ORIGINAL RESEARCH ARTICLE | UHD JOURNAL OF SCIENCE AND TECHNOLOGY

Study the Characteristics of Finite Impulse Response Filter Based on Modified Kaiser Window

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ABSTRACT

Finite impulse response (FIR) plays an important part between all other types of filters. There are many types windows used to design of FIR filters. Most important types are as follows: Hanning, hamming, rectangular, triangular, Blackman, Kaiser, etc. The characteristics of these filters depend on the number of generated coefficients in addition to the side lobes of the filter spectrum. The aim of this work is to study and evaluate Kaiser Window type depends on the variation of its factors applied for resizing the impulse response to reach a suitable size the filter. Kaiser Window is an important filter window that can be used to get many types of windows depending on their parameters. The proposed filter approach is designed and implemented through mixing of many filter factors. The filter characteristics are achieved using different values of filter size and attenuation. The implementation of the proposed Kaiser filter window provides an adequate and easy way to measure the window coefficients and maximum side lobe levels. The benefit of Kaiser Window that you can generate many types of window depending on the parameters change.

Index Terms: Filter Coefficients, Finite Impulse Response Filter, Kaiser Filter, Sidelobes

1. INTRODUCTION

There are two main categories of filters; infinite impulse response (IIR) and finite impulse response (FIR) [1]. These two categories of filters can be implemented for one-dimensional signals and two-dimensional (2D) signals, these filters can be implemented through software and hardware algorithms [2]. In addition, these can be implemented through time domain or frequency domain. Many of filter algorithms are implemented by hybridizing both software and hardware to achieve flexibility and real-time processing [3]. According to the characteristics of the impulse response, filters can be divided into four types; low-pass filter (LPF) that passes low frequencies, high-pass filter that passes high frequencies, band-pass filter that passes a band of frequencies, and band-stop filter that stop a band of frequencies [4]. The implementation of filter is realized through the convolution process between the input signal and the impulse response. To generate a FIR filter, the coefficients of time domain filter must be limited in number by multiplying by a window function of a finite width [5]. The basic idea of applying windows through the design of filters is to truncate the sequence of the filter to be limited to certain values. Various types of windows are used for resizing the signal into limited values [6].

Many algorithms are implemented considering the factors of FIR filters to frequency domain specifications [7]. A narrow band LPF with good filter gain is introduced [8].

Access this article online

DOI: 10.21928/uhdjst.v1n2y2017.pp1-6

E-ISSN: 2521-4217

P-ISSN: 2521-4209

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Received: 14-05-2017 Accepted: 19-06-2017 Published: 29-08-2017

A hardware digital filter is implemented based on minimizing of computational expenses [9]. An adaptive analog filter is introduced based on echo cancelation [10]. An efficient pipelined filter is implemented using many structures [11]. A wideband efficient linear phase filter is implemented [12]. A high quality frequency domain applied in analog speech scrambler is reconstructed [13]. An efficient filter design based on FIR structure is implemented [14]. An optimized filter design technique is achieved [15]. A filter bank structure based on low complexity is implemented [16]. A programmable architecture applied programmable logic device (PLD) and FFT [17], then introduced PLD and digital signal processor [18]. Fast algorithm of filter design based on LUT is performed [19], then for 2D applications [20]. A real time digital filter processing based on hardware processor is realized [21].

2. RELATED WORK

Several researches have touched on this subject, the following are number of published works:

Mohindru *et al.* designed a new mathematical model to find the transfer function of LPF FIR based on Fourier transform and rotation angle. In contrast with the variation of the angle of rotation from 0 to $\pi/2$, then it is possible to find the width of the transition band and the attenuation of the band. By raising the length of the filter, you can find the frequency response of the digital filter. The response characteristics of FIR filter can be organized through the operation, by controlling only one parameter and remaining other coefficients fixed [22].

Kumar *et al.* designed a novel approach for the structure of filters with variable band of finite linear impulse response band using different polynomials. In this work, the transfer function of a variable bandwidth filter, which is a linear combination of the linear phase filters with fixed coefficients and previous polynomials, is operated separately as control parameters to control the bandwidth of the filter. The proposed method also provides a better performance of tunable BPFs with larger filter captures compared to previously published results [23].

Pak *et al.* proposed a new approach to manage the horizon size of the nonlinear FIR filter. The proposed approach is to carry out an estimation of the state through a FIR filter bank. In this approach, the state estimate is obtained by weighting the average of several estimates of a bank of impulse responses using different horizon sizes. The filter size used for this approach is selected to maximize the likelihood function. The simulations of this approach gives good results compared with the conventional approach [24].

Leighton P. Barnes, George C. Verghese, studied the relationship between the Wiener filter and the coherence function, then defines the causal relationship between the wide-sense stationary (WSS) process. This causal consistency is interpreted in a modeling context and is used to show that a measure depends on the frequency of causality it can and cannot represent. In addition, the Wiener causal convergence of FIR filters with the Wiener causality of IIR filter is studied because the length of the filter passes the infinity. The main results show Lp convergence frequency responses under certain conditions of continuity support in the power spectra, and the upper asymptotic limits for the convergence error. Then, under the same conditions, the uniform convergence of AR approximations shows power spectra as the order model tends to infinity [25].

Huang *et al.* presented a filter design in the closed form based on the compensation of transfer characteristics. In this approach, a new filter design based on a convolution of window is presented and the relationship between window spectrum and filter performance is developed. Then, a schematic design of three step filter is designed these are designing an irregular filter, designing a compensation filter, and the sum of filter. This scheme can be simplified into a closed form characterized by two analytical formulas by merging the intermediate steps [26].

Pak *et al.* proposed an efficient nonlinear FIR. The proposed least square extended FIR filter is derived using a least squares criterion and one through the property. This approach is a special filter designed for the constant speed motion model and does not require noise information such as Gaussian noise covariance. If the noise information is very uncertain, this approach can provide a constant performance, whereas the non-linear estimators of the existing state, such as the extended Kalman filter and particle filter, degradation of performance often in the same condition. The simulations results indicated the robustness of this approach against the uncertainty of the noise model [27].

Boukharouba implemented a new technique for the FIR filters, where the desired frequency response is a smooth rectangular function. In this approach, go directly to smooth out the ideal desired response in the frequency domain not in time domain. The impulse response of the filter is a sampled version of the inverse Fourier transform of the frequency response. This approach achieved the best performance results of filter specifications compared with the results of other works [28].

3. KAISER WINDOW DESIGN

Kaiser Window have many parameters that affect the overall filter design. The ripple parameter α enabling the designer to trade-off the transition and ripple. It is defined in the interval – M ≤ n ≤ M, and otherwise it is zero. The equations of the Kaiser Window and their parameters are shown below:

Measuring the ripple (δ) factor and then measure the attenuation factor (A) as the following:

$$A = -20 \log_{10} \delta \tag{1}$$

Measure the value of α according to the value of attenuation (A) factor as the following:

$$\propto = 0.1102\,(A-8.7)\quad A > 50$$

$$\alpha = 0.5842\,(A-21)^{0.4} + 0.07886\,(A-21)\quad 21 \le A \le 50 \tag{2}$$

$$\alpha = 0\quad A < 21$$

Find the window size (M) according to the calculated attenuation value (A) and the transition width (TW) as below:

$$M \ge \frac{(A-7.95)}{28.72 * \mathrm{TW}} \tag{3}$$

Kaiser Window is calculated using the following equation:

$$w(n) = \frac{I_0(\alpha\sqrt{1-(n/M)^2}}{I_0(\alpha)} \quad -M \le n \le M \tag{4}$$

And the zero order Bessel function I_0 (α) is measured as below:

$$I_0(x) = 1 + \sum_{n=1}^{\infty}\left[\left(\frac{x}{2}\right)^n \frac{1}{n!}\right] \tag{5}$$

4. PROPOSED FIR FILTER ALGORITHM

There are many parameters affected the filter design, these parameters are ripple factor, attenuation factor, TW, number of coefficients, and window type in addition to the required accuracy.

The length of the filter concentrated on the number of values of impulse response samples in the FIR filter, then the impulse response is varied from $n = 0$ to $n = M-1$, where M is the filter length.

The proposed filter approach is demonstrated in Fig. 1 and is shown below:

Step 1: Ripple (δ) measurement, considering pass band ripple and stop band ripple.
Step 2: Attenuation measurement depending on the ripple value.
Step 3: TW measurement depending on the number of coefficients.
Step 4: Filter length measurement based on the TW.
Step 5: Bessel function $I_0(x)$ measurement depending on the calculated parameters.
Step 6: Kaiser Window elements measurement.
Step 7: Filter coefficients measurement depending on the given cutoff frequency.
Step 8: Filter coefficients truncation based on the required Kaiser Window.
Step 9: Realize the filter according to the measured parameters.

5. RESULTS AND DISCUSSIONS

To realize and study the FIR filter characteristics, it is better to generate different types of windows. These windows are; Barthann, Bartlett, Blackman, Blackman-Harris, Bohman, Chebyshev, Flattop, Gaussian, Hamming, Hanning, Nuttall, Parzen, Rectangular, Taylor, Triangular, Tukey, and Kaiser. To study the shape characteristics of these windows, these are generated using different length number of window. Fig. 2 shows these windows in time domain and frequency domain generated using 20 values. Fig. 3 shows these windows in time domain and frequency domain generated using 50 values. These two figures illustrated that the time domain shape varying from rectangular shape to different type of Sinc function shape or hat shape. In addition, these two

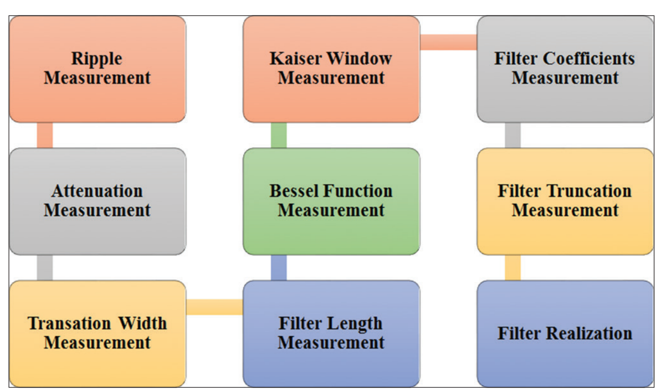

Fig. 1. Proposed finite impulse response filter algorithm

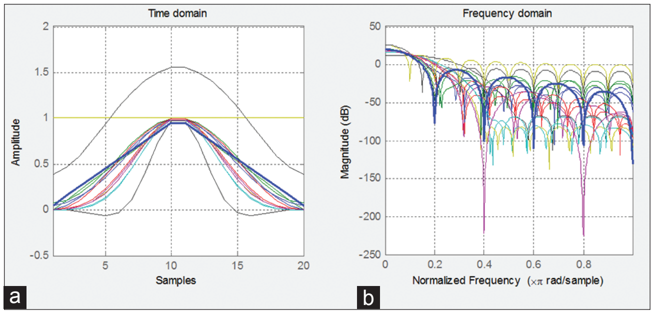

Fig. 2. (a and b) Time domain and frequency domain representation of 20 values

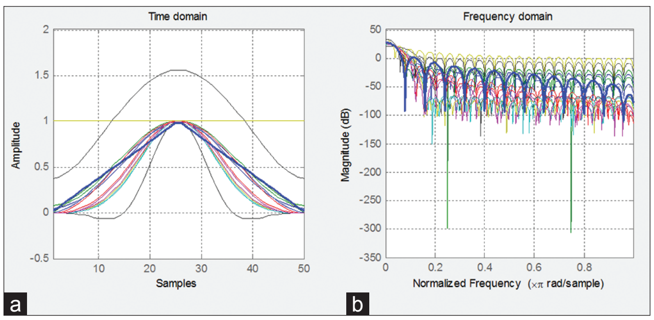

Fig. 3. (a and b) Time domain and frequency domain representation of 50 values

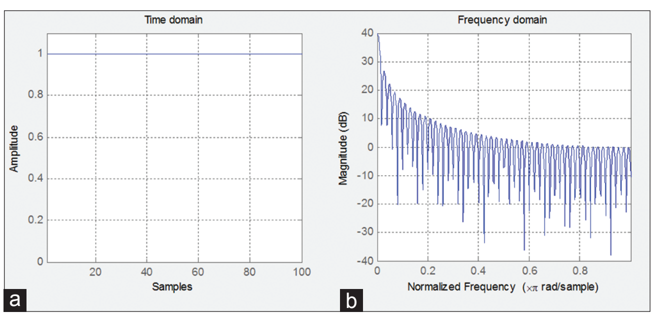

Fig. 4. (a and b) Kaiser characteristics beta = 0

figures in frequency domain show that the relative sidelobe concentrated on −26.8 dB where the latest on appears on about −75 dB.

Kaiser Window is an important filter window and it have weighting factor parameter. The value of beta is the parameter of Kaiser Window that affects the sidelobe attenuation in the frequency domain of the window. Fig. 4 indicates the Kaiser characteristics when beta equal to zero in which indicated that the leakage factor equal to 9.26% and the sidelobe attenuation equal to −13.3 dB. Fig. 5 indicates the Kaiser characteristics when beta equal to 0.5 in which indicated that the leakage factor equal to 8.48% and the sidelobe attenuation equal to −13.6 dB. Table I demonstrates the Kaiser parameters of low beta values in which shows the variation of beta value from 0 to 1 with the increment of 0.1 in which the leakage factor varying from 9.26% to 6.53%. In addition, the values of sidelobe attenuation is varied from −13.3 to −14.7 dB, with main lobe width at −3dB equal to 0.017578 for all values of beta.

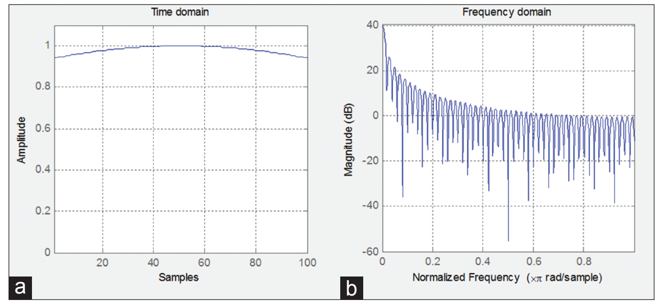

Fig. 5. (a and b) Kaiser characteristics beta = 0.5

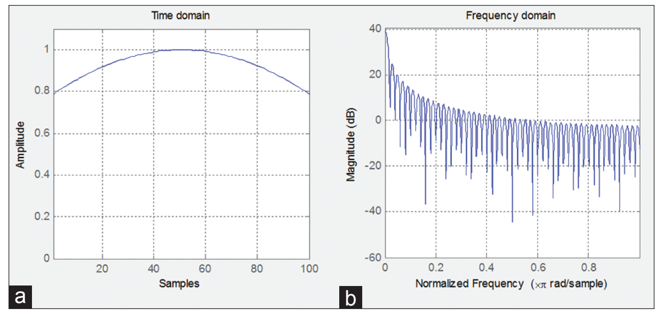

Fig. 6. (a and b) Kaiser characteristics beta = 1

TABLE I
Kaiser Parameters of Low-beta Values

Beta value	Leakage factor (%)	Sidelobe attenuation dB	Main lobe width −3 dB
0	9.26	−13.3	0.017578
0.1	9.22	−13.3	0.017578
0.2	9.13	−13.3	0.017578
0.3	8.97	−13.4	0.017578
0.4	8.75	−13.5	0.017578
0.5	8.48	−13.6	0.017578
0.6	8.16	−13.8	0.017578
0.7	7.80	−14.0	0.017578
0.8	7.40	−14.2	0.017578
0.9	6.98	−14.4	0.017578
1.0	6.53	−14.7	0.017578

Fig. 6 indicates the Kaiser characteristics when beta equal to one in which indicated that the leakage factor equal to 6.53% and the sidelobe attenuation equal to −14.7 dB. Fig. 7 indicates the 10 in which indicated that the leakage factor equal to 0% and the sidelobe attenuation equal to −74.1 dB,

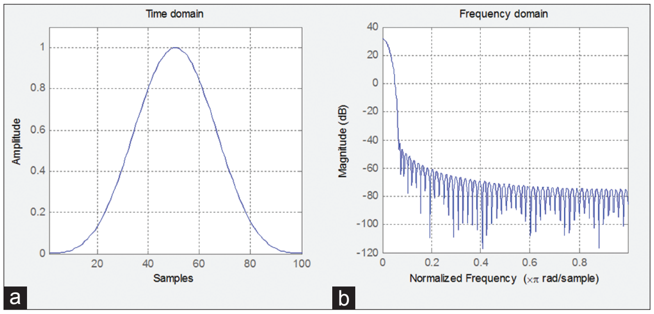

Fig. 7. (a and b) Kaiser characteristics beta = 10

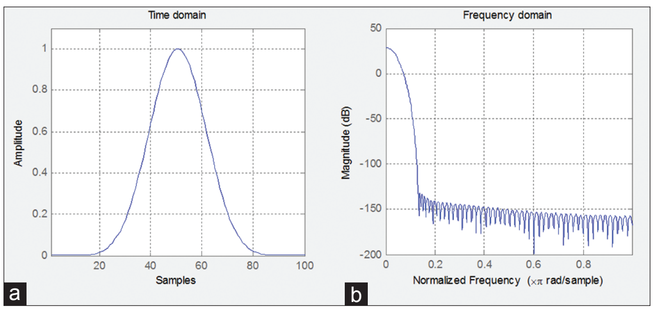

Fig. 8. (a and b) Kaiser characteristics beta = 20

in this case, the shape of the window is approach to a hat. Fig. 8 indicates the Kaiser characteristics when beta equal to 20 in which indicated that the leakage factor equal to 0% and the sidelobe attenuation equal to −154.9 dB, in this case, the shape of the window is stretch hat. Table II demonstrates the Kaiser parameters of high-beta values in which shows the variation of beta value from 1 to 20 with the increment of 1, in this case, the leakage factor varying from 6.53% to 0%, considering that the zero value of leakage factor starting when beta equal to 6. In addition, the values of sidelobe attenuation is varied from −14.7 to −154.9 dB, with main lobe width at −3 dB varying from 0.017578 to 0.048828 corresponding to the variation of the value of beta.

6. CONCLUSION

In this work, an adequate algorithm of Kaiser parameters is measured to be ready for the filter implementation. The framework to study the characteristics of FIR filter based on modified Kaiser Window is implemented through two scenarios. Two scenarios are implemented; First scenario used low Beta values and second scenario used high Beta values. The obtained results indicated that the implemented Kaiser Window filter has good characteristics and stable. The comparison of sidelobe attenuation and the values of beta it is possible to select an adequate and optimal values to be used for the filter design. The obtained results indicated that an improvement of the filter characteristics during the attenuation value will increase. In addition, an improvement happens on the filter performances when filter length will increase, that is increase the number of coefficients. The implemented Kaiser Window indicated that is easy to generate many types of windows through simple modification. The sidelobe attenuation of Kaiser Window is about −14 dB when beta value is 1 and this value will increase vastly up to −154.9 when beta reach to 20.

TABLE II
Kaiser Parameters of High-beta Values

Beta value	Leakage factor (%)	Sidelobe attenuation dB	Main lobe width −3 dB
1	6.53	−14.7	0.017578
2	2.47	−18.6	0.019531
3	0.61	−24.0	0.021484
4	0.12	−30.4	0.023438
5	0.02	−37.1	0.025391
6	0.00	−44.0	0.027344
7	0.00	−51.0	0.029297
8	0.00	−58.4	0.031250
9	0.00	−66.1	0.033203
10	0.00	−74.1	0.035156
11	0.00	−82.3	0.035156
12	0.00	−90.3	0.037109
13	0.00	−98.1	0.039063
14	0.00	−105.8	0.041016
15	0.00	−113.8	0.041016
16	0.00	−122.3	0.042969
17	0.00	−130.4	0.044922
18	0.00	−138.7	0.044922
19	0.00	−147.1	0.046875
20	0.00	−154.9	0.048828

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