International Journal of Electrical, Electronics and Mechanical Controls

A Scientific and Scholarly Resource for Publication of Research Manuscripts

http://www.ijeemc.com

Volume 1 Issue 1 November 2012

IMPLEMENTATION OF FIR FILTER USING EFFICIENT WINDOW FUNCTION AND ITS APPLICATION IN FILTERING A SPEECH SIGNAL

Saurabh Singh Rajput, Dr.S.S. Bhadauria

Department of Electronics, Madhav Institute of Technology and Science , Gwalior

E-mail: sourabhrajput@yahoo.com, saritamits_61@yahoo.co.in

Abstract—Digital filtering is one of the main fundamental aspect of Digital signal processing, So Digital filters are widely used in many digital signal processing applications. In this paper low-pass FIR filter is implemented using an efficient adjustable window function based on Blackman window function. In this adjustable window function for a fixed length the bandwidth of main lobe and side lobe amplitude can be varied by changing the value of α. In the frequency spectrum of window function for a efficiently selected value of α, this window provides higher side lobe attenuation comparison to Hamming window and the width of the main-lobe is slightly greater than Hamming window function. In some applications such as FFT, signal processing and measurement where higher side lobe attenuation is required compared to Hamming window this this type of filters can play very significant role. The Blackman window function can also be used for these types of applications but the Blackman window has a wider main lobe width. Designed low-pass FIR filter is used in the speech signal processing application. In this work, firstly speech signal was recorded in wave format after that designed FIR filter applied to this recorded speech signal. Signal comparison between the original speech and filtered speech signal shows that the high frequency component of speech signal has significantly removed by using this FIR low-pass FIR filter.

Keywords- Blackman window, FIR filters, FFT, IFFT, Low-pass filter, Speech Signal, Window function

International Journal of Electrical, Electronics and Mechanical Controls

A Scientific and Scholarly Resource for Publication of Research Manuscripts

http://www.ijeemc.com

Volume 1 Issue 1 November 2012

1. 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. 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. The basic filter types can be classified into four categories: low-pass, high-pass, band-pass, and band-stop. On the basis of impulse response, there are two fundamental types of digital filters: Infinite Impulse Response (IIR) filters, and Finite Impulse Response (FIR) filters [1].

Finite Impulse Response digital filter has strictly exact linear phase, relatively easy to design, highly stable, computationally intensive, less sensitive to finite word-length effects, arbitrary amplitude-frequency characteristic and real-time stable signal processing requirements etc. Thus, it is widely used in different digital signal processing applications [1,2].

FIR filter is described by differential equation. The output signal is a convolution of an input signal and the impulse response of the filter.

$$\mathrm{y(n)} = \sum_{k=0}^{N-1} (\mathrm{h(k)x(n-k)}) \qquad (1)$$

x(n) is the input signal

h(n) is the impulse response of fir filter

International Journal of Electrical, Electronics and Mechanical Controls
A Scientific and Scholarly Resource for Publication of Research Manuscripts

http://www.ijeemc.com

Volume 1 Issue 1 November 2012

The transfer function of a causal FIR filter is obtained by taking the z-transform of impulse response of FIR filter h(n).

$$\mathrm{H(z)} = \sum_{k=0}^{N-1} \left(\mathrm{h(k)} z^{-k}\right) \qquad (2)$$

Most Common type filters include a low-pass filter, which pass through the frequencies below their cutoff frequencies, and progressively attenuates frequencies above the cutoff frequency of a signal according to desired requirements. There are many straightforward techniques for designing FIR digital filters to meet arbitrary frequency and phase response specifications, such as window design method or frequency sampling techniques. The Window method is the most popular and effective method because this method is simple, convenient, fast and easy to understand. The main advantage of this design technique is that the impulse response coefficient can be obtained in closed form without the need for solving complex optimization problems [4].

Window functions can be divided into two categories; Fixed and Adjustable window functions. Mostly used Fixed window functions are; Rectangular window, Hanning window, Hamming window and Blackman window. On the other hand the Kaiser window is a kind of adjustable window function. In the literature survey, these different widows are used for the Digital FIR filter designing and spectral performance analysis [5-7]. FIR filter design using a new window function is given in [8]. The Performance Enhancement Study of FIR Filters Based on Adjustable Window Function is given in [9]. A novel window function yielding suppressed main lobe width and minimum side lobe peak is described in [10]. In the study of Fourier transform of these different Fixed window functions, for the fixed length the Rectangular window provides smallest main lobe width but the highest peak of side lobe among them, So Rectangular window is not widely used in digital signal processing applications. The Hanning and Hamming window provides good side lobe attenuation compare to rectangular window, so these windows are commonly used in different DSP applications. For higher side lobe attenuation Blackman window is used but the Blackman

International Journal of Electrical, Electronics and Mechanical Controls
A Scientific and Scholarly Resource for Publication of Research Manuscripts
http://www.ijeemc.com
Volume 1 Issue 1 November 2012

window has a wider main lobe width compare to Hanning and Hamming window. The Kaiser window is a kind of adjustable window function which provides independent control of the main lobe width and ripple ratio. But the Kaiser window has the disadvantage of higher computational complexity due to the use of Bessel functions in the calculation of the window coefficients [8].

In some applications such as FFT, signal processing and measurement, higher side lobe attenuation is required compared to a Hamming window [10]. The Blackman window function can also be used for these types of applications but the Blackman window has a wider main lobe width and if the main lobe width of any window function increases the ability to distinguish two closely spaced frequency components decreases.

In this paper an efficient adjustable window function based on Blackman window function is used for designing an FIR filter. For an efficient value of α, this window function provides higher side lobe attenuation comparison to Hamming and Hanning windows and the main lobe width of this window function is slightly greater than the hamming window. In this window function for a fixed length the main lobe width and amplitude of side lobe can be varied in the frequency domain by changing the value of α, which provides greater flexibility according to different applications. the design filter is compared for some different values of α. The application of this filter in speech signal filtering is also presented in this paper.

2. FIR FILTER DESIGN METHOD

In actual procedure for designing digital FIR filters first, the desired filter responses are characterized, and the filter coefficient values are calculated for a causal FIR filter. There are different methods to find the coefficients of digital filter from frequency specifications.

1. Fourier series method
2. The window method
3. Frequency sampling method

International Journal of Electrical, Electronics and Mechanical Controls
A Scientific and Scholarly Resource for Publication of Research Manuscripts

http://www.ijeemc.com

Volume 1 Issue 1 November 2012

4. Optimal filter design method

A simple and efficient way to design an FIR filter is window method. In the Window Design Method, The unit impulse response of ideal filter was obtained by applying inverse Fourier transform to the ideal frequency characteristics of digital filter. Then this unit sample response must be truncated at some point, this process is equivalent to multiplying it by a finite length window function. After truncation and windowing, an FFT is used to generate the corresponding frequency response of FIR filter. The frequency response can also be modified by choosing different window functions.

LOW-PASS FIR FILTER DESIGNING PROCEDURE USING WINDOW METHOD:-

1. Desired magnitude response of the ideal filter is given by the equation [12].

$$H_d(e^{jw})=1, \quad \text{when } |w| \leq w_c$$
$$H_d(e^{jw})=0, \quad \text{when } wc \leq |w| \leq w_s/2 \qquad (3)$$

2. Ideal Impulse response $h_d(n)$ of filter was obtained by applying inverse Fourier transform to the desired ideal frequency response $H_d(e^{jw})$ of digital filter [13].

$$h_d(n) = \frac{1}{2\pi}\int_{-wc}^{wc} e^{jnw}\, dw \qquad (4)$$

$$h_d(n)=\begin{cases}\frac{\sin[wcn]}{\pi n}, & n \neq 0\\ \frac{wc}{\pi}, & n = 0\end{cases} \qquad (5)$$

3. The window function w(n) is selected according to requirements of transition bandwidth and stop-band attenuation. In this paper an efficient window function based on Blackman window function given in equation (9) is used for designing FIR filter.

International Journal of Electrical, Electronics and Mechanical Controls

A Scientific and Scholarly Resource for Publication of Research Manuscripts

http://www.ijeemc.com

Volume 1 Issue 1 November 2012

4. After windowing the impulse response of the FIR filter can be described as:

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

5. The transfer function of designed FIR filter is obtained by taking the Fourier transform of h(n).

$$\mathrm{H}(e^{jw}) = \sum_{n=0}^{N-1}(\mathrm{h(n)}e^{-jnw}) \qquad (7)$$

3. AN EFFICIENT ADJUSTABLE WINDOW FUNCTION BASED ON BLACKMAN WINDOW

Rectangular, Hanning, Hamming and Blackman windows are types of fixed window functions. In fixed window functions main-lobe width of the filter can be varied by changing the length of the filter. On the other hand Adjustable windows have two or more independent parameters such as window length and one or more additional parameters that can control the other window's characteristics [4]. Hanning and Hamming windows are both commonly used in narrowband applications and other digital signal processing applications such as spectral analysis of different types of signal. But in some applications such as FFT, signal processing and measurement, higher side lobe attenuation is required compared to a Hamming window [10]. The Blackman window function can also be used for these types of applications but the Blackman window has a wider main lobe width. In spectral analysis of different types of signals small main lobe width of the window function is required. A Blackman window and a generalized adjustable Blackman window function are given in [11,14]. In generalized window function the width of main lobe can be varied by changing the value of α for a fixed length of the filter according to different requirements. For an efficient value of α, this efficient window function provides a higher side lobe attenuation comparison to Hamming window and the main lobe width is slightly greater than Hamming window. This generalized window is referred to as Blackman window for α = 0.16 and Hanning window for α = 0.

$$w(n) = \begin{cases} \frac{1-\alpha}{2} - 0.5\cos\left(\frac{2n\pi}{M-1}\right) + \left(\frac{\alpha}{2}\right)\cos\left(\frac{4n\pi}{M-1}\right), & 0 \le n \le N-1 \\ 0, & \text{otherwise} \end{cases} \quad (8)$$

4. APPLICATION OF LOW-PASS FIR FILTER IN FILTERING A SPEECH SIGNAL

FIR filters using the window method are widely used in different speech processing applications such as speech filtering, speech enhancement, noise reduction and frequency boosting. Speech filtering is a basic application of speech signal processing. This type of Digital filter is used to modify the frequency response of a speech signal according to desired speech processing application. Hanning and Hamming window function are commonly used in different signal processing applications but in some application higher side lobe attenuation is required. Designed low-pass FIR filter using efficient window (α=0.07) is used to eliminate the high-frequency spectrum of the speech signal. Simulation results show that for a efficiently selected value of α, this window function provides higher side lobe attenuation compare to Hamming window and the width of the main-lobe is slightly greater than Hamming window function. Fast Fourier transforms (FFT) is used for measurement and analysis purpose. Simulation results show that high-frequency part of speech signal was greatly reduced after applying design low-pass FIR filter.

PARAMETER SPECIFICATION:

International Journal of Electrical, Electronics and Mechanical Controls
A Scientific and Scholarly Resource for Publication of Research Manuscripts
http://www.ijeemc.com
Volume 1 Issue 1 November 2012

Firstly a speech signal is recorded in wave format and load into Matlab by using the following command

[x   fs]=wavread('speech.wav')

x is the sample speech signal and fs is the sampling frequency of speech signal. the Sampling frequency of recorded speech signal 8000 Hz.

Suppose we want to remove the high-frequency components above 1200 Hz of the speech signal. So

The cutoff frequency of the low-pass FIR filter is selected f_c= [1200] Hz.

The normalized cutoff frequency of the filter is obtained as $w_c = f_c/(f_s/2) = 0.3$

Consider order of the filter is 31.

The cutoff frequency and order of the filter can be changed according to desired specifications.

5. SIMULATION RESULTS

The comparison of design low-pass FIR filter using Hanning, Hamming, Blackman and adjustable generalized window function for α=0. 07 is shown in figure 2. Consider the length of the filter M=31 and normalized cutoff frequency is w_c=0.3. For length M=31 and normalized cutoff frequency w_c=0. 4, this efficient window function provides higher side lobe attenuation compare to Hanning and Hamming windows however the main-lobe width is slightly increased.

International Journal of Electrical, Electronics and Mechanical Controls

A Scientific and Scholarly Resource for Publication of Research Manuscripts

http://www.ijeemc.com

Volume 1 Issue 1 November 2012

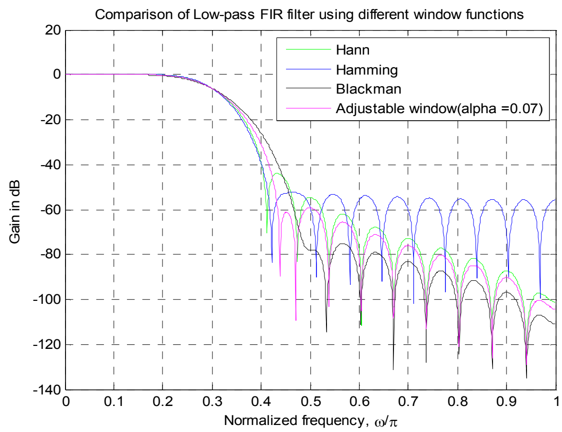

Figure2. The amplitude frequency characteristic curve of FIR filters for different window function for the length

M=31.

Designed low-pass FIR filter is used to eliminate the high-frequency spectrum of the speech signal. the Amplitude frequency characteristic curve of low-pass FIR filter is shown in fig3.

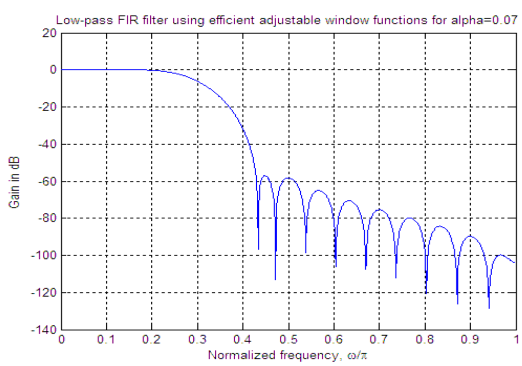

Figure3. Amplitude frequency characteristic curve of low-pass FIR filter for M=31.

Representations of recorded speech signal in time domain and frequency domain before and after filtering are shown in fig 4 and 5.

International Journal of Electrical, Electronics and Mechanical Controls

A Scientific and Scholarly Resource for Publication of Research Manuscripts

http://www.ijeemc.com

Volume 1 Issue 1 November 2012

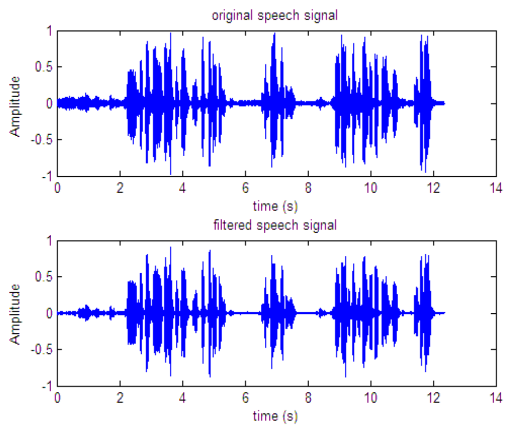

Figure 4.spectrum of input and output speech signal in time domain.

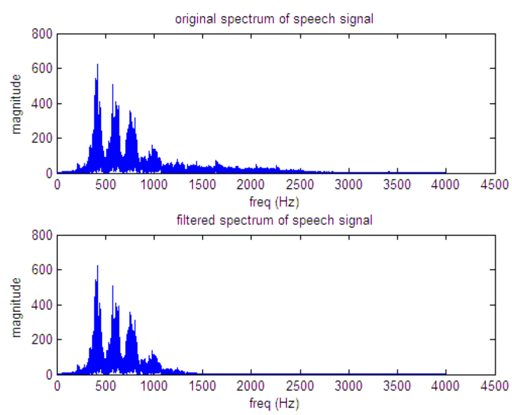

Figure 5.Frequency spectrum of original and filtered speech signal.

As shown in Figure 5, the designed low-pass filter significantly reduces high-frequency components, above 1,200 Hz, while letting the signal pass through the filter. A cutoff frequency of low-pass filter can be varied according to different speech processing applications.

International Journal of Electrical, Electronics and Mechanical Controls
A Scientific and Scholarly Resource for Publication of Research Manuscripts
http://www.ijeemc.com
Volume 1 Issue 1 November 2012

6 CONCLUSION

Digital filter plays a very important role in different digital signal processing applications. In this paper, low-pass FIR filter designed using efficient adjustable window function. These types of window functions are simple in operation, convenient and provides greater flexibility in digital signal processing applications. In some applications such as FFT, signal processing and measurement, higher side lobe attenuation are required compared to Hamming window. In the range $0 \leq \alpha \leq 1$, for small values of α, as the value of α increases the main lobe width is continuously increasing compared to Hanning and Hamming windows; however the amplitude of side lobe is also decreased. α is selected according to desired specifications based on different applications. Digital filters can also play a major role in speech signal processing applications. Application of designed low-pass filter in speech filtering is also presented in this paper. This filter can also be used for other speech signal processing applications. In digital filters different parameters such as filter order and the cutoff frequency can also be changed According to given specifications.

REFERENCES

1. J.G.Proakis and D.G.Manolakis, “Digital Signal Processing Principles, Algorithms and Applications” third edition Prentice-Hall,2002.
2. Sanjit K. Mitra, “Digital Signal Processing: A computer-base approach”, Tata McGraw-Hill, 2nd Ed,2001
3. Oppenheim, R. Schafer, and J. Buck, “Discrete-Time Signal Processing” second edition, Prentice-Hall,1999.
4. T. Saramaki, “Finite impulse response filter design,” in Handbook for Digital Signal Processing, Edited by S. K. Mitra and J. F. Kaiser, IBSN 0-471-61995-7 John Wiley &Sons Inc, 1993.

5. Sonika Gupta, Aman Panghal Performance "*Performance Analysis of FIR Filter Design by Using Rectangular, Hanning and Hamming Windows Methods*" International Journal of Advanced Research in Computer Science and Software Engineering Volume 2, Issue 6, June 2012.
6. Prof.GopalS.Gawande,Dr.K.B.Khanchandani,T.P.Marode "*performance analysis of fir digital filter design techniques*" International Journal of computing & Corporate research volume 2 issue 1 january 2012.
7. S. M. Shamsul Alam , Md. Tariq Hasan "Performance Analysis of FIR Filter Design by Using Optimal, Blackman Window and Frequency Sampling Methods". International Journal of Electrical & Computer Sciences IJECS-IJENS Vol: 10 No:01.
8. Mahrokh G. Shayesteh and Mahdi Mottaghi-Kashtiban "FIR filter design using a new window function" 978-1-4244-3298-1/09,2009 IEEE.
9. Najat Sh.Jasim Mohammed "The Performance Enhancement Study of FIR Filters Based on Adjustable Window Function" Eng a& Tech Journal vol30,no.5,2012.
10. Md Abdus Samad "A novel window function yielding suppressed mainlobe width and minimum sidelobe peak" International Journal of Computer Science, Engineering and Information Technology (IJCSEIT), Vol.2, No.2, April 2012.
11. FREDRIC J.HARRIS "On the Use of Windows for Harmonic Analysis with the Discrete Fourier Transform" processing of IEEE vol.66 no.1 january 1978.
12. Andreas Antoniou " Digital filters analysis, Design and applications ", Tata McGraw-Hill, 2nd Ed. 1999.
13. S.Salivahanan, A.Vallavaraj,C. Gnanapriya, Digital Signal Processing, Tata McGraw-Hill, 2000.
14. Window Function, Wikipedia, the free encyclopedia 2012 http://en.wikipedia.org/wiki/Window_Function.