CONDITIONING THE LOW LEVEL SIGNALS FROM A VOLTAGE CONTROLLER

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Cu ajutorul unui program de instrumentație virtuală s-a dezvoltat o aplicație soft pentru achiziționarea şi prelucrarea semnalelor analogice de nivel mic. Avantajul major este acela că instrumentația virtuală poate modifica cu uşurință setările prin intermediul programului. Operațiile de mediere şi filtrare (filtrele Butterworth si Chebyshev) elimină cu succes zgomotul din semnalele de nivel mic (de ordinul milivolților).

The program described below develops many possibilities in order to facilitate reading and analyzing analog and digital signals. Using data acquisition boards, signals can be analyzed and measurement instruments can be created or simulated. Main advantage of virtual instrumentation is that it can be easily modified. Averaging and filtering (Butterworth and Chebyshev filters) operations are successfully applied for low-level signals.

Keywords: analog signals, data acquisition, filtering

Introduction

DC voltages, DC currents and resistances are often measured with digital multi-meters (DMMs). For low-level signals, more sensitive instruments must be used. Low-level measurements are those close to theoretical limits and outside the range of most DMMs. An important aspect of making good low-level measurements is a proper understanding of instrument specifications (noise, speed and resolution). Making measurements close to theoretical limits, all considerations are very important. Virtual instrumentation can acquire and process any level signals with good speed and efficiency [1].

Because of many advantages of digital signal processing, analog signals are converted to digital form before they are processed with a computer. User must convert an analog signal into its digital representation using an analog-to-digital (A/D) converter. The acquired data does not always immediately convey useful information because it can have as result a sum of the useful signal and noise. One must remove noise disturbances, correct for corrupted data from faulty equipment or compensate for environmental effects. Essential purpose of signal processing is study, conception and realization of processing systems for signals. There are

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many application areas: radar, audio, telecommunications, images, vocal signal, teledetection [2].

Signals filtering represent a main operation in information's processing. It can be done analogically or numerically. The numerical filter is a processing system. Signals are represented through sequence of numbers at discrete time intervals. The processing is linear and the signal applied at the numerical filter's input has as result another signal with a different waveform. Specific frequencies are deleted or reduced. Stability (for limited input must result a limited in time output) and causality (output signal is not before the input one) are two important properties for numerical filters [3-4].

This paper presents low-level data acquisition and processing for a single record. LabVIEW soft is used in order to do these operations [5-6]. We will analyze data affected by noise. We want to separate the noise from the original acquired signals. For this reason, we choose to filter the signals.

1. LabVIEW soft

Using acquisition boards for analog or digital data from various transducers, signals can be analyzed or conditioning and measurements instruments can be created or simulated (virtual instrumentation).

The name LabVIEW is the abbreviation for: "Laboratory Virtual Instrument Engineering Workbench". It represents a graphical alternative soft to the conventional programming designed for instrumentation. It is equipped with all necessary tools for replace classical measurement systems. LabVIEW is an environment designed in order to create flexible and scalable test, to measure and to control many applications, with a minimal price. Using this software greatly reduces the development time for any data acquisition and control application.

LabVIEW uses a generally graphical language for programming called „G", containing libraries with specific functions. The programs are called virtual instruments and are made from two parts, distributed in two windows: "Front Panel" (necessary elements for interactive operations and the display of the results) and "Block Diagram" (source code, containing the corresponding instructions, constants, functions and pointers). Flowing data are determined in block diagram using links represented by lines between icons.

A/D converters are an integral part of National Instruments DAQ boards. One of the most important parameters of an analog input system is the rate at which the DAQ device samples an incoming signal. A fast sampling rate acquires more points in a given time and can form a better representation of the original signal. It is known that in order not to introduce errors (this effect is catted aliasing) the sampling rate should be at least twice the highest spectral component in the considered signal.

2. Acquisition and Processing Program

We built an application, called “DAP” (**D**ata **A**cquisition & **P**rocessing). We will read, store and compute analog and digital signals, particularly currents and voltages.

The 6024E board is a high-performance multifunctional analog, digital and timing I/O board for PCI, PXI, PCMCIA and CompactPCI bus computers. Supported functions include analog input, analog output, digital I/O and timing I/O. The 6024E features 16 channels of analog input (ACH), two channels of analog output, a 68-pin connector and 8 lines of digital I/O (DIO). This device uses the National Instruments DAQ – system-timing controller (STC) for time-related functions. It consists of 3 timing groups that control analog input, analog output and general-purpose counter/timer functions. These groups include a total of seven 24-bit and three 16-bit counters. The DAQ-STC makes possible such applications as buffered pulse generation and equivalent time sampling.

The device has a bipolar input range with programmable gain. One can program each channel with a unique gain of 0,5 (input range ±10V, precision 4,88mV), 1 (input range ±5V, precision 2,44mV), 10 (input range ±500mV, precision 244,14μV) or 100(input range ±50mV, precision 24,41μV) to maximize the 12-bit analog-to-digital converter resolution. With appropriate gain setting, one can use the full resolution of the A/D.

The acquisition board DAQ 6024E can operate with a maximum 200000-scans/second analog scan rate. For application “DAP” we need at least 1000-readings/second scan rate, meaning one millisecond data reading. Board admits an independently scanning for each channel. Processing operations can be on line or off line. We use off line processing because we want to see and compare effects from different filtering operations.

In order to acquire and to process data we acquired the transitory phenomena “Automat mode, working with load, damage stop” of Automatic Voltage Controller Equipment 45Vcc/21Acc (AC/DC converter used for industrial consumers supply with floating batteries, with a view to obtain sources of electrical energy using a synchronous generator), in Poiana Teiului power station, HG2 generator. This station is on Bistrița river and generator has as nominal values: apparent power S=7,5MVA, active power P=5,5MW, cosφ=0,9.

The acquisition board contains both analog and digital channels. Only two analog channels will be used here (low level voltage signals, millivolt order): **Urotor** voltage (regulator voltage supply an exciter with 9 rotary diodes; at its output is about 250Vdc, measured with a 1000: 1 ratio) and **Iex** current (regulator current is measured directly on a shunt 25V/75mV). These are scaled in the program, corresponding to their real values (tens of amperes for the field current, tens of volts for the rotor voltage).

The front panel of “DAP” program is presented in Fig. 1 and it contains next buttons:

- shift factors for OY axis;
- scaled factors for each channel;
- number of points for averaging (**m**);
- filter delay factor (**kf** - in milliseconds);
- acquired data graphic (at a scan rate of 1000 scans/sec);
- averaging data graphic;
- scaled data graphic;
- filtering data graphic.

We will present two sequences from the bloc diagram, one for averaging algorithm and one for filtering method (Figs. 2 and 3).

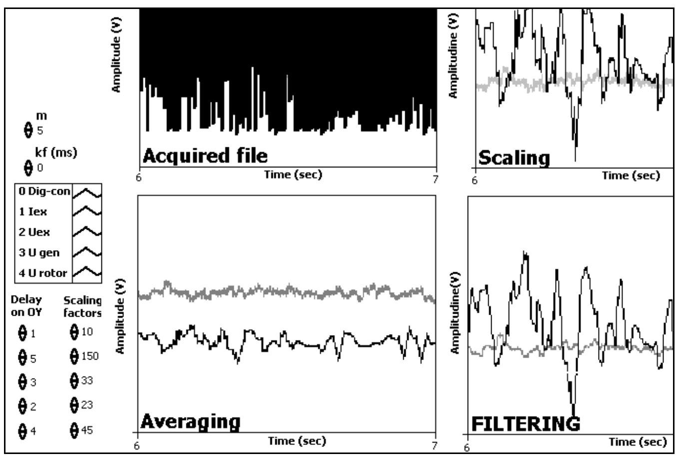

Fig.1 Front panel.

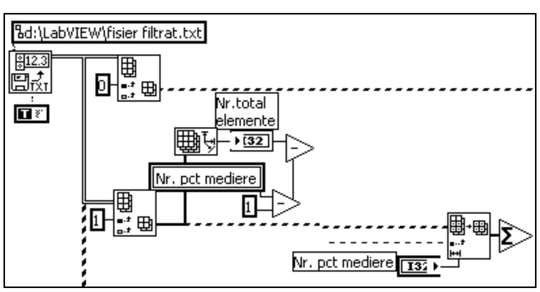

Fig.2 Bloc diagram – averaging sequence.

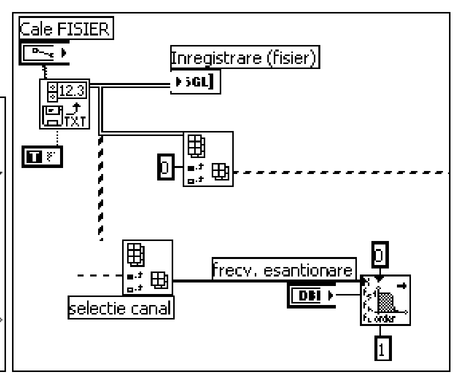

Fig.3 Bloc diagram – filtering sequence.

Initial acquired data file (Fig. 4: "1Iex" for current and "2Urotor" for voltage) presents an acquisition on 2 analog channels, for about 25 seconds. Data present the „ Automat mode, working with load, damage stop" acquisitioned phenomena.

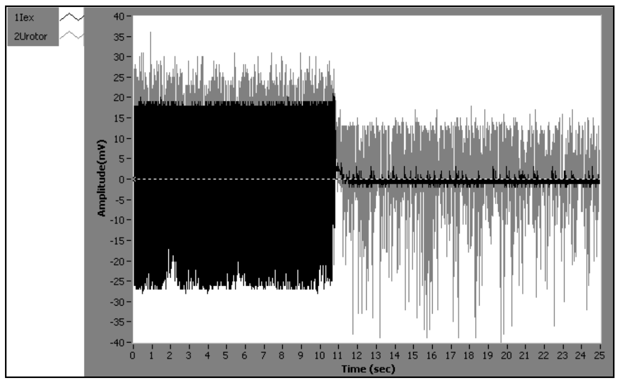

Fig.4 Initial acquired data file.

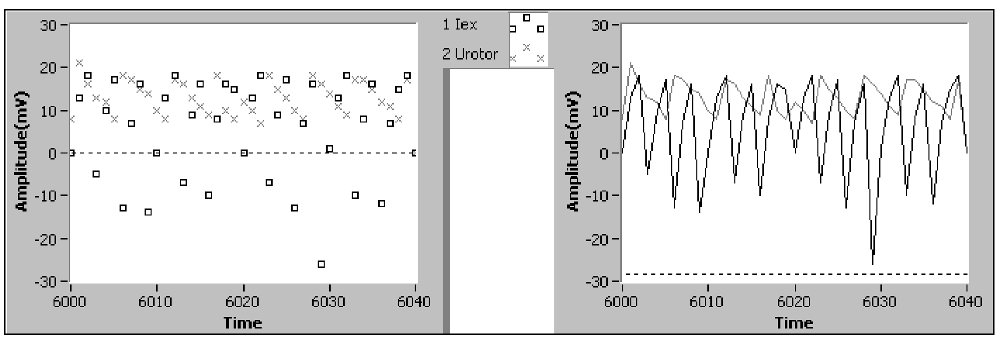

Fig.5 Initial data zoom (a-acquired points; b-linear interpolation for acquired points).

The first operation is averaging the acquired data. We set the "average" parameter (denoted with **m**). It does the arithmetical mean for the first (0÷m-1) values. It results the first value displayed on the graphic. We average the values (1÷m) and it results the second value displayed on the graphic, etc.

Instantaneous DC measurements of a noisy signal can vary randomly and significantly. One can measure a more accurate value by averaging out the noise that is superimposed on the desired DC level.

For a continuous signal, the averaged value between two moments is defined as the signal integration between the two moments, divided by the measurement time.

For a sampled signal, the average value is the sum of the voltage samples divided by the measurement time in samples, or the mean value of the measurement samples. One can improve the measurement accuracy by using a longer averaging time, equivalent to the integration time or measurement time.

The second operation is scaling for acquired averaged data, corresponding to „scaling factor" parameters. We must scale data in order to predict the real acquired values. This means a simple product with a certain real number that is the scale factor. This is different for each channel. Scaling operation is applied for all values from every channel.

The third operation is filtering for acquired scaled, averaged data. IIR (Infinite Impulse Response) filters are filters that may or may not have ripple in the pass-band and/or the stop-band. Digital IIR filter design derives from the classical analog designs (as Butterworth, Chebyshev, Elliptic and Bessel).

A smooth response at all frequencies and a monotonic decrease from the specified cut-off frequencies characterizes the frequency response of Butterworth filters. Butterworth filters are maximally flat, the ideal response of unity in the pass-band and zero in the stop-band. The half power frequency or the 3dB down frequency corresponds to the specified cut-off frequencies.

Following illustration (Fig. 6) shows the response of a low-pass Butterworth filter. The advantage of Butterworth filters is a smooth, monotonically decreasing frequency response.

After one sets the cut-off frequency, the steepness of the transition proportional to the filter's order it is set. Higher-order Butterworth filters approach the ideal low-pass filter response.

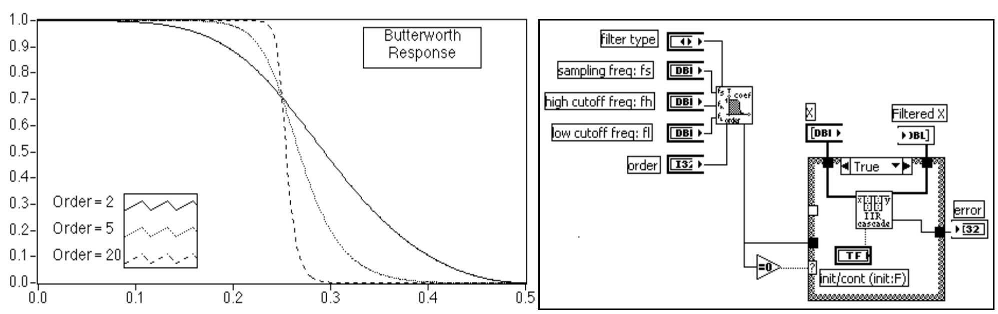

Fig.6 Butterworth filter – filter's response depending on its order;
Bloc diagram for Butterworth filter.

„Filter's time constant" parameter (denoted **kf**), set from the program, is the response of the delay of the filter for one step unit input. Graphic tangent intersects OX time axis in „time constant" parameter value.

Butterworth filters do not always provide a good approximation of the ideal filter response because of the slow roll off between the pass-band (the part of interest in the spectrum) and the stop-band (the unwanted part of the spectrum).

Chebyshev filters minimize peak error in the pass-band by accounting for the maximum absolute value of the difference between the ideal filter and the filter response you want (the maximum tolerable error in the pass-band).

The frequency response characteristics of Chebyshev filters have an equiripple magnitude response in the pass-band, monotonically decreasing magnitude response in the stop-band and a sharper roll off than Butterworth filters.

Fig. 7 shows the response of a low-pass Chebyshev filter. The equiripple response in the pass-band is constrained by the maximum tolerable ripple error and that the sharp roll off appear in the stop-band.

The advantage of Chebyshev filters over Butterworth filters is that first ones have a sharper transition between the pass-band and the stop-band with a lower-order filter. This produces smaller absolute errors and higher execution speeds.

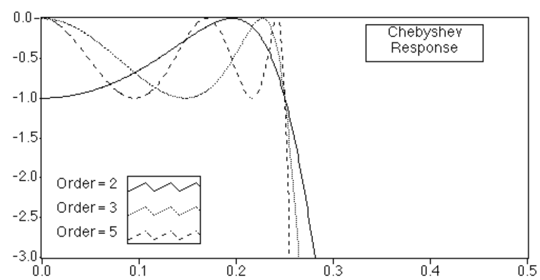

Fig.7 Chebyshev Filter – filter's response depending on its order

We choose for processing a one-second interval, so we can presume that both signals are continuous. For **m** and **kf** parameters we choose next values: 20 (a period length is 20ms), 100 and 1000 values (period multiples values).

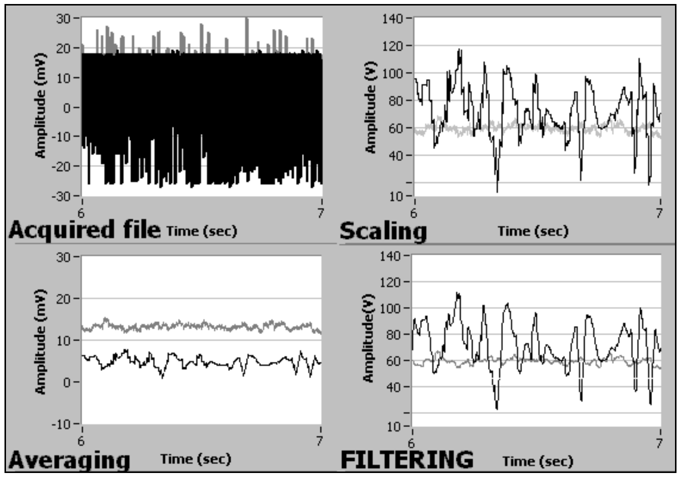

Fig.8 Data processing for **m = 20** and **kf = 20.**

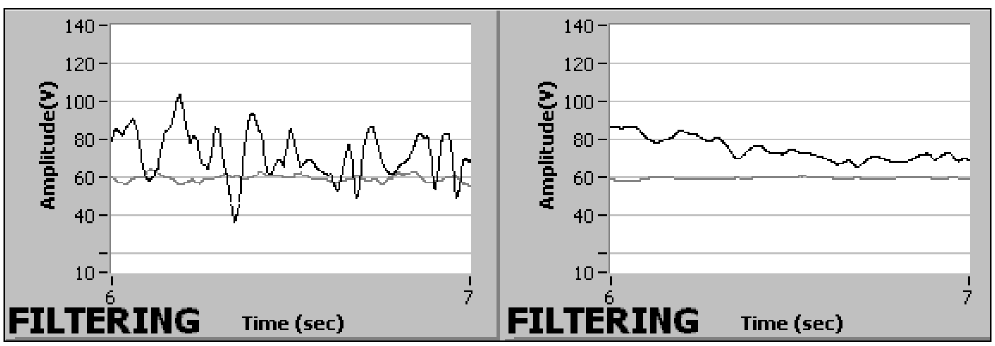

Fig.9 Final data for **m = 20** and **kf = 100**. Fig.10 Final data for **m = 20** and **kf = 1000.**

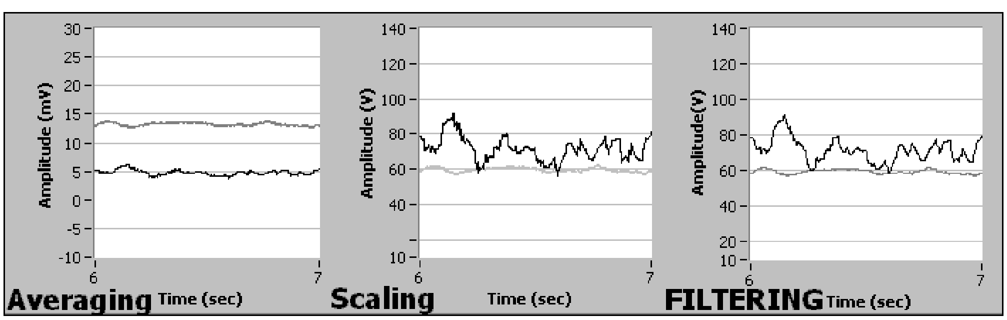

Fig.11 Data processing for **m=100** and **kf=20**.

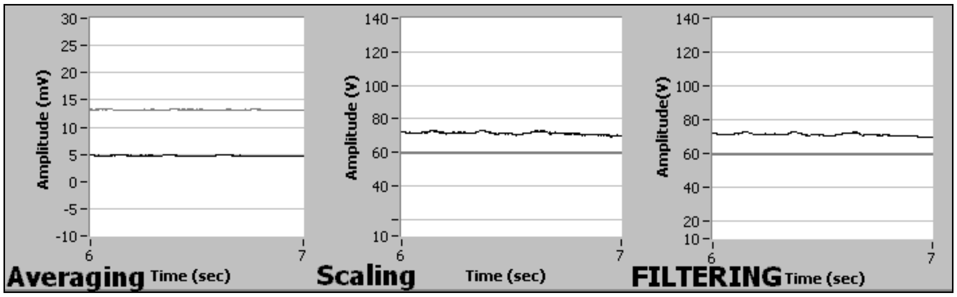

Fig.12 Data processing for **m =1000** and **kf=20**.

Filtering and averaging operations order can be changed in the program. Proceeding like this, we will obtain Fig. 13.

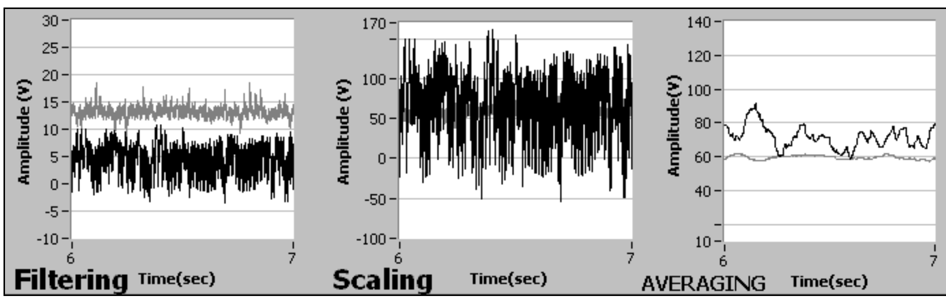

Fig. 13 Data processing for **kf = 20** and **m = 100**.

The comparison between low-pass Butterworth and Chebyshev filter, for the same parameters, follows. The same data file is analyzed. Figs. 15 and 16 present filtering data with low-pass Butterworth filter, time constant of 20 and 100 ms. Figs. 17 and 18 present filtering data with low-pass Chebyshev filter, time constant of 20 and 100 ms. One can observe that, for the same parameters, Butterworth filter is more efficient than Chebyshev filter in 'cutting' high frequencies for low-level data.

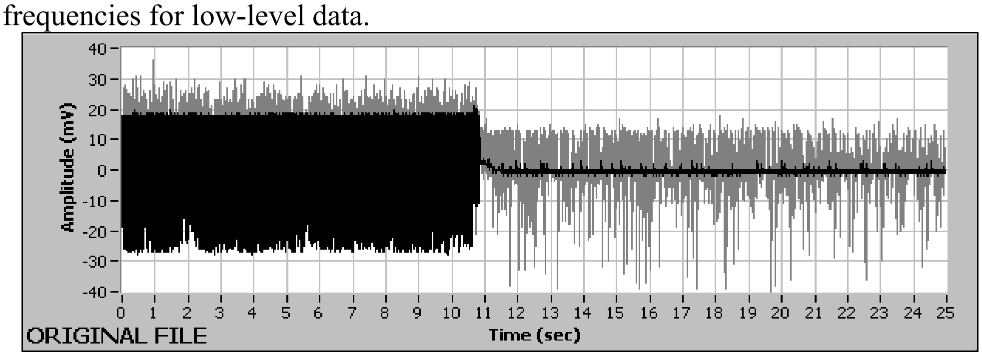

Fig.14 Acquired data file "Automat mode, working with load, damage stop"

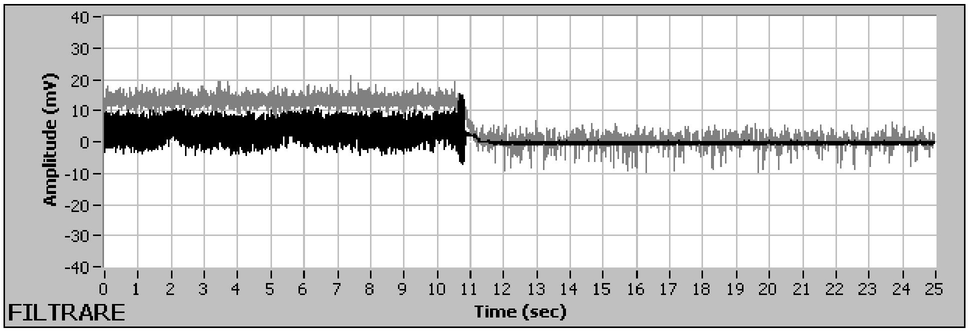

Fig.15 Low-pass Butterworth filter; **kf** = 20ms.

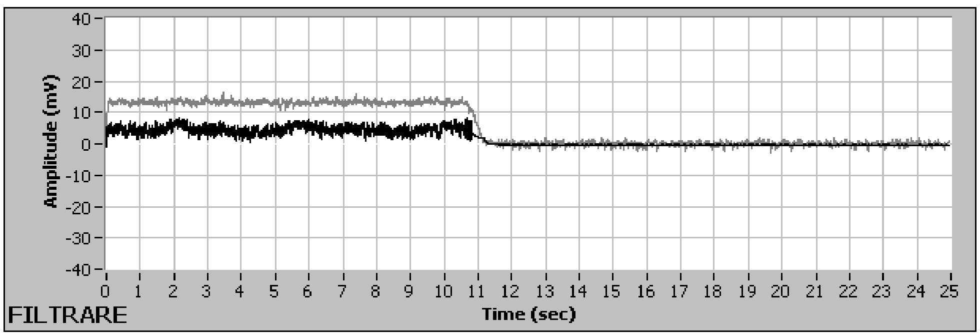

Fig.16 Low-pass Butterworth filter; **kf** = 100ms.

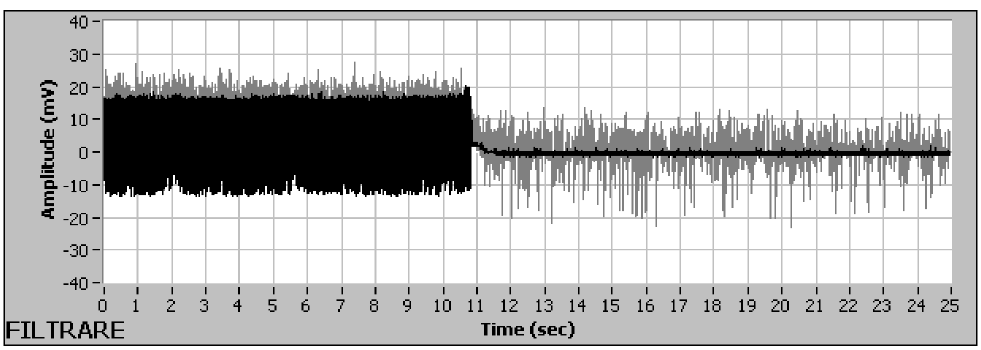

Fig.17 Low-pass Chebyshev filter; **kf** = 20ms.

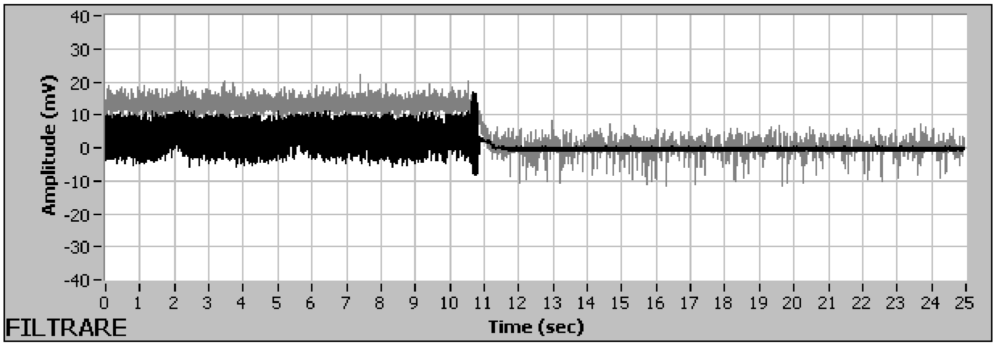

Fig.18 Low-pass Chebyshev filter; **kf** = 100ms.

Conclusions

- For **m** constant, increasing **kf** gives a better linearity for final representation. For **kf** constant, increasing **m** gives also a better linearity for final representation.
- For **m** higher (1000, 2000), filtering operation effects are not significant (**kf** can be chosen anywhere inside 0÷2000 interval). For **kf** higher (1000, 2000), averaging operation effects are visible.
- Operations’ order counts and it is recommended the following succession: first averaging, than filtering.
- We approximate the noise with the subtraction between acquired waveform and the averaging one is about the same with increasing **m**, without significant changing.
- The waveform resulted from the subtraction between averaged and scaled waveform and the filtering one is changing with increasing **kf**. If we keep **m** constant the variation domain will increase, but it can become spiky if we increase **m**.
- Histogram analysis gives as conclusion: the noise obtained from the subtraction between averaging and filtering operations is gaussian. Most of classical statistical theories suppose that variables have gaussian distributions so the noise has to be gaussian too.
- Some of the factors affecting the choice of a suitable filter are: operator requires linear phase, it can tolerate or not ripples and it requires or not a narrow transition band. In practice, one may need to experiment with several different options before finding the best one. Here, low-pass Butterworth filter seems to have good performances, if we compare it with other types of filters for the same **kf**. So, it is often used in this kind of applications.
- After one chooses the type of filter, he must specify the design parameters. The first filter design parameter to consider is the sampling rate. The

maximum frequency component of the signal of interest usually determines the sampling rate. A common rule of thumb is to choose a sampling rate that is 10 times the highest frequency component of the interest signal. In practice, a particular sampling rate is chosen and adjusted only if there are problems.

We can affirm that this analyzing program, “DAP”, has utility in numerical data acquisition, computation and processing. Acquired data can be obtained from any type of electrical equipments and than can be processed.

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