ORIGINAL ARTICLE

Application of Digital Filtering Technology in Random Vibration Signal Processing

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Abstract: This paper discusses the signal processing methods of various filtering technologies in signal processing, and it introduces the design methods of several kinds of low-pass digital filters, such as IIR digital low pass filter and FIR digital low pass filter. Some digital filtering techniques are introduced, such as median filtering, recursive average filtering, and arithmetic average filtering.

Keywords: Signal Processing; Digital Filtering Technology; Filter Design

1. Introduction

Noise is common in signal generation, and too small signal noise ratio will cause signal distortion. Therefore, it is particularly important for noise processing, and digital filtering technology plays an important role in signal processing. With the development of signal processing, digital filtering technology is more and more widely used. Digital filtering technology plays a significant role in track detection[1], medical image denoising[2], construction engineering detection[3] and other aspects. It has lots of advantages, including suppressing noise, extracting useful information, and making the collected data conform to the real situation. Filters can be divided into classical filters and modern filters. Classical filters can be divided into low pass filter (LP), high pass filter (HP), band pass filter (BP) and band stop filter (BS)[4]. The basic purpose of filter is to remove noise, but the purpose of filter is different. The choice of filter depends on the characteristics of the signal and the required part of the frequency as well as the required specific properties.

This article first introduces several digital filtering techniques, and then it introduces in detail the design methods of IIR digital low pass filter and FIR digital low pass filter and their characteristics.

2. Digital filtering technology

2.1 Arithmetic mean filtering method

Digital filtering technology refers to the way of filtering out the noise contained in the collected data in software. The data is processed to make the it conform to the real situation.

The arithmetic mean filtering method first takes data for N times continuously and it performs arithmetic mean operation on the sampling value. The larger the number of sampling times, the smoother the signal and the lower the sensitivity. This method is suitable for filtering random interference signals. The resulting signal is an average with a small range of fluctuation. This method is not suitable for slower parameters.

The filter using this filtering method is called the average filter, and its transfer equation is:

$$H(z)=\frac{1}{N}\sum_{n=0}^{N-1} z^{-n}=\frac{1}{N}\frac{1-z^{-N}}{1-z^{-1}} \tag{1}$$

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2.2 Median filtering method

In this method, N data are continuously sampled, and then the N sampling values are arranged according to the size. Finally, the median value is taken as the effective value of this measurement. This method can overcome the pulse interference caused by accidental circumstances, and it is suitable for the measured parameters which change slowly, but not for the parameters which change quickly.

2.3 Median average filtering method

This method takes N consecutive samples without the minimum and maximum values, and then it calculates the average value of the remaining data. This method combines two kinds of arithmetic mean filtering method and median filtering method, which can effectively eliminate the influence brought by random pulse interference. The disadvantage is that the speed is slow and it is not suitable for the parameter with fast changing speed.

2.4 Recursive average filtering method

This method is suitable for the system with a slow sampling speed. N measured data are taken as a queue with a fixed length of N. During each sampling, the measured data enter the end of the queue in accordance with the first-in, first-out principle. The method can suppress periodic interference and it is suitable for high frequency oscillation system. However, this method has low sensitivity and it is not suitable for dealing with errors caused by random burst pulses.

2.5 Limiting filtering method

The method is detecting the amplitude of the measured signal through the program, so as to eliminate the burst pulse signal interference. First, the maximum allowable deviation value for each two sampling times is determined, and a judgment is made each time the data is detected. If the difference between the detected value and the original value is greater, the value is invalid, and it continues to use the last sampling value. The method can effectively filter out the random burst pulses. However, it is not applicable to filter out periodic interference, but for measuring situations where the change is relatively slow.

3. Design methods and characteristics of some low-pass digital filters

3.1 Design principle of digital filter

Filter is an important device to realize filtering technology. Filter should be designed and decided according to the actual situation. For example, for low pass filter, sampling response h(n) is a descending sine function, it goes from minus infinity to plus infinity, instead of just having a unit impulse at n equals 0, which is not possible in the physical, so the design of the filter is in certain conditions of ideal filter approximation unceasingly. When the mutation frequency response in ideal state needed, physically need to add a transition zone, the frequency response to realize the transition from one frequency band to another frequency band. As a result, even the bands that should be blocked will have a smaller allowable range.

The design of IIR low-pass digital filter and FIR low-pass digital filter can be roughly divided into three steps:

(1) Provide technical indexes α and ω etc., required by the filter;

(2) Design a transfer function H(z) that approximates the required index;

3. Concrete implementation of H(z)

3.2 IIR low-pass digital filter design method

Through the application of low pass filtering method[5] made low pass filter, and the design of the digital filter based on G(s) of analog filter design, the technical indicators ω, alpha through $\omega=\Omega T_s$ to calculate the corresponding technical indicators Ω analog filter, according to the linear relationship between ω and Ω available IIR transfer equation.

The transfer equation of IIR digital filter is:

$$H(z) = \frac{\sum_{r=0}^{M} b_r z^{-r}}{1+\sum_{k=1}^{N} a_k z^{-k}} \tag{2}$$

Let g(s) be the unit impulse response of G(s), and the corresponding digital unit response is:

$$h(nT_s) = g(t)|_{t=nT_s} = g(t)\sum_{n=0}^{\infty}\delta(t-nT_s) \tag{3}$$

The corresponding digital transfer system is:

$$H(z) = \sum_{n=0}^{\infty} h(nT_s) z^{-n} \tag{4}$$

The corresponding frequency response is:

$$H(e^{j\omega}) = \frac{1}{T_s}\sum_{k=-\infty}^{\infty} G(j\Omega - jk\Omega_s) \tag{5}$$

This conversion mode from G(s) to H(z) is called "impulse response invariance method"[6]. It transforms the stable G(s) into the stable H(z), thus transforming the design of digital filter into the design of corresponding analog signal[7]. Therefore, using Ω and alpha design low-pass analog filter G(s), using formula transformation be corresponding H(z), so as to complete the design of IIR digital filter.

IIR filter has the following advantages: the filtering requires a small amount of computation, which can obtain better stopband and passband attenuation, more accurate edge frequency of stopband and passband. IIR filter is suitable for the case that only need to minimize noise and other aspects are not required.

3.3 FIR low pass filter design method and characteristic

The characteristic of FIR system is that its $H(z)=\frac{B(z)}{A(z)}$ corresponding denominator coefficient is all 0. FIR filter design principle is constantly approximate to the ideal frequency filter frequency characteristics.

The unit response formula of the ideal low-pass digital filter is:

$$h_d(n)=\frac{1}{2\pi}\int_{-\pi}^{\pi}H_d(e^{j\omega})e^{j\omega n}d\omega=\frac{1}{2\pi}\int_{-\omega_c}^{\omega_c}e^{j\omega n}d\omega=\frac{\sin(\omega_c n)}{\pi n} \tag{6}$$

Since $h_d(n)$ is a descending sine function, it is physically impossible. So the method of truncating $h_d(n)$ is used to get an approximate function that can be realized. Only the point from $h_d\left(-\frac{M}{2}\right)$ to $h_d\left(\frac{M}{2}\right)$ is taken, and the intercepted $h_d(n)$ is shifted to get:

$$h(n)=h_d\left(n-\frac{M}{2}\right)\quad, n=0,1\ldots M \tag{7}$$

Then his transfer function can be obtained as follows:

$$H(z)=\sum_{n=0}^{M}h(n)z^{-n} \tag{8}$$

The H(z) can be seen as $H_d(e^{j\omega})$ approximation.

Because of the truncation of $h_d(n)$, when the M value becomes larger, there will be Gibbs phenomenon, that is, ripples in the passband. $h_d(n)$ is truncated to M length, which is equivalent to adding a rectangular window with M+1. Rectangular window spectrum have bigger side lobe, the Gibbs phenomenon comes from $H_d(e^{j\omega})$ and the side lobe of convolution. Therefore, selecting the window function with smaller edge lobes can reduce the Gibbs phenomenon, but the transition zone will also be widened. For example, hamming window is adopted to replace rectangular window[8].

There are many other design methods for IIR filters, such as Chebyshev optimal uniform approximation.

FIR filter and IIR filter characteristics are different. Allowing the design of multi-pass filter, its system is always stable, which is easy to achieve linear phase. However, it is not easy to obtain good stopband and passband attenuation characteristics and to realize that timely.

4. Conclusion

Several digital filtering techniques, the design methods and characteristics of IIR and FIR digital filters are discussed. Digital filtering technology varies greatly, and its application is also different. For example, IIR filter can be selected when better passband is needed; FIR filter can be selected when the system is stable; when real-time and linear phase are needed, average filter and smooth filter can be selected.

Digital filtering technology is widely used in signal processing. There are also a variety of filter technology theories and filter design methods, and new theories are also constantly put forward. In the face of the advent of 5G era, more new filters will be generated, and digital filtering technology will also be applied to all walks of life with a more flexible, fast and convenient development.

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