PROCESSING OF PULSE DETECTOR SIGNAL OF IONIZING RADIATION WITH FILTERS OF FINITE IMPULSE RESPONSE

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Abstract: For the processing of pulse detector signal exposed in a field of ionizing radiation, generally linear algorithms are used. They are based on moving average value. In this paper are presented results obtained when digital filters with finite impulse response are used. The signal is treated with four different type of filters with various shapes: rectangular, triangular, saw tooth and binomial. Response time variation due to filter length starting from 10 and ending with 40 coefficients, was analyzed. Filter performances were determined using both deterministic and random signals. Random signal generation was performed utilizing previously acquired signals from real radiation source ¹⁹²Ir with step variation. Response time and cumulative error in the output result were obtained using deterministical step signal. A close coincidence in the results for both analyzes was achieved. In this paper, some aspects of implementation of analyzed algorithms are discussed, in case of their application in microprocessor based measurement systems.

1. Introduction

The first step towards the processing of the impulse detector signal for ionizing radiation is the treatment of the number of registered pulses N_{ℓ} during the time of *i*-th measuring period of time ΔT_{i} . The processing of the detector circuit pulses depends on the chosen way of operation of the measuring circuit: counting in fixed measuring interval or finding an average value of certain number of pulses in variable interval. The result of the averaging has a statistic uncertainty characterized with fractional standard error. Therefore, at constant rate of counting of input impulses, the accuracy with which the average value is determined, enlarges with the duration of the interval. In case of variable input rates, the moment of change of the average value of the signal, principally is unsynchronised with the measuring interval. The change of the input signal average value results in additional error. The problem of synchronization can be minimized with a choice of shorter elementary averaging intervals (optimized in

compromise with the statistical fluctuation of the result) and the solution should be found in combining more results processed with adequate algorithm.

The usable algorithms can be realized by implementing the principles of the hardware solution of the existing measuring circuits in adequate software. Such is the case with the quasi-exponential algorithm, adequate to the solution based on the digital measuring circuits which use principle of analog circuits with dosing capacitor. Several variations of the linear algorithm are known [1], floating mean algorithms used in newly built microprocessor-controlled rate meters [2], algorithms with ponderous moving average values [3] or the family of algorithms for evaluation of the average value of finite block of samples [4].

In this paper, the processing of the signal is performed with digital filters with finite impulse response (FIR). The proposed algorithms are applicable in microprocessor based dose meters. The analyzes are made through adequate simulations, with the aid of the software package for digital signal processing DADiSP [5].

2. Filtering of the signal with FIR filters

Generally, the linear filter from the stream of input samples x_k produces a stream of output samples according to the relation [6]:

$$y_{n} = \sum_{k=0}^{M} b_{k} \cdot x_{n-k} - \sum_{k=1}^{N} a_{k} \cdot y_{n-k}$$
 (1)

In the relation (1), (M + 1) coefficients b_k and N coefficients a_k are fixed and they define the impulse response filter. The filter produces a new output sample from the existing, M-th previous input samples and from the N samples of the filter output.

The second term in the relation (1) is not used for the filter with finite impulse response. In such case, the output filter samples can be obtained in the process of direct discrete convolution of the input samples (x_k) and the finite impulse response of the filter $\{h_k\} = \{b_k\}$;

$$y_n = \sum_{k=0}^{M} h_k \cdot x_{n-k} = \sum_{k=0}^{M} x_k \cdot h_{n-k}$$
 (2)

If only FIR filters are considered where all the coefficients are the same $b_k = 1/(M+1)$, from the relation (2) it is really found:

$$y_n = \sum_{k=0}^{M} b_k \cdot x_{n-k} = \frac{1}{M+1} \sum_{k=0}^{M} x_{n-k}$$
 (3)

Relation (3) presents a floating mean value of a block of M + 1 samples. At the rectangular filters the samples pass through an rectangular "window" (constant impulse res-

ponse). But, with adequate choice of the "window" shape different values of weighted coefficients can be obtained. In this way we can directly influence to the average value of filter output [7]. The only limitation of the values of these coefficients is:

$$\sum_{k=0}^{M} b_k = 1$$

Besides the algorithms with rectangular filter, three others are tested, whose coefficients are with shape "two on exponent". The impulse responses of the respective filters are with triangular, saw-tooth and binomial shape. As an input signal for the test purposes, a random signal is used, acquired from a real source of ionizing radiation source ¹⁹²Ir [8]. The signal is combination from two data streams with different average values. It is shown on Fig. 1. In order to determine the dynamics of the analyzed algorithms, the response time and the cumulative error are defined. For this purpose a deterministic step signal is used (ideal test signal), with levels which correspond to the average values of the random test signal.

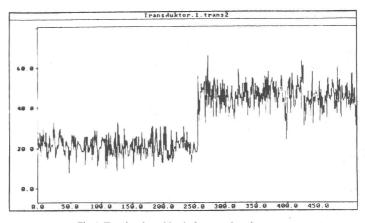
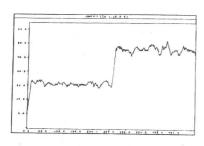


Fig. 1. Test signal combined of two random data streams

3. The analyzes of the obtained results

On the figure.2(a,b) are presented the results from the simulation with a rectangular filter (variant I) whose coefficients are:

$$h_n = \frac{1}{N+1}$$
, for $n = 0, 1, ..., N$



with:

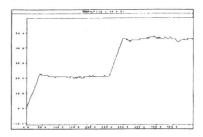


Fig. 2 Processing of the signal with FIR filter - rectangular a) length of the filter N = 10; b) length of the filter N = 40

On the figures 3 (a,b,c), are presented the results from the signal processing

- Triangular filter (variant II) with coefficients:

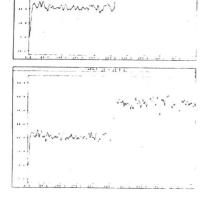
$$h_n = \frac{4n}{N^2}$$
, for $n = 0, 1, ..., \frac{N}{2}$ and $h_n = h_{N-n}$, for $\frac{N}{2} + 1$, $+ N$

- Saw-tooth filter (variant III) with coefficients

$$h_n = 2 \frac{N-n}{N(N+1)}$$
, for $n = 0, 1, ..., N$

- Binomial filter (variant IV) with coefficients.

$$h_n = \frac{1}{N^2} \binom{N}{n}$$
, for $n = 0, 1, \dots, N$



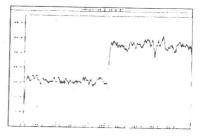
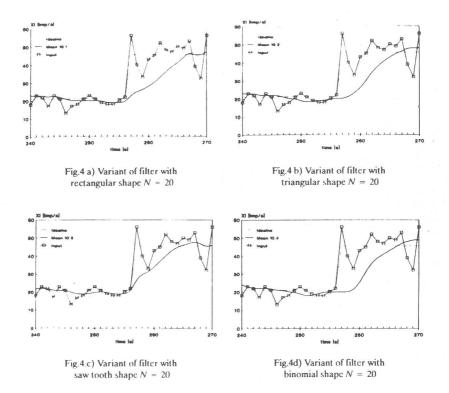


Fig.3 Signal processing with FIR (N=10) a) triangular shape b) saw tooth shape c) binomial shape

On the figures 4 (a,b,c,d), are presented details in the surrounding of the step random test signal for each of the described variants with the filters of same length N = 20.



On the Fig.5, the response of the four algorithms with deterministical input are presented. The test signal now is arranged by two step functions, and can be expressed as:

$$x(t) = 21.11 x_H(t) + 45.65 x_H(t - 256 \Delta t)$$
 (4)

were:

21.11 and 45.65 are the average values of random step signal,

 Δt – elementary interval,

256 - number of input data samples.

The surface under each curve is a measure for quantitative characterization of each algorithm.

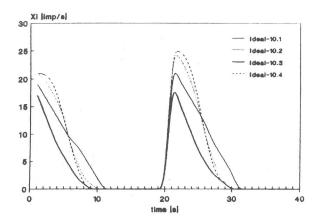


Fig.5 Responses obtained with different algorithms for deterministical signal

In the table 1, the review of the response time and cumulative error, calculated by relation (5) is presented.

$$\varepsilon = \int_{0}^{\infty} \left[y(t) - x(t) \right] dt \tag{5}$$

T able 1. Review of the response time and cumulative error

Algorithm (variant)	I	Ш	Ш	IV
Response time [s]	10	9	8	8
Cumulative error [pulse]	108.8	106.8	63.83	109.5

To synthesize the filter, the number and values of the coefficients h_k should be determined using one or other of several available techniques. Filters with linear phase are simpler for implementation. Due to the symmetry of the coefficients h_k of the finite impulse response filter it is possible the number of multiplications then to be divided by two; it becomes apparent that with one extra accumulator and control logic it may be possible to double the filter bandwidth.

4. References

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