MULTIFUNCTIONAL ANALYZER

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We describe here the hard and software complex intended for studying characteristics of the stochastic processes in the turbulent atmosphere. Its own database allows the initial data and results of processing to be stored. A variety of digital filters extends the potentialities of the spectral analysis. The spectral, correlation, and principal statistical characteristics are calculated.

Propagation of optical radiation through the atmosphere makes up a number of demands on the instrumentation for acquiring the atmospheric parameters and programs ensuring input, storage, and processing of information, namely, simultaneous input of information on several channels, wide dynamic range of the analog-to-digital converter (ADC), high frequency resolution, sufficiently large volume of input information, operative archiving of the initial data and results of processing, and adaptable capabilities for making statistical processing.

To solve these problems the hard and software complex "Analyzer of random signals" based on computer, 8-channel system of analog information input, and 12-digit analog-to-digital converter was developed.

The complex is prepared for functioning before the experiment. The preparation consists of calibration of all channels including introduction of coefficients and dimensions and program correction, removing additive errors of the devices connected with the complex.

A software of the complex is a package of independently operating programs (Fig. 1).

The program for information input allows the data array up to 64 Kb to be entered during one cycle of information acquisition for calculation of histograms to be accumulated when the measurements take several hours. The information may be entered in different ways, i.e., the oscillographic mode with the external and internal synchronization and the mode when a sample length exceeds 512 values (just as the length of the graphic screen where the entered information is presented), in this case the data are recorded when the screen is turned off. Sampling frequency, number of channels, and consecutive number of the channel are chosen visually by a user. The range of sampling frequency is from 0.001 Hz up to several tens of kHz.

The system has its own data archive. Operation of the archive is based on the original structure of data storage organization: the series consists of the experiments which, in their turn, consist of separate

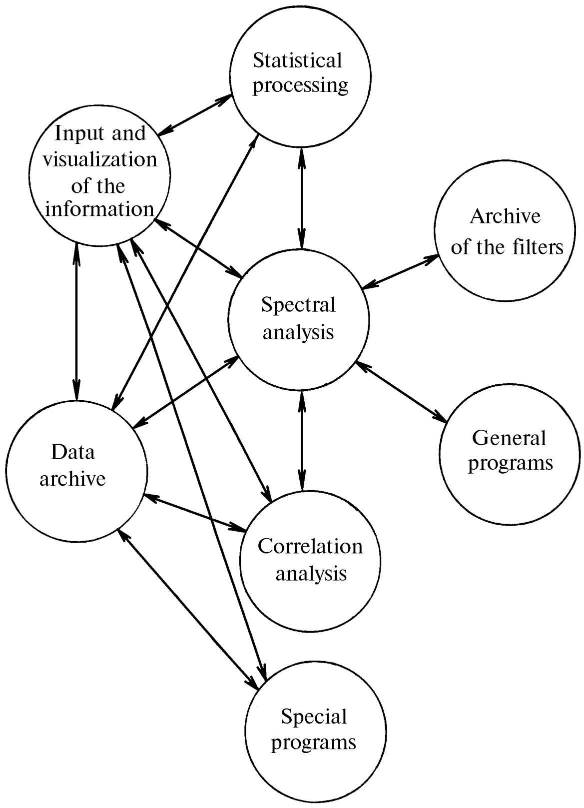

FIG. 1. Structural scheme of the software package "Random Signals Analyzer".

inputs. In the final result the data input is a file where the data array and accompanying information, i.e., sampling frequency, sample length, and recording channel numbers are written. First the labels of the series and experiment are created, i.e., the necessary test information on the experimental conditions is recorded. The program automatically creates the directories of the series, experiments, and inputs which are accessible also from out of the program and makes it possible to record on the hard disk up to thousand inputs during the experiment. Check of the stored

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inputs is possible on the hard and floppy disks and poor inputs can be erased if necessary. Extracting data file from the archive or direct data entry from the ADC assumes entering the program for the statistical and spectral processing of the information acquired.

The program calculating the signal power spectrum by the Fast Fourier Transform operates with the sample from one channel. But the program calculating the histograms allows the histograms of the samples of any number of channels to be created and simultaneously displayed on the screen. Here the statistical characteristics of samples are also calculated and correlation analysis is carried out.[1,2] In the mode of initial information processing, the curve presenting the input sample can be filtered or differentiated. Moreover, the parameters for calculation of the filter pulse characteristic are assigned by the user himself. Having filtered or differentiated the curve, the user can look at the spectrum or histogram of the curve. The curve itself, results of filtration and differentiation can be formed into an array which is a final result. There is also a possibility to create an archive where most frequently used pulse characteristics of the filters can be stored.

Various types of the digital filters with the finite pulse and linear phase characteristics are created in the program. The weighting method[3] was used to calculate these characteristics. The four-term "minimal" cosine window[4] was used as the weighting function. This window has an extremely low level of side lobes (the first side lobe has a value of −98.17 dB) that allows such filters to be successfully applied to extract weak signals on the background of strong noise. As a results, the following filters with the symmetric pulse characteristics (PC) were synthesized: the low-pass and high-pass, band-pass, rejection, and also much more complicated multiband filters with several bands of transmission and rejection. The differentiating filters of the first (with the antisymmetric PC) and second (with the symmetric PC) orders were also calculated. Moreover, the differentiation is carried out within the requested frequency band of the signal presence with the simultaneous suppression of noise in the other frequency bands. The corresponding cut-off frequencies, sampling frequency, and PC length are the input parameters of the procedures of the filter calculations. Quality of the filters is tested by calculating their amplitude–frequency characteristics over the entire Nyquist interval. The filtering itself is carried out following an unstable nonrecursive scheme, i.e., by direct convolution method allowing for the PC symmetry type with simultaneous neutralization of the time delay introduced by the filter (Fig. 2).

The program allows the user to exchange the information with the other widely used programs (for example, STATGRAF, GRAPHICS, etc.). At the same time, all graphic screens being formed in the program can be printed out.

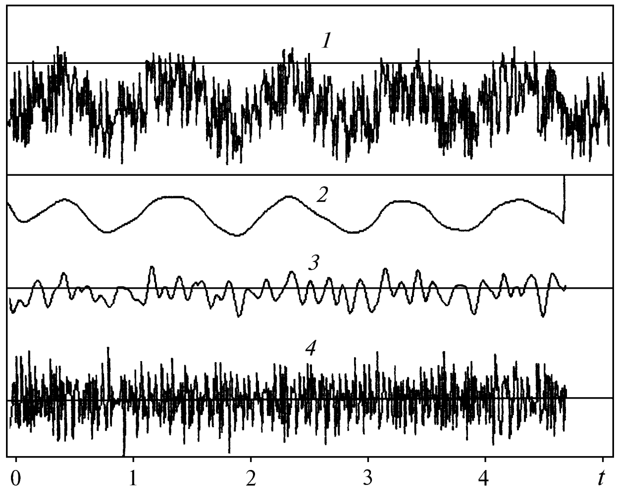

FIG. 2. Versions of the initial signal filtration: the initial curve (1); the result of processing of the initial curve by the low-pass filter with the cut-off frequency of 2 Hz *(2); the result of processing of the initial curve by band-pass filter with the transmission frequency of* 2–10 Hz *(3); the result of processing of the initial curve by the high-pass filter with the cut-off frequency of* 10 Hz *(4).*

The program provides a user the following modes to study the curve visually in all graphic screens: point-by-point analysis, scaling of image, magnification. Architecture of the program is such that it can include additional modules without essential changes of the basic program.

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