PRESET TIME COUNT RATE METER USING ADAPTIVE DIGITAL SIGNAL PROCESSING

by

Aleksandar D. ŽIGIĆ

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Two presented methods were developed to improve classical preset time count rate meters by using adaptable signal processing tools. An optimized detection algorithm that senses the change of mean count rate was implemented in both methods. Three low-pass filters of various structures with adaptable parameters to implement the control of the mean count rate error by suppressing the fluctuations in a controllable way, were considered and one of them implemented in both methods. An adaptation algorithm for preset time interval calculation executed after the low-pass filter was devised and implemented in the first method. This adaptation algorithm makes it possible to obtain shorter preset time intervals for higher stationary mean count rate. The adaptation algorithm for preset time interval calculation executed before the low-pass filter was devised and implemented in the second method. That adaptation algorithm enables sensing of a rapid change of the mean count rate before fluctuations suppression is carried out.

Some parameters were fixed to their optimum values after appropriate optimization procedure. Low-pass filters have variable number of stationary coefficients depending on the specified error and the mean count rate. They implement control of the mean count rate error by suppressing fluctuations in a controllable way.

The simulated and realized methods, using the developed algorithms, guarantee that the response time shall not exceed 2 s for the mean count rate higher than 2 s^{-1} and that controllable mean count rate error shall be within the range of 4% to 10%.

Key words: preset time count rate meter, adaptable signal processing, low-pass filter, FIR filter, mean count rate

INTRODUCTION

Fluctuations of the results of processing of signals from radiation detectors in order to obtain mean count rate are caused by random variations of the spacing between successive input pulses even in steady state [1]. This property of both analog and digital rate meters is present irrespective of which measurement principle, preset time or preset count, is applied.

The properties of preset count algorithms have been analyzed in detail [2], applying classical methods of analysis. The improvements as regards statistical fluctuations by modifying spacings between input pulses [3], have been implemented in practice [4, 5]. The corresponding analyses of preset time algorithms have also been reported [6].

No attempts have been made so far to apply adaptable digital signal processing methods in the analysis and design of digital rate meters even though some indications of the equivalence of FIR (Finite Impulse Response) and IIR (Infinite Impulse Response) filters and some of digital rate meter algorithms have been reported [7].

The purpose of this paper is to present the application of digital filters [8], and adaptable digital signal processing [9], methods to the design of digi-

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Authors' address:
VINČA Institute of Nuclear Sciences
P. O. Box 522, 11001 Belgrade, Serbia and Montenegro

Author's e-mail:
alex@vin.bg.ac.yu

tal systems for calculation of mean count rate. Two methods for improving the performance of classical preset time algorithms are proposed. The first method starts with a longer preset time of 10 s assuming stationary rates corresponding to background radiation levels and then uses an adaptation algorithm to adjust the duration of the preset time interval if the mean count rate is changed. The second method starts with a shorter preset time of 1 s, being immediately prepared to react to sudden changes of the mean count rate, but switches to a longer preset time interval of 10 s if stationary rates corresponding to background radiation levels are maintained.

Both methods use an optimized detection algorithm to sense the change of mean count rate, adaptable low-pass filter to implement mean count rate error control by controllable suppression of fluctuations and an optimized algorithm of adaptation of preset time interval based on the current value of the mean count rate. Relative standard deviation, defined as the ratio of the standard deviation of the mean count rate and the mean count rate, is used as the performance criterion for the selection of the optimum parameter values. Optimum values are those that minimize the relative standard deviation.

The methods differ in the strategy of execution of the algorithm for adaptation of the preset time interval. In the first method, the preset time interval adaptation algorithm is executed after the low-pass filters. This allows obtaining shorter preset time intervals for higher stationary rates. The second method executes the adaptation algorithm before the low-pass filters, which enables sensing of the rapid changes of the average rate before fluctuation suppression is carried out.

Both methods were simulated by a self-designed software package and implemented in a newly developed portable rate meter using a standard 8-bit microcontroller.

PROBLEM DESCRIPTION

The stochastic nature of pulse arrivals suggests that the fluctuations from the average number of arrivals from a pulse counter within a fixed time interval (preset time) will also represent a random process. Should the fluctuations follow normal distribution then the standard deviation could represent the appropriate measure of the mean count rate error.

It would be appropriate to find a convenient low-pass filter to suppress the fluctuations of the mean count rate. In order to be able to control the mean count rate error, low-pass signal processing needs to have one or more parameters whose adjustments would keep the fluctuations within defined limits.

The intrinsic property of the preset time algorithms for counting rate meters is the presentation of the mean count rate in fixed time intervals:

shorter time intervals enable faster readouts at the expense of larger fluctuations, while longer time intervals imply slower readouts that correspond to reduced fluctuations. This calls for an optimum solution, and

for stationary or very slowly varying conditions keeping the time interval fixed would yield satisfactory results. However, should the rate change considerably, an adaptable digital signal processing algorithm should enable faster readouts for increased rates. In this case there are two possibilities:

– to execute the adaptation algorithm after the low-pass filters to enable shorter time periods for higher stationary rates, and
– to execute the adaptation algorithm before the filters, which would make it possible to sense the rapid change of mean count rate before the suppression of fluctuations is carried out.

THE STATISTICS OF THE FLUCTUATIONS FROM THE AVERAGE NUMBER OF ARRIVALS

Pulse arrivals from pulse counters, such as GM tubes, represent a random process. Time intervals between adjacent pulses for a Poisson random process are given in [1]:

$$I(t)\ \mathrm{d}t \quad r\,\mathrm{e}^{\ rt}\,\mathrm{d}t \tag{1}$$

where: r is the mean count rate, t is a time interval and $I(t)$ is the distribution function for time intervals between adjacent pulses. The fluctuations of the number of pulses from the mean during preset time interval will form a random process. The fluctuations of the mean count rate can be approximated with Gaussian probability density function. Therefore, the standard deviation is the appropriate measure of the mean count rate error and the relative standard deviation is used as the error estimate of the proposed preset time algorithm.

SIGNAL PROCESSING FOR FLUCTUATIONS SUPPRESSION

The most salient features of count rate meters are the accuracy of calculation of the mean count rate and the response time to the fast changes of the mean count rate. The first requirement concerning accuracy comes down to the need to have one or more rate meter parameters the adjustments of which would keep the fluctuations within defined limits. This calls for very narrow bandwidth low-pass filtering. The second requirement is directly opposed to the first requirement since it calls for a high bandwidth signal processing circuit that is capable of responding with sufficient speed to the mean count rate non-stationarities.

The conciliation of the two opposed requirements makes it necessary to implement an adaptable signal processing method which would provide low and controllable error when the mean count rate stays within certain predefined limits from its true value, and quick response to rapid changes of the mean count rate outside the mentioned limits. The adaptable adjustment of the readout time will follow.

In count rate meters, regardless whether they are using preset time or preset count algorithm to estimate the mean count rate, the most important piece of signal processing circuit is the low-pass filter used to suppress the fluctuations of the mean count rate. Filters, in general, can be described as devices realized in the form of physical hardware or computer software applied to a set of noisy data in order to extract information about a prescribed quantity of interest [9]. In the framework of the radiation level measurements using count rate meters, the quantity of interest is the mean count rate and the random fluctuations of the mean count rate represent the noise to be suppressed using low-pass filtering. Here, the preset time count rate meter was realized using adaptable digital signal processing, and the low-pass filter was realized as a linear time-variant (adaptive) discrete system with FIR and IIR filter was considered for realization.

Adaptable digital filters for the preset time count rate meter

The linear time-invariant discrete system can be described in time domain as [8]:

$$y(n) = \sum_{k=0}^{M} b_k\, x(n-k) - \sum_{k=1}^{N} a_k\, y(n-k) \qquad (2)$$

where: $x(n-k)$ is the filter input delayed by k sampling time intervals, $y(n-k)$ is the filter output delayed by k sampling time intervals, $y(n)$ is the present filter output, b_k the k-th coefficient of the feedforward part of the filter and a_k the k-th coefficient of the feedback part of the filter. The alternative representation in z-domain is given by [8]:

$$H(z) = \frac{Y(z)}{X(z)} = \frac{\sum_{k=0}^{M} b_k z^{-k}}{1 + \sum_{k=1}^{N} a_k z^{-k}} \qquad (3)$$

where $X(z)$ is the z-domain excitation in z-domain, $Y(z)$ is the response in z-domain and $H(z)$ is the filter transfer function in z-domain.

In the case of the FIR filter, the eq. (2) reduces to:

$$y(n) = \sum_{k=0}^{M} b_k\, x(n-k) \qquad (4)$$

and the eq. (3) to:

$$H(z) = \frac{Y(z)}{X(z)} = \sum_{k=0}^{M} b_k z^{-k} \qquad (5)$$

where b_k $(0 \le k \le M)$ are the coefficients and, at the same time, the impulse response of the FIR digital filter, while $M+1$ is the number of coefficients or the length of the filter, which is here the adaptable filter parameter. The direct form FIR filter, also called the tapped delay line, is shown in fig. 1a.

Unlike the FIR filter, the transfer function of the IIR filter (eq. 3), has both the numerator and the denominator, *i. e.* both zeroes (like the FIR filter) and poles (unlike the FIR filter). Here, two types of IIR filters were considered: digital realization of Butterworth approximation of the ideal low-pass filter using single second order section with direct canonical nontransposed structure and digital realization of inverse Chebyshev approximation of the ideal low-pass filter using single second order section with direct canonical nontransposed structure. The direct canonical nontransposed structure is shown in fig. 1b.

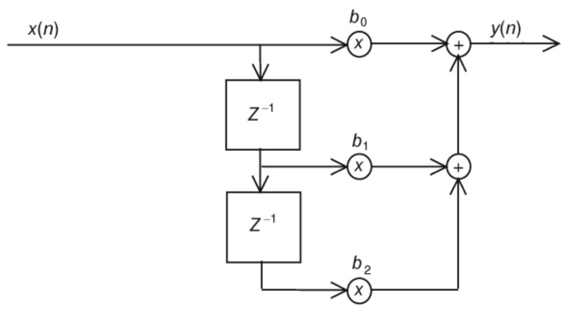

Figure 1a. The direct form FIR filter (tapped delay line) with 3 coefficients

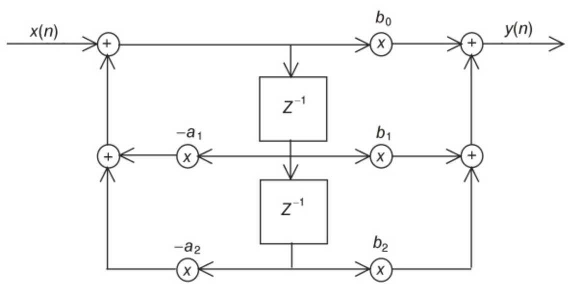

Figure 1b. The direct canonical nontransposed structure of a single-section second order IIR filter

Figures 2 and 3 depict the quantitative performance of three selected filter systems together with the case of no filter. The criterion used for performance evaluations was relative standard deviation. The filters used were: Butterworth second-order single-section IIR filter ($w_{passband}$ = 0.001 rad/sample, $w_{stopband}$ = 0.15 rad/sample, $A_{passband}$ = 1 dB, $A_{stopband}$ = 80 dB), inverse Chebyshev second-order single-section IIR filter ($w_{passband}$ = 0.001 rad/sample, $w_{stopband}$ = 0.1 rad/sample, $A_{passband}$ = 1 dB, $A_{stopband}$ = 80 dB) and FIR (moving average) filter with 20 coefficients equal to unity. As can be seen from fig. 2, the FIR filter performs better than the IIR filters and therefore it is used for further evaluations.

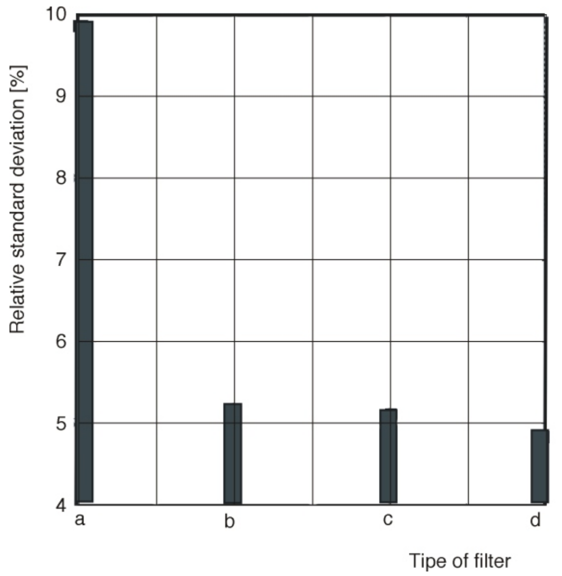

Figure 2. Relative standard deviation for the mean count rate of 10 s^{-1} and preset time of 10 s, for: (a) no filter, (b) Butterworth IIR, (c) Chebyshev type 2 IIR filter, and (d) FIR filter with 20 taps

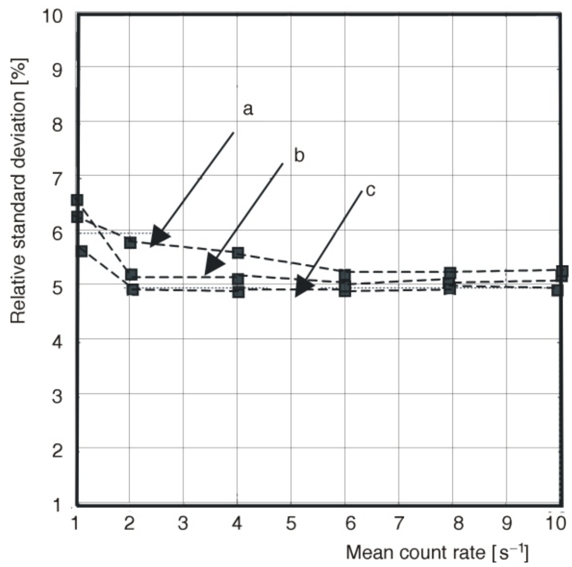

Figure 3. Relative standard deviation against mean count rate for: (a) Butterworth IIR, (b) Chebyshev type 2 IIR filter, and (c) FIR filter with 20 taps

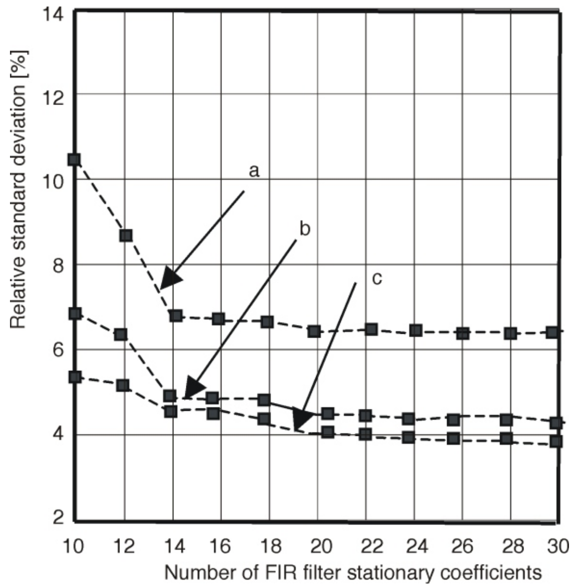

Figure 4. Results of the optimization procedure for the number of FIR filter stationary coefficients for: a) mean count rate = 1 s^{-1}, (b) 2 s^{-1} and (c) 10 s^{-1}

The most significant error reduction of over 2 times compared with the no filter case is achieved by the FIR. Figure 4 shows the results of the optimization procedure for the number of stationary coefficients of the FIR filter. As can be seen from fig. 4, the optimum number of 20 FIR filter stationary coefficients was used for performance comparisons in figs. 2 and 3.

THE REALIZED PRESET TIME METHODS

The realized methods pertain to background radiation level measurements with possible sudden radiation level variations. It is supposed to provide accurate measurements of the mean count rate with the predefined count rate fluctuation of ±5 - ±10% in the stationary conditions, and guaranteed maximum 1 s response time for sudden increase of the count rate up to 100 times above the background level.

The proposed two methods differ in the strategy of execution of the adaptation algorithm for the calculation of the new value of the preset time interval.

The flow chart that describes the signal flow realized by the first method is given in fig. 5.

After start-up and initialization the preset time is set to its default value of 10 s and the error as specified. Then the sequence Poisson random pulses is generated. The sequence is fed to the preset time algorithm and mean count rate is calculated. After that, the algorithm for detection of change of the mean count rate compares the calculated rate with the thresholds obtained by the optimization proce-

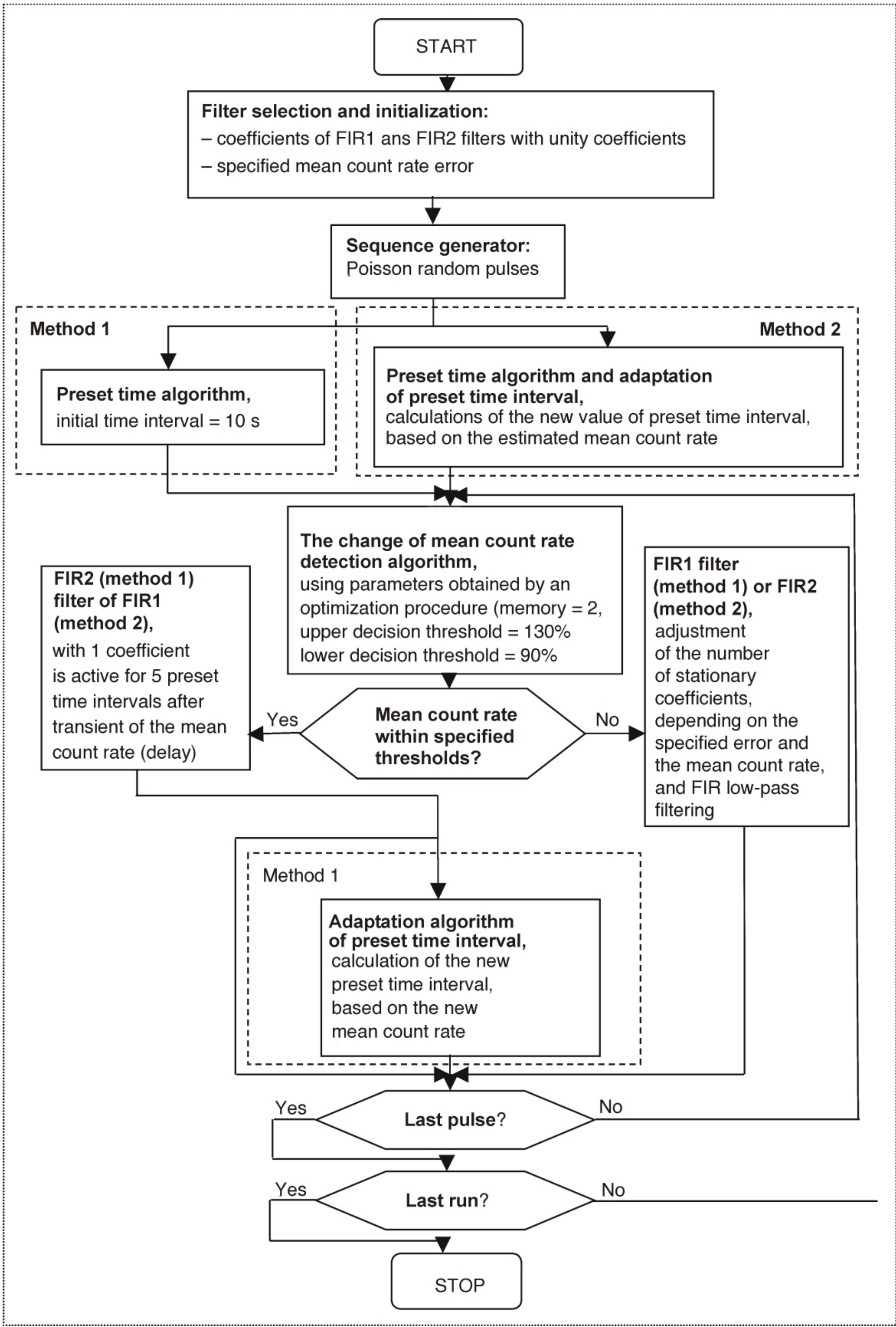

Figure 5. The combined flow chart of the first and second method

dures. To avoid errors caused by statistical nature of the mean count rate, a memory in the detection algorithm has been introduced: the calculated mean count rate is compared with two earlier values. The optimum upper threshold of 130% of the mean count rate, the optimum lower threshold of 90% of the mean count rate, and the optimum number of previous two values were obtained by carrying out an optimization procedure the results of which are shown in figures A1, A2, and A3 in the Appendix. If the change of the mean count rate has not been detected, the FIR filtering for stationary state is performed (FIR filter 1). The number of stationary coefficients of the FIR filter is variable depending on the specified error and the mean count rate. The relationships between the number of FIR filter 1 stationary coefficients and the mean count rate for the considered range of the specified errors is illustrated in fig. 6. Figure 6 shows that for any mean count rate greater or equal to 1 s^{-1}, 20 stationary coefficients are sufficient for the mean count rate error of approximately 4%; if the specified error is increased to 5%, then 15 stationary coefficients are sufficient. The flat portions of the curves are due to the adaptation of the preset time interval with the mean count rate given in fig. 9, where the preset time intervals shorten with the increase of the mean count rate which keeps the average number of pulse arrivals within the preset time approximately constant.

If a change of the mean count rate has been detected, the second FIR filter takes over and performs very little signal processing during transient (FIR filter 2). The number of FIR filter coefficients is only 1 in order to preserve fast response time of the measurement. When the stationary FIR filter 1 with 20 taps takes over the signal processing immediately after the end of the transient, then the mean count rate in the first several preset time intervals would exhibit large fluctuations due to the averaging of count rate values from the previous stationary and transient states, as shown in fig. 7a. However, if a delay of several preset time intervals is introduced in the low-pass filtering in the stationary state, no significant fluctuations are present and the specified error is preserved. Fig. 7b shows the case of a delay of 5 preset time intervals.

Following the signal processing in the transient state, the adaptation algorithm of the preset time interval performs calculations of the new value of preset time interval based on the estimated new value of the mean count rate, according to the curve given in fig. 8.

While the first method executes the adaptation algorithm after the low-pass filters to enable shorter preset time intervals for higher stationary rates, the second method executes the adaptation algorithm before the filters which would allow sensing a rapid change of the mean count rate before fluctuation suppression is carried out. Thus, the most significant change in the proposed adaptable

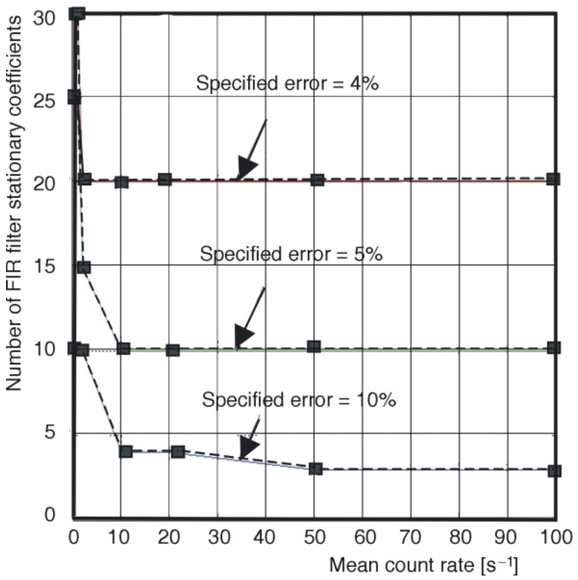

Figure 6. The number of FIR filter stationary coefficients against the mean rate for 3 specified values of the error

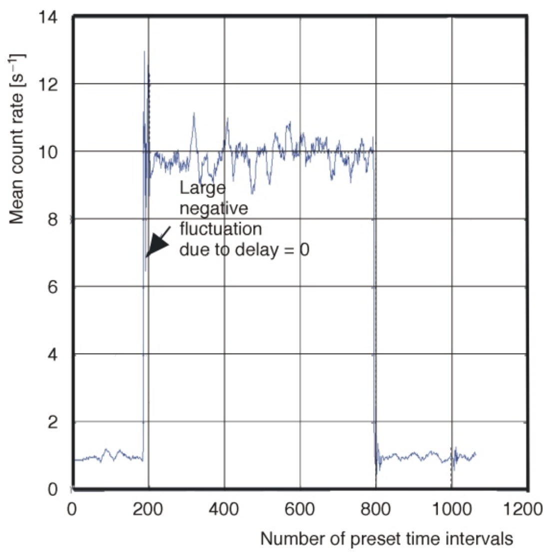

Figure 7a. FIR stationary averaging with no delay

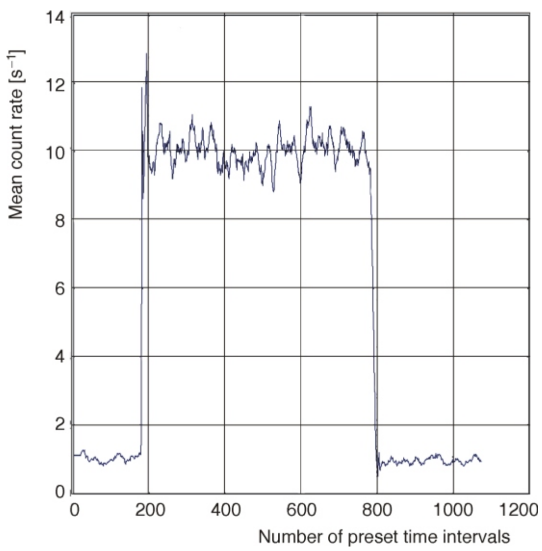

Figure 7b. FIR stationary averaging with a delay of 5 preset time intervals

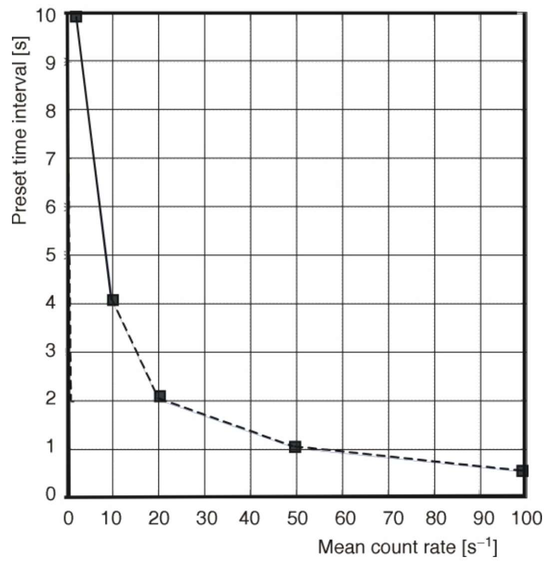

Figure 8. The relation between the preset time interval and the mean count rate implemented in the adaptation algorithm

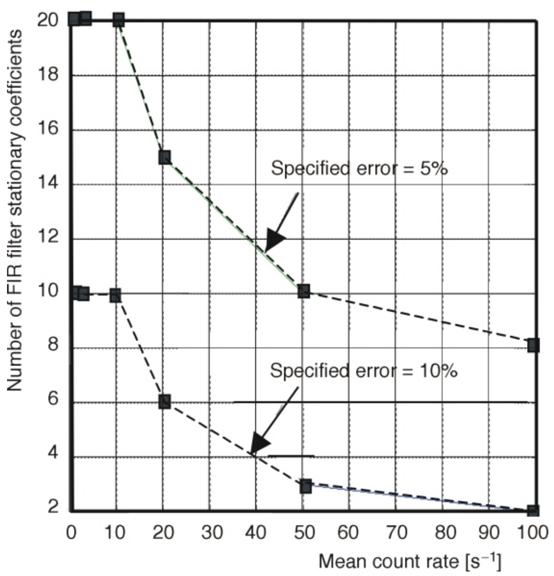

Figure 9. The number of FIR filter stationary coefficients against the mean count rate for 2 considered values of the specified error

digital signal processor is the function block following the generator of Poisson random pulses.

The preset time algorithm and the adaptation of the preset time interval in the second method constitute a unified signal processing block. The adaptation algorithm first sets the lowest value for the initial preset time interval, that is 1 s (Appendix, A4a), which corresponds to the count rates higher or equal to 10 s^{-1}. Since it has been shown that the transient period lasts for maximum 2 preset time intervals, the maximum response time is 2 s. Hence, a swift response to a sudden change of the mean count rate is provided. If the number of pulses arrived at within 1 s is less than an optimized threshold value of 6 counts (Appendix, A5a,b), then the background radiation with typical mean count rates of 1 or 2 s^{-1} are assumed and the preset time interval is set to 10 s (Appendix, A4b). The adaptation algorithm permanently switches between the two mentioned states and sets the preset time interval to 1 s or 10 s depending on the estimated mean count rate.

Again, as in the first method the number of stationary FIR filter coefficients is variable depending on the specified error and the mean count rate. The number of FIR filter stationary coefficients against the mean count rate is given in fig. 9. Figure 9 shows that for any mean rate higher or equal to 1 s^{-1}, 20 stationary coefficients are sufficient for errors of approximately 5% and if the specified error be increased to 10%, then 10 stationary coefficients are sufficient.

CONCLUSIONS

By applying adaptable signal processing tools, two methods are presented for improving the classical preset time rate meters: accurate control of the mean count rate error and adaptable preset time interval. They share an optimized detection algorithm that senses the change of the mean count rate and low-pass filters of FIR structures. FIR filters implement the control of the mean count rate error by suppressing the fluctuations in a controllable way. FIR filters have a variable number of stationary coefficients depending on the specified error and the mean count rate.

The methods differ regarding the execution of the adaptation algorithm for the calculation of the new value of the preset time interval. In the first method this execution takes place after the low-pass filters, which makes it possible to obtain shorter preset time intervals for higher stationary rates. The first method assumes stationary rates pertaining to background radiation levels, sets the initial preset time interval to 10 s, and if the mean count rate is changed, it adjusts the duration of the preset time interval based on the new value of the mean count rate. In the second method the execution takes place before the low-pass filters, which enables sensing of the rapid changes of the mean count rate before the fluctuation suppression is carried out. The fast reacting second method sets the initial preset time to 1 s corresponding to higher count rates, changing to 10 s if background radiation levels are present, taking the first (1 s) or the second (10 s) preset time interval value depending on the estimated mean count rate.

The parameters of the filters and adaptation algorithms that are not variable depending on the specified error and the mean count rate, were fixed to their optimum values after appropriate parameter optimization procedures had been performed.

The specified error and acceptable response time of the rate meter determine the appropriate method, filter structure and filter parameters. The simulated and realized methods with developed algorithms guarantee response time not in excees of 2 s for mean count rates higher than 2 s^{-1} with controllable rate errors between 4% and 10% with the number of FIR filter stationary coefficients not higher than 20.

APPENDIX

Standard optimization procedure for parameters and filters considered in this paper is described using the procedure for selection of the optimum value for the lower threshold of the detection algorithm that senses the change of the mean count rate. The lower decision threshold is varied over the range of values from 70% to 95% of the mean count rate. The lower limit of 70% was chosen because it is symetrical with the selected upper decision threshold of 130% of the mean count rate. In order to preserve the response time (rate of convergence) of the algorithm, values below the 70% were not considered. All other parameters of the algorithm were held fixed. The variation of the lower threshold was carried out for three typical values of the mean count rates: 1, 2, and 10 s^{-1}. The relative standard deviation was used as the performance criterion. The lower decision threshold values of 80% to 90% of the mean count rate were shown to yield minimum relative standard deviation for majority of the considered mean count rates. The results of the optimization procedure are shown in fig. A1. For example, the curve for the mean count rate of 1 s^{-1} (the background radiation level) shows very clearly the range of the optimum values for the lower threshold to be between 80% and 90% of the mean count rate. The value of 90% was selected as the result of the optimization procedure.

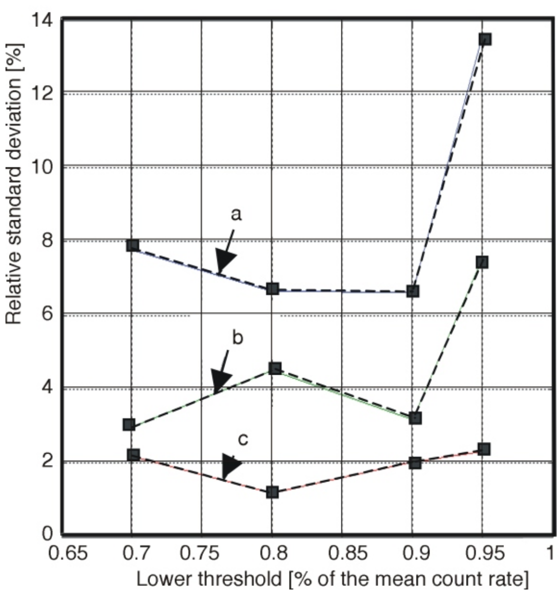

Figure A1. FIR filter relative standard deviation with respect to the lower threshold, for: (a) mean count rate = 1 s^{-1}, (b) 2 s^{-1}, and (c) 10 s^{-1}

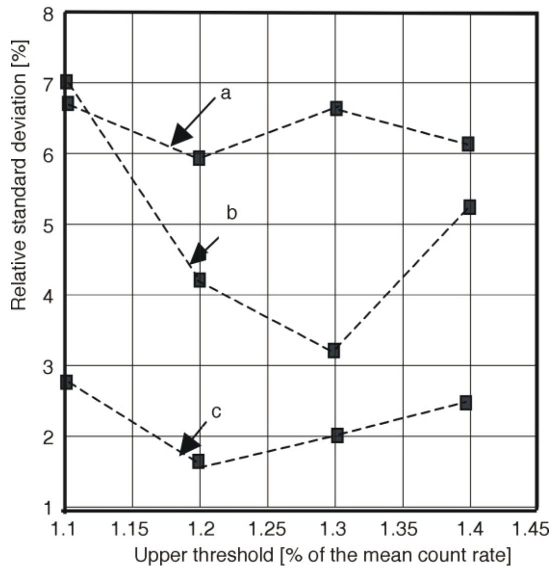

Figure A2. FIR filter relative standard deviation with respect to the upper threshold of the detection algorithm, for: (a) mean count rate = 1 s^{-1}, (b) 2 s^{-1}, and (c) 10 s^{-1}

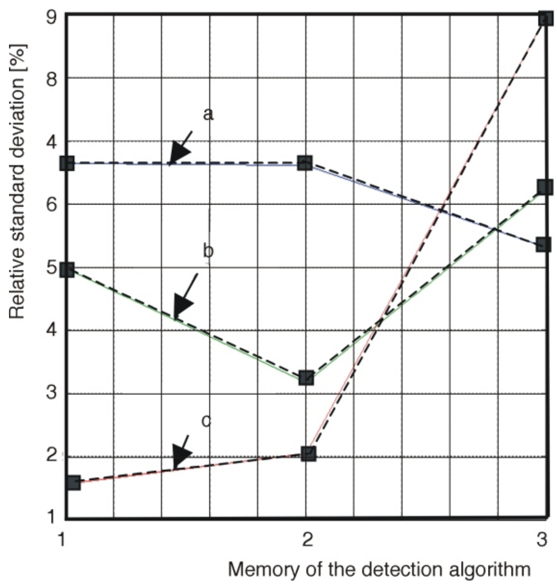

Figure A3. FIR filter relative standard deviation against the memory of the detection algorithm (the number of previous values of the mean count rate), for: (a) mean count rate = 1 s^{-1}, (b) 2 s^{-1}, and (c) 10 s^{-1}

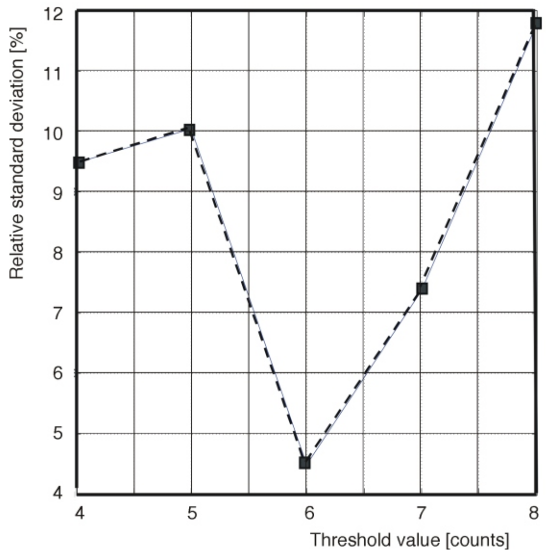

Figure A4a. Relative standard deviation against the threshold value of the adaptation algorithm of the preset time interval. The mean count rate is 10 s^{-1}. The adaptation algorithm is carried out before low-pass filtering

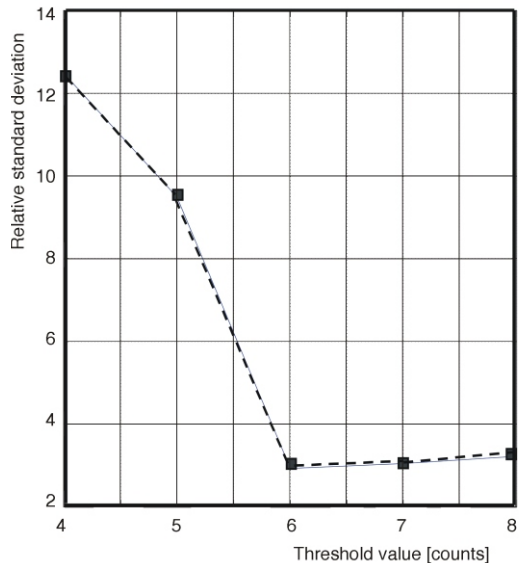

Figure A4b. Relative standard deviation against the threshold value of the adaptation algorithm of the preset time interval. The mean count rate is 1 s^{-1}. The adaptation algorithm is carried out before low-pass filtering

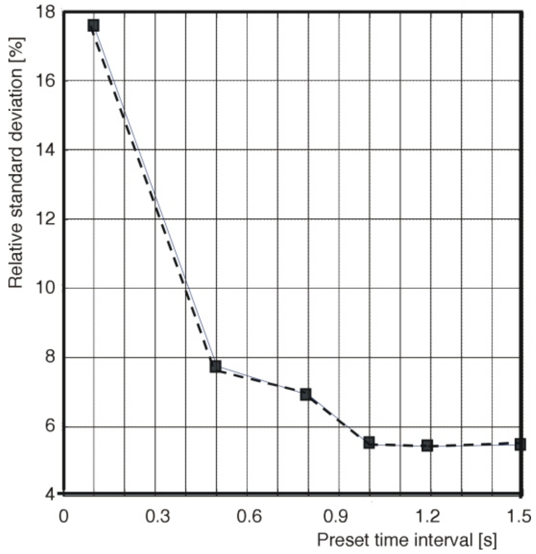

Figure A5a. Relative standard deviation against the preset time interval. The mean count rate is 10 s^{-1}. The adaptation algorithm of the preset time interval is carried out before low-pass filtering

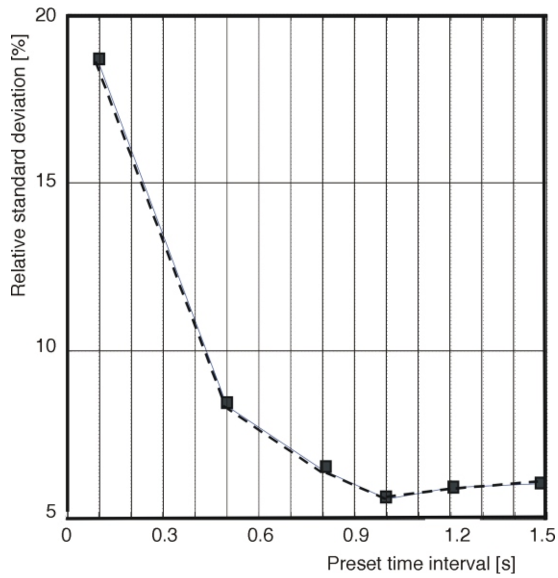

Figure A5b. Relative standard deviation against the preset time interval. The mean count rate is 1 s^{-1}. The adaptation algorithm of the preset time interval is carried out before low-pass filtering

REFERENCES

[1] Knoll, G. F., Radiation Detection and Measurement, John Wiley & Sons, New York, 1999, p. 142

[2] Arandjelović, V., Koturović, A., Vukanović, R., The Dynamic Properties of the Preset Count Digital Rate Meter Algorithms, *Review of Scientific Instruments, 69* (1998), 8 pp. 2902-2908

[3] Arandjelović, V., Koturović, A., Vukanović, A., A Software Method for Supressing Statistical Fluctuations in Preset Count Digital Rate Meter Algorithms, *IEEE Trans. on Nuclear Science, 49* (2002), 5, pp. 2561-2566

[4] Žigić, A., Šaponjić, Dj., Jevtić, N., Radenković, B., Arandjelović, V., HYPERION Network: A Distributed Measurement System for Background Ionizing Radiation Monitoring, *Book of Abstracts*, The First International Meeting on Applied Physics, Badajoz, Spain, October 13–18, 2003, p. 51

[5] Žigić, A., Šaponjić, Dj., Arandjelović, V., Žunić, Z. S., The Concept of a Unified Meteorological and Radiological Monitoring System Based on Hyperion Technology, *Nuclear Technology and Radiation Protection, 19* (2004), 2, pp. 74-76

[6] Arandjelović, A., Koturović, A., The Optimum Dynamic Parameters of Digital Rate Meter Algorithms, *Review of Scientific Instruments, 68* (1997), 12, pp. 4609-4611

[7] Fehlau, P. E., Comparing a Recursive Digital Filter with the Moving-Average and Sequential Probability-Ratio Detection Methods for SNM Portal Monitors, *IEEE Trans. on Nuclear Science, 40* (1993), 2, pp. 143-146

[8] Antoniou, A., Digital Filters: Analysis and Design, McGraw-Hill Inc., New York, 1979

[9] Haykin, S., Adaptive Filter Theory. Prentice-Hall, Englewood Cliffs, NJ, USA, 1986

Александар Д. ЖИГИЋ

МЕРАЧ СРЕДЊЕГ ОДБРОЈА НА БАЗИ ПРЕДОДРЕЂЕНОГ ВРЕМЕНА КОРИШЋЕЊЕМ АДАПТИВНЕ ДИГИТАЛНЕ ОБРАДЕ СИГНАЛА

У овом раду су представљене две методе које користе адаптивну дигиталну обраду сигнала за побољшање карактеристика класичног мерача средњег одброја на бази предодређеног времена. Обе методе користе оптимизовани алгоритам за детекцију промене средњег одброја. У раду су разматрана три типа ниско-пропусних филтара различитих структура који реализују функцију контроле грешке средњег одброја, потискујући флуктуације средњег одброја на контролисани начин. Од три разматрана, изабран је један ниско-пропусни филтар који је коришћен код обе методе.

Прва метода користи адаптациони алгоритам за израчунавање нове вредности предодређеног времена који се извршава после ниско-пропусног филтрирања. Овај алгоритам омогућава да се добију краћи интервали предодређеног времена за веће стационарне средње одброје.

Друга метода користи алгоритам за израчунавање нове вредности предодређеног времена који се извршава пре ниско-пропусног филтрирања. Овај алгоритам омогућава детекцију брзе промене средњег одброја пре потискивања флуктуације средњег одброја од стране ниско-пропусног филтра.

Неки параметри реализованих метода су фиксни и добијени су као резултати извршавања одговарајућих оптимизационих процедура. Ниско-пропусни филтар има променљив број стационарних коефицијената у зависности од захтеване грешке и вредности средњег одброја. Ниско-пропусни филтар реализује функцију контроле грешке потискујући флуктуације средњег одброја на контролисани начин.

Развијене и реализоване методе побољшања карактеристика средњег одброја, које користе развијене алгоритме, гарантују: временски одзив на промене средњег одброја који није дужи од 2 секунде за средње одброје веће од 2 импулса по секунди и контролисану грешку средњег одброја која је у границама од 4% до 10%.

Кључне речи: мерач средњег одброја на бази предодређеног времена, адаптивна обрада сигнала, ниско-пропусни филтар, ФИР филтар, брзина средњег одброја