Controlling the Stop Band Attenuation in FIR Filter

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ABSTRACT

The Stop band attenuation can be controlled by the stop band weight by keeping pass band weight constant. The improvement in stop band attenuation can be obtained by compensation in pass band ripples. In this paper different simulation results are obtained with stop band weights variations and keeping the pass band weight constant. It can be observed that significant improvements are obtained by variations in stop band weight at the expense of pass band ripple. There is an improvement of 0.0709 % in the stop band attenuation.

Keywords: *Bartlett window, Blackman window, Equiripple, FFT, FIR Filter, Hamming window, Windowing Technique.*

I. INTRODUCTION

Digital signal processing is the processing of digital signals which consists a stream of numbers in binary form. DSP has its existence on the truth that it is possible to represent a signal in digital form [1]. This activity is done by sampling the voltage level at regular time intervals and manipulating the voltage level at that instant into a digital number proportional to the voltage. This process is performed by a device called as Analogue to Digital converter, A to D converter or ADC. ADC is presented with a steady voltage whilst it is taking its sample, a sample and hold circuit is required to sample the voltage just before the conversion. As it completes the sample and hold circuit is ready to update the voltage again ready for the next conversion. In this method a succession of samples is made.

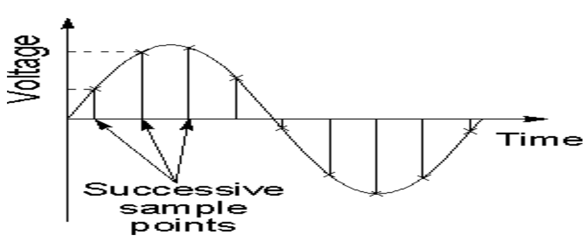

Fig. 1 Waveform sampling for Digital Signal Processing.

Once in a digital format the real DSP is able to be undertaken. The digital signal processor performs complicated mathematical calculations upon the representation of the signal. However to use the signal it then regularly needs to be converted back into the analogue form where it can be amplified and passed into a loudspeaker or headphones. The arrangement that performs this function is called a digital to analogue converter, [2] D to A converter or DAC.

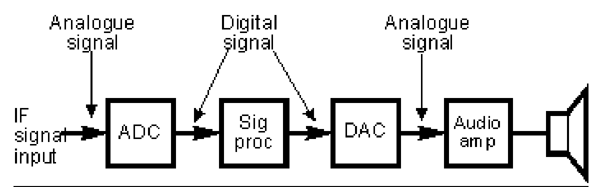

Fig.2 Digital Signal Processor block diagram.

Digital filters serve two basic purposes. One is separation of signals that have been combined and the other is restoration of signals that have been distorted in some way. Analog filters can be used for these tasks however; digital filters can achieve far better results.

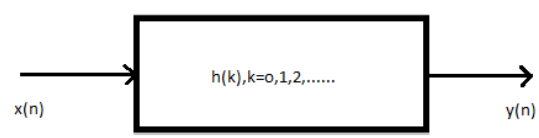

Fig.3 Digital Filter.

Where x(n) is input sequence, y(n) is output sequence and the impulse response of the filters can be represented by the sequence h(k). Mainly the digital filters are of two types. One is Finite Impulse Response or FIR filter and the other one is Infinite Impulse Response or IIR filter. Filters are also known as non recursive filters where IIR filters are recursive filters [1], [2]. IIR filters are difficult to control and have no particular phase, whereas FIR filters make a linear phase always possible. IIR can be unstable, whereas FIR is always stable. IIR when compared to FIR can have limited cycles, but FIR has no limited cycles. IIR is derived from analog, whereas FIR has no analog history. IIR filters make polyphase implementation possible, whereas FIR can always be made casual. The output response of FIR filter is given by the equation:

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

The output response of IIR filter is given by the equation

$$y(n) = \sum_{k=0}^{\infty} h(k)x(n-k) \qquad (2)$$

II. DESIGN METHODS

FIR filter is a filter whose response to any finite length input is of finite duration, as it settles to zero in finite time. This is in contrast to IIR filters, which may have internal feedback and may continue to respond indefinitely, usually decaying.

Properties of FIR filters

FIR Filter can easily be designed to be "linear phase". Put simply, linear-phase filters delay the input signal but don't distort its phase. FIR filters are simple to implement. On most DSP microprocessors, the FIR calculation can be done by looping a single instruction. FIR filters are suited to Multirate application.

Types of FIR Filter

Lowpass filter, Highpass filter, BandPass filter, Band reject filter.

Designing of FIR Filter

Here our main emphasis is on Band rejected FIR filter. A band rejected filter is an electronic device or circuit that rejects signals between two specific frequencies to stop, but that discriminates against signals at other frequencies. The well known methods of design techniques for linear phase FIR Band reject filter are [2], [3], Fourier method, Window method, Frequency sampling method**.**

Window 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 [3]. The method is to make an ideal filter in the frequency domain, and then translate it into the discrete time domain. However this will give an infinite impulse response. To compensate for this, a window function is multiplied into the ideal impulse response.

Design Procedure

The basic design procedure is first decide the desired frequency response of the filter $H_d(e^{jw})$ then calculate the desired impulse response $h_d(n)$ by inverse fourier transform of $H_d(e^{jw})$, because $h_d(n)$ is infinitely long [4], we have to deal with it by window function to get to the unit impulse response h(n).

Now it is written as

$$h(n) = w(n).\, h_d(n) \qquad (3)$$

Where w(n) is the window function. Window techniques used for the designing of FIR Band Reject filter [4], [5] are Hamming Window, zHanning Window, Bartlett Window, Blackman Window etc.

Hamming Window	$w(n) = \alpha - \beta\cos(\frac{2\pi n}{N-1})$, $\alpha=0.54$, $\beta= 1-\alpha=0.46$
Hanning Window	$w(n) = 0.42-0.5\cos\frac{2\pi n}{N-1}+0.08\cos\frac{4\pi n}{N-1}$
Bartlett Window	$w(n) = 1-\frac{2(n-\frac{N-1}{2})}{N-1}$
Blackman Window	$w(n)= 0.42 - 0.5\cos\frac{2\pi n}{M-1} + 0.08\cos\frac{4\pi n}{M-1}$

The frequency response of designed FIR filter is obtained by taking Fourier transform of h(n) [6].

$$H(ejw) = \sum_{n=0}^{N-1}(h(n)e^{-jnw}) \qquad (4)$$

III. LOW-PASS EQUIRIPPLE FIR FILTER DESIGN

The equiripple FIR filter design is based on classical polynomial approximation theory with respect to a performance metric which is different than the least squares metric used by Fourier series design methods i.e. we are concerned with minimizing the maximum error over a set of frequencies. The Equiripple design method is referred to as Minimax Method. In an equiripple FIR, the local extrema in the an equiripple Low Pass filter, the passband and stopband edges must be specified along with the maximum passband ripple and maximum stopband ripple.

The equiripple filter which meets these specifications will have the smallest possible order of any FIR filter. The solution to the equiripple FIR filter problem is based on famous alteration theorem from polynomial approximation theory. Based on this theorem, the Parks-McClellan techniques make use of an iterative algorithm called Remez Exchange Algorithm to find the Equiripple FIR and the details are complicated. Most digital filter design packages incorporate the Parks-McClellan technique for equiripple FIR Design. Using the Parks-McClellan technique, it is possible to design not only lowpass filter but high pass filters, bandpass filters, arbitrary magnitude response, Hilbert transform and differentiator filter as well.

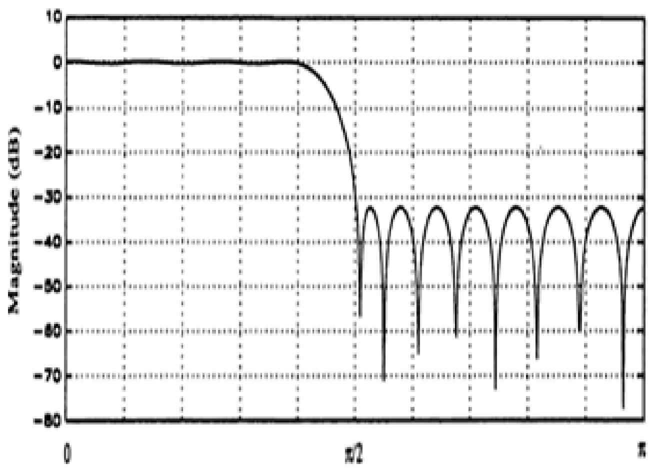

Fig.4 Frequency Response for Equiripple FIR filter

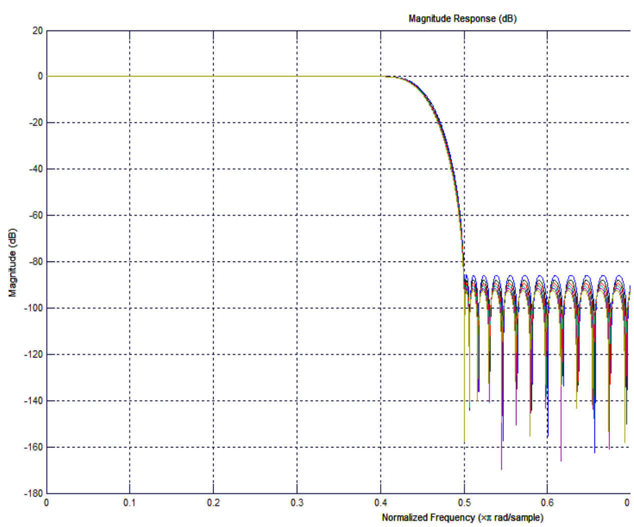

Fig. 5 Combined Magnitude Response for different weights

IV. RESULT ANALYSIS

Comparison of different Passband weights with varying stopband weights in terms of filter parameters without changing the order is observed from the different magnitude response of the Equiripple filter is shown in table.1.

Table1. Comparison of different Weights

Order	PassBand Weight	StopBand Weight	Normalized Frequency	Magnitude Response
N=100	1	1	0.5241	-85.816
	1	1	0.5236	-87.938
	1	3	0.5234	-89.3936
N=100	1	4	0.5233	-90.4532
	1	5	0.5229	-91.4834
	1	6	0.5227	-92.3735

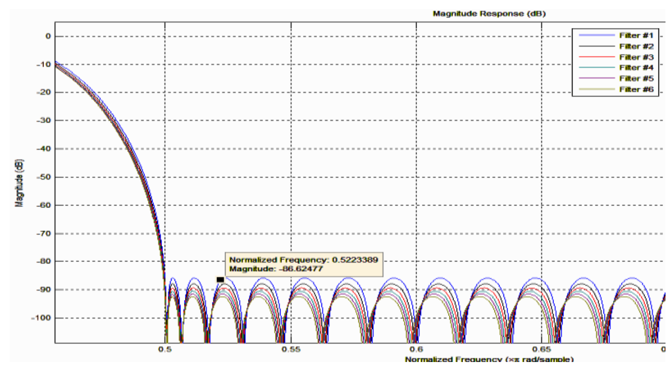

Fig.6 Stop Band Attenuation for Diffetent weights.

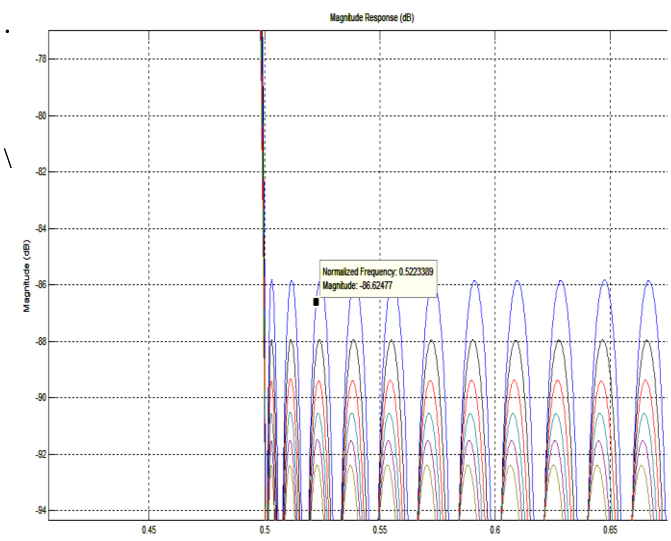

Fig. 7 StopBand Attenuation.

V. CONCLUSION

Stopband and Passband Weights specify the relative importance of the passband ripple versus stopband attenuation. By default, passband and stopband are equally weighted to 1. Now if we fix the passband weight and increase the stopband weight, the stopband attenuation decreases at the expense of increased passband ripples. In the comparison shown above the stopband weight is increased up to the maximum value '6' which resulted, the minimum sideband attenuation and the maximum bandpass ripples among all other results.

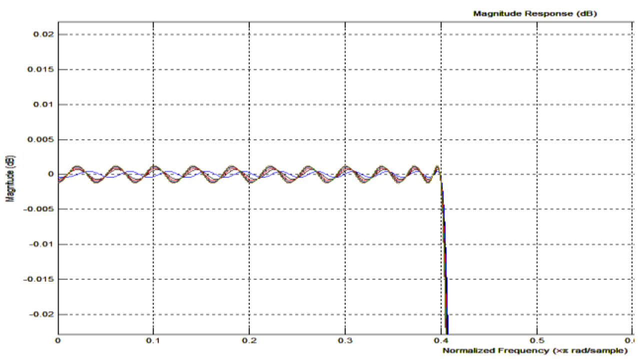

Fig.8 Passband Ripples

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