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A VIRTUAL PERIODMETER FOR SIGNALS BURIED IN NOISE

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Abstract: The paper presents an instrument designed to measure the period of signal buried in noise and having a high degree of distortions. It works as a virtual instrument implemented on a personal computer that receives the digitized signal from a digital acquisition card or from a digital oscilloscope through the GPIB interface. Its operation principle is based on a new algorithm developed for assessing the period of the input sampled signal by comparing a witness segment of samples with the rest of the samples that compose the signal, till they overlap the best. The method, as well as the instrument was tested in laboratory and the results were compared with those obtained by using other known methods.
Keywords: noise, period, virtual instrument

1 INTRODUCTION

There are a lot of situations when knowing period is of great importance, especially when it is difficult to assess due to irregularities of the signal caused by noise and strong distortion. Some of them can be enumerated: signal recovery from noise by using digital lock-in amplifiers, speech recognition, event diagnostic and prognosis in electric power systems, etc. Another example where knowing period is also crucial will be detailed further in this introduction.

It is well known that application of DFT or FFT to a sampled signal that have the length different from an integer number of periods leads to incorrect calculation both of amplitude and phase of spectral components, due to leakage phenomenon [1,2]. An amelioration of it can be achieved by employing smoothing windows or by synchronous or coherent sampling, in which the sampling frequency must be synchronized to the fundamental frequency of the signal to be analyzed [3]. The better the fundamental period is known, the more accurate Fourier coefficients can be established.

Another method for eliminating the leakage errors of periodic signal spectra is proposed by Sedlacek [4,5]. This method is based on finding first the signal period and then interpolating and resampling the measured waveform in such a way that there are exactly N samples per period, where N is the number of FFT points used. The interpolation is generally linear, but also polynomial and cube spline functions can be used. This method provides very good results, but some time is wasted in interpolating and resampling phases. However, this is the most used method for signal spectrum determination when it is no requirement for speeding the processing.

Several methods for period determination exist. Among them, autocorrelation, digital filtering, digital Fourier transform, power spectrum, Cepstral analysis and Hilbert transform are more used [6]. All of these were implemented in the same virtual instrument in order to compare their performances with those of the new method proposed in this paper.

2 PRINCIPLE OF THE METHOD

The period determination is made by calculus performed by a computer. The signal is supposed to be sampled and stored into the computer memory in a vector form. All the calculus is accomplished in time domain.

We assume that we have already in the computer the signal under investigation, formed by n samples acquired with the sampling frequency f_s . The method consists in comparing a witness sample sequence (denoted here as witness segment) that generally represents the first m samples of the acquired signal, with the rest of n-m samples that compose the signal, till they overlap the best. The overlapping degree is assessed by using two criteria: mean square error and mean absolute error. A suggestive graphical representation of the method is shown in figure 1.

Assuming N to be the number of samples in a period, the total number of samples acquired, n, must fulfill the condition

$$n \geq N + m \tag{1}$$

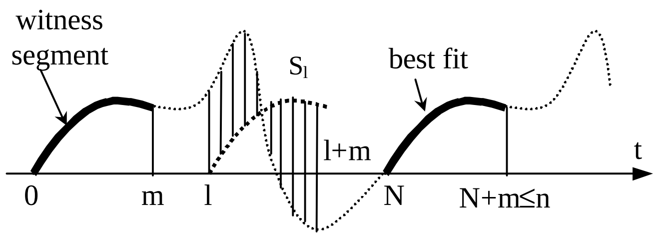

Figure 1. Graphical illustration of the method

The method requires the calculation of one of the following two parameters, at choice:

$$S_{l_{ms}} = \frac{\sum_{i=1}^{m} (s_{i+l} - s_i)^2}{m} \tag{2}$$

in the case of mean square error (MSE) criterion, and

$$S_{l_{ma}} = \frac{\sum_{i=1}^{m} |s_{i+l} - s_i|}{m} \tag{3}$$

in the case of mean absolute error (MAE) criterion. In the above equations, $0 \le i \le m-1$ and $k \le l \le n-m$ for every *m* adjacent samples in the signal. We also symbolized s_i as the i^{th} sample of the signal.

Once the array S_l is built (further denoted as *error array*), we choose from it the smallest value that represents the place where the witness segment overlaps the best upon the signal. The index corresponding to this place represents the desired period.

Obviously, this place may not be that of the witness segment. That is why we choose to begin the array calculation not from 0 but from an integer *k*, where *k* is not allowed to exceed *N*. On the other hand, if $n \ge 2N + m$ (i.e. one acquires more than two periods), the smallest value of S_l might correspond to the second period acquired, especially when the signal we deal is very noisy. In order to avoid this mistake, we have to distinguish the first minimum from the second one, which is in fact a very difficult task. The things become more complicated if the parameter S_l has more minimums having very closed values, as shown in figure 2. Due to the stochastic character of the noise, any of the points 1, 2, 3, 4 or 5 could be the global minimum, thus leading to erroneous results. This problem can be overcome if one increases the length of the witness segment *m*. In figure 2 b) the error array is calculated taking m = 300 samples, whilst in figure 2 c) the same array is obtained with m = 1000 samples. The longer is *m*, the more pregnant is the minimum corresponding to the signal period among other local minimums. The reverse of this outcome is an increasing in the computing time.

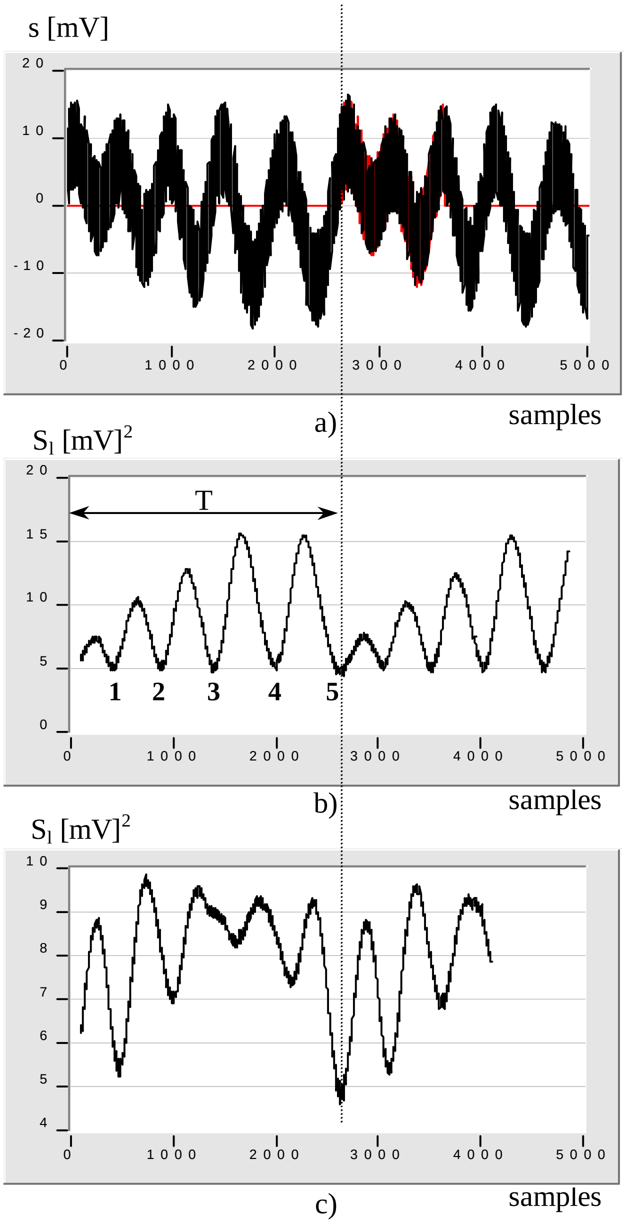

Figure 2. a) The signal represented in time domain (5000 samples), b) error array when m = 300 samples and c) error array when m = 1000 samples

3 INSTRUMENT DESCRIPTION

Two variants of the instrument were tested in the laboratory. They differ to each other by the device employed for acquiring the input signal.

3.1 Variant I

In this case, a digital oscilloscope type Keithley TDS220 is utilized for acquiring the signal with the sample rate up to 1 GHz and for sending the samples to the host computer via GPIB interface. The oscilloscope samples the signal by using an 8-bit ADC. The total number of samples acquired is 2500 per frame. The advantages of this variant are: the possibility of adjusting the time base of the oscilloscope so that one acquires at least one period of the signal for fulfilling condition (1) and large range of input signal frequency to be measured (up to 20 MHz if working in real time). The disadvantage consists in more time consuming for sending data to the host computer via GPIB interface and in higher price, since two additional devices (oscilloscope and GPIB controller) are involved.

3.2 Variant II

This time, a plug-in digital acquisition board (DAQ) type National Instruments PCI-MIO-16E-1 performs the signal acquisition. Due to the lower sampling rate of this device (1.25 MHz), the dynamic range of the input signal frequency is lower than in variant I (depending on how accurate the period measurement is desired). However, owing to DMA data transfer, this task is almost instantaneously accomplished, allowing the computer to provide faster the result.

3.3 Virtual instrument presentation

The virtual instrument was designed in the National Instruments LabVIEW 5.1 package. Its main functions are: i) driving the data acquisition from oscilloscope and DAQ board, ii) processing the signal according to the algorithm in order to obtain the period; iii) displaying the results. In addition to these functions, some other utilities were added to the instrument in order to control and test its capabilities. Measuring computing time and error assessment when using other methods, graphical time domain signal and error arrays displaying and distortion coefficient computation are some of these additional functions.

The logic diagram of the instrument is presented in figure 3. In figure 4, the front panel of the virtual instrument operating under variant II is shown. It allows the operator to select n, m, f_s, k as well as the DAQ parameters like channel, buffer length, amplification, trigger level. One of the following methods: autocorrelation, filtering, cepstrum, Hilbert transform, Fourier transform and our proposed algorithm can be chosen from a controller. As output indicators, signal period and frequency, computing time, method error, signal distortion coefficient, signal to noise ratio, as well as waveform graphs for signal and error arrays are established on the front panel.

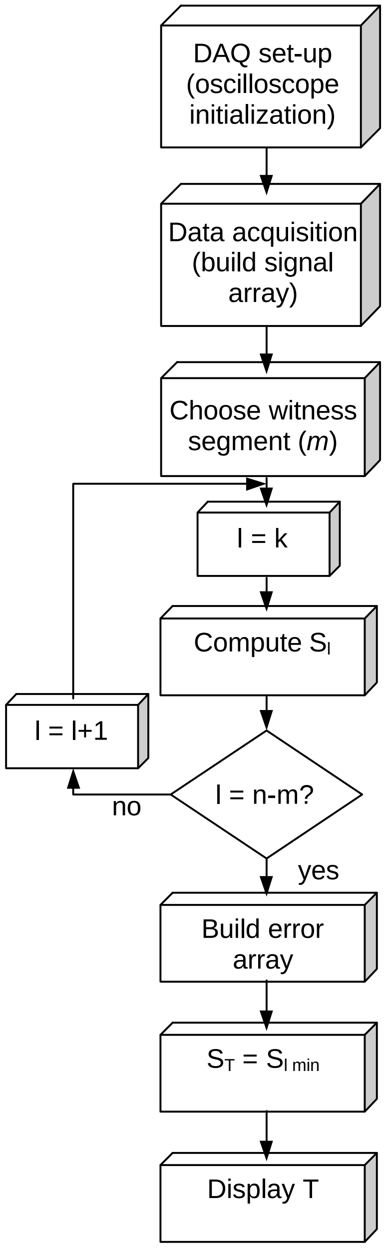

Figure 3. Flow chart of the virtual instrument

4 RESULTS AND ERRORS

Both variants finally provide the sampled signal available in the computer memory as a vector. As we have already shown, it is necessary to approximately know the signal period in order to fulfill the condition (1). This condition can be written in terms of the signal period T, the sampling frequency f_s and the number of samples to be acquired n as

$$n - m \geq T f_S \tag{4}$$

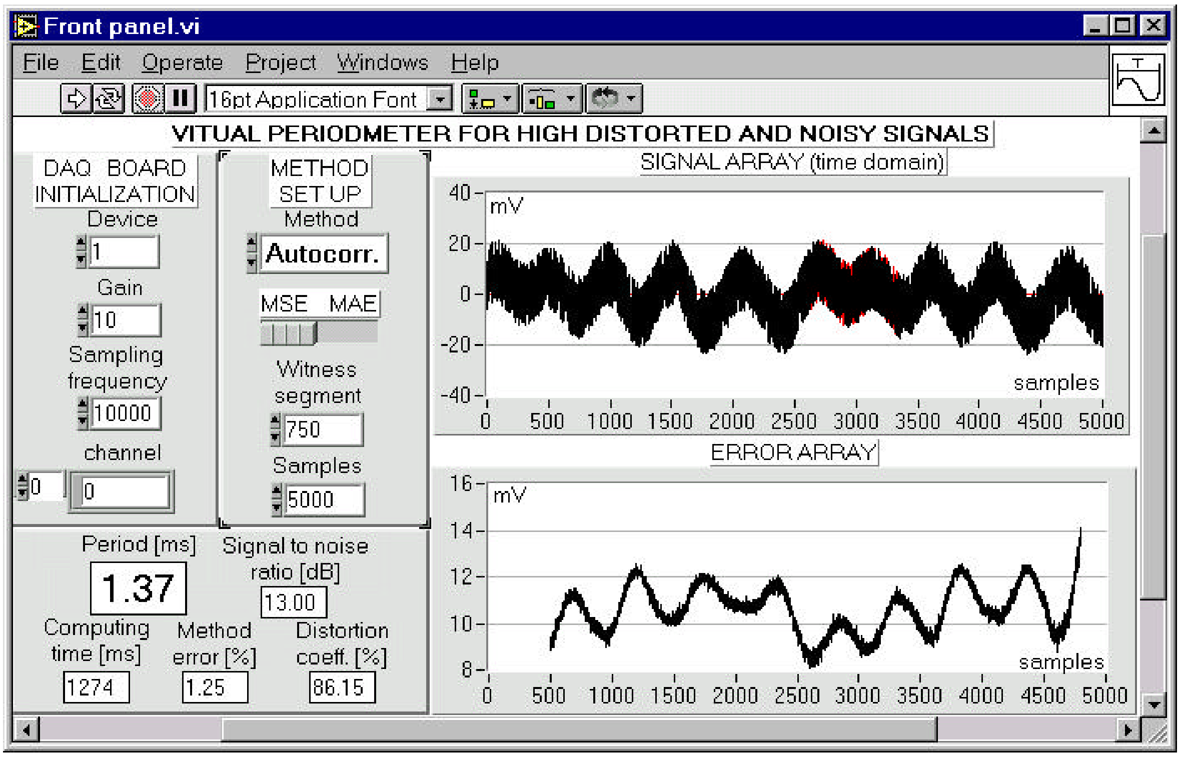

Figure 4. Front panel of the virtual instrument

On the other hand, from the Nyquist condition, Tf_s ³ *2.* However, we have found from experiments that at least 100 samples per period should be taken.

The instrument testing was performed by using in input the signal delivered by an arbitrary function generator type Stanford Research DS345. On the front panel presented in figure 4, the first graph represents a signal having 86.15 % distortion and 10 zero crossings during a period. A uniform white noise of 7.3 V in amplitude is added to this signal. The signal to noise ratio thus obtained is 13 dB. In the same figure, the trace representing the array of variances S_l is drawn. The point where S_l takes a minimum value corresponds to the signal period. In table 1, the computing time and accuracy in period determination process for the classical methods implemented in the testing instrument in comparison with the proposed method are indicated. The parameters used were n = 10,000 and n = 20,000, m = 1,000, k = 400 and f_s = 100 kHz.

Table 1. Computing time [s] / error [%] obtained for different methods in comparison with the proposed algorithm.

# of samples	Autocorrelation	FIR Filter	IIR Filter	Cepstrum	Hilbert transform	DFT	Our method
10 000	4.23/2.35	2.75/1.96	1.38/2.24	3.36/3.22	11.5/3.00	1.65/2.33	0.98/1.78
20 000	7.98/1.78	4.15/1.35	1.59/1.67	6.15/1.98	21.5/1.55	2.55.1.45	1.87/1.70

The overall error essentially depends on the signal shape. If the signal has numerous zero crossings, the witness segment must be long, *m* taking values as much as *N*, thus resulting in a longer computing time. If this condition is not fulfilled, the witness segment might fit the signal in other points inside the period, and the result becomes completely wrong.

It should be noted from table 1 that in our algorithm, the error remains the same when increasing *n*, whilst in other methods the error decays when *n* grows. Figure 5 displays the dependencies of the computing time (a) and error (b) on the number of samples, measured for the same signal as above. Every time, *m* was taken 50 % of the number of samples per period. It can be observed that *n* can be decreased to about 2000 samples without significant increasing of error, but with serious computing time saving. It means that the signal frequency to be measured can be quite high even when employing DAQ boards, as described in variant II. This is one of the relevant advantages of the method.

However, the decisive role in minimizing the time and error is played by *m*. As can be concluded from relation (2), we need to perform *m* squares and *2m* additions for every element in the S_l array.

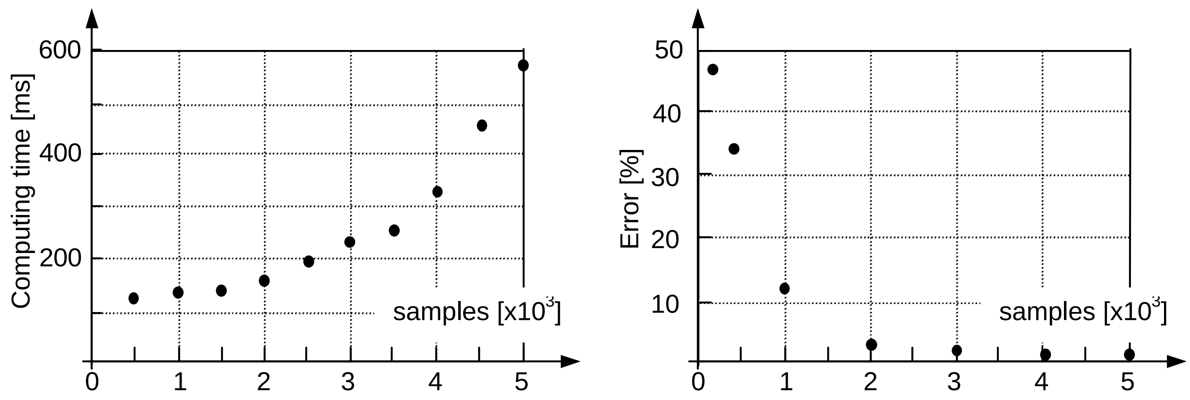

Figure 5. a) Computing time and b) error level vs. number of samples

Considering the squaring computing time much longer than that for addition, the overall computing time for a single element is therefore proportional to *m*. The entire array S_l, which has the dimension *n-m-k*, will be calculated in *m(n-m-k)* squaring times. This expression has a maximum at m_{max} = (n-k)/2.

On the other hand, the error decreases with the increasing of *m* because the longer *m* is the more information about the signal shape is stored in the witness segment. In table 2 the error *(å)* and the computing time *(ct)* are given in the conditions: n = 5000 , k = 300. One can therefore conclude that increasing the method's efficiency implies increasing *m* as much as possible, taking care to not exceed *N* and provided that the condition (1) is accomplished.

Table 2. Error (å) and computing time (ct) for different m when n=5000 samples and k=300.

m [samples]	250	500	750	1000	1500	2000	2500
å [%]	69.9	17.5	1.76	1.22	0.97	0.54	0.50
ct [ms]	286	529	755	956	1196	1378	1127

Regarding the criterion employed, MSE provides better results in terms of about 25 % error level lower than using MAE criterion, but is more computing time consuming, caused by the squaring operations. This difference becomes important when the signal under investigation has a very low signal to noise ratio.

5 CONCLUSIONS

An instrument designed to measure period for very noisy and distorted signals was presented in the paper. The instrument works according to an original algorithm that was proved to be less computing time consuming and more accurate than other algorithms known in literature. The virtual environment employed (LabVIEW) allows the instrument to operate in relation to different variants of hardware, leading to flexibility and low cost.

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