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Design, comparison and analysis of low pass FIR filter using window techniques method

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ABSTRACT

In digital communication, a filter is any device that selectively attenuates any unwanted component from a received signal. It essentially changes or reshapes the waveform of a signal in the desired manner. In this work, we have designed and studied low pass filter using Rectangular, Bartlett, Hamming, Hanning, Tukey and Kaiser Window algorithms and compared them with each other for further analysis. The parameters we used for designing the filter are sampling frequency, cut off frequency, filter order and a variable parameter, α. MATLAB toolbox for DSP, namely FDA Tool and FV Tool were employed for the realization of the design.

1. Introduction

In digital communication, a filter is any device that selectively attenuates any unwanted component from a received signal. It essentially changes or reshapes the waveform of a signal in the desired manner. Filters are widely used in the field of telecommunication and electronics [1,2]. Signal processing, signal restoration and separation are the primary tasks of any filter [3,4]. In signal processing, digital filters are more preferable to analog filters because digital filters have some characteristics that are not possible with analog filters such as linear phase response. Digital filters are commonly used to suppress the additional noise in a signal which has a negative effect on the performance of the system. For example, in a telephone system, there is no need to transmit high frequencies since most speech frequencies fall within the band of 0.4 to 3.4 kHz [5]. Digital filters are divided in terms of their impulse response; Finite Impulse Response (FIR) and Infinite Impulse Response (IIR).

Linearity, stability, efficient realization on hardware and flexibility in controlling the waveform and magnitude response provide the motive of using FIR filters in most application such as data compression, speech and image processing etc. FIR filters are employed in filtering problems where there is a requirement for a linear-phase characteristic within the passband of the filter, otherwise, an IIR or FIR can be used [6,7]. In this work, Fourier series method is used in FIR design. A spontaneous oscillation occurs during trimming the impulse response. This abrupt oscillation is removed by time limited weighted function w(n) known as window function to cause a change in spectral efficiency, sampling frequency cutoff frequency, sampling rate, and cycling prefix.

2. Design methods of FIR

There are several methods for designing FIR filters. FIR filter is completely defined by its corresponding difference equation or transfer function. The former can be written as:

$$y(n) = \sum_{k=0}^{M-1} b_k x(n-k)$$

where M is the filter length, M-1 represents the order of the filter, b_k is the set of filter coefficients, x(n) and y(n) are the filter input and output. From the equation above the FIR filter response depends on the present and past input samples [2].

Filters specifications i.e. pass- band ripples, stop band ripples, maximum attenuation, minimum attenuation of the stop band, cut off frequencies (f1 and f2) and the transition bandwidth are

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often specified in frequency domain [8]. The various techniques for the design of FIR filters are Frequency sampling method, Fourier series method, and Window method.

2.1. Window method

In the Fourier series method of FIR filter design, when truncating the infinite duration impulse response, an abrupt oscillation occurs which make the method undesirable. This undesirable oscillation can be reduced by using a time limited weighting function w (n), called window functions, to improve the Fourier coefficients. The Fourier series relation is given by

$$H_d(e^{jw}) = \sum_{n=-\infty}^{\infty} h_d(n)e^{-jwnt}$$

where,

$$h_d(n) = \frac{1}{2\pi}\int_0^{2\pi} H(e^{j\omega})e^{j\omega n}d\omega$$

In general, truncating $h_d(n)$ to a length M−1 is equal to multiplying a window function, w(n) with $h_d(n)$. Therefore, he FIR filter becomes

$$h(n) = h_d(n)w(n)$$

Various window-based methods that are used as precures for designing this filter are listed below:

1. Jains method: This method provides best side lobe efficiency compared to Blackman and Hamming window, but it consists of large transition band that is not spectrally efficient
2. Motaghe Kashtibans method: Narrower main lobe compared to Hamming window
3. Rakshits method: Higher sidelobe roll off ratio compared to Hamming window
4. Kumars method: Best stop band performance compared to Hamming window but width of main lobe increases
5. Martin method: Best signal to overall interference ratio compared to Blackman window

3. MATLAB implementation and analysis of FIR filter using window method

3.1. Design procedure

Procedure for designing an FIR filter using window method is given as follows: Describe the specifications of a filter, find the window function from the specification w(n), evaluate the order of the filter (M-1), find the coefficients of the window function, evaluate the ideal filter coefficients, and evaluate the coefficients of the FIR filter.

3.2. Low pass filter

We design low pass filter with the following specifications: sampling frequency of 44,100 Hz, and a cut-off frequency of 10,800 Hz with filter order of 30. We have compared two filters with common characteristics and the better one is chosen, compared with other sets and the best is chosen. In this case the rectangular is compared with Bartlett, Hamming with Hanning, and Kaiser with Tukey.

4. Result and discussion

We have designed, compared and analysed six types of window functions for Low Pass filter. From Table 1, we can deduce that rectangular window has the highest leakage factor (9.15%) and Hamming window with the least value (0.04%). In terms of relative side lobe, hamming window has the least value (−41.7 dB) while rectangular and Kaiser have (−13.3 dB) and (−13.6 dB) respectively. However, in the Main lobe, both Kaiser and rectangular have the same main lobe width. This means that as the length of filer is increased, the width of the main lobe becomes narrower and narrower, and the transition is reduced considerably. However, the attenuation in the side lobe is independent of the length and is a function of the type of the window. We can then deduce that the width of the main lobe is inversely proportional to the length of the filter [2].

From Fig. 1, it is observed that the rectangular window has more ripples at the stopband compared to Bartlett, however Bartlett window has a flatter response at the passband. Fig. 2 shows

Table 1
MATLAB Simulation Result.

Window Techniques	Leakage Factor %	Relative Side Lobe dB	Main Lobe Width dB
Rectangular	9.15	−13.3	0.054688
Hamming	0.04	−41.7	0.085938
Barlett	0.28	−26.5	0.085938
Hanning	0.05	−31.5	0.09375
Tukey	3.58	−15.1	0.078125
Kaiser	8.34	−13.6	0.054688

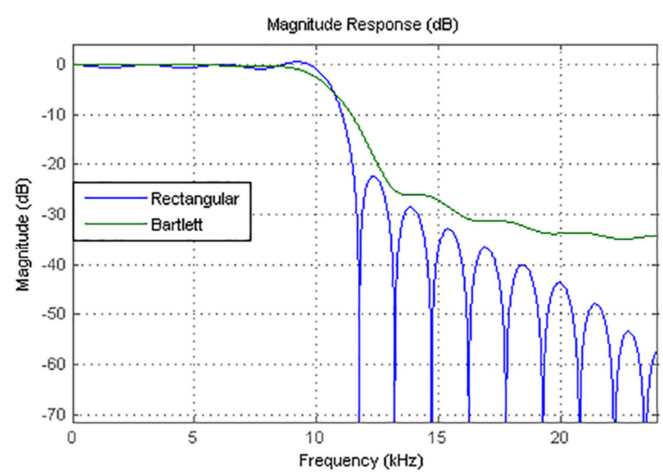

Fig. 1. Magnitude of Rectangular and Bartlett window.

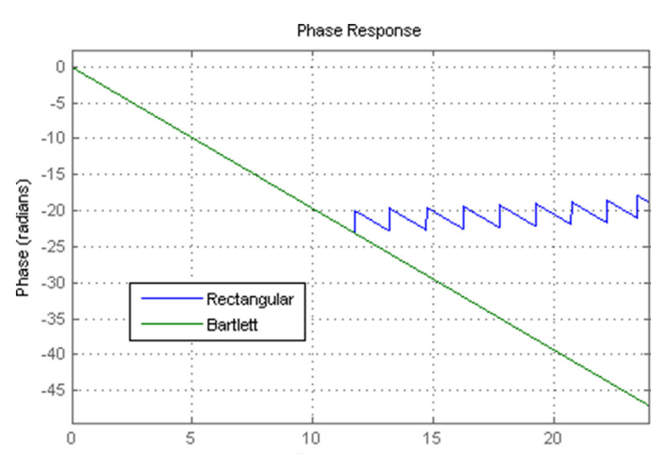

Fig. 2. Phase Response Rectangular and Bartlett window.

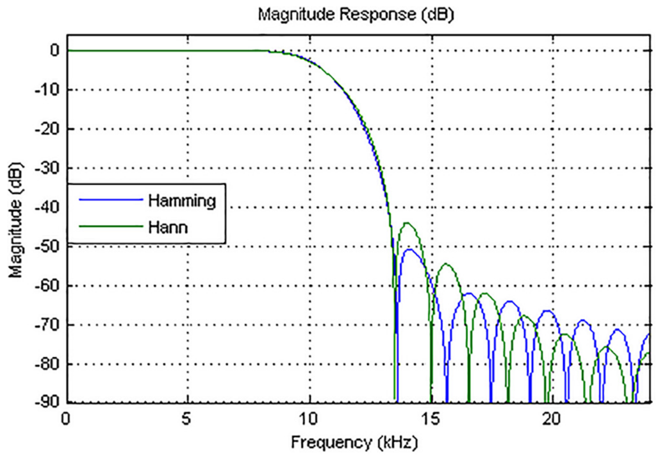

Fig. 3. Magnitude of Hamming and Hanning window.

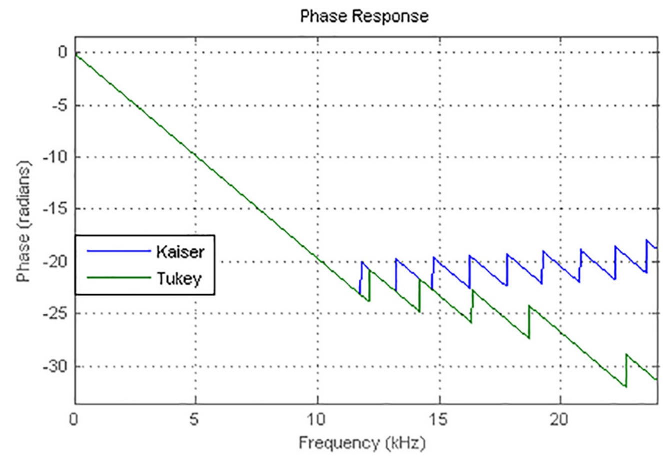

Fig. 6. Phase Response of Kaiser and Tukey window.

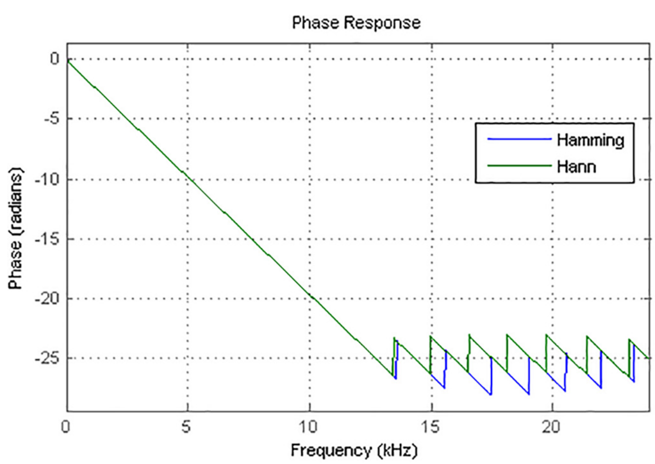

Fig. 4. Phase Response of Hamming and Hanningwindow.

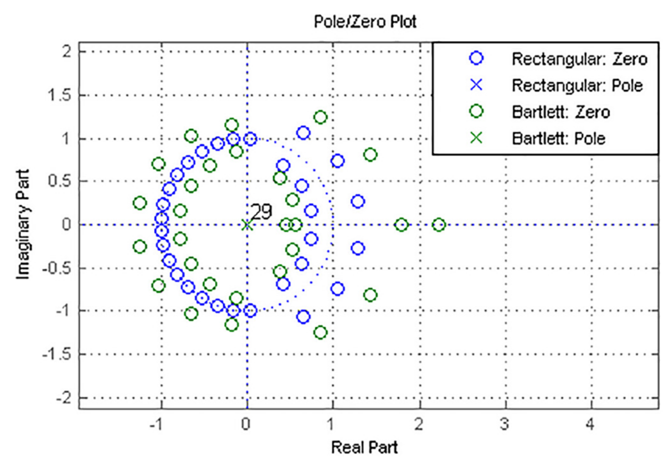

Fig. 7. Zero/Pole plot of Rectangular and Bartlett.

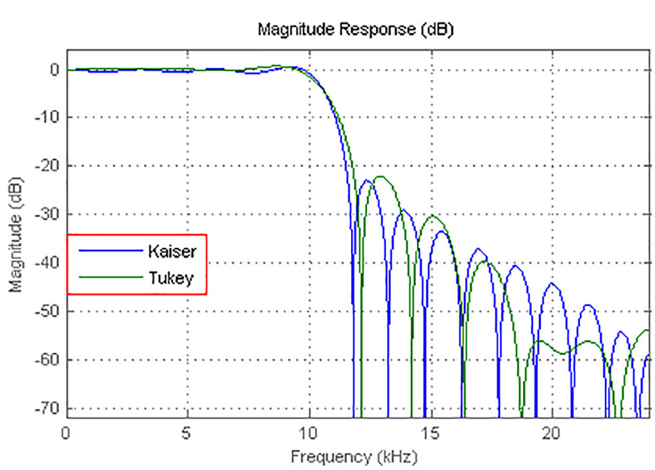

Fig. 5. Magnitude of Kaiser and Tukey window.

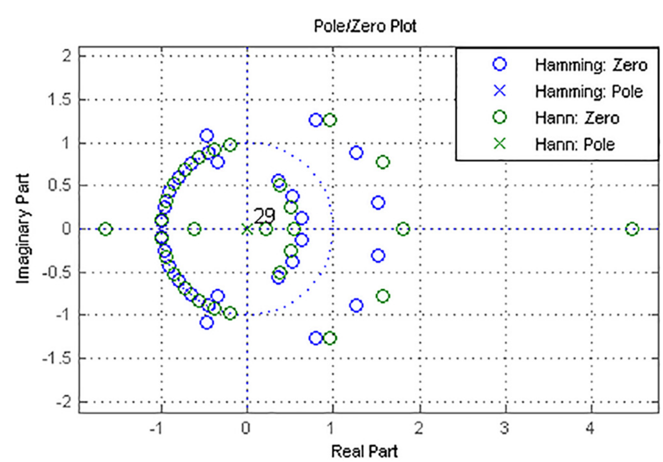

Fig. 8. Zero/Pole plot Hamm and Hann.

Phase Response Rectangular and Bartlett window. Fig. 3 shows the magnitude response of Hamming and Hanning window. Hamming Window has a better stopband attenuation, and they have the same flat response at the passband. Fig. 4 shows Phase Response of Hamming and Hanning window. In the Magnitude Response plot (Fig. 5), it is obvious that Kaiser window has a better transition band than the Tukey window. Fig. 6 shows Phase Response of Kaiser and Tukey window.

As shown in the Figs. 7–9, (the zero/pole plot of the window functions), by inspecting carefully we can see that most of the zeros are within the unit circle, some lie on the unit circle, and

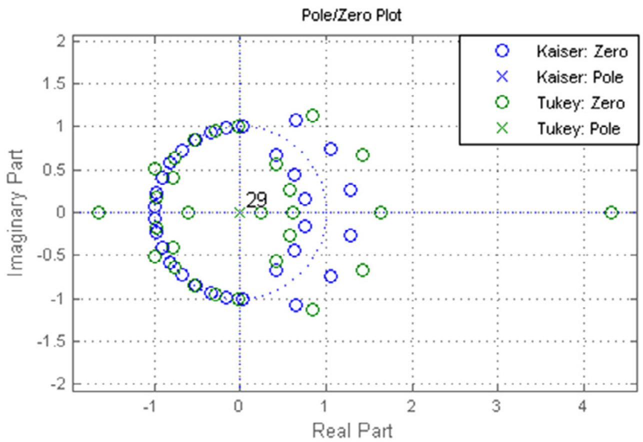

Fig. 9. Zero/Pole plot of Kaiser and Tukey.

few of the zeros are outside the circle. All the plots have a single pole at origin which means it always come with default zero pole and as inherent to all FIR filters, they have no pole. This is as a result of the increase in the component of the impulse response; hence all the functions are observed to be stable.

Figs. 10 and 11 show the overall magnitude and frequency response of all the window functions used in this paper. It can also be observed from the figure that Kaiser window has the smallest transition band, ideal frequency response and a desired stop band attenuation which can be controlled by a parameter, α (it is a normalized that ranges from 0 to 1). Hence, Kaiser window is suitable for design of a low pass filter irrespective of the choice of the filter order.

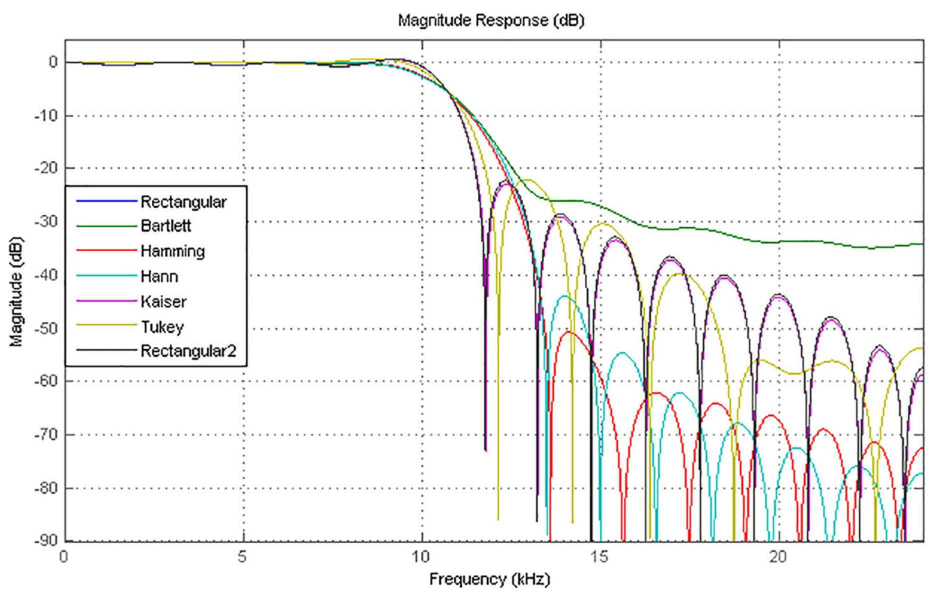

Fig. 10. Overall Magnitude Response.

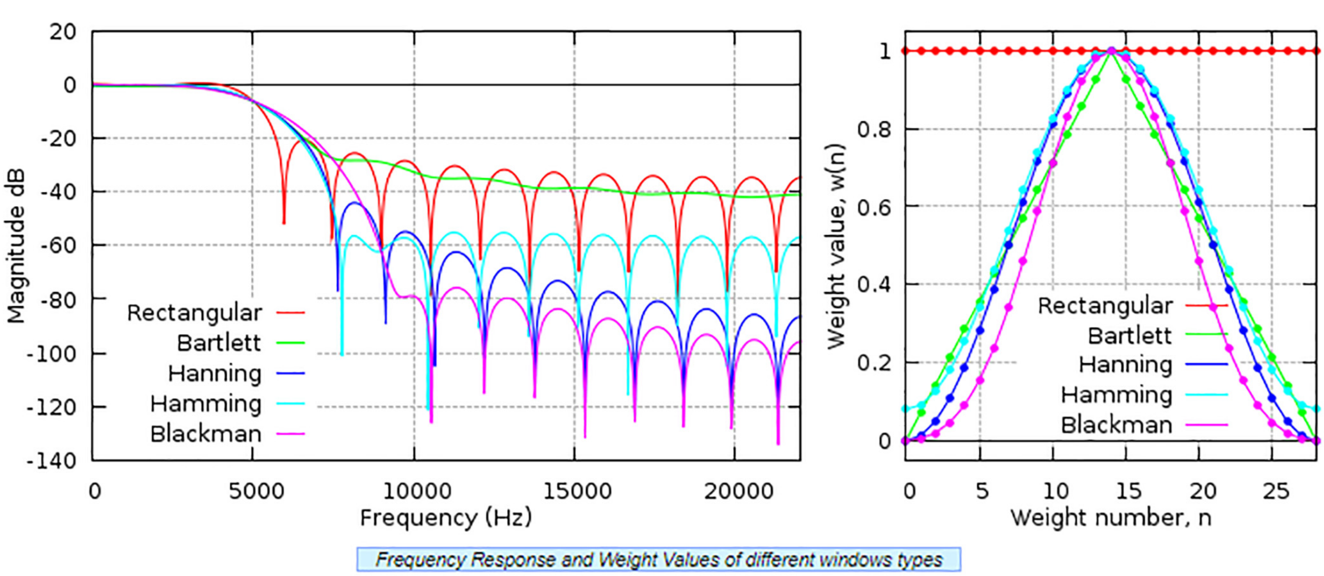

Fig. 11. Overall frequency response.

5. Conclusion

In conclusion, out of all the compared window functions, Kaiser window has the smallest transition band, ideal frequency response and a desired stop band attenuation which can be controlled by a parameter, α (it is a normalized value that ranges from 0 to 1). Hence, Kaiser window can be used in a design where the same ripple size is used in both the passband and stop band.

CRediT authorship contribution statement

Ibrahim Abdulhadi Sulaiman: Formal analysis, Writing - original draft. **Hussain Mohammad Hassan:** Methodology, Software, Formal analysis, Writing - original draft. **Mohammad Danish:** Writing - review & editing. **Munendra Singh:** Validation, Conceptualization, Writing - review & editing. **P.K. Singh:** Validation, Supervision. **Manisha Rajoriya:** Conceptualization, Methodology, Resources, Supervision.

Declaration of Competing Interest

The authors declare that they have no known competing financial interests or personal relationships that could have appeared to influence the work reported in this paper.

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