Ministry of Education and Science of Ukraine

# ODESA NATIONAL UNIVERSITY OF TECHNOLOGY

International Competition of Student Scientific Works

# BLACK SEA SCIENCE 2023

# PROCEEDINGS

ODESA, ONUT 2023

Ministry of Education and Science of Ukraine

Odesa National University of Technology

International Competition of Student Scientific Works

# BLACK SEA SCIENCE 2023

## Proceedings

Odesa, ONUT
2023

# DETERMINATION OF INTERVALS OF DISCRETIZATION OF TIME SERIES OF MEASUREMENTS OF TECHNOLOGICAL PROCESS PARAMETERS IN ASK TP

**Authors:** Chychkan Alina
Tkachenko Karyna
**Advisor:** Abramenko Ivan
State University of Biotechnology (Ukraine)

***Abstract.*** *At the current stage of development of the rural economy of our country, its products are not fully competitive. One of the reasons for this is the high cost of production. Automation of agricultural production increases the reliability and extends the service life of equipment, facilitates and improves working conditions, improves labor safety, reduces the use of labor and economic costs, and increases the quantity and quality of products.*

*Therefore, the improvement of the existing methods of obtaining information for the purposes of management by substantiationdetermination of sampling intervalstime series of technological process parameter measurements is an important and urgent task.*

***The goal of the work****. The purpose of this work is to increase the efficiency of automation systems by technological processes based on the use of modern computer technologies.*

***General characteristics of work.*** *A comparative evaluation of the effectiveness of using known techniques for determining the discretization interval of a time series of technological process parameter measurements in automation systems was performed.A new method for determining the sampling interval for automation systems is proposed, which is based on the use of the discrete Fourier transform for a smoothed digital signal and filtered noise and the Kotelnikov calculation theorem. Obtained calculation formulas.*

***Key words:*** *measurement channel, Kotelnikov's theorem, Fourier transform, sampling interval, numerical filtering.*

## I. **INTRODUCTION**

Any reasonable type of human activity is based on information about the properties of the state and behavior of that part of the real world with which this activity is connected.

Information in the technical system arises in the process of production and management. In management, information plays a decisive role, developing and changing under the influence of economic factors, scientific and technological progress, the external environment, production needs, and the organization of the management process.

When using information, a very important characteristic is its adequacy, which means the degree of correspondence of the image created with the help of the received information to a real object, process, phenomenon. In technical systems, the correctness of decision-making depends on the degree of adequacy of information.

In the production process, unacceptable deviations of technological parameters from the specified (normal) values may occur, caused by various reasons. Manufacturing facilities that provide the technological process cannot always eliminate these deviations on their own. Therefore, they are equipped with appropriate control automation systems, which ensure the normal flow of the technological process.

Automation of various types of production is an important direction of scientific and technical development of society. It ensures an increase in labor productivity, the release of a person from the production process, and an increase in the quality of products [1].

Therefore, the improvement of the existing methods of obtaining information through justificationdetermination of sampling intervalstime series of measurements of technological process parameters in ASK TP is an important and urgent task.

## II. THEORETICAL BASICS

### 2.1. Measurementtechnological parameters in ASK TP

The measurement of technological parameters in the ASK TP is implemented with the help of measuring channels.

The measuring channel is a structurally or functionally separated part of the information system, which performs a sequence of operations from the perception of the measured quantity to obtaining the result of its measurements, which is displayed by a number or a code corresponding to it.

A typical structure of the measuring channel is presented in fig. 1 [2, 3].


Fig. 1. Typical structure of the measuring channel

Here $x(t)$ - measured parameter a $y(t)$ - measurement information signal.

Primary transducers (sensors) are measuring devices that perceive the controlled parameter and convert it into a signal (usually electrical) suitable for transmission over communication channels, further conversion, processing and storage, which is not directly perceptible by the observer.

If the sensor signal parameters do not agree with the analog-to-digital converter (ADC) input parameters or do not meet the standard (for example, the ADC input is a voltage in the range $0 \div 10$ B, and the sensor (thermocouple) has an output voltage in the range from 0 to 100 mV), then a normalizing converter is used, which provides normalization of the sensor signal (bringing it to unified change ranges), galvanic isolation of the output circuits of the primary converters and the input circuits of the input module the controller.

A unified signal is a signal of remote transmission of information with unified parameters that provides an interface between units, devices and installations.

Unified values of the measuring signal at the ADC input $z(t)$ with the linear static characteristic of the normalizing converter are determined by the ratio [4]:

$$y_{\text{ун}}(t) = \frac{y(t)(Y_{\text{ун,max}} - Y_{\text{ун,min}}) + Y_{\text{ун,min}}\, y_{\max} - Y_{\text{ун,max}}\, y_{\min}}{y_{\max} - y_{\min}},$$

where: $y(t)$ - current value of the measuring signal; $Y_{\text{ун,min}}, Y_{\text{ун,max}}$ - the minimum and maximum value of the unified signal; $y_{\min}$, $y_{\max}$ - the minimum and maximum value of the measuring signal.

The structure of the measuring channel is determined by the measuring task it implements and the general methodology of building the system. In general, the measuring and informational parts can be distinguished in the channel. The measuring part includes: primary measuring transducer, normalizing transducers, switches. All elements of the measuring part have standardized metrological characteristics, and directly affect the total error of the channel. The information part includes ADCs, databases, software components, etc. Measurement information in the information part of the channel is presented in digital form [2, 3, 4].

The components of the measuring part are included in the signal $y(t)$ a certain fallibility $\varepsilon(t)$.

Signals of measurement channels of technological parameters are functions of time $z = f(t)$. Polling period of sensors of analog variables $\Delta t$ depends on the dynamic characteristics of the signal and the measuring channel. This problem is solved taking into account two contradictory considerations. Magnification $\Delta t$, on the one hand, reduces the load on the processor, and on the other hand, the error of determining the values increases $\overline{x}_i(i\Delta t)$.

### 2.2. Existing methods of determining the survey period of analog variables

Time discretization step $\Delta t$ can be determined by the Kotelnikov calculation theorem (in the English-language literature - the Nyquist-Shannon theorem) [5], whichsays that the analog signal $z(t)$ ,whathas a limited frequency spectrum, can be unambiguously and losslessly restored in its own waycalculations, taken with the sampling frequency $f_{\text{д}}$, which should be at least twice the maximum frequency of the spectrum $f_{\max}$:

$$f_Д \geq 2 f_{max} .$$

Then the value $\Delta t$ it is necessary to choose no more than half a period of the maximum frequency of the spectrum $f_{max}$

$$\Delta t \leq 1/(2 f_{max}) .$$

Implementation length $T$ at the same time is

$$T = (N-1)/f_д ,$$

and the frequency of the first harmonic is in hertz

$$f_1 = f_Д/(N\text{-}1) .$$

However, the subtraction theorem is applicable to the case where the signal started infinitely long ago and never ends, and has no breakpoints. Real signals do not meet these conditions because they are finite in time and usually have discontinuities in time realization. Accordingly, the width of their spectrum is infinite. In this case, full recovery of the signal is impossible [6, 7].

Therefore, in each specific case, it is necessary to find a compromise solution based on the nature of the signal, the required accuracy of its restoration, the characteristics of the applied smoothing filter, and other factors. All this leads to the fact that in real automation systems, the sampling frequency is empirically selected in $k$ times higher than the frequency $2 f_{max}$ :

$$f_Д = k \cdot 2 f_{max} /$$

Values are usually used $k$ in the range of 2-5, but clear criteria for determining the value $k$ currently does not exist for a specific time series in literary sources.Given that the time series of real technological parameters have the maximum frequency of the spectrum $f_{max} \leq 0{,}5$ Hz,value $\Delta t$ with this approach varies within 0.04-0.5 s, and the number of counts per period of maximum frequency within 2-250 [2, 6].

Determination of the acceptable value of the sensor polling interval is also based on the following considerations.

Sinceelements of the information and measurement channel have inertia, which is characterizedconstant time $T$, theyislow-frequency signal filters.going outfrom this, inmeasuring channelsthe element with the maximum value is selectedconstant timeand,reducingher at $5 \div 10$time,determinethe assumed value of the polling period of the sensors from the ratio $\Delta t \leq T_{max} / (5 \div 10)$ [2, 3].

Determination of the value of the acceptable period of survey of the sensors is also carried out on the basis of the autocorrelation function $R_{\bar{x}}(\tau)$ [7] (Fig. 2).

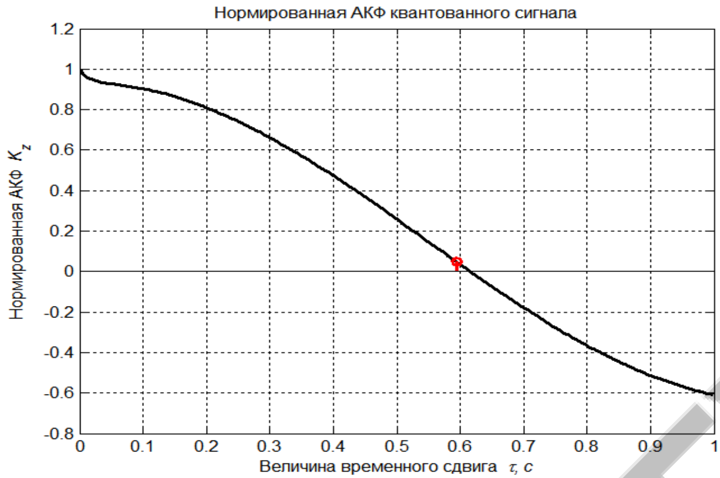



Fig. 2. Normalized autocorrelation function of the measurement signal

For a number series $\overline{x}_1, \overline{x}_2, \ldots, \overline{x}_n$ with known mathematical expectations $M_{\overline{x}}$ and dispersion $D_{\overline{x}}$ the autocorrelation function is defined by the formula

$$R_{\overline{x}}(k) = \frac{1}{(N-1)D_{\overline{x}}} \sum_{i=1}^{N-k} (\overline{x}_i - M_{\overline{x}})(\overline{x}_{i+k} - M_{\overline{x}})$$

To define $\Delta t$ in this case, the ratio is used

$$\Delta t \le 4\tau_{cn} D_{\overline{x}-x} / D_x$$

where $\tau_{cn}$ is the decay time of the autocorrelation function, that is, the time it takes for the autocorrelation function to reach its values $R_{\overline{x}}(\tau_{cn}) \le 0{,}05 R_{\overline{x}}(0)$; $D_{\overline{x}-x}$ - dispersion of measurement error; $D_x$ - dispersion of the measured signal.

Minimum value $\tau_{cn}$ is determined by the implementation of a random process (Fig. 3) according to the formula:

$$\tau_{cn,\min} = \tau_p / N_{\text{пер}}$$

where $N_{\text{пер}}$ – the number of crossings by the signal of the line of mathematical expectation in the time interval $\tau_p$; $\tau_p$ is the duration of the implementation of the random process, which is chosen so that $N_{\text{пер}} \ge 100$ [2, 4].

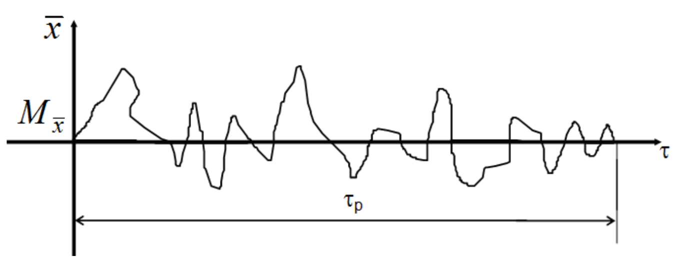



Fig. 3. Implementation of the measuring signal

## III. JUSTIFICATION OF THE DETERMINATION OF DISCRETIZATION INTERVALS OF THE TIME SERIES OF THE MEASUREMENTS OF THE TECHNOLOGICAL PROCESS PARAMETERS

### 3.1. Definition of a mathematical apparatus

All of the above approaches to determining the sensor polling interval are not unambiguous. Therefore, the development of a more comprehensive approach is an urgent task.

Signal $i$-th dimension $z_i(t)$ is comingin the corresponding controller modules, where it is converted into digital form. However, the information obtained in this way is not suitable for use in control tasks due to the presence of interference, and therefore it is necessary to perform its preliminary processing, which is called primary.

The list of stages of this processing is determined by the sequence of transformations that the measurement information signal undergoes in the ADC, a typical algorithm of which is presented in Fig. 4 (see appendix) [3, 9].

Most signals in automation systems are analog in nature. They change continuously over time and can take any values within a certain range. It is impossible to enter such a signal into a computer and process it, because it has an infinite number of values at any time interval. Therefore, in digital processing systems, the signal is presented in discrete digital form.

A digital signal can be obtained from an analog signal by time sampling, level quantization, and encoding. The values of a digital signal are called counts.

In practice, the method of representing a signal by a sequence in which readings are located at equal time intervals has become widespread $\Delta t$ .

In this case, the signal is determined by the sequence of its values $z(0), z(\Delta t), z(2\Delta t), \ldots$ . This sequence is called a discrete sequence $z_i(i\Delta t)$ .

Interval $\Delta t$ called the sampling period, or the sampling interval. The inverse of the sampling period is called the sampling frequency, or the sampling frequency $f_{Д}$ .

The conversion of an analog signal into a discrete sequence of values is called time sampling.

It is obvious that the representation of the signal by a discrete sequence of readings leads to the loss of information about the behavior of the signal in the intervals between readings. To minimize these losses, the sampling period $\Delta t$ must be reduced. However, reducing the discreteness period leads to an increase in the number of deductions and, as a result, to an increase in the volume of calculations. Therefore, when choosing a discretization period, you have to look for a compromise solution.

Operations of time discretization and level quantization are performed in ADC analog-to-digital converters. Elements of a digital sequence $z_{Ц}(i)$ can take only a number of discrete values $z_{ц,1}, z_{ц,2}, \ldots, z_{ц,N-1}$, the number of which depends on the number of digits used.

Thus, signal quantization is associated with the occurrence of an additional error.

To determine the sampling intervals of the measurement signal, we will use its frequency representation.

The frequency representation can be obtained by using the methods of harmonic analysis - a section of mathematical analysis in which the properties of periodic non-sinusoidal functions are studied using their representation in the form of Fourier series or integrals [7, 8].

In practical applications, we deal with finite samples of values. For the purposes of spectral analysis of a discrete signal, the continuous Fourier transform cannot be used, since it is intended for continuous signals. In this case, the discrete Fourier transform of the DFT is used.

The DFT is performed on the finite sequence $N$ of calculations or over a periodic sequence in which the period consists of $N$ calculations Assume that the signal is sampled at equal time intervals $\Delta t$, and as a result, a finite discrete sequence is obtained:

$$z_i = z(i\Delta t) = z(0), z(\Delta t), z(2\Delta t), \ldots, z[(N-1)\Delta t], i = 0,1,\ldots, N-1.$$

DFT sequence $z(i\Delta t)$ is defined as a discrete sequence $Z_k$ from $N$ calculations in the frequency domain:

$$Z_k = Z(kf_1) = Z(0), Z(\mathrm{f}_1), Z(2f_1), \ldots, Z[(N-1)f_1], k = 0,1,\ldots, N-1,$$

where $f_1$ - the frequency of the first harmonic, which is calculated from the expression

$$f_1 = 1/[(N-1)\Delta t].$$

Sequence $Z_k$ in trigonometric form is determined by the ratio:

$$Z_k = \frac{1}{N-1}\sum_{i=0}^{N-1} z_i e^{-jk\frac{2\pi}{N-1}i} = \frac{1}{N-1}\sum_{i=0}^{N-1} z_i \left[\cos\frac{k2\pi i}{N-1} - j\sin\frac{k2\pi i}{N-1}\right].$$

According to the last relation, the meaning of the DPF is that the original non-periodic function of an arbitrary shape is represented as a set of sines or cosines with different frequency and amplitude. In other words, a complex function will turn into many simpler ones.

Each sinusoid (or cosine) with a certain frequency and amplitude obtained as a result of the DFT is called a spectral component or harmonic. The spectral components form the Fourier spectrum.

Visually, the Fourier spectrum is presented in the form of a graph on which the frequency is plotted along the horizontal axis $f$, and vertically – the amplitude of the spectral components (Fig. 5).

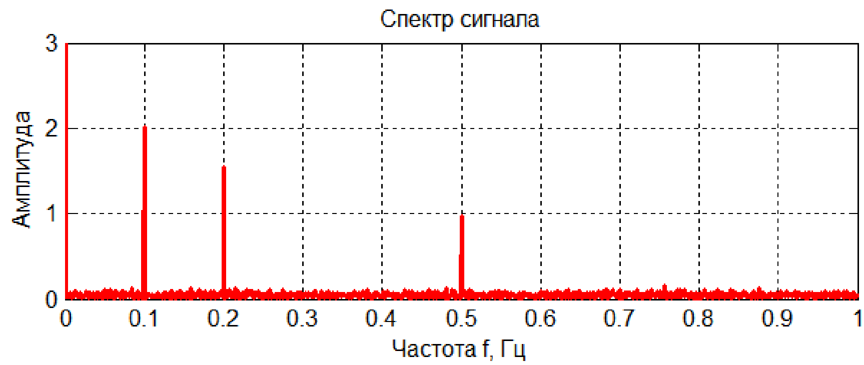



Fig. 5. Frequency spectrum of the signal

Then each spectral component can be represented as a reference, the horizontal position of which corresponds to its frequency, and the height to its amplitude. A harmonic with zero frequency is called a constant component (in time representation it is a straight line).

The spectrum of the DFT signal in practice is determined only for positive frequencies in the range from 0 to $f_Д$.

Frequency $f_Н = f_Д/2$ received the name Nyquist frequency. Nyquist's higher frequency components are a mirror image of the lower frequencies.

Therefore, when analyzing signals, it is advisable to analyze the spectrum only in the range from 0 to $f_Н$, by increasing the modules of the DPF bins by 2 times.

For the correct use of DPF, the following two important features should always be remembered:

- to expand the band of spectral analysis, you need to increase $f_Д$, i.e. take time-based bills more often;
- in order to improve the frequency resolution without changing the bandwidth of the spectral analysis, it is necessary to increase $N$, that is, to analyze a longer sample of the signal.

There is no single correct spectrum of any signal. The DFT decomposes the signal not into the harmonics of the signal, but into its own harmonics.

Strictly speaking, the decomposition result cannot be interpreted as a signal spectrum. Decomposition is simply data, applied to which the inverse transformation can be applied to obtain an output signal. However, with a sufficiently small value of the frequency spectrum discretization step, the spectrum graph displays the frequencies of the output signal adequately.

To ensure functioning on a real-time scale, the full calculation of the DFT must be performed in the time interval corresponding to the accumulation of one data packet. It is assumed that while the DFT calculation of the current data packet is being performed, the PLC processor is accumulating data for the next packet.

### 3.2. Definition of the filtering operation algorithm

When defining the time step of discretization $\Delta t$ the filtering operation that is should also be taken into accountis the most complex of the operations of primary signal processing of measurement information, and its efficiency is determined by the value $\Delta t$ .

In general, filtering can be performed in hardware using RC or LC filters before feeding the signal to the controller, or in software using various filtering algorithms. Exponential smoothing algorithms have become the most widely used in ASK TP (about 90% of filtering tasks are solved with the help of these algorithms) [2, 9].

Algorithms of exponential smoothing belong to the class of adaptive smoothing methods, the main characteristic of which is the ability to continuously take into account the evolution of the dynamic characteristics of the measured parameter, adjust to this dynamic, giving, in particular, the greater weight and the higher the information value to the measurements that are closer to the current moment in time. They allow you to update the smoothing results with minimal delay and with the help of relatively simple mathematical procedures. The advantages of exponential filtering algorithms include the low complexity of calculations and the small amount of controller memory required for data storage.

Two such algorithms are usually used in PLCs - the algorithm of simple exponential smoothing, which is used in the absence of a certain tendency of changes in indicators (the so-called trend) in the time series of measurements, i.e. for stationary or quasi-stationary time series, and the Holt algorithm in the presence of a trend (non-stationary time rows).

The simple exponential smoothing algorithm (Brown's method) is defined by the recurrence relation of the following form

$$\hat{z}_i = \hat{z}_{i-1} + \alpha\left(z_{i-1} - \hat{z}_{i-1}\right),$$

where $\hat{z}_i$ and $\hat{z}_{i-1}$ – current and previous value of the smoothed signal; $z_{i-1}$ - previous value of the measured signal; $\alpha$ is the smoothing coefficient, which is chosen a priori ( $0 < \alpha < 1$ ).

Thus, the current value of the smoothed signal is determined as the sum of the previous smoothed value and corrections for the error of the previous value of the signal measurement. The weight of the correction determines how "sharply" the algorithm will react to changes.

The closer the value $\alpha$ to 1, the more the influence of the last measurement values is taken into account. If $\alpha$ closer to 0, the weights by which the measurement values are weighted decrease slowly, that is, a larger number of previous measurements are taken into account.

Of course the value $\alpha$ is within $0{,}05 \div 0{,}3$ . A value of 0.5 is almost never exceeded.

It is recommended to set a value for a slowly changing time series $\alpha$ in the range $0{,}05 \div 0{,}2$, and for rapidly changing - in the range $0{,}3 \div 0{,}5$.

Simple exponential smoothing of time series containing a trend leads to a systematic error associated with the lag of the smoothed values from the actual levels of the time series.

To take into account the linear trend in non-stationary series, a special two-parameter exponential smoothing using the Holt method is used). If there is an upward or downward trend in the time series, then along with the assessment of the current level, an assessment of the slope is also necessary. In the Holt method, both the time series and the trend are simultaneously smoothed using two different parameters.

The two-parameter smoothing method includes two equations. The first is intended for smoothing a series of measured values, and the second - for smoothing the trend

$$\hat{z}_i = \alpha_1 z_i + (1-\alpha_1)(\hat{z}_{i-1} + T_{i-1})$$
$$T_i = \alpha_2(\hat{z}_i - \hat{z}_{i-1}) + (1-\alpha_2)T_{i-1},$$

where $\hat{z}_i$ and $\hat{z}_{i-1}$ - current and previous value of the smoothed signal; $z_i$ - current value of the measured signal; $T_i$ and $T_{i-1}$ - current and previous value of the trend; $\alpha_1$ and $\alpha_2$ – time series and trend smoothing coefficients, the values of which are selected a priori in the range $0 \div 1$.

The adequacy of the smoothed time series can be most easily determined by calculating the average relative error of approximation [8, 9]:

$$\bar{\Delta}_{\text{ап}} = \frac{100 \cdot N}{\sum_{i=1}^{N} z_i} \sqrt{\frac{\sum_{i=1}^{N}(z_i - \hat{z}_i)^2}{N}},$$

where $N$ - the number of measurement points stored in the PLC memory.

Approximation error within $5 \div 7$ % indicates a good fit of the model to the original data. Permissible limit of values $\bar{\Delta}_{\text{ап}}$ - not more $8 \div 10$ % (sometimes up to 15 %).

### 3.3. The essence of the developed methodology

The technique is based on the simultaneous use of DFT for a smoothed digital signal and filtered noise.

To determine the time step of the discretization $\Delta t$ is suggested by comparing the harmonic amplitudesmaximum spectrum frequency $f_{\max}$ choose the optimal value in different components of the signaldiscretization step $\Delta t$ for a specific measured parameter.

Considering the standard accuracy of automatic control systems, it can be assumed that the share of this amplitude in the filtered noise $A_{\max,\text{шум}}$ should be no more than 5% of the amplitude of the smoothed signal $A_{\max,\text{згл}}$, i.e $100\, A_{\max,\text{шум}} / A_{\max,\text{згл}} \le 5$ %.

## IV. PRACTICAL IMPLEMENTATION OF THE METHODOLOGY

To test the proposed approach and compare it with known approaches, a simulation model of the measuring channel and ADC was developed using the Matlab program (appendix).

Temporal number accurate measurementsparameterwas formed from 3 harmonicsonformula

$$y(t) = y_0 + a_1 \sin(2\pi f_1 t) + a_2 \sin(2\pi f_2 t) + a_3 \sin(2\pi f_3 t)$$

where $a_1, a_2, a_3$, $рад/с$ and $f_1, f_2, f_3$, Hz - amplitudes and frequencies of harmonics that are randomare determined from acceptable lists of values $0,8 \div 1,9$ and $0.05 \div 0.5$ Hz.

The measurement information signal at the input of the ADC was formed according to the formula $z(t) = y_{ун}(t) + \varepsilon(t)$ where $\varepsilon(t)$ - the error of the measuring channel, distributed according to the normal law of probability distribution. Noise filtering was performed using a simple exponential smoothing method.

The frequency representation of the signals of the measuring channel is shown in Fig. 6 (see appendix).

Dependencies of the fate of amplitudes of signal harmonics in noise fromvaluesdiscretization step $\Delta t$ for different harmonics are shown in fig. 7.

The processing of the obtained results made it possible to obtain a formula for an unambiguous and reasonable determination of the step size $\Delta t$ by known value $f_{\max}$ (Fig. 8, and see the appendix)

$$\Delta t = 0,0045 + 0,138 \cdot e^{-17,14 f_{\max}},$$

or the formula for determining the coefficient $k$ for Kotelnikov's theorem in the form of notation $f_{Д} = k \cdot 2 f_{\max}$ (Fig. 8, b, see the appendix)

$$k = 208,3\left(1 + 0,33 \cdot e^{-125 f_{\max}} - 1,33 \cdot e^{-31,25 f_{\max}}\right),$$

which corresponds to the specified error of 5% (with other values of the specified error, similar dependencies can be obtained)

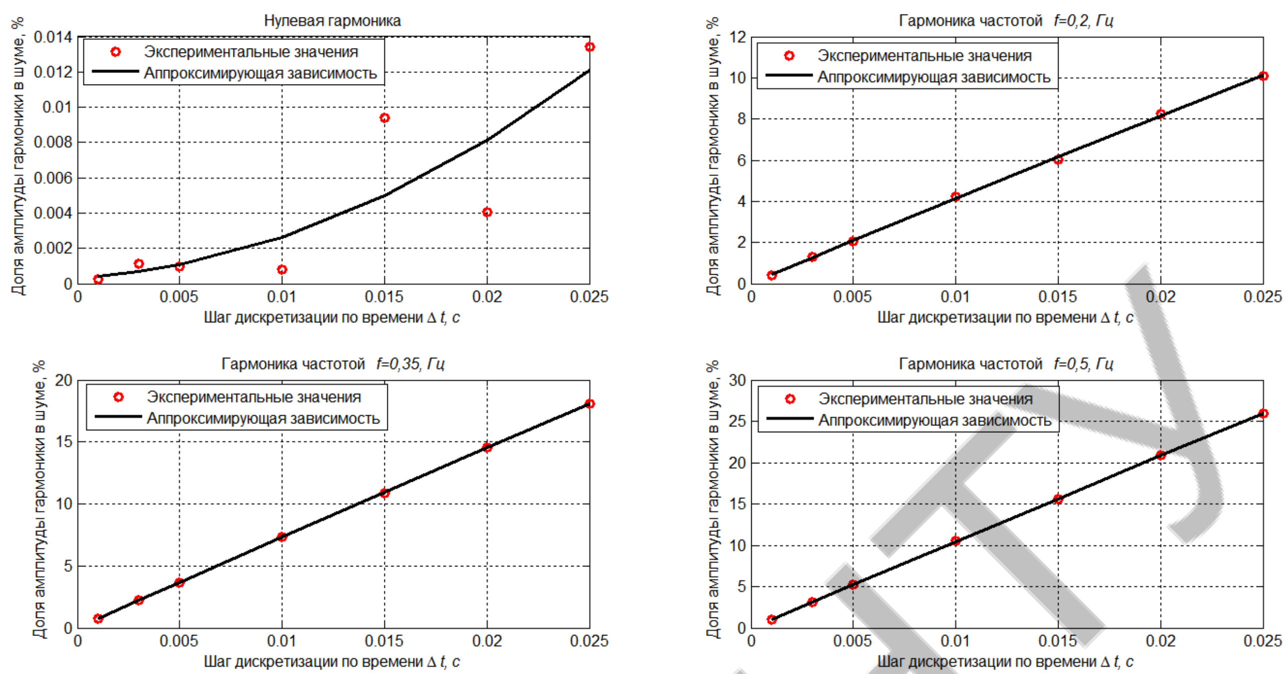



Fig. 7. Dependencies of the fate of amplitudes of signal harmonics in noisefroma step $\Delta t$

The analysis of the obtained results shows that the value $\Delta t$ at $f_{\max} = 0{,}05 \div 0{,}5$ Hzvaries depending on $f_{\max}$ within 0.005-0.063 s, and the number of counts per period of maximum frequency - within 300-406.

The obtained results can be used to measure any real parameter and differ significantly from the recommendations of known approaches.

Therefore, to compare the accuracy of restoration of the discretized signal, it was established that at the maximum value $\Delta t = 0{,}1$c, which is determined by Kotelnikov's theorem, harmonics with frequencies of 0.1, 0.2, 0.3, 0.4, and 0.5 Hz are distinguished with errors of 20.6, 41, 63, 83, and 100%, respectively. At the minimum value $\Delta t = 0{,}04$ c, which is determined by Kotelnikov's theorem, these same harmonics are distinguished with errors of 8.3, 16.7, 25, 33.4, and 41.8%, respectively, which is unacceptable.

To evaluate the effectiveness of the filtering operation, a dependency was also determinedaverage relative error of approximation $\bar{\Delta}_{\text{ап}} = f(\Delta t)$, where $\Delta t$ were calculated according to formula (1) (Fig. 9).

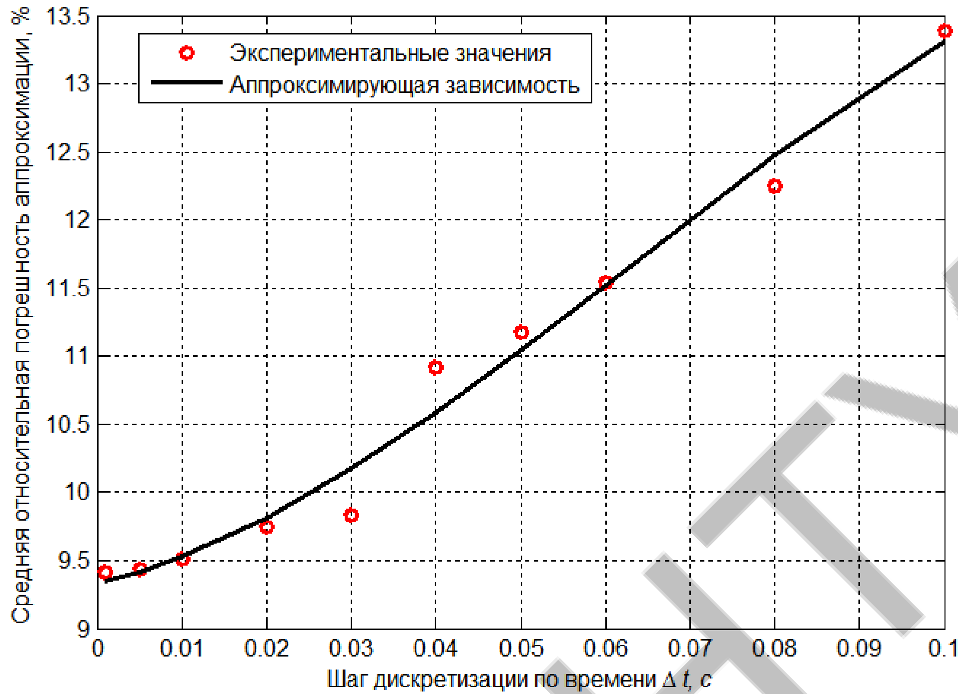



Fig. 9. Dependencyaverage relative filtering error fromdiscretization step $\Delta t$

In the case of using the autocorrelation function of the signal, a significant ambiguity in the selection of the value was found $\Delta t$ with this approach (Fig. 10) and a significant overestimation of its values.

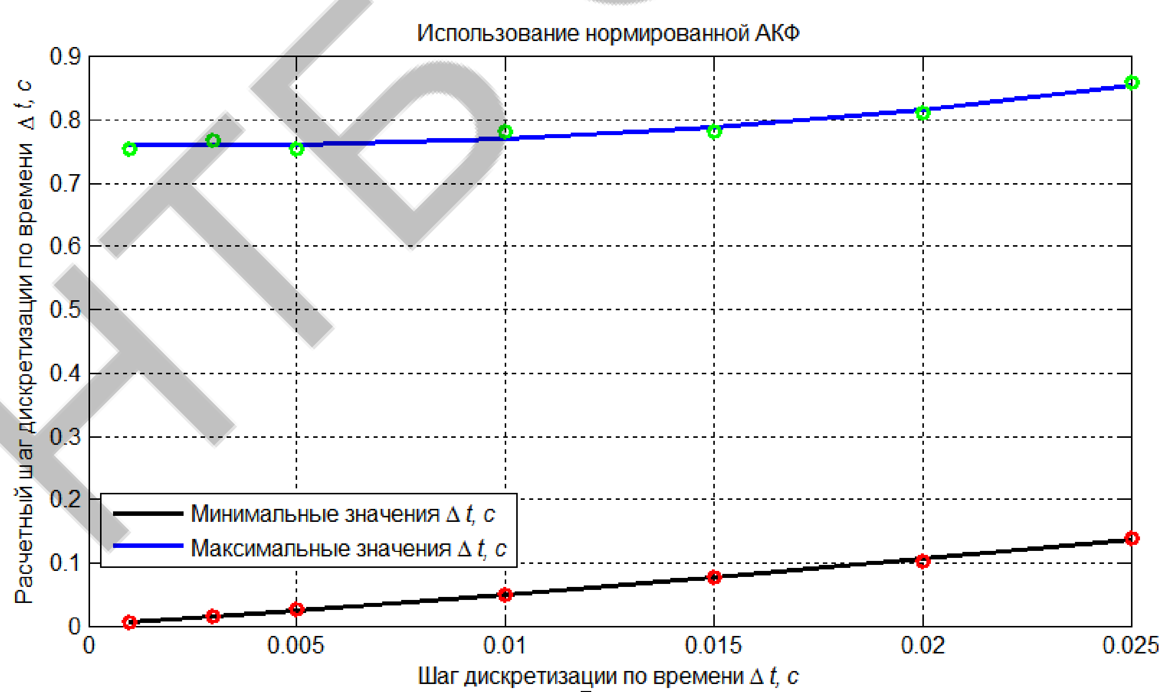



Fig.10. Definition of the sampling step $\Delta t$ by the autocorrelation function

## V. CONCLUSIONS

A new technique for determining the discretization interval of a time series of parameter measurements for automation systems has been developed, which is based on the use of discrete Fourier transformation for a smoothed digital signal and filtered noise and the Kotelnikov calculation theorem.

The use of the proposed method allows you to ensure the necessary accuracy of obtaining measurement information for any real parameter of the technological process and reasonably formulate the requirements for the used controller and measurement tools.

In addition, the proposed technique can be used to select the parameters of smoothing digital filters, as well as for a comparative analysis of their effectiveness.

## LITERATURE


1. Буличов, А. (2014). Контроль стану ізоляції мережі 6-10 кВ з ізольованою нейтраллю. *Вісник Вологодського державного технічного університету*, (4), 45–49.
2. Дороніна, О., & Гребченко, Н. (2013). Оперативний контроль стану ізоляції приєднань 6-10 кВ із двигунами. *Автоматизація об'єктів та технологічних процесів*, (6), 40–44.
3. Качесов, В. (2015). Метод визначення зони однофазного замикання у розподільчих мережах під робочою напругою. *Електрика*, (6), 13–15.
4. Шкрабець, Ф. П., Місяць, Є. П., & Кириченко, М. С. (б. д.). Структура системи безперервного контролю параметрів ізоляції у мережах напругою 6 – 10 кВ гірничих підприємств. У *Ефективність та якість електропостачання промислових підприємств* (с. 18–22).
5. Благовіщенська, М. М. (2015). *Інформаційні технології систем управління технологічними процесами*. Вища школа.
6. Конюх, В. Л. (2013). *Комп'ютерна автоматизація виробництва*. КузДТУ.
7. Айфічер, Е. (2014). *Цифрова обробка сигналів. Практичний підхід*.
8. Kotel'nikov V. A. (2016) On the bandwidth of "air" and wires in telecommunications // Advances in Physics: Journal. (7), 762-770.
9. Blagoveshchenskaya, M. M. (2015). Information technologies of control systems of technological processes. Higher. Shk.

# ADDITION

## The program of the simulation model of conducting experiments

```
close all;clear all;clc
% ЗАДАНИЕ ИСХОДНЫХ ПАРАМЕТРОВ РАСЧЕТА
Tanaliz=600;% Длительность интервала анализа сигнала, с
dt=0.04;% Шаг дискретизации по времени, с
N_d=1200;% Постоянная составляющая измеряемого параметра
skoSUM=1.5;% Ср. кв. отклонение шума
f=[0.3,0.4,0.5];a=(N_d*pi/30)*[0.025,0.03,0.015]; % Параметры зависимости w(t)
k_br=0.08;% Коэффициент передачи тахогенератора (первичный преобразователь)
yUmin=4;yUmax=20;% Диапазон изменения унифицированного сигнала y_ун(t), мА
n=10;% Количество двоичных разрядов АЦП
% РАСЧЕТ
t=0:dt:Tanaliz;N=length(t);% Вектор отсчетов времени
p=[3,1,2];
%p=randperm(3) % Вектор случайной выборки параметров w(t)
%p=input('Введите вектор виборки параметров w(t) в формате [p1,p2,p3] -')
% Задание зависимости w=f(t)
Ag=[a(p(1)), a(p(2)), a(p(3))];
Us_d=N_d*pi/30;% Номинальная угловая скорость вращения ЭД
x=Us_d+Ag(1).*sin(2*pi*f(1)*t)+Ag(2).*sin(2*pi*f(2)*t)+Ag(3).*sin(2*pi*f(3)*t);
y=k_br*x;% Вектор измерительного сигнала y(t)
% НОРМАЛИЗАЦИЯ СИГНАЛА
% Вектор значений унифицированного сигнала y_ун(t)
y_max=max(y); y_min=min(y);Dy=0.5*(y_max-y_min);y_max=y_max+Dy; y_min=y_min-Dy;
yUn=(y*(yUmax-yUmin)+yUmin*y_max-yUmax*y_min)/(y_max-y_min);
% ДОБАВЛЕНИЕ ШУМА
% Вектор значений шума на входе АЦП
SUM=skoSUM*((y_max-y_min-2*Dy)/2)*randn(1,N);% Нормальный закон распределения
%k=1;SUM=k*(-skoSUM+2*skoSUM.*rand(1,N));% Равномерный закон распределения
z=yUn+SUM;% Вектор сигнала на входе АЦП z(t)
% КВАНТОВАНИЕ ПО УРОВНЮ
Nkv=2^n;% Количество уровней квантования
Skv=(yUmax-yUmin)/Nkv;% Шаг квантования
Ukv=[yUmin:Skv:yUmax];% Вектор значений уровней квантования
for i=1:N
if z(i)<Ukv(1);z(i)=Ukv(1);end
if z(i)>Ukv(length(Ukv));z(i)=Ukv(length(Ukv));end
j=1;while z(i)>Ukv(j);j=j+1;end;Zkv(i)=Ukv(j);
if (Ukv(j)-z(i))>0.5*Skv;Zkv(i)=Ukv(j-1);end
end
% ФИЛЬТРАЦИЯ ШУМА
% Сглаживание методами Брауна 1, 2 порядка и методом Хольта
alfaB1=0.3;alfaB2=0.3;alfaH1=0.3;alfaH2=0.2; % Значения коэффициентов сглаживания
% Инициализация фильтров
sBr(1)=mean([Zkv(1),Zkv(2),Zkv(3),Zkv(4),Zkv(5)]);sBr2(1)=sBr(1);sH(1)=sBr(1);TH(1)=0;
```

```
% Реализация алгоритмов фильтров
for i=2:N
sBr(i)=sBr(i-1)+alfaB1*(Zkv(i-1)-sBr(i-1));
sBr2(i)=sBr2(i-1)+alfaB2*(sBr(i-1)-sBr2(i-1));
sH(i)=alfaH1*Zkv(i)+(1-alfaH1)*(sH(i-1)+TH(i-1));
TH(i)=alfaH2*(sH(i)-sH(i-1))+(1-alfaH2)*TH(i-1);
end
% Сглаживание методом скользящего среднего
sSS(1)=Zkv(1);sSS(2)=mean([Zkv(1),Zkv(2)]); sSS(3)=mean([Zkv(1),Zkv(2),Zkv(3)]);
sSS(4)=mean([Zkv(1),Zkv(2),Zkv(3),Zkv(4)]);
for i=5:N;sSS(i)=mean([Zkv(i),Zkv(i-1),Zkv(i-2),Zkv(i-3),Zkv(i-4)]);end
Zsgl=sBr;% Выбор метода фильтрации
% ОПРЕДЕЛЕНИЕ АВТОКОРРЕЛЯЦИОННОЙ ФУНКЦИИ СИГНАЛА
vs=mean(Zsgl); % Определение выборочного среднего
vd=cov(Zsgl);% Определение выборочной дисперсии
for k=0:round(1/dt)
su=0;for ik=1:N-k;su=su+(Zsgl(ik)-vs)*(Zsgl(ik+k)-vs);end
tau(1+k)=k*dt;AKF(1+k)=su/((N-1-k)*vd);end
ik=1;while AKF(ik)>0.05*AKF(1);ik=ik+1;end;
% ГАРМОНИЧЕСКИЙ АНАЛИЗ
fd=1/dt;% Частота дискретизации, Гц
fnaikv=fd/2;% Частота Найквиста, Гц
df=fd/(N-1);% Шаг дискретизации по частоте, Гц
Sign=[yUn;Zkv;Zsgl;Zsgl-Zkv];% Матрица анализируемых сигналов
for i=1:4
DPF(i,:)=abs(fft(Sign(i,:),N));% Амплитуды преобразования Фурье сигналов
DPF(i,:)= 2*DPF(i,:)./N; % Нормировка спектра по амплитуде
end
DPF(1:4,1)= DPF(1:4,1)./2; % Нормировка постоянной составляющей в спектре
f1=0:df:1*fnaikv;dv=length(f1);% Вектор частот отображения спектра
vs_A=mean(DPF(3,:));vsko_A= std(DPF(3,:));
j=1;for i=1:dv;if DPF(3,i)>(vs_A+1*vsko_A);Ng_F(j)=i;j=j+1;end;end
% РЕЗУЛЬТАТЫ РАСЧЕТА
format short g;
Rrasc(1)=Tanaliz;Rrasc(2)=dt;Rrasc(3)=N;Rrasc(4)=fd;Rrasc(5)=df;
Tanaliz_dt_N_fd_df=Rrasc
% Время спада автокорреляционной функции
sc=0;Z1=z- mean(z);for i=1:N-1;if sign(Z1(i)*Z1(i+1))<1;sc=sc+1;end;end
Tau_sp_min=Tanaliz/ sc
Tau_sp_max=tau(ik)
Cg_F=f1(Ng_F)% Частоты гармоник спектра полезной составляющей
Ag_0=(Us_d*k_br*(yUmax-yUmin)+yUmin*y_max-yUmax*y_min)/(y_max-y_min);
Ag=((Us_d+Ag)*k_br*(yUmax-yUmin)+yUmin*y_max-yUmax*y_min)/(y_max-y_min)-Ag_0;
TAg=[Ag_0,Ag] % Точные значения амплитуд спектра полезной составляющей
AGDPF=sprintf('Амплитуды гармоник спектра полезной составляющей по ДПФ сигналов [yUn;
Zkv; Zsgl; Zsgl-Zkv]')
Ag_F=[DPF(1:4,Ng_F)]
% ОПРЕДЕЛЕНИЕ АДЕКВАТНОСТИ АППРОКСИМАЦИИ
% Определение средней относительной погрешности аппроксимации
```

```
SOPA=100*(sqrt((sum((Zkv-Zsgl).^2))/N))/mean(Zkv)
% По времени спада автокорреляционной функции
dt_min=4*Tau_sp_min*cov(Zsgl-Zkv)/cov(z)
dt_max=4*Tau_sp_max*cov(Zsgl-Zkv)/cov(z)
SGSP=100*Ag_F (4,1:4)./Ag_F (3,1:4) % Содержание гармоник в шуме в % по ДПФ
%Sss=(SGSR(2)+SGSR(3)+SGSR(4))/3
% ВЫВОД ГРАФИКОВ
% График нормированной автокорреляционной функции Zkv (черный цвет)
plot(tau,AKF,'Color','k','LineWidth',2);hold all;grid on
stem(tau(ik),AKF(ik),'Color','r','LineWidth',2)
title('Нормированная АКФ квантованного сигнала');
xlabel('Величина временного сдвига  \it\tau, c'); ylabel('Нормированная АКФ \it  K_{z}');
It=700;% Количество точек вывода графиков
figure(2)
% График квантованного временного ряда, измеренного датчиком (зеленые точки)
plot(t(1:It),Zkv(1:It),'o','Color','g','LineWidth',2);hold all;grid on
% График точного изменения измеряемого параметра (черные точки)
plot(t(1:It),yUn(1:It),'o','Color','k','LineWidth',2)
% График сглаженного временного ряда (красный цвет)
plot(t(1:It),Zsgl(1:It),'Color','r','LineWidth',2.5)
title('Анализируемые временные ряды');
xlabel('Время \it  t (c)'); ylabel('Измеряемый параметр \it  z');
legend('Реальный ряд','Точный ряд','Отфильтрованный ряд')
figure(3)
% График спектра точного временного ряда (черный цвет)
subplot(2,2,1),plot(f1(1:dv),DPF(1,1:dv),'Color','k','LineWidth',2);grid on;
title('Спектр точного временного ряда');xlabel('Частота \it  f, Гц'); ylabel('Амплитуда');
% График спектра измеренного временного ряда (зеленый цвет)
subplot(2,2,2),plot(f1(1:dv),DPF(2,1:dv),'Color','g','LineWidth',2);grid on;
title('Спектр измеренного временного ряда');xlabel('Частота \it  f, Гц'); ylabel('Амплитуда');
% График спектра отфильтрованного временного ряда (синий цвет)
subplot(2,2,3),plot(f1(1:dv),DPF(3,1:dv),'Color','b','LineWidth',2);grid on;
title('Спектр отфильтрованного временного ряда');xlabel('Частота \it  f, Гц'); ylabel('Амплитуда');
% График спектра шума(красный цвет)
subplot(2,2,4),plot(f1(1:dv),DPF(4,1:dv),'Color','r','LineWidth',2);grid on;
title('Спектр отфильтрованного шума');xlabel('Частота \it  f, Гц'); ylabel('Амплитуда');
```

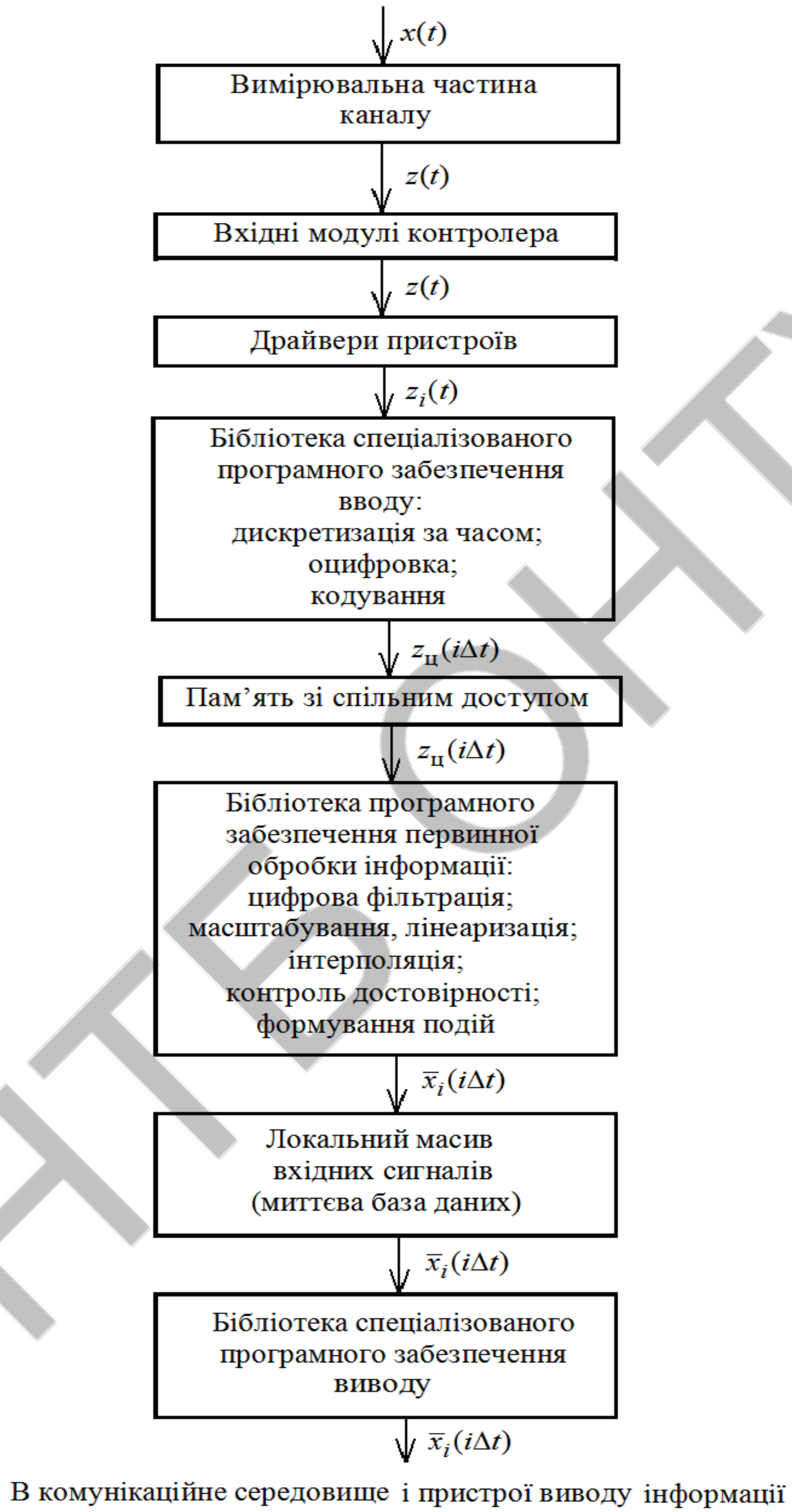



Fig.4. A typical algorithm for primary processing of measurement information

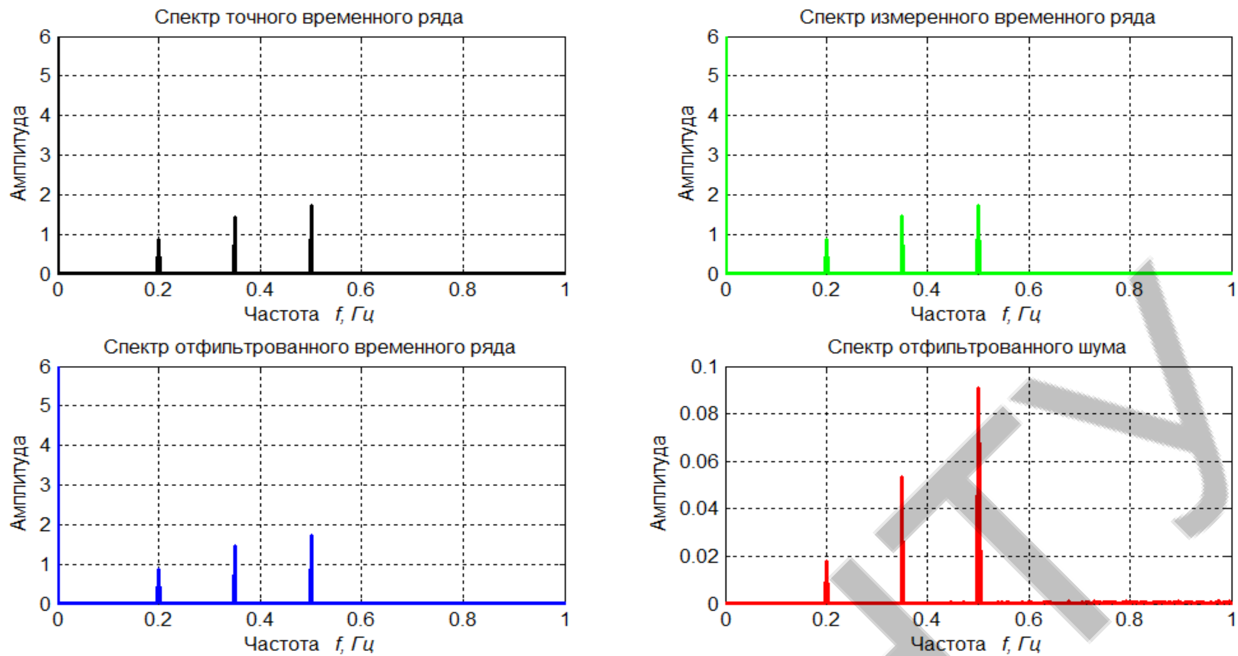
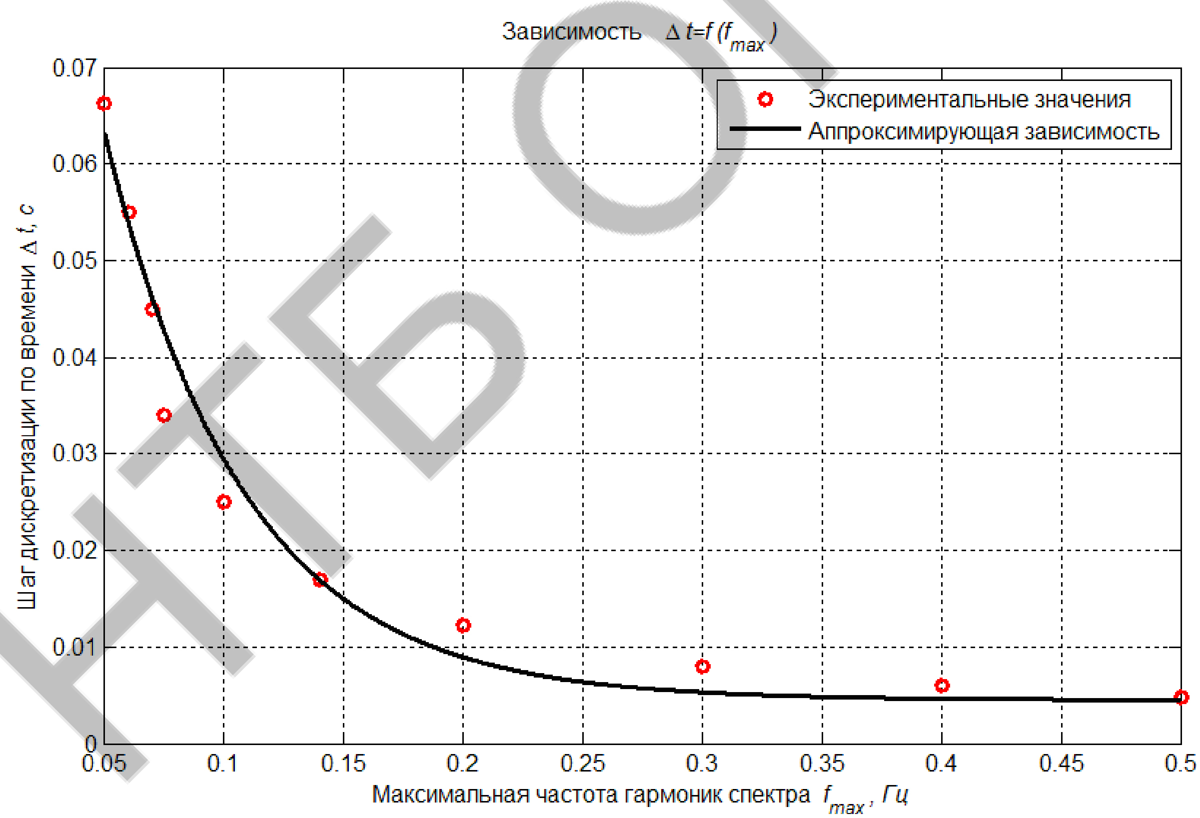
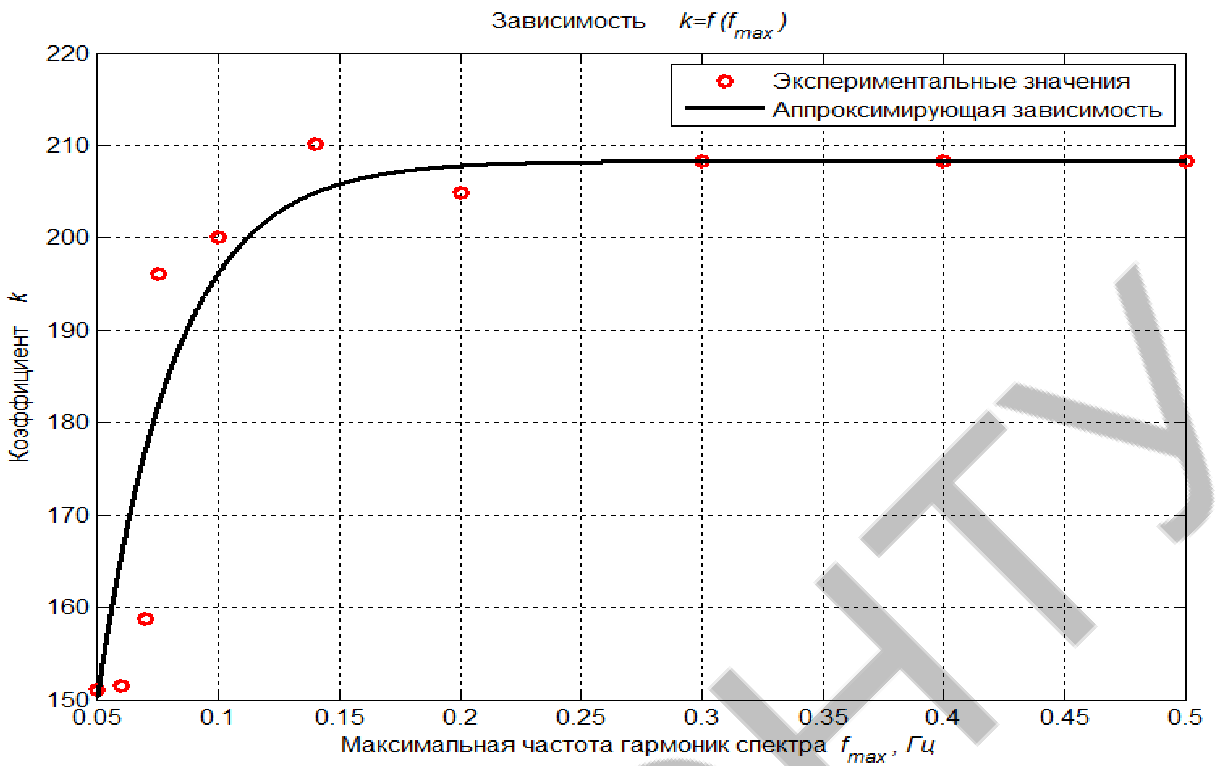



Fig. 6. Frequency spectra of signals




a)




b)

Fig.8. Results of computational experiments obtained with $100\,A_{\text{max,шум}} / A_{\text{max,згл}} = 5\ \%$