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Comparative Performance Study of Finite Impulse Response Bandpass Filters Designed Using Hamming and Kaiser Windows

Conrad Onésime Oboulhas Tsahat, Charmolavy Goslavy Lionel Nkouka Moukengue, Kiminou Barol Léonard Mafouna, A Ndzeli Ngoulou

Electrical and Electronic Engineering Laboratory, National Polytechnic School, Marien Ngouabi University, Brazzaville, Republic of the Congo
Email: ln999kouka@gmail.com

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Abstract

In the field of digital communications, a filter is a device that selectively attenuates any unwanted component of a received signal and essentially changes the signal waveform in the desired manner. In this paper, we study a finite impulse response (FIR) bandpass filter using Hamming and Kaiser window methods and compare them for further analysis using Matlab software. The parameters used for the design of this filter are sampling frequency, cutoff frequency and filter order. This study shows good conformity of these two methods at the passband level and a slight difference at the lower and upper attenuation band level.

Keywords

Bandpass Filter, Finite Impulse Response, Hamming Window, Kaiser Window

1. Introduction

Digital filtering is much more widely used in digital signal processing. Digital filters are used in several fields such as seismic signal processing systems, biomedical systems, audio and video processing systems, and communication systems [Adel Jalal] [1]. Depending on the type of impulse response, digital filters can be divided into two categories: finite impulse response (FIR) filters and infinite impulse response (IIR) filters [2]. In recent years, several studies have been carried out with the aim of reducing lower lobe levels and several windowing methods have been developed to design digital filters.

In 2013, Tahseen Flaih *et al.* [2] designed a multiple accumulation (MAC) FIR digital filter from a MATLAB program. These results with Xilinx MAC filter show a significant reduction in the multiplication process, hence low power consumption. In 2015, Mohsin Iqbal *et al.* [3] analyzed the filtering effect of different digital filters using the designed FIR digital filters to process the input signal.

In 2018, Adel Jalal Yousif *et al.* [1] designed a digital finite impulse response (FIR) filter involving multiparameter optimization. These simulation results show that the proposed WIPSO algorithm is better than GA and PSO in terms of magnitude response accuracy and convergence speed for the design of a 24th-order high-pass FIR filter.

In [4], FIR filters are used for ECG signal processing, dealing with the de-noising of the signal itself. This leads to an increase in the signal-to-noise ratio (SNR) and decrease in the bit error rate (BER). The use of FIRs in the field of hearing aid design turns out to be a fairly recent topic. In ref. [5], the authors propose a system of selective noise suppression in hearing aids, in order to improve the sound that is within the range of audible frequencies for the patient in question.

Regarding radar tracking systems, FIR filters [6] are used for moving target segmentation (based on target velocity).

In 2020, Ibrahim Abdulhadi Sulaiman [7] designed and studied a low-pass filter using Rectangular, Bartlett, Hamming, Hanning, Tukey, and Kaiser Window algorithms and compared them with each other for further analysis with matlab software.

In 2023 R. Avanzato *et al.* [8], proposed a new window-based approach for the design of FIR filters, the proposed method provides a very good solution for the control of critical frequencies.

The design of digital filters with good frequency performance (steep slope, effective attenuation, etc.) and efficient implementation is a classic but still current problem, particularly in signal processing, telecommunications and embedded systems.

In order to explore methods for expanding the bandwidth of a filter, we have made a comparative study of finite impulse response bandpass filters using the hamming and kaiser methods with the help of Matlab software.

2. Theoretical Study

The synthesis of RIF filters amounts to calculating its coefficients (the coefficients of H(z)), there are two main methods of synthesis of RIF filters the time windowing method and the sampling method [9].

2.1. Design Methods of FIR Filter

There are several methods of FIR filter, for example: window function design method, optimization design method, frequency sampling design method. Window function design technique is one of the main FIR filter design methods, because of its simple operation and easy physical meaning, window function method has become a method for widely use in engineering practice.

There are six kinds of basic window function; they are Rectangular window, Triangular window, Han window, Hamming window, Blackman window and Kaiser Window.

The basic idea of all window function design method is to select the filter on the basis of suitable and ideal frequency characteristics, and then its impulse response was truncated to obtain a FIR filter of linear-phase and cause and effect. Therefore the focus of this method is to select an appropriate window function and a suitable ideal filter [10] [11].

Suppose the ideal response of desired filter is

$$h_d\left(e^{j\omega}\right)=\sum_{n=-\infty}^{\infty} h_d(n)e^{-jn\omega}$$

The design of FIR filter lies in finding a transfer function as shown in **Figure 1**.

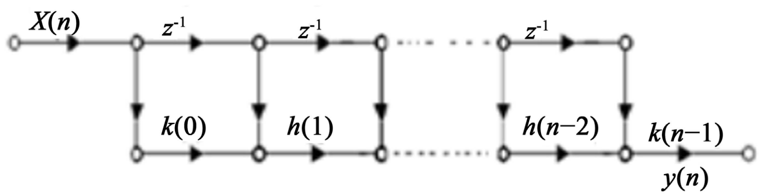


Figure 1. Direct structure of FIR filter [10] [11].

$$H\left(e^{j\omega}\right)=\sum_{n=0}^{N-1} h_d(n)e^{-jn\omega} \quad (1)$$

To approximate $H_{ad}\left(e^{j\omega}\right)$, suppose

$$h_d(n)=\frac{1}{2\pi}\int_{-x}^{x} H_d\left(e^{j\omega}\right)e^{j\omega}d\omega \quad (2)$$

The rectangular frequency characteristics of $H_d\left(e^{j\omega}\right)$ so $h_d(n)$ must be an infinite sequence and non-causal. The $h(n)$ of FIR filter to be designed is inevitable finite; infinite $h_d(n)$ was approximated by using finite $h(n)$.

The most effective way is to cut off $h_d(n)$, or $h_d(n)$ was intercepted by using finite window function sequence $w(n)$, *i.e.*

$$h(n)=h_d(n)\omega(n) \quad (3)$$

So the shape and length of Window function sequence were very critical. In the design process, window function $w(n)$ was selected according to requirements of transition bandwidth and stop-band attenuation of FIR filter [10] [11].

2.2. Synthesis by the Window Method

FIR filter design methods are based on a direct approximation of the desired frequency response of the discrete-time system. For this method, the filter coefficients correspond to the impulse response of the filter to be designed. To be able to process different signals, it is necessary that these signals be finite. However, this is not always the case for all signals. The window method is often used to create these types of finite signal sequences from infinite sequences. This "cutting" of an infinite sequence to create a finite sequence, however, affects the frequency range. Non-ideal effects, which are observed due to the finite number of filter coefficients, can be mitigated by using a weighting window. The principle is that the filter coefficients in the middle are weighted more heavily than the coefficients at the beginning and end. There are several window functions that define the maximum achievable stopband attenuation.

2.2.1. Rectangular Window

The limited knowledge of the signal is equivalent to multiplying it by a time window function called a rectangular window represented by:

$$w_R(n)=\begin{cases}1 & 0\le n\le N-1\\ 0 & \text{elsewhere}\end{cases} \tag{4}$$

The function $H_R(\Omega)$, is the Fourier transform of the rectangular window and called spectral window.

$$H_R(\Omega)=\sum_{n=0}^{N}\mathrm{e}^{-j\Omega n} \tag{5}$$

This truncation of infinite signals to obtain signals of finite duration leads to undesirable effects. A truncated signal is therefore the multiplication of a signal of infinite duration by a rectangular window, that is to say [Adel Jalal]:

$$h_N(n)=h(n)\omega_R(n)=\begin{cases}h(n) & 0\le n\le N-1\\ 0 & \text{Elsewhere}\end{cases} \tag{6}$$

2.2.2. Triangular or Bartlett Window

The Bartlett window provides a maximum attenuation of 25 dB and a transition band of width

$$\Delta\omega=\frac{6.1\pi}{N}$$

$$\omega_R(n)=\begin{cases}\dfrac{2n}{N-1} & 0\le n\le\dfrac{N-1}{2}\\ 2-\dfrac{2n}{N-1} & \dfrac{N-1}{2}\le n\le N-1\\ 0 & \text{Elsewhere}\end{cases} \tag{7}$$

2.2.3. Blackman Window

The Blackman window provides a limiting attenuation of 74 dB and a transition band of width

$$\Delta\omega=\frac{11\pi}{N}$$

$$\omega_R(n)=\begin{cases}0.42-0.5\cos\left(\dfrac{2\pi n}{N-1}\right)+0.08\cos\left(\dfrac{4\pi n}{N-1}\right) & 0\le n\le N-1\\ 0 & \text{Elsewhere}\end{cases} \tag{8}$$

2.2.4. Hamming Window

The Hamming window provides a maximum attenuation of 53 dB and a transition band of width

$$\Delta\omega = \frac{6.1\pi}{N} \tag{9}$$

$$\omega_R(n) = \begin{cases} 0.54 - 0.46\cos\left(\dfrac{2\pi n}{N-1}\right) & 0 \le n \le N-1 \\ 0 & \text{Elsewhere} \end{cases} \tag{10}$$

2.2.5. Hanning Window

The hanning window provides a limiting attenuation of 44 dB and a transition band of width

$$\Delta\omega = \frac{6.2\pi}{N} \tag{11}$$

$$\omega_R(n) = \begin{cases} \dfrac{1}{2}\left[1 - \cos\left(\dfrac{2\pi n}{N-1}\right)\right] & 0 \le n \le N-1 \\ 0 & \text{Elsewhere} \end{cases} \tag{12}$$

2.2.6. Kaiser Window

The width of the main lobe is inversely proportional to the length of the filter. The attenuation in the side lobe is, however, independent of the length and is function of the type of the window. A complete review of many window functions and their properties was presented by Harris [12] [13]. Therefore the length of the filter must be increased considerably to reduce the main lobe width and to achieve the desired transition band. Kaiser has chosen a class of windows having properties closely approximating those of the prolate spheroidal wave functions. This family of windows, known as the Kaiser windows is defined by

$$w_k(\beta, n) = \begin{cases} \dfrac{I_o\left\{\beta\left[1 - \left(\dfrac{2n}{N-1}\right)^2\right]^{\frac{1}{2}}\right\}}{I_o(\beta)} & -\dfrac{N-1}{2} \le InI \le \dfrac{N-1}{2} \\ 0 & \text{otherwise} \end{cases} \tag{13}$$

where N is window length and $I_o(x)$ is the modified Bessel function of the first kind of order zero, given by:

$$I_o(x) = \sum_{k=0}^{x}\left[\frac{\left(\dfrac{x}{2}\right)^2}{k!}\right]^2 \tag{14}$$

The Kaiser window provides the designer considerable flexibility in meeting the filter specifications.

Finding the value of β in the Kaiser window is a tuning process that involves adjustments to achieve the desired characteristics of the window for signal processing applications. This is often done using practical or theoretical methods to select a suitable β value based on the requirements of the task or signal analysis. [14]

$$\beta = \begin{cases} 0.1102\left(A_{sb} - 8.7\right) & \text{for } A_s > 50 \\ 0.5842\left(A_{sb} - 21\right)^{0.4} + 0.07886\left(A_{sb} - 21\right) & \text{for } 21 \le A_s \le 50 \\ 0 & \text{for } A_s < 21 \end{cases} \quad (15)$$

Determining the filter order in the Kaiser window is a crucial step in designing a digital filter or other filters. The calculation of the filter order often involves specifying the frequency response characteristics desired for the filter

$$N = \frac{2.056A_{sb} - 16.4}{2.285\left(\Delta\omega\right)} \quad (16)$$

Kaiser Window β = 4.54

The Kaiser window provides a maximum attenuation of 50 dB and a transition band of width

$$\Delta\omega = \frac{5.8\pi}{N} \quad (17)$$

$$\omega_K\left(n\right) = \begin{cases} \dfrac{I_o \beta \sqrt{N^2 - 4\left(n - \dfrac{N-1}{2}\right)^2}}{I_o\left(\beta\right)} & \text{for } 0 \le n \le N-1 \\ 0 & \text{Elsewhere} \end{cases} \quad (18)$$

where I_o is the Bessel function modifying the zero order and β the parameter characterizing the energy exchange between the main lobe and the secondary lobes.

$$I_o\left(x\right) = 1 + \sum_{m=1}^{\infty}\left[\frac{1}{n!}\left(\frac{x}{2}\right)^n\right]^2 \quad (19)$$

3. Results and Discussions

The window method allows the synthesis of finite impulse response filters based on specifications primarily targeting the bandwidth, the attenuation band and the transition band. This implementation always leads to parasitic ripples in the frequency response. Its main advantage is the simplicity of calculating the coefficients.

For this, we will limit our work to the study of the two Hamming and Kaiser windows for the synthesis of low-pass and band-pass filters from the point of view of ripple $A_a = 50\ \text{dB}$.

The synthesis approach using the windowing method of RIF filters knowing the specifications follows the following procedure:

- Choose the ideal filter you want;
- calculation of characteristic pulsations and frequencies (Ω_c et F_c) located at the center of the transition bands;

- search for the RI $h_R(n)$ of the ideal filter; if this is not known, it can be calculated by the Fourier transform;
- choice of window $w(n)$ satisfactory from the point of view of attenuation of the stop band;
- calculation of the order of the filter *N*knowing the width of the transition band $\Delta\omega$;
- Calculation of the filter coefficients by multiplying the RI by the chosen window.

The low-pass and band-pass filters we are going to design meet the following specifications respectively:

➢ for the low-pass filter

$$\begin{cases} F_e = 1\,\text{kHz} \\ \omega_p = 0.3\pi:\ 0\,\text{dB} \\ \omega_a = 0.4\pi:\ A_a = 50\,\text{dB} \end{cases}$$

With F_e is the sampling frequency

ω_p est la pulsation passante

ω_a is the attenuation pulse

➢ For the bandpass filter

$$\begin{cases} \omega_{p1} = 0.35\pi; w_{p2} = 0.65\pi:\ 0\,\text{dB} \\ \omega_{S1} = 0.2\pi; w_{S2} = 0.8\pi:\ 50\,\text{dB} \end{cases}$$

With ω_{p1} is the first passing pulsation

ω_{p2} is the second passing pulsation

ω_{s1} is the first attenuation pulsation

ω_{s2} is the second attenuation pulsation

3.1. Bandpass Filter

- Calculation of characteristic pulsations and frequencies (ω_c et F_c) located at the center of the transition bands:

  $\omega_{c_i} = \left(\omega_{p_i} + \omega_{a_i}\right)/2 \quad \Rightarrow \quad \omega_{c_1} = 0.275\pi$ rad/s and $\omega_{c_2} = 0.725\pi$ rad/s

  And $f_{c_i} = \dfrac{\omega_{c_i}}{2\pi} \quad \Rightarrow \quad f_{c_1} = 0.14\,\text{Hz}$ and $f_{c_2} = 0.36\,\text{Hz}$

  $f_{p_i} = \dfrac{\omega_{p_i}}{2\pi}$ and $f_{a_i} = \dfrac{\omega_{a_i}}{2\pi}$ with

  $$\begin{cases} f_{p_1} = 0.175\,\text{Hz} \\ f_{p_2} = 0.325\,\text{Hz} \end{cases} \text{and} \begin{cases} f_{a_1} = 0.1\,\text{Hz} \\ f_{a_2} = 0.4\,\text{Hz} \end{cases}$$

- Calculation of the width of the transition band:

  $\Delta\Omega_{c_i} = 2\pi \dfrac{\Delta F_{c_i}}{F_e}$ with $\Delta F_{c_i} = \left| f_{a_i} - f_{p_i} \right|$

Since we have two different stop bands, the calculations are done with the stop band with the highest attenuation. Similarly, the width of the transition band chosen is the one with the smallest value.

- Finding the RI $h_{pB}(n)$ of the ideal filter:

$$h_{pB}(n) = \frac{\sin(\omega_{c_2} n) - \sin(\omega_{c_1} n)}{\pi n}$$

So

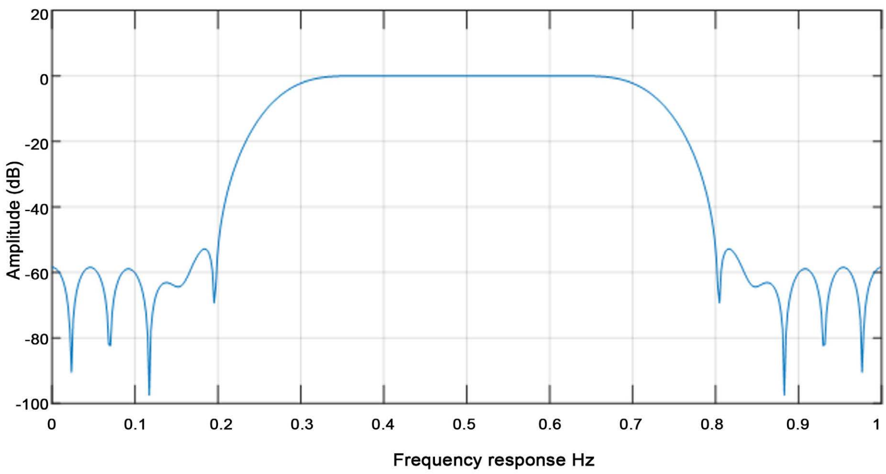
$$h_{pB}(n) = \frac{\sin(0.725\pi n) - \sin(0.275\pi n)}{\pi n}$$

- Calculation of the filter coefficients $h(n)$ by multiplying the RI by the chosen window

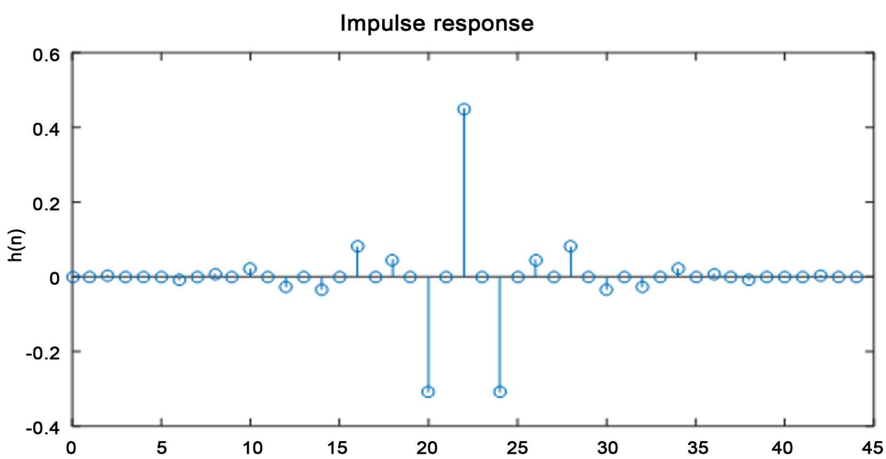
$$h(n) = \frac{\sin(0.725\pi n) - \sin(0.275\pi n)}{\pi n} w(n)$$

3.1.1. Hamming Windowing Method

The Hamming window provides a maximum attenuation of 53 dB and a transition band of width



(a)



(b)

Figure 2. Hamming bandpass filter of order $N = 45$ (a) Frequency response; (b) Impulse response.

$$\Delta\Omega = \frac{6.6\pi}{N} \Rightarrow N = \frac{6.6\pi}{\Delta\Omega}$$

Figure 2 shows the evolution of the amplitude and pulsation of the bandpass filter as a function of frequency for $N = 45$. We see that the level of the side lobes of the filter is around 50 dB. The width of the main lobe is 0.6 Hz.

As shown in **Figure 1** we have two pulsations respectively whose attenuation band $\omega_{s1} = 0.2\pi$ et $\omega_{s1} = 0.8\pi$ and the cut-off pulse is $\omega_c = 0.5\pi$ what shows a perfect agreement between the simulation and the specifications.

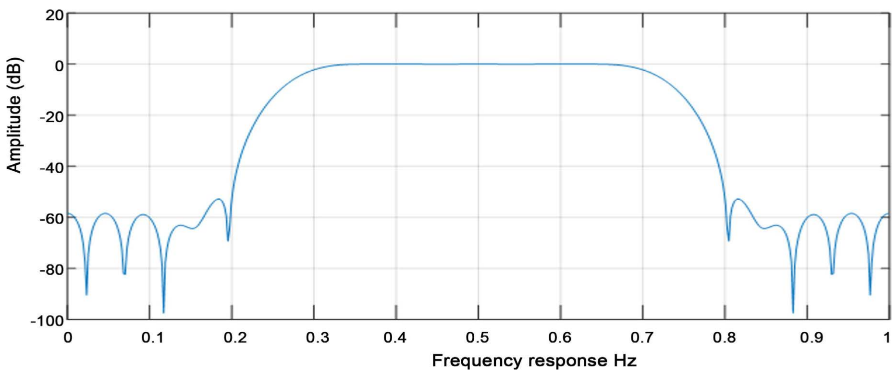
3.1.2. Kaiser Windowing Method

The Kaiser window provides a maximum attenuation of -50 dB and a transition band of width

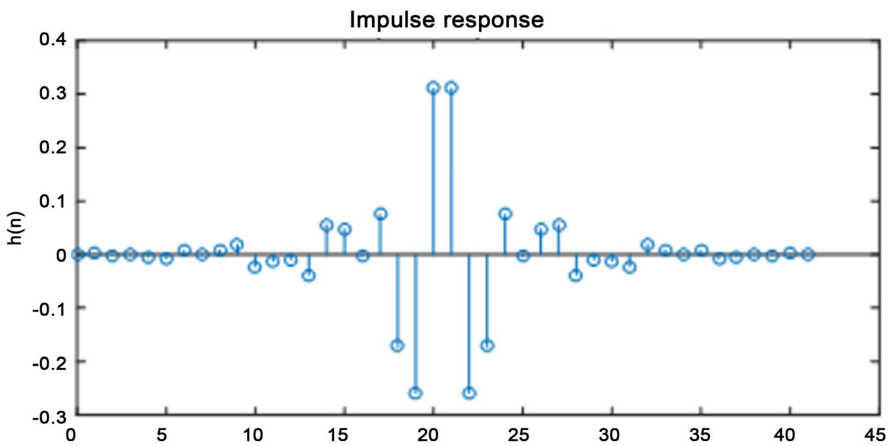
$$\Delta\omega = \frac{5.8\pi}{N}.$$

This window is one of the most effective $N = \dfrac{5.8\pi}{\Delta\Omega}$

Figure 3 shows the evolution of the amplitude and pulsation of the bandpass filter as a function of frequency For $N = 42$.



(a)



(b)

Figure 3. Kaiser bandpass filter of order $N = 42$ (a) Frequency response; (b) Impulse response.

Figure 3(a) illustrates the evolution of the amplitude of the $N = 42$ order bandpass filter as a function of frequency.

As shown in Figure 3(b), we have two pulsations respectively in the attenuation band $\omega_{s1} = 0.2\pi$ et $\omega_{s1} = 0.8\pi$, $\omega_{p1} = 0.35\pi$ and $\omega_{p2} = 0.65\pi$ in the bandwidth with the cut-off pulse $\omega_c = 0.35\pi$.

3.2. Comparative Study of Filters

Figure 4 shows a comparison of the bandpass filter with both Hamming and Kaiser windows.

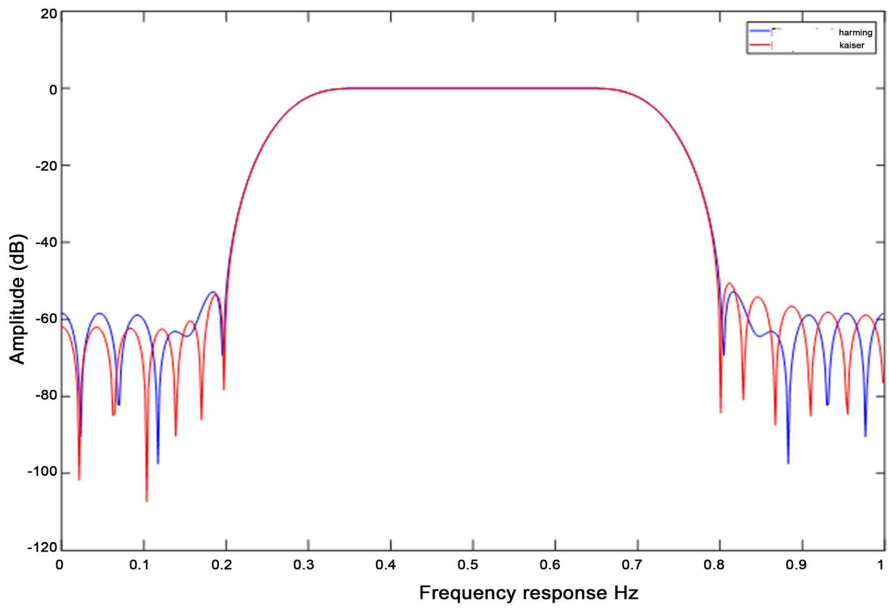


Figure 4. Bandpass filter by the Hamming and Kaiser windowing method.

The two hamming (blue) and kaiser (red) curves are the same at the passband level and slightly different at the lower and upper attenuation band level.

4. Conclusion

In this article, we are interested in the synthesis of the bandpass filter with the two windows Hamming and Kaiser, then a comparative study was made. This study was made from the Matlab software. First, we studied separately this bandpass filter of Hamming and Kaiser, then we compared the responses of the Hamming and Kaiser filters on the same figure, a good agreement is observed at the level of the bandwidth and a slight difference at the level of the lobes.

Conflicts of Interest

The authors declare no conflicts of interest regarding the publication of this paper.

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