}$$

Here, N-1 defined as window length and value of *n* change -N/2≤ n≤ N/2 between as integer. Presented in Eq.4.2.1 proposed window function analysed at different filter types. Comes from 3 variable proposed window function has better spectrum; narrower mainlobe and lower sidelobes.

4.3 Well Known Window Functions' Spectrums on Ideal Band Pass Filter

This part, three variable proposed window functions' results of efficiency, defined in Eq. 4.2.1, aimed to discuss with constant window length and different filter types and different variables.

In proposed window function the modified model of Hamming and Gaussian window functions are used by changing their coefficients. In this way, the function is formed by three variables. Mainlobe and side lobes can be controlled by independent to each others.

In thesis, the quality of proposed window function applying results of not only function's spectrum but also analysed at ideal filter samples to compare well known window functions, which are Blackman window function, Hamming window function, Kaiser window function, Bartlett window function and Gauss window function.

With changing different interval of known as three values α, β, η observed to controllable changings at every three different type of ideal filter characteristic. A lot of different window functions are tested at ideal band pass filter and between this functions, it is seen that, Hamming window function gives better results. In the light of the results of working in η (Denoted in Eq. (4.2.1)) chosen at optimum mainlobe. Same filter, at same window length is tested by using the proposed conventional window functions.

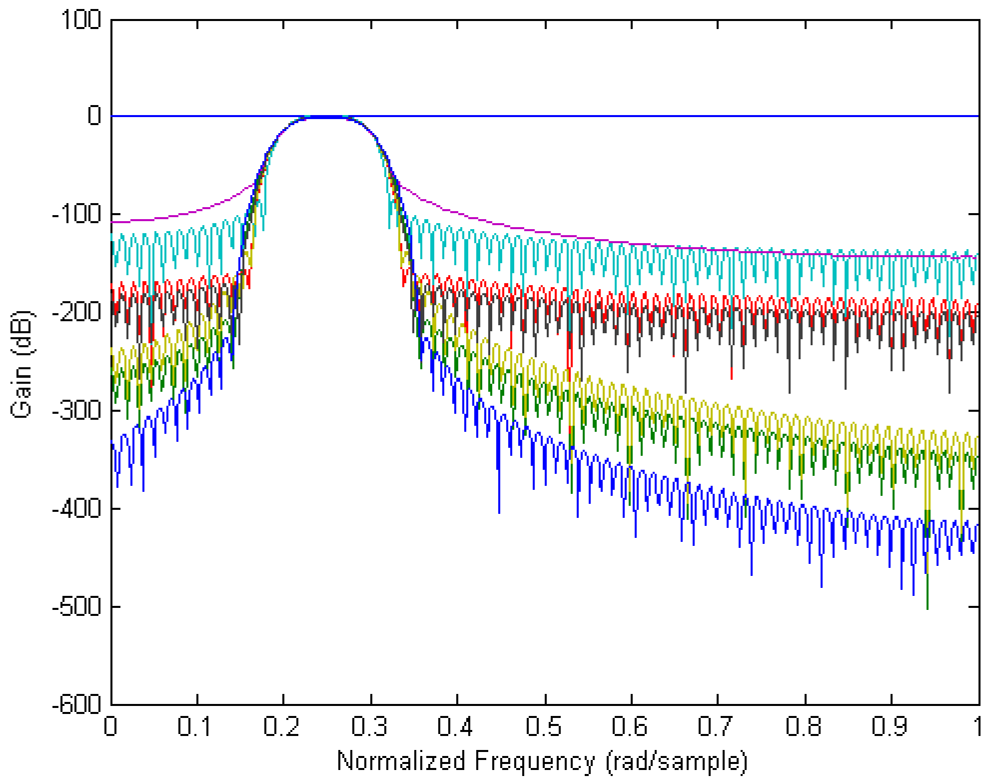

Figure 4.1 Comparison of proposed window function and well-known window function

The spectrums of Blackman window function (green), Hamming window function (red), Kaiser window function (light blue), Bartlett window function (violet), Gauss window function (black) are given in Fig. 4.1. Vertical axis points the gain (dB), horizontal axis points the normalized frequency (rad/sample). In this study, proposed window function is tested for different ideal filter gain lobes and reached similar results. For example, in Fig.4.1, the passband frequency is taken as 1/8 of sampling frequency, the bandwidth is equated to 1/20 of sampling frequency and the window length of band pass filter is set to 151.

Change of α, β and η in fixed values lead to better performance on mainlobe and sidelobes of bandpass filter. To watch over these changings, the conditions of Fig. 4.1. are taken on base values η =2.4, α = 2, β = 8. This numeric values determined using of Tuning method. In other words, using of 'for loop' in Matlab, η taken 1 to 10 periodically increased 0.01, and then doing spectrum analysed and then determined η,likewise α and β.

For example, for same condition used η = 2.4 in Fig. 4.2. and η = 3 in Fig. 4.3. For η = 3, show better result at sidelobes,worse results at mainlobe to the η = 2.4. Discussion of these results η determined as 2.4, likewise α and β. Thus, these values on window function show better performance to the well known window functions.

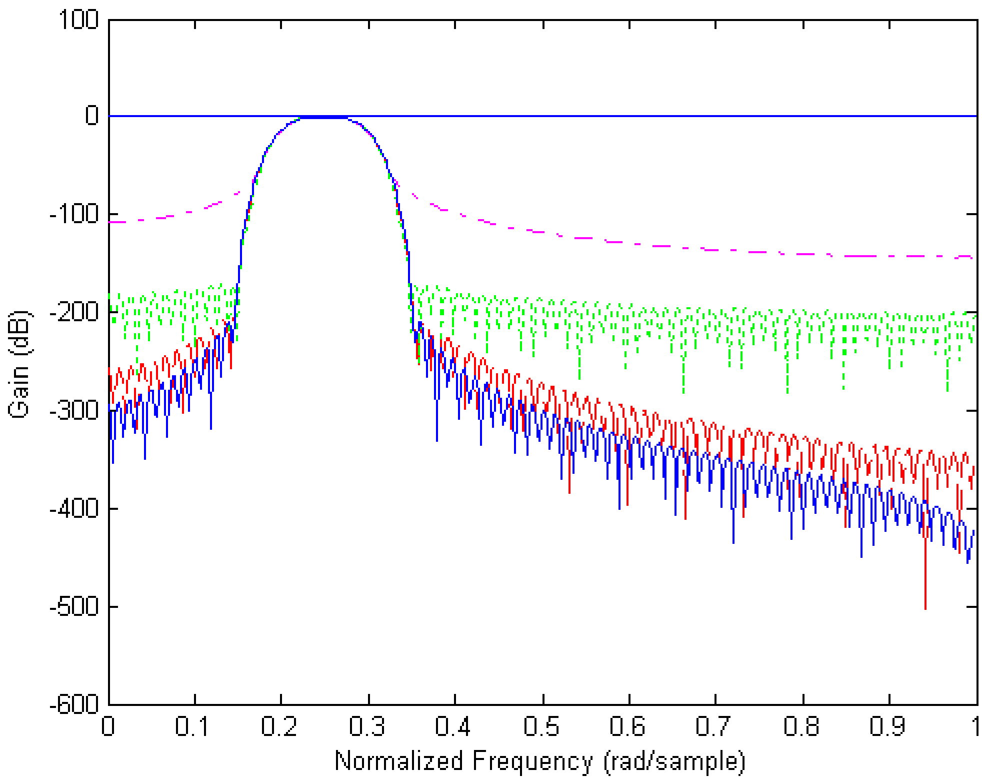

Figure 4.2 Comparison with Proposed window function and well known window function for η = 2.4 on bandpass filter

In Figure 4.2 and 4.3, Bartlett window function is plotted with -.-.-.-. and violet line, Gauss window function is plotted with …… and green line , Blackman window function is plotted with ---- and red line, Proposed window function is plotted with ___ and blue line.

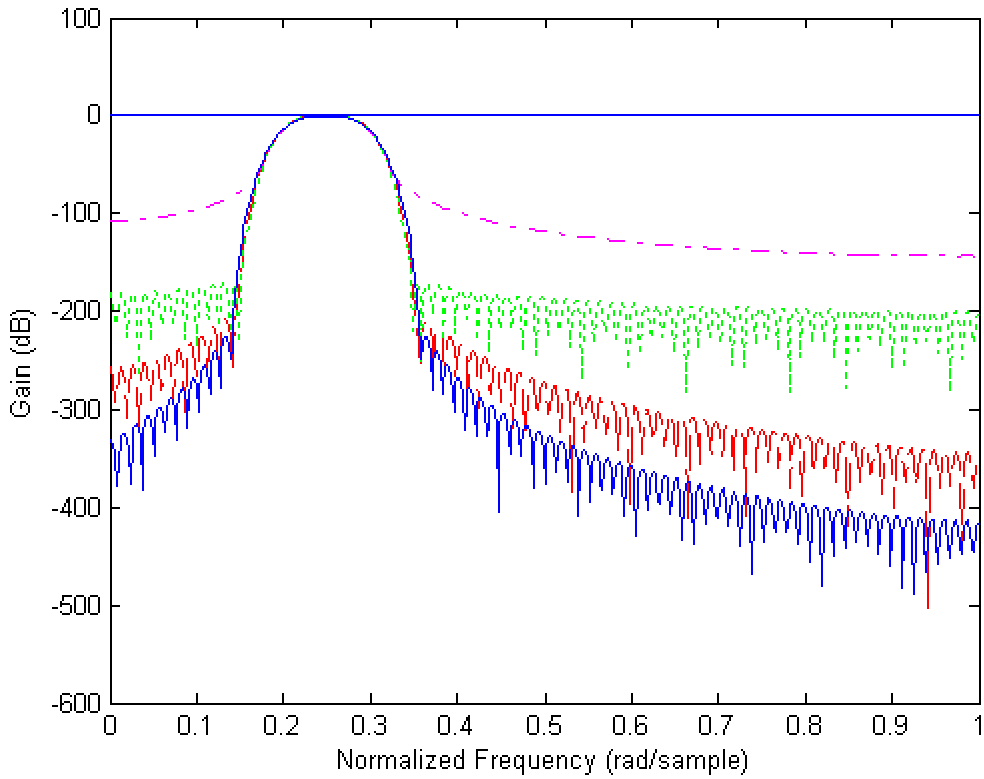

Figure 4.3 Comparison with Proposed window function and well known window function for η = 3 on bandpass filter

4.4 Using of Different Window Length on Proposed Window Function

Aim of proposed window function, the efficiency is taken as optimum values for α =2, β = 8 and η= 2.4, compared to the well known window function in literature at different window length. Results are shown at Fig. 4.4. Fig. 4.5. and Fig.4.6.

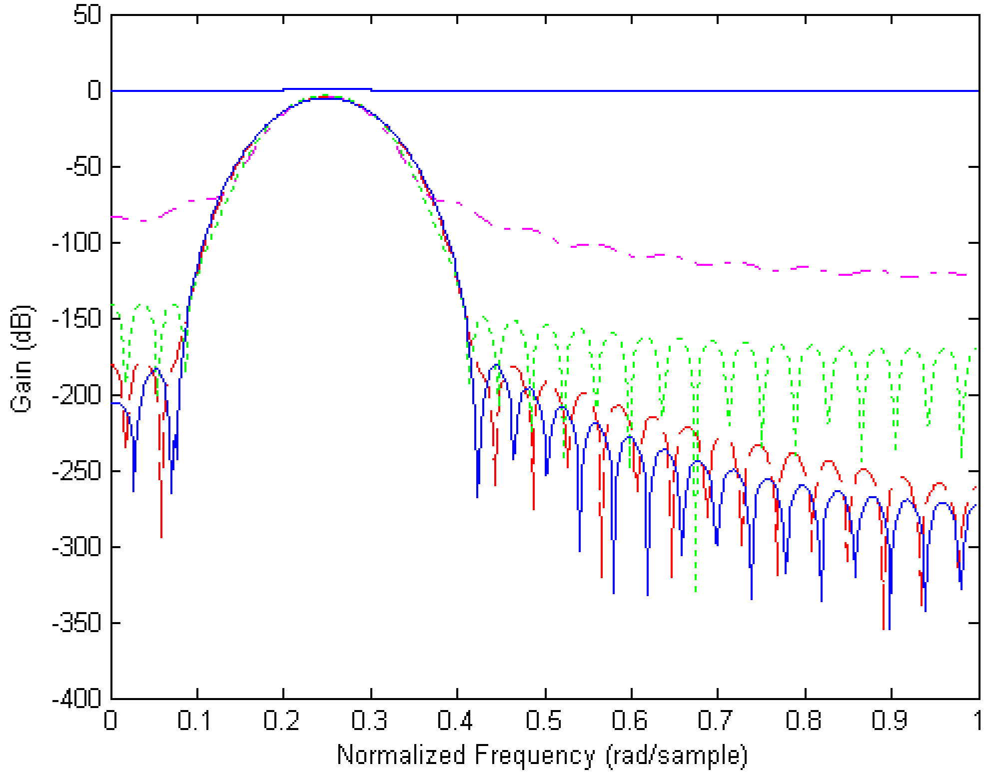

Figure 4.4 Comparison with Proposed window function and well known window functions for window length (N+1) = 51 on bandpass filter

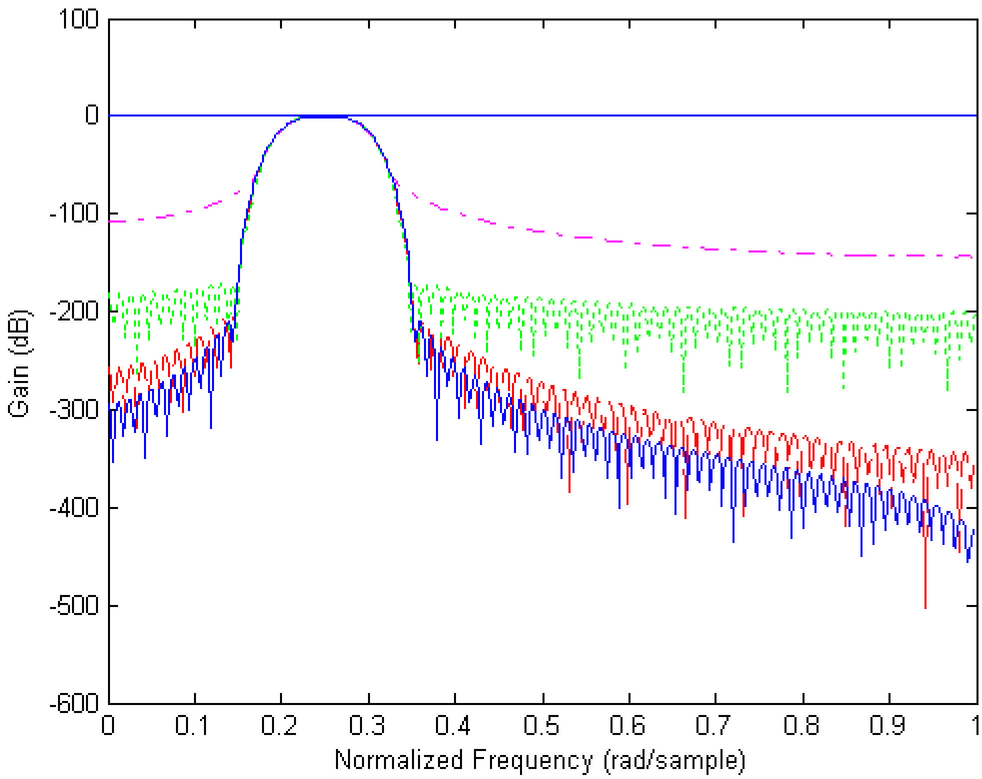

Figure 4.5 Comparison with Proposed window function and well known window functions for window length (N+1) = 151 on bandpass filter

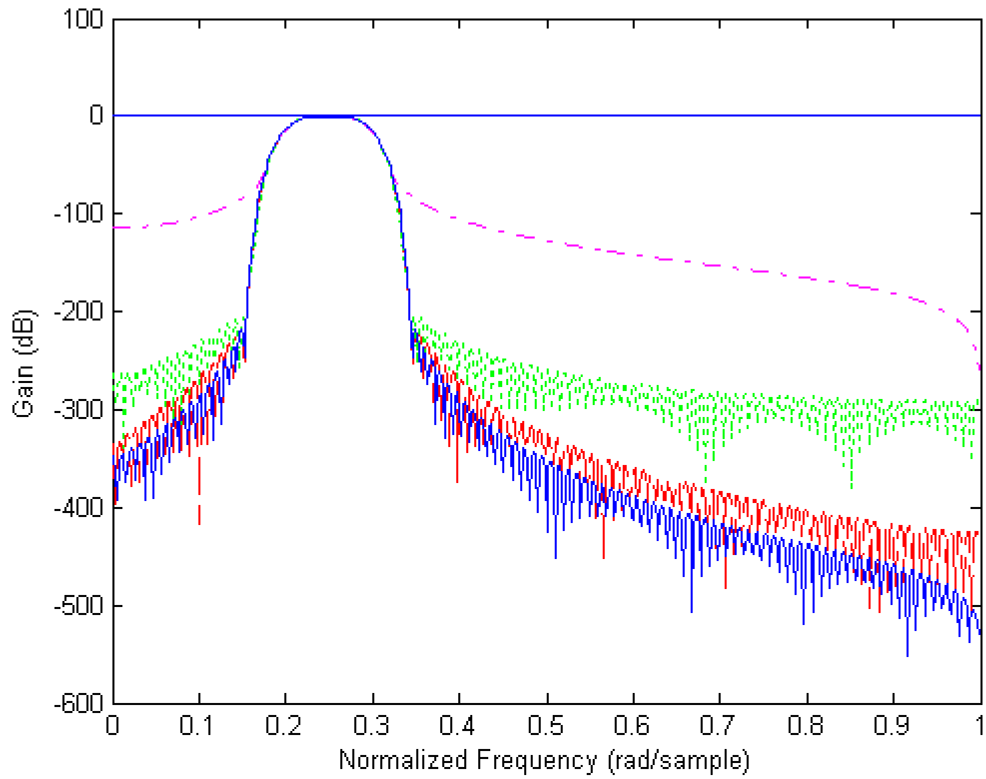

Figure 4.6 Comparison with Proposed window function and well known window functions for window length (N+1) = 202 on bandpass filter

It is clearly seen that proposed window function is more successful comparing to literature window function in terms of sidelobes in Fig 4.4, Fig 4.5 and Fig 4.6. In other words, conclusion of these comparison is pointed out changing of window length does not affect success of proposed window function.

4.5 Using of Different Filter Types on Proposed Window Function

Proposed window function gain lobes is anaysed at low pass, band pass and high pass filter types and compared with the conventional window functions gain lobes in this section.

Every three samples, used $\alpha = 2$, $\beta = 8$ and η =2.4. For the success of study taken as window length (N+1) 151. Results are shown in Fig. 4.7, Fig. 4.8. and Fig. 4.9.

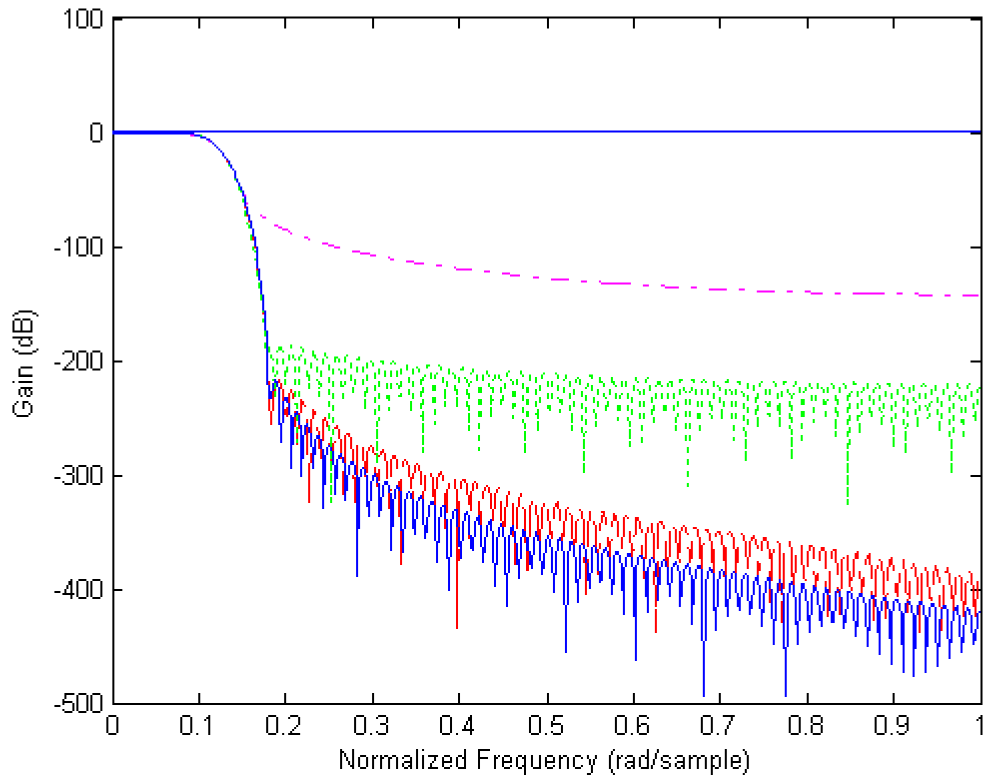

Figure 4.7 Comparison with gainlobes of Proposed window function and well known window on lowpass filter

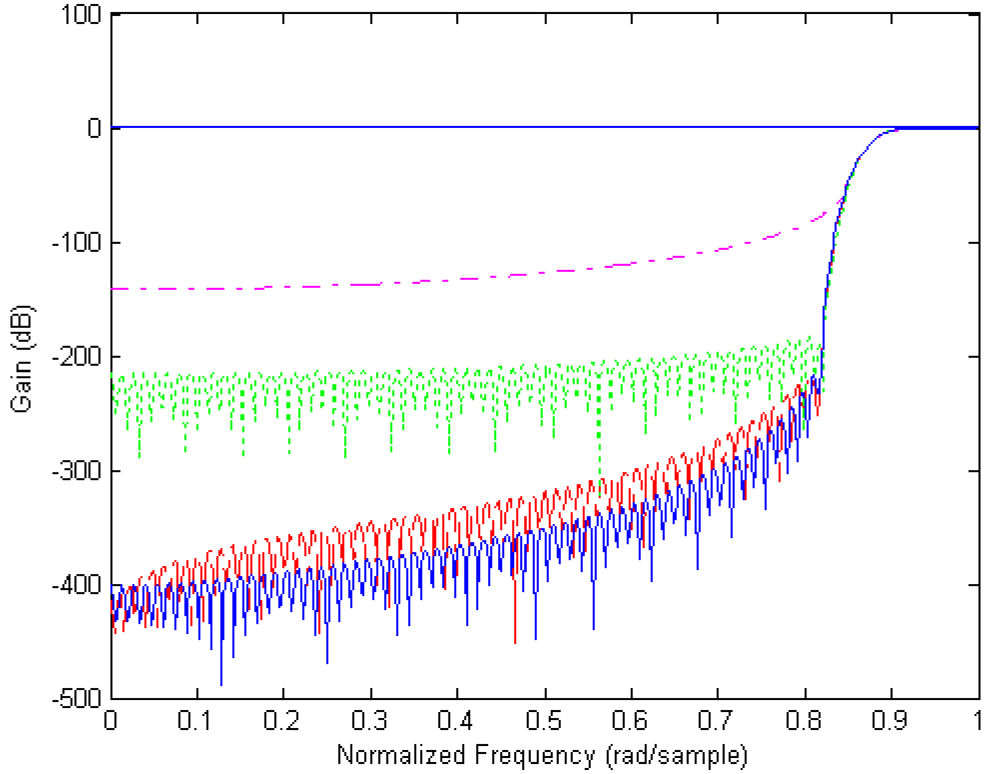

Figure 4.8 Comparison with gainlobes of Proposed window function and well known window on high pass filter

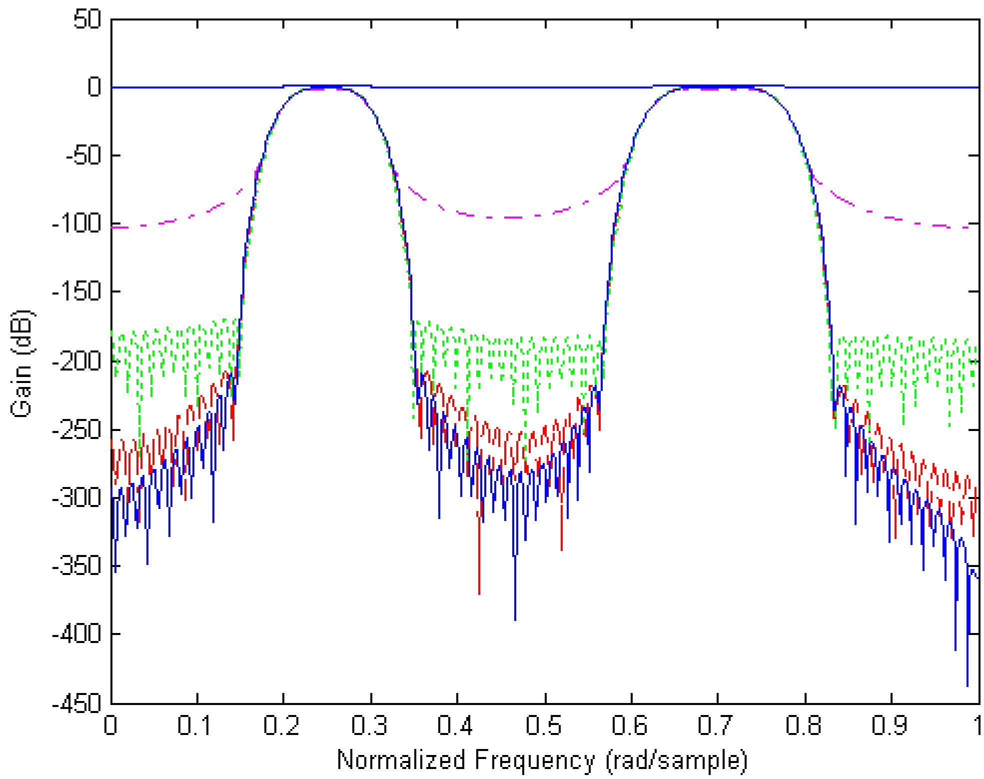

Figure 4.9 Comparison with gainlobes of Proposed window function and well known window on band pass filter

In the light of the examples results show that in Fig. 4.7, Fig. 4.8. and Fig. 4.9, change of filter types does not effect the success of proposed window function when it is compared with well known literature window functions. Proposed window function is more successful to Bartlett, Gauss and Blackman window funtion.

CHAPTER 5

CONCLUSION AND FUTURE STUDY

In this chapter, a brief summary for the results of thesis work is presented. This thesis presents a new window function and its application in the design of nonrecursive digital filters.

The window characteristic parameters give important information for the resultant filter design characteristic. It's known from the literature that smaller mainlobe width of a spectrum causes smaller transition width between the passband and stopband regions in a filter; and smaller ripple ratio causes smaller ripples in the passband and stopband regions.

In thesis, a new window function is formed Gaussian window function and Hamming window function combining the coefficients of the FIR filter are obtained by using Fourier Series Method. Combination of Gaussian and Hamming windows has resulted with more free parameters, which means the controlling of window function is more flexible, so it has better performance.

To find the optimum values of the parameters some numerical examples are studied on different filter characteristics and different window length. From the result of these examples it has been clearly shown that; a new window function reaches narrower main lobe and less amplitude side lobes.

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APPENDIX

In this section, some MATLAB programs are given which are used in this thesis for developing a new window function. Some explanation are given after nomenclature of the symbol “%”. Programs calculate and plot the window spectrums and filter amplutude responses for Rectangular, Blackman, Hamming, Kaiser, Bartlett, Hann, Gaussian and Proposed window function.

```
clear all; close all ;clc

f=[0 0.05 0.1 0.15 .2 .2 .25 .3 0.3 .35 .4 .45 .5 .55 .6 .65 .7 .75 .8 .85 .9 .95 1];

%m=[0 0 0 0 0 1 1 1 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0] %band pass filter

%m=[1 1 1 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0] % low pass filter

%m=[0 0 0 0 0 1 1 1 0 0 0 0 0 0 0 1 1 1 0 0 0 0 0] %second order band pass filter

%m= [0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 1 1 1] %high pass filter

%m is specified filter type, which is lowpass filter, highpass filter and bandpass filter.

n=150;

%n is defined window length

figure(1);

plot(f,m)

b1 = fir2(n,f,m,blackman(n+1));

[h1,w1]=freqz(b1,128);
```

```
figure(2);

plot(f,m,w1/pi,20*log(abs(h1*125)));

%plot rectangular and blackman window function

b2 = fir2(n,f,m,hamming(n+1));

[h2,w2]=freqz(b2,128);

figure(3);

plot(f,m,w1/pi,abs(h1*125),w2/pi,20*log(abs(h2*125)));

%add to plotted hamming window function

b3 = fir2(n,f,m,kaiser(n+1));

[h3,w3]=freqz(b3,128);

figure (4); plot(f,m,w1/pi,20*log(abs(h1*125)),w2/pi,abs(h2*125),w3/pi,abs(h3*125));

% add to plotted kaiser window function

b4 = fir2(n,f,m,bartlett(n+1));

[h4,w4]=freqz(b4,128);

figure(5);

plot(f,m,w1/pi,abs(h1*125),w2/pi,abs(h2*125),w3/pi,abs(h3*125),w4/pi,abs(h4*125));

%add to bartlett window function

b5 = fir2(n,f,m,hann(n+1));

[h5,w5]=freqz(b5,128);
```

```
figure (6);
plot(f,m,w1/pi,abs(h1*125),w2/pi,abs(h2*125),w3/pi,abs(h3*125),w4/pi,abs(h4*125),w5/pi,abs(h5*125))

%add hann window function

b6 = fir2(n,f,m,gausswin(n+1)); [h6,w6]=freqz(b6,128);

N=n;

alfa=2

%increasing alfa value, lead to better side loop and wider main loop.

beta=8;

% increasing beta value, lead to better side loop and wider main loop.

zeta=2.4;

% increasing zeta value, lead to better side loop and wider main loop..

brec = fir2(N,f,m,rectwin(N+1)); [hrec,wrec]=freqz(brec,128);

for a=1:N+1; n= a-(N+2)/2;

   w(n+((N+2)/2))=((cos((pi)*(n/N)))^zeta)*alfa^(((-1)*beta)*((n/N)^2));

   b(a)=brec(a)*w(n+((N+2)/2)); [h8,w8]=freqz(b,128);

end

figure (9);

plot(f,m,w1/pi,20*log(abs(h1*125)),'r--',w4/pi,20*log(abs(h4*125)),'m.',w6/pi,20*log(abs(h6*125)),'g:',w8/pi,20*log(abs(h8*125)),'b-')
```

```
xlabel('Normalized Frequency (rad/sample)')

ylabel('Gain (dB)')

figure (10);
plot(f,m,w1/pi,20*log(abs(h1*125)),w2/pi,20*log(abs(h2*125)),w3/pi,20*log(abs(h3*1
25)),w4/pi,20*log(abs(h4*125)),w5/pi,20*log(abs(h5*125)),w6/pi,20*log(abs(h6*125)),
w8/pi,20*log(abs(h8*125)))

%plots all window function

xlabel('Normalized Frequency (rad/sample)')

ylabel('Gain (dB)')
```