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Measuring System for Automation of Experiments

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Abstract

A LabVIEW system that shows high flexibility and may be adjusted to realize wide range of digital signal processing algorithms is described. Analyses of signals both in time and frequency domains is possible, including DFT, signal filtering, analysis of amplitude probability distribution and autocorrelation function of the signals.

Keywords: Real time digital signal processing, filtration, discrete Fourier transform, autocorrelation.

1. INTRODUCTION

Carrying out researching of properties of new materials, physical phenomena and various signals need to designing of measuring systems, which in each case have various structure and parameters. Such system can be created on the basis of:

- Set of measuring devices;
- High-level languages;
- Specialized software.

Each of methods has the merits and lacks. If measuring system is assembled on the base of standard measuring devices, the basic problem is writing of the software that unites elements into system and realizes necessary algorithm of parameters measuring,

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and, if it is necessary – control the investigated object. In this case, changing research object leads to essential reorganization of instrumental system, including replacement of devices, as the range of measured units or analyzed parameters will be changed. In such systems the minimum works on programming of hardware equipment is necessary.

Realizing a measuring system at program level with use of standard algorithmic languages, one of the basic problems is the software designing continuity. If the team of developer's will be changed, it's easier, in many cases, to write new corresponding program modules, than to understand ready codes of programs. The second problem is calibration of gauges and converters of physical signals into electric signals and digital codes. It is necessary to add, that these systems provide the higher flexibility, if will be altered the values to be measured or range of their parameters. However, realization of this idea needs much more time, comparing the first method.

Advantage of the third method is the maximum simplicity in description of measurement processes, as a typical system is represented as a simple connection of standard units in the form of graphic elements that may be changed, if the system structure will be modified. Such systems can be created with use of special graphic elements in environment of SIMULINK or LabVIEW, specially focused on use of these systems for carrying out of various physical experiments.

Essential advantage of LabVIEW is the possibility of creating of the system which is represented on computer display screen as a typical measuring device. Another advantage is the possibility of modular construction of system by principle "from simple to complex".

In the given paper the example of realization of the system created in the LabVIEW environment that provide carrying out of measurements of parameters of electric signals both in time and in frequency domains is considered. The offered system possesses high flexibility and can easily be reconstructed in a wide range of parameters, which will be measured.

2. MEASUREMENT OF SIGNAL PARAMETERS

2.1. Description of the system.

The system is constructed by a modular principle. Each of modules provides special algorithm of signal processing according to certain algorithm. The system includes following modules:

- Spectral analysis unit;
- Filtering unit;
- Display unit;

- Autocorrelation function unit;
- Probability density unit;
- Control unit;
- Signal simulator unit.

Parameters of each module may be changed in wide range both to change mode of operation, and to change the mode of indication of results. Adjustment and calibration of the system is carried out by means of a simulator of signals, in which three sinusoidal signal oscillators and the noise generator (with uniform and normal spectrum distribution) are present. Composition of signals is applied to the input of the filter's unit or may be directly connected to the input of spectral analysis unit. The effect of the filtration can be observed both in time, and in frequency domains. Parameters of the spectral processing of signals may variate in wide range, changing number of the samples in analyzed array and the type of a time window. Calculation of autocorrelation function and probability distribution of a signal are additional options of the system.

2.2. Research of spectral characteristics of a signal

In the given system researching of signal characteristics in frequency domain is based on Fast Fourier Transform (FFT) algorithm. The spectrum of the signal presented in discrete time is described by equation (the designations of [1] hereinafter are applied)

$$X[k] = \sum_{k=0}^{N-1} x[n] W_N^{kn}, \qquad k, n = 0,1,\ldots, N-1, \tag{1}$$

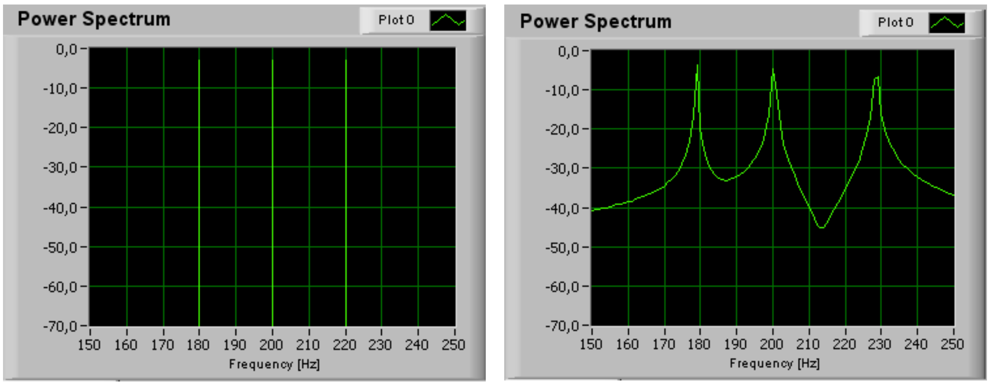
where k, n – the integer variables, x [n] – array of samples in time domain, X [k] – array of spectral samples, N – number of samples in analyzed sequence, W – weighting coefficient (complex exponentials).

If the signal contains the only sinusoidal component of frequency f_s, and the following condition is satisfied

$$f_S = p\Delta F\ , \tag{2}$$

where p – integer variable, ΔF – frequency resolution; then its spectrum will be presented by the only sample in frequency domain. If (2) is not satisfied, the view of the spectrum essentially changes.

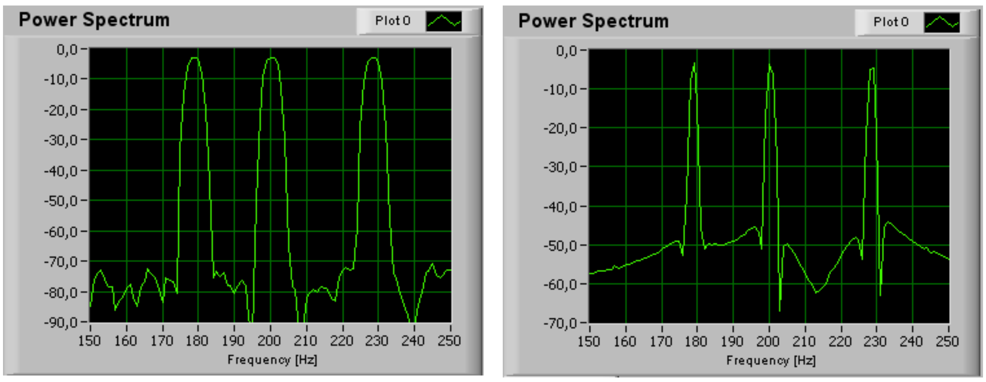
The signal consisting of the sum of three sinusoids with frequencies 180 Hz, 200 Hz and 220 Hz each of which satisfies (2) is applied to the unit input to demonstrate this effect (Fig.1a), and the spectrum of a similar signal where this condition is not upholded is shown on Fig. 1b.



(a) (b)

Fig. 1. A spectrum of the signal containing of three sinusoidal components

Such effect of spectrum transform visualization is well known to experts in digital processing of signals. To minimize it is recommended to weigh signal in discrete time applying various time windows. In the given system nine special windows can be applied, which parameters are well studied, for example, in [2]. The Fig. 2 illustrates spectrum of the signal weighted by Flat Top (a) and Hamming (b) windows. The signal, which spectrum is shown in Fig. 1b is analyzed. Expansion of the main petal of a spectrum of each component of a signal is observed, but also increases the dynamic range of a spectrum representation. Each window gives specific shape of spectrum, and its choice is carried out by experimental selection, or on the basis of empirical experience.

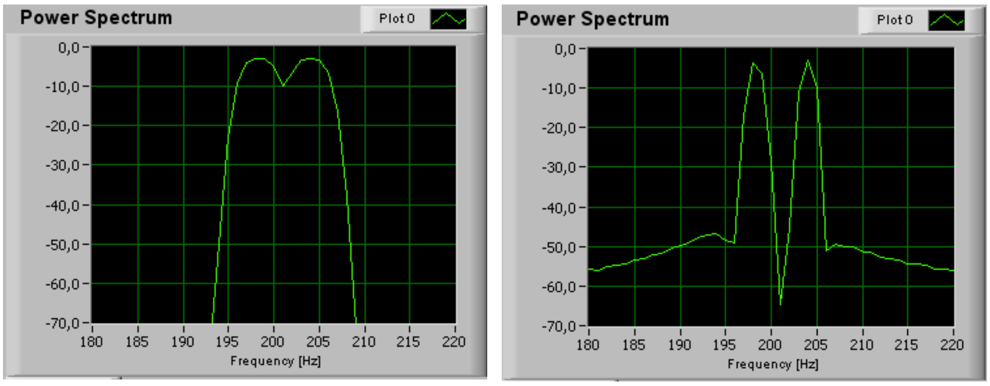


(a). Flat Top Window (b). Hamming Window

Fig. 2. Influence of a time window on the spectrum shape

The choice of a time window is especially important, for example, if the signal components are very much close on frequency. As an example, Fig. 3 shows spectra of a mix of sinusoidal signals with frequencies f_1 = 198.22 Hz and f_2 = 204.47 Hz.

Additional option of the system is possibility of flexible control of process of averaging, which is important if the signal composed of significant level of noise. The researcher has possibility to choose averaging parameters experimentally, observing results of processing in real time.



(a). Flat Top Window (b). Hamming Window

Fig. 3. Effect of a time window on spectrum resolution

2.3. Signals filtering

The filtration of signals is the important element of research of parameters and characteristics of signals. By means of filtering it is possible to eliminate an unnecessary or needless part of a spectrum of a signal, before the subsequent analysis of the allocated part of a spectrum will be started. This kind of processing has especially great importance if useful components of a signal have considerably smaller amplitude, than the other part of a signal, and if the analyzed signal and disturbances (noise) are placed in various regions of a frequency range. Realization of the digital filter can be carried out in several ways. The general view of the mathematical description of the filter with the infinite impulse response characteristic (IIR) looks like

$$y[n] = \sum_{k=0}^{N} b_k x[n-k] + \sum_{k=1}^{M} a_k y[n-k], \tag{3}$$

where $x\,[n\text{-}k]$ – sequence on a filter input; $y\,[n\text{-}k]$ – set of samples on a filter output; M, N, n and k – integers; a_k and b_k – weighting filter coefficients. If all factors a_k are equal to zero, then expression (3) describes the filter with the finite impulse response characteristic (FIR) [3]. The signal on a filter output can be defined in frequency domain. In this case the output signal is defined from synthesis Discrete Fourier Transform (DFT) equation

$$y[n] = \frac{1}{N} \sum_{k=0}^{N-1} (X[k]F[k]) W_N^{-kn} \ , \tag{4}$$

where F [k] – frequency response samples of the filter. Changing shape of the frequency response and observing output signal both in frequency and in time domains, it is possible to define optimal parameters and type of the filter that is necessary for analyzed signal.

The wide range of filters types is included in system. An easy method in their adjustment and evident visualization of results of processing gives very good results for any signals. The control panel of filter parameters, containing a minimum quantity of elements is shown on Fig. 4.

To illustrate work of the filtering unit, the signal containing a composition of three sinusoids of identical amplitude with frequencies 50.63, 250.85 and 450.97 Hz is applied to the system input.

Fig. 5 illustrates spectrum of a signal before filtration (a) and after filtration with low pass filter (b), high pass filter (c) and band pass filter (d). System display simultaneously shows both frequency response of the filter and a spectrum of a filtered signal, additional display demonstrate the oscillogram of signal on a filter output in time domain, that allows to estimate results of signal processing in real time (not illustrated here).

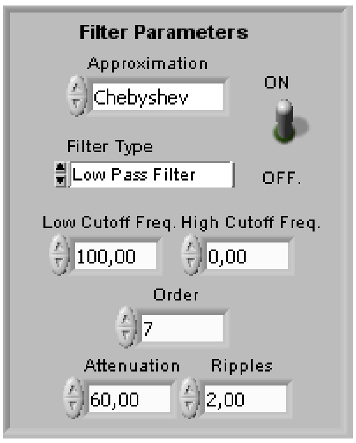


Fig. 4. The control panel of the filtering unit

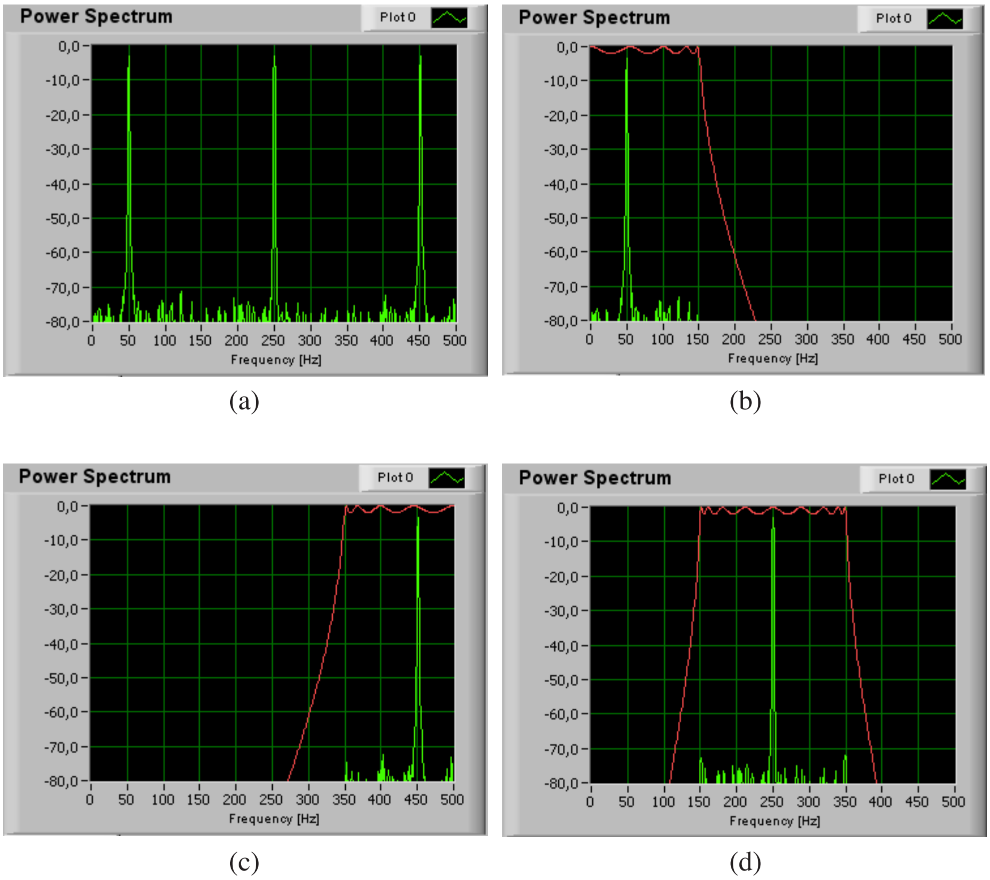


Fig. 5. A spectrum of a signal before and after application of the filters

7-th order filters on the base of Chebyshev approximation with out of pass-band attenuation of –60 dB and pass-band ripples of 2 dB are applied to all filters used in shown examples. Totally five various kinds of frequency response approximation are realized in system. Choosing a filter order, out of pass-band attenuation and in pass-band ripples it is possible to provide any requirements on clearing of real noised signals.

2.4. Calculation of the correlation characteristic of a signal.

Correlation characteristics of a signal have great importance for many practical applications. The estimation of autocorrelation characteristic $\phi_{xx}[m]$ can be defined in direct mode according to equation

$$\phi_{xx}[m] = \frac{1}{Q}\sum_{n=0}^{Q-|m|-1} x[n]x[n+|m|]\,, \tag{5}$$

where Q - integer variable and also is observed $|m| < N$-1. If the estimation of autocorrelation function is carried out in frequency domain, should be realized DFT synthesis equation algorithm

$$\phi_{xx}[m] = \frac{1}{N}\sum_{k=1}^{N-1} |X[k]|^2\, e^{j(2\pi/N)km}\,. \tag{6}$$

In described system the autocorrelation function on the base of N samples of a signal is calculated in real time and it is immediately displayed. The signal, which autocorrelation function is calculated may be thought as function weighed by a rectangular window. The example of autocorrelation function of the sinusoidal signal of 24 Hz is shown on Fig. 6, where triangular envelope shows effect of weighing of a signal by rectangular window.

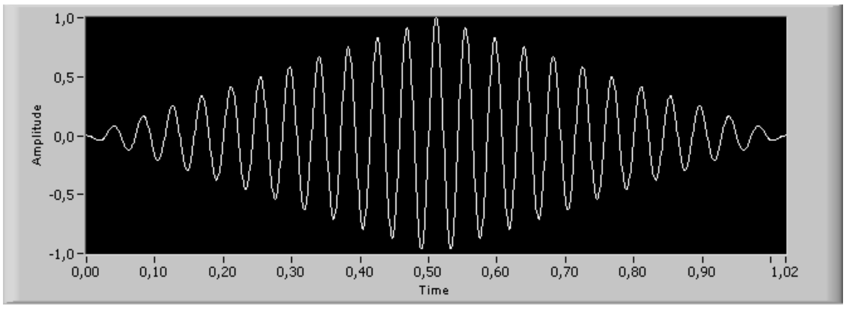
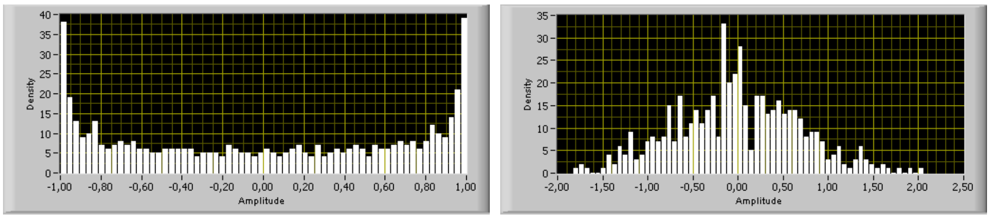


Fig. 6. The autocorrelation function of a sinusoidal signal

2.5. Probability density distribution of a signal.

Analyses of the stochastic process is important to learn probabilistic characteristics and the moments (mean and variance). The system provides displaying of the distribution function and calculation of the first and second moments of the investigated process. For the majority practical cases it is enough to understand properties of a signal.

Fig. 7 illustrates distribution of a sinusoidal signal (a) and white noise (b). Changing subsystem parameters allows establishing necessary accuracy of calculations and visualization of results. Example shows signal magnitude segmentation on 64 levels, which is sufficient to estimate the real signals.



(a) (b)

Fig. 7. Histograms of density distribution of a signal

3. CONCLUSIONS

The considered signal processing system realized in LabVIEW shows efficiency of the software for working out and designing of the systems oriented on measurement of parameters and characteristics of signals, researching new processes, materials and working out the technique of the tests of created products. The considered technique can be successfully applied to the spatial multidimensional analysis of signals [4]. Working out of such systems does not demand the big expenses of time and finances, and at the same time provides the high flexibility if should be made any changes in system configuration.

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