Design and Analysis of Fir Filters Based on MATLAB

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Abstract - **In digital control system, interference, which is mixed in the input signal, has a great influence on the performance of the system. Therefore, processing of input signal has to be done to get useful signal. Finite impulse response (FIR) filter plays an important role in the processing of digital signal.**

Designing the FIR filter by MATLAB can simplify the complicated computation in simulation and improve the performance. By using the methods of window function, frequency sampling and convex optimization techniques, the design of FIR filter has been processed by MATLAB. In the view of the designed program of MATLAB, the amplitude-frequency characterization can be achieved. By using the FIR digital filters which have been designed to process the input signal based on the MATLAB function, the filtering effect of different digital filters is analyzed by comparing the signal's amplitude-frequency diagrams which have been generated.

The experimental results show that the FIR filters designed in this paper are effective.

Index Terms - **FIR filter, MATLAB, window function, frequency sampling, optimization, amplitude-frequency characterization.**

I. INTRODUCTION

In signal processing, 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. This is in contrast to the other major type of electronic filter, the analog filter, which is an electronic circuit operating on continuous-time analog signals.

The digital filter is a discrete system, and it can do a series of mathematic processing to the input signal, and therefore obtain the desired information from the input signal. The transfer function for a linear, time-invariant, digital filter is usually expressed as

$$H(z) = \frac{\sum_{j=0}^{M} b_j z^{-j}}{1 + \sum_{i=1}^{N} a_i z^{-i}}$$

Where, a_i and b_i are coefficients of the filter in Z-transform.

1.2 FIR filter

In signal processing, a finite impulse response (FIR) filter is a filter whose impulse response (or response to any finite length input) is of finite duration, because it settles to zero in finite time. This is in contrast to infinite impulse response (IIR) filters, which may have internal feedback and may continue to respond indefinitely (usually decaying). [3]

The impulse response (that is, the output in response to a Kronecker delta input) of an Nth-order discrete-time FIR filter lasts exactly N + 1 samples (from first nonzero element through last nonzero element) before it then settles to zero.

FIR filters can be discrete-time or continuous-time, and digital or analog. The finite impulse response (FIR) filter is one of the most basic elements in a digital signal processing system, and it can guarantee a strict linear phase frequency characteristic with any kind of amplitude frequency characteristic. Besides, the unit impulse response is finite; therefore, FIR filters are stable system. The FIR filter has a broad application in many fields, such as telecommunication, image processing, and so on.

The system function of FIR filter is.
$$H(z) = \sum_{n=0}^{L-1} h[n] z^{-n},$$
Where L is the length of the filter and h[n] is the impulse response.

An FIR filter has a number of useful properties which sometimes make it preferable to an infinite impulse response (IIR) filter. FIR filters:

II-SIMULATION OF ELECTRONIC COMMUNICATION SYSTEM

The concept of telecommunications and electronic systems simulation

Telecommunication is the exchange of signs, signals, messages, words, writings, images and sounds or

information of any nature by wire, radio, optical other electromagnetic systems.

Telecommunication occurs when the exchange of information between communication participants includes the use of technology. It is transmitted through a transmission medium, such as over physical media, for example, over electrical cable, or via electromagnetic radiation through space such as radio or light. [10]

System simulation technology refers to computer simulation technology, which developed since 1970 combining modern computers and simulation software. Computer simulation has high precision, versatility, good repeatability, rapid modeling, and low-cost advantages.

III-DESIGN OF FIR FILTER

FIR filters are filters having a transfer function of a polynomial in z - and is an all-zero filter in the sense that the zeroes in the z-plane determine the frequency response magnitude characteristic.

FIR filters are particularly useful for applications where exact linear phase response is required. The FIR filter is generally implemented in a non-recursive way which guarantees a stable filter.

FIR filter design essentially consists of two parts.
a. Approximation problem
b. Realization problem

The approximation stage takes the specification and gives a transfer function through four steps. They are as follows:
a. A desired or ideal response is chosen, usually in the frequency domain.
b. An allowed class of filters is chosen (e.g. the length N for a FIR filters).
c. A measure of the quality of approximation is chosen.
d. A method or algorithm is selected to find the best filter transfer function.
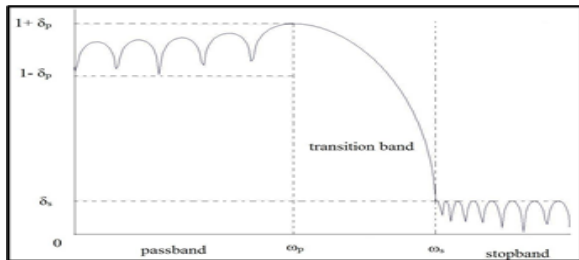

Figure 4.1 Amplitude-frequency characteristic of lowpass filter

As the figure shows, the transition band of the filter is between the passband and the stopband. The frequency ω_p denotes the edge of the passband, and the band-edge frequency ω_s defines the edge of the stopband. So, the difference of ω_s and ω_p is the width of the transition band, i.e. $\omega_t = \omega_s - \omega_p$.

The ripple in the passband of the filter is denoted as δ_p, and the magnitude of the filter varies from $1-\delta_p$ to $1+\delta_p$. δ_s is the ripple in the stopband. Usually we use a logarithmic scale to show the frequency response, hence, the ripple in the passband is $20\log10\delta_p$ dB, and the ripple in the stopband is $20\log10\delta_s$ dB.

III-RESULTS

(A)THE FIR FILTER DESIGN BASED ON MATLAB

The realization of window function method by Matlab
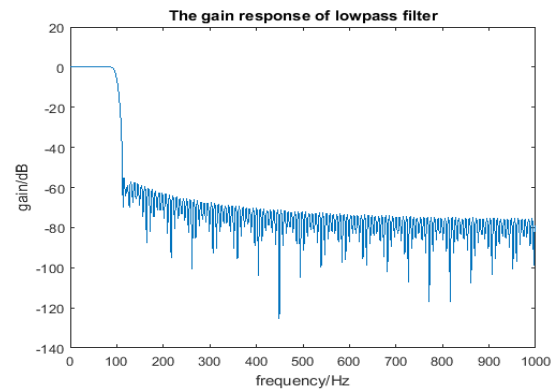1. Design a lowpass filter using the window function method.

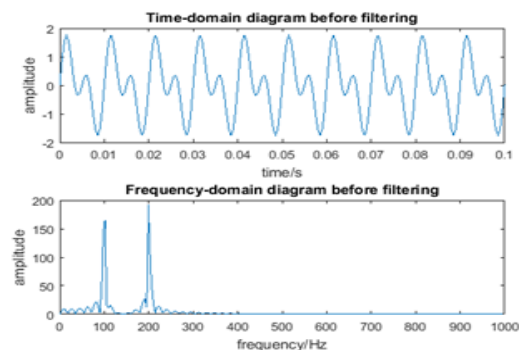Figure 5.1 Gain response of lowpass filter using Hamming window.

Figure 5.2 Time-domain and frequency-domain diagram before filtering

2. Design a bandpass filter using the window function method.
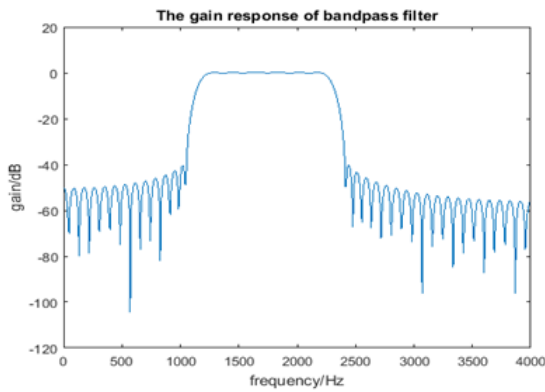
Requirements:

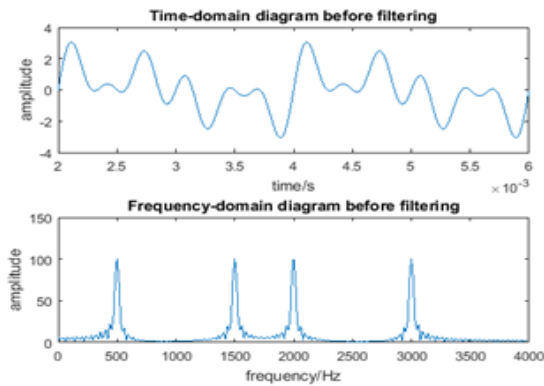Figure 5.3 Gain response of bandpass filter using Kaiser Window.

Figure 5.4 Time-domain and frequency-domain diagram before filtering

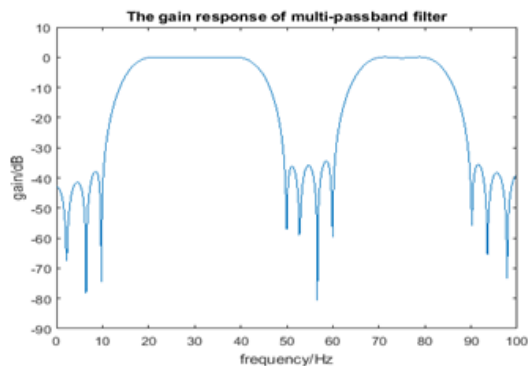3. Design a multi-passband filter using the window function method.

Figure 5.5 Gain response of multi-passband filter using Kasier window.

(B) The realization of frequency sampling method by MATLAB

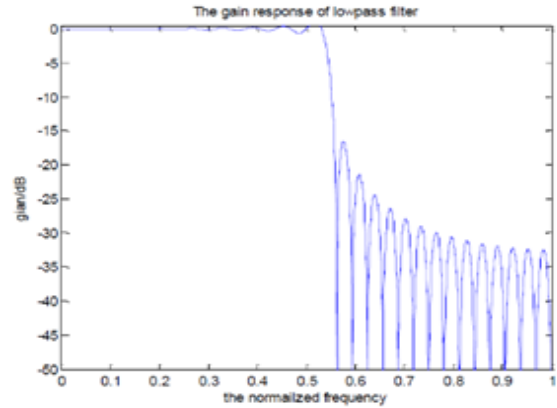1. Design a lowpass filter using frequency sampling method.

Figure 5.6 Gain response of lowpass filter using frequency sampling method
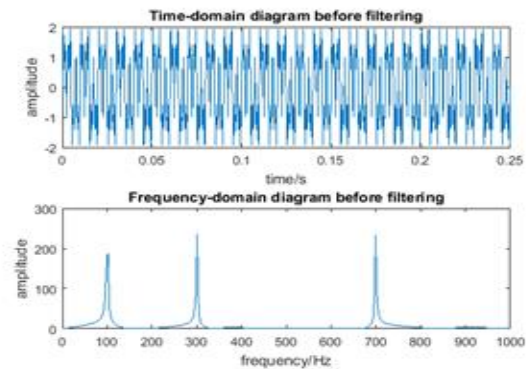

Figure 5.7 Time-domain and frequency-domain diagram before filtering

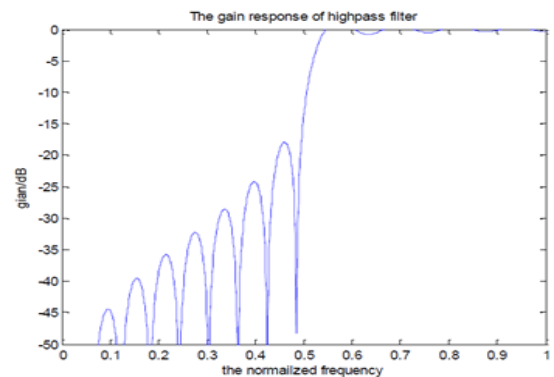2. Design a high pass filter using frequency sampling method.

Figure 5.8 Gain response of highpass filter using the frequency sampling method.

(C)The realization of Optimized Equiripple Method by MATLAB

1. Design equiripple lowpass filter using function Remez.

Requirements:
- The passband cutoff frequency is 0.5π, stopband cutoff frequency is 0.6π and the sample frequency is 2000Hz.
- The stopband attenuation is greater than or equal to 40dB, passband ripple is 0.1710 and stopband ripple is 0.01.
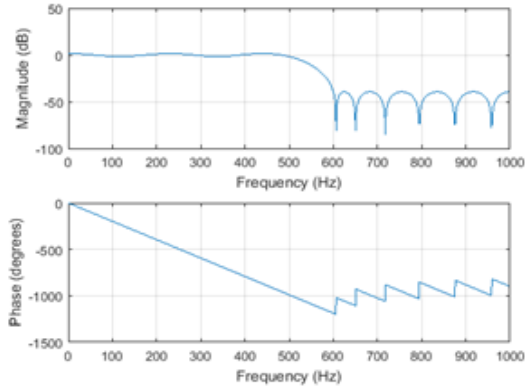

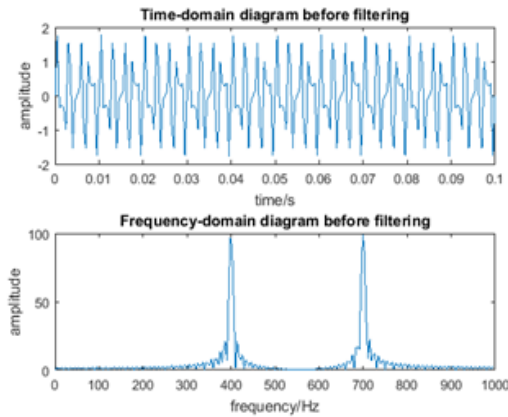Figure 5.9 Gain response of equi-ripple lowpass filter

Figure 5.10 Time-domain and frequency-domain diagram before filtering
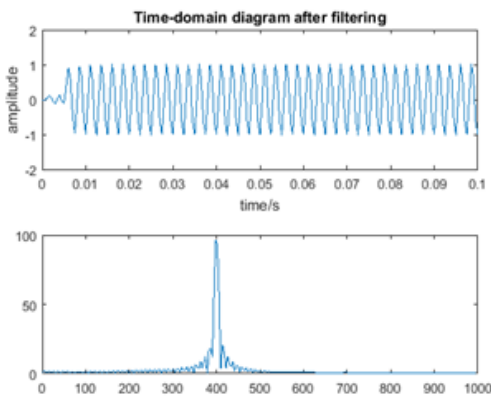

Figure 5.11 Time-domain and frequency-domain diagram after filtering

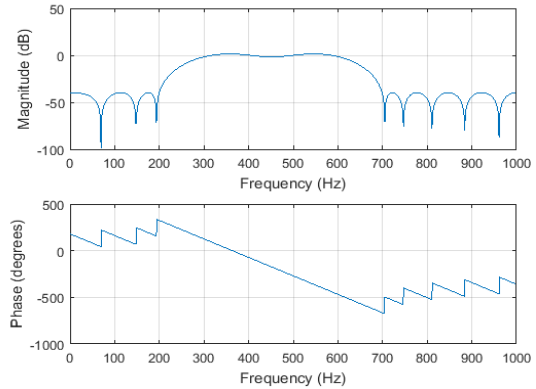2. Design equiripple bandpass filter using function Remez.

Figure 5.12 Gain response of equiripple bandpass filter
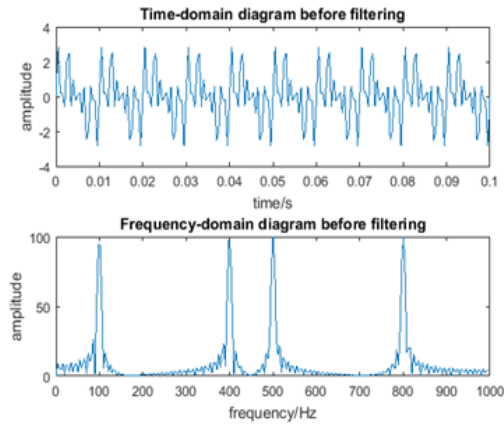

Figure 5.13 Time-domain and frequency-domain diagram before filtering

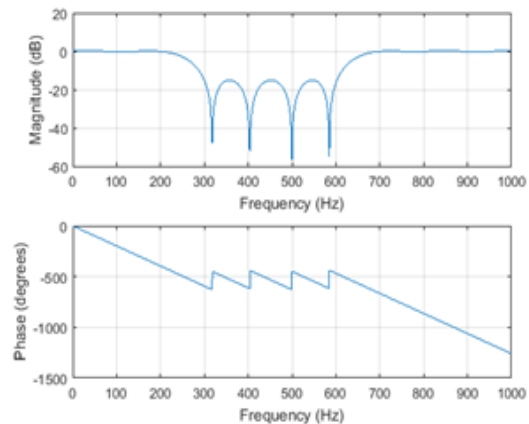3. Design equiripple band stop filter using function Remez.

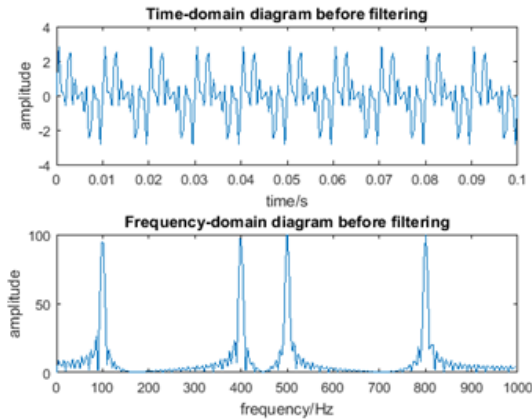Figure 5.14 Gain response of euiripple band stop filter.

Figure 5.15 Time-domain and frequency-domain diagram before filtering

FIR filter with CVX

The response of the FIR filter is given by.

$$H(\Omega) = \sum_{n=0}^{N-} h(n)e^{-i\Omega n}$$

The convex optimization criterion is given by.

Minimize $||H(\Omega)||_\infty$ or $||H(\Omega)||_2$

Subject to $||H(\Omega_p)||_\infty$

CVX is used to design a filter to achieve optimization. The number of samples has been set as N=50. So, there are 50 samples, and the desired frequency is 2000Hz. The passband starts from 1500Hz to 2500Hz, and the stopband starts from 0 to 1500Hz, and 2500Hz to 4000Hz.

IV-CONCLUSION

FIR filter design essentially consists of two parts.
1. Approximation problem
2. Realization problem

There are essentially three well-known methods for FIR filter design namely:
1. The window method
2. The frequency sampling technique
3. Optimal filter design methods

The above three the major advantages of using window method is their relative simplicity as compared to other methods and ease of use. The fact that well defined equations are often available for calculating the window coefficients has made this method successful.

The basic idea behind the window design it to choose a proper ideal frequency-selective filter (which always has a non-causal, infinite-duration impulse response) and then to truncate (or window) its impulse response to obtain a linear-phase and causal FIR filter.

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