The Performance Enhancement Study of FIR Filters Based on Adjustable Window Function

Najat Sh.Jasim Mohammed
Electromechanical Engineering Department, University of Technology/ Baghdad
E-mail:najatjasim@yahoo.com

ABSTRACT

The aims of this study are enhancement the performance of low pass FIR (Finite Impulse Response) Filters by using adjustable window design method, and reduce the amplitude of side lobes and Gibbs phenomenon problems in these filters,also study the effect of different orders of low pass FIR filters (20th ,70th, 200th) on its performance when the parameter β in fixed value in the time domain and frequency domain, and investigate the effect of different values of parameter $\beta=0.5,3,6,9$ on its performance when the filter order in fixed value in the time domain and frequency domain.These filters are used in the ECG signal applications to remove high frequency noise.

The simulation of design these filters is implementing by using FDA tool (filter design and analysis tool) and wintool (window tool) from matlab (R2010a) program. The study results have been obtained the lowest value of relative side lobe attenuation= -66.2dB with thin main lobe width = 0.016602 when the filter order and the parameter β are equal to 200th, 9 respectively, we find that Kaiser window function is the optimum window to reduce side lobe amplitude and to reduce the Gibbs phenomenon with control of the main lobe width as compared with other windows such as Hamming, Bartlett, Blackman and Rectangular, and gives much better stop band attenuation than the Parks-McClellan algorithm.

Keywords: FIR Filters, Windowing, Kaiser Window, Gibbs phenomenon.

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دراسة تحسين اداء المرشحات ذات استجابة النبضة المحددة بالأعتمادعلى دالة نافذة قابلة للتنظيم

الخلاصة

تهدف هذه الدراسة الى تحسين اداء مرشحات المرور الواطئ ذات استجابة النبضة المحددة باستخدام نافذة قابلة للتنظيم(نافذة كيسر)،وتقليل سعة الحزم الجانبية ومشاكل ظاهرة جبس لهذه المرشحات وكذلك دراسة تاثير اختلاف درجات هذه المرشحات (20th ,70th, 200th) علـــى ادائها عندما يكون المعامل β ثابت القيمة في المجال الزمني والمجال الترددي ، وفحص تـــاثير اختلاف قيمة المعامل β ($\beta=0.5,3,6,9$) على اداء هذه المرشحات عندما تكون درجة المرشح ثابتة في المجال الزمني والمجال الترددي ،هذه المرشحات تستخدم في تطبيقات اشارة تخطـــيط ضربات القلب لازالة ضوضاء الترددات العالية.المحاكاة لتصميم هذه المرشحات انجزبواســطة

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اداة تصميم وتحليل المرشح و اداة النافذة من الاصدار (R2010a) لبرنامج المـــاتلاب، نتـــائج
الدراسة حصلت على اوطأ قيمة للتوهين النسبي للحزم الجانبية = dB66.2- وبعرض حزمـــة
رئيسية ضيقة = 0.016602 عندما تكون درجة المرشح=th200 و معامل β= 9، حيث وجدنا
ان دالة نافذة كيسر هي افضل نافذة لـتقليل سعة الحزم الجانبية وتقليل ظاهرة جبس مع السيطرة
على عرض الحزمة الرئيسية مقارنة مع بقيـــة النوافــذ مثـــل نافــذة هامنـــك، بارتلِيــت،بلاك
مان،ريكتانكلر،وتعطي احسن توهين لـحزمة التوقف مقارنة مع خوارزمية باركسمك كليلان.

INTRODUCTION

T he digital filters are used in a wide variety of signal processing
applications such spectrum analysis, and pattern recognition, and it has
been used in different fields like the military fields, processing the speech
and pictures, industrial, engineering fields, and medical technology[1][2].
Digital filters eliminate a number of problems associated with their classical analog
counterparts and thus are preferably used in place of analog filters. Digital filters
belong to the class of discrete-time LTI   (Linear Time Invariant) systems,which
characterized by the properties of causality, reusability, and stability. They can be
characterized in the time domain by their unit-impulse response sequence, and in
the transform domain by their transfer function. Obviously, the unit-impulse
response sequence of a causal LTI (Linear Time Invariant) system could be of
either finite or infinite duration and this property determines their classification
into either Finite Impulse Response (FIR) or Infinite Impulse Response (IIR)
system[1].A key element in processing digital signals is the filter, filters perform
direct manipulations on the spectra of signals. To completely describe digital
filters, three basic elements   (or building block) are needed: an adder, a multiplier,
and a delay element. The adder has two input and one output, and it simply adds
the two inputs together. The multiplier is a gain element, and it multiplies the input
signal by a constant. The delay element delays the incoming signal by one
sample [3].

THEORY

Digital filter plays an important role in today's world of communication and
computation ,on the other hand to design Finite Impulse Response (FIR) filter
satisfying all the required conditions is a challenging one, filtering is a process that
passes certain frequency components in a signal through the system and attenuates
other frequency components, the range of frequencies that is allowed to pass
through the filter is called the pass band, and the range of frequencies that is
attenuated by filter is called the stop band .If filter is defined in terms of its
magnitude response, there are four different types of filter: low pass ,high pass,
band pass, and band stop filters[4][5].Conventional FIR filters consist of cells
equal in number to the length of the filter i.e. the number of data taps. Each cell
consists of a storage register, a second register and a multiplier. The storage
register stores the data tap values, which are digital samples of the signal being
processed by the filter. The second register stores the filter coefficients for a
particular tap and the multiplier generates the product of the two register contents.

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The latter product serves as the output of the cell, and the weighted sum that constitutes the FIR filter output is generated by adding the outputs of all of the cells[6].The digital filtering process can be described, in a very simplified manner, as the result of a convolution between the input (sampled) signal and the coefficients (also called "parameters", or "tap") of digital filter[7].Before a filter can be designed, a set of filter specifications must be defined, suppose that we would like to design a low-pass filter with a cutoff frequency ω_c .The frequency response of an ideal low-pass filter with linear phase and a cutoff frequency ω_c , is

$$H_d\left(e^{j\omega}\right) = \left\{ \begin{array}{ll} e^{-j\alpha\omega} & |\omega| \leq \omega_c \\ \mathbf{0} & \omega_c < |\omega| \leq \pi \end{array} \right\} \qquad \text{... (1)}$$

which has a unit sample response:

$$h_d(n) = \frac{sin(n - \alpha)\omega_c}{\pi(n - \alpha)} \qquad \text{... (2)}$$

The specifications for a low-pass filter
will typically have the form as:

$$1 - \delta_\mathrm{p} < \left|H\left(e^{j\omega}\right)\right| \leq 1 + \delta_\mathrm{p} \quad \mathbf{o} \leq |\omega| < \omega_p \qquad \text{... (3)}$$

$$\left|H\left(e^{j\omega}\right)\right| \leq \delta_s \quad \omega_s \leq |\omega| < \pi \qquad \text{...(4)}$$

Thus, the specifications include the pass band cutoff frequency, ω_p ,the stop band cutoff frequency,ω_s ,the pass band deviation,δ_p,and the stop band deviation, δ_s. The pass band and stop band deviations, are often given in decibels (dB) as follows:

$$\alpha_p = -20\log 1 - \delta_p \qquad \text{...(5)}$$
$$\alpha_s = -20\log(\delta_s) \qquad \text{...(6)}$$

The interval ω_p ,ω_s is called the transition band. The frequency response of an N th-order causal FIR filter is:

$$H\left(e^{j\omega}\right) = \sum_{n=0}^{N} h(n)\, e^{-jn\omega} \qquad \text{...(7)}$$

Let $h_d(n)$ be the unit sample response of an ideal frequency selective filter with linear phase,

$$H_d\left(e^{j\omega}\right) = \mathbf{A}\left(e^{j\omega}\right)\mathbf{e}^{-\mathrm{j}(\alpha\omega-\beta)} \qquad \text{... (8)}$$

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Because $h_d(n)$ will generally be infinite in length, it is necessary to find an FIR approximation to $H_d(e^{j\omega})$. With the window design method, the filter is designed by windowing the unit sample response,

$$h(n) = h_d(n)w(n) \qquad\qquad \dots (9)$$

where $w(n)$ is a finite-length window that is equal to zero outside the interval $0 \leq n \leq N$ and is symmetric about its midpoint,

$$w(n) = w(N - n) \qquad\qquad \dots (10)$$

The effect of the window on the frequency response may be seen from the complex convolution theorem,

$$H(e^{j\omega}) = \frac{1}{2\pi}\ H_d(e^{j\omega}) * W(e^{j\omega})$$
$$= \frac{1}{2\pi}\int_{-\pi}^{\pi} H_d(e^{j\theta})\, W(e^{j(\omega-\theta)})d\theta \qquad\qquad \dots (11)$$

Thus, the ideal frequency response is smoothed by the discrete-time Fourier transform of the window, $W(e^{j\omega})$.

There are many different types of windows that may be used in the window design method[8].

DESIGN METHODS OF FIR FILTERS

At present, digital signal processing represents mostly used way of constructing the radio communication systems. One of the most important parts of the digital signal processing is the analysis of digital filters.FIR digital filters are widely used in the field of signal processing due to its distinguishing features such as: the stability, linear phase and easiness for realization .Traditionally, there exist some methods for FIR digital filters design, such as window method, frequency sampling method and best uniform approximation. Unfortunately, each of them is only suitable for a particular application[9][10].FIR filters are filters having a transfer function of a polynomial in *z*- and is an all-zero filter in the sense that the zeroes in the *z*-plane determine the frequency response magnitude characteristic. The *z* transform of a *N*-point FIR filter is given by:

$$\sum_{n=0}^{N-1} h(n)z^{-n} \qquad\qquad \dots (12)$$

FIR filters are particularly useful for applications where exact linear phase response is required. The FIR filter is generally implemented in a non-recursive way which guarantees a stable filter.FIR filter design essentially consists of two parts:
1.Approximation.
2.Realization problem.

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The approximation stage takes the specification and gives a transfer function through four steps. They are as follows:

a. A desired or ideal response is chosen, usually in the frequency domain.
b. An allowed class of filters is chosen (e.g. the length *N* for a FIR filters).
c. A measure of the quality of approximation is chosen.
d. A method or algorithm is selected to find the best filter transfer function[11]. To design a filter means to select the coefficients such that the system has specific characteristic .The required characteristics are stated in filter specifications. Most of the time filter problem specifications refer to the frequency response of the filter. There are different methods to find the coefficients from frequency specifications [12]:

1.Window design method.
2.Frequency Sampling method.
3.Weighted least squares design.
4.Minimax design.
5.Equiripple design.

FIR SYSTEM STRUCTURES

Consider a simple FIR system given by the following difference equation [13].
$$y(n) = x(n) + \frac{5}{6} x(\text{n-1}) + \frac{1}{6} x(n-2) \qquad \qquad \textbf{... (13)}$$

The Z transfer function can be obtained by taking the ZT of the above equation. This system has the order of 2.

$$H(Z) = \frac{Y(Z)}{X(Z)} = 1 + \frac{5}{6}Z^{-1} + \frac{1}{6}Z^{-1} \qquad \qquad \dots \textbf{(14)}$$

DIRECT FORM REALIZATIONS

Direct form realization of the system, given by Eq.(13),is the direct implementation of difference equation using the delays expressed as Z^{-1}.The direct form realization is demonstrated in Figure (1).The vertical line in the Figure (1) is delay line with tap points taken to obtain the values of x(n-1), x(n-2), etc.This realization is called direct form I realization and is also termed as tapped delay line implementation or transversal realization .In general, for an M-order system ,there would be M taps and M-1delay elements. This M-order system will require M multiplications and M-1 additions.When the FIR system has a linear phase ,the impulse response has either even or odd symmetry that is given by:
$$b(n) = \mp b(M-1-n) \qquad \dots\dots\dots\dots (15)$$

In the case of even symmetry the FIR filter is said to be symmetric and in case of odd symmetry, it is called anti-symmetric filter[13].
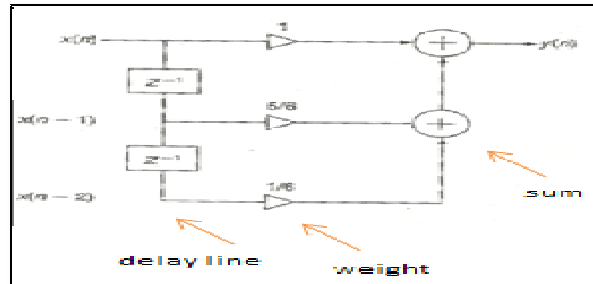
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Figure (1) Direct form realization for FIR system.

IDEAL LOW PASS FILTER IMPULSE RESPONSE

The frequency response of an ideal low pass filter is shown in the Figure (2).The frequency axis is normalized with respect to the sampling frequency. The cut-off, or transition frequency (f_t) is always between 0 and 0.5, as 0.5 represents the Nyquist frequency. As you would expect from a low pass filter, all frequencies below f_t are passed, where-as all those above are stopped. The impulse response of this ideal low pass filter is shown in Figure (3), it is a sinc function[14].
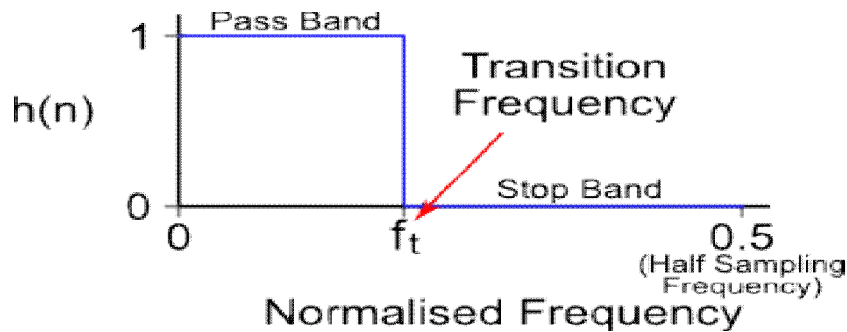
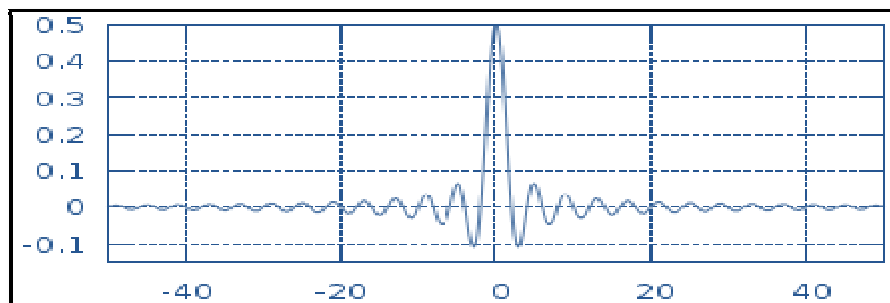**Figure (2) The frequency response of an ideal low pass filter**

Figure (3) The impulse response of Ideal LPF.

GIBBS PHENOMENON

We have truncated the impulse response of the filter and have used the response between −Q and +Q. This truncation is equivalent to multiplication of the sync function by the rectangular window as shown in Figure (4). In frequency domain, there will be convolution of Fourier transform of sync function, that is, ideal brick −wall frequency response of filter with Fourier transform of the rectangular window, that is , a sync function.The resulting convolved output is as shown in Figure (5).In frequency domain, there will be convolution of ideal filter response and the Fourier transform of the rectangular window. This convolution gives rise to a ripple in the pass band and large oscillations in the stop band as shown in Figure (5).The stop band oscillations are called spectral leakage and the oscillations near the band edge of the filter are called Gibbs phenomenon [13].
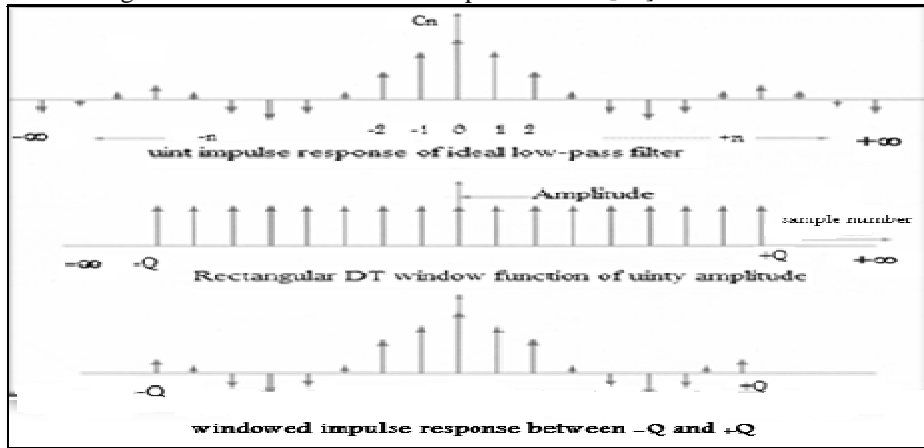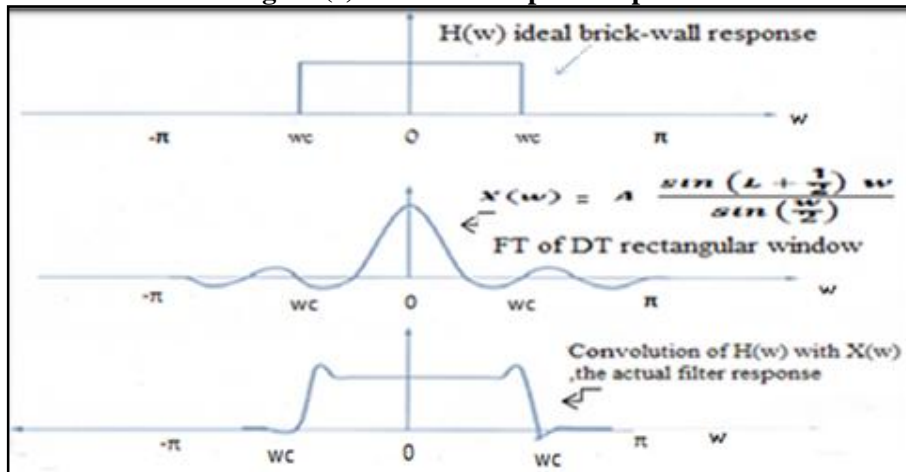

Figure (4) Windowed impulse response

**Figure (5) The Convolution of FT of impulse response filter
With FT of DT rectangular window.**

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WINDOWING

Windows are time-domain weighting functions that are used to reduce Gibbs' oscillations resulting from the truncation of a Fourier series, Very recently, windows have been used to facilitate the detection of irregular and abnormal heartbeat patterns in patients in electrocardiograms[15]. Medical imaging systems, such as the ultrasound, have also shown enhanced performance when windows are used to improve the contrast resolution of the system[15][16]. Windows have also been employed to aid in the classification of cosmic data and to improve the reliability of weather prediction models [15].one of the first class of FIR filters is that based on the use of a "smoothing window". This window, constructed to have only N non-zero points, is multiplied point by point by an impulse response of infinite duration which has the "perfect" frequency response.This multiplication or windowing has the effect of making the filter impulse response finite in duration (hence FIR),but also has the effect of smearing the desired frequency response. The stop band ripple specification is obtained by using a window capable of suppressing all side lobes to the desired degree[17].Applying a window to the sinc function weights provides extra control over the characteristics of the filter. The Figure (6) illustrates the process of windowing. First, the normal sinc weights are calculated, and then the window weights are calculated, in this case a Hamming Window has been used; as defined in equation (16). The two sets of weights are multiplied together to create the final set of filter weights.

$$w(n) = 0.54 - 0.46\cos\left(\frac{2\pi n}{M}\right)... \{ \textbf{Hamming } \text{Windowing Equation} \} \qquad ... \quad (16)$$

Once again, M is the order of the filter, which is equal to the filter length - 1. The trick is to select the window type and filter length that will give a filter with the the equations for different window types [14]. Ideally, the main-lobe width should be narrow, and the side-lobe amplitude should be small. However, for a fixed-length window, these cannot be minimized independently. Some general properties of windows are as follows : 1. As the length N of the window increases, the width of the main lobe decreases, which results in a decrease in the transition width between pass bands and stop bands. This relationship is given approximately by N Δf = c where Δf is the transition width, and c is a parameter that depends on the window.2. The peak side-lobe amplitude of the window is determined by the shape of the window, and it is essentially independent of the window length.

3. If the window shape is changed to decrease the side-lobe amplitude, the width of the main lobe will generally increase [8]. The amplitude of the side lobe can be reduced by using a smooth window function. Note that in frequency domain, FT of the window function gets convolved with the ideal filter response. The main lobe in the spectrum of the window lead to broadening of the transition band and the side lobes lead to Gibbs ringing. We would like to have a window function with thin main lobe and small side lobes[13].
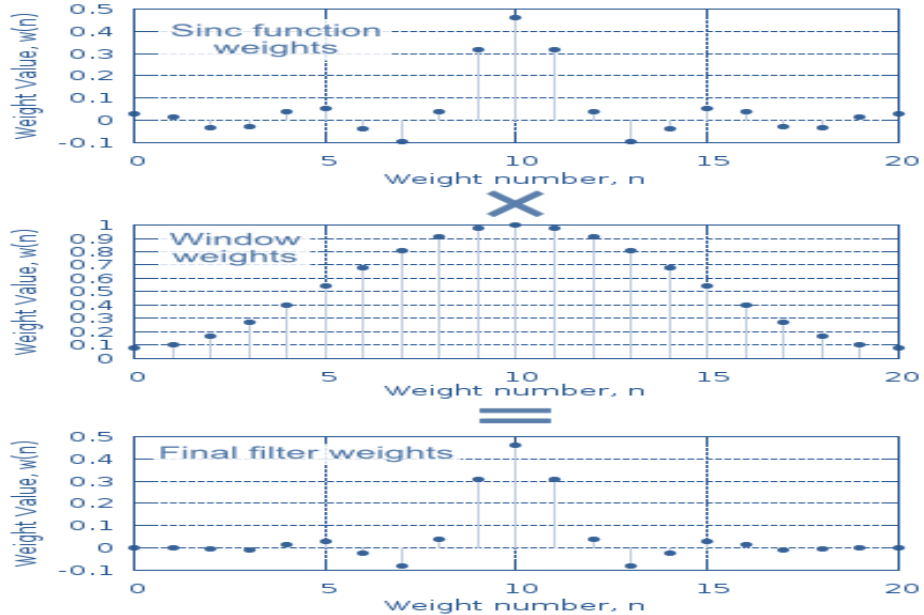
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Figure (6) the process of windowing.

Table (1) shows the equations for window types.

Window Type	Weight Equation
Rectangular	$w(n) = 1$
Bartlett	$w(n) = 1 - \dfrac{2\left\|n - \frac{M}{2}\right\|}{M}$
Hanning	$w(n) = 0.5 - 0.5\,cos\left(\dfrac{2\pi n}{M}\right)$
Hamming	$w(n) = 0.54 - 0.46\,cos\left(\dfrac{2\pi n}{M}\right)$
Blackman	$w(n) = 0.42 - 0.5\,cos\left(\dfrac{2\pi n}{M}\right) + 0.08\,cos\left(\dfrac{4\pi n}{M}\right)$

CHARACTERIZATION OF WINDOWS

Windows are frequently compared and classified in terms of their spectral characteristics. The frequency spectrum of a window is given by

$$W\left(e^{j\omega T}\right) = e^{-j\omega(N-1)T/2}\,W_0\left(e^{j\omega T}\right) \qquad ..(17)$$

where $W_0\,(e^{j\omega T})$ is called the amplitude function, N is the window length, and T is the interval between samples. The amplitude and phase spectrums of a window are given by $A(\omega) = /W_0(e^{j\omega T})/$ and $\theta(\omega) = -\omega(N-1)T/2$, respectively ,and $/W_0(e^{j\omega T})//W_0(e^0)$ is a normalized version of the amplitude spectrum. The

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normalized amplitude spectrum of a typical window is depicted in Figure (7).Two parameters of windows in general are the null-to null width B_n and the main-lobe width B_r . These quantities are defined as $B_n = 2\omega_n$ and $B_r = 2\omega_r$, where ω_n and ω_r are the half null-to-null and half main-lobe widths, respectively, as shown in Figure(7).An important window parameter is the ripple ratio *r* which is defined as:

$$r = \frac{\text{maximum side}-\text{lobe amplitude}}{\text{main}-\text{lobe amplitude}} \qquad \dots (18)$$

The ripple ratio is a small quantity less than unity and, in consequence, it is convenient to work with the reciprocal of *r* in dB, that is,

$$R = 20\log\left(\frac{1}{r}\right) \qquad ..(19)$$

R can be interpreted as the minimum side-lobe attenuation relative to the main lobe and $-R$ is the ripple ratio in dB. Another parameter that may be used to quantify a window's side lobe pattern is the side-lobe roll-off ratio, *s*, which is defined as:

$$S = \frac{a1}{a2} \qquad \dots(20)$$

where *a*1 and *a*2 are the amplitudes of the side lobe nearest and furthest, respectively, from the main lobe   (see Figure 7).If S is the side-lobe roll-off ratio in dB, then *s* is given by [15]:
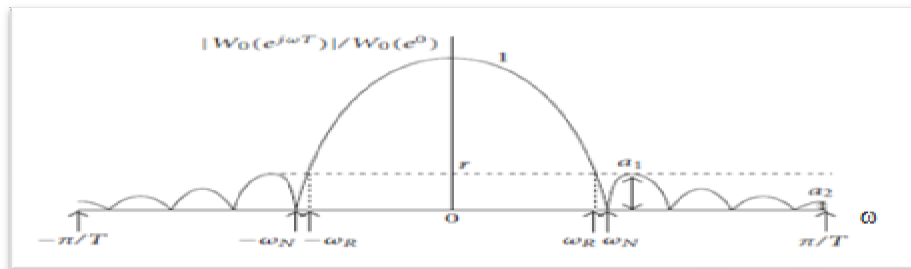
$$s = 10^{S/20} \qquad \dots(21)$$

**Figure (7): A typical window's normalized amplitude spectrum
and some common spectral characteristics.**

KAISER WINDOW

   Windows can be categorized as fixed or adjustable, Fixed windows have only one independent parameter, namely, the window length, which controls the main-lobe width. Adjustable windows have two or more independent parameters, namely, the window length, as in fixed windows, and one or more additional parameters that can control other window characteristics. The Kaiser and

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Saram¨aki windows have two parameters and achieve close approximations to discrete prolate functions that have maximum energy concentration in the main lobe[15]. The Kaiser Window is specified differently to the other windows. Rather than specifying the filter order, the amount of ripple and the transition band width are specified. By using the equations below you can calculate the values for *A* and *tw*. These values can be used to calculate the filter order, *M* and a further parameter, *β*. In these equations the transition width and sampling frequency are in Hz.

$$A = -20.0 \log_{10}(ripple) \qquad \cdots (22)$$

$$tw = 2\pi \frac{Transistion\ Width}{Sampling\ Frequency} \qquad \ldots (23)$$

$$M = \begin{cases} \text{ceil}\left(\frac{A - 7.95}{2.285 * t\omega}\right) & A > 21 \\ \text{ceil}\left(\frac{5.79}{t\omega}\right) & A \leq 21 \end{cases} \qquad .. (24)$$

$$\beta = \begin{cases} 0.0 & A \leq 21 \\ 0.5842 + 0.07886(A - 21)^{0.4} & 21 < A \leq 50 \\ 0.1102(A - 8.7) & A > 50 \end{cases} \qquad \ldots\ldots (25)$$

Once you have values for *M* and *β*, you can finally calculate the actual window weights. The Kaiser window equation makes use of another function I_0, this is a Zeroth Order Modified Bessel Function. Although it expands to infinity, the denominator quickly becomes very large, therefore you only really need to calculate $I_0(x)$ up to around i=20 [14].

$$\omega(n) = \frac{I_0\left(\beta\sqrt{1 - \left(\frac{2n}{M} - 1\right)^2}\right)}{I_0(\beta)} \qquad \ldots\ldots\ldots (26)$$

$$I_0(X) = \sum_{i=0}^{\infty} \frac{\left(\frac{X}{2}\right)^{2i}}{(i!)^2} = 1 + \frac{\left(\frac{X}{2}\right)^2}{(1!)^2} + \frac{\left(\frac{X}{2}\right)^4}{(2!)^2} + \frac{\left(\frac{X}{2}\right)^6}{(3!)^2} \qquad \ldots\ldots\ldots\ldots\ldots (27)$$

PARKS - MCCLELLAN ALGORITHM

The Parks-McClellan algorithm, published by James McClellan and Thomas Parks in 1972, is an iterative algorithm for finding the optimal Chebyshev Finite Impulse Response (FIR) filter.The goal of the algorithm is to minimize the error in the pass and stop bands by utilizing the Chebyshev approximation. The Parks-McClellan algorithm is a variation of the Remez algorithm or Remez exchange algorithm, with the change that it is specifically designed for FIR filters and has become a standard method for FIR filter design[18].

FILTER DESIGN WITH FDA TOOL

The Filter Design and Analysis Tool, or FDA Tool works with Mat lab and the Signal Processing Toolbox to provide a complete environment for start to finish filter design. Its graphical user interface, the FDA Tool supports many advanced techniques not available in SP Tool, Use the FDA Tool to: Design filters, quantize filter, analyze filters, modify existing filter designs, realize simulink models of quantized direct form FIR filters, perform digital frequency transformations of filters[19].

FILTERS SPECIFICATIONS
The low pass FIR filters design with sampling frequency =1000Hz,cutoff frequency =100Hz,and normalized frequency=0.2, based on Kaiser window design method with four values of beta (0.5, 3, 6, 9),and three values of specify order (20th, 70th, 200th).These filters used to remove the high frequency noise from ECG signal.

RESULTS AND DISCUSSION
The design of low pass FIR filters based on adjustable window design method implementing by using FDA tool and win tool from mat lab (R2010a) programs. Tables (2,3,4,5) show the simulation results of study the effect of four values of beta (0.5, 3, 6, 9),and three values of specify order (20th, 70th, 200th) on the FIR filters performance, where: Leakage factor = ratio of power in the sidelobes to the total window power,but the relative sidelobe attenuation = difference in height from the mainlobe peak to the highest sidelobe peak, and mainlobe width (-3dB) = width of the mainlobe at 3 dB below the mainlobe peak.

**Table (2): show the result of Kaiser Window with different length
of low pass FIR filter and β=0.5.**

Filter order	Relative side lobe attenuation (dB)	Main lobe width(-3dB)	Leakage factor
20	-13.6	0.085938	8.42%
70	-13.6	0.025391	8.59%
200	-13.6	0.0087891	8.49%
Filter order	Relative side lobe attenuation (dB)	Main lobe width(-3dB)	Leakage factor
20	-25.3	0.10938	0.42%
70	-24.2	0.03125	0.60%
200	-23.9	0.010742	0.64%

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**Table (3) show the result of Kaiser Window with different
Length of low pass FIR filter and *β*=3.**

Filter order	Relative side lobe attenuation (dB)	Main lobe width(-3dB)	Leakage factor
20	-44	0.14063	0%
70	-44	0.039063	0%
200	-43.9	0.013672	0%

**Table(4) show the result of Kaiser window with
different length of low pass FIR filter and *β*=6.**

Filter order	Relative side lobe attenuation (dB)	Main lobe width(-3dB)	Leakage factor
20	-66	0.17188	0%
70	-66	0.046875	0%
200	-66.2	0.016602	0%

**Table (5) show the result of Kaiser window with different
Length of low pass FIR filter and *β*=9.**

Filter order	Relative side lobe attenuation (dB)	Main lobe width(-3dB)	Leakage factor
20	-66	0.17188	0%
70	-66	0.046875	0%
200	-66.2	0.016602	0%

Table(6) shows the simulation results of comparison between the Kaiser window of 200th low pass FIR filter with some others windows such as Hamming, Bartlett, Blackman and Rectangular.

**Table (6) show the result of comparison between Kaiser
Window of 200th low pass FIR filter with other windows.**

Window type	Relative side lobe attenuation (dB)	Main lobe width (-3dB)	Leakage factor
Kaiser :beta=9	-66.2	0.01660	0%
Hamming	-42.7	0.01269	0.04%
Bartlett	-26.5	0.01269	0.28%
Blackman	-58.1	0.01562	0%
Rectangular	-13.3	0.00878	9.26%

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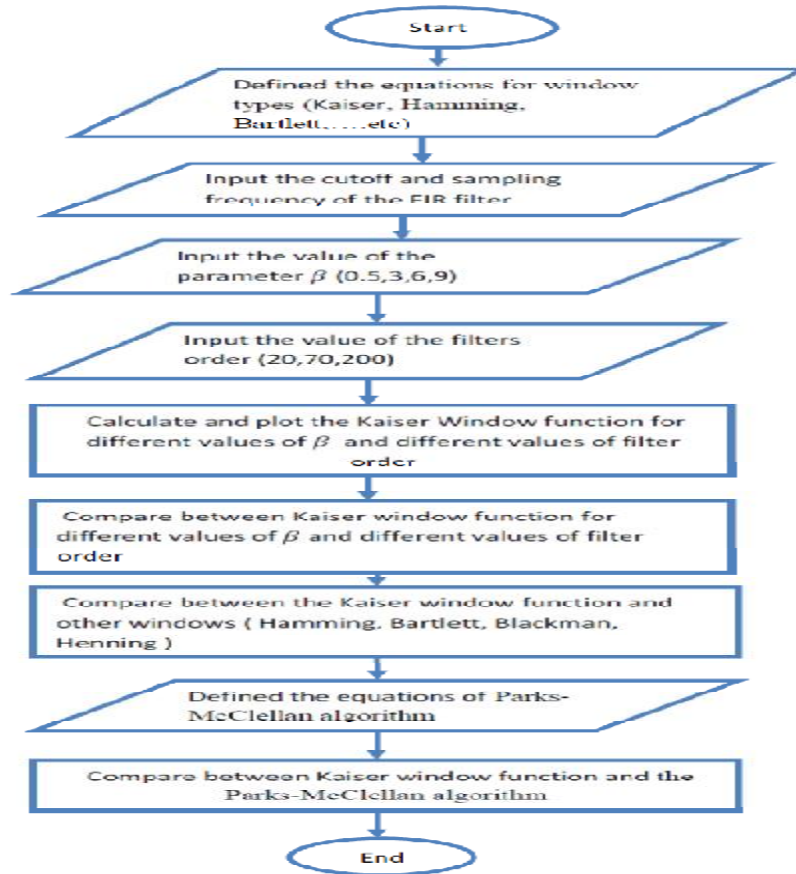Study of FIR Filters Based on
Adjustable Window Function**

Figure (8) The flow chart of simulation process of low pass FIR filter.

Figures (9,10,11) show the effect of three values of specify order of low pass FIR filters (20th ,70th, 200th) respectively on its performance when the parameter β in fixed value by using Kaiser window visualization in frequency domain and time domain .From Figure (9) when β=0.5 and increasing the filter order from (20 to 200)th ,the main lobe width(-3dB) decrease from (0.085938 to 0.0087891) and relative side lobe attenuation equal to (-13.6)dB.

And from Figure (10) when β=6 and increasing the filter order from (20 to 200)th ,the main lobe width (-3dB) decrease from (0.14063 to 0.013672) and relative side lobe attenuation increase from (-44 to -43.9) dB.

But from Figure (11) when β=9 and increasing the filter order from (20 to 200)th ,the main lobe width (-3dB) decrease from (0.17188 to 0.016602) and relative side lobe attenuation decrease from (-66 to -66.2) dB
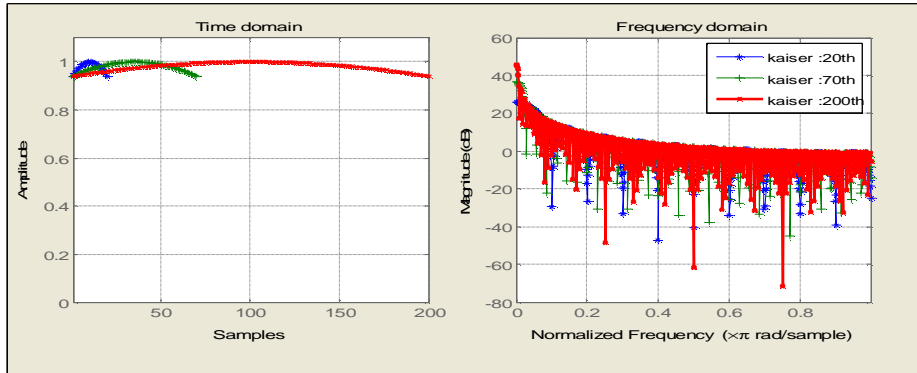
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**Figure (9) Kaiser Window visualization with different length
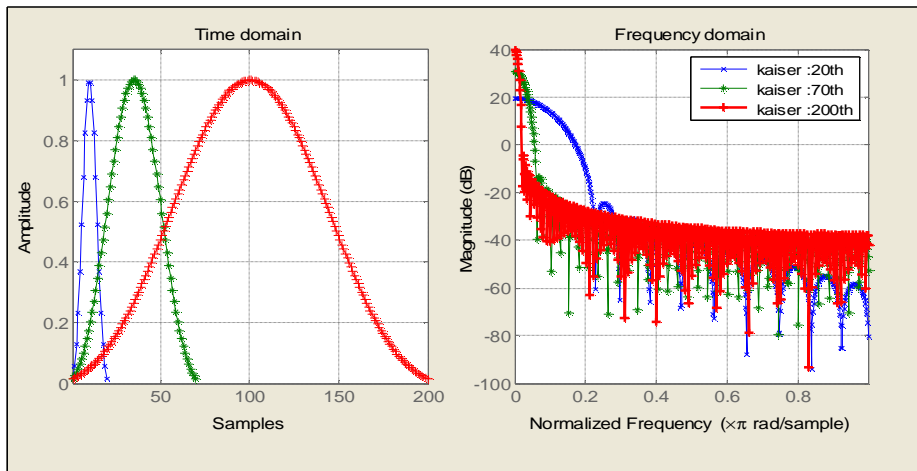of low pass FIR filter and *β=0.5*.**

**Figure (10) Kaiser Window visualization with different length
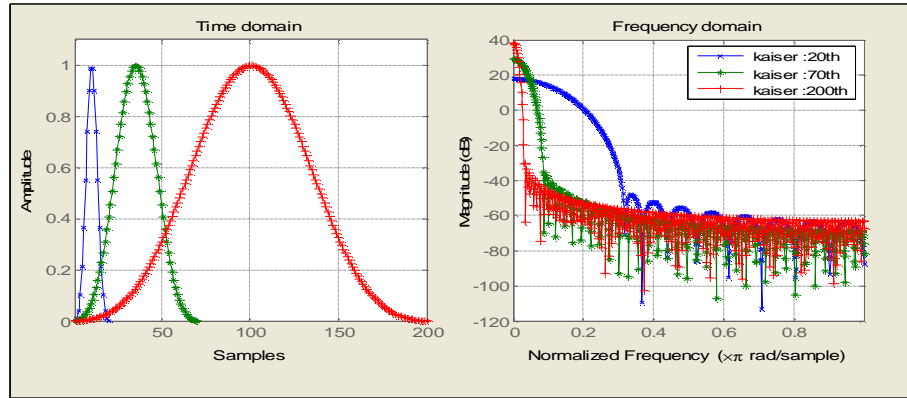Of low pass FIR filter and *β=6*.**

Figure (11):Kaiser window visualization with different length of low pass FIR filter and *β=9*.

Figures (12,13,14) show the effect of four values of parameter *β=0.5*,3,6,9 respectively on the performance of low pass FIR filters when the filter order in fixed value, in time domain and frequency domain.

Figure (12) shows the Kaiser window visualization of 20th low pass FIR, when increasing *β* from (0.5 to 9) the main lobe width (-3dB) increase from (0.085938 to 0.17188) and relative side lobe attenuation decrease from   (-13.6 to -66) dB.

Figure (13) shows the Kaiser window visualization of 70th low pass FIR, when increasing *β* from (0.5 to 9) the main lobe width (-3dB) increase from (0.025391 to 0.046875) and relative side lobe attenuation   decrease from   (-13.6 to -66) dB.

Figure (14) shows the Kaiser window visualization of 200th low pass FIR, when increasing β from (0.5 to 9) the main lobe width (-3dB) increase from (0.0087891 to 0.016602) and relative side lobe attenuation decrease from   (-13.6 to -66.2) dB .

From Figures (12,13,14) we see the increasing *β* increase the main lobe width and decrease the   height of the side lobes because the parameter *β* controls the trade-off between main-lobe width and side-lobe amplitude.
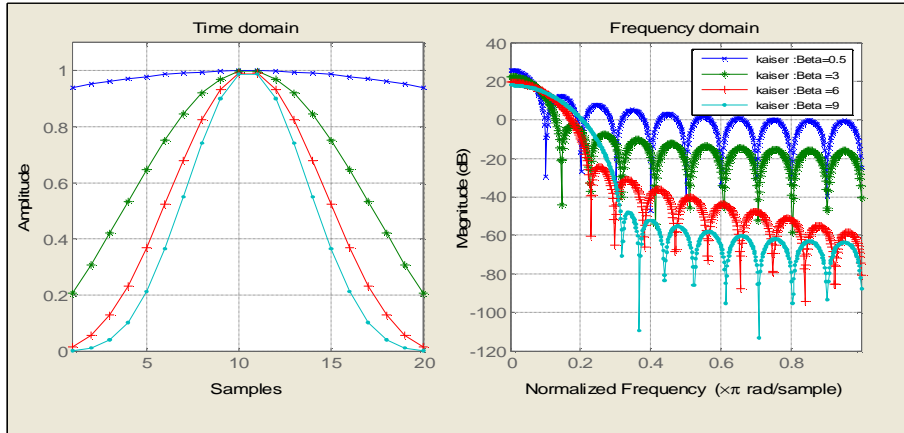
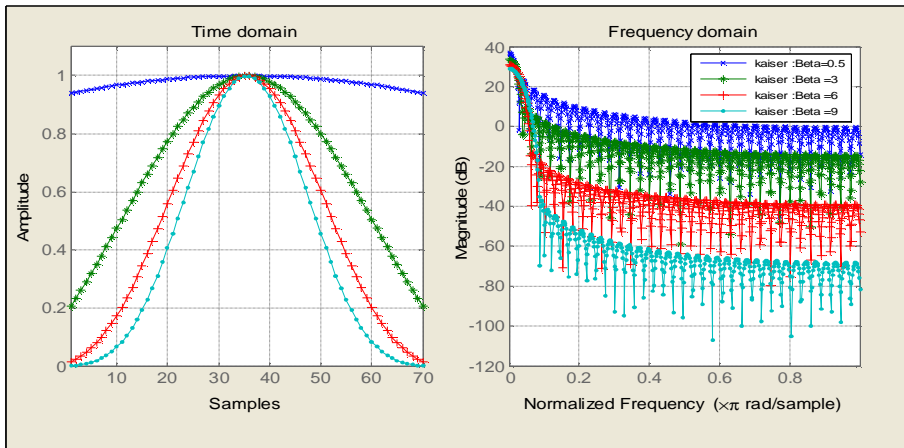**Figure (12) Kaiser window visualization of 20th low pass FIR filter with four values of *β*.**

Figure (13) Kaiser Window visualization of 70th low pass FIR filter with four values of *β*.
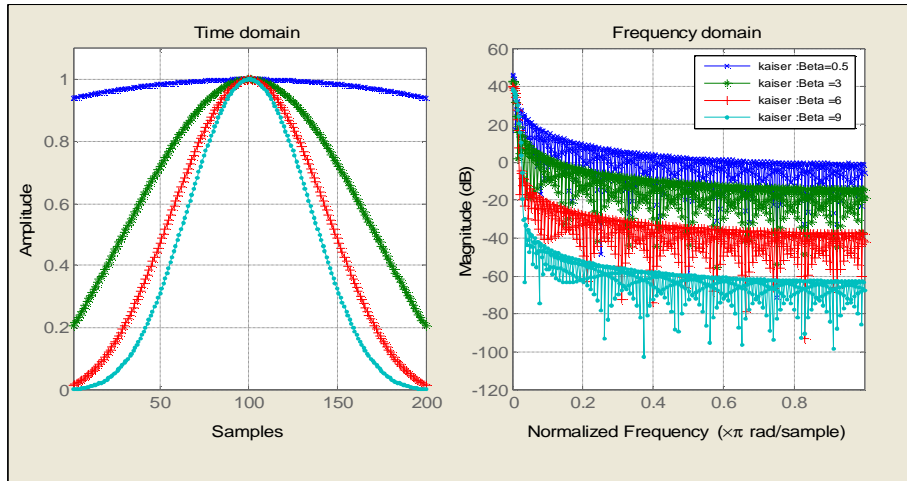
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**Figure (14): Kaiser window visualization of 200th low pass FIR
filter with four values of *β*.**

Figures (15,16) show the comparison between the Kaiser window of 200th low pass FIR filter with other windows such as Hamming, Bartlett ,Blackman and Rectangular. We see enhanced performance when Kaiser Window design method is used.
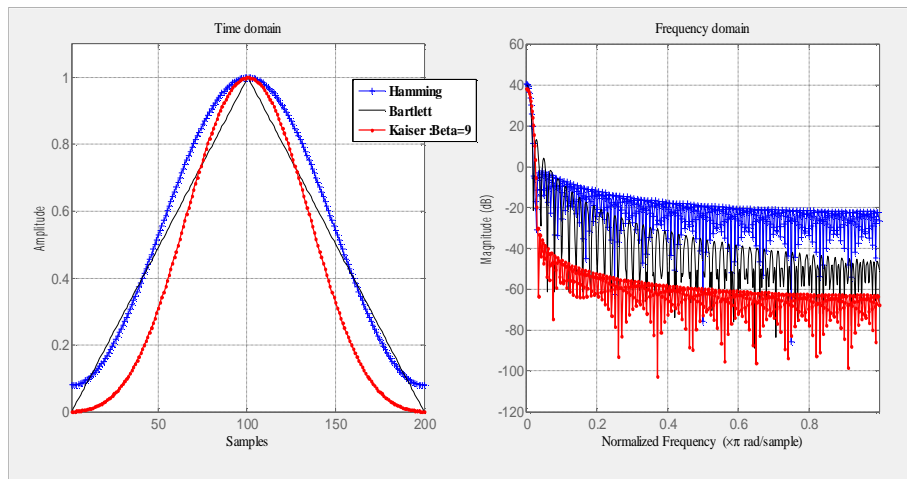

**Figure (15)comparison between Kaiser window of 200th low pass FIR
filter with Hamming window and Bartlett window.**

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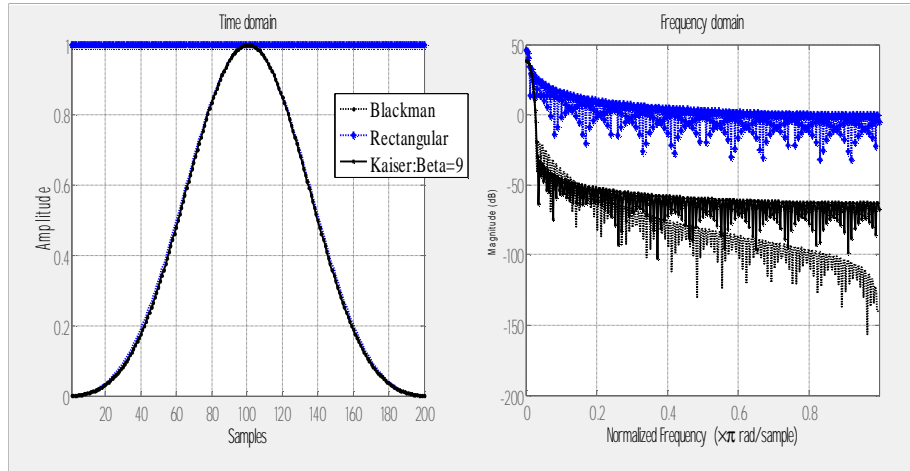Adjustable Window Function**

**Figure (16) comparison between Kaiser window of 200th low pass FIR
filter with Blackman window and Rectangular window.**

Figure (17) shows the comparison between the magnitude response of 200th
low pass FIR filter based on Kaiser window which designed in this paper and the
standard method (Parks-McClellan algorithm),that's used in reference[4]. We see
the Kaiser window gives much better stop band attenuation and flat pass band
response, but the Parks-McClellan algorithm gives narrow transition band and has
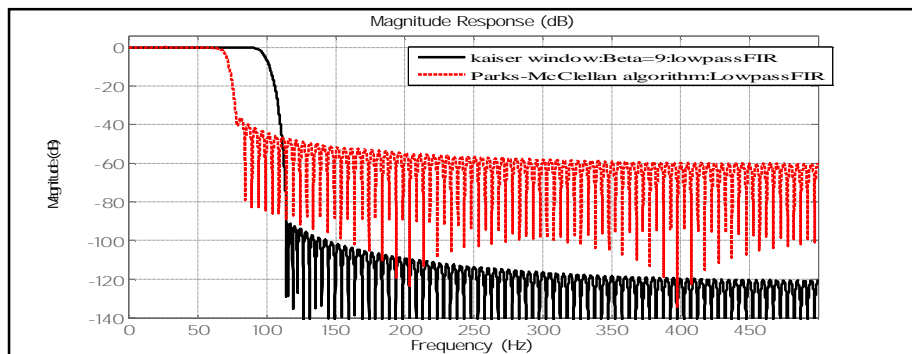large ripples near the cutoff frequency.

**Figure (17) comparison between the magnitude response of 200 low pass FIR
filter based on Kaiser window and Parks-McClellan algorithm
(cutoff frequency=100Hz).**

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CONCLUSIONS

1-The Kaiser Window function is optimum method to reduce side lobe amplitude and reduce the oscillations near the band edge of the filter (Gibbs phenomenon) with control of main lob width.

2- The lowest value of relative side lobe attenuation= -66.2 dB with thin main lobe width(-3dB) = 0.016602, when the filter order and the parameter β are equal to 200th, 9 respectively.

3-The Kaiser Window function that can have compromise between the side lobe amplitude and the transition width.

4-The parameter β determines the shape of the window and thus controls the trade-off between main-lobe width and side-lobe amplitude, we see that if the side lobe amplitude is small, then the width of the main lobe and hence the transition width increase and vice versa.

5-   The Kaiser window gives much better stop band attenuation than the Parks-McClellan algorithm.

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