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Implementing method of optimum front-end conditioner based on Butterworth filter

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Abstract The front-end conditioner is an essential part of digital systems of nuclear spectrometer, which functions in two ways: (1) prevents saturation of the subsequent ADC; (2) limits the bandwidth of frequency to realize anti-aliasing. To realize the above-mentioned functions, an optimum front-end conditioner for a resistive feedback charge-sensitive preamplifier is designed. In the conditioner, the pole-zero compensation (P/Z compensation) technique was used to effectively filter signals from the preamplifier. The Butterworth filter was improved after the pole-zero position was optimally set up to shape the wave of output, which tallied with the whole system. The front-end conditioner can resolve the aberration of waveform of nuclear signals in a regular Butterworth filter. Compared with the traditional triple-pole filtering circuitry, the circuitry of this conditioner is more compact and flexible. Moreover, its output waveform is more symmetrical and the signal-to-noise ratio (SNR) is higher. The improvement in the resolution of spectrometer is also significant.

Key words Digital nuclear spectrometer, Pole-zero compensation, Butterworth filter

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1 Introduction

A traditional multichannel analyzer based on nuclear spectrometer systems have been developed for digitalized systems. Fig. 1 shows a type of high- resolution system termed as the unattached digitalized nuclear spectrometer system (UDNSS), which has been widely investigated and applied [1, 2].

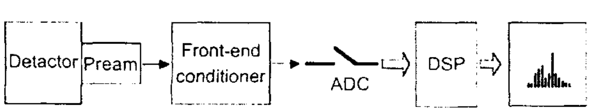

Fig. 1 The structure of UDNSS.

The main advantages of this circuit include no specific requirements for ADC sampling rates, simplified structures, and easy realization. However, regular ADCs with limited sampling frequency and dynamic range prevent the nuclear spectrometer from achieving high resolution. Therefore, a front-end conditioner is introduced in the digitalized systems. The conditioner critically pretreats output signals from the preamplifier. Hence, a high-energy resolution could be attained even for inputs at high counting rates and with large dynamic range.

The front-end conditioner has two basic functions. First, it shapes outputs with long exponential attenuating tail that come from the preamplifier into narrow pulses with optimum waveform. Saturation of ADC can be avoided during the overlapping of pulses. Thus, it is possible to modify the system's resolution. Second, a low-pass filter is used to limit the maximum

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frequency of signals in the frequency domain, which follows the principle of Nyquist sampling, realizing anti-aliasing of frequency. The pretreated signals are sent to the ADC for digitalizing, and then handled by a digital signal processing (DSP) system. Therefore, a high resolution of nuclear-energy spectrum can be obtained.

This article discusses the characteristics of nuclear signals in time domains and frequency domains. A novel type of front-end conditioner is proposed to better optimize the performance of the UDNSS and enhance the system's energy resolution. The composition of the conditioner, the characteristics of impulse response, and the realizing method are also described. The wave shapes and the results of energy spectra measurement are also dealt with in this study.

2 Characteristics of nuclear signals

In measuring systems of nuclear energy spectra, a sensing part usually contains a detector and a charge-sensitive preamplifier. A general charge-sensitive preamplifier is of a resistive feedback type. The output signal $V_i(t)$ is an exponential signal function with a fast-rising edge and a long tail. For easy analysis, the signal $V_i(t)=U(e^{-t/\tau_f}-e^{-t/\tau_r})$ is scale-normalized and can be estimated as [3]:

$$V_i(t)=e^{-t/\tau_f} \tag{1}$$

where τ_r is rise time constant and τ_f is fall-time constant of the biexponential signal.

The equation of frequency domain is:

$$V_i(s)=\frac{1}{s+\frac{1}{\tau_f}} \tag{2}$$

For reducing the impulse's tail lumping and preventing ADC saturation under high count rates, a pole-zero compensation (P/Z compensation) technique is used to obtain short tail-signals. For this, a transfer function can be composed [4]:

$$H_1(s)=\frac{s+\frac{1}{\tau_f}}{s+\frac{1}{\tau_p}} \tag{3}$$

where $\frac{1}{\tau_p}$ is the new pole.

After $H_1(s)$ acts on $V_i(s)$, a complete P/Z compensation can be realized. The output signal in frequency domain is expressed by

$$V_1(s)=\frac{1}{s+\frac{1}{\tau_p}} \tag{4}$$

And the formula of time domain of $V_1(s)$ is

$$V_1(t)=k_1 e^{-t/\tau_p} \tag{5}$$

Under the complete realization of P/Z compensation, $\delta(t)$ can be considered as a signal passing through a filtering network of $1/(s+\Omega_p)$, in which Ω_p equates $1/\tau_p=2\pi f_p$.

It is necessary to add several poles after the P/Z compensation network to realize wave filtering. Theoretically, as the number of poles of a filter increases, its impulse will be closer to the shape of the optimized filter. Hence, there will be an improvement in the SNR of treated signals. A current method is adding several real mutiple poles or conjugate complex poles [3].

The frequency response of a low-pass filter with m real poles is

$$H_{\text{r-pole}}(s)=\frac{1}{\tau^m(s+1/\tau)^m} \tag{6}$$

and the frequency response of a filter with n complex poles is

$$H_{\text{c-pole}}(s)=\frac{1}{[s+(a_1+j\omega_1)][s+(a_1-j\omega_1)][s+(a_2+j\omega_2)]}\times\frac{1}{[s+(a_2-j\omega_2)]\ldots\ldots[s+(a_n+j\omega_n)][s+(a_n-j\omega_n)]}$$
$$=\frac{k_1}{s+a_1-j\omega_1}+\frac{k_1^*}{s+a_1+j\omega_1}+\frac{k_2}{s+a_2-j\omega_2}+\frac{k_2^*}{s+a_2+j\omega_2}+\ldots\ldots+\frac{k_n}{s+a_n-j\omega_n}+\frac{k_n^*}{s+a_n+j\omega_n} \tag{7}$$

Combining a complete P/Z compensation filter with a low-pass filter network, a $\delta(t)$ signal can be considered to pass through $(m+1)$ rank real-pole or $(2n+1)$ rank complex-pole low-pass filter. Therefore, output signal shapes are totally determined by impulse response of systems corresponding to Eqs. (6) and (7). The impulse response of $(m+1)$ rank real-pole filter can be written as