DESIGN ANALYSIS OF HIGH PASS FILTER USING DIFFERENT FILTER TECHNIQUES

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ABSTRACT

Filter is one of the most efficient part of digital processing system. In this paper, high pass filter has been designed and analysed using different design techniques. The developed High pass filter is designed using equiripple, generalized equiripple and constrained equiripple. The performance of all the design filters has been analysed in terms of magnitude response, phase response and filter order. It can be observed from the simulated results that performance of generalized equiripple is better as compared to the other techniques in terms of filter order, number of ripples in the spectrum and transition width with less computational complexity.

***Key Words:* Digital Filters, DSP, Equiripple Filters FIR, High Pass Filter, MATLAB**

I. INTRODUCTION

Digital signal processing (DSP) is technique of performing mathematical manipulation of an information signal in digital domain. It is characterized by the representation of discrete time, discrete frequency, or other discrete domain signals by a sequence of numbers and the processing of these signals. DSP performs amplification, attenuation, filtering, transformation, correlation, spectral analysis etc.operations in digital domain. For this separation of spectral envelope much of current work is to design and development of digital filter. The main objective of filtering is to alter the spectrum by the given specifications. Digital filters are classified by their use & implementation. These are called time domain or frequency domain based on their use, and finite impulse response (FIR) and infinite impulse response (IIR). The most important DSP technique is FIR filter. In the FIR system, the impulse response sequence is of finite duration, i.e.it has a finite number of nonzero terms. FIR filters are digital filters with finite impulse response. They are also known as non-recursive digital filters as they do not have the feedback. High pass filters are designed using Different types of FIR designed methods using the Software tool of MATLAB. The system with the impulse response

$$H(n) = a^n u(n) \qquad (1)$$

is a non zero for n>0.

The FIR filter depends only on the present and the past input samples. FIR filters are useful where the exact linear response is required. A general FIR filter does not have a linear phase response but this property is satisfied when

$$H(n)= h(M-1-n) \qquad (2)$$

For n=0,1…….(M-1)

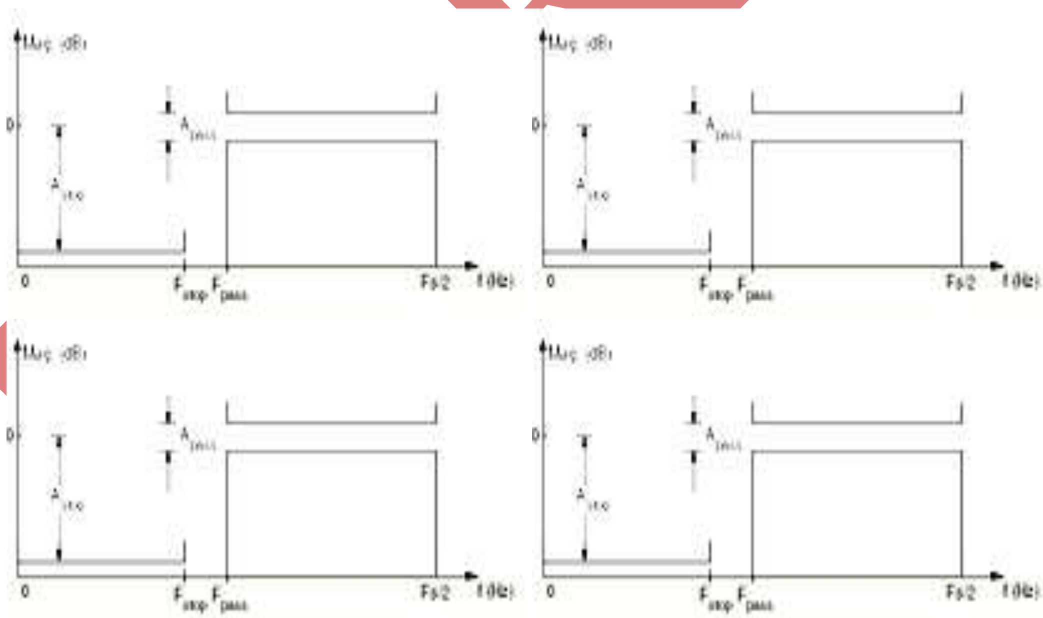
In case of IIR filter an algorithm is used in frequency domain to minimize mean square approximation error. The frequency and the Magnitude specifications are used in design but with different orders for each filter type. FIR filters also have a low sensitivity to filter coefficient quantization errors. This is an important property to have when implementing a filter on DSP processors or on an integrated circuit. The FIR filter of length M is described by the convolution of the unit sample response h(n) of the system with input signal x(n) and is represented by equation (3).

$$y(n)= h(k).x(n-k) \qquad (3)$$

Thus impulse response of the filter denotes the Coefficient of FIR filter. FIR filter design based on windows is simple and robust; however, it is not optimal:

i) The resulting pass-band and stop-band parameters are equal even though often the specification is more strict in the stop band than in the pass band unnecessary high accuracy in the pass band.

ii)



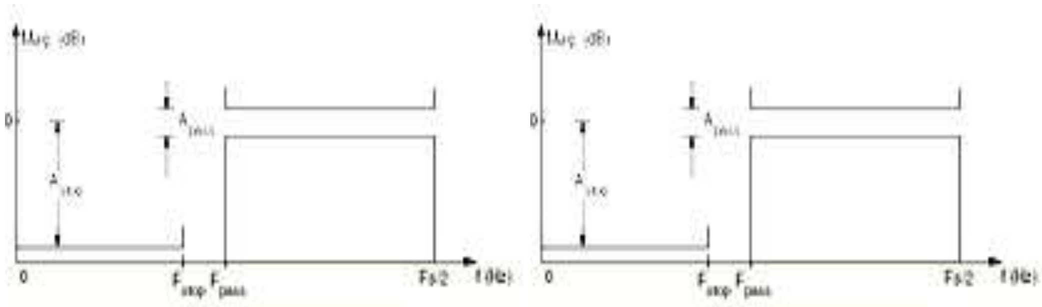


Fig.1 High Pass FIR Filter

iii) The ripple of the window is not uniform (decays as we move away from discontinuity points according to side-lobe pattern of the window) by allowing more freedom in the ripple behavior we may be able to reduce filter's order and hence its complexity.

II. EQUIRIPPLE FILTER TECHNIQUE

For an efficient and optimized digital FIR filter design, there are two methods available broadly, Equiripple filter design & Least Squares filter design. A general method for designing a filter is also Frequency Sampled FIR filter design but it is not an optimized design for error minimization. The basic knowledge I have is that Equiripple filter, as the name suggests, has equal ripples in passband & stopband, which means the signal distortion that happens at the edge of the passband due to presence of a large ripple is avoided in Equiripple design But, Equiripple design has a large transition band, thus limiting the total passband width. On the other hand, in a Least Squares design, the transition band width is smaller than for Equiripple design, hence the passband width is more, but the passband ripple are not equi-ripple & exhibit a spike at the passband edge due to Gibbs phenomenon, which causes signal distortion at the edge. The optimal method of calculating FIR filter coefficients is very powerful, very flexible and very easy to apply.

The least-square criterion of minimising

$$\varepsilon^2=\int E^2(\omega)d\omega \qquad (4)$$

is not entirely satisfactory. A better approach is to minimize the maximum error at each band

$$\varepsilon=\max_{\omega} |E(\omega)| \qquad (5)$$

an equiripple filter – a filter which amplitude response oscillates uniformly between the tolerance bounds of each band. The various optimal filter design methods are Least square, Equiripple, Maximally flat, Constrained Equiripple,Generalized equiripple, Constrained band equiripple, etc. The basic initiative in each method is to design the filter coefficients till the response is finest. The optimal method is based on the concept of the equiripple pass-band and stop- band. Equiripple design produces most efficient filter with minimum possible order.

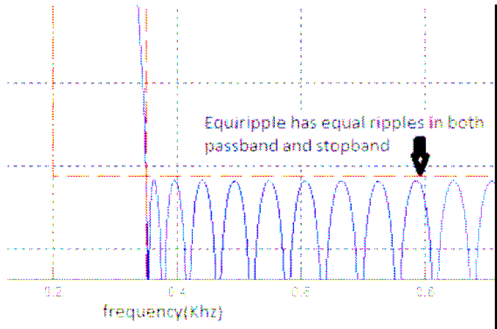


Fig-2 Equiripple response showing ripples in PB & SB

III. DESIGN SIMULATIONS

Table 1 shows the parameter specifications and Table 2 shows the Orders at methods of designed the FIR filter. All the methods of FIR filters are implemented by software package MATLAB at the same sampling frequency and the same pass-band and the same stop- band frequency. In this paper different methods used for design the FIR filter at different orders. The generalized equiripple is better than the other equiripple filter techniques. Based upon the study of all equiripple filters, equiripple filter with magnitude response, phase response are shown in figure 2.It is viewed in optimum design criterion that the weighted approximation error between the desired frequency response and the actual frequency response is spread across the pass band and stop band of the filter minimizing the maximum error. The resulting filter designs have ripples in both the pass band and stop band. As the same frequency & same filter density of pass band and stop band, the filter order is larger than the generalized equiripple filter shown in fig.5. Length of the Equiripple Filter L= 17.

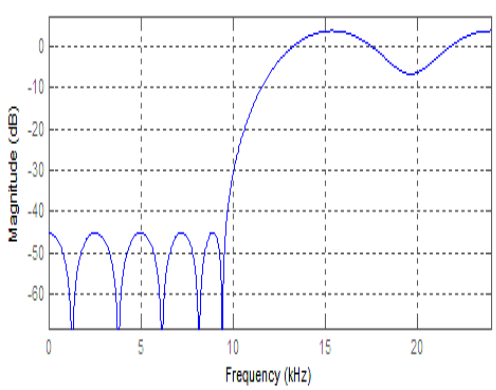


Fig.3 Equiripple Filter magnitude response

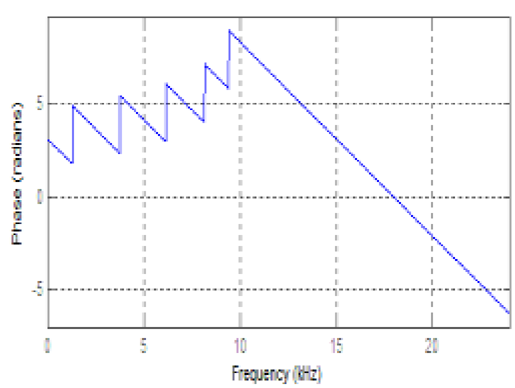


Fig.4 Equiripple Filter Phase response

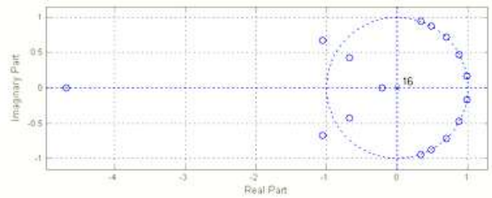


Fig.5 Equiripple Filter Pole Zero Plot

Length of the generalized equiripple ,L= 5

In generalized equiripple, filter can be realised efficiently in hardware. It has the passband attenuation Apass = 35 and Astop = 40 with density factor of 16 which gives the realized response approximately equal to ideal High pass FIR filter. Generalized equiripple also has linear transition band with less error than other equiripple filters. The phase response and pole zero filter response are given in fig. 6 and fig 7. Length of the constrained equiripple ,L= 5

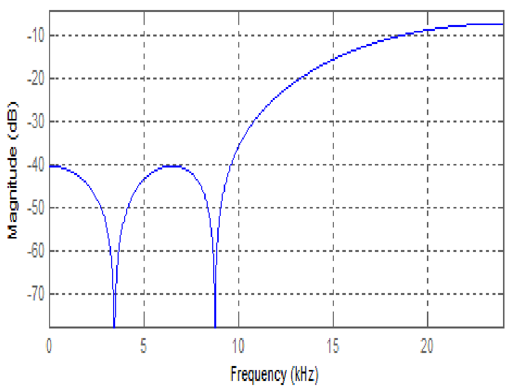


Fig.6 Generalized Equiripple Magnitude response

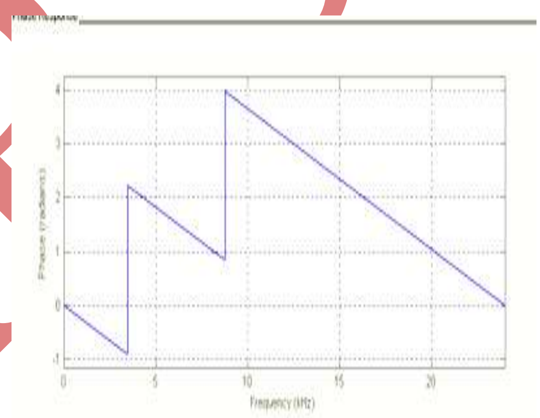


Fig.7.Phase Response

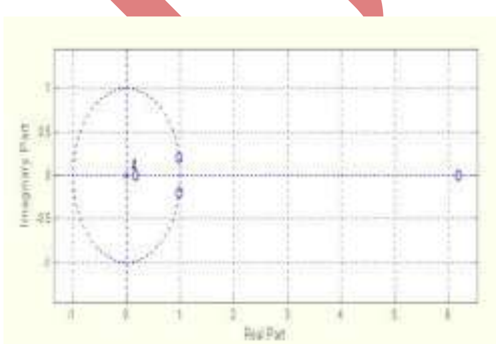


Fig.8 Pole Zero Plot Diagram

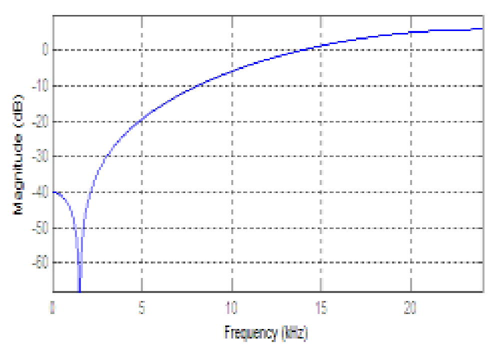


Fig.9: Constrained Equiripple Magnitude Response

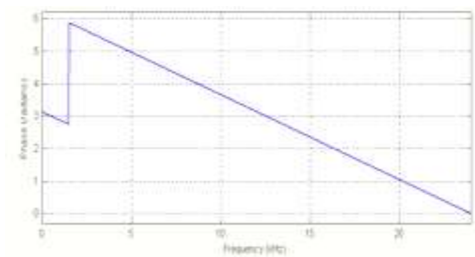


Fig. 10: Phase Response

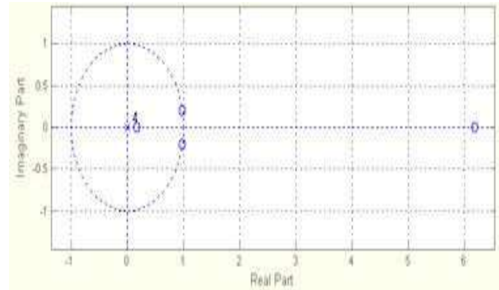


Fig.11: Pole Zero Plot

IV. RESULT ANALYSIS

Based on the filters specifications considering the Sampling frequency 'Fs' as 48000 Hz, Passband Frequency 'Fpass' as 9600Hz, Stopband Frequency 'Fst' as 12000Hz, Passband Attenuation 'Ap' as 35dB, and Stopband Attenuation' Ast' as 40 dB. Taking density factor as 16 for all the High Pass FIR filter cases we have simulated the results below on MATLAB.

Parameters	Values
Normalized Sampling Frequency	48000Hz
Normalized Pass-band Frequency	9600Hz
Normalized Stop-band Frequency	12000Hz

Table1: Parameter Specification

In order to obtain the ideal response of High Pass FIR filter, the filter order should be minimum. In above all equiripple factors the number of multipliers and adders are more in other equiripples as compared to generalized filter and also has the sharpest response with required transition width.

FIR design method	No. of multipliers	No. of adders
Equiripple	**17**	**16**
Generalized Equiripple	**5**	**4**
Constrained Equiripple	**5**	**4**

Table 2: Performance Analysis

S.No	FIR Filter DesignTechniques	Pass band Attenuation	Stop band Attenuation	Order of Filter
1.	Equiripple	35	40	16
2.	Generalized Equiripple	35	40	4
3.	Constrained Equiripple	35	40	4

Table 3: Resource Requirement

The magnitude response of constrained band Equiripple has the minimum passband attenuation which tells that it will be computationally complex to implement it on hardware and also it will consume large time for design but it will have the good response amongst all the Equiripple techniques.

V. CONCLUSION

There are several techniques available for designing linear phase FIR filters. Every method has its own advantages and disadvantages. The choice of technique depends on the decision of designer whether to compromise accuracy of approximation or ease of design. The window technique is most suitable for prototype filters like the low-pass, high- pass, band-pass etc. The frequency sampling technique is suitable for designing of filters with arbitrary frequency response. Optimal filter design techniques give best filters for given length of the FIR filter with transition width, but these techniques are more complex.

VI. ACKNOWLEDGEMENT

The authors would also like to thank Director, National Institute of Technical Teachers' Training & Research, Chandigarh, India for their constant inspirations and support throughout this research work.

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