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Estimation and Tuning of FIR Lowpass Digital Filter Parameters

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Abstract— Finite impulse response (FIR) digital filters are known to have many distinguishable features such as stability, linear phase characteristic at all frequencies and digital implementation as non-recursive structures. FIR filter design can be considered as an optimization problem. In this paper an estimation method of FIR filter parameters is proposed. The method relies on establishing a relationship between the signal input parameters and the filter parameters. An FIR lowpass filter was implemented and tested using various parameter values. The results showed efficient performance characteristics of FIR lowpass filters.

Keywords— FIR lowpass filter, transition bandwidth, sampling frequency, window length, filter order, and stopband attenuation.

I. INTRODUCTION

Designing digital filters involves the determination of a set of filter coefficients to meet a set of design specifications. Digital filters can be classified in two categories: finite impulse response (FIR) filters and infinite impulse response (IIR) filters. By varying the weight of the coefficients and number of filter taps, virtually any frequency response characteristics can be realized with an FIR filter. FIR filter is an attractive choice because of the ease in design, linear phase shift property and stability [[1], [2]].

Designing an FIR digital filter require specifying passband, stopband, and transition band. In passband, frequencies are needed to be passed unattenuated. In stopband, frequencies needs to be passed attenuated. Transitionband contains frequencies which are lying between the passband and stopband. Therefore, the entire frequency range is split into one or even more passbands, stopbands, and transition bands. In practical, the magnitude is not necessary to be constant in the passband of a filter. A small amount of ripple is usually allowed in the passband. Similarly, the filter response does not to be zero in the stopband. A small, nonzero value is also tolerable in the stopband as shown in Fig 1. The transition band of the filter as shown in Fig.1 is between the passband and the stopband. The frequency ω_p denotes the edge of the passband, and the band-edge frequency ωs 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 δp, and the magnitude of the filter varies from 1-δp to 1+ δp. δs are the ripple in the stopband.

Usually we use a Logarithmic scale to show the frequency response, hence, the ripple in the passband is 20log10δp dB, and the ripple in the stopband is 20log10δs dB.[[3], [4]]

Different methods are used to design a lowpass filter such as: window method, frequency sampling method, and optimization method.

In window method, a truncated ideal lowpass filter with a certain bandwidth is generated, and then we use a chosen window to get certain stopband attenuation. The filter order L can be adjusted to meet a specified roll-off rate in the transition band. Any finite-length (order) of the ideal lowpass impulse response may be considered as the product of the infinite-length lowpass impulse response and a window function *W*, which has a finite number of contiguous nonzero-valued samples [[1]-[4]]

$$b = \frac{\sin(\omega_c[n-M])}{\pi[n-M]} W_L[n-M],$$

where L is the window length, M=(L-1)/2, 0≤ n ≤ L-1, and WL[n] is generally a function $F_E[n]$ which has even symmetry about *M* defined as

$$W_L[n] = \begin{cases} F_E[n] & 0 \ll n \ll L-1 \\ 0 & otherwise \end{cases}.$$

The result is a finite-length or truncated lowpass filter.

The frequency sampling method will work in following way, we start in the frequency domain, and sample the desired frequency response $H(e^{j\Omega})$ with N evenly-spaced samples instead of a continuous frequency, and get $H_d(e^{j\Omega})|_{\Omega=2\pi k/N}$ (k=0,1,…, N-1). Then, let H(k)= Hd(k)= $H_d(e^{j\Omega})|_{\Omega=2\pi k/N}$, we get the unit impulse response, h(n) = IDFT[H(k)], where IDFT is Inverse Discrete Fourier Transform. The inverse DFT then yields an impulse response which will lead to a filter whose frequency response the same as that of the specification exactly at the location of the frequency samples. The advantage of this method is that we can design filters directly in the frequency domain, but the disadvantage is that the sampling frequency can only be integer times of 2π/N, and we cannot ensure a random cutoff frequency. Lacking precise control of the specified frequencies, like ωp and ωs, is the most serious disadvantage of the window function method in the design of a lowpass FIR filter. The frequency sampling method is better than the window method in the aspect that the real-valued frequency response characteristics *Hr (ω)* is specified, which can be either zero or unity at all frequencies, except the transition band. The Chebyshev approximation method offers completely control of the filter requirements. As a result, this method is more preferable than the other two. It is based on the Remez exchange algorithm, which minimizes the error with respect to the max-norm [[1]-[4]].

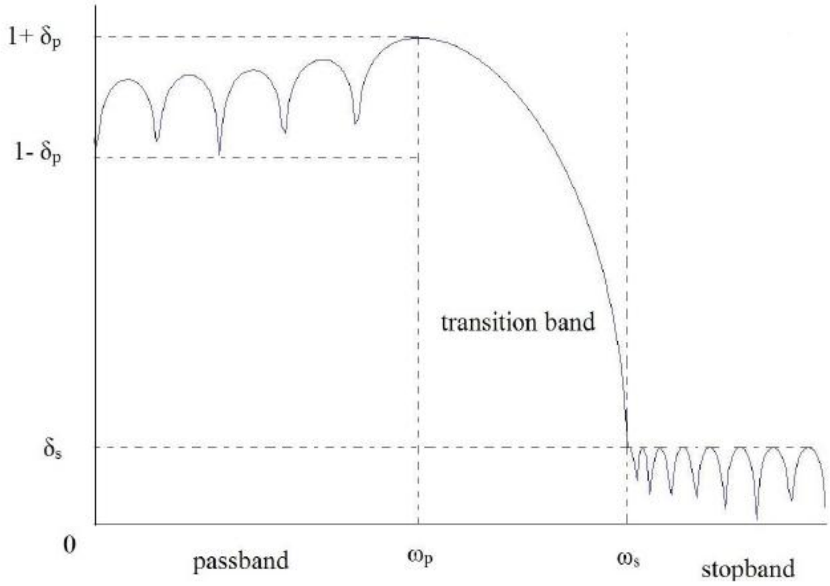

Fig. 1 Amplitude-frequency characteristics of lowpass filter

II. LITERATURE REVIEW

Designing FIR digital filter is an important field in digital signal processing. Several conventional and new methods have been suggested to design efficient digital filters. Here we present some of new researches in this field.

In [5] the researchers presented the design and implementation of a lowpass, highpass and a handpass Finite Impulse Response (FIR) Filter using SPARTAN-6 Field Programmable Gate Array (FPGA) device. The filter performance is tested using Filter Design and Analysis (FDA) and FIR tools from Mathworks. The FDA Tool is used to define the filter order and coefficients, and the FIR tool is used for Simulink simulation.

In [6] the FIR filter is designed and simulated using Matlab. 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.

In [7] the researchers used an evolutionary algorithm to design linear phase FIR lowpass filter. They presented a novel approach using cuckoo Search Algorithm (CSA). Cuckoo-inspired algorithms are population based new optimization purposes. A CSA is proposed to achieve flat passband and high stopband attenuation in FIR filter design.

III. EXPERIMENTAL REALIZATION OF LOWPASS FILTER

Designing a lowpass filter using the window function method with these requirements [8]

- Using the Hamming window and the sampling frequency is 2000Hz.
- Cutoff frequency of passband is 0.1π, and cutoff frequency of stopband is 0.17π.
- The passband attenuation is less than or equal to 0.1dB, and the stopband attenuation is greater than or equal to 50dB.

Here, the cutoff frequency is all normalized frequency, we get this from ω=2πf/fs, where fs is the sampling frequency.

The order of the digital filter must be carefully selected in order to get the required filter.

A matlab code was written to design a lowpass filter and it was implemented using different filter orders. (The implementation results are shown in Figures 2, 3 and 4.

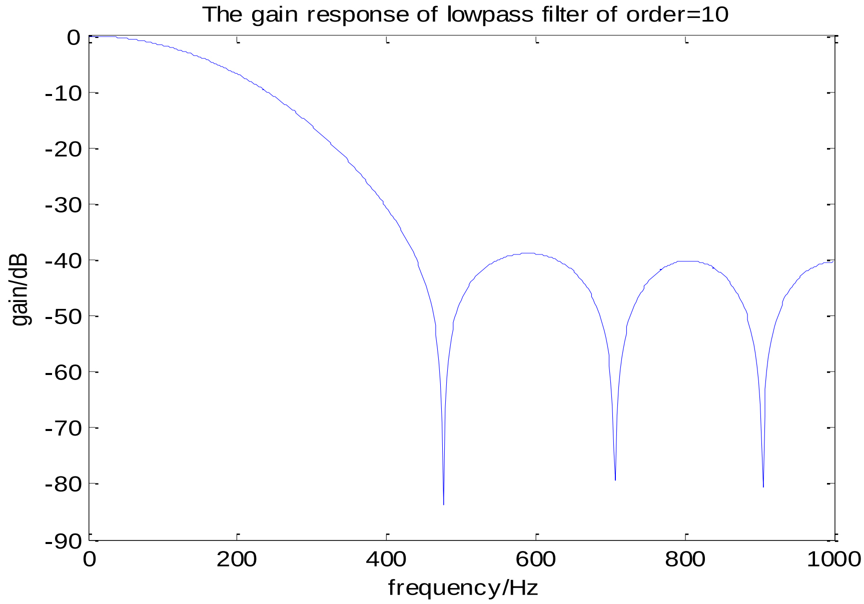

Fig. 2 Gain response of lowpass filter of order=10

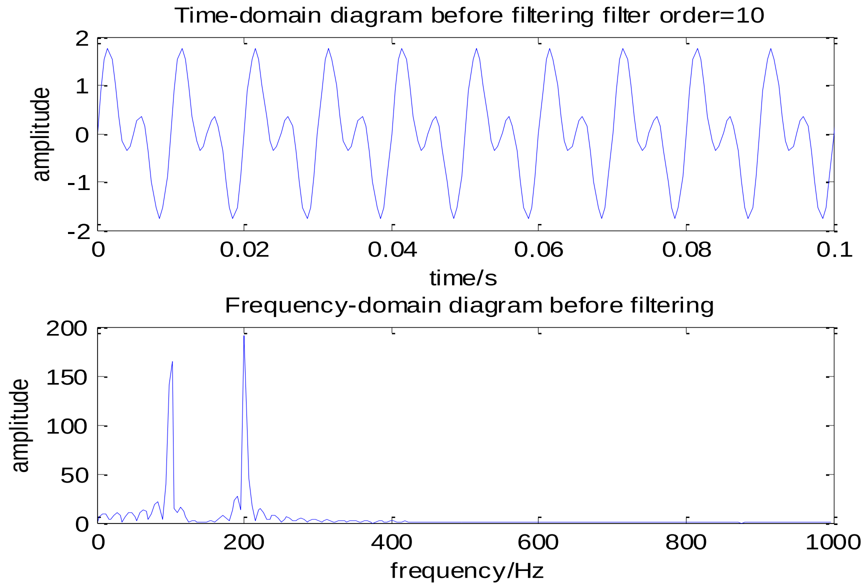

Fig. 3 Time and frequency domain before filtering

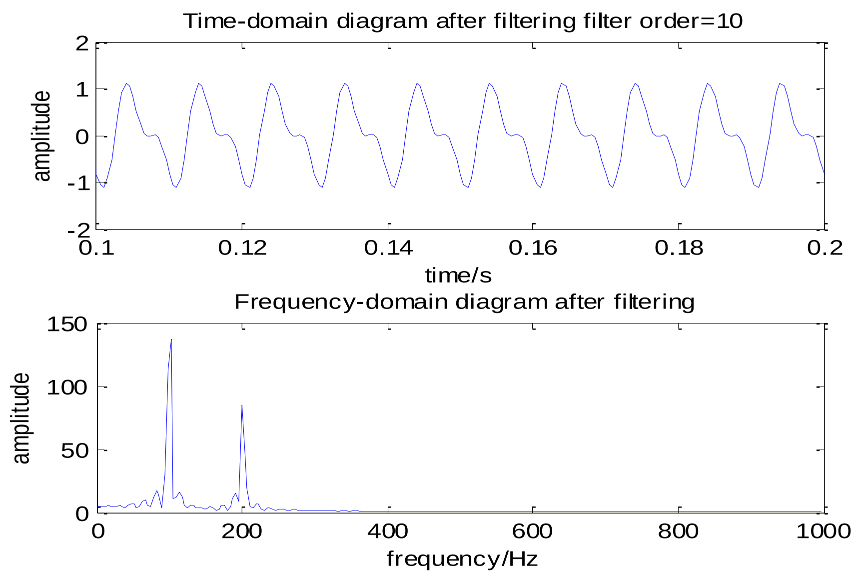

Fig. 4 Time and frequency domain after filtering

Here from Fig. 4 we can see that the filter failed to give one band frequency.
Increasing the filter order to 100 we can get the following results as shown in Fig.s 5, 6 and 7.

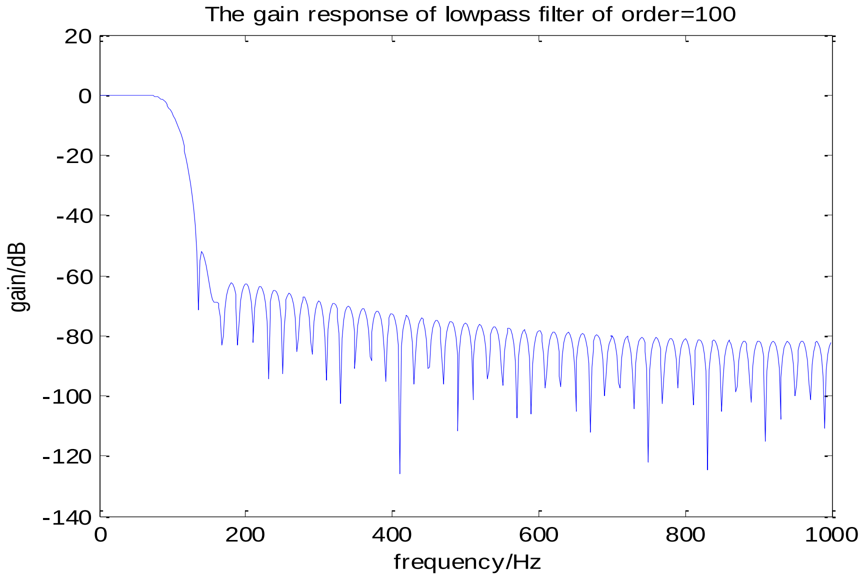

Fig. 5 Gain response of lowpass filter of order=100

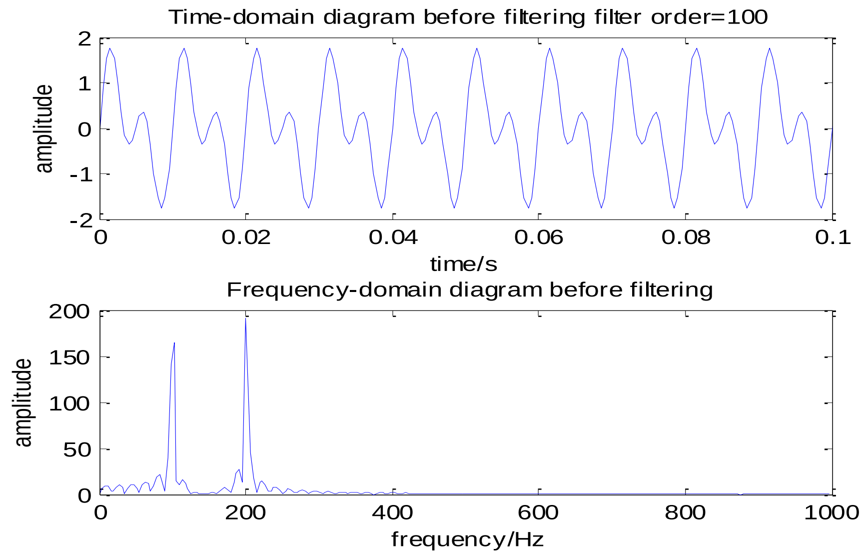

Fig. 6 Time and frequency domain before filtering

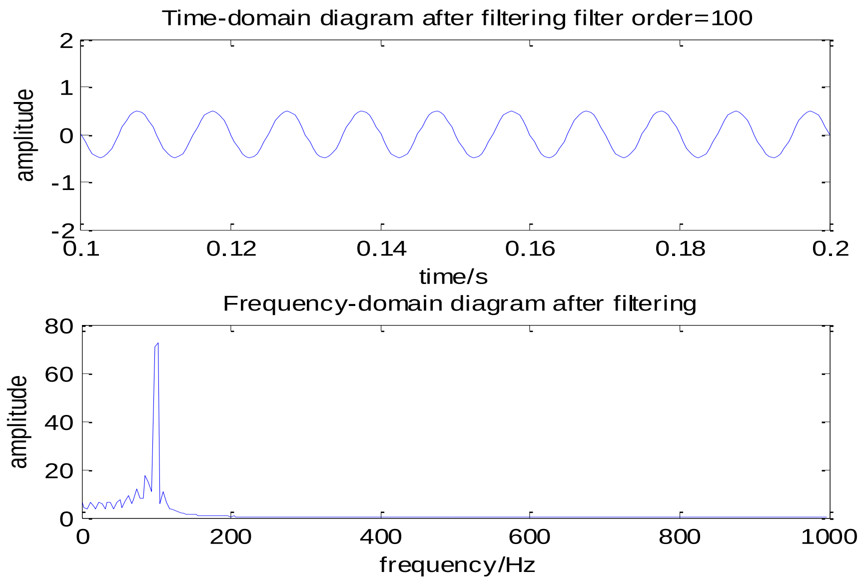

Fig. 7 Time and frequency domain after filtering

Comparing Fig. 6 and 7, it is easy to see that the input signal is made up of two superposition signals with different frequencies. The passband is 0-100 Hz, and stopband starts from 170Hz, and there are two frequencies of 100Hz and 200Hz needing to be filtered. The signal with frequency of 100Hz which is in the range of passband is kept, while the signal with frequency of 200Hz which is in the range of stopband is filtered.

The minimum stopband attenuation values for hamming window are 78dB, which meets the requirements.

A matlab program was built and tested to evaluate digital filter parameters taking in consideration the following:

- Two sines frequencies (100 and 200 Hz) where used as an inputs to the filter.
- The filter has to keep 100 Hz signal and to filter the 200 Hz signal.
- The program accepts: sampling frequency, transition bandwidth, window length and filter order as input parameters.
- The input parameters where used to filter the signal and calculate the stopband attenuation.

The results of program running are shown in Table 1.

IV. RESULTS AND DISCUSSIONS

Referring to results shown in Table 1 we can conclude the following facts and recommendations:

- The successful operation of the digital lowpass filters to remove the unwanted frequency highly depends on the selected values of transition width, window length and filter order.
- Transition band length must be small and less than 1.
- Increasing transition band width toward 1 leads to increase stop band attenuation, while decreasing it value leads to decreasing stop band attenuation.
- Window length and filter order are related to transition width, sampling frequency and the frequency of the signal to be kept.
- Choosing random values to filter parameter may leads to filter failing by keeping the unwanted frequency.
- Experimentally we estimated FIR lowpass filter parameters by using regression and the following equations obtained:

  m=(c*f1)/(fs/2), M=round (8/m) and N=M-1

  Where **m** is the transition bandwidth, **c** small value coefficient less than 1, **f1** frequency to keep after filtering, **fs** sampling frequency, **M** window length and **N** filter order.

These equations were tested using various values for f1, fs, and c. The calculation and simulation results show the efficient use of the digital filter by means of filtering the unwanted frequencies and controlling the stop band attenuation.

Table 1: Experimental results

Transition bandwidth(m)	Window length(M)	Filter order(N)	Amplitude of the signal to be kept	Amplitude of the signal to be filtered	Remarks
0.1	10	10	145	80	Failed to remove 200 Hz signal
0.1	20	10	148	80	Failed to remove 200 Hz signal
0.1	10	20	98	14	Failed to remove 200 Hz signal
0.1	10	30	83	0	succeeded
0.1	10	40	80	0	succeeded
0.1	10	50	80	0	succeeded
0.1	10	60	75	0	succeeded
0.1	10	100	73	0	succeeded
0.1	20	100	78	0	succeeded
0.1	20	120	76	0	succeeded
0.1	20	140	73	0	succeeded
0.01	20	140	73	0	succeeded
0.001	20	140	73	0	succeeded
0.1	140	140	73	0	succeeded
0.03	267	266	72	0	succeeded
0.001	7999	8000	0.13	0	succeeded
0.002	3999	4000	0.28	0	succeeded
0.004	1999	2000	1	0	succeeded
0.5	15	16	120	35	Failed to remove 200 Hz signal
0.5	30	31	82	0	succeeded
0.5	40	41	78	0	succeeded
0.5	50	51	78	0	succeeded
0.5	80	81	78	0	succeeded

V. CONCLUSIONS

A digital lowpass filter was implemented and a relationship between the input parameters(sampling frequency, frequency to be kept , and transtion band width) and the calculated filter parameters(window length and filter order) was formed. The equations obtained are used to estimate the FIR digital lowpass filter parameters. As a result, the performnace chractaristics of the FIR filters has been improved.

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