Design of FIR Low-pass Filter Based on MATLAB

Weiwei Tian [a], Zixuan Ni [b,]*
Lanzhou University, Lanzhou 730107, China;
[a]weiweitian96@163.com, [b]nzx18069832761@163.com
Corresponding author: Zixuan Ni

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Abstract. With the rapid development of the global electronic market, the filter has been greatly promoted. And the development of digital technology also create a good technical foundation for the filter. This paper introduces the basic concepts of digital filters, the brief introduction of MATLAB Software, and the design method of FIR low-pass filter. Taking MATLAB as working platform and development tools, the design and simulation of FIR low-pass filter are realized by MATLAB program. And the practicability of low-pass filter for audio processing is proved by tests.

1. Introduction

Digital signal processing technology has entered the mature stage at present. In the process of digital signal processing, digital filtering is a very important part. The digital filtering converts the digital signal through some arithmetic logic, and eliminates some frequencies or improves the relative occupancy ratio of some frequencies at the same time, so as to eliminate the interference frequency [1]. Compared with traditional filtering, digital filter has great advantages both in stability and flexibility. Therefore, the comprehensive analysis of digital filters is of great significance for the promotion and application of digital signal technology. In this paper, FIR low-pass filter designed by MATLAB software is used to realize low-pass filtering of audio files and the high frequency is filtered to get the bass part finally.

2. Overview of Digital Filter

Digital filter is an algorithm or device made up of digital multiplier, adder and delay unit. The function of digital filter is to handle digital code of the discrete signal input, in order to change the signal spectrum. There are many kinds of filters. Functionally, filters can be divided into low pass, high pass, band stop, and band pass. In terms of the unit impulse response, the digital filter can be divided into finite impulse response filter and infinite impulse response filter. In the process of using the filter, it is necessary to make a comprehensive analysis of the two filters before deciding what kind of filters to use. In general, the FIR filter achieves the goal of filtering through the superposition algorithm, so there is no ready-made formula for it to use in the design process. And the IIR filter can complete the design work only through the analog filter design parameters, but in the flexibility, the IIR filter has the obvious inferiority compared with the FIR filter.

3. Brief Introduction of MATLAB Software

MATLAB language is an engineering application software which can integrate scientific computing, data visualization and program design. It has become the essential basic software for the subject of computer to have aided engineering analysis, design, simulation as well as teaching, which is composed of MATLAB main package, simulink module and toolbox with various functions [2]. MATLAB has efficient numerical computation and symbolic computation function which can relieve users from mathematical analysis in complex. And it has a complete graphics processing function which can realize the visualization results and programming. Its rich function (such as signal processing toolbox, communication toolbox, etc.), provides a number of convenient and practical tools for users.

4. Design of The FIR low-pass Filter

4.1 The Design Method of FIR Low-pass Filter

Usually MATLAB is used to design and simulate the FIR filter coefficients, the usual way is optimization design, FDA TOOL design and window function design [3]. For the method of optimization design, Park-Mc-Clellan algorithm can be used to design linear phase FIR filter which makes the maximal error between actual frequency response and the expected response frequency minimum, but this design method cannot directly set the filter resistance band attenuation parameter. FDA Tool (filter design & analysis tool) is a tool specifically for the design and analysis of filter in the MATLAB toolbox which provides an interactive design environment for the entire process. Users design almost all types of filters by setting amplitudes and zeros and poles, including type, order, frequency and other parameters. The window function method is to set the unit impulse response sequence h(n) by adding a window according to the given frequency response feature. The window functions commonly used in the project include rectangular Windows, hanning Windows, hamming Windows, cassett Windows, blackman Windows and bartlett Windows and so on. The function commands based on fir1 and fir2 can easily get the filter coefficients.

4.2 Design and analysis of FIR low-pass filter based on MATLAB

The fast algorithm of FFT discrete Fourier transform in MATLAB can transform a signal into the frequency domain. After FFT, N sample points can get the FFT results of N points. In order to facilitate FFT operations, N usually takes integer power of 2. Assuming that the sampling frequency is F_s, the signal frequency F and the sampling point is N, so the result of FFT is a complex number of N points. Each point corresponds to a frequency point. The module value of this point is the amplitude characteristic of the frequency. For example, the frequency of a point n is: $F_n = \frac{(n-1)\cdot F_s}{N}$. If the sampling frequency is 1024Hz, the sampling point is 1024, then it can be identified by 1Hz. In other words, sampling the signal for a second time and conducting FFT, the result can be analyzed to 1Hz and if the sample is sampled for two seconds and the conducting FFT, the result can be analyzed to 0.5 Hz. If you want to increase the frequency resolution, you have to increase the number of samples, which is the sampling time. Frequency resolution and sampling time are reciprocal. The most common FIR implementation architecture is called the direct-form FIR [4].

Low-pass filter is to take low-frequency component out and abandon the high frequency part of the digital signal, and the band-pass filter is take digital signal in a certain range of frequencies. So the band-pass filter after the translation is a low-pass filter. The structure of the low-pass filter is shown in figure 1.

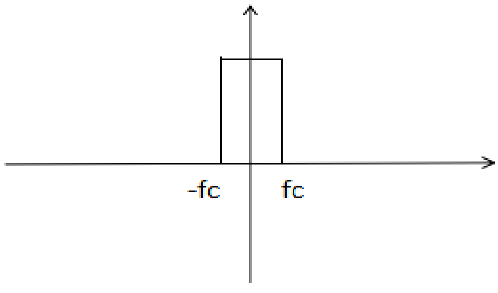

Figure. 1 Structure of the low-pass filter

Its function expression is:

$$h_0(f) = U(f + f_c) - U(f - f_c) \tag{1}$$

And the low-pass filter is shifted by the band-pass filter. Its function expression is:

$$h(f) = U(f + f_c - f_1) - U(f - f_c - f_1) + U(f + f_c + f_1) - U(f - f_c + f_1) \tag{2}$$

The Discrete Fourier Transform of the input digital signal is:

$$X(f)=\sum_{n=N}x(n)e^{-j2\pi nf} \tag{3}$$

The Fourier transform of the output signal is:

$$Y(f)=X(f)H(f)=\sum_{n=N}x(n)e^{-j2\pi nf}\cdot[U(f+f_c-f_1)-U(f-f_c-f_1)+U(f+f_c+f_1)-U(f-f_c+f_1)] \tag{4}$$

Take the inverse Fourier Transform and we can get output signal:

$$y(n)=\sum_{n=N}\frac{2x(n)\sin(2\pi(t-n)f_c)\cos(2\pi(t-n)f_1)}{\pi(t-n)} \tag{5}$$

This is the output of the pass-through filter.

4.3 The filtering effect of audio signal

Because f_c and f_1 are normalized, the period of frequency is 1, so when the frequency is greater than 0.5, it is a high frequency component, whereas less than 0.5 is the low frequency part. For a low-pass filter, the sum of f_c and f_1 is less than 0.5. Taking a signal, the sample frequency of which is 11400hz. First, the spectrum and digital signal of the experiment audio are obtained which were shown in figure 2.

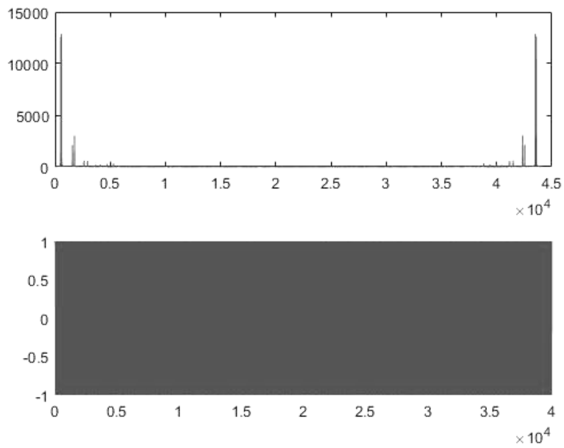

Figure. 2 The digital signal and its spectrum

Then obtained the following results after pass the designed low-pass filter in figure 3 and figure 4. The analysis of the audio shows that the lower part of the audio is retained and higher part is removed through the low-pass filter.

(f_c is 1109/88200Hz, f_1 is 1091/88200Hz)

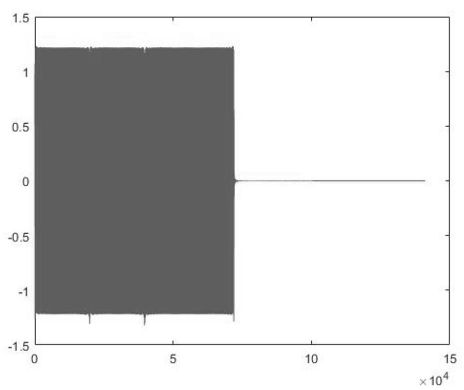

Figure. 3 Audio data

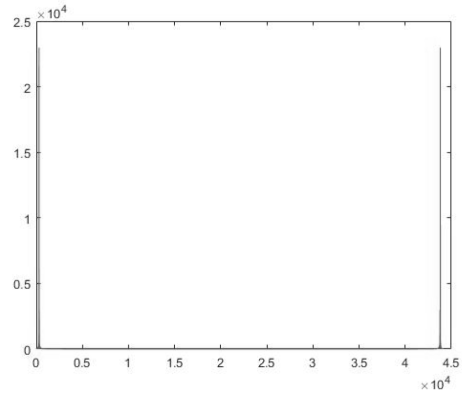

Figure. 4 Spectrum

And then obtained the following results after pass the designed band-pass filter in figure 5 and figure 6. The analysis of the audio shows that the audio becomes thinner and deeper than before, as the audio through the band-pass filter loses a portion of the frequency.

(f_c is 1109/88200Hz, f_1 is 10091/88200Hz)

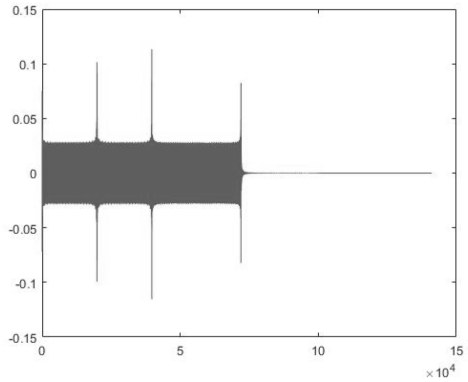

Figure. 5 Audio data

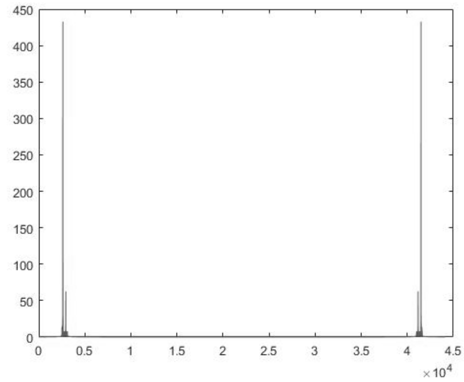

Figure. 6 Spectrum

5. Conclusion

MATLAB is a large and powerful software that is suitable for multi-disciplinary, multi-working platform. It has many functions include mathematical statistics, digital signal processing, time series analysis, dynamic system simulation and so on. In this paper, the low-pass filter designed by MATLAB can be used for signal selection. It can be used to select the signal according to the frequency, to avoid interference of high-frequency signals, and it also can be used in instruments, acoustics, signal processing and other circuits.

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