

Comparative Performance of the Sign Windowed Filtering Technique with Han, Hamming, Kaiser and Black Manwindowed Filtering Technique in Artifact Removing

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ABSTRACT-Artifacts such as 50Hz power line noise should be removed from ECG signal of a patient in a hospital in order to guarantee correct clinical information concerning the patient. The technique for removing 50Hz power line interference artifact from human ECG was developed and implemented. The technique consists of sine window function for use in static filter coefficient. The corrupt signal was made to pass through Finite Impulse Response adaptive filter. Based on the developed window function, filter order of 100, sampling frequency of 1000Hz, pass band frequency of 40/60Hz and rejection frequency of 47.5/52.5Hz, processing of the electrocardiographic signal for the removal of 50Hz power line interference was extensively performed. The filter order, sampling frequency and pass band frequency were widely varied to determine their optimum values and they were ultimately determined to be 100, 900Hz and 40/60Hz respectively. The sine window function removed the targeted 50Hz noise from the ECG signal. When comparing the sine window with other windows, there was an improvement in signal quality over other windows with a SNR ratio of 33.98dB. The Kaiser window has SNR of 33.40dB while the Rectangular window has the least performance with SNR of 12.48dB.

KEY WORD: Transfer Function of FIR and Sine window function

Date of Submission: 09-03-2018

Date of acceptance: 24-03-2018

I. INTRODUCTION

Digital filter plays an important role in digital signal processing applications. Digital filters are widely used in digital signal processing applications, such as digital signal filtering, noise filtering, signal frequency analysis, speech and audio compression, biomedical signal processing and image enhancement etc. A digital filter is a system which passes some desired signals more than others to reduce or enhance certain aspects of that signal (Saurabh and Bhaduria, 2012). It can be used to pass the signals according to the specified frequency pass-band and reject the frequency other than the pass-band specification. In signal processing, a window function (also known as an apodization function or tapering) is a mathematical function that is zero-valued outside of some chosen interval. For instance, a function that is constant inside the interval and zero elsewhere is called a *rectangular window*, which describes the shape of its graphical representation. When another function or waveform/data-sequence is multiplied by a window function, the product is also zero-valued outside the interval: all that is left is the part where they overlap, the "view through the window". The areas of application of window function include: Spectral analysis, the design of an impulse response filter, beamforming and antenna design. There are many window functions that are being in every day design. Kaiser window is one-parameter family of window function used for digital signal processing. Kaiser-Bessel derived is an off shoot from Kaiser Window (Harris F.J, 1978). It is designed as modified discrete cosine transform. Kaiser window has the disadvantage of higher computational complexity due to use of Bessel functions in the calculation of the window coefficient (Mahrokh and Mahdi 2009). When designing FIR digital filter using Hamming window function, there is a provision of smaller lobe width and sharp transition band compare to Hanning (Er and Er 2015). The rectangular window is the simplest window, equivalent to replacing all but N values of a data sequence by Zeros, making it appear as though the waveform suddenly turns on and off (Aayushi and Chetna 2017). Hanning window named after Julius von Hann and it's similar in name form to Hanning window

(Shushank and Narinder 2014). The sine window function is a cosine without the $\pi/2$ phase offset. So, the sine window is sometimes called cosine window. The autocorrelation of a sine window produces a function known as the Bhman window. The triangular window is the 2nd order B-spline window and can be seen as the convolution of two $N/2$ width rectangular windows. The Fourier transform of the result is the squared values of the transform of the result is the squared values of the transform of the half-width rectangular window. In this research our interest is to embark on analytical study of these windows and compare them with sine window to be able to know its performance in artifact removing. The artifact to remove is 50Hz power line interference.

II. RELATED WORKS

Shahana et al (2007) presented a performance analysis of FIR digital filter design based on Residue Number System (RNS) on one hand and traditional method on the other hand. They compared the performance in terms of speed and area requirement for various filter specifications. According to the authors, RNS based FIR filters operate more than three times faster and consume only about 60% of the area than traditional filter when number of filter taps is more than 32. The area for RNS filter is increasing at a lesser rate than that for traditional, resulting in low-power consumption. An FIR filter is described by equation (1),

$$Y(n) = \sum_{k=0}^N H(k)X(n - k) \quad (1)$$

where $x(n)$ is the input to the filter, $H(k)$ represents the filter coefficients, N is the order of the filter and $Y(n)$ stands for the output from the filter. For very large N , filters implemented in the traditional binary weighted number system suffer from the disadvantages of the carry propagation delay in binary adders and multipliers. In RNS a large integer is broken into smaller residues which are independent of each other, and each digit is processed in parallel channels without any carry propagation from one to another. This leads to significant speed up of multiply and accumulate (MAC) operations which in turn results in high data rate for RNS based FIR filters. This RNS has also been applied in the design of IIR filters and the same effect of increased speed is realized.

Pranab et al (2008a) used what they described as window method to design and implement digital filters for audio signal processing. The window method is actually a Fourier series method that applies window to reduce Gibbs Phenomenon. The phenomenon is an error arising from truncation of the infinite Fourier series of the desired filter response to make it finite. In other words they designed and implemented finite impulse response digital filters. Rectangular window was used to design three different types of digital filters (low pass, high pass and band pass) while Matlab programming was used to implement them. In the low pass, the cut-off frequency is 3.4 KHz while the cut-off frequency for the high pass is 600Hz. In the band pass the pass band frequencies are 600Hz and 3.4 KHz.

Pranab et al (2008b) implemented an finite impulse response filter using DSP blocks. They considered three digital filters (low pass, high pass and band pass) in which the high and low frequency components of real-time voice signals were removed. The filters were designed using Kaiser Window in one instance and triangular window in another instance. The performance evaluation of the filters resulting from the two designs was carried out in terms of the filtered outputs and the frequency response curves. Experimental results indicate that a maximum number of ripples appear in stop band for Kaiser Window, while minimum number of ripples appear in triangular window for different filters. Therefore, real-time FIR digital filters using triangular window gives optimal result to process the audio signals. It should be noted that both Kaiser Window and triangular window are parts of the process in the Fourier series method of design of digital filters. Arshan and Roberts (2001) presented "A CMOS digitally programmable current steering semi-digital FIR reconstruction filter". They targeted a low-power, area efficient, single bit finite impulse response (FIR) reconstruction filter for delta-sigma applications based on current steering approach. The filter coefficients are made programmable with discrete values from -8 to 8, thus allowing for various filter responses on the same chip. The filter is implemented in a 0.25 μ m standard CMOS process and incorporates 2.09mm² of active area and a 2.5V supply. Three different filter functions are implemented to consist of a voice band low pass filter, an audio band low pass filter and a band pass filter. The audio band example achieves a dynamic range of 78dB for a signal bandwidth of 20 kHz and 65 dB over a 100 kHz bandwidth. Mahesh et al (2006) used Kaiser Window instead of rectangular window to design low pass, high pass and notch filters to be used in cascade for ECG processing. The Kaiser window corrects the problem of Gibb's phenomenon that is associated with rectangular window and therefore offers better removal of noise in ECG and less modification of QRS complex. The sampling frequency is 1000Hz and the order of each filter is 100. From the analysis of the proposed sine window function, it can be seen that there is an average reduction in the side lobe peak compared to other windows because sine window gives less lobe width. Also, filter design using the new window achieves less ripple ration compared to the filters obtained using other windows.

III. PROPOSED SINE WINDOW FLITER

In this paper, the main idea is to remove the 50Hz power line interference from the desired signal. The desired is Electrocardiographic (ECG) signal of a patient in a hospital. Finite Impulse Response (FIR) was designed using a new sine window function in other to remove 50Hz power line interference. FIR filter designed using the new window function has the same main lobe as the other windows. There is an average reduction in side lobe peak of the proposed window compared to that of the Hamming, Kaiser, Triangular, Blackmann and Han window. It is obvious that the new window performs better than other window. The length of the proposed window is a little more than the length of the filter obtained by Parks-McClella algorithm while the new window has simple form as shown in figure 1.

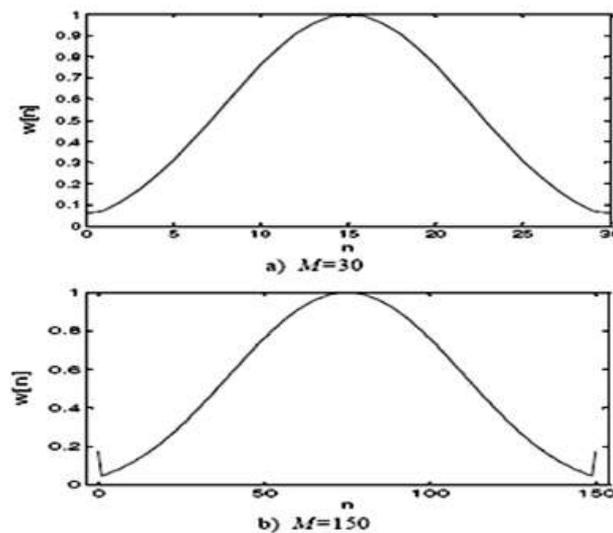


Figure 1: The new window

IV. METHODOLOGY

There are several methods that can be used to design and implement digital filters. The choice of method depends on the impulse response nature of the filter and frequency response nature of the desired filter. Digital filters are described by two types of transfer functions: transfer functions of finite impulse response (FIR) filters and those of infinite impulse response (IIR) filters.

V. TRANSFER FUNCTION OF FIR FILTER

The transfer function of FIR filters is stated in (2) (Sarkar, 2003).

$$H(z)=h(0)+(h1)z^{-1}+(h2)z^{-2}+(h3)z^{-3}+(h4)z^{-4}+(h5)z^{-5}+.....+(hn)z^{-N} \tag{2}$$
 Where n varies from 0 to N and N is the order of the filter while h(0) to h(n) are the filter coefficients. The transfer function of IIR filters is stated in (3) (Mbachu & Nwosu, 2014).

$$H(z)=\frac{b_0 + b_1Z^{-1} + b_2Z^{-2} + b_3Z^{-3} ++ b_NZ^{-N}}{1 + a_1Z^{-1} + a_2Z^{-2} + a_3Z^{-3} ++ a_NZ^{-N}} \tag{3}$$

where $b_0, b_1, b_2, \dots, b_N$ and a_1, a_2, \dots, a_N are filter coefficients. The order of the filter is N either in the numerator or denominator polynomials, which ever one is higher.

The proposed filter is a bandstop type of filter with narrow stopband. The analogue amplitude response of a bandstop filter of symmetrical form with frequencies in hertz is shown in figure 2.

- F₁ = low cut off frequency
- F₂ = high cut off frequency
- F₃ = low rejection frequency
- F₄ = high rejection frequency
- F₀ = centre frequency

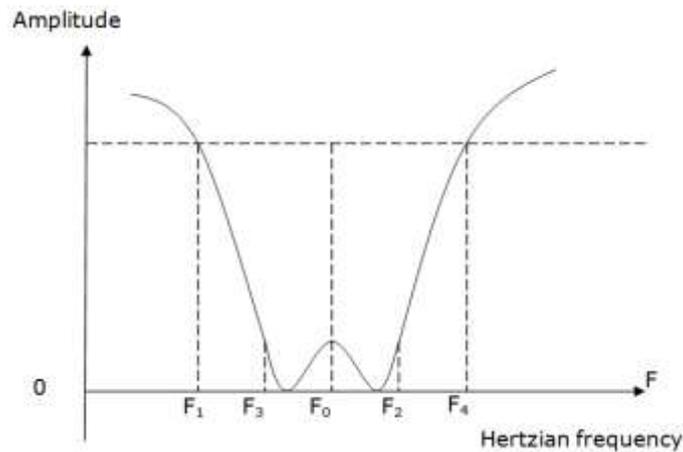


Figure 2: Amplitude Response of Bandstop Filter

The analogue magnitude response of symmetrical banstop filter with frequency axis denoted by the normalised frequency Ω or $\Omega T/2$ is illustrated in figure 2, where

$$\Omega_0 = \sqrt{\Omega_3 \Omega_4} \tag{4}$$

α_{max} is maximum passband attenuation and occurs at Ω_1 and Ω_2 while α_{min} is the minimum stopband attenuation and occurs at Ω_3 and Ω_4

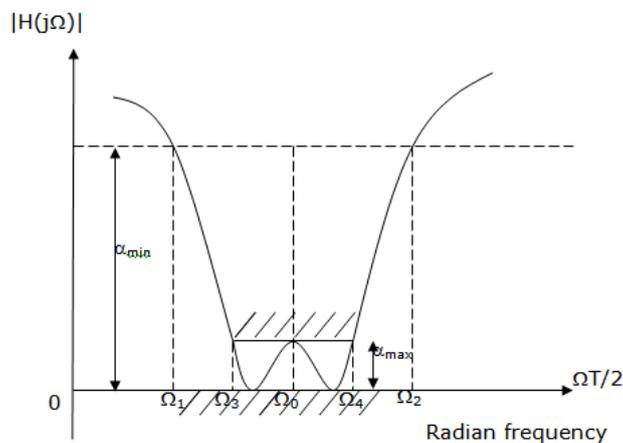


Figure 3: Magnitude Response of Bandstop Filter with Normalized Frequency Ω or $\Omega T/2$

VI. IMPLEMENTATION OF SINE WINDOW NOTCH FILTER

Ten steps are involved in this design and implementation as follows; mathematical modeling of the sine window function, calculation of the order of the filter based on selected sampling frequency and attenuation values, obtaining responses of the filter based on the calculated order and cut-off frequencies, determining the optimum order of the filter, determining the optimum sampling Frequency of the Filter, determining the optimum passband frequencies, structural realization of the filter, generation of results, calculation of signal to noise ratio of the filter, and finally comparative analysis of the proposed window and other windows in use for the processing of ECG signal.

VII. ANALYSIS AND COMPARATIVE PERFORMANCE OF THE PROPOSED SINE WINDOW WITH OTHER WINDOWS

In this section, the proposed window is validated by comparing its performance with other existing windows in filtering ECG signals. The windows include hanning, hamming, Blackman, Kaiser, rectangular and triangular windows. The filtered ECG signals when the filter is designed with the different windows are presented in figure 4 to figure 10 below. It can be clearly seen that these windows reduced substantially the 50Hz powerline noise in the ECG signal. A computation of the signal to noise ratio will provide the extent of the

performance of each window in removing the 50Hz powerline noise. The matlab command for creating the object of the filter and effecting the filtration is presented as follows (A):

```

1 b1=fir1(L,wn,'stop',hanning(L+1));
2 b2=fir1(L,wn,'stop');
3 b3=fir1(L,wn,'stop',blackman(L+1));
4 b4=fir1(L,wn,'stop',kaiser(L+1,3.3953));
5 b5=fir1(L,wn,'stop',rectwin(L+1))
6 b6=fir1(L,wn,'stop',triang(L+1));
7 y1=filter(b1,1,d,si);
8 y2=filter(b2,1,d,si);
9 y3=filter(b3,1,d,si);
10 y4=filter(b4,1,d,si);
11 y5=filter(b5,1,d,si);
12 y6=filter(b6,1,d,si);

```

Where L is order of the filter, wn is a vector of the two passband frequencies of the filter, d is the corrupt ECG signal and si ensures that the filter taps are zeroed initially, while stop indicates that the filter is a bandstop filter. Numbers 1 to 6 creates the object of each filter designed with the corresponding windows while numbers 7 to 12 filters the powerline out from the corrupt signal for each filter designed with the corresponding window as indicated by the corresponding objects of the filters. The sine window filter removed the targeted 50Hz noise from the ECG signal as shown in 4. From figures 5-10, it can be deduced that each of the six windows were used to validate that the work substantially filtered out interference signal. The sine window technique outperformed all of them because it yielded better signal to noise ratio (see table 1) and power spectral density in comparing to other windows.

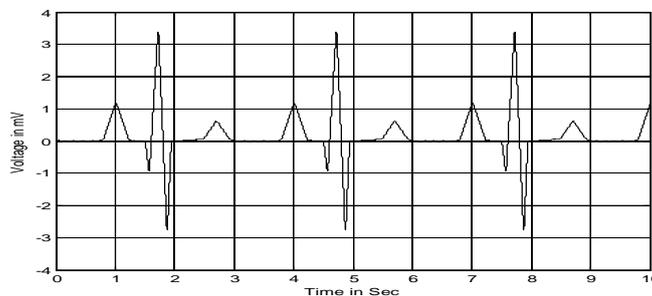


Figure 4: ECG Signal Filtered with the Proposed Window

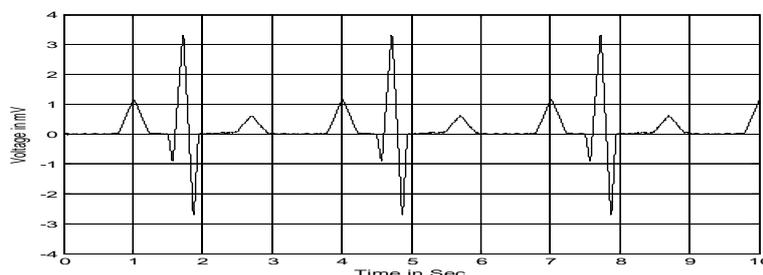


Figure 5: ECG Signal Filtered with Han Window

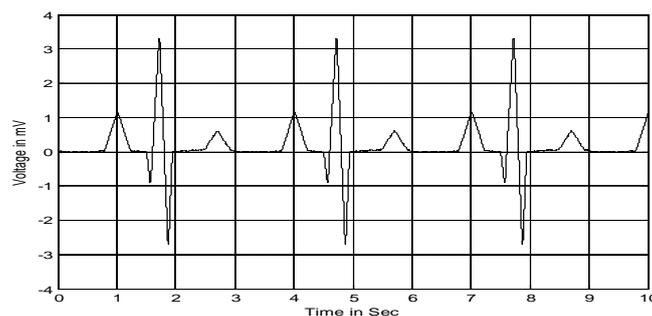


Figure 6: ECG Signal Filtered with Hamming Window

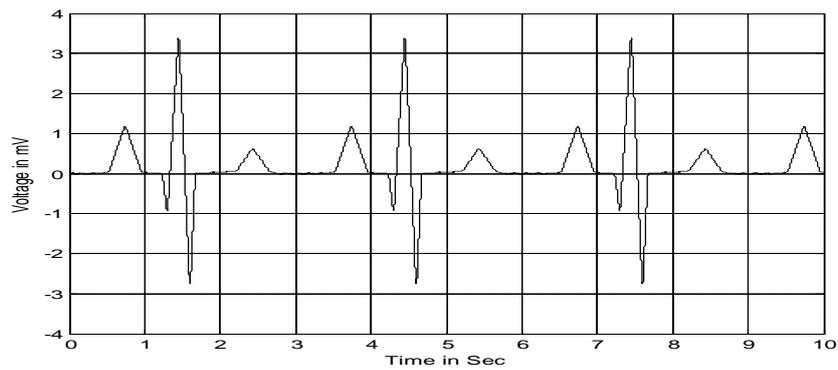


Figure 7: ECG Signal Filtered with Blackman Window

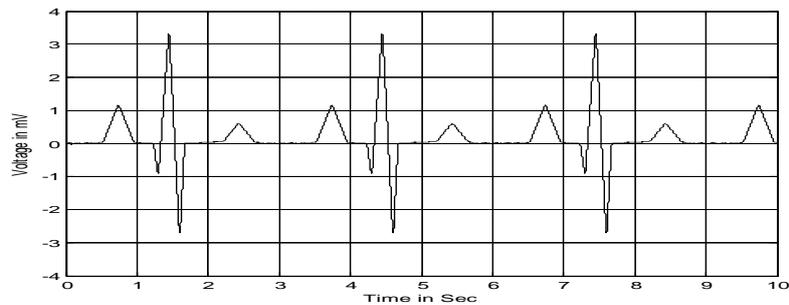


Figure 8: ECG Signal Filtered with Kaiser Window

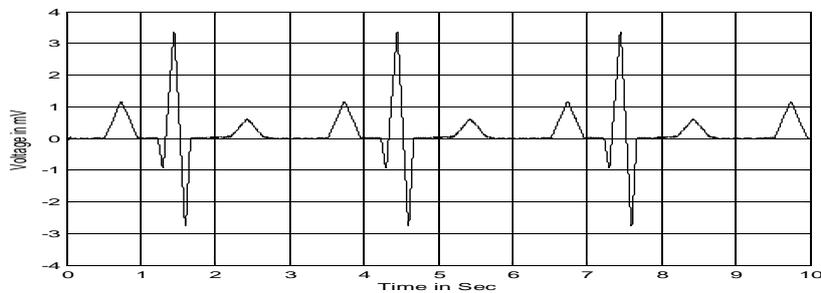


Figure 9: ECG Signal Filtered with rectangular Window

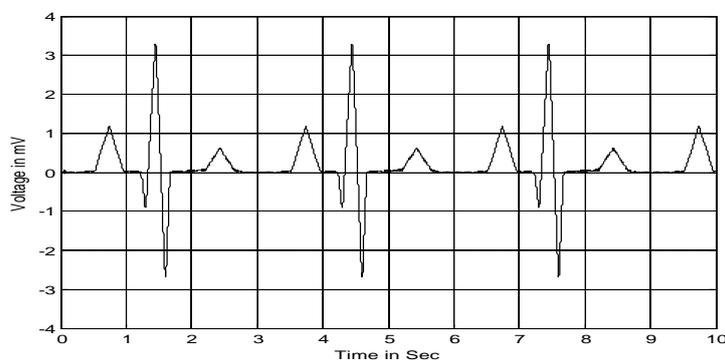


Figure 10: ECG Signal Filtered with Triangular Window

4.3.1 Signal to Noise Ratio (SNR)

In order to estimate performance of each window the signal to noise ratio of the filtered ECG signal for each window is calculated. The equation for calculating signal to noise ratio (SNR) is as in (4) (Mbachu, 2015, Mbachu & Nwosu, 2014)

$$SNR_o = 10 \log \frac{\sum S_F^2}{\sum N_o^2} \quad (5)$$

where N_o , the output noise power, is the noise power present in the filtered signal power, while S_F is the power of the filtered signal. Output noise power N_o is given by

$$N_o = S - S_F \quad (6)$$

where S is the power of the corrupt signal. Using (5) in (6) gives (7)

$$SNR_o = 10 \text{Log} \frac{\sum S_F^2}{\sum (S - S_F)^2} \quad (7)$$

The signal to noise ratios are presented in table 1 below

Table 1: Signal to Noise Ratios (SNR) of the Filtered ECG

| Windows | Proposed window | Han window | Hamming window | Blackman window | Kaiser window | Rectangular window | Triangular window |
|---|-----------------|------------|----------------|-----------------|---------------|--------------------|-------------------|
| Power of the corrupt ECG at 50Hz in dB | 11.26 | 11.26 | 11.26 | 11.26 | 11.26 | 11.26 | 11.26 |
| Power of the filtered ECG at 50Hz in dB | -27.25 | -6.83 | -7.95 | -1.46 | -22.14 | -3.88 | -5.18 |
| Signal to noise ratio of the filtered ECG at 50Hz in dB | 33.98 | 18.02 | 19.16 | 12.48 | 33.40 | 14.49 | 16.34 |

Considering signal to noise ratios of the contaminated signal and the filtered signals as computed from the power spectral densities of the contaminated signal and filtered signals at normalised frequency of 0.1 which translates to 50Hz in this work, and tabulated as in table 1 shows that the sine window has an improvement on the signal quality over other windows with a SNR of 33.98dB, followed by Kaiser window with SNR of 33.40dB while the rectangular has the least performance with SNR of 12.48dB.

VIII. CONCLUSION

In this paper, a robust sine window function has been presented as an efficient window applicable to FIR filtering. A complex signal such as ECG signal was filtered of 50Hz power line interference artifact. In the filtration analysis, the sine window filter removed the targeted 50Hz noise from the ECG signal. This can be seen by comparing the original ECG signal with the ECG signal filtered with other windows. From the signal to noise ratio (SNR) of the contaminated signal and filtered signals, there is an improvement on the signal quality over other windows with a 33.98dB. The Kaiser window has SNR of 33.40dB while the rectangular window has the least performance with SNR of 12.48dB.

IX. RECOMMEDATION FOR FUTHER STUDIES

The ultimate practical implementation of this work is programming the functions into digital signal processor using the appropriate language and interfacing it to the peripheral hardware like spectrum analyzer and ECG transducer. Simulation results only give insights on, for instance, stability properties or qualitative filtration strength of the designed filters and do not provide practical issues like real time processing speed, device cost, and maximum filter order and quantization effects. Recommendation is therefore made of the implementation of this project using real time systems Digital Signal Processor (DSP).

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Ojo,O.S. Comparative Performance of the Sign Windowed Filtering Technique with Han, Hamming, Kaiser and Black Manwindowed Filtering Technique in Artifact Removing.” American Journal of Engineering Research (AJER), vol. 7, no. 3, 2018, pp.225-232.