Optimal FIR filters Analysis using Matlab

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ABSTRACT

In this paper optimal FIR filter techniques are used to design any type of FIR filter. This techniques design the smallest-order FIR filter for a given set of specification. It is based on the concept that the filter exhibits ripples of equal length in the band-pass and band-stop. The design algorithm of these Equiripple FIR filter uses iterative polynomial interpolation. Input signal has a great influence on the performance of the system in digital control system. Processing of input signal is important to get useful signal. In the processing of digital signal Finite impulse response (FIR) filter plays an important role. Using Matlab the FIR filter is designed and simulated. Different methods like frequency sampling, window function and convex optimization technique are processed using Matlab in the design of FIR filter. By comparing the signal's amplitude-frequency diagrams which have been generated the filtering effect of different digital filters are analyzed by using FIR digital filters which are designed to process the input signal based on the Matlab function. This paper shows the experimental results show the FIR filters designed [7].

Keywords: Matlab, Optimal, Remez method, optimization, FIR filter, Equiripple Design.

1. INTRODUCTION

In general case filter is applied to illustrate a device that distinguishes, according to some attribute of the objects applied at its input. For frequency domain concept this is used to pass the frequencies of pass-band and attenuates all other frequencies outside the pass-band. So the properties of the filtering depend on the frequency response function which settles on the choice of system parameters. Indeed FIR filters are used to design a filter where there is a requirement for a linear phase characteristic within the pass band of that filter. The performance of different methods depends on the similarities of its frequency response with the desired frequency response. To follow this philosophy, there are threez techniques to design FIR filters. In this paper, the spectral responses of those methods are analyzed [9].

2. DESIGN OF OPTIMAL FIR FILTER

To design a low pass filter, the desired frequency response is given by

$$H_d(\omega) = \begin{cases} 1 & 0 \le \omega \le \omega_p \\ 0 & \omega_s \le \omega \le \pi \end{cases}$$

For this system, consider N = 51, ω S = 1 rad/s and the corresponding frequency response, determined by from,

$$H_1(\omega) = h(25) + 2\sum_{n=0}^{24} h(n)\cos\omega(25-n)$$

There are various methods to determine the filter coefficients. Among these, the most widely used method is that of McClellan, Parks and Rabiner. Their program is capable of designing optimal FIR filters including low-pass; high-pass, band-pass and band reject filters. We wish to design a low pass linear phase FIR frequency response such that it has very flat pass-bands and low stop-bands ripples. Also assume that the response has very sharp attenuation slop. To determine the filter coefficients, it is required to sample the desired frequency response considering non uniform frequency spacing and put the values of those samples. The values of filter impulse response coefficients, h(n) are cited in the Table 1. From these values the desired equation of |H(ω)|, that is H1(ω) is found in terms of impulse response, h(n) and which show flat, that is ripple less pass-band and there are ripples in the stop-band which are gradually decreasing [9].

3. FIR Design by Optimisation: Least-Square Method

We now present a method that approximates the desired frequency response by a linear-phase FIR amplitude function according to the following optimality criterion. The integral of the weighted square frequency-domain error is given by

$\varepsilon 2 = \int E2(\omega)d\omega$

and we assume that the order and the type of the filter are known. Under this assumptions designing the FIR filter now reduces to determining the coefficients that would minimize ε^2.

3.1 Equiripple Design

The least-square criterion of minimizing is not entirely satisfactory. A better approach is to minimize the maximum error at each band

$\varepsilon \ 2 = \int E2(\omega)d\omega$

$\varepsilon = \max\omega \ |E(\omega)|$

The method is optimal in a sense of minimizing the maximum magnitude of the ripple in all bands of interest, the filter order is fixed.

It can be shown that this leads to an Equiripple filter – a filter which amplitude response oscillates uniformly between the tolerance bounds of each band.
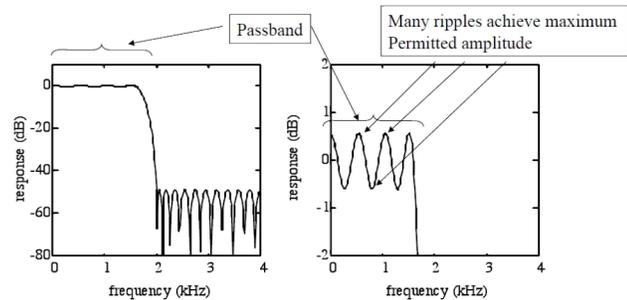
3.2 Remez Method

There exists a computational procedure known as the Remez method to solve this mathematical optimization problem. There are also exist formulae for estimating the required filter length in the case of low-pass, band-pass and narrow transition bandwidths. However, these formulae are not always reliable so it might be necessary to iterate the procedure so as to satisfy the design constraints [5].

4. FIR FILTER DESIGN USING MATLAB

4.1 Realization of Optimized Equiripple Method using Matlab

We use remez function in the optimized Equiripple method.

Function c=remez(n,f,a,w,'ftype') where
n shows filter order;
f shows a vector, and is a positive number between 0 and 1;

a shows a vector, and represents the amplitude in the specified frequency domain;
w shows the weighted value of each frequency band;
b shows a vector whose length is n+1.

4.2 Designing Equiripple low-pass filter using Remez function

Requirements:

- Stop-band cutoff frequency is 0.6π and Pass-band cutoff frequency is 0.5π, and the sampling frequency is 2 kHz.

- The stop-band attenuation is $\geq$40dB. Stop-band ripple is 0.0 and pass-band ripple is 0.1710.
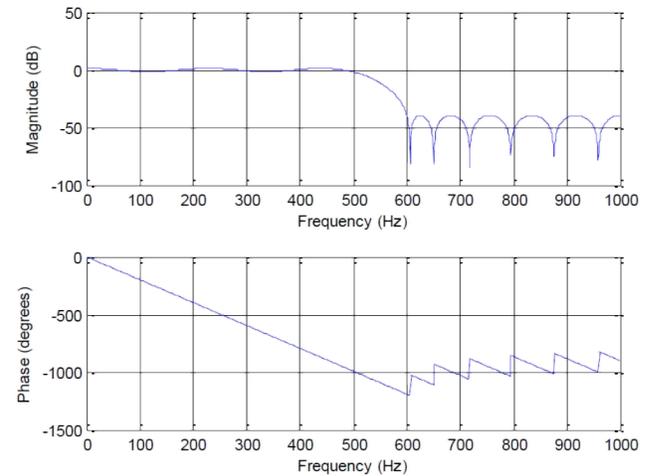

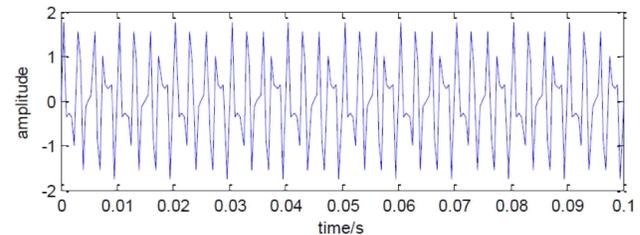*Fig. 4.2a Equiripple Low-pass filter gain response.*

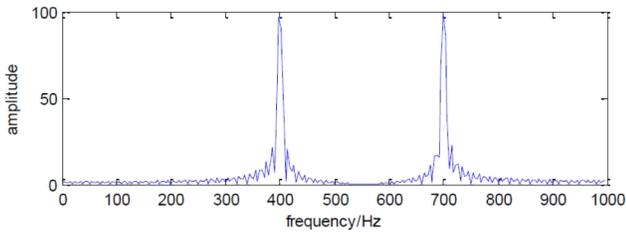*Fig. 4.2b Time domain before filtering*

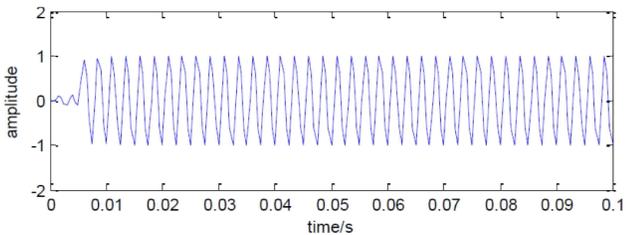M. Iqbal

Fig. 4.2c Frequency domain before filtering

Fig. 4.2d Time domain after filtering.

Fig. 5.2e Frequency domain after filtering

Fig. 4.3a Equiripple Band-pass filter gain response.

Fig. 4.3b Time domain before filtering

Fig. 4.3c Frequency domain before filtering
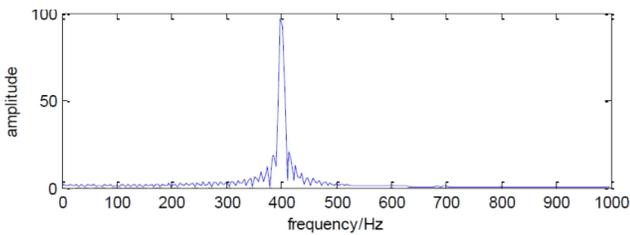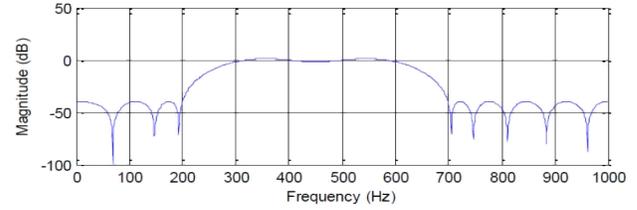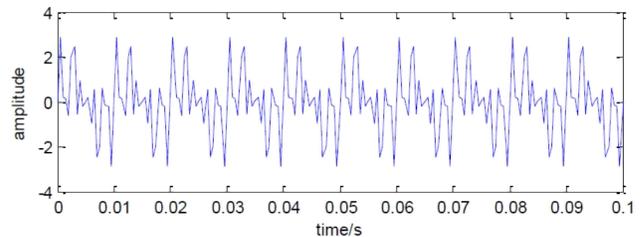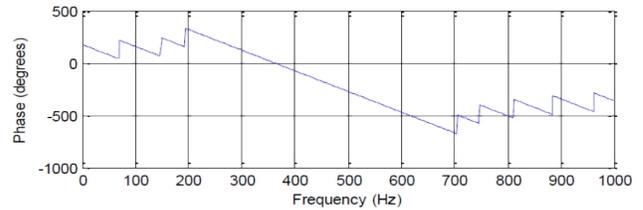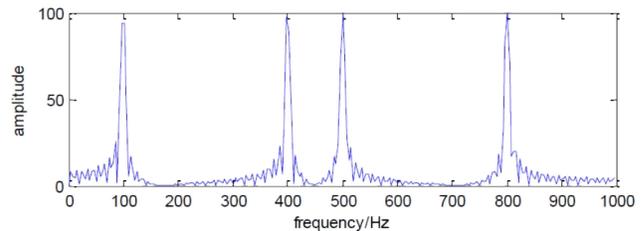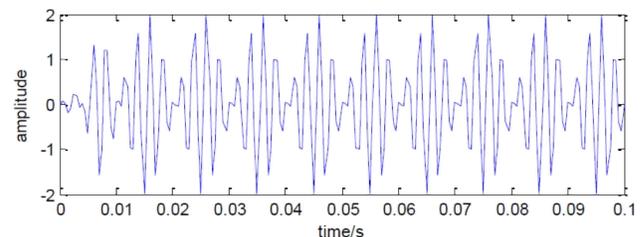

Fig. 4.3d Time domain after filtering

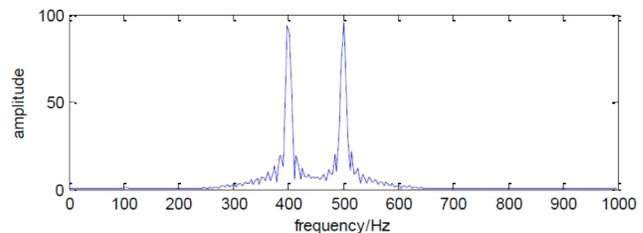*Fig. 4.3e Frequency domain after filtering*

Comparing Figure 4.2b-2d and 4.2c-e, we see that that input signal is made up of two superposition signals with different frequencies The pass-band is from 0 to 0.5kHz and stop-band is from 05k to 1 kHz. When filtering is done so the signal with frequency 0.4 kHz is not filtered because it is in the range of band-pass filter while the signal with frequency 0.7 kHz is filtered because it is in the range of band-stop filter.

4.3 Designing Equiripple band-pass filter using Remez function.

Requirements:
- Sampling frequency is 2 kHz. Stop-band cutoff frequency is 0.2 and 0.7π and Pass-band cutoff frequency is 0.3 and 0.6π.
- The stop-band attenuation is $\geq$40dB. Stop-band ripple is 0.01 and pass-band ripple is 0.17.

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In the above figure we see that input signal is made up of four superposition signals having different frequencies. The stop-band is from 0 to 0.2 kHz and 0.7k to 1 kHz and the band-pass is from 0.3k to 0.6 kHz. So those signal whose frequency are in this range of band-pass will be kept and those signal whose frequencies are not in this range will be filtered out i.e. in figure 5.3e frequency 0.4 and 0.5 kHz are in range of band-pass so they are kept but frequency 0.1k, 0.8 kHz is in the range of band-stop so it is filtered out.

4.4 Designing Equiripple band-stop filter using Remez function.

Requirements:

- Sampling frequency is 2 kHz. Pass-band cutoff frequency is 0.2 and 0.7π and Stop-band cutoff frequency is 0.3 and 0.6π.

- The stop-band attenuation is $\geq$15dB. Pass-band ripple is 0.01 and stop-band ripple is 0.17.
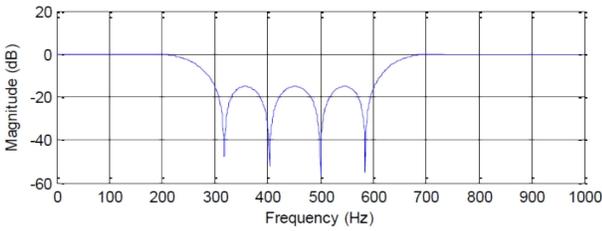

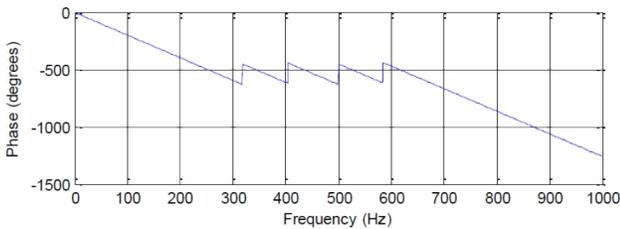*Fig. 4.4a Equiripple Band-stop filter gain response.*

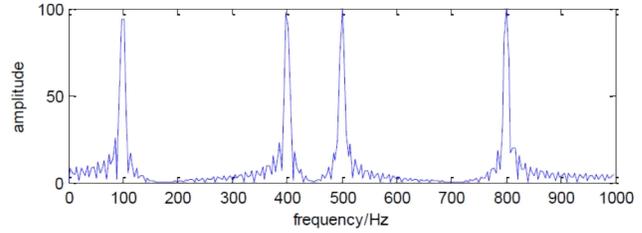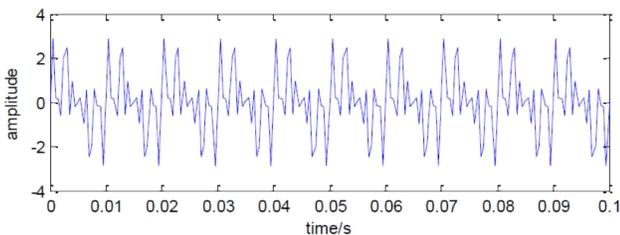*Fig. 4.4b Time domain before filtering.*

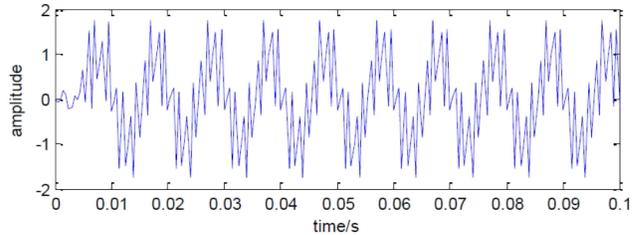Figure 4.4c Frequency domain before filtering.

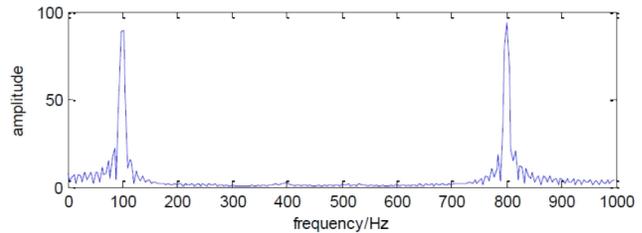*Fig. 4.4d Time domain after filtering.*

Fig. 4.4e Frequency domain after filtering.

In the above figure we see that input signal is made up of four superposition signals having different frequencies. The pass-band is from 0 to 0.2 kHz and 0.7k to 1 kHz and the stop-pass is from 0.3k to 0.6 kHz. So those signal whose frequency are in this range of band-pass will be kept and those signal whose frequencies are not in this range will be filtered out i.e. in figure 5.3e frequency 0.1 and 0.8 kHz are in range of band-pass so they are kept but frequency 0.4k, 0.5 kHz is in the range of band-stop so it is filtered out.

5. CONCLUSIONS

The design of linear phase optimal FIR filters, with very flat pass-bands, can be done by solving the frequency response equations for different frequency components. To solve the filter parameters, one can use an iterative algorithm like Remez exchange algorithm [3]. The response of Remez algorithm. We can also do this, by sampling the desired frequency response with non-uniform frequency spacing as shown the response in Fig. 1. Here we see that the degree of flatness in pass-bands and ripples in stop-bands depend on the transition bandwidth. If we set this bandwidth low then ripples increases and vice versa. We can get better stop-band attenuation using Blackman

window method. In summary, the optimal solution is not always a good solution to the filter design problem. The major disadvantage of the window design method is the lack of precise control of the critical frequencies, such as and, in the design of a low pass FIR filter and its value depends on the type of window and the filter length N [3]. The frequency sampling method provides an improvement over the window design method, since $H_r(\omega)$ is specified at the frequencies and transition band is a multiple of [3]. This filter design method is particularly attractive when the FIR filter is realized either in the frequency domain by means of the DFT or any of the frequency sampling realizations.

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