A UNIQUE APPROACH ON THE PERFORMANCE ANALYSIS IN DSP USING THE MSK FIR WINDOW

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ABSTRACT

The impulse response is eliminated by using the windowing approach for resizing the filters. The two most widely used window approaches are the Keystone and Kaiser windows. By using those specific windows, we may realize a variety of window kinds based on their attributes and ripple factor specification. In order to compare its parameters with these two-parent windows KW and TW, we have presented a novel modified window technique called MSK FIR window. By combining KW and TW, the new MSK FIR window that is being proposed is realized. For 101 window size, the new MSK FIR Windows' attributes have been confirmed. It goes without saying that performance will improve and become more reliable following the installation of FIR filter with new MSK FIR window.

Keywords: Tukey Window (TW), Kaiser Window (KW), Impulse response (IR), Adaptive Filter, Finite Impulse Response (FIR) and MSK, digital signal processing (DSP), Fourier transformation (FT), window function (WF)

1. INTRODUCTION

For digital filter creation, the window method is quick, easy, and reliable—but often not as reliable as it could be. This trait makes it very simple to grasp in terms of convolution theory for Fourier transforms, which makes studying FT theorems and windows for spectrum examination later on useful. Smooth curves, non-negative rectangle and triangle functions, and other distinct function types that are frequently utilized in analysis are the window functions used in predictable applications.

Windowing amplitude rectifies the input signal to increase the amount of spectral leakage on bucket signals and decrease it on off-bucket signals. Windowing enhances leakage by varying the amplitude of sample signals at start and finish of window. The input signal is multiplied by a specific windowing function to accomplish windowing. An input signal is something that can have multiple dimensions or be a complex number. The complex multiplication is needed in order to convolution complex values.

The window approach simply involves using $w(n)$or another appropriately selected window function to window a $h(n)$ theoretically ideal filter IR.

$$h_w(n) = h(n).w(n), \quad n \in z \quad (1)$$

Windowing a simple waveform enables its FT to have a non-zero values at frequencies that are varied from ω. The frequencies tend to be highest close to ω and lowest at substantial distances from ω. When two sinusoidal signals with distinct frequencies are present, leakage might regulate and provide spectral form characterization for them. When one sinusoidal signal's amplitude is too little compared to another sinusoidal signal, leakage will be minimized if their frequencies are different. In other words, leakage from the higher amplitude component may conceal the function's spectral component. However, even though the two sinusoidal signals have the same amplitude, leakage may be desired to interact while the frequencies were close to one another, making the signals unresolvable.

Among the prominent functions are windowing functions. The precise meaning of this Greek term is eliminating the feet. That is the technical name for using a function with zero value outside of a specific interval to changesignal form. To smooth out the transitions, a tapering function is used. There are different kinds of windows, including: Rectangular, Triangular, Kaiser, Tukey, Chebyshev, Hamming, Bohman, and Modified Bartlett-Hann window and the minimum 4-term specified by Nuttall Blackman-Harris, Gaussian, Bartlett, Hann (Hanning), Taylor, Gaussian, and Flat Top weighted windows are among the windows that can be generated. Other windows can also be generated based on production of coefficients.

The characteristics of FIR digital filter include an arbitrary amplitude-frequency characteristic, real-time steady signal processing requirements, a very stable, computationally demanding, and precisely linear phase. As such, it finds extensive applicability in many DSP applications [1, 2].

2. LITERATURE REVIEW

The two types of WFs, they are Fixed and Adjustable. The fixed WF that are most frequently used are the Hann, Blackman, Blackman, and rectangular windows. The Kaiser window, on the other hand, has an adjustable window feature. These various widows are employed in spectral performance studies and digital FIR filter design [3].

With MATLAB's powerful computational capabilities, the design and simulation survey of a digital filter may be completed fast and effectively. Users not only feel comfortable with the digital filter's performance characteristics, but also find the calculation process easier because of its simplification. Simulink is a MATLAB signal processing box with robust functionality and an intuitive user interface. When used in tandem with MATLAB, Simulink facilitates the easier and more efficient construction of simulations for users [4].

Rectangular windows, triangle windows, Han windows, Hamming windows, Blackman windows, and Kaiser windows are the six types of basic window functions. The fundamental principle behind all WF design methods is to choose a filter based on appropriate and ideal frequency features, then truncate its IR to produce a linear-phase, cause-and-effect FIR filter. Thus, choosing a suitable ideal filter and window function is the main goal of this strategy [5].

P. P. Vaidyanathan and Yuan-Pei Lin [6] suggested restricting the prototype filters' search to class of filters that could be found using Kaiser windows. One parameter optimization is all that remains of the design procedure. A demonstration will be provided to demonstrate that excellent designs can still be obtained despite search boundaries. The prototype filter's design is approached as an optimization problem for cutoff frequency in the Kaiser window design. A plan based on carry save array multiplier's construction was put out by Paraskevas Kaliva et al. [7], in which each cell performs the bit-level calculation of a FIR filter. The latency caused by this structure is independent of the quantity of filter taps. Compared to alternative techniques based on discrete multipliers, the suggested scheme uses less hardware and is pipelined at bit level as well as systolic at cell level.

Two kinds of linear phase FIR filters with frequency response masking were introduced by Hakan Johansson [8] for use in decimation and interpolation by arbitrary integer factors. Low complexity sharp transition linear phase FIR interpolation and decimation filters are the filters that are suggested. The locations of stopband and passband boundaries can be chosen with greater freedom and less complexity with the new ones.

The design of VDFs with discrete coefficients was studied by Hai Huyen Dam et al. [9] in an effort to achieve low complexity and effective hardware implementation. With a limit on complete number of power-of-two for filter coefficients, the filter coefficients have been stated as sum of signed power-of-two terms. In order to produce an optimal quantized solution, an effective design strategy is suggested that includes an enhanced technique for managing VDF coefficients quantization for least-squareand min–max criteria.

3. FILTER DESIGN

There are numerous digital filter types and classification schemes. Four categories can be used to group common filter types.

1: Band Pass Filter (BPS)
2: High Pass Filter (HPF)
3: Low Pass Filter (LPF)
4: Band Stop Filter (BSF)

Basically, there are 2 kinds of filters depending on the features of the time domain and IR.

1-Finite Impulse Response (FIR)
2-Infinite Impulse Response (IIR).

The approaches for IIR filter design are developed from transformation of filters, specifically in continuous time, whereas techniques for FIR filter design are entirely based on discrete-time domain. Other Producing a sufficient number of filter coefficients to arrive at an ideal filter design is the primary goal of FIR filter design (based on Windowing techniques).The following procedures can be utilized to execute FIR filter design: Filter type selection Filter architecture, Windows method design, filter analysis, filter implementation, filter structure, and computation of IR. The properties of time, frequency domain, or both domains are the starting point for FIR filterdesign. The demand for the frequency domain filter design is centered on the magnitude response, while time domain filter design seeks to produce a distinct IR. Figure 1 illustrates the stop band and pass band ripple. This LPF's impulse response is determined by the sinc function.

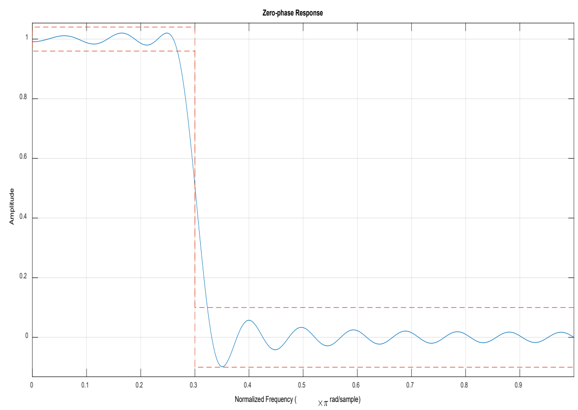

Fig1 .Normalized LPF

SINC FUNCTION

The sine function sinc(x) is also called as sampling function. This function is commonly encountered in fields of DSP and FT theory. For this function, two conditions have been applied. The Normalize sinc function expressed below

$Sinc(x) = \sin x / x$

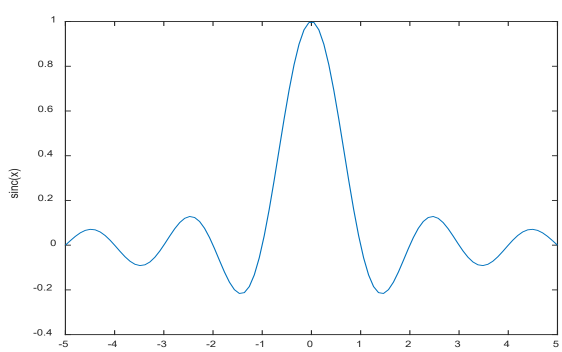

Fig.2. plot for sinc function

TUKEY WINDOW

A tapering cosine rectangle window, or Tukey window, is defined as having the initial and final r/2 percent of the samples equal to cosine functionfragment. The unique features are provided by the window function's characteristics. A function that reduces to zero until the window's end is included in the taper function. designed to lessen the impact of breaks in the time series between its beginning and conclusion. While it is not possible to completely prevent spectral leakage, it can be effectively reduced by changing the taper function's shape in order to lessen the significant interruptions near the window margins. The cosine taper might be expressed mathematically as taper ratio r. The cosine window attempts to reset the data to zero at boundary conditions, although this does not considerably lower the window transform's level.

$$W(n) = \begin{cases} \frac{1}{2}\left\{1+\cos\left(\frac{2\pi}{r}[n-r/2]\right)\right\}, & 0 < n < r/2 \\ 1, & \frac{r}{2} \le n < 1-r/2 \\ \frac{1}{2}\left\{1+\cos\left(\frac{2\pi}{r}[n-1+r/2]\right)\right\}, & 1-\frac{r}{2} < n < 1 \end{cases} \quad (2)$$

The cosine-tapered section length divided by total window length is known as the parameter r. Tukey window: segments of a phase-shifted cosine with period 2r = 1, where r is a real number among 0 and 1, make up half of the window's length. If we enter r ≤ 0, a rectangle window will appear.

We enter r ≥ 1, we have Hann window r = 0.5 is the Tukey window function's default value.
The following consequences will be caused by this tapering: For the aforementioned cosine tapers, it will decrease the spectral power leakage from a spectral peak to distant frequencies and coarsen the spectral resolution by factor (x – r/2).

KAISER WINDOW

The Kaiser window is a well-known window because of its useful features. The filter length and MLW have the opposite relationship. The side lobe attenuation is a type of WF that is independent of filter's length. Pro Harris provided a thorough explanation of the many kinds of window functions and their attributes. In order to achieve the ideal transition band and reduce the MLW, filter length should be significantly increased. The Kaiser has nominated a group of windows with properties that resemble the elliptical spheroidal wave functions. The KW, a well-known family of windows, is distinguished by its designated purpose:

$$w(n) = \frac{I_0\left(\pi\alpha\sqrt{1-\left(\frac{2n}{N-1}-1\right)^2}\right)}{I_o(\pi\alpha)} \quad (3)$$

Furthermore, two significant parameters that can be utilized as a primary component of the filter design are the transition width (TW) and the ripple parameter (δ).

The following lists the steps of the Kaiser Window algorithm:
Step 1: using the minimum ripple value (δ) modification to calculate the attenuation value.

$$\alpha = -20\log_{10}(\delta)$$

where $\delta = \min(\delta_s, \delta_p)$

Step 2: window parameterβ estimation as per attenuation value.

$$\beta = 0.1102(A-8.7), \qquad A > 50$$

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

$$\beta = 0.0 \qquad A < 21$$

Step 3: transition bandwidth (TW) estimation as per number of well-defined unique coefficients that differsamong pass and stop band.

$$TW = \omega_s - \omega_p \quad (4)$$

Step 4: cutoff frequency (ω_c) estimation.

$$\omega_c = (\omega_s + \omega_p)/2 \quad (5)$$

Step 5: window size(N) estimation.

$N \geq (\frac{\alpha - 7.98}{2.872 * TW})$

Step 6: Bessel function - Io(x) estimation.

$$I_0(x) = \sum_{n=0}^{\infty} \left[\frac{(\frac{x}{2})^n}{n!} \right]^2 \quad (6)$$

Step 7: KW estimation in well-defined interval of n.

$$w(n) = \frac{I_0\left(\pi\alpha \sqrt{1 - \left(\frac{2n}{N-1} - 1 \right)^2} \right)}{I_o(\pi\alpha)} \quad (7)$$

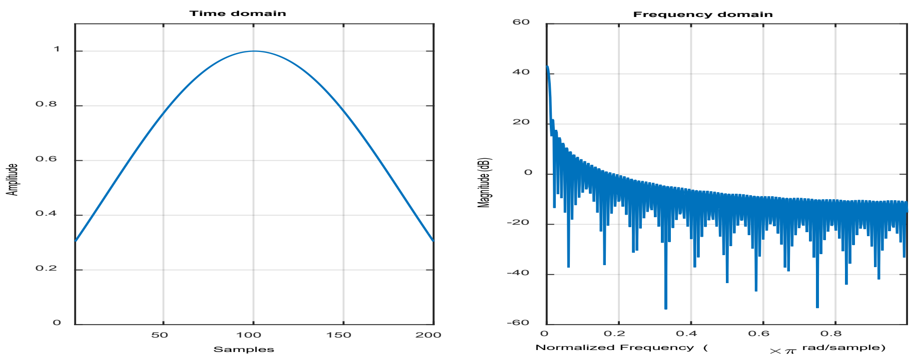

Fig.3.200 point Kaiser window

We could estimate the comparison of KW function at numerous values of β&N in below table

Variable parameter β	window function length (N)	Time domain		Frequency domain	
		Min Amp	MaxAmp.	RSLA dB	MLW
0	38	1	1	-13.3	0.042969
	48			-13.3	0.035156
1	38	.08	1	-14.7	0.046875
	48			-18.7	0.035156
2	38	.04	1	-18.8	0.050781
	48			-18.7	0.0390630
3	38	.02	1	-24.5	0.054688
	48			-24.3	0.0429690
4	38	.01	1	-31.0	0.625000
	48			-30.8	0.046875

Table.1. Comparison of KW at diversebeta and N values

4. WINDOW MEASUREMENTS

We will discuss a few of the key variables that may be adjusted to increase the effectiveness of the various kinds of window filters.

Coherent Gain (CG):However, the FT-computed signal magnitudes will drop whenever a window is applied to signal. The Coherent Power Gain measures the signal's attenuation caused by the window function. A window w(k) of length N has CG, which is determined by:

$$CG = \frac{1}{N} \sum_{n=0}^{N-1} w(n) \quad (8)$$

Equivalent Noise Bandwidth (ENBW):The ENBW of a filter is its bandwidth when it passes the same amount of power through a perfect rectangular filter as cumulative bandwidth of channel selective filters in receiver. As LPF's cutoff frequency increases, ENBW rises.

$$ENBW = \frac{N\sum_{n=0}^{N-1}\left|w(n)\right|^2}{\left(\sum_{n=0}^{N-1}w(n)\right)^2} \quad (9)$$

Processing Gain (PG): In DSP processes are selected to enhance specific signal quality approaches, by adjusting the disparities between the signals and the corrupting impacts. When additive random noise corrupts a sinusoidal signal. The system's or window's PG, which may be determined using the provided equation

$$PG = \frac{1}{ENBW} \quad (10)$$

The PG also expressed in decibels:

$$PG = 10\log_{10}\frac{1}{ENBW} \quad (11)$$

Leakage Factor(LF):
The power ratio in the sidelobesto complete window power.
Relative Side Lobe Attenuation(RSLA):
The variance in height from main lobe peak to highest side lobe peak.
Main Lobe width(MLW) (-3db):
The MLW at 3db below main peak.

The Proposed FIR Filter Process
The efficient window strategy for FIR design that is suggested relies on choosing between Kaiser and Tukey windows based on a number of different factors.
Step 1: assume that the pass band ripple (δp) and the stop band ripple (δs) are equal.
Step 2: taking the pass band frequency (wp) into account.
Step 3: Determine the parameters that are needed to realize the window.
Step 4: determine the parameters for KW.
Step 5: Compute the TW parameters.
Step 6: based on a number of parameters, choose the preferred window type.
Step 7: Applying the filter coefficients.
Step 8: Concatenate the filter coefficients of chosen window function.

5. RESULTS AND ANALYSIS

The CG, PG, ENBW, LF, RSLA, and MLW are the measurement factors for the window that are displayed in Table 2. For both Kaiser and Tukey window functions, the suggested FIR filter algorithm is put into practice and tested at different attenuation levels. The chosen Tukey window's frequency and time domain representations for following values of alpha (α) are exposed in Figure 4: 0, 0.25, 0.5, 0.75, and 1. Anddifferent values of alpha alpha(α), we could discover the time and frequency domain representations of chosen KW: 0, 0.25*5, 0.5*5, 0.75*5, & 1*5. This is the Tukey window function's multiple of 5. The estimated time and frequency domain description of newly suggested window, which is dependent upon Kaiser and Tukey Windows, are shown. These comparison graphs are useful for determining filter efficiency when choosing an appropriate window function for a given application.

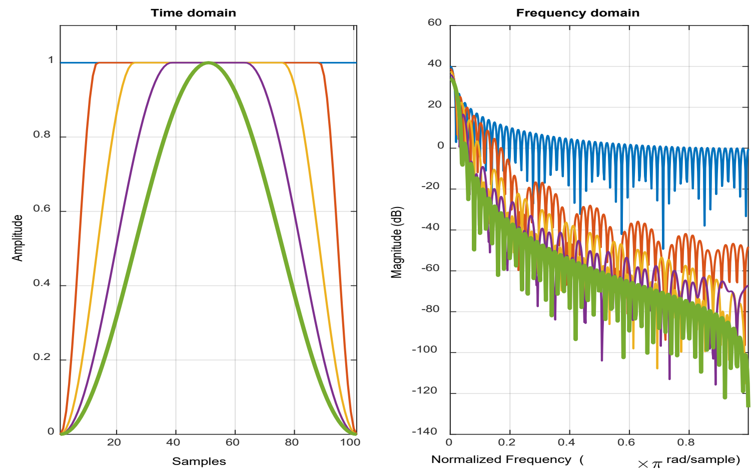

Fig.4. Tukey window

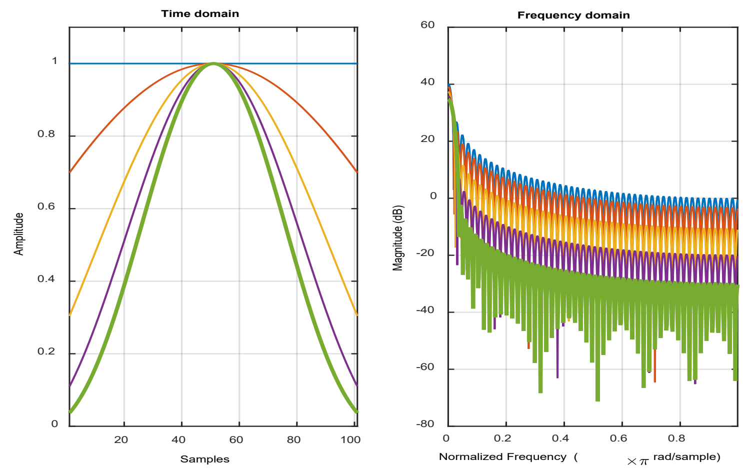

Fig.5.Kaiser Window

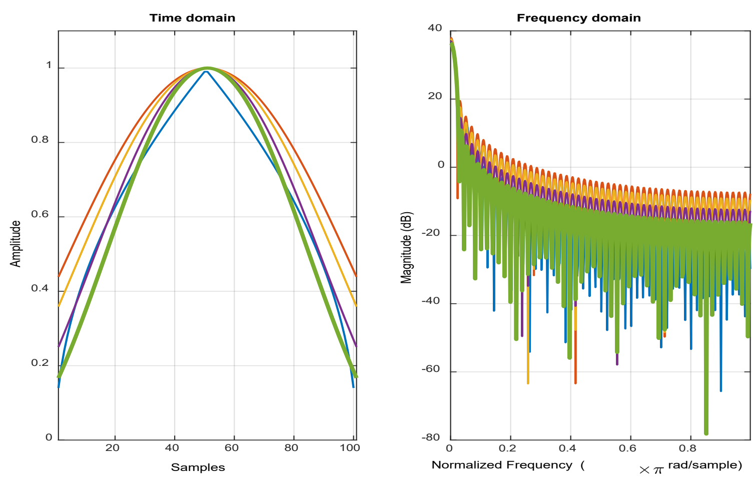

Fig.6. MSK FIR Window

Information	Value of α	CG	ENBW	PG (dB)	LF	RSLA (dB)
Tukey window	α1=0.00	1.0	1.0	0.0	9.26%	-13.3
	α1=0.25	0.866 3	1.1131	-0.465	6.49 %	-13.6
	α1=0.50	0.742 6	1.2344	-0.9147	3.61 %	-15.1
	α1=0.75	0.618 8	1.3736	-1.3786	1.15 %	-19.4

	α1=1.0	0.495 0	1.5150	-1.8041	0.05 %	-31.5
Kaiser window	α2=5*0.0	1.000 0	1.0000	0.000 0	9.26 %	-13.3
	α2=5*0.25	0.893 9	1.0107	-0.0463	5.40 %	-15.5
	α2=5*0.50	0.731 3	1.0912	-0.3791	1.28 %	-21.2
	α2=5*0.75	0.616 3	1.2268	-0.8877	0.18 %	-28.7
	α2=5*1.00	0.539 8	1.3708	-1.3698	0.02 %	-37.0
New MSK² Window	α1 & α2	0.750 0	1.0370	-0.1580	3.18 %	-19.0
	α1 & α2	0.777 4	1.0444	0.1886	2.70 %	-19.3
	α1 & α2	0.756 2	1.0654	-0.2753	1.88 %	-20.3
	α1 & α2	0.706 4	1.1062	-0.438	1.00 %	-22.9
	α1 & α2	0.654 8	1.1604	-0.6460	0.46 %	-27.0

Table.2.Window measurements for Tukey,Kaiser& MSK FIR Windows

6. CONCLUSION

The combination of KW and TW parameters yields the suggested efficient window technique MSK. This special technique can be used with the Kaiser and Tukey windows to obtain the predicted filter parameters. From these combinations of KW and TW parameters, we can obtain the suggested MSK window values. This work proposes a novel MSK window and examines the parameters of the Tukey and Kaiser windows correlatively. We measured and compared the parameters of the KW, TW, and proposed MSK window approach, including CG, ENBW, LF, PG, and RSLA, in this study comparing window functions. The further analysis of these variables and the development of the MSK Window technique demonstrate improvements in the filter's attenuation, and the final MSK filter findings demonstrate increased stability and efficiency.

CONFLICT OF INTERESTS

None.

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None.

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