www.ADClab.ru

LTD«ADClab»

Data viewing and processing software

LookALF

Technical description

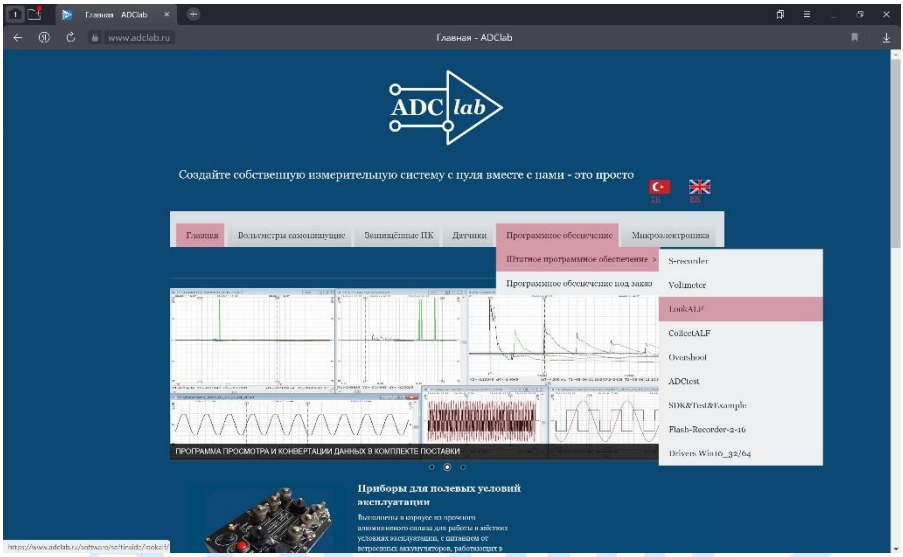
Moscow
2023

Content

Introduction

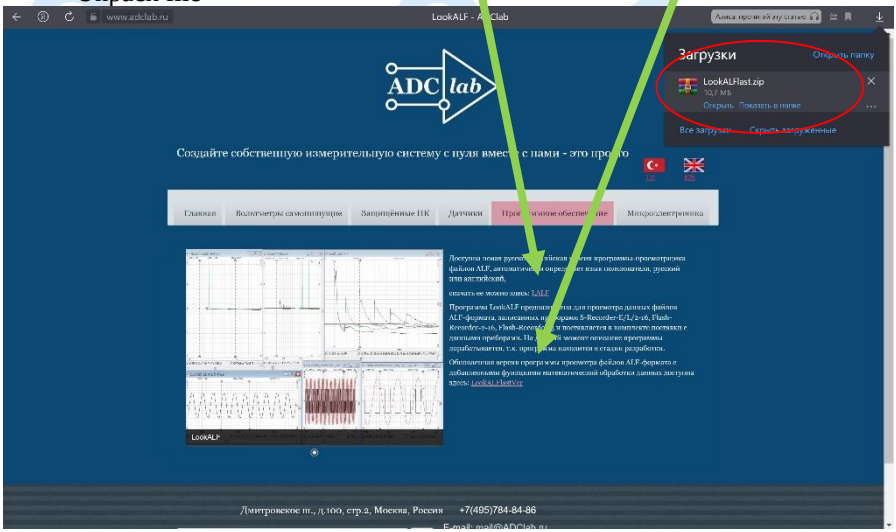
The LookALF software package is designed for viewing, mathematical processing, graphical analysis and documentation of test results recorded using external analog-to-digital conversion devices of ADClab LTD.
List of compatible devices: Voltmeter S-Recorder-E, S-Recorder-L, S-Recorder-2-16, Flash Recorder-2-16, Flash Recorder MITX series, Flash Recorder-3 M1-6 modifications.
Preparing to work with the program.
LookALF software is supplied with a measuring device of our company.
You can download the current version of the software on the official website:
www.ADClab.ru

1. **To start working with the software package, you should:**
   Open the link: www.ADclab.ru ;
   Select «Software – LookALF»



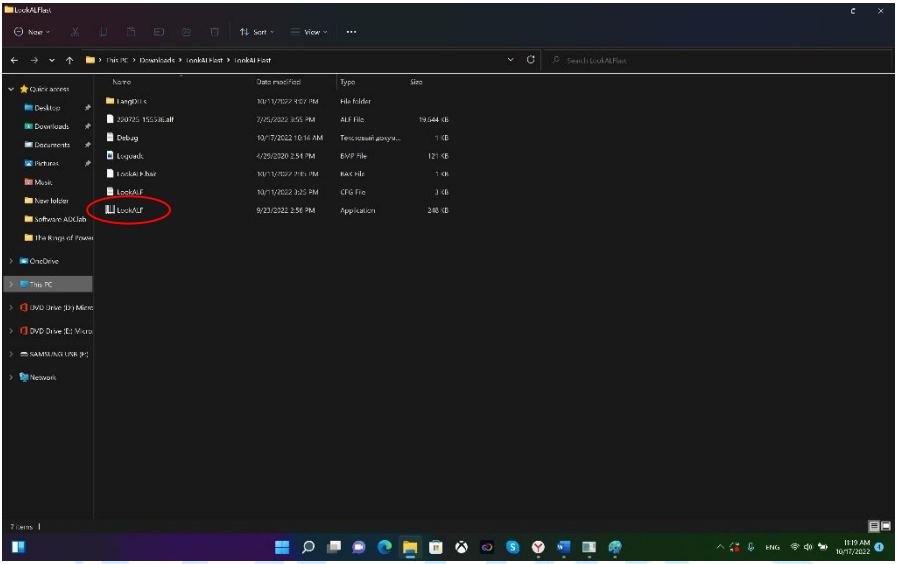
Pic. 1

2. Download LALF or LookALFlastver
   Unpack file



Pic. 2

3. Open the app LookALF



Pic. 3

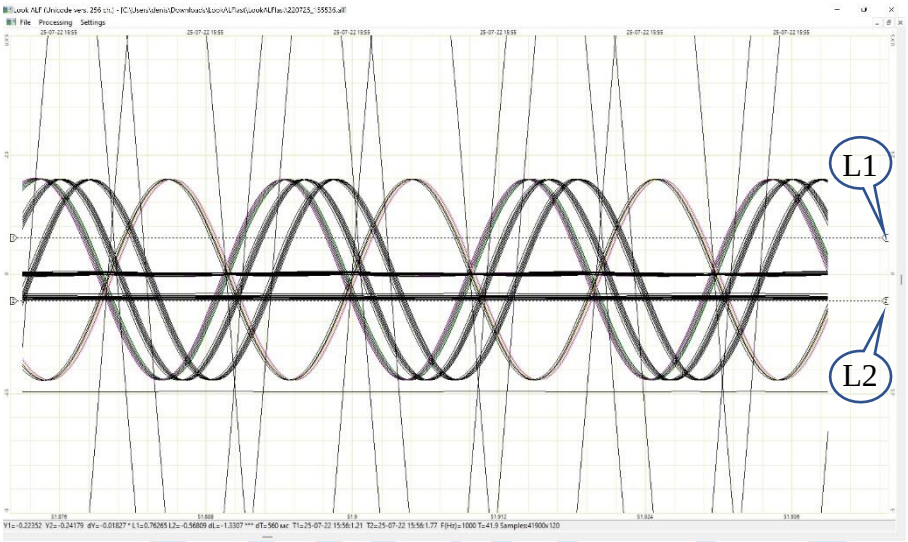
The LookALF software package is a universal software tool for post-processing and correct analysis of the recorded data of the conducted experiment.

Thanks to a variety of functions, LookALF allows you to present experimentally obtained data in graphical form, display time and parametric dependencies, perform graphical measurements, compare measurement graphs with each other and with graphs of various measuring devices.

Simultaneous work with several windows inside the program is possible, also with a large number of graphs, it is possible to work on several monitors for the convenience of locating and comparing a large number of necessary parameters.

The software package has a set of functions for mathematical data processing. One-time processing can be carried out using various methods for sequential multi-stage data processing. The processed data is also displayed on the screen in graphical form.

Software design:



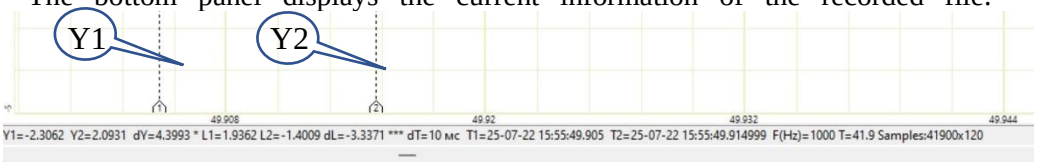
Pic. 4 Окно графиков

The program window is a space for placing a large number of graphs. With the help of labels placed on the graph, the range and amplitude of the signal, its duration are measured. You can also see the sampling rate for each channel.

There are tabs on the top panel:

- File:
- "Open a file" in .alf format, open previously viewed files "History", "Close the graph window", "Close all graph windows", "Save the processed file", "Save a file fragment (selected channels)", "Exit the program";Обработка (математические функции) – отдельный раздел описания;
- Settings:
  «Settings» - is meant for working with devices and channels,
- «About» (Actual software version).

The bottom panel displays the current information of the recorded file:



Pic.5 The bottom panel

Tab. 1

Наименование параметра	Описание	Корректировка
Y1	Mark №1 axis Y chosen channel Shows the Y-axis value at a specific point	You can set the mark in the "Parameters" section by selecting the necessary channel. Moves with the left mouse button held down
Y2	Mark №2 axis Y chosen channel Shows the Y-axis value at a specific point	You can set the mark in the "Parameters" section by selecting the necessary channel. Moves with the left mouse button held down
dY	The difference between the values of the marks Y1 and Y2 on the ordinate axis	Depends on the set marks Y1, Y2
L1	Horizontal mark №1 (line). It is not fixed to a specific point, you can track the change of the graph over the entire duration of time	Moves with the mouse and the Shift key held down
L2	Horizontal mark №2 (line). It is not fixed to a specific point, you can track the change of the	Moves with the mouse and the Shift key held down

	graph over the entire duration of time	
dL	The difference between the values of the L1 and L2 marks along the abscissa axis	Depends on the set marks L1, L2
dT	The time passed from Y1 to Y2, assigned to the real time of the experiment	Depends on the set marks Y1, Y2
T1	Current time of the set mark Y1	The time of the set mark Y1 from the beginning of the experiment, or the real time of the study, depending on the selected value in the "Parameters" window
T2	Current time of the set mark Y2	The time of the set mark Y2 from the beginning of the experiment, or the real time of the study, depending on the selected value in the "Parameters" window
F(Hz)	Sampling rate	-
Точки	Duration of measurements (number of points of the recorded file)	-

There is also a file scroll slider on the bottom panel.

The program can open several files in different windows, save the file in text and alf formats as fully graphics and separate channels, remember the history of previously viewed files, perform mathematical transformations, and work with devices and channels together and separately.

Tab. 2

Формат файла ALF float

Offset байт	Длина байт	Тип	Значение
0	24	Строка	Всегда содержит "ADCLABFFS" и пробелы, начало заголовка файла
24	8	int64	Восемь нулевых байт
32	24	Строка	Всегда содержит "SAMPLES_FORMAT"
56	8	Int64	Всегда 0x00000014(20) и 0x00000000, длина структуры
64	4	Unsigned	Маска, всегда равна 7
68	4	int	Число записанных каналов в файле, далее обозначается как "К"
72	8	double	Частота выборки на каждый канал (синхронную пару)
80	1	Байт	Тип выборки, для float alf всегда равен 1, файл содержит 4-х байтовые выборки типа float
81	1	Байт	Номер младшего значащего бита, для float alf всегда = 0
82	1	Байт	Число значащих бит, для float alf всегда = 0
83	1	Байт	Режим = 0
84	24	Строка	Всегда содержит "CHANNELS_INFO_HEADER" и пробелы
108	8	Int64	Всегда для float alf содержит 0x00000014(20) и 0x00000000, длина структуры
116	4	Unsigned	Маска, для float alf = 0x00000003
120	2x8	Double[2]	Диапазон сигналов в каналах, мин и макс
136	24	Байты	Всегда содержит "CHANNELS_INFO" и далее пробелы
160	8	Int64	Длина структуры, = 20 x К (число записанных каналов)
168	20xK	Структуры	Тут лежат структуры для каждого из К записанных в файле каналов. Структура описана ниже
168+20xK	24	Строка	Всегда содержит "SAMPLES_RECORD_INFO" и пробелы
192+20xK	8	Int64	Длина структуры, = 8
200+20xK	8	Int64	Смещение, = 0
208+20xK	24	Строка	Всегда содержит "SAMPLES_RECORD" и пробелы
232+20xK	8	Байты	Всегда 8 байт 0xff
240+20xK	Много X 4 байта, до конца файла	float	Область отсчетов (измерений). Каждый отсчет по 4 байта типа float. Область отсчетов содержит последовательные выборки. Одна выборка включает по одному отсчету для заданного числа каналов(пар). В поле со смещением 68 (см. выше) определяется, сколько каналов включено в выборку. Длина одной выборки в байтах = 4 x число каналов. В структурах, расположенных ниже смещения 168, определяется, какие каналы включены в выборку

Формат структуры для канала

Offset байт	Длина байт	Тип	Обозначение	Значение
0	4	int		Номер канала
4	2x8	Double[2]		Диапазон сигнала для канала, мин и макс

The ADCLABFF file format is used to store the data received during the collection process. Files in this format have the extension *.alf.

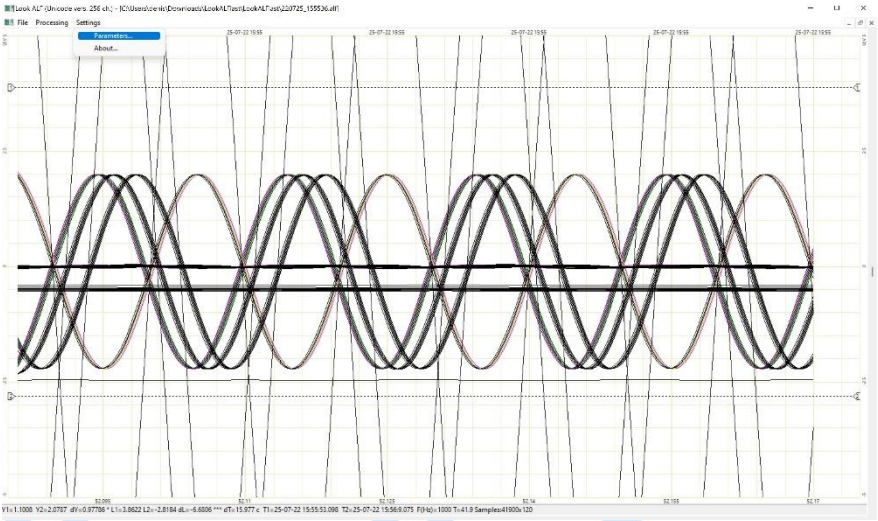
ADCLABFF is a specialized binary file format developed by ADClab specifically for storing large amounts of data. This format provides the possibility of fast sequential data writing and quick access to data when reading.

The software product can be used independently of the hardware of the software and hardware complexes, providing a service for working with alf files. When storing data in alf files, the subsequent possibility of viewing and analyzing them by the software product is provided, as well as the possibility of converting them into files of other formats.

Functionality

- **Operating with graphs**

For efficient operation with data displayed the device cannel sorting should be made as it is shown below:

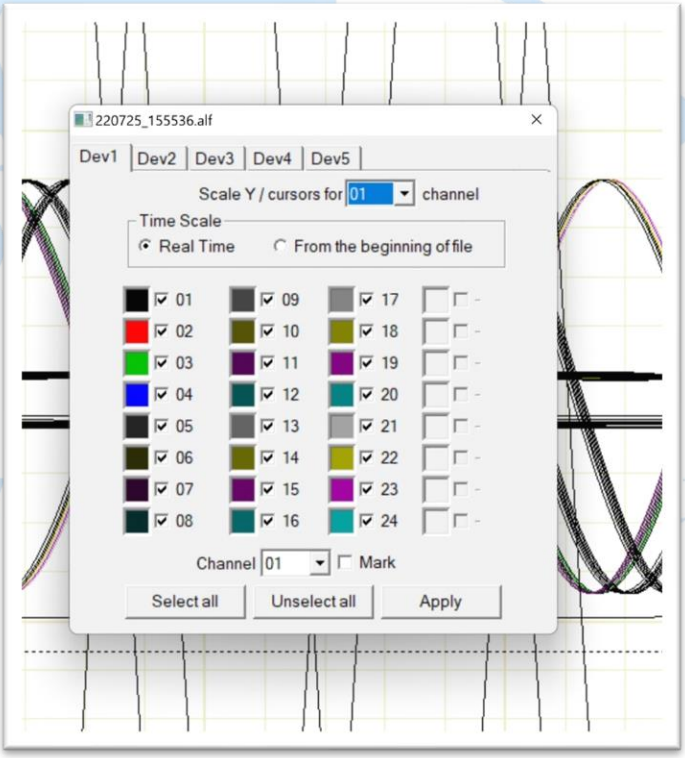


Pic. 5 Settings

The software package allows you to place up to 8 measuring devices with 32 channels in each. Depending on the test performed, LookALF will select the number of devices that participated in the recording of this file. Figure 6 (below) shows 5 devices (Div1, Div2, Div3, Div4, Divx 5), which corresponds to the study being conducted. Signals from the generator were sent to 424 channels of each device, which can be seen in the parameters window.

«Settings» interface (Pic. 6) provides:

1. Switching between devices,
2. *Choose the device (example Dev1-5);*
3. Display or hide signal curve on any available device,
   1-24 numbers correspond channels of chosen device. Hide/unhide channel by clicking left mouse button;
4. Choose color by pressing;
5. Sett the cursor on the channel in question;
6. Find a channel and highlight it;
7. Choose time set;
8. One can choose or cancel all channels via corresponding buttons in the settings window;
9. Press apply after introducing alterations.



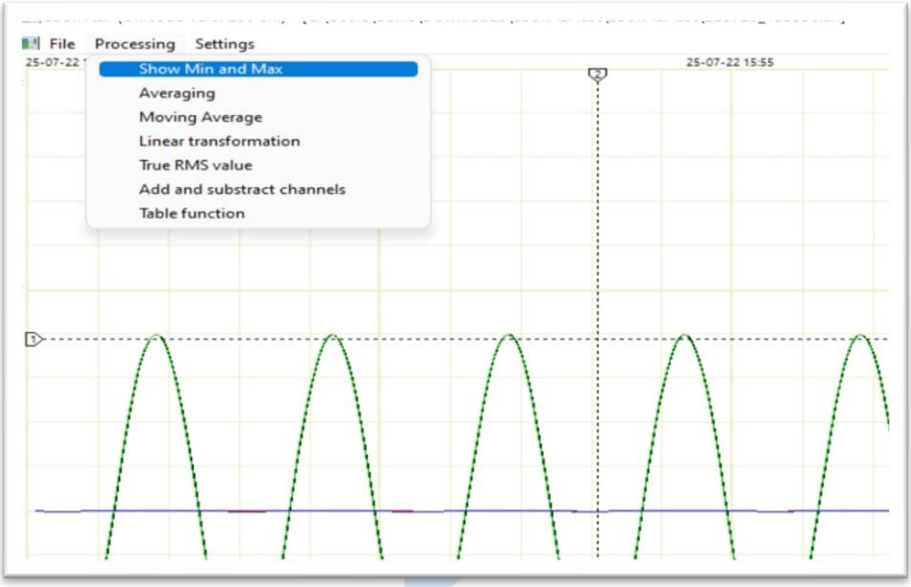
Pic.6 «Settings»

Math functions and conversion

The software package provides a wide range of mathematical transformations and possibilities for building dependencies and calculating values using various functions.

Open «Processing» tab as it is shown on the Picture №7.

Attention!!! Each math function is performed only for the area between the exposed marks Y1 и Y2.



Pic. 7 «Processing»

1. **«Show min-max» function**

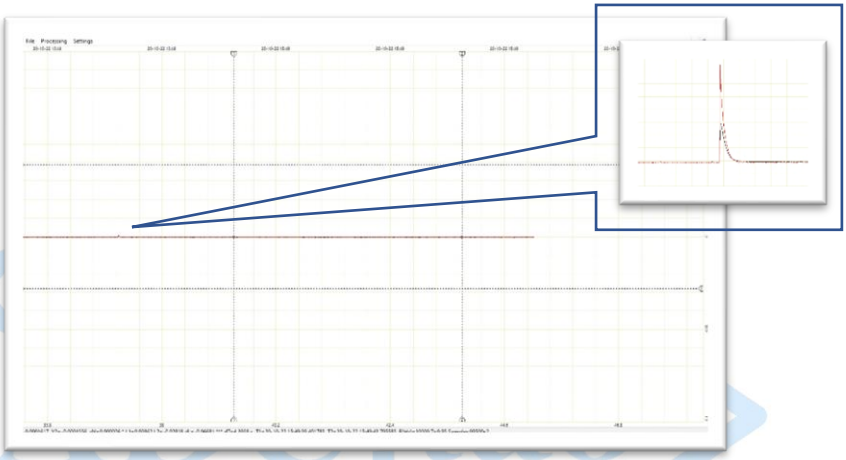
LookALF software can open big data files with huge data arrays. To display high speed and short-term processes, for example impulses, failures and fluctuations, which are difficult to search. It is much easier to apply «Show min-max» function which operates in the following way:

The whole data array M is divided into 1920 R pieces in case with HD resolution, after which minimal and maximal values are calculated in M1, M2…Mn arrays and display afterwards.

Here is the formula to submit the results:

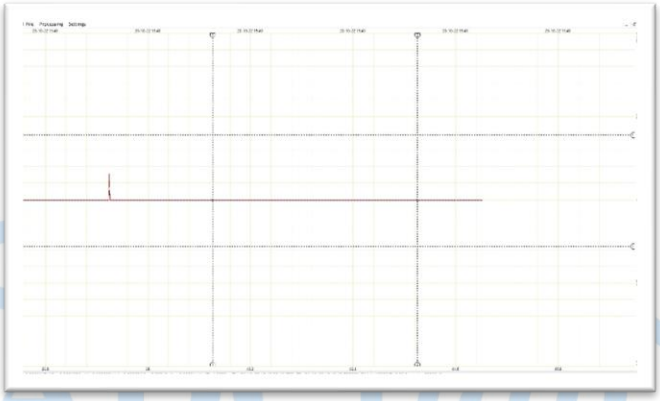
$$Min\left\{\frac{M_n}{2R}\right\}; Max\left\{\frac{M_n}{2R}\right\}$$

Here is the example «Show min-max» function:



Pic. 8 Recorded file

The graph shows (pic. 8) continuous file as the result of test conducted. Here you can see impulse across the record length thanks to min-max function. This computational function can display pulse peaks along the entire length of the recording, which avoids the loss of some important values. This processing is useful for long-term recording with a high sampling rate, which is typical for monitoring constant measurements.



Pic. 9 Completed «Min-max»

When performing this function, all invisible pulses are displayed for a long period and are easier to find.

2. «Averaging» function

Averaging – is equal to the sum of all the numbers of the set divided by their number.

$$\bar{x} = \frac{1}{n}\sum_{i=1}^{n} x_i = \frac{1}{n}(x_1 + \cdots + x_n)$$

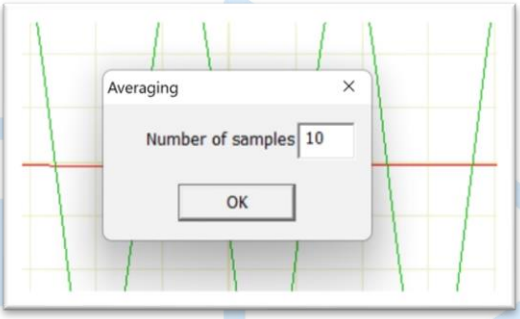
где $n - averaging\ number$

For your information, each four-fold averaging gives an additional level of accuracy of the measurements with respect to the random noise component.

This method is effective in the case of a significant excess of the sampling frequency over the upper frequency of the measured signal, this method is called Oversampling and confirms the advantage of a high sampling rate of our devices.

Note. When performing this function, it is necessary to take into account the selected averaging interval and the sampling frequency on the site. The number of points between the markers Y1, Y2 during averaging should be at least twice as many as the averages. For example, between the labels Y1 and Y2 100 milliseconds at a sampling frequency of 1000 Hz, respectively, 100 points inside the interval and the number of averaging should be no more than 50, which corresponds to a result of 2 points, otherwise the program will not be able to build a curve.

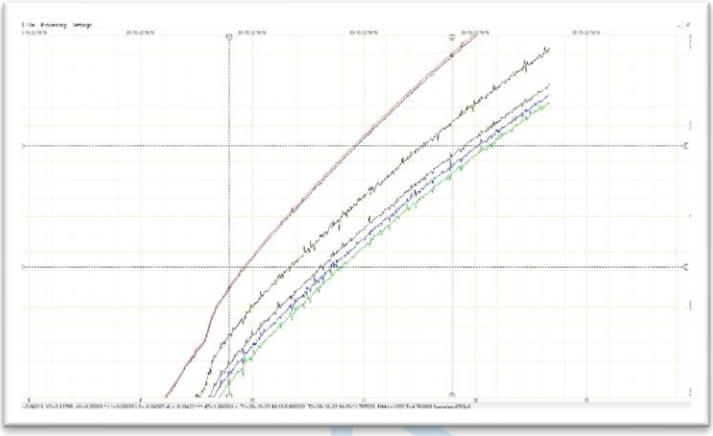
Picture 8 shows the window for selecting the number of averages.



Pic. 10 Averaging window

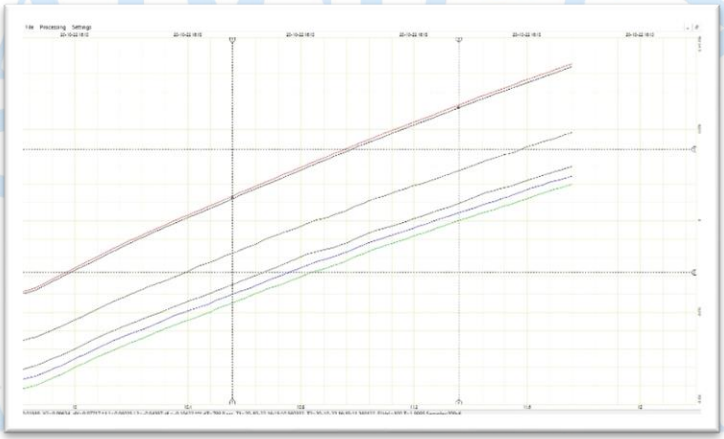
When measuring slow-moving processes with the ability to record a high sampling rate, you can use the averaging function to track the trend of signal changes over time.

Picture 11 shows a signal that needs to be filtered by time and track the trend of the value change over time.



Pic. 11 Recorded signal

Averaging provides to visually and without noise research the process, as shown in Pic. 12.



Pic. 12 Math function «Averaging» result with the number of averages equal to 10

3. «Moving average» function

This transformation is used to smooth out short-term fluctuations in order to determine long-term dependence.

An even more interesting function differs from the simple averaging in that the number of measurements does not decrease after this conversion, best shown here:

https://excel2.ru/articles/skolziashchee-srednee-v-ms-excel

Math formula:

$$Y(t) = \frac{Y(t-1) + Y(t) + Y(t+1)}{3}$$

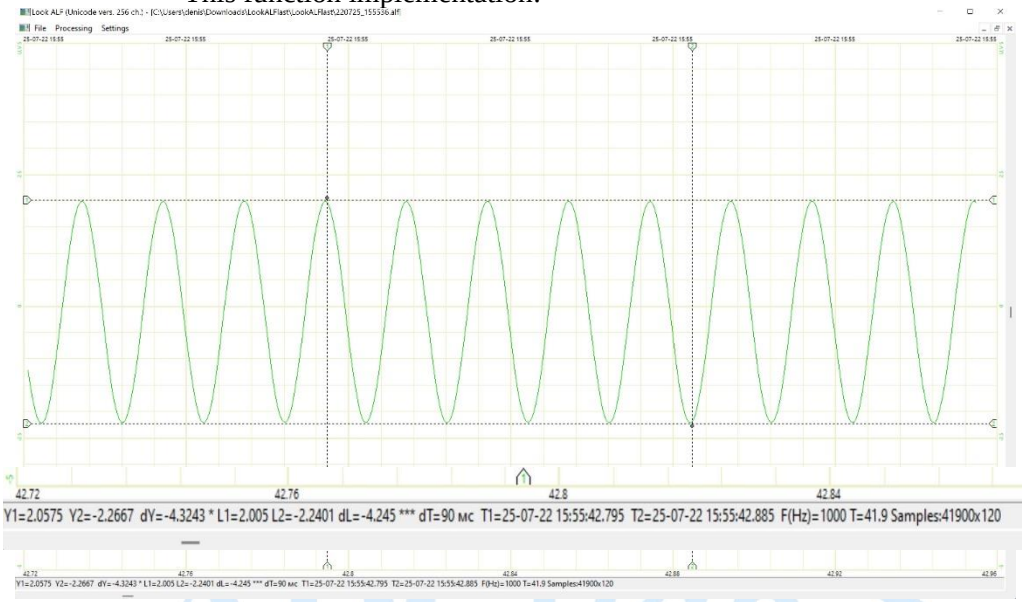
4. «Linear transformation» function

The main property of this function is that the increment of the function is proportional to the increment of the argument.

Math formula:

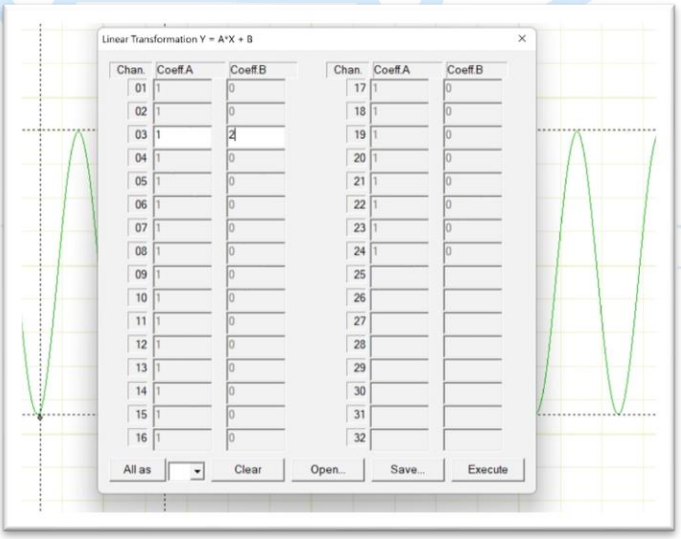
$$Y = aX + b$$

This function implementation:



Pic. 13 Chosen period

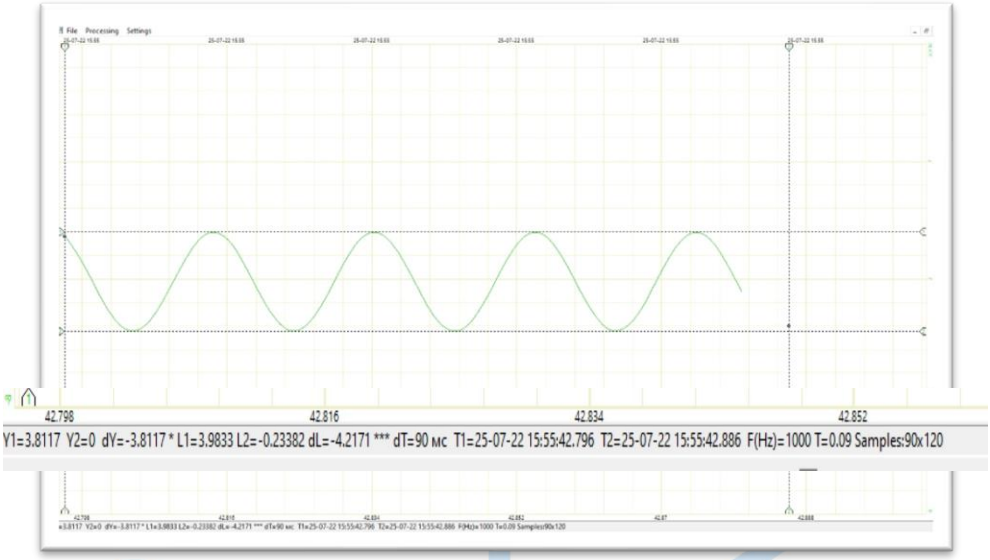
Picture 14 shows a sinusoid with parameters on the bottom panel. Let's apply The "Linear Transformation" function:



Pic. 14 «Linear Transformation» parameters window

$$Y = aX + b$$

Insert values a=1, b =2,
The result:

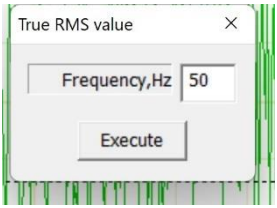


Pic. 15 Calculated result

Thus, we obtained the offset using the simplest linear transformation. You can also compare studies with other channels and record the result.

5. «True RMS value» function

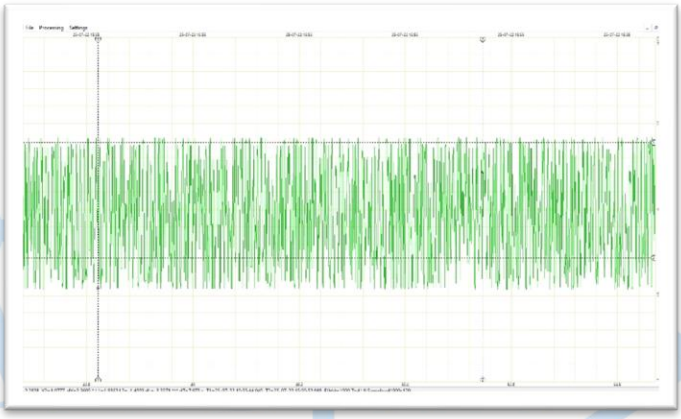
The effective (effective) voltage value is a DC voltage that will emit the same power as the measured alternating voltage at the same resistive load. Accordingly, the effective value of the current strength is such a value of the DC power, when passing through the resistive load, the same power will be released as when passing the measured current.



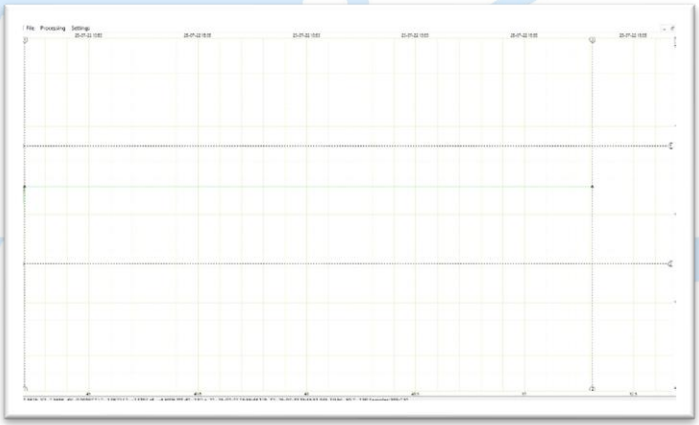
To perform calculations using this function, you need to enter the voltage frequency in Hertz.

Enter a known signal frequency, as a rule, 50, 60, 120, 400, ... Hz is used in the power grid.

To perform this function, a measurement area of 1-2 seconds is required.



Pic. 16 The sinusoid of the recorded signal, long-lasting



Pic. 17 The result of calculating the effective voltage value at a signal frequency of 50 Hz

The mathematical formula of the effective value used in the program:

$$Y(t) = \sqrt{\frac{1}{t}\int_0^t y^2\,dt} = \frac{U}{\sqrt{2}}$$

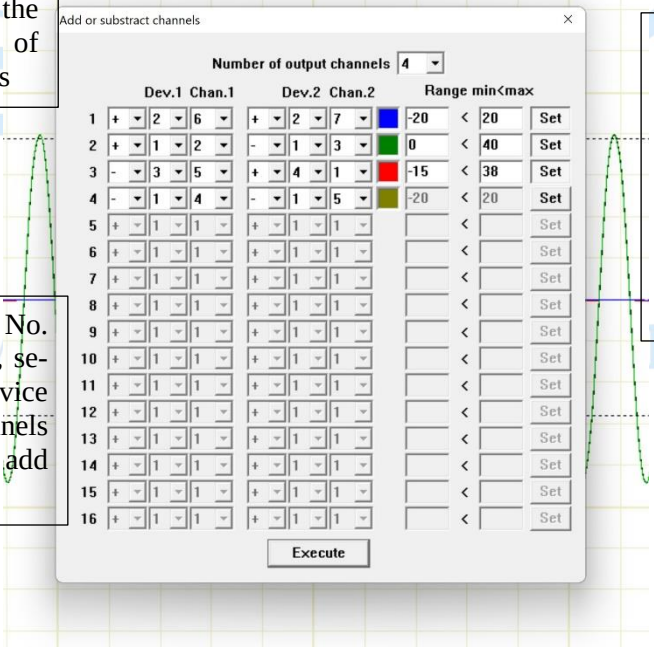
$\frac{1}{t} = F$ – sampling rate

6. «Addition and subtraction» function

This conversion provides to operate with signals recorded in channels. It is possible to add channels to each other and subtract them from each other.

n the settings window of this function, you can select the +/- argument, the device and channel number, and assign a new range of the Y scale based on the expected result.

Select the number of calculations



We select an approximate range from the minimum value to the maximum along the Y axis, which will correspond to the future calculation

In columns No. 1 and No. 2, select the device and channels that we will add or subtract

Pic. 18 «Addition and substraction» parameters window

As a result of calculations (Pic. 18), we will get 4 curves on the graph in a new file, processed using the mathematical function "Addition and subtraction".

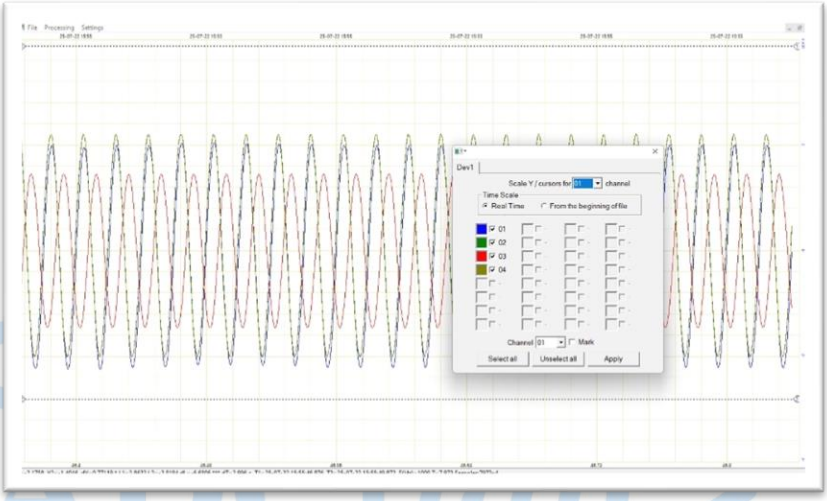
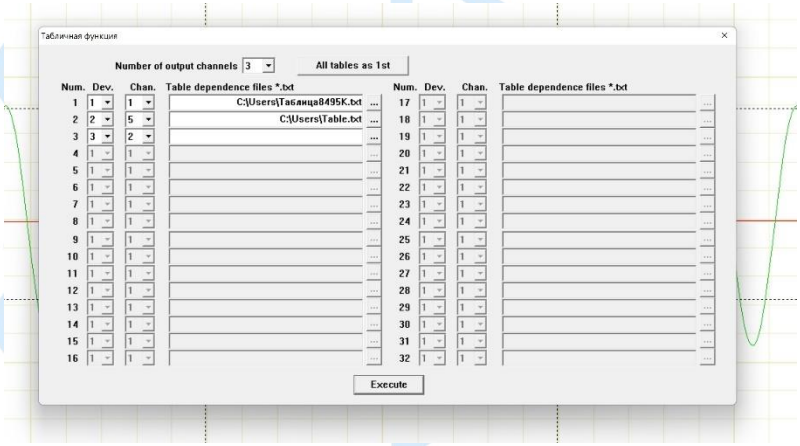


Рис. 18 Results

7. «Table function»

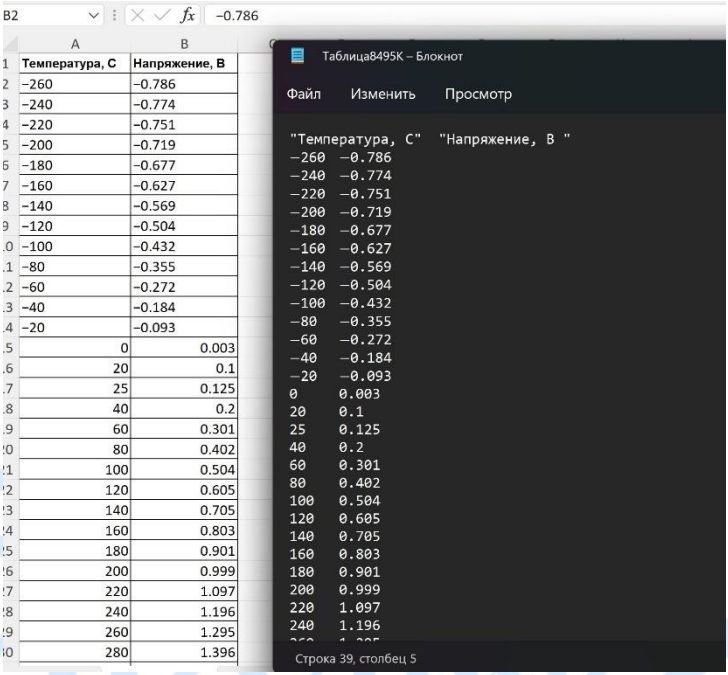
The function of transformation and correction of the measured process. Measuring instruments perceive and output information from different types of sensors in a certain type of measurement units. For the convenience of working with various units of measurement and displaying graphs in a form suitable for the researcher, it is possible to bring the graphs to the required form. As a rule, sensors have calibration tables for comparing the measured voltages with a certain unit of measurement.

Each sensor has clear characteristics and a calibration table with which it is possible to carry out these calculations.



Pic. 20 «Table function» parameters window

In the calculation window, you can select the device and channel number and their number for conversion. The file of the calibration table must be in the format .txt and match the format given below.



B2 −0.786

	A	B
1	Температура, С	Напряжение, В
2	−260	−0.786
3	−240	−0.774
4	−220	−0.751
5	−200	−0.719
6	−180	−0.677
7	−160	−0.627
8	−140	−0.569
9	−120	−0.504
10	−100	−0.432
11	−80	−0.355
12	−60	−0.272
13	−40	−0.184
14	−20	−0.093
15	0	0.003
16	20	0.1
17	25	0.125
18	40	0.2
19	60	0.301
20	80	0.402
21	100	0.504
22	120	0.605
23	140	0.705
24	160	0.803
25	180	0.901
26	200	0.999
27	220	1.097
28	240	1.196
29	260	1.295
30	280	1.396

Таблица8495K – Блокнот

Файл Изменить Просмотр

```
"Температура, С"  "Напряжение, В "
−260  −0.786
−240  −0.774
−220  −0.751
−200  −0.719
−180  −0.677
−160  −0.627
−140  −0.569
−120  −0.504
−100  −0.432
−80   −0.355
−60   −0.272
−40   −0.184
−20   −0.093
0     0.003
20    0.1
25    0.125
40    0.2
60    0.301
80    0.402
100   0.504
120   0.605
140   0.705
160   0.803
180   0.901
200   0.999
220   1.097
240   1.196
```

Строка 39, столбец 5

Pic. 21 Example of a calibration table

When working with tables in Microsoft Excel, the second column corresponds to the results, the unit of measurement of which is the same as when conducting research. The data in the experiment will be converted to the first column.

Saving a file from Excel format must be done in the form .txt (Unicode text) and opened in the "Tabular Function" window. As a result, we will get a scale and curves corresponding to the new unit of measurement.

For example, you work with thermocouples that have a non-linear scale for converting voltage to temperature in the form of a table that you can use in this conversion, each channel can be assigned its own table or the same for all channels.

8. «Spectral analysis» function

The function describes the amplitude coefficients when decomposing the original function into elementary components - harmonic oscillations with different frequencies.

The Fourier transform is integral and is given by the following formula:

$$\hat{f}(\omega) = \frac{1}{\sqrt{2\pi}} \int_{-\infty}^{\infty} f(x) e^{-ix\omega} dx$$

$\hat{f}(\omega)$ –spectrum, ω – cyclic frequency of the signal in increments $d\omega$, $f(x)$ - signal under study,

In this function of the LookALF software package, the "Windowed Fourier Transform" is used, which is determined by:

$$F(t, \omega) = \int_{-a}^{b} f(\tau) W(\tau - t) e^{-i\omega\tau} d\tau$$

$W(\tau - t)$ $-$ windowed function, t – time, ω – signal friquency, τ – window position, a,b – spectrum frequency limits, $f(\tau)$ – converted signal,

Window function in case of discrete transformation:

$$F(m, \omega) = \sum_{n=-a}^{b} f[n] \omega[n - m] e^{-j\omega n}$$

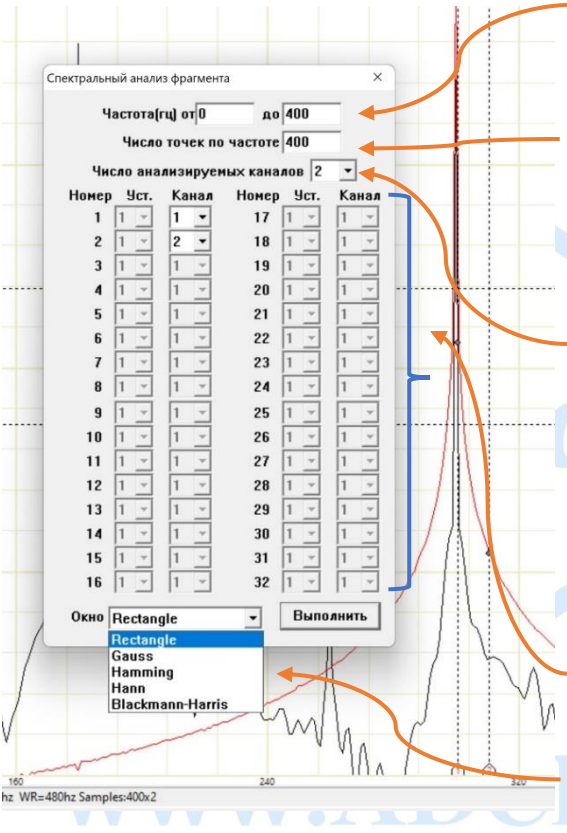
$f[n]$ – converted signal, $\omega[n - m]$ – window function, ω – cyclic frequency of the signal

To study the local distribution of frequencies, while preserving the original real—time variable, a generalization of the Fourier transform is used - the window Fourier transform. To begin with, you need to select some window function W, and this function must have a well-localized spectrum.

This section is intended for the periodic component of the signal.

A signal of a given duration is divided into a number of intervals using a sliding window of one type or another. This makes it possible to obtain, investigate and construct dynamic spectra in the form of spectrograms and analyze their behavior over time.

The spectral analysis window consists of:



1. Selecting the frequency range of the signal breakdown by spectrum in dB;
2. The interval of the spectrum window breakdown (number of points), preferably set as a multiple of the sampling frequency;
3. Selection of the number of analyzed channels. Depending on the number of analyzed channels, the type of window function used and the power of the computer, the processing time of the recorded signal changes;
4. Selection of the device and the processing channel;
5. Selection of the processing method.

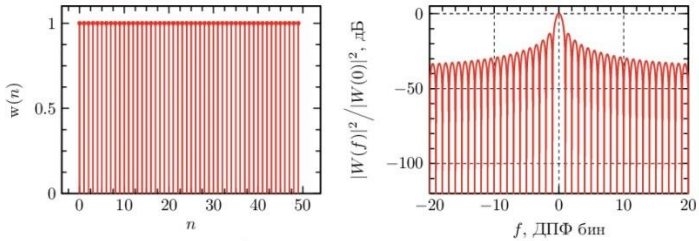
Pic. 22 «Spectrum analysis» parameters window

In order to reduce the spreading of the signal on the side lobes of the spectrum, it is necessary to reduce the level of the side lobes. Therefore, to reduce the spreading effect, it is necessary to change the window function, eliminating jumps at the beginning and end.

Functionality and types of window transformations:

- **«Rectangle» windowed function**

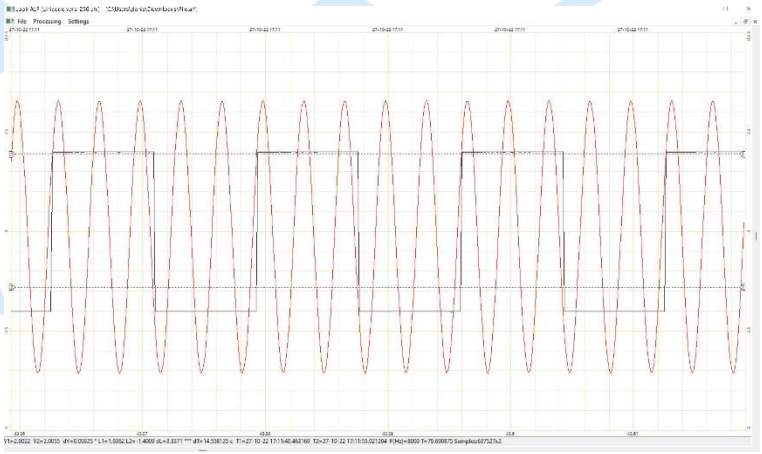
The rectangular window has a single value for all frequencies
n = 0…N-1;
The view of the rectangular window and its spectral energy density are shown below:



Pic. 23 «Rectangle»

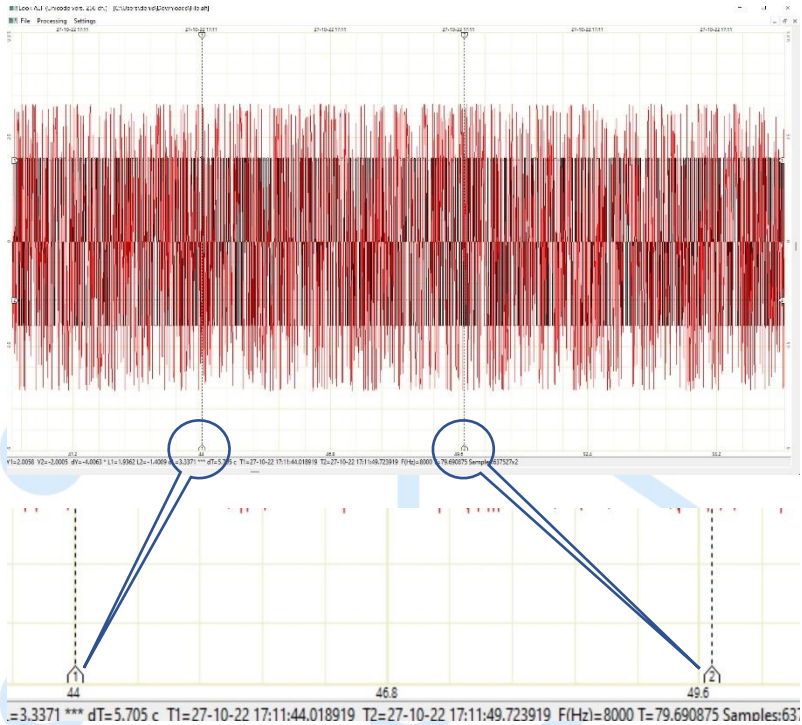
Let's consider an example of performing this window transformation:

We open previously recorded signals in the form of sinusoid and rectangular pulses,

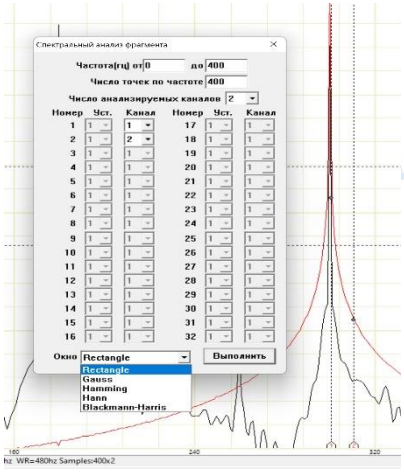


Pic. 24 Recorded signals

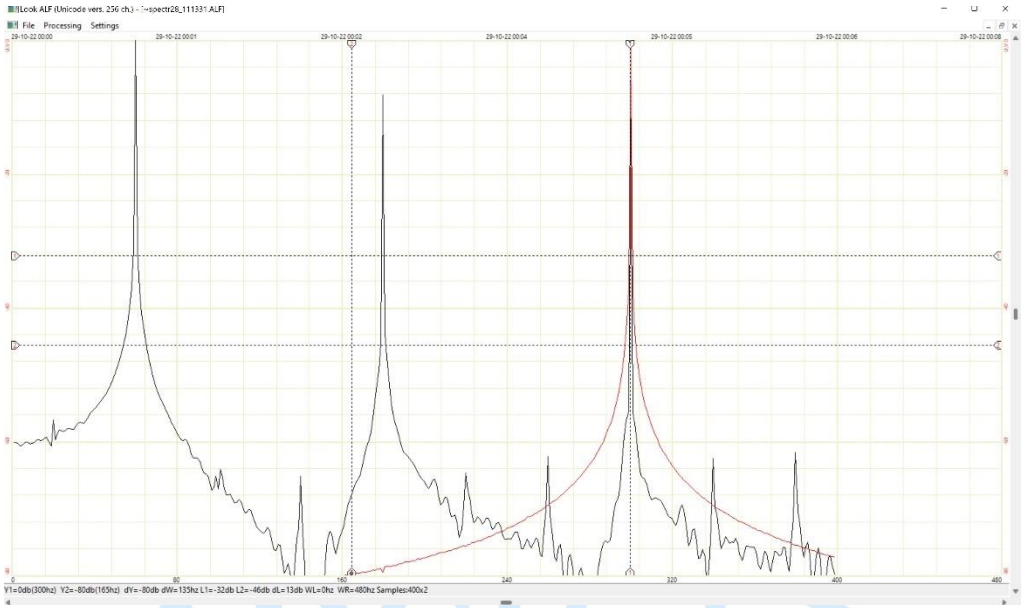
We put marks on the processed time interval,



Pic. 25 Time marks Y1, Y2 of the investigated recording interval



- Enter the estimated frequency of spectrum decomposition (in the example from 0 to 400 Hz);
- Number of points per unit of time (preferably a multiple of frequency)
- Select the number of analyzed channels (2);
- Select the device and the number of the interest channel (in this case, there is one device, the number of channels is 2, we selected the first and second channels, respectively);
- Select the window function "Rectangle" and click “Execute”.



Pic. 26 The result of the spectral analysis function "Rectangle»

As a result, we obtained a signal spectrum decomposed by frequency from 0 to 400 Hz (along the abscissa axis) by amplitude in dB (ordinate axis).

The sine spectrum (the red curve of the graph) has one peak with smooth spreading of the side lobes, where 0 dB is at the peak. The spectrum of rectangular pulses (black curve) has several peaks at different frequencies.

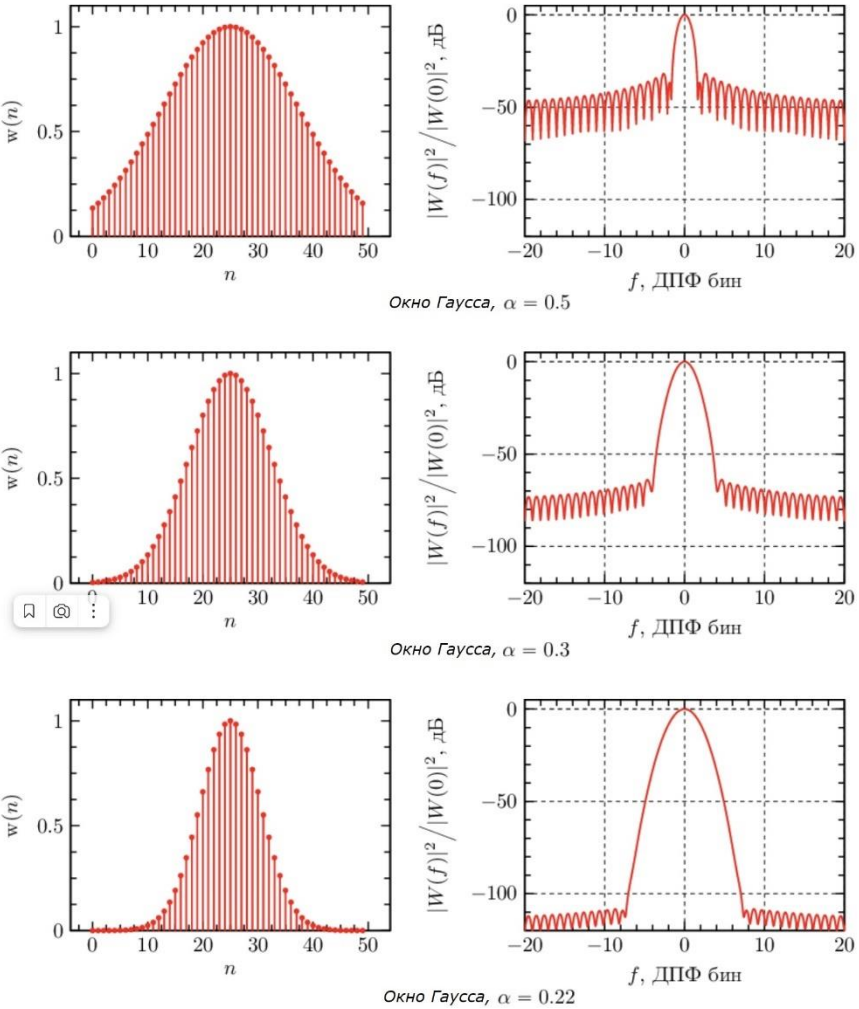
When performing spectral analysis functions, it is desirable to process only signals that are similar in parameters, or individually. In the case of processing signals of different types, incorrect scaling is possible, since the program scales the Y scale according to the signal that is selected in the "Parameters" section to set the labels Y1, Y2.

- **«Gauss» windowed function**

Parametric Gauss windows allow you to change the width of the window and obtain a different level of side lobes of the spectral energy density of the window function using the parameter α:

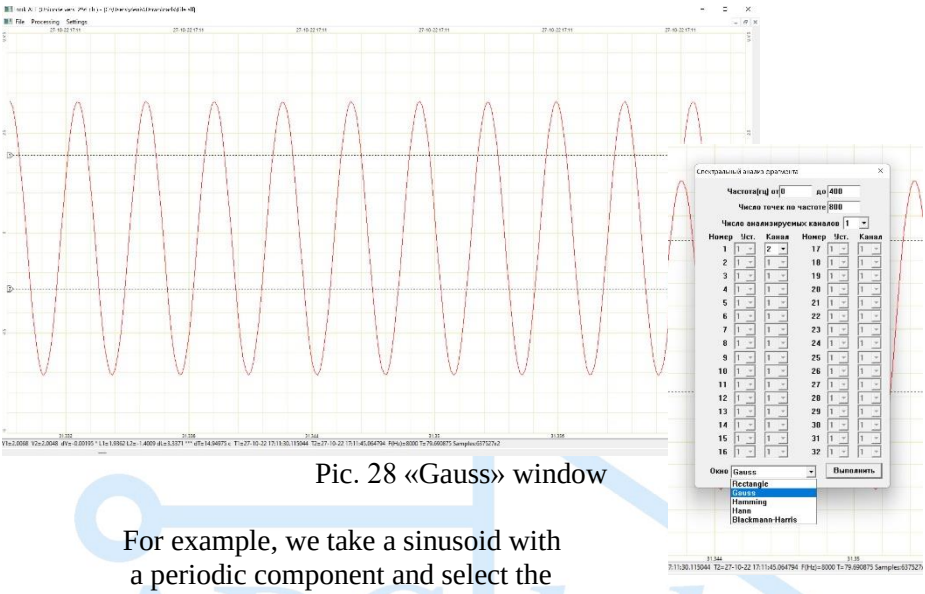
$$W(n) = e^{-\frac{1}{2}(\alpha\frac{n}{N/2})^2} \quad -\frac{N}{2} \le n \le \frac{N}{2}$$

N – number of counts, α – set parameter, n – frequency
The windows width is inversely proportional to the parameter *α*



Pic. 27 Changing the spectrum by parameter *α*

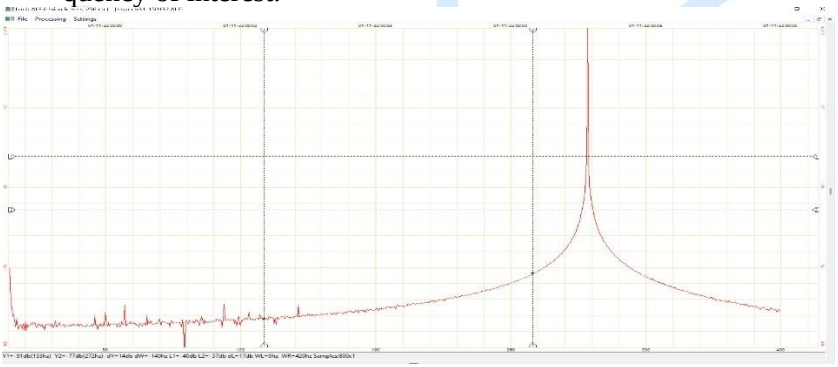
In the window of the fragment spectral analysis, similarly to the previous function, we configure the parameters and select the "Gauss" window



Pic. 28 «Gauss» window

For example, we take a sinusoid with a periodic component and select the appropriate channel, the frequency of spectrum expansion, scaling (number of points) and the Gauss Window in the parameters.

As a result, we obtain a signal spectrum decomposed by the frequency of interest:



Pic. 29 Results

- **«Hemming» windowed function**

The Hamming window uses the period of the raised cosine. In this case, the displacement coefficients and cosine amplitude are equal 0,5:

$$W(n) = \alpha_0 - (1 - \alpha_0) \times cos(\frac{2\pi n}{N})$$

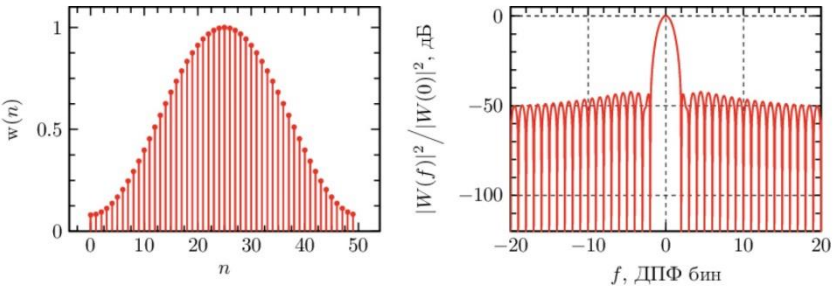


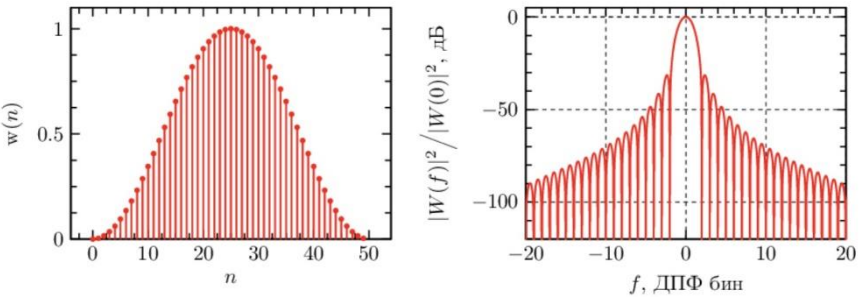
Рис. 30

The Hemming window has good frequency properties – a low level of side lobes and a small window width.

- **«Hann» windowed function**

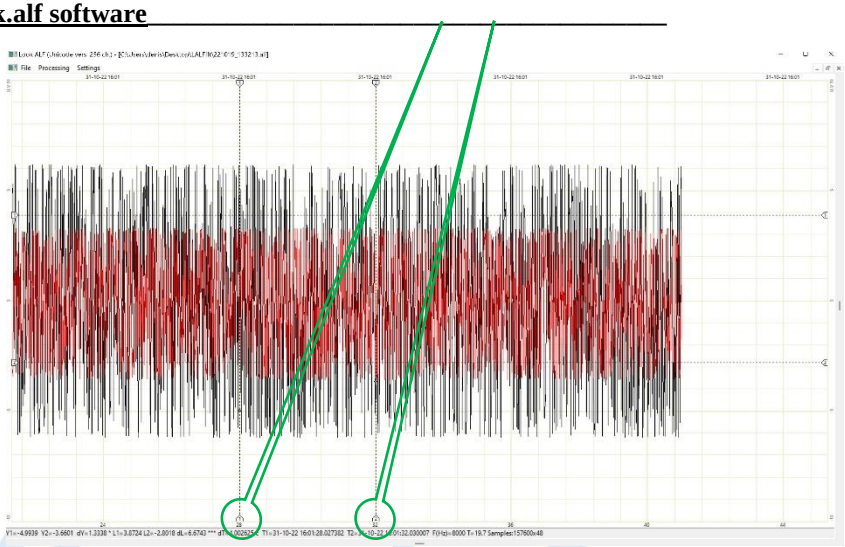
Hann window represents one period of the raised cosine.

$$W(n) = 0{,}5(1 - \cos(2\pi \frac{n}{N})), 0 \le n \le N$$



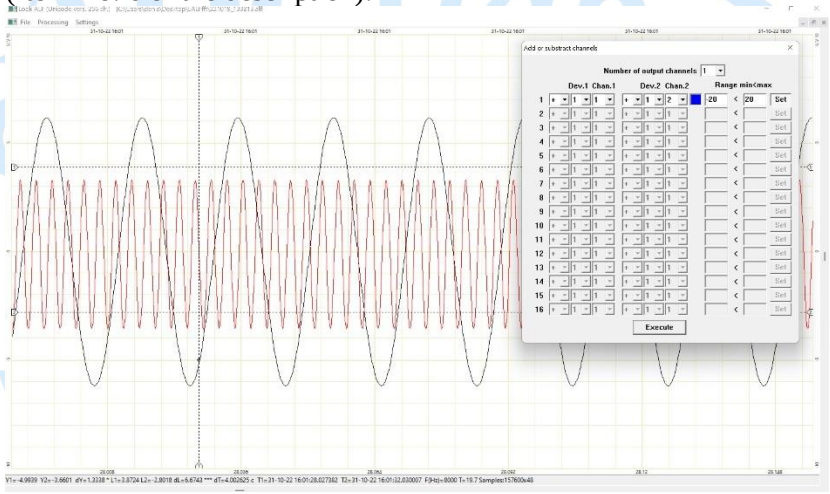
Pic. 31

Let's perform the task of decomposing two sinusoidal signals into a spectrum by adding them into one. To do this, we open the recording of two signals with different sampling frequency and amplitude. We set the marks Y1, Y2 for the analyzed area in time:



Pic. 32 Setting marks Y1, Y2 in time

For clarity, we scale the signal and perform the "Addition" function (Item No. 6 of the description):



Pic. 33 Addition of the first and second channels

We get the result of the addition function in a new window in the form of a single signal:

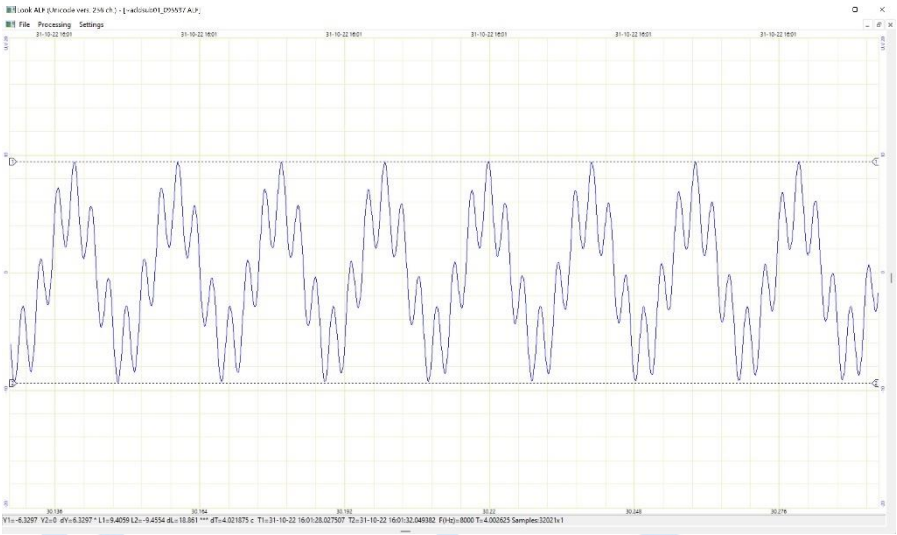
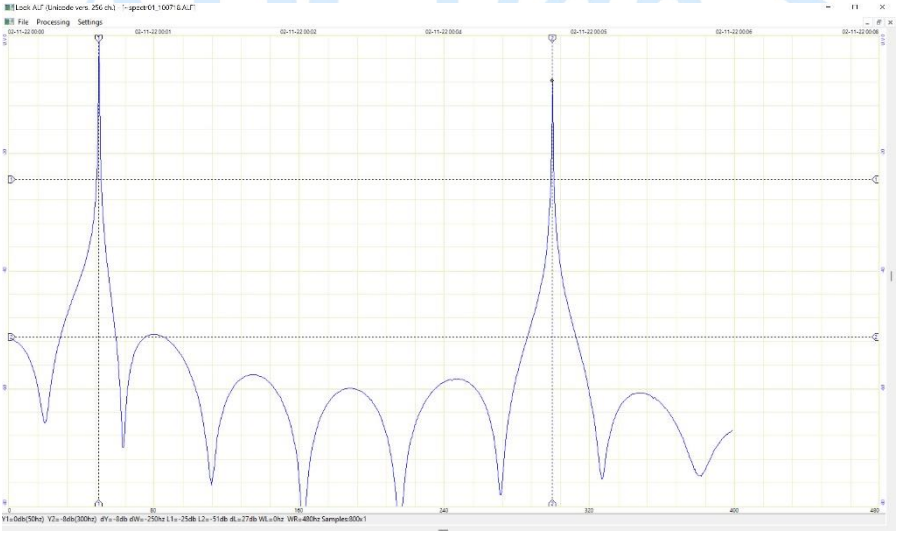


Рис. 34

We decompose the signal using the spectral analysis function « Hann»:

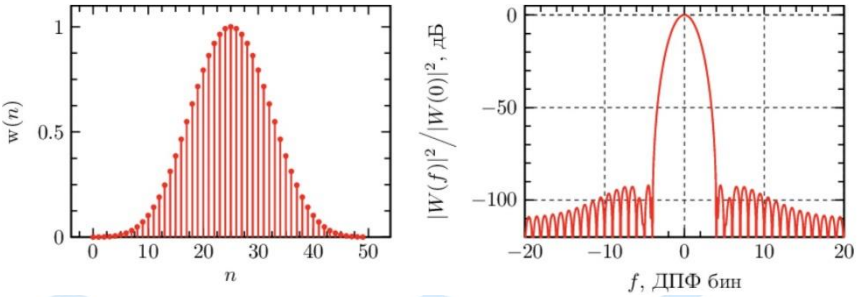


Pic. 35 The result of signal processing (pre-folded sinusoidal signals) with peaks when decomposed by frequency from 0 to 400 Hz

- **Windowed function «Blackman-Harris»**

Adding cosine summands and optimizing coefficients with four summands. The level of the side lobes of the spectral plane of energy decreases, but the width of the main lobe increases.

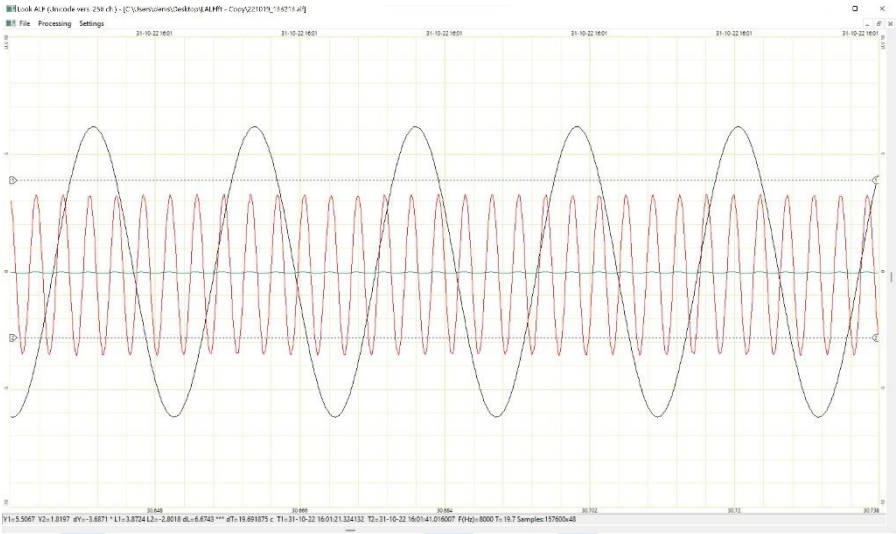
$$W(n) = \alpha_0 - \alpha_1 \cos\left(\frac{2\pi n}{N}\right) + \alpha_2 \cos\left(\frac{4\pi n}{N}\right) - \alpha_3 \cos\left(\frac{6\pi n}{N}\right)$$



Pic. 36

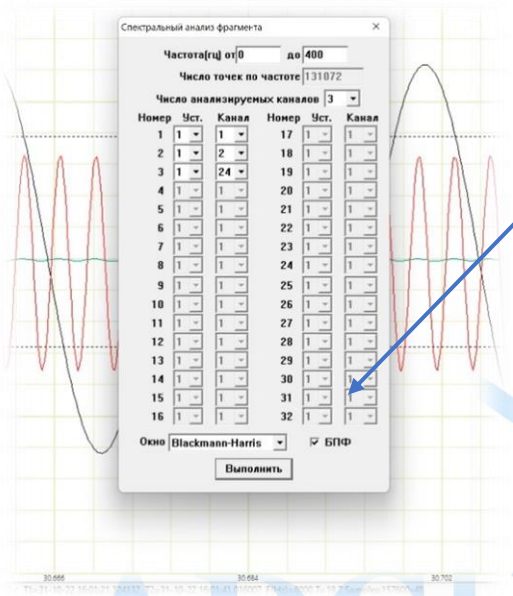
- **Fast Fourier transformation (FFT)**

Fast Fourier transform is an algorithm for accelerated calculation of the discrete Fourier transform, which allows to obtain the result in a short period of time. Allows you to express the length of the DFT through a certain number of transformations of shorter length using recursion, thus reducing the complexity of calculations.



Pic. 37 Signals to be processed using discrete Fourier transform

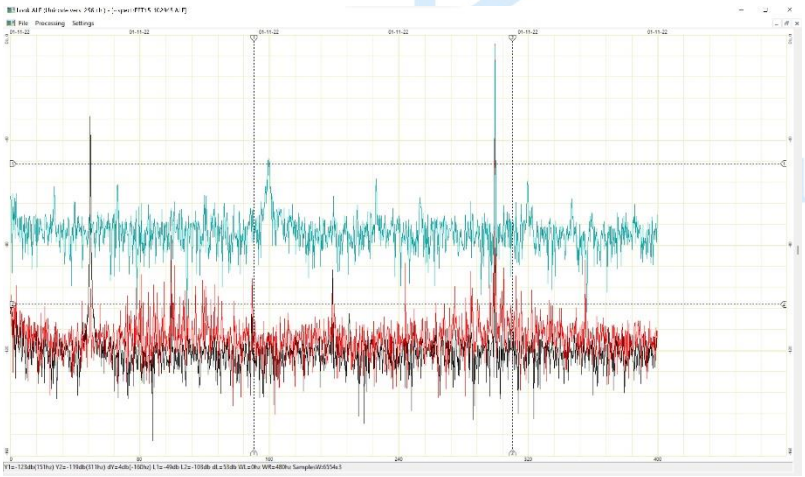
It is advisable to use the FFT to calculate a large number of channels with a long interval (it is necessary to set the marks Y1, Y2). It is also possible to use FFT with weak computing capabilities of a working PC.



To perform the fast Fourier transform function, it is necessary to press «v»

Pic. 38 Using the FFT

When using the fast Fourier transform, the number of points decomposed by frequency is set in an automatic mode. Calculations are performed in accelerated mode.



Pic. 39 Results

The program provides the ability to save recorded and processed files in several formats:

- .alf – ADClab file format that can be opened and processed in various programs;
- .txt – a text format that is convenient for converting information in various programs from simple Notepad, MS Excel to more complex viewing and processing tools.
- .wav – audio file format to be processed in a large number of tools.