PERFORMANCE ANALYSIS OF FIR DIGITAL FILTER DESIGN BY USING HAMMING AND KAISER WINDOW TECHNIQUES

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Abstract: Digital signal processing (DSP) is all of digital filters. Finite Impulse Response (FIR) digital filter is preferred than Infinite Impulse Response (IIR) for many key advantages. In this article, FIR digital filter is designed using Hamming and Kaiser Window techniques and their performance is evaluated. The filter coefficients, frequency response (both amplitude and phase) graph and frequency and time-domain plot before and after filtering are demonstrated. The simulation result shows that the Kaiser window provides better filtering performance compared to Hamming because it can provide better minimum stop band attenuation for fixed filter length and transition width in comparison with Hamming. Thus Kaiser Window is the favourite window that can be employed for many filtering applications.

Keywords: DSP, FIR digital filter, Hamming window, Kaiser window, MATLAB

Introduction

Digital filter plays pivotal role in the field of Digital Signal Processing (DSP). It's broadly used than analog for its improved signal to noise ratio (Ferdous & Ali, 2013) easy implementation and flexibility under software control (Gupta & Panghal, 2012). Its design and analysis is easily and efficiently achieved using computing capabilities of MATLAB. Digital filters are used to modify the frequency characteristics of the input signal to meet certain specific design requirements. Digital filter may be Finite Impulse Response (FIR) or Infinite Impulse Response (IIR) according to their impulse response length. FIR filters are preferred than IIR filters, because of its linear phase, stability, non-recursive structure and arbitrary amplitude-frequency characteristic (Saiful *et al*., 2014).

Generally, four methods are used to design FIR digital filters: Windowing, Optimization, Numerical and Frequency sampling (Gupta & Panghal, 2012; Saiful *et al*., 2014; Avci, 2016). Windowing is used in this research for its straightforwardness and minimal computational burden. Windows were categorized based on the number of independent window parameters in their functions as, fixed or adjustable (Saramaki, 1993; Nouri, 2011; Avci, 2016). Fixed windows have only one independent parameter, the window length which controls the main-lobe width. Adjustable windows have two or more independent parameters, the window length, and one or more additional parameters that can control other window's characteristics. Rectangular, Triangular (Bartlet), Hanning, Blackman, Hamming and Kaiser are well-known fixed windows with an exception of Kaiser which is adjustable with an additional parameter, α for controlling the sidelobe level.

Gupta & Panghal (2012) shows that Hamming window is stable and has linear phase than Hanning and rectangular window techniques. Kaiser window is proved to be better than the other techniques (Ferdous & Ali, 2013; Kumar, 2014).

The two fundamental objectives of this research are; first to design an FIR digital filter using Hamming and Kaiser window techniques. Secondly, to implement, analyze, evaluate and compare their performance.

Materials and Methods

Window method is the straightforward and the simplest technique for FIR digital filter design (Tiwari, 2014). The design of FIR filter is performed by selecting the specifications of the filter (the desired impulse response in accordance with the frequency response). Then the designed filter is implemented using MATLAB.

The output y(n) of FIR filter with length M for an input signal x(n) is given by the convolution of unit sample response of the system:

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

Where: coefficient of FIR filter is indicated by the impulse response of the filter h(n).

In the window method, the desired frequency response specification $H_d(\omega)$, corresponding unit sample response $h_d(n)$ is determined using the following relations:

$$h_d(n) = \frac{1}{2\pi}\int_{-\pi}^{\pi} H_d(\omega).e^{jwn}d\omega \quad (2)$$

Where: $H_d(\omega) = \sum_{-\infty}^{\infty} h_d(n).e^{-jwn}$ (3)

In general, unit sample response $h_d(n)$ obtained from the above relation is infinite in duration, so it must be truncated at some point say n = M-1 to yield an FIR filter of length M (i.e. 0 to M-1). Then h(n) the impulse response of a preferred FIR filter is given by:

$$h(n) = \begin{cases} h_d(n) & 0 \le n \le M-1 \\ 0 & otherwise \end{cases} \quad (4)$$

The Frequency response of the desired FIR filter is

$$H_d(e^{jw}) = \sum_{n=0}^{M-1} h_d(n).e^{-jwn} \quad (5)$$

Direct truncation of $h_d(n)$ to M terms to obtain h(n) leads to the Gibbs phenomenon effect which manifests itself as a fixed percentage overshoot and ripple before and after an approximated discontinuity in the frequency response due to the non-uniform convergence of the Fourier series at a discontinuity.

Thus the frequency response obtained by using above equation contains ripples in the frequency domain. In order to lessen the ripples, $h_d(n)$ is multiplied by a window function w(n) whose duration is finite:

$$h(n) = h_d(n).w(n) \quad (6)$$

The window function and filter order are both specified according to the desired normalized frequencies (ωc, ωc1, ωc2), transition width and stopband attenuation (Ferdous & Ali, 2013).

The commonly used windows are (Proakis & Manolakis, 2011; Saramaki, 1993; Dogra & Sharma, 2014):

1. Rectangular window:

$$w(n) = \begin{cases} 1 & 0 \le n \le M-1 \\ 0 & otherwise \end{cases} \quad (7)$$

2. Triangular (Bartlett) window:

$$w(n) = \begin{cases} 1 - 2\left|\frac{n-\frac{M-1}{2}}{M-1}\right| & 0 \le n \le M-1 \\ 0 & otherwise \end{cases} \quad (8)$$

3. Hanning window:
Causal:

$$w(n) = \begin{cases} 0.5 - 0.5cos\left(\frac{2\pi n}{M-1}\right) & 0 \le n \le M-1 \\ 0 & otherwise \end{cases} \quad (9)$$

Non-causal:

$$w(n) = \begin{cases} 0.5 + 0.5cos\left(\frac{2\pi n}{M-1}\right) for\ |n| & 0 \le M-1 \\ 0 & otherwise \end{cases} (10)$$

4. Hamming window

Causal:

$$w(n) = \begin{cases} 0.54 - 0.46cos\left(\frac{2\pi n}{M-1}\right) & 0 \le n \le M-1 \\ 0 & otherwise \end{cases} \quad (11)$$

Non-causal:

$$w(n) = \begin{cases} 0.54 + 0.46cos\left(\frac{2\pi n}{M-1}\right) for\ |n| & 0 \le M-1 \\ 0 & otherwise \end{cases}$$

(12)

5. Blackman window

Causal:

$$w(n) = \begin{cases} 0.42 - 0.5cos\left(\frac{2\pi n}{M-1}\right) + 0.08cos\left(\frac{4\pi n}{M-1}\right) & 0 \le n \le M-1 \\ 0 & otherwise \end{cases} (13)$$

Non-causal:

$$w(n) = \begin{cases} 0.42 + 0.5cos\left(\frac{2\pi n}{M-1}\right) + 0.08cos\left(\frac{4\pi n}{M-1}\right) for\ |n| & 0 \le M-1 \\ 0 & otherwise \end{cases} (14)$$

6. Kaiser window.

$$w(n) = \begin{cases} \dfrac{I_0\left[\propto\sqrt{\left(\frac{M-1}{2}\right)^2 - \left(n - \frac{M-1}{2}\right)^2}\right]}{I_0\left[\propto\left(\frac{M-1}{2}\right)^2\right]} & 0 \le n \le M-1 \\ 0 & otherwise \end{cases} (15)$$

Results and Discussion

The results of the simulation is depicted in the figures below using Hamming with N=25 and Kaiser with N=25 and ∝=4.

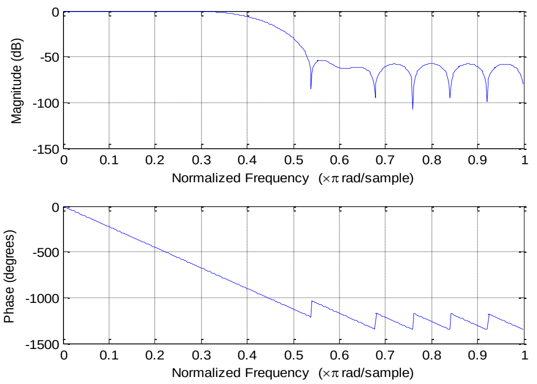

Fig. 1a: Frequency response (magnitude and phase) of lowpass filter using Hamming window

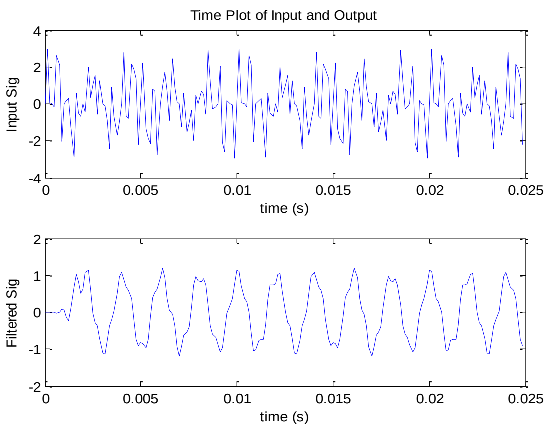

Fig. 1b: Time-domain diagram for Hamming window before and after filtering.

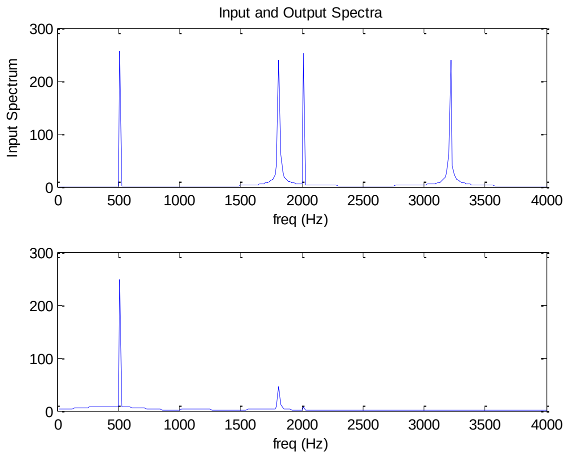

Fig. 1c: Frequency-domain diagram for Hamming window before and after filtering

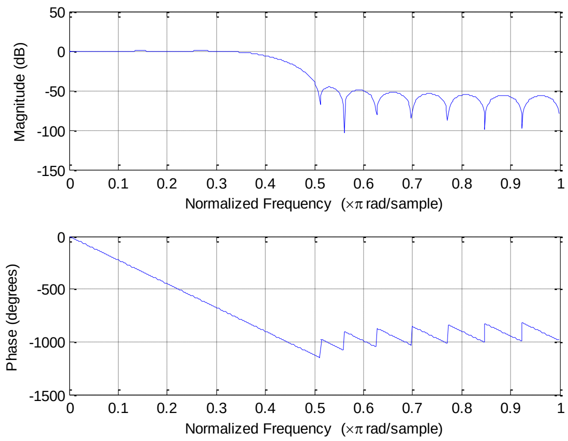

Fig. 2a: Frequency response (magnitude and phase) of lowpass filter using Kaiser window

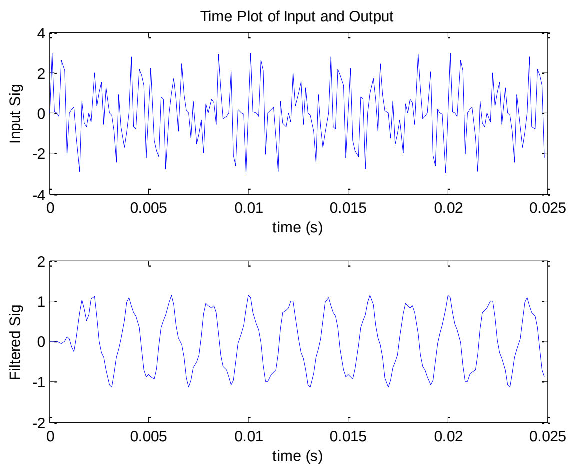

Fig. 2b: Time-domain diagram for Kaiser window before and after filtering.

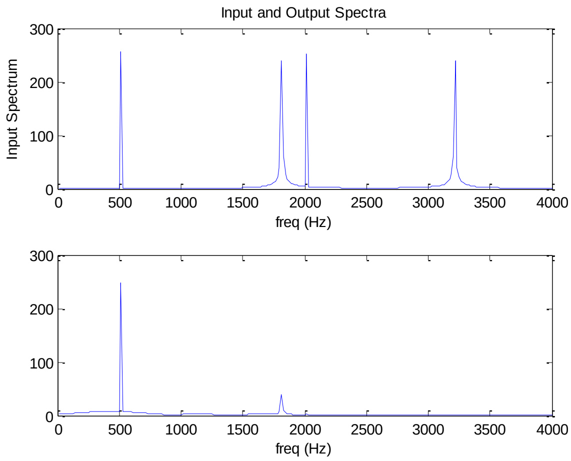

Fig. 2c: Frequency-domain diagram for Kaiser window before and after filtering

The magnitude of the frequency response of the filter designed using Hamming window in Fig.1 generates less oscillation in the sidelobes than the Kaiser window (Fig. 2) but that in turn increase the sidelobe peak and mainlobe width. Meanwhile the additional parameter $\propto$ in the Kaiser window reduces the sidelobe level and main lobe peak as shown in Fig. 2. Thus Kaiser window is the preferred window for the design of FIR digital filter as shown by Kumar (2014) and Ferdous (2013).

Conclusion

The result of the FIR digital filter designed in this paper shows that as compared with Hamming window, Kaiser window is the favorite window function to be employ in designing FIR digital filter. The additional parameter in Kaiser allows adjustment of compromise between the overshoot reduction and transition region width spreading.

References

Avci K 2016. Design of FIR filters using exponential–Hamming window family. *Turkey J. of Elect. Eng. & Comp. Sci..* 24: 2513 – 2524.

Dogra S & Sharma N 2014. Comparison of different techniques to design of filter. *Inter. J. of Comp. Applications.* 97(1). ISSN: 0975-8887.

Ferdous J & Ali S 2013. A Survey report for performance analysis of finite impulse response digital filter by using different window techniques. *Inter. J. of Research inEngr & Tech.* 2(12). eISSN: 2319-1163.

Gupta S & Panghal A 2012. Performance analysis of fir filter design by using rectangular, hanning and hamming windows methods. *Inter. J. of Advanced Research inComp. Sci. & Software Engr.* 2(6). ISSN: 2277 128X.

Kumar KS, Yazdanpanah B & Raju, GSN2014. Performance comparison of windowing techniques for ECG signal enhancement. *Inter. J. of Engineering Research*. 3(12): 753-756. ISSN:2319-6890 (online),2347-5013(print).

Nouri M, Sajjadi S & Falahati GA 2011. An improved window based on cosine hyperbolic function. *Multidisciplinary J. in Sci. & Tech., J. of Selected Areas in Telecom*.

Proakis JG & Manolakis DG 2011.Digital Signal Processing: Principles, Algorithms and Applications. Pearson Prentice-Hall, 4th edition.

Saiful I, Shariful I, Rahman SK, Ferdous NS & Jyoti JS 2014. Design of FIR filter using hamming window. *Inter. J. of Emerging Research in Mgt &Tech.* 3(2). ISSN: 2278-9359.

Saramaki T 1993. Finite impulse response filter design. In: Mitra K & Kaiser JF (Eds.) Handbook for Digital Signal Processing. Wiley & Sons, New York, NY, USA.

Tiwari A, Thakre V & Markam K 2014. Review: Performance analysis of FIR digital highpass filters. *Inter. J. of Comp. & Comm. Eng. Research.* 2 (2). e.ISSN: 2321-4198.