Digital Filter Design Based on MATLAB Environment

Yang LIU[1], Chenghao SONG[1]

[1](School of Information Science & Electric Engineering, Shandong Jiaotong University, China)

Abstract: Traditional digital filter design process complex and calculation workload big, filtering features adjustment difficulties, once the design is complete, can't easily modify filtering characteristic curve, is inconvenient, thus greatly limits the development and application of digital filter. This paper mainly introduces the use of the MATLAB Signal Processing Toolbox (Signal Processing Toolbox) rapid effective design by software of conventional design method of digital filter, using MATLAB language to program design and interface design by means of Signal Processing Toolbox FDA Tool tools detailed steps. Contrast can be realized at any time using the MATLAB design filter design request and the filter characteristic adjustment parameter, so as to make improvements and greatly reduce the workload, is advantageous to the optimization of filter design. Finally, by calling the function modules in the Simulink simulation block diagram of digital filter can double-click on each function module in the simulation process, change the parameters at any time, to get the simulation results under different condition.

Keywords: Digital Filter, Finite Impulse Response (FIR), Infinite Impulse Response (IIR), MATLAB

I. INTRODUCTION

The original filter is made up of passive components such as resistors, inductors and capacitors [1][2]. It is called a passive filter. Since the integration technology has emerged, the prospect of active filters has become broad. With the development of science and technology, large-scale integration of electronic systems is becoming more and more important. Since the filter is used most in the electronic system, that is to say, the performance of the filter determines the quality of the electronic system, the passive filter, the mechanical filter and the RC active filter cannot achieve comprehensive large-scale integration. So, people began to explore new paths. As early as the 1950s, a new concept was proposed: switched capacitor filters, but they were not taken seriously [3]. Until the 1970s, a scientist in the United States experimentally changed this concept into an experimental one. In reality, he replaced the resistor R with a switch and a capacitor, so he called it a switched capacitor filter. In his experiment, he came to the conclusion that the performance of the switched capacitor filter is only related to the ratio of the capacitor and is independent of the absolute value of the capacitor. In this way, people began to pay attention to this new and significant thing[4].

Since the 1960s, due to the development of computer technology, integrated technology and materials industry, the development of filters has reached a new level, towards low power consumption, high precision, small size, multi-function, stable and reliable, and low price[5]. Efforts to develop, among which high precision, small size, multi-function, stability and reliability have become the main direction after the 1970s, leading to the rapid development of various filters such as digital filters, RC active filters, switched capacitor filters and charge transfer devices. By the late 1970s, the monolithic integration of the above-mentioned filters had been developed and applied. From the 1990s to the present, the main focus was on applying various types of filters to the development and development of various products. Of course, the filters the research itself is still going on[6].

From the formal point of view, digital filtering has linear filtering and nonlinear filtering. Linear filtering refers to convolution filtering, which is divided into frequency domain filtering and time domain filtering. It is divided into recursive filtering and non-recursive filtering according to the filtering method in the real domain. Nonlinear filtering mainly refers to homomorphic filtering, which linearizes nonlinear problems by using logarithmic methods. Due to the development of electronic computer technology and large-scale integrated circuits, digital filters can be implemented by computer software, or can be realized in real time by large-scale integrated digital hardware. When digital signals are used to process analog signals, the input analog signals must first be band-limited and sampled. And analog-to-digital conversion, in order to obtain analog signals, the digital signal processed by the digital filter must be digitally, analog-converted, and smoothed. The digital filter has high precision, high reliability, programmable change characteristics or multiplexing, and easy integration. Advantages: Compared with traditional analog filters implemented by hardware circuits, digital filters have several advantages: (1) simplifying the design of hardware circuits, improving the integration and reliability of hardware circuits; (2) interference The signal suppression ability is obviously improved, which promotes the control precision and stability of the system; (3) The parameter adjustment of the digital filter is more convenient and flexible than the analog filter.

II.   DESIGN AND IMPLEMENTATION OF DIGITAL FILTER

A. Program design of IIR filter

IIR low-pass filter: the pass band cutoff frequency is 500Hz, the maximum attenuation in the pass band is 0.5, the pass band ripple is 0.5, the minimum attenuation in the stop band is 60, and the sampling frequency is 8 KHz;

IIR band pass filter: the width of the pass band is 500Hz, the cutoff frequency of the stop band is 500Hz and 1000Hz, the maximum attenuation in the pass band is 0.5, the minimum attenuation in the stop band is 60, and the sampling frequency is 8 KHz. The choice of filter order occupies a very important position and role in the actual design of the whole filter. It affects the processing speed and the complexity of implementation. Therefore, in the design and use of the actual filter, we often face the choice of order. The problem needs to be based on the actual selection of the appropriate filter order. In MATLAB, the order selection function is provided for different types of filters. The selection functions for the Chebyshev I-type filter order are as follows.

[N, wpo]=cheb1ord(wp,ws,rp,rs), this function returns the minimum order of the digital filter N and the Chebyshev type I filter passband cutoff frequency wpo. The design requirement is that the attenuation in the passband does not exceed rp, the attenuation in the stopband is not lower than rs, and the cutoff frequencies of the passband and the stopband are wp and ws, respectively, and the unit is rad.

[N,wn]=cheb1ord(wp,ws,rp,rs,'s'), this function returns the minimum order N of the analog filter and the frequency wn inherent to the Chebyshev I-type filter, wp, ws is Rad/s is the unit [10].

Processing speed and implementation complexity are very important to the design of the system. The higher the order of the filter, the better the processing effect and the more complicated the implementation. Therefore, the IIR low-pass filter in this paper will first select the design to meet the design. The minimum order of the target is 3, and the program runs as follows.
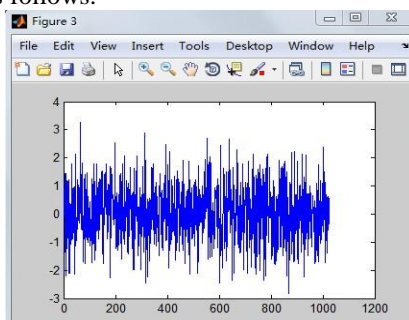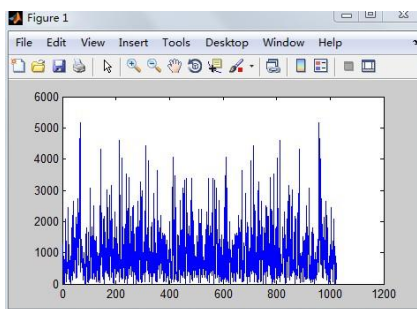

Fig. 1 The random signal

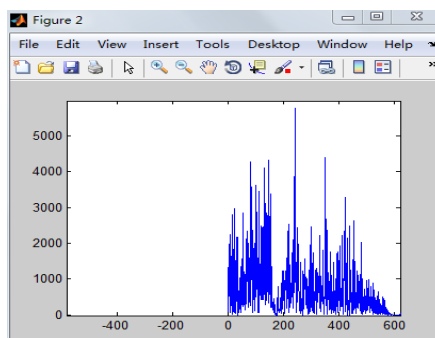Fig. 2 Before filtering the signal frequency spectrum

Fig. 3 After filtering the signal frequency spectrum

B. IIR type filter interface implementation

First design the filter according to the requirements. The interface after starting the FDATool is the filter design interface. To enter the filter design interface from other interfaces, click the last icon in the list of tool buttons on the lower left side of the FDATool window (filter design), you can enter the design interface, in which the filter design is to be done, first select the type of filter, the type of filter has high pass, low pass, band pass and band stop, and many other types. After selecting the type, you should choose the design method. The design method can be divided into two categories from the big category: one is FIR and the other is IIR. There are many different methods in each category, for different design methods. There are different options, then the filter order. If it is an 11th-order filter, you can enter 10, you can also choose the minimum order. After setting these, you need to set the frequency characteristics of the filter. Amplitude characteristics, where Fs is the sampling frequency, Fp is the pass band cutoff frequency, Fs is the stop band cutoff frequency, and the latter (Apass) is the maximum allowable attenuation in the pass band (Astop) is the minimum allowable attenuation in the stop band. After setting all the parameters, press (Design Filter) to view the design results. The interface shown below is the design interface and the running result - phase amplitude characteristics.
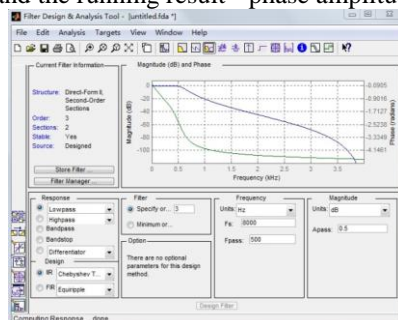

Fig. 4 Low-pass filter run results interface

The interface design of the bandpass filter is the same as that of the lowpass filter. Here, the parameters of the interface are modified. The following is the operation result interface of the bandpass filter.
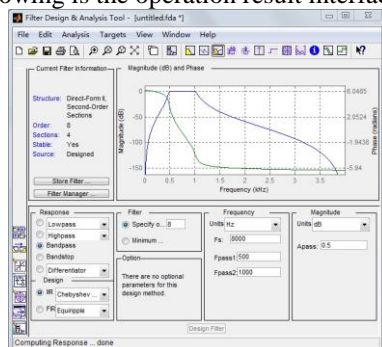

Fig. 5 Band-pass filter run results interface

The results of the operation shown in the figure show the amplitude and phase response of the IIR band pass filter. The above is the result of the interface design of the IIR filter through the FDATool tool. It is very convenient to design and analyze the filter using the FDATool tool. The programming can be programmed. Direct simulation analysis, FDATool has no way to simulate but can save the designed filter, through the Realize Model in the menu bar to import the filter into Simulink for simulation, in general, it is very convenient to design the filter through FDATool.

III. CONCLUSION

The content of this research is to use MATLAB Signal Processing Toolbox to quickly and effectively design the design method of conventional digital filter composed of software. It is the FDATool tool which uses MATLAB language for programming and uses signal processing toolbox. The interface design is carried out. Finally, the system simulation is performed by adjusting the simulation block diagram of the digital filter by adjusting the function modules in Simulink. In MATLAB, the design of various filters has a corresponding function to calculate the amplitude response. Write the program according to the index parameters and various functions. Run the program in MATLAB environment to get the filter amplitude and phase response. Equal curve and filter coefficients. The FDATool interface is divided into two parts, one part is Design Filter, the

lower part of the interface is used to set the filter design parameters, and the other part is the characteristic area. In the upper part of the interface, the filter is used to display each filter. Characteristics. In this paper, the function block of Simulink is used to construct the simulation block diagram of the digital filter. In the simulation process, you can double-click each function module to change the parameters at any time to obtain the simulation results in different states.

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