A Hybrid Window Function to Design Finite Impulse Response Low Pass Filter with an Improved Frequency Response

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***Abstract*— In large amount applications appreciate Finite Impulse Response filters, Fast Fourier Transform, Signal Processing Technique and measurements, forced upon ~45dB or ultimately less side lobe magnitudes. However, the setback is prompt in window based FIR filter design lies in its side lobe magnitude that are higher than the specification of application. The expected hybrid window Technique has transcend attitude like minimum side lobe peak and higher side lobe peak attenuation compared to the several generally used windows with main lobe width(-3dB) of 0.11328. The proposed window has reliant side lobe attenuation of -72.7 dB and peak side lobe amplitude of -98 dB for the filter term of N=35. The studied results also unmask significant performance upgrading of the coming window compared to Rectangular, Hanning, Hamming, Blackman and Kaiser Windows. Finally, the approaching window is applied to design Low pass FIR filter prove the simplicity of the window**

Keywords*— *Digital signal processing, Digital filter, Finite Impulse Response Lowpass filter, Window Functions- Rectangular, Hanning, Hamming, Blackman and Kaiser, Hybrid Window.

I. INTRODUCTION

Digital filter plays very important practice in today's hand one is dealt of communication and computation. Digital filters are effective of performing extensive review which are intensively very difficult, to move up in the world by the whole of an analog implementation. It can be implemented in hardware and software and it gave a pink slip process the real-time all hail and off-line (recorded) signals. Beside the inherent advantages one as steep accuracy, reliability, small physical period of time and reduced resentment to foundation tolerances or drift, digital filters are allows achieving certain characteristics which are not possible with analog implementations one as interchangeable linear phase and stability. In addition, the characteristics of digital filters boot be re assigned or adapted by seldom changing the coefficients of the filter. Many digital system consider all hail filtering to go back on one word unwanted noise, spectral shaping, signal detection and analysis.

Two types of filters laid at one feet these trade are Finite Impulse Response filter (FIR) and Infinite Impulse Response filter (IIR). FIR filters are selected in approximately of signal processing academic work than IIR filter. Since FIR filter have many suitable features one as settled stability, linear phase detailed at generally told the frequencies and implementation of filter as non-recursive structures. To design a digital FIR filter that useful all the required requirement is a challenging one. Design of FIR filter is to obtain the coefficients one that the system meets the specific characteristics. The different methods secondhand for FIR filter design are Windowing approach, Frequency Sampling approach and Optimal Filter Design Method. In the debate, it is desired for a window function to have characteristics of smaller ripple ratio and narrow main lobe width. However, these two requirements are contradictory. For the admit of comparison with length, Hamming window offers the smallest peak of side lobe as well as main lobe width compared to Hanning window. The Blackman window has wider main lobe width yet smaller side lobe peak compared to Hamming window. The Kaiser window is tunable function and there is a trade-off surrounded by side lobe peak and main lobe widths and can be customized. The Kaiser window has the disadvantages of higher computational complication for absorbed window coefficients. There has been great high on the hog into the design of beautiful window to rival the desired requirement for different application. In this freebie Hybrid window work is exposed which is a agglomeration of Hamming and Blackman windows.

The rest of the paper is organized as follows: The Review of FIR filter design is discussed in Section 2. In Section 3, the proposed Hybrid window is presented in details. In Section 4, Proposed window and other windows were simulated using MATLAB and also simulated the Lowpass FIR filter design. Finally the performance analysis was done on Side lobe peak attenuation and peak amplitude of side lobe for various window Technique in Section 5 before concluding paper in Section 6.

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II. REVIEW OF FIR FILTER DESIGN

A brief review of some significant researches related with design of FIR filter are as follows:

Sonika Gupta and Aman Panghal [1], proposed a FIR filter design using window methods such as Hanning window, Hamming window, Blackman window, Bartlett window and Kaiser Window. Using Bartlett window, proposed approach reduces the overshoot occurs in the pass band and stop band but spreads the transition region considerably. The proposed filter design by Hanning, Hamming and Blackman windows provided the smooth truncation of the ideal impulse response that gives better frequency response than Bartlett window but these window methods uses more complicated cosine terms in window function. Results shown that Kaiser window based FIR filter design gives best frequency response than other window by choosing proper value for the parameter β, that allows the adjustment of compromise between the overshoot reduction and transition region width spreading. Drawback of proposed method is that the minimum stop band attenuation is fixed for each function.

S.K. Shome, S.R.K.Vadali et.al [2], proposed the Simple Moving Average (SMA) and Exponentially Weighted Moving Average (EWMA) based filter techniques and are applied on corrupted signal having different signal to noise ratio. Performance evaluation of these techniques had been carried out using MATLAB and DSP TMS320C6713. The authors founds that signal average techniques have less complexity than conventional digital filter design. The performance of EWMA filter design is better than SMA when the signal was corrupted with more noise. For Higher throughput the proposed method require further optimization.

Tao Zhang[3], analyze the performance of FIR filter with order 4, 8 and 12 using window method. In the proposed approach, the frequency response of the filter much closer to the desired frequency response and have better filter effect & stronger ability to filtering the interference signal, when the order of the filter is high. High order filter design requires more calculation and memory space. Atul Bhargava and Ravindra Pratap Narwaria [4] developed least square method neural network algorithm to design linear phase FIR filter based on approximation of a magnitude response. The proposed approach minimized the sum of squared error between the amplitude response of the desired FIR filter and that of designed by the single layer feed-forward neural network algorithm by considering the high order and gives better result i.e. minimum pass band ripples and stop band ripples. G. Jovanovic-Dolecek et.al [5] proposed a one simple method for the design of multiplier less finite Impulse Response filter by the repeated use of the same filter. The proposed filter uses a cascade of second order Recursive Running Sum (RRS) filter known as a cosine filters and its corresponding expanded version. Therefore, the proposed approach doesn't required multiplier to implement this filter. The proposed method was intended for the narrow band filter design. Narendra Singh Pal et.al [6] described the implementation of highly efficient multiplier less serial and parallel Distributed Arithmetic (DA) algorithm for FIR filters. The results of the proposed filter are analyzed for 3 – tap and 16 – tap FIR filter using partitioned input based LUT implemented on Xilinx. The speed performance and area efficiency of parallel DA superior to Serial DA algorithm. The proposed parallel DA algorithm achieved less area, power consumption and high speed for smaller tap filter. For larger tap filter, the proposed parallel DA algorithm small area and high speed at a cost of power consumption. Mohamed Al Mahdi Eshtawie and Masuri Bin Othman [7] proposed an algorithm for modifying the values and number of non – zero coefficients used to represented the FIR filter response. The proposed algorithm represent FIR filter response using new set of filter coefficients (Half of the coefficients are original) and results are compared with filter response designed by original coefficients that had been shown error probability was same in both cases. In the proposed algorithm, decrease in number of non-zero coefficient decreases the number of adder into half. This have a very much influence on increased system speed and reduced the hardware complexity but increased the power consumption as the order of the filter increased.

Hui Zhao and Juebang Yu [8] developed the neural network-based digital filter Design using continuous hopfield neural network (CHNN) and neural network based filter equation. The proposed algorithm provided the relation between MSE criterion and the Lyapunov function. Authors compared the proposed method to the other technique with few linear phase FIR design examples and finally the proposed algorithm prove the Neural Network Optimization (NNO) technique based filter provides a better result. The proposed technique have the advantages of high computational efficiency and suitable for hardware implementation for the real-time processing purpose. Wu-Sheng Lu[9] designed an equiripple FIR filter, based on the sequential quadratic programming (SQP) algorithms. The proposed algorithm proved that sequential quadratic programming based FIR filter had low group delay. Filter designed by using sequential quadratic programming (SQP), the degree of flatness varies with the length of filter. They used MATLAB toolkit functions for implementation of proposed algorithm Because of its simplicity, it can be written in computer code very easily.

Amanpreet Kaur [10] developed a Modified Particle Swarm Optimization (MPSO) based FIR filters with low delay. The proposed algorithm proved that gradient based optimization techniques are not effective for designing filter. In order to reduce error the different optimization technique for FIR filter design are presented where in the remaining frequency samples are chosen to satisfy an optimization criterion, the main advantage of the Modified Particle Swarm Optimization (MPSO) algorithms is sharper transition band responses of the filter .This method could controlled the overshoot phenomenon near the passband and stop-band edges of the designed filter. Filter designed by using Modified Particle Swarm Optimization (MPSO), the degree of flatness varies with the length of filter. Lo-Chyuan Su, Yue-Dar Jou, Fu-Kun Chen[11] described a neural network implementation based technique for designing of digital filters. To demonstrate the feasibility of the Neural network design approach, a model is chosen based on the Hopfield neural network. The proposed method proved that the computational requirement and the required number of neurons was significantly smaller than the Bhattacharya and least-squares (LS) method and hardware cost also greatly reduced.

S. M. Shamsul Alam [12] designed a digital finite impulse response (FIR) filter using different method and generate different curves and finally compared with ideal response curve. They designed a FIR filter using Remez exchange algorithm with Blackman window method, Frequency sampling method and Optimal method. It was shown that the response curve of FIR filter depend on the width of transition band. The proposed technique had the advantages of high computational efficiency . Sheenu Thapar [13] designed Low pass FIR filter using artificial neural network with genetic algorithms and founds that the artificial neural network (ANN) optimized with genetic algorithms(GA) is met the performance goal in just seven iterations. They compared the proposed approach with Kaiser window method and shows that, not only the computational complexity of the proposed neural architecture, but the hardware cost also can be greatly reduced. Yong Ching Lim[14] presented a novel fast convergent weighted least squares algorithm for quasi-equiripple FIR and IIR filter designs. For deriving the weighted squares frequency response, a novel iterative algorithm is used by the proposed method. The proposed algorithm proved that the designed filter has better response and converges at a speed several times faster than the commonly used Lawson's algorithm.V. Ralph Algazi [15] designed a Finite duration filters using least-square method. Designed algorithm was developed to control the overshoot phenomenon near the pass-band and stop-band edge of the designed filter. The proposed algorithm converged at a speed several times faster than the commonly used algorithm. Because of its simplicity, it could be written in computer code very easily and could be quite attractive in image processing.

III. PROPOSED HYBRID WINDOW

FIR filter design by window method is simple and easy. Since well define equations are available for calculating the window coefficients. The different types of window functions are Rectangular window, Triangular window, Hanning window, Hamming window, Blackman window and Kaiser Window which are differed by their desirable features.

The desirable features of window functions are

- The main lobe of the frequency response should contains most of the energy and should be narrow for sharp transition region
- The peak side lobe magnitude should be small for minimum pass band and stop band ripple
- The magnitude of side lobe of the frequency response should decrease as ω tends to π.

All existing window functions for Finite impulse response (FIR) filters design are developed to meet desired specification such as minimum pass band and stop band ripple and sharp transition region.

A. Procedure for design of Linear Phase FIR filter

Step 1: Choose the desired frequency response Hd(eiω) of the filter.

Step 2: Obtain filter coefficients by taking inverse Fourier transform of desired frequency response

$$h_d(n) = \frac{1}{2\pi}\int_{-\pi}^{\pi} H_d(e^{i\omega})e^{i\omega n}\,d\omega \tag{1}$$

Step 3: Convert infinite length filter coefficients“ (1)” into finite length filter coefficients by multiplying the infinite impulse response with a finite length window function w(n)

$$h(n) = h_d(n) \text{ x } w(n) \tag{2}$$

Step 4: Obtain transfer function H(z) by taking z-transform of h(n).

Step 5 : Realize the transfer function.

The different types of window functions are

B. Existing Windows

1) Rectangular window

$$w(n) = 1, \quad 0 \le n \le N\text{-}1$$

$$0, \quad \text{otherwise}$$

2) Hanning window

$$w(n) = 0.5\text{-}0.5\cos(2\pi n/N\text{-}1), \quad 0 \le n \le N\text{-}1$$

$$0, \quad \text{otherwise}$$

3) Hamming window

$$w(n) = 0.54\text{-}0.46\cos(2\pi n/N\text{-}1), \quad 0 \le n \le N\text{-}1$$

$$0, \quad \text{otherwise}$$

4) Blackman window

$$w(n) = 0.42\text{-}0.5\cos(2\pi n/N\text{-}1) + 0.08\cos(4\pi n/N\text{-}1), \quad 0 \le n \le N\text{-}1$$

$$0, \quad \text{otherwise}$$

5) Kaiser window

$$w(n) = \frac{I_0\left(\alpha\sqrt{\left(\frac{N-1}{2}\right)^2 - \left(n-\frac{N-1}{2}\right)^2}\right)}{I_0(\alpha)}, \quad 0 \le n \le N\text{-}1$$

$$0, \quad \text{otherwise}$$

C. Proposed Hybrid Window

The proposed hybrid window is the combination of Hamming and Blackman window.

The proposed window function is

$$w(n) = [0.54-0.46\cos(2\pi n/N-1)]*[0.42-0.5\cos(2\pi n/N-1)+0.08\cos(4\pi n/N-1)], \quad 0 \le n \le N-1$$

$$0, \quad \text{otherwise}$$

$$w(n)=[0.3418-0.4816\cos(2\pi n/N-1)+0.1582\cos(4\pi n/N-1)-0.0184\cos(6\pi n/N-1)], \quad 0 \le n \le N-1$$

$$0, \quad \text{otherwise}$$

Direct truncation of infinite number of samples of Impulse Response $h_d(n)$ into N number of samples h(n) by Rectangular window leads to the **Gibbs phenomenon** effect i.e. overshoot and undershoot in pass band and leakage in stop band due to the non-uniform convergence of the Fourier series at a discontinuity. Thus the frequency response obtained by Rectangular window contains ripples in the frequency domain. In order to reduce the ripples, instead of multiplying $h_d(n)$ with a rectangular window w(n), $h_d(n)$ is multiplied with a window function that contains Cosine terms which is taper and decays toward zero gradually, instead of abruptly as it occurs in a rectangular window. The multiplication of impulse response $h_d(n)$ and w(n) in time domain is equivalent to convolution of $H_d(\omega)$ and $W(\omega)$ in the frequency domain which have the effect of smoothing the frequency response $H_d(\omega)$. The several effects on the frequency response of filter by windowing the filter coefficients are as follows:

- A major effect is that discontinuities at the edge of window function leads to ripples in the frequency response, H(w) of the filter.
- The width of the transition bands depends upon the width of the main lobe of the frequency response of the window function, w(n)
- As the length of the window function N increases, the main lobe width of W(n) is reduced which reduces the width of the transition band, but this increases the ripple in the frequency response.

IV. SIMULATION RESULTS

The existing windows and proposed window function are simulated in MATLAB with Order N=35. *Fig. 1,2,3,4,5 and 6* shows the frequency response of Rectangular window, Hanning window, Hamming window, Blackman window, Kaiser window and Hybrid window in Time domain and frequency domain. The proposed Hybrid window achieved maximum relative sidelobe attenuation of -72.7 dB compared to existing window. The frequency response of Low Pass FIR filter using existing and proposed window techniques are shown in *Fig.7,8,9,10,11,12.*

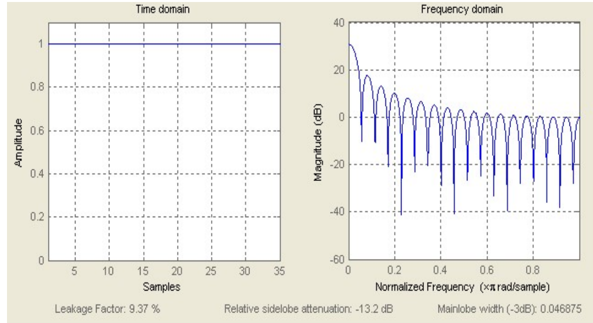

Fig. 1. Rectangular Window Frequency Response

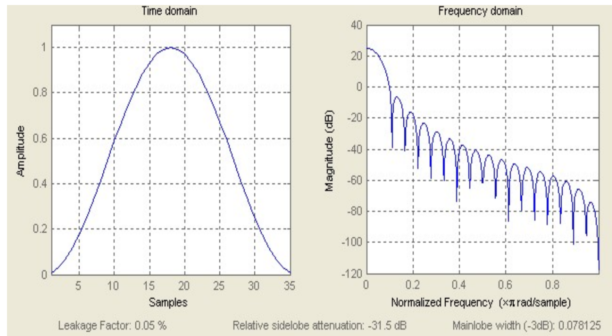

Fig. 2. Hanning Window Frequency Response

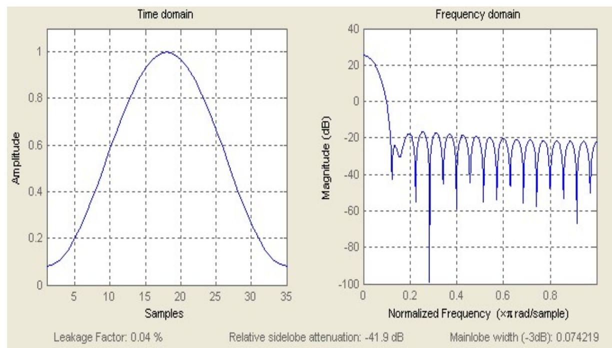

Fig. 3. Hamming Window Frequency Response

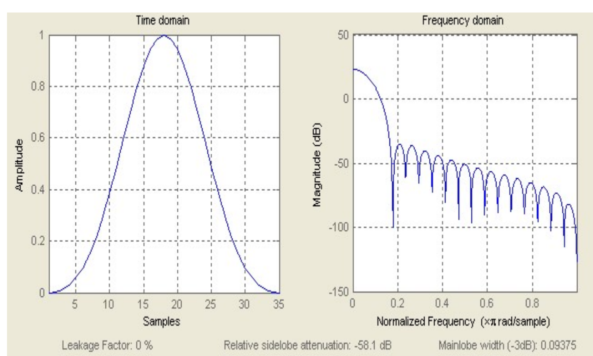

Fig. 4. Blackman Window Frequency Response

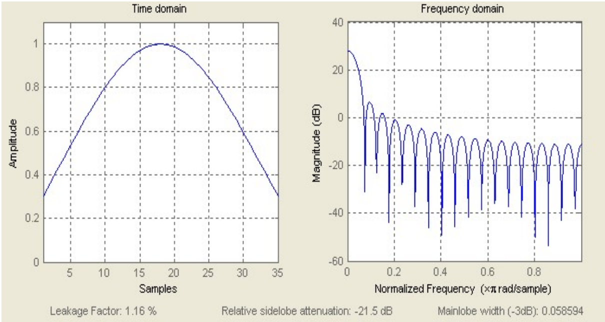

Fig. 5. Kaiser Window Frequency Response

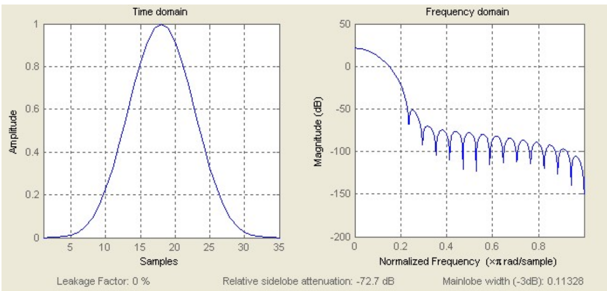

Fig. 6. Hybrid Window Frequency Response

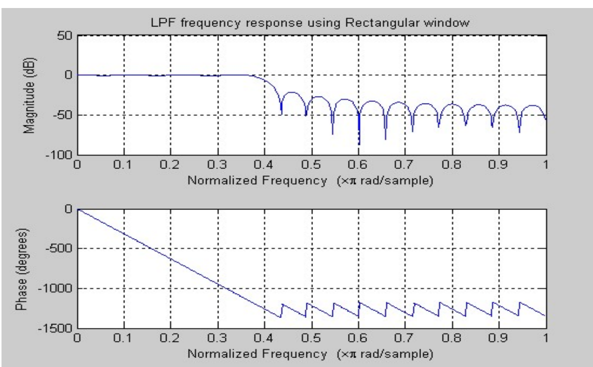

Fig. 7. FIR- LPF Frequency Response using Rectangular Window

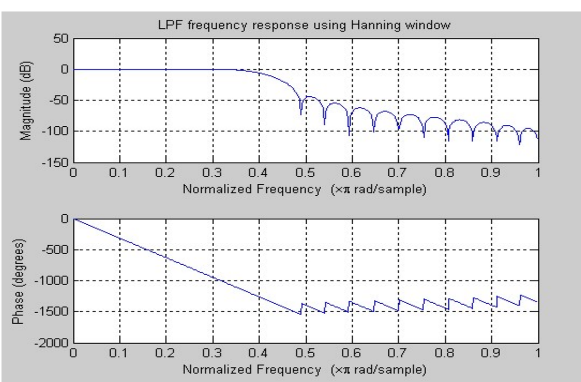

Fig. 8. FIR- LPF Frequency Response using Hanning Window

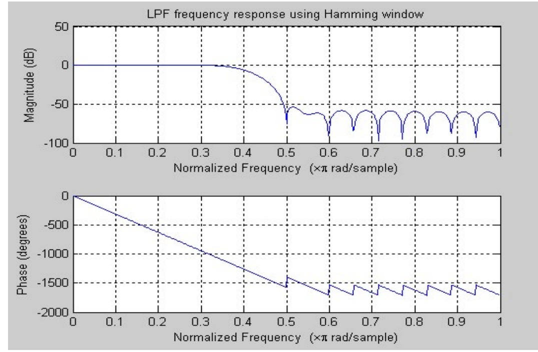

Fig. 9. FIR- LPF Frequency Response using Hamming Window

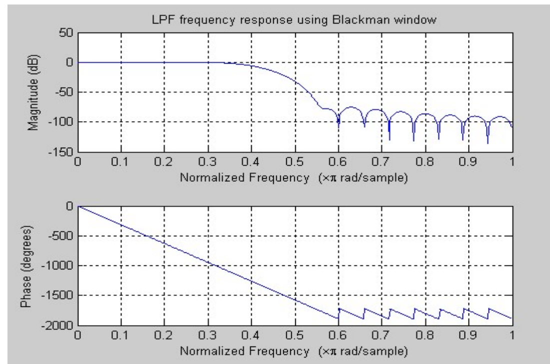

Fig. 10. FIR- LPF Frequency Response using Blackman Window

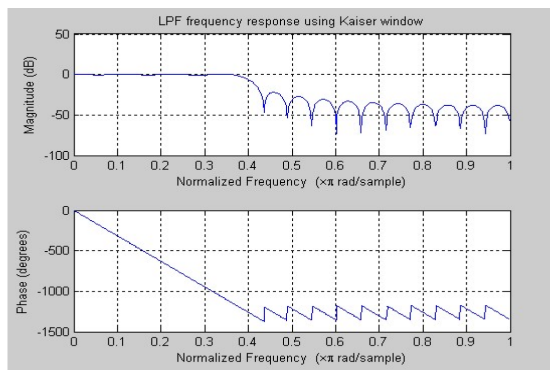

Fig. 11. FIR- LPF Frequency Response using Kaiser Window

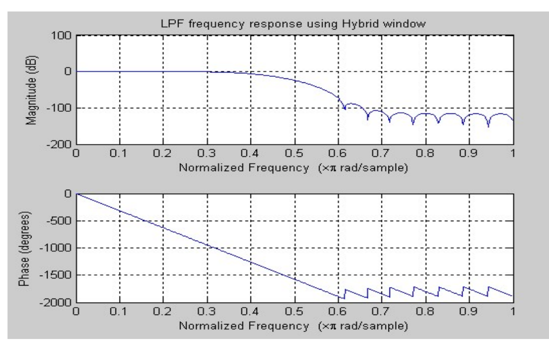

Fig. 12. FIR- LPF Frequency Response using Hybrid Window

V. PERFORMANCE ANALYSIS

Designing of FIR filter by rectangular window has a minimum transition width but the attenuation of side lobes in stop band is very less compare to other windows. Filter design by Hanning and Hamming windows have the same transition width which is higher than rectangular window but the side lobe attenuation is more than rectangular window. Designing of filter by Blackman window having the highest transition region width and side lobe attenuation in stopband compare to other windows. In Kaiser window attenuation of side lobe in stopband can be achieved by increasing the value of parameter α as a result increases in transition width. So the existing window techniques, achieved either minimum transition width to obtain sharp transition or maximum side lobe attenuation. The proposed hybrid window has at least -31.2 dB improvement on side lobe reduction compared to Hamming window and -14.6 dB improvement on side lobe reduction compared to Blackman window while offering slightly increased main lode width ~0.01953to ~0.039. The performance of proposed window technique is compared with existing windows is shown in *Table.I* .

TABLE I. PERFORMANCE COMPARISON

Window Type	Relative sidelobe attenuation(dB)	Peak amplitude of sidelobe(dB)
Rectangular	-13.2	-25
Hanning	-31.5	-48
Hamming	-41.5	-52
Blackman	-58.1	-75
Kaiser	-21.5	-26
Hybrid	-72.7	-98

VI. CONCLUSION

The proposed window is symmetric function and shows better equiripple property. Performance analysis of the proposed window compared to that of Rectangular, Hanning, Hamming, Blackman and Kaiser Window shows that the proposed window offers less side lobe peak and more side lobe reduction. The Designed Lowpass FIR filter using proposed window achieves less ripple ration compared to above mentioned window filters. Finally for the same specification, the filter length of N=35 the proposed window gives more side lobe reduction of -72.7 dB with slightly increased main lobe width (-3dB) of 0.039 in comparison with Hamming window and 0.01953 with Blackman window. The Designed Lowpass FIR Filter can be used for removing the noise contained in speech signal based on the requirement of filter characteristic parameters such as filter order and cut-off frequency. In future the proposed window can be make it as adjustable window in order to vary the main lobe width and amplitude of the side lobe with a fixed length.

REFERENCES

[1] Sonika Gupta and Aman Panghal, "Performance Analysis of FIR filter Design by using Rectangular, Hanning and Hamming window Methods", International Journal of Advanced Research in Computer Science and Software Engineering, ISSN: 2277-128X, Volume 2, Issue 6, June 2012.

[2] S. K. Shome, S.R.K. Vadali, U.Datta, S.Sen and A.Mukherjee, "Performance Evaluation of Different Averaging based filter Design using Digital Signal Processor and its synthesis on FPGA", International Journal of signal processing, Image processing and Pattern Recognition,Volume 5, Issue 3, september 2012.

[3] Tao Zhang, "Research on Design FIR Digital filter using MATLAB and window function method", Journal of Theoretical and Applied Information Technology, ISSN: 1992- 8645, No.1, Feb 2013.

[4] Atul Bhargava, Ravindra pratap Narwaria, " Design of FIR Digital filters using Least Square Method Based on Neural Network",IJSETR,ISSN: 2278-7798, Volume 1, Issue 5, Nov 2012.

[5] G.Jovanovic – Dolecek, M.M-Alvarez and M.Martinez," One Simple Method for the design of Multiplier less FIR filters", Journal of Applied Research and Technology, Volume 3, No.2, August 2005.

[6] Narendra Singh Pal, Harjit Pal Singh, R.K.Sarin and Sarabjeet Singh, " Implementation of High speed FIR filter using Serial and Parallel Distributed Arithmetic Algorithm",IJCA, ISSN:0975-8887, Volume 25, No.7, July 2011.

[7] Mohamed Al Mahdi Eshtawie and Masri Bin Othman, " An algorithm proposed for FIR filter coefficients Representation", World Academy of science Engineering and Technology, Feb 2007.

[8] Hui Zhao and Juebang Yu, " A Novel Neural network based approach for designing digital filters", IEEE International symposium on circuits and systems, 9-12, 1997.

[9] Xiaohua Wang and Xianzhi Meng, " A Novel Neural Network – Based Approach for designing FIR filters", Proceedings of the 6th world congress on intelligent control and automation, 21-23, June 2006.

[10] Amanpreet Kaur, " Design of FIR filter using Particle Swarm Optimization Algorithm for Audio Processing", IJCSN, Volume 1, Issue 4, August 2012.

[11] Lo-Chyuan Su, " Neural Least – Square Design of FIR Digital filters with Trigometric properties", WHAMPOA an Interdisciplinary Journal, Volume 54, 103-110, 2008.

[12] S.M. shamsul Alam, " Performance Analysis of FIR filter Design by using Optimal, Blackman window and Frequency sampling methods", IJECS, Volume 10, Issue 1, 13-18, 2007.

[13] Sheenu Thapar, " A low pass FIR filter Design using Genetric Algorithm Based Artificial Neural Network",IJCTEE, Volume 2, Issue 4, August 2012.

[14] Yong Ching Lim, " A Weighted Least Square Algorithm for Quasi Equiripple FIR and IIR digital filter Design", Transaction on signal processing, Volume 40, Issue 3, 551-558, 1992.

[15] V. Ralph Algazi, " Frequency weighted least square design of finite duration filters", IEEE Transaction on circuits and system, Volume 22, 2007.