

## Modeling and Simulation of Digital Filter

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**Abstract.** Digital filter is a basic algorithm to speech, image processing, pattern recognition, spectrum analysis and so on. It plays a very important role in digital signal processing. At present, many designers use software tools such as Matlab to model and simulate the digital filter. This paper described the design and simulating process of a FIR low pass filter. The Equivalent ripple approximation algorithm is used for the design. Instead of using codes in command or m-file, the FDA tool and Simulink are adopted to complete the test system. Filtering processing for tested signal shows that the design method is convenient for engineering and education.

### Introduction

Digital filter is now becoming an import engineering and education subject. Digital filters are a very important part of digital signal processing [1]. In fact, their extraordinary performance is one of the main reasons that DSP has become so popular. Digital filters are used for two general purposes: separation of signals that have been combined, and restoration of signals that have been distorted in some way. Analog filters can be used for the same purposes; however, digital filters can achieve far superior results. The digital filters are usually divided into two types: IIR and FIR [2]. The FIR filter is widely used since it has the advantages of ensure linear phase, and is easily to implement [3]. Presently, the close approximation to ideal frequency response characteristics is what most methods strive for. The common design methods of FIR digital filter is mainly based on the window function, the frequency sampling and Equivalent ripple approximation (Remez exchange algorithm). These methods can be implemented by virtue of Matlab. The traditional method was using the input command in the command window of Matlab [4]. In the course of digital signal processing, the theory and conclusions are mostly derived from the mathematical derivation, the more abstract and difficult to understand. In the course of digital signal processing, the theory and conclusions are mostly derived from the mathematical derivation, the more abstract and difficult to understand. The experimental teaching is commonly used the MATLAB software. The processing procedure can take advantage of the FDA Tool and Simulink of Matlab. This method is more direct and convenient and is effectively for teaching and engineering. The DSP Builder component is convenient to realize the traditional simulation, and it also can generate HDL code [5]. It has been used in the course teaching.

### Equivalent Ripple Approximation

According to the window function and the frequency sampling methods, the approximation error in the whole frequency distribution is not uniform. If the design index is just achieved at the band with the most error, the index at the band with the least error will be far better than the design commands. And then the resources will be wasted. However, The Equivalent ripple approximation (Remez exchange algorithm) has minimized the maximum, and the approximation error in the whole frequency distribution is uniform. This design method makes the filter order minimum with the same index; or makes the most approximation minimum with the same order [6].

If  $H_d(\omega)$  is used to denote the amplitude characteristic wished to approximate, and is used to denote the amplitude characteristic of the practical designed FIR filter, the weighted error function is that:

$$E(\omega) = W(\omega)[H_d(\omega) - H_g(\omega)] \quad (1)$$

Where  $W(\omega)$  is the weighted function, and it is used to control the approximation precision of different frequency bands.

The basic idea of Remez exchange algorithm can be summarized as: calculate the filter coefficients by the principle which minimize the most approximation error of  $E(\omega)$ . The solution of this algorithm is complicated. Fortunately, it can be simplified with the help of Matlab tool.

### Filter Design

The coefficients of Remez exchange algorithm are given as:

Cut-off frequency of pass band:  $\omega_p$

Maximum attenuation of pass band:  $\alpha_p$

Cut-off frequency of stop band:  $\omega_s$

Minimum attenuation of stop band:  $\alpha_s$

Generally, the ripple amplitude  $\delta_1$  and  $\delta_2$  are also acquired. They can be obtained by  $\alpha_p$  and  $\alpha_s$  :

$$\alpha_p = -20 \lg \frac{1 - \delta_1}{1 + \delta_1} = 20 \lg \frac{1 + \delta_1}{1 - \delta_1}, \quad \alpha_s = -20 \lg \frac{\delta_2}{1 + \delta_1} \approx 20 \lg \delta_2 \quad (2)$$

then

$$\delta_1 = \frac{10^{\alpha_p/20} - 1}{10^{\alpha_p/20} + 1}, \quad \delta_2 = 10^{-\alpha_s/20} \quad (3)$$

The low pass FIR filter is designed for instance. The design coefficients were given as:  $\omega_p=200\text{Hz}$ ,  $\omega_s=400\text{Hz}$ ,  $\alpha_p=3\text{dB}$ ,  $\alpha_s=10\text{dB}$ . Then the oscillation ripple amplitude coefficients were calculated as:  $\delta_1=0.171$ ,  $\delta_2=0.316$ .

The amplitude and phase frequency characteristic of this low pass filter is shown in figure 1.

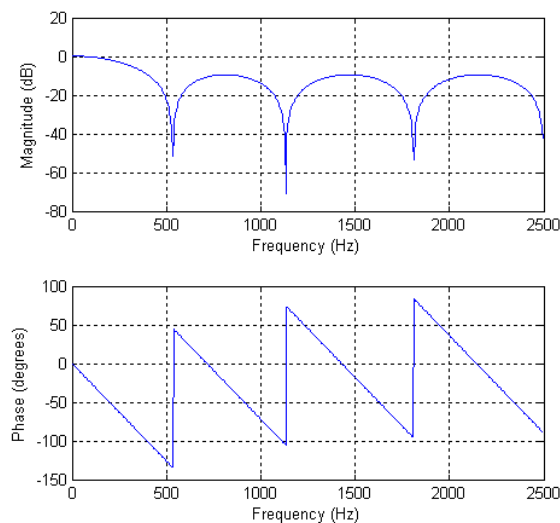


Fig.1. The frequency characteristic of the low pass filter

The traditional design method is used m-file or command window with the help of codes. The FDA tool is interactive graphic signal processing in Matlab, and it is convenient to implement the filter design and response analysis. Simulink is another powerful tool. The whole test circuit can be constructed and simulated in Simulink environment.

The same low pass filter designed by FDA tool and Simulink follow these main steps:

- (1) Set the response type, design method, filter order and filter coefficients.
- (2) Run “design filter”, the response of this designed filter can be obtained.

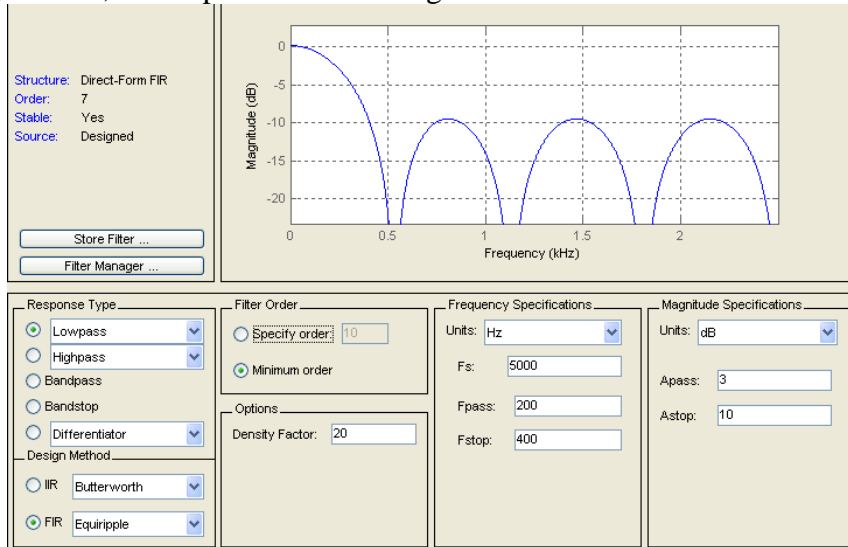


Fig.2. The designed result interface

(3) Export the designed filter by FDA tool to Simulink model, and complete the whole test circuit model. The sweep sine was set as original input signals. The model was shown in figure 3.

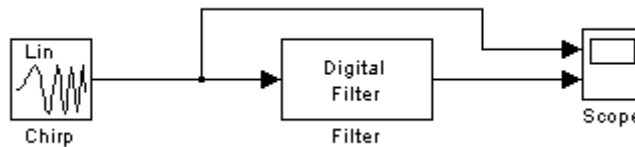


Fig.3. The whole test circuit

(4) Run the simulation and obtain the test result, and the result was shown in figure 4.

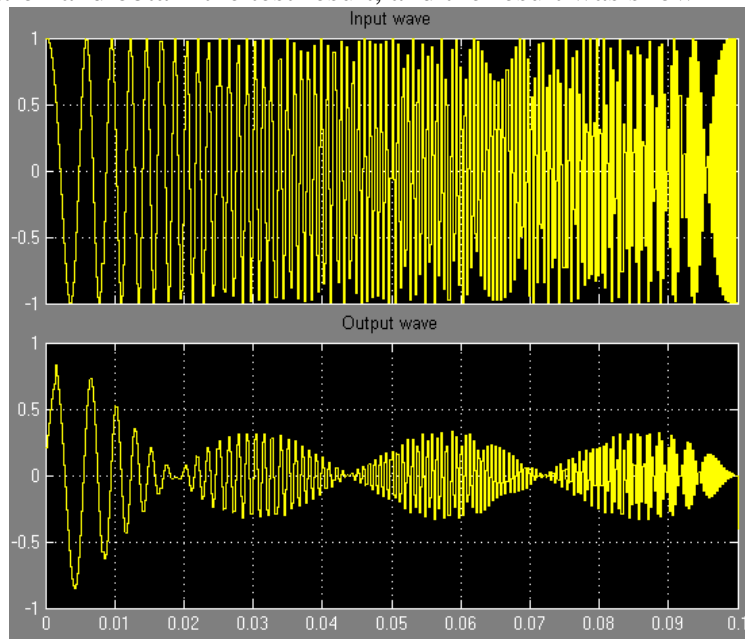


Fig.4. The simulation result

The above process is still far from the hardware implementation. The digital filter by FPGA is usually achieved by code. This is difficult for the beginners, and the teachers also feel more laborious. The DSP Builder tool can simplify the process without HDL code. Figure 5 presented the three order FIR filter constructed by DSP Builder components.

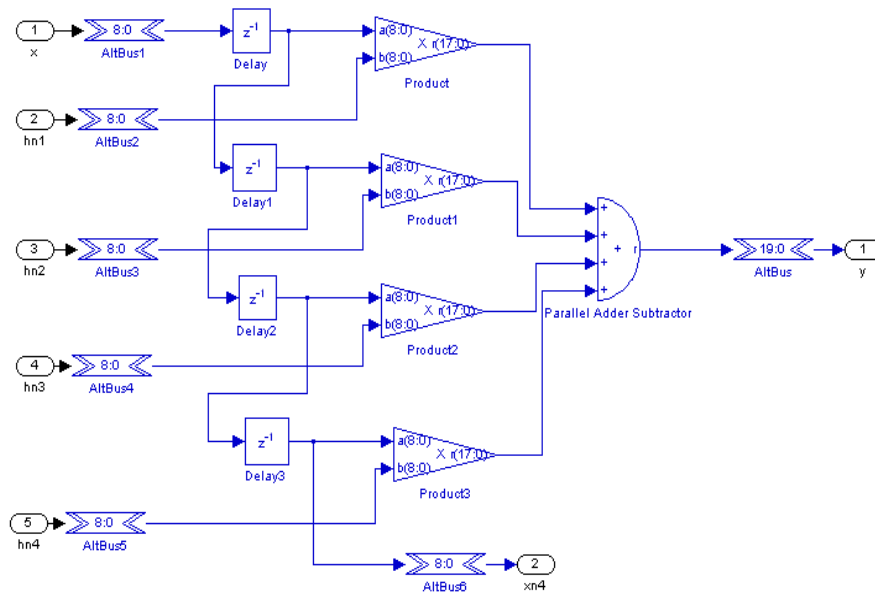


Fig.5. Three order filter

Then two above filters constructed the seven order filter. It was shown in Figure 6.

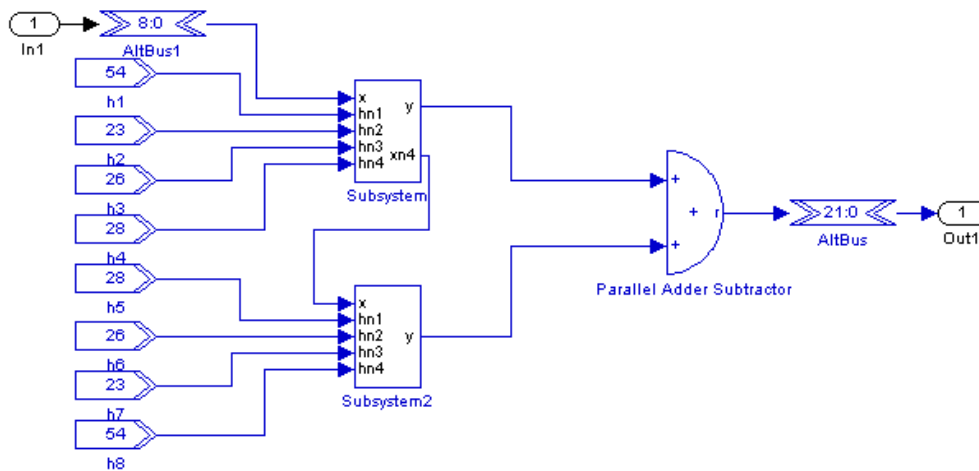


Fig.6. Seven order filter

The whole test model was shown in Figure 7, and the test result was shown in Figure 8.

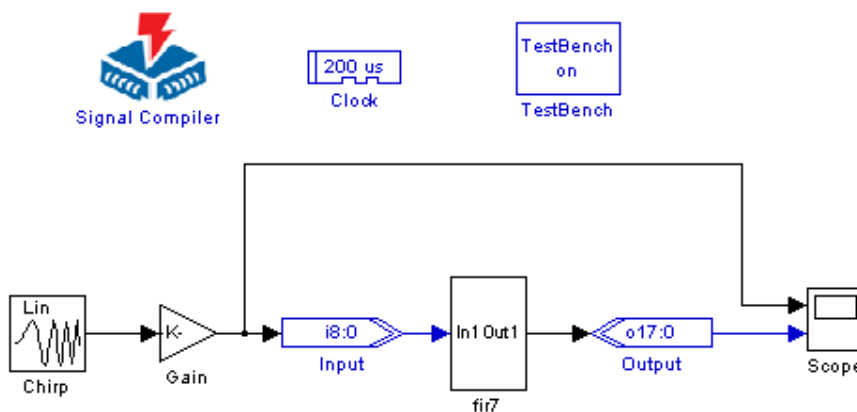


Fig.7. The whole test model

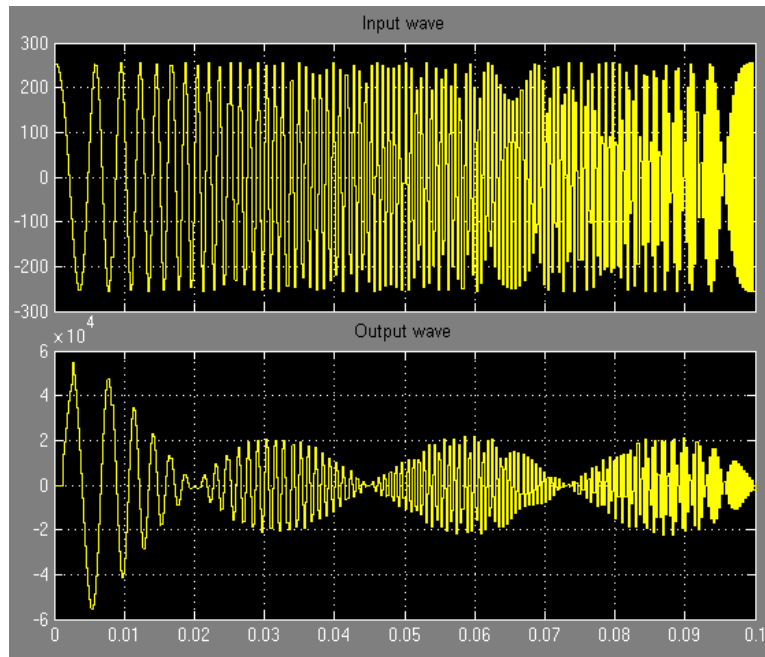


Fig.8. The test result

The upper half part of Figure 8 shows the sweep signal, and the lower half shows the processed signal through the digital filter. This result was same as the Figure 4. This means the filter model constructed by DSP Builder components achieve the same effect. Then the model can generate HDL code and the Quartus II project. It also can be simulated in Modelsim by virtue of DSP Builder. The Modelsim simulation result was shown in Figure 9.

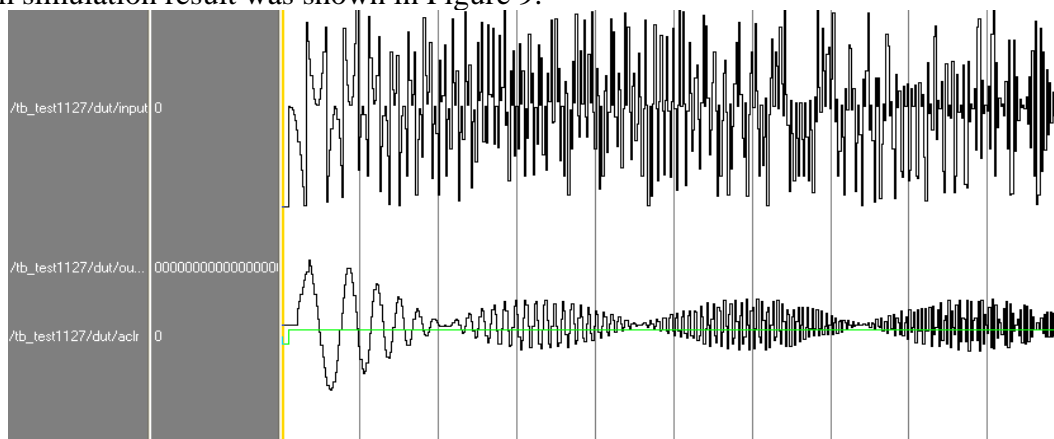


Fig.9. The Modelsim test result

## Conclusion

This paper has described the implementation of digital filter design and simulation of a toolkit which is capable of converting a code based design pattern into a graphic and model tool. The commonly used low pass filter is designed by FDA tool and the whole simulation system is tested by Simulink. The traditional deigned pattern is using codes in command or m-file. The design and test method based on FDA tool and Simulink become more and more popular in engineering and education at colleges and universities. In addition, the design can be developed to the hardware by the Simulink and DSP Builder. This processing procedure avoids the code writing, and contacts several courses.

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## References

- [1] Liu Huiheng, Tian Ling. Reform and practice of teaching methods to Digital Signal Processing [J]. Research in Teaching, 2008, 31(3): 237-239.
- [2] Liang Yingbo, Zhang Lihong. Simulation and implementation of FIR digital filter using window function [J]. Journal of Zhoukou Normal University, 2013, 30(5): 51-53.
- [3] Guo Sumin, Guo Sufan, Wu Bo. Exploration on the Teaching Reform and Implementation of Digital Signal Processing Course [J]. Research in Teaching, 2006, 29(5): 453-455.
- [4] Xie Liming, Zheng Rui. Matlab-based Design of IIR Digital Filter [J]. Machinery & Electronics 2011, (1): 28-30.
- [5] Lu Yizhe, Meng Meihai, Zhang Feng, Zhang Ke. FPGA DDS Design Based on DSP Builder [J]. Journal of Detection & Control, 2011, 33(6): 42-45.
- [6] Chen Minge, Jiao Zhanya, An Xiaogang. A method of designing of equiripple linear-phase FIR filter [J]. Journal of Xi'an University of Science and Technology , 2008, 28(1): 199-202.