Design of Adaptive Filter for Weak Signal Acquisition System

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Abstract. In order to solve the contradiction between the accuracy of weak signal acquisition and the limited frequency band, a weak signal acquisition system based on adaptive filter is designed in this paper. According to the change of the signal frequency, the proposed adaptive filter based on the technology of hardware adjustable narrowband filtering decides the frequency of the input signal, realizes the narrowband filtering and the large dynamic variation of specific frequency signal. And the high precision acquisition of weak signal can be obtained through the processing of filtering technologies under complex noise conditions, which includes adaptive frequency search, digital low-pass filtering, adaptive adjustment, etc. Finally, the test experiment of the designed filter has been implemented, and the experimental results show that the adaptive filter of the weak signal acquisition system can meet the design requirements and show high performance of data acquisition for weak signal.

Keywords. Adaptive filter; data acquisition; weak signal.

1. Introduction

With the development of electronic technology and sensor technology, data acquisition systems have been widely used in military, radar system, aerospace and other fields. Different performance requirements of the data acquisition system will be put forward by different demands, and how to design a high precision data acquisition system should be carefully considered especially when the object is microvolt level voltage variable with a higher dynamic range and wider frequency band [1-2]. If the signal collected by the data acquisition system is in a single frequency, then the high signal-to-noise ratio (SNR) will be achieved through narrowband filter [3]. However, if the signal frequency belongs to a certain frequency band, the filter mentioned above will never be able to meet the requirements of both bandwidth and acquisition precision at the same time [4]. The reason why the accuracy of the traditional data acquisition system is limited mainly account for large noise energy [5]. However, the noise can be eliminated if the parameter of the adaptable filter could be dynamically changed under various outside conditions [6]. And the acquisition accuracy would be improved effectively in a certain frequency band by adaptive filtering technology, that is the automatic regulation of parameter of adaptive filter to satisfy the narrowband filtering requirement decided by the characteristics of the input signal. In this paper, the problem of accuracy in weak signal acquisition system will be addressed by the design of adaptive filter.
2. **The Scheme of Adaptive Filtering Technology**

The adaptive filtering technology is a kind of filtering method developed rapidly in recent years providing an optimal filtering solution for uncertain research objects, which has high performance of denoising in practice [7]. It is well known that the spectrum distribution of noise is very broad, therefore, it is inevitable for noise to pass through the filter during data acquisition especially when the frequency bandwidth of the filter in hardware design is too large. Besides, it is impossible for the filter with broadening frequency bandwidth to acquire a high SNR when the input signal is sine wave with single frequency. So it is an effective way of applying narrowband filter to improve SNR and suppress noise. And according to the change of the signal frequency, the adjustable narrowband filter can automatically set the central frequency point, realizing narrowband filtering of the specific frequency signal as well as the large dynamic range of frequency shift. The adaptive filter processing framework is shown in figure 1.

![Diagram](image_url)

**Figure 1.** The block diagram of adaptive filtering.

As shown in figure 1, the adaptive filtering process is divided into three parts: the adaptive frequency searching, the digital low-pass filter and the adaptive regulation. And the function of the adaptive adjustment part is that when the upper computer obtains the frequency which has been fed back to the hardware narrowband filter, there would be deviation between the central frequency given by the hardware narrowband filter and the measured one. Subsequently, the actual frequency $f$ is acquired through the adaptive frequency searching, then it is able to naturally determine the adjustable range of the hardware narrowband filter which would be in the section of $[f - 1kHz, f + 1kHz]$. So there must be the maximum amplitude of the signal in the above section, which could be measured in the upper computer by the feedback adjustment filter. After then, the deviation mentioned above could be eliminated.

3. **Design of Low-Pass Filter with FIR**

3.1 **The Filtering Principle of FIR**

For an ideal filter, its cutoff frequency is abrupt, that is, the frequency response of the low-pass filter with ideal linear phase is shown in equation (1):
and the unit sampling response in time domain, corresponding to the frequency response in equation (1) processed by discrete time Fourier inverse transform, can be expressed by equation (2).

\[ h_d(n) = \frac{1}{2\pi} \int_{-\pi}^{\pi} e^{-jn\tau} e^{jwn} dw = \begin{cases} \sin\left(\frac{w_c(n-\tau)}{\pi(n-\tau)}\right) & , n \neq \tau, \tau = \frac{N-1}{2} \\
\frac{w_c}{\pi}, & n = \tau \end{cases} \]  

(2)

It is well known that the FIR filter is finite length, which means that the infinite unit impulse response has been truncated. However, there would be ripples and choppy transition bands appeared in pass-band and stop-band in frequency domain when there is truncation processed by rectangular window in time domain. The amplitude of the input signal would be seriously affected by the ripple size of pass-band, particularly for the high precision acquisition system, if the ripple size cannot be lower than the engineering requirements that will lead to the attenuation or amplification of the amplitude gain.

In order to attenuate the undesirable effect of the ripple of pass-band and stop-band as far as possible, the suitable window function and its length N should be carefully designed to meet the engineering requirements. The core steps of window function design are as follows: at first, the ideal frequency response should be determined according to the cutoff frequency, then the impulse response \( h_d(n) \) in time domain should be figured out by discrete-time inverse Fourier transform. Subsequently, the impulse response \( h_d(n) \) could then be truncated into finite sequence by the window function \( w(n) \), which can be shown in equation (3):

\[ h(n) = h_d(n) * w(n) \]  

(3)

Finally, it is necessary to check whether the frequency response meets the design requirements or not by calculating equation (3). If not, it needs to go back to change the window shape or the interception length, then repeat the calculation and test until the requirements are satisfied.

### 3.2 Calculation of FIR Filter Coefficients Based on Indicator Requirements

According to the requirement of hardware index, the precision of DC signal is 0.002%, that is, the ripple size of pass-band should be lower than 0.002%. Taking the actual precision of pass-band ripple must be lower than the precision mentioned above into account, the maximum ripple \( \delta \) of pass-band should be calculated via equation (4):

\[ \delta = 0.002%/1000 = 0.0000002 \]  

(4)

Then the minimum attenuation of the stop-band could be obtained by the maximum ripple \( \delta \) of the pass-band, which can be calculated through equation (5):

\[ A_s = -20 \log(\delta/(1+\delta)) = 153.98 \text{dB} \]  

(5)

When the hardware sampling rate \( f_s \) is 625 KSPS, the pass-band should be in the range of 0-9KHz, and the pass-band cut-off frequency of the digital low-pass filter would be 9KHz, then after normalization the pass-band cut-off frequency \( w_p \) is calculated and shown as equation (6):

\[ w_p = f_p / f_s * 2\pi = 0.0288 \pi \]  

(6)

The cutoff frequency \( w_c \) and the transition band \( \Delta w \) of ideal filter are determined by the stop-band cutoff frequency \( w_{st} \) of digital filter, and the transition bandwidth is determined by the main lobe width of window. And the relationship between the main lobe width and the length of sequence window
showing that the longer the window is, the narrower the transition band is, following the longer the filter calculation period is. Therefore, it is necessary to reduce the transition bandwidth as far as possible while to meet the requirements of data refreshed display combining with the running speed of the upper computer, so that the sequence length of the window has been chosen as 500. Then it firstly needs to select the kinds of window to calculate the transition bandwidth, the minimum attenuation of the stop-band must be greater than or equal to 153.98dB when choosing the shape of the window, however, the numerical value of common window function mentioned above is below 74dB. Fortunately, the stop-band attenuation of Kaiser window could be adjustable by parameter $\beta$ to meet the requirement of minimum attenuation, and the parameter $\beta$ could be obtained by using the existing empirical equation (7):

$$\beta = 0.1102 \left( A_s - 8.7 \right) = 16.01$$  \hspace{1cm} (7)

When the length $N$ of window is 500, the transition bandwidth $\Delta w$ can be calculated by the empirical formula of Kaiser window which is shown in equation (8):

$$\Delta w = \frac{A_s - 7.95}{2.285(N - 1)} = 0.0408 \pi$$  \hspace{1cm} (8)

Furthermore, the cut-off frequency of the ideal filter can be obtained by equation (9):

$$w_c = w_p + \frac{1}{2} \Delta w = 0.0432 \pi$$  \hspace{1cm} (9)

Substituting the cut-off frequency $w_c$ into the Kaiser window expression, the following equation will be got and shown as equation (10):

$$w(n) = \frac{I_0(\beta \sqrt{1 - (1 - \frac{2n}{N-1})^2})}{I_0(\beta)} R_n(n) = \frac{I_0\left(16.01 \sqrt{1 - (1 - \frac{2n}{499})^2}\right)}{I_0(16.01)} R_n(n)$$  \hspace{1cm} (10)

The frequency response $h_d(n)$ of the ideal filter can be obtained by substituting the cut-off frequency $w_c$ into equation (1), and then the filter coefficient based on the index requirement will be achieved by substituting $w(n)$ and $h_d(n)$ into the equation (3). Checking the response frequency figured out weather meets the index calculated above or not, if not, it is necessary to calculate and verify again by changing the truncation length $N$ until the requirement is met. However, in practice, it is a hard work to calculate the design parameters mentioned above just by manual computation. Luckily, the powerful auxiliary tool provided by MATLAB plays a key role in simplifying the plenty of calculation of design, which is a tool called FDA. There is a popping design surface where the low-pass filter can be designed by selecting window function, then it needs to input the parameters into the window, including the cut-off frequency of FIR filter, the sampling frequency and the parameter $\beta$, where the cutoff frequency in digital domain needs to be normalized to figure out the analog frequency. If the Kaiser window is selected, the design surface is shown in figure 2.

By pressing the design button in figure 2, the amplitude response of the filter will be displayed. It can be seen intuitively whether the parameters meet the design requirements or not, if not, it just needs to regulate the data of Specify order in figure 2. And the amplitude-frequency response of the low-pass filter calculated by this tool is shown in figure 3.

Through the value on the cursor, it can be seen that when the frequency is 9.002268KHZ, the amplitude gain is -2.82*10^7dB, and the precision requirement could be satisfied with such a small pass-band ripple. After obtaining the required frequency response, the unit sampling response sequence can be derived in the form of header file which can be directly added to the engineering file in the upper computer by slightly modified, waiting to be invoked. The time domain convolution
operation is compiled into C language code in the upper computer program, then the digital signal uploaded from the lower computer and the coefficient of FIR filter are processed by convolution to complete the filtering processing.

![Figure 2. The interface of parameter settings.](image)

![Figure 3. Amplitude-frequency curve of low-pass filter.](image)

4. Test Validation
This experiment is mainly to show the effects of digital low-pass filtering through the output signals before and after filter treatment for validating the effect of the design in this paper. The input signal with amplitude of 0.5V from digital calibration of FULKE5502A has been selected in the experiment, which has been collected by AD data acquisition system. The raw input signal is shown in figure 4a. Then, the input signal has been processed through low-pass filtering, which is shown in figure 4b. It is obvious that the amplitude of fluctuation of random noise has been completely eliminated, illustrating that the proposed algorithm is feasible. And it can be seen from the comparison of the two figures 4a and 4b that after adaptive filtering, the noise accompanying with the weak signal is eliminated, whereas the weak signal is completely overwhelmed by the noise in the raw data. So, after adaptive filtering, the weak signal is abstracted from the environment noise perfectly and the performance of adaptive filter is outstanding. It is verified that the design in this paper has superior advantages of filtering for weak signal which can be used in weak signal acquisition system.
5. Conclusion
In this paper, the data acquisition system with the function of low-pass filtering based on adaptive filtering technology has been designed, which is suitable for the acquisition of weak signal. The algorithm of frequency measurement based on the theory of DTFT has been compiled to determine the parameter of narrowband filter. And the adaptive filtering process of digital signal has been realized, including the digital low-pass filtering and the adaptive filtering based on hardware narrowband filter. Finally, the filter designed in this paper is tested and validated. And the experimental result shows that the adaptive filter of the weak signal acquisition system can meet the design requirements and realize the high precision acquisition function of the weak signal. Hence, it can be used to obtain accurate band-pass filtering in low frequency bands.

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