Design and Analysis of Audio Frequency Filter with Hamming Window

1st Narsimhulu Lingampelly
Dept of ECE
Christu Jyothi Institue of Technology and Sciences
Jangaon,Telangana
narasimhalingampelly@gmail.com

2nd Rajitha Aripirala
Dept of ECE
Christu Jyothi Institue of Technology and Sciences
Jangaon,Telangana
aripirala06@gmail.com

Abstract –Audio frequency digital filters are pervasive in the modern communication systems hence good digital filter performance is important and hence to design a digital finite impulse response (FIR) filter satisfying all the required conditions is a demanding one. This report deals with the design of FIR digital filter using hamming window technique. Hence this type of filter plays very important role in spectral analysis of different types of audio signals.

Keywords—Audio Frequency Filters, FIR, Hamming Window

I. INTRODUCTION

The digital signal processing has become an extremely important subject. A fundamental aspect of digital signal processing is filtering. A digital filter is a system that performs mathematical operations on a sampled, discrete-time signal to reduce or enhance certain aspects of that signal. There are two types of digital filters on the basis of the impulse response of the filter: Infinite Impulse Response (IIR) filters, and Finite Impulse Response (FIR) filters. Digital filters with infinite duration impulse response referred to as IIR filters. IIR filters are recursive type filters where by the present output depends on the present input, past input and past outputs. Digital filters with finite duration impulse response referred to as FIR filters. FIR filters are non-recursive type filters where by the present output depends on present input and past inputs. FIR filters are widely used than IIR filters, because FIR digital filters have an exactly linear phase, always stable, non-recursive structure and arbitrary amplitude-frequency characteristics. In view of the design and simulation analysis, the design of digital filter is quickly and efficiently achieved by using powerful facilities like MATLAB.

II. DESIGN METHODOLOGY

To design the FIR filters the simple and effective way is window method. In this method infinite impulse response of the ideal prescribed filter is truncated by using a window function. The main advantage of this design technique is that the impulse response coefficient can be obtained in closed form and can be determined very quickly. The window method is simple in operation, easy to understand and very convenient method for designing digital FIR filter. The most popular and widely used window functions are rectangular window, hanning window, hamming window and Kaiser Window. The Rectangular window response provides side lobes which gives rise to ripples in pass band and stop band. The amplitude of the ripples is determined by the amplitude of the side lobes. For the rectangular window, the amplitude of the side lobes is unaffected by the length of the window. But the main lode width of rectangular window is narrower and higher. For fixed length, the hanning window has significantly lower side-lobe amplitude but the main lobe width is wider compared to Rectangular window. The Hamming window also has the same main lobe width of hanning window but it generates lesser oscillations in the side lobes than hanning window. Hence Hamming window is generally preferred rather than hanning 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. With regard to these window methods a Hamming window technique is implemented here. FIR filter design using Hamming window function provides sharp transition band. This type of filter is very useful in spectral analysis of audio signals.

III. Design And Implementation

A low pass filter with a cut off frequency (w_c) has an impulse response as sinc function, which extends up to time [t=$\pm\infty$] but finite impulse response filter considers only some portion of this impulse response. This abrupt truncation of impulse response creates oscillations in pass band and stop band which is called gibb's phenomenon. Truncation of impulse can be achieved by multiplying with a window function like hamming window , hanning window etc.

$h_d(n) \rightarrow impulseresponse\ of\ ideal\ LPF, w(n) \rightarrow window\ function\ , h(n) \rightarrow Truncated\ impulse\ response$

Considering hamming window

$$w(n) = 0.54 - 0.46\sin\left(\frac{2\pi n}{N}\right), \qquad h(n) = h_d(n) * w(n)$$

$N \rightarrow order\ of\ filter;\ N = (L-1);\ L \rightarrow length\ of\ impulse\ response$

Designing the filter for audio frequency requirements fp=20khz;Δf=200hz;fs=50kh, fc= fp+ $\Delta f/2$

$$\Delta f = 200hz, \Delta F = \frac{200}{50 \times 10^3} = 0.004;\ f_c = 20khz + 100; f_c = 20100hz$$

$f \rightarrow refers\ to\ original\ frequency; F \rightarrow Refers\ to\ normalised\ frequency$ Or refers to digital frequencies

Assume $A_p = 0.1db\ and\ A_s = 50db$

$$A_p = 20log_{10}\left(1+\delta_p\right), 10^{0.1/20} = 1 + \delta_p, \delta_p = 10^{0.1/20} - 1, \delta_p = 0.011$$

$$A_s = -20log_{10}\delta_s, 50 = -20log_{10}\delta_s, \delta_s = 10^{-50/20}, \delta_s = 0.0033$$

Satisfied window is hamming window

$$\Delta f = \frac{3.3}{L}, 0.004 = \frac{3.3}{L}, L = \frac{3.3}{0.02}, L = 825, order\ N = L - 1 = 824$$

IV. Simulation And Results

The design of FIR filter using hamming window function for different values of ripple and frequency are shown in the figure below. we considered a pass band ripple of 0.011 and a stop band ripple of 0.0033, a pass band frequency of 20khz and a stop band frequency of 2200khz , Considering these values the order of the filter becomes 824.The proposed FIR filter has implemented using MATLAB and results are shown in FIG.1 and FIG.2.

V. Conclusions

In this paper, an FIR filter has been designed using hamming window function. This window function is simple in operation and provides greater flexibility in digital signal processing. The digital filters are easily designed and also easy to use in several of signal filtering applications. The choice of technique to design the filter depends heavily on the decision of designer whether to compromise accuracy of approximation .FIR filter design by using hamming window is stable as compare to rectangular and hanning window techniques. Ripples in pass band are less in hamming as compared to rectangular and hanning and also hamming has a linear phase than rectangular and hanning windows.

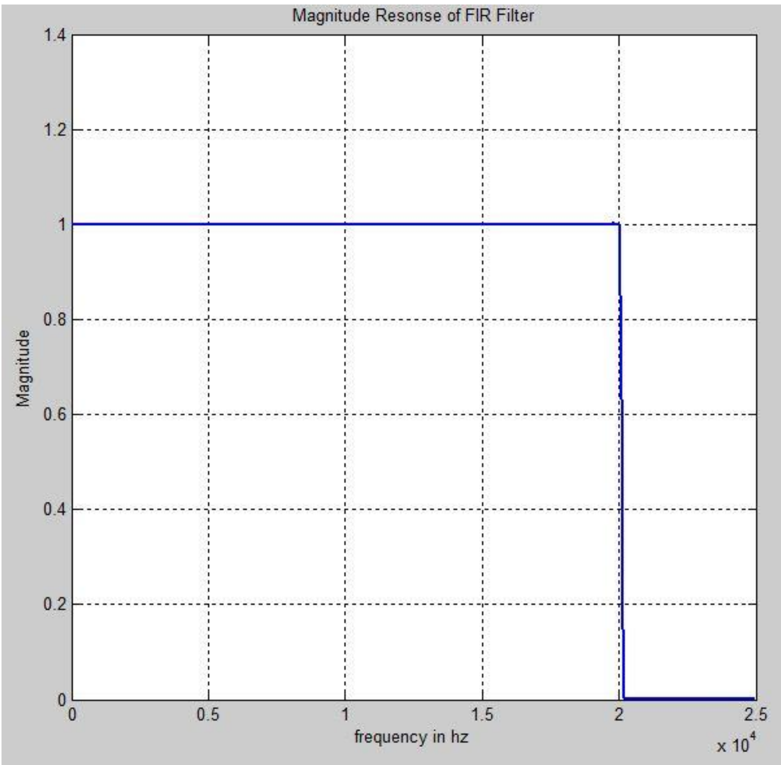

FIG.1.MAGNITUDE RSPONSE OF FIR FILTER

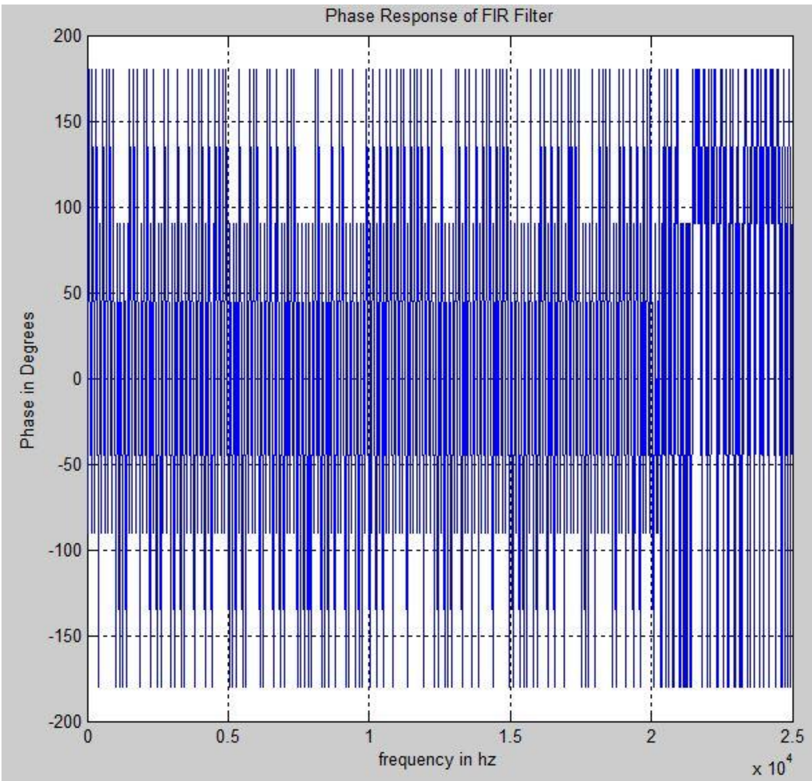

FIG.2.PHASE RSPONSE OF FIR FILTER

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