CHAPTER III
DESIGN AND DEVELOPMENT

Noise is a major contribution to energy resolution degradation in nuclear radiation spectroscopy. It can be improved through signal processing using wave-shaping network in spectroscopy amplifier with optimum processing time adjustment. Generally, the signal performance index is identified by the signal to noise ratio. Such a modern signal processing technique is used to analyze the optimum condition of the system by the relation of signal and noise power spectral density in connection with the transfer function of the shaping network at proper processing time. In this research, the power spectral densities of signal and noise are calculated by sampling preamplifier's output at low background level for simulation technique. Transfer function of shaping network is derived from the output response of spectroscopy amplifier using step-input signal. These processes can be achieved via a high sampling rate digital oscilloscope with sufficient bandwidth, where each signal is digitized and transferred to a microcomputer for simulation. In this section, the development of an optimum shaping time estimation program for energy resolution enhancing in nuclear spectroscopy and data sampling technique design are described.

3.1 Concept of optimum shaping time estimation

From the theoretical point of view and the review of previous literatures, the problem for searching an optimum shaping time of spectroscopy amplifier without an experiment carried out manually can be solved by simulation method. The simulation method needs the input from several parameters such as noise from preamplifier, counting rate at a specific energy of radiation and transfer function of filter network in a spectroscopy amplifier. The simulation of signal processing chain must be performed in a computer and generate the signal pulse like virtual amplifier output. The histogram of amplitude distribution will be reconstructed and the deviation of signal amplitude output becomes the figure of merit for optimization. The shaping time which gives the minimum deviation of pulse amplitude under noise and count rate condition could be the best operating point for the nuclear spectroscopy at each operating environment. The diagram below shows the concept for searching the best shaping time to operate the nuclear spectroscopy system at each environment by simulation method.

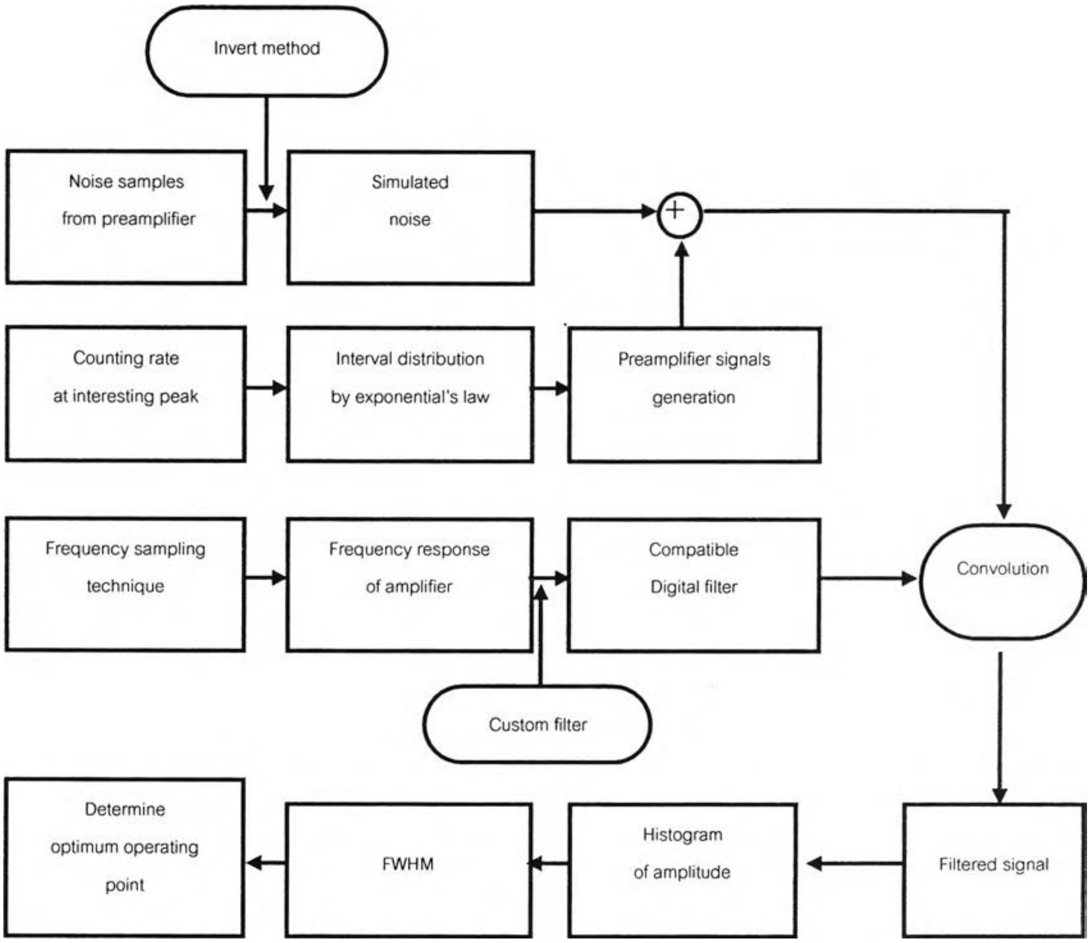


Fig. 3.1 The concept for searching the best shaping time by simulation of the nuclear spectroscopy system at each environment.

There are four steps for completion of optimum condition searching as follows: First, sample of noise sequence is acquired from the preamplifier output and used as the input for noise simulation by Monte Carlo's inverse method. Second, the counting rate at the peak of interest is measured and used as the mean count rate to calculate the time interval between pulses. Third, frequency response of amplifier is transformed into a digital filter by custom filter design method. Finally, the noises are added to preamplifier signal and convoluted with digital filter. The amplitude distribution of filtered signals is represented as a histogram and the corresponding FWHM is determined. The process continues and the FWHM at each shaping time value are plotted for optimum operating point estimation.

3.2 Front-end signal pulse simulation.

The front-end signal pulse is a complex signal composing of electronic noise from preamplifier, variation of charge collection and signal pulse pile up due to the random nature of radiation emission. Therefore, the simulation of this signal pulse will be integrated with specific signal function at each operating condition under MATLAB program, as a calculation tool. The empirical cumulative distribution function (ecdf) of MATLAB program referred to appendix A, is used to create the signal simulation source.

3.2.1 Time domain simulation of noise. In general nuclear spectroscopy system, the time profile of the detector signal and response of the pulse shaper are used in energy spectrum analysis as for analyzing the pile-up and ballistic deficit [1] in contrast to noise analysis which is usually done in frequency domain. In this study, the method for generating time-domain electronic noises with the simple technique is conducted to simulate the same physical characteristic in time and frequency domain of noise referred to preamplifier output used in each spectroscopy system. In order to simulate the time-domain noise in each nuclear spectroscopy system, the probability distribution function of noise amplitude is needed for the inverse transform method [31,32]. The favorite method in Monte Carlo is employed to converts the uniform probability distribution of the random numbers between 0 to 1, U(0,1), into the random number according to the desired probability distribution, f(x), where x is signal amplitude, before carrying out a simulation of each nuclear spectroscopy system. Let F(x) be cumulative distribution function of f(x) and F^{-1} is the inverse function of F(x), so x can be found from Eq. (3.1)

$$x = F^{-1}(u) \tag{3.1}$$

Generally, most noise has a Gaussian or normal distribution of instantaneous amplitude with time [33], including the situation of noise from preamplifier output. The amplitude distribution of a sequence of noise in theoretical model can not fit to a realistic distribution due to unpredictable interference at each environment. Therefore, the most straightforward and popular method for generating truly random number is to amplify a noise signal and then sampling it at a constant sampling rate [34]. The probability distribution function of noise amplitude, F (x), that is a fraction of the total time for which noise amplitude is less than or equal to x [35,36] is measured based on the fact that noise signal is ergodic process that sample's ensemble average equal to time average. Ergodicity implies that all the statistical properties of process are invariant in time and that these properties are deducible from measurements made in time [35]. The unknown distribution function of noise amplitude is estimated via the empirical cumulative distribution function, Fn(x), that is its natural estimate of F(x) according to the Glivenko-Contelli theorem [37]. If Xi is a sequence of noise samples drawn from preamplifier output where i = 1,2,...,n. Thus Fn(x) is showed by Eq.(3.2) .

$$Fn(x) \equiv \frac{1}{n} 1(Xi \leq x) \tag{3.2}$$

3.2.2 Random photon signal generation. Poisson's law usually presents the random nature of radiation decay from radioisotope source. This law is popularly used in simulation of pile up in nuclear spectroscopy system. The mean counting rate of radiation is required for signal simulation based on photon generating model, to calculate the time interval between the two consecutive radiations induced pulses. In signal simulation, the interval distribution function, f(Δt), is given by

$$f(\Delta t) = \lambda e^{-\lambda \Delta t} \tag{3.3}$$

Where Δt is time interval and λ is the true counting rate. Time intervals are chosen randomly by using the cumulative distribution function, cdf, obtained by integrating the probability distribution function from 0 to Δt. the cdf is

$$F(\Delta t) = \int_0^{\Delta t} f(x)\,dx = 1 - e^{-\lambda \Delta t} \tag{3.4}$$

This function is set equal to the cdf of uniform random number to obtain Δt from the uniform random number C, which can be written

$$F(\Delta t) = C \tag{3.5}$$

The time interval between the two consecutive pulses can be obtained by sampling the random number from uniform distribution as the input to Eq. 3.4 and the photon generation time is shown in Fig. 3.2

$$\Delta t = -\frac{1}{\lambda}\ln(1 - C) \tag{3.6}$$

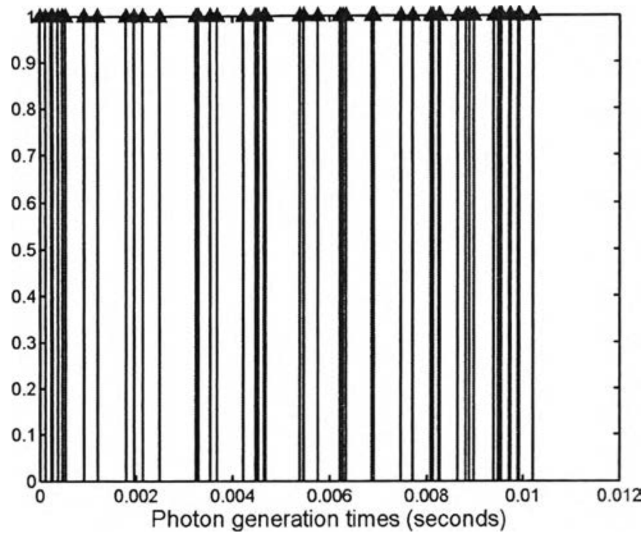


Fig. 3.2 Illustrate the photon generation time according to Eq. 3.6 for the counting rate 5000 counts per sec. with measured time 0.01 sec. The arrows represent a photon that occurs along the generation time.

3.2.3 **Preamplifier signal generation**. The new generation of radiation detectors and their electronics design for specific applications needs system simulation in time domain to study the behavior or test the design of the system. The classical modeling of radiation signal is defined in Eq. (3.7)

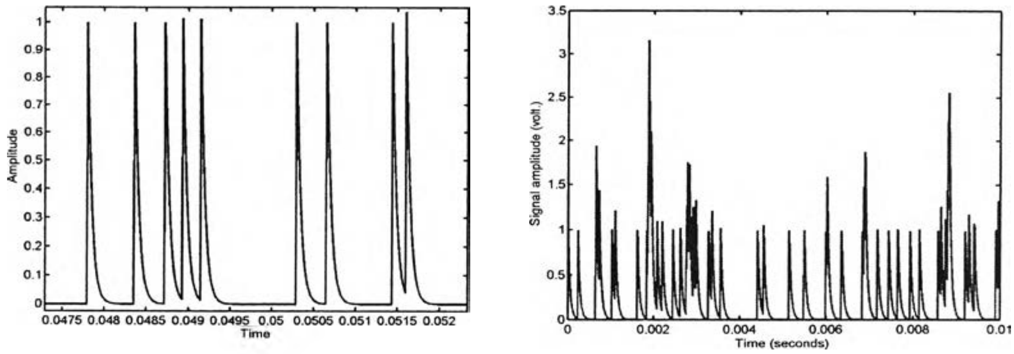
$$s(t) = \sum_k A_k \, p(t - t_k) + n(t) \tag{3.7}$$

Where s(t) represents the radiation signal and A_k stands for the amplitude of pulse proportional to the energy of radiation energy. p(t) represent the preamplifier impulse response, t_k is an arrival times of detected radiation which is distributed according to Poisson's law of emission rate of radiation, while the electronics noise n(t) is characterized differently in each system.

The charge sensitive impulse response, p(t), with the time constant pulse decay time,τ, can be written in Eq. 3.8

$$p(t) = Ae^{-t/\tau} \tag{3.8}$$

Where A is arbitrary value corresponding to signal voltage due to amount of charge.



a) Low counting rate simulated b) High counting rate simulated

Fig.3.3 Simulated radiation signals from preamplifier.

Fig.3.3 shows the simulated radiation signals from preamplifier output. For the detector which is designed for specific application, this model has to be modified. The amplitude of signals originated from the detector is in proportion to energy of radiation and the distribution of signals amplitude represents the quality and quantity of interesting energy. Time interval between consecutive exponential pulses follows Poisson law and lead to pulse pile-up when time interval is too short. The simulated noise and preamplifier signal output is added together to generate the composite preamplifier output.

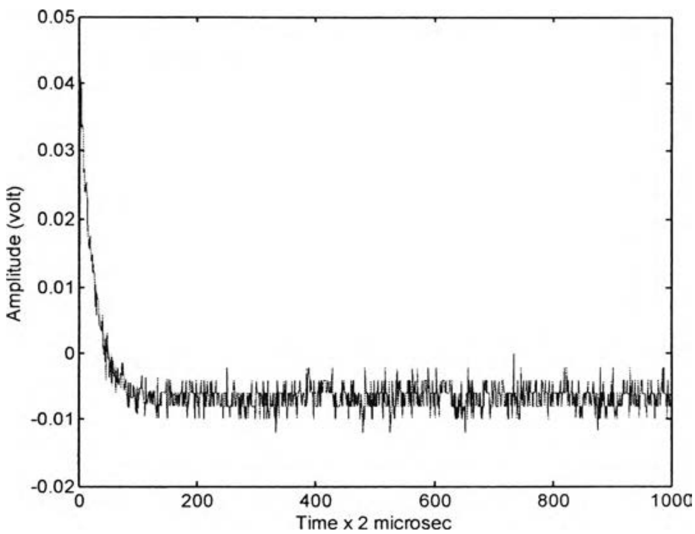


Fig. 3.4 The preamplifier signal output and noise.

3.2.4 Front-end noise sampling technique. Actually, front-end noise arises from detector voltage bias and preamplifier. To conduct the calculation of front-end noise, power spectral density and limits of sampling frequency as well as the system for data acquisition are designed as shown in Fig. 3.5. The digital storage oscilloscope at a sampling rate of 5 MS/s is employed for system testing. The data are taken from a preamplifier output with the detector under bias, at low background level and sent to a microcomputer via RS-232 serial port for data manipulation using the scrip command of the program.

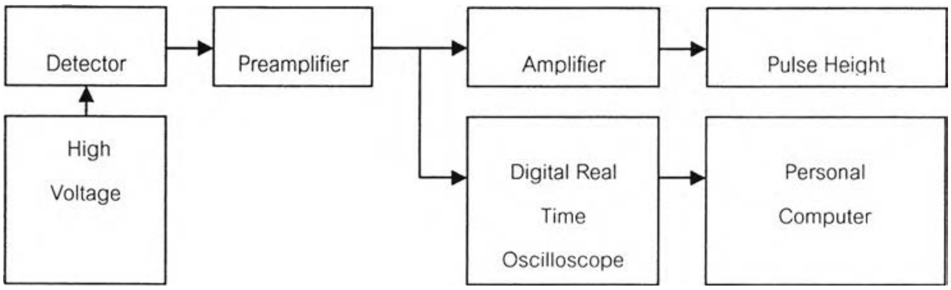


Fig. 3.5 System design for the estimation of the front-end power spectral density.

Two types of preamplifiers for proportional detector Canberra model 2006 and the locally developed one are chosen for the study. The tested results show a different spectrum profile of power spectral density as illustrated in Fig. 3.6. The frequency distribution characteristics for different shaping times and feature of the front-end system under test are utilized for a network modeling.

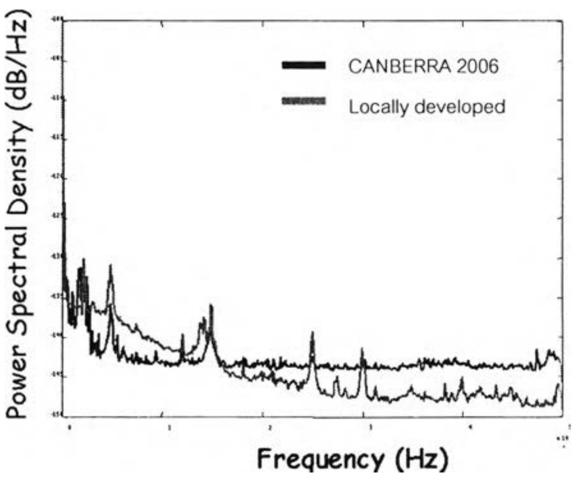


Fig. 3.6 The front-end noise power spectral density of CANBERRA 2006 and locally developed preamplifiers.

3.3 Spectroscopy amplifier modeling.

As mentioned previously, a spectroscopy amplifier is supposed to be a complex system and can be defined as a linear time-invariant model. The characteristics for noise reduction is determined by its power transfer function or the frequency response obtained from the Fourier transform of time domain output and input signal of the spectroscopy amplifier with reference to equation 2.9

3.3.1 Frequency response sampling technique. The step impulse response is applied to investigate the frequency response of spectroscopy amplifier. A system arranged for data sampling is shown in Fig 3.7, a step output of a function generator is applied to the spectroscopy amplifier under study and both of input and output signals are sampled by a high sampling digital oscilloscope. Those sampled data are sent to a microcomputer via RS-232 port for power spectral density calculation using the scrip command of the program.

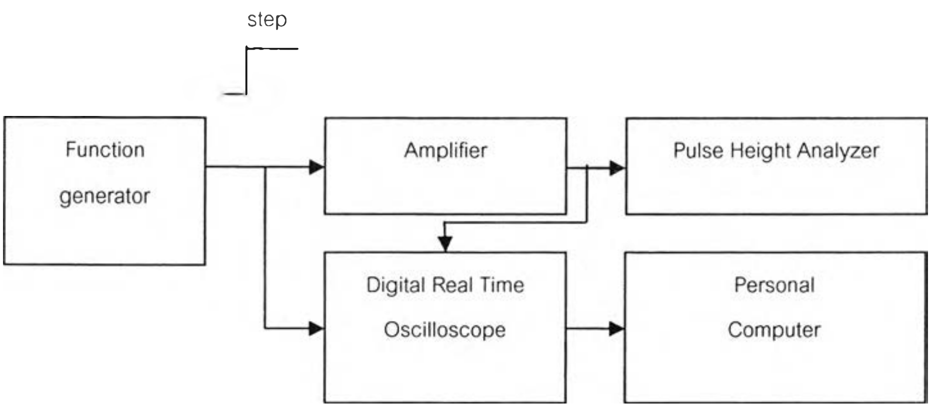
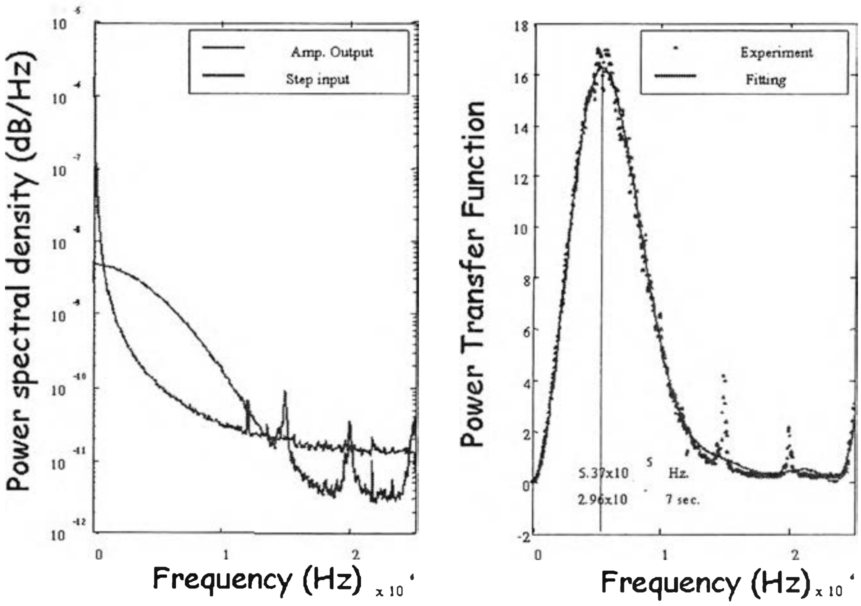


Fig. 3.7 System setup for the estimation of the frequency response of spectroscopy amplifier.

3.3.2 Power transfer function of band pass filter estimation. Canberra 2020 spectroscopy amplifier is the test pieces for frequency response estimation. A frequency response of step input at 0.25 microsecond shaping time is set for creation of both input and output power spectral density (PSD) as shown in Fig. 3.8a. The power transfer function of the spectroscopy amplifier is calculated by dividing a PSD output with PSD of step input and the frequency response is obtained as shown in Fig. 3.8b. The results show a narrow band pass characteristics of the spectroscopy amplifier. A linear scale plot of the above mentioned frequency response gives a sharp peak of frequency position, which is convenient to determine a corner frequency or a reciprocal term of shaping time.

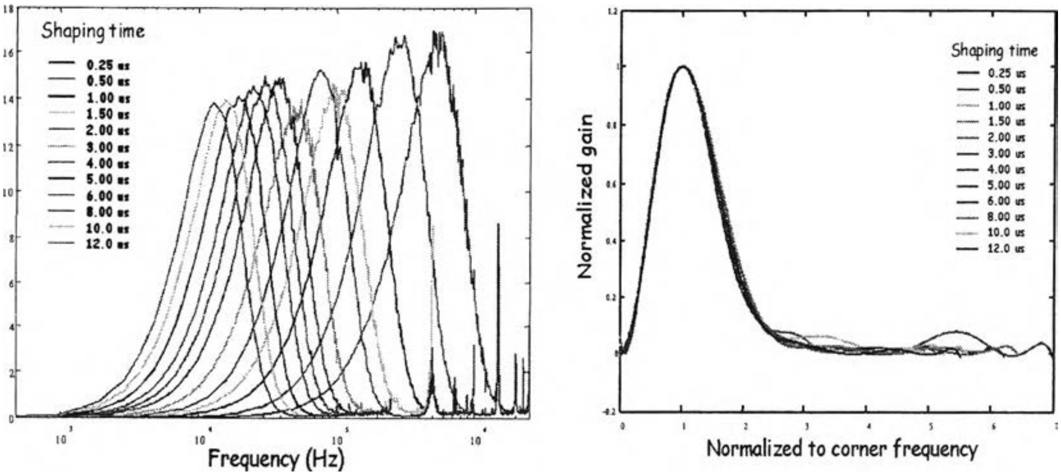
Power transfer function of Canberra 2020 (0.25 μs)



a) b)

Fig. 3.8 Show the calculated frequency response of Canberra 2020 spectroscopy amplifier a) The PSD of step input and amplifier output. b) The frequency response at 0.25 microsecond shaping time

The various shaping time values provided for optimum noise reduction, in Canberra 2020 spectroscopy amplifier, was studied from 0.1 μs up to 12 μs. Linear scale plot of the frequency response of a cascade stage high pass and low pass filter at different corner frequency shows the bandwidth limit at each shaping time setting, as shown in Fig. 3.9a. All of frequency responses are then normalized by their corresponding corner frequencies and plotted. The results show the same response profile and it may conclude that a filter is characterized by a specific filter order and bandwidth independent from the shaping time settings as shown in Fig. 3.9b.

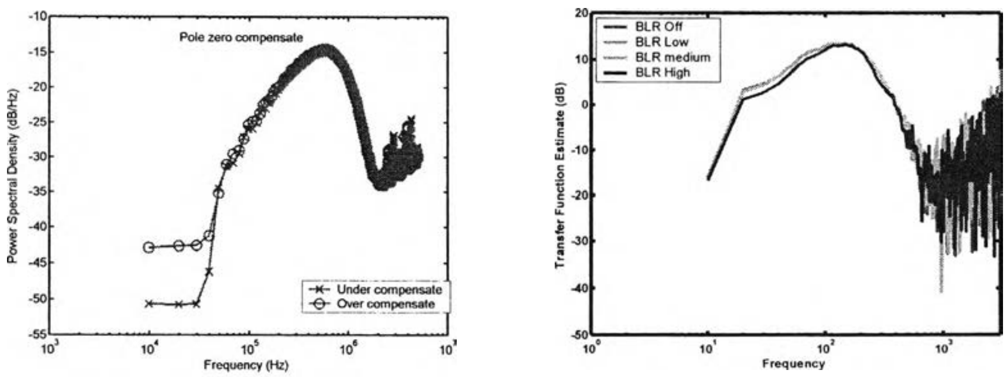


a) Frequency response at different shaping time

b) Normalized profile

Fig. 3.9 Frequency response of Canberra 2020 spectroscopy amplifier at 0.25 to 12 μs shaping time.

Besides, the characteristic frequency response shape can also be applied to determine the pole-zero compensation and baseline restoration effects. As shown in the frequency response curve of Fig. 3.10a, the frequency response in log scale shows a different value of power transfer function at low frequency when the degree of pole-zero compensation is changed to over or under compensation. Fig. 3.10b shows the different profile shapes of frequency response in log scale plot for different degrees of baseline restoration.



a) Pole-zero effect

b) Baseline restorer effect

Fig. 3.10 Frequency responses of spectroscopy amplifier at different degree of pole zero compensation and baseline restoration.

3.3.3 Designing of digital filter from frequency response. Any linear system can be analyzed in frequency domain by using Fourier transform method. This means that system is absolutely described by a changing amplitude and phase of output signal, at constant input signal, passing through it at any frequency, called the frequency response of system. A similar analysis can be done in time domain by using convolution of the input signal with impulse response. Because the impulse response and frequency response represent the information about the system, thus, they have a direct relation between both of them. The relationship between the impulse response and the frequency response is one of basis method of signal processing. Generally, it can be state that a system's frequency response is the Fourier transform of its impulse response.

In this section the impulse response function, called digital filter which represent the characteristics of nuclear spectroscopy will be designed. This method, in other words called custom filter, is used to design digital filter with an arbitrary frequency response fit to the needs of any applications. The designing steps can be written in flow chart in Fig. 3.11, with the developed program in appendix A.

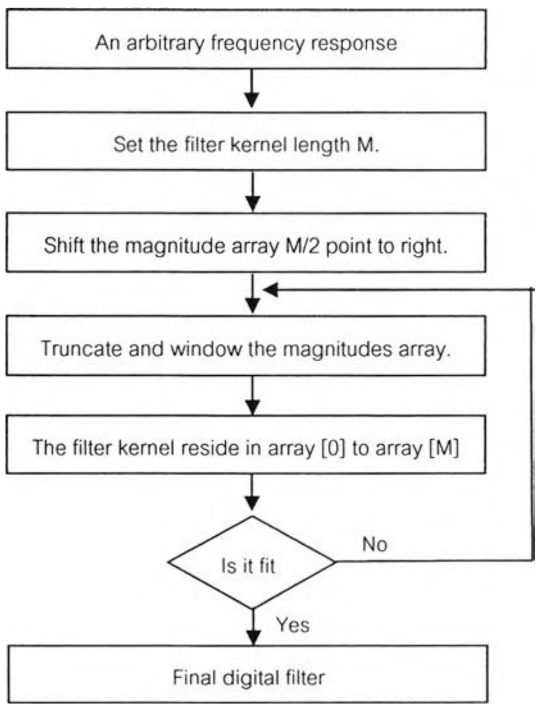


Fig. 3.11 The flow chart of designing the digital filter from arbitrary frequency response.

The desired frequency response of a system, H[k], is inserted to the inverse Discrete Fourier transform, DFT, in order to transform a frequency response from frequency domain into impulse response in time domain. The equation used to calculate the inverse DFT or synthesis equation can be written as:[38]

$$h[n] = \sum_{k=0}^{N/2} \mathbf{Re}\,\overline{H}[k]\cos(2\pi kn/N) + \sum_{k=0}^{N/2} \mathbf{Im}\,\overline{H}[k]\sin(2\pi kn/N) \qquad (3.9)$$

Where;

$$\mathbf{Re}\,\overline{H}[k] = \frac{\mathbf{Re}\,H[k]}{N/2}$$

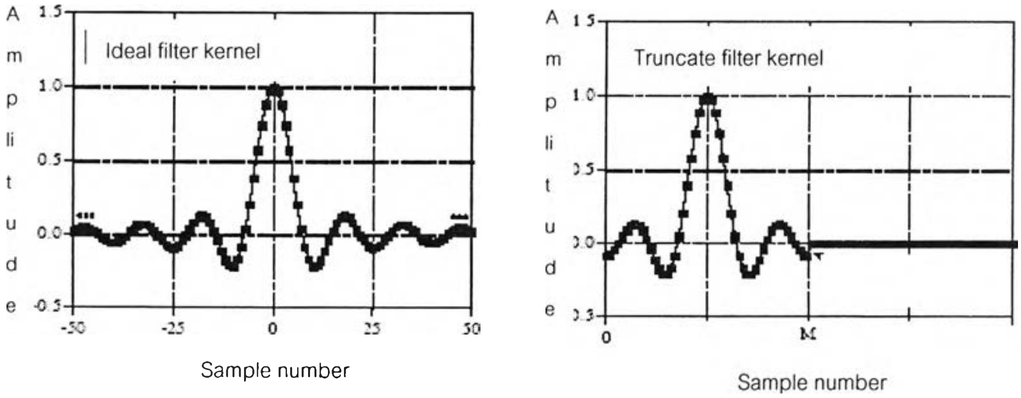
$$\mathbf{Im}\,\overline{H}[k] = \frac{-\mathbf{Im}\,H[k]}{N/2}$$

$$\mathbf{Re}\,\overline{H}[0] = \frac{\mathbf{Re}\,H[0]}{N}$$

$$\mathbf{Re}\,\overline{H}[N/2] = \frac{\mathbf{Re}\,H[N/2]}{N}$$

In equation (3.9), h[n] is impulse response being synthesized and the index, n, running from 0 to N-1 represent the number of impulse response coefficients in time domain. ReH[k] and ImH[k] stand for real and imaginary parts of frequency response, respectively while k runs from 0 to N/2. This equation needs Re$\overline{H}$[k] and Im$\overline{H}$[k] rather than ReH[k] and ImH[k] because they are slightly different from those in frequency domain and need for scaling before using to synthesize the impulse response.

The impulse response that corresponds to the desired frequency response, h[n], is not suitable for use as a filter kernel because an ideal impulse response is a continuous function with sample numbers spread over negative and positive area without dropping to zero amplitude as shown in Fig 3.12a and is impractical for numerical calculation by computer. To avoid this problem, the impulse response needs to be modified by truncating and shifting as shown in Fig. 3.12b. Accordingly, the impulse response is truncated into M points of desired filter kernel and replacing the impulse response outside M point with zero and the remaining portion is shifted to the right. This allows the filter kernel to be moved and only positive indices are used.



a) The real impulse response from inverse DFT.

b) The modified impulse response.

Fig.3.12 The impulse response modified by truncating and shifting.

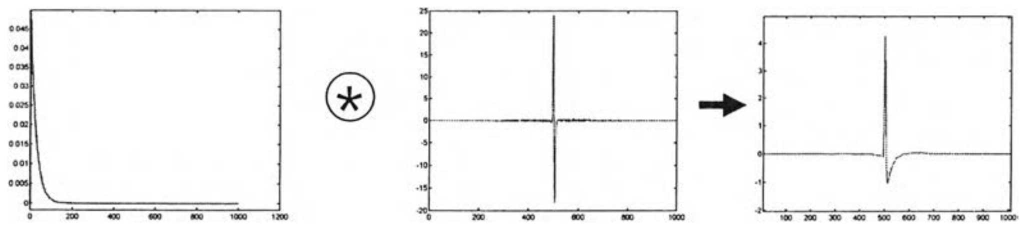
After the impulse response is modified, the ripple in the pass band and poor attenuation resulted from truncation is obtained. This problem can not be solved by increasing the length of filter kernel. The simple method for improvement is to smooth its curve or a fluctuation between the defined frequency response by multiply the modified impulse response with the smoothing curve function such as Blackman windows and Hamming window. The smoothing curve function are given by Eq (3.10) and Eq (3.11), respectively [38]

$$w[n] = 0.42 - 0.5\cos(2\pi n / M) + 0.8\cos(4\pi n / M) \quad (3.10)$$

$$w[n] = 0.54 - 0.46\cos(2\pi n / M) \quad (3.11)$$

The frequency response of the final impulse response function or digital filter can be tested before use. The test method is done by padding the digital filter with zero as for making the same filter length like the original one and taking Fourier transform to obtain its frequency response.

The digital filter or filter kernel which is derived from frequency response of spectroscopy amplifier can be used for time domain wave shaping simulation. This filter function is convoluted with simulated signal from radiation detector, noise and signal pulse noise and output signal like the output from spectroscopy amplifier is obtained as shown in Fig 3.13. These pulse height distributions of simulated output signals are analyzed for FWHM's as the data for performing the optimum shaping time searching.



Preamplifier output (convolution) Filter kernel (output) Amplifier output

Fig. 3.13 Simulated output produce from convolution between simulated preamplifier output and filter kernel.