SAMPLE AUTO-RANGING DATA ACQUISITION USING NATIONAL INSTRUMENTS SUPPORT

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Abstract – *In the process of data acquisition the analogic gain is usually determined by the peak value of the input signal. For all samples the quantization error is theoretically less than 1/2 LSB. However for samples having small value, the relative error due to the quantization is high, and it goes higher with the decreasing of the sample value. The relative measuring error can be reduced by dynamically setting the proper gain. This is usually achieved with a supplementary hardware arrangement. The solution proposed in this paper consists in a numeric algorithm, developed in LabView, which replaces the hardware. The idea is to measure the same signal with many channels, set up with different gains, and to rebuild the signals, choosing the samples measured with the maximum accuracy. Sample translation must be performed due to the inter-channel delay.*

Keywords – data acquisition, sample, quantization error.

1. SAMPLE AUTO-RANGING DATA ACQUISITION

Computer based data acquisition systems are widely used in laboratory and industry measurements and automation. Configured and driven by G programming languages (LabView, HPVEE, TestPoint) they fast and flexible solutions for a large variety of applications.

The classical structure of a Data acquisition system (DAQ) consists of a multiplexer, a programmable gain amplifier (PGA), an sample and hold circuit (SH) followed by the analog to digital converter (ADC), as it is shown in

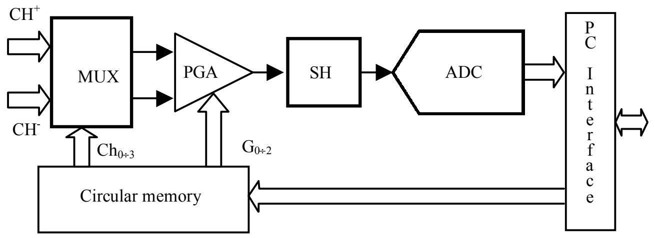

Fig. 1 – Classic structure of data acquisition system

figure 1. Most actual configurations allow to independently set-up the gain for every channel.

When measuring time variable signals, because the gain is fixed for every channel, signal's samples will be measured on the same scale. It means that small samples will be measured with greater relative error. Of course, these errors are cumulated when signal analysis is performed.

The authors proposed in [1], a different architecture for acquisition, which dynamically changes the gain, in order to measure every sample on the proper scale and to minimize the error. The auto-ranging is performed using a one, two or three bit flash ADC to determine the appropriate gain for measuring the current sample. The sample auto-ranging system [2] is presented in figure 2.

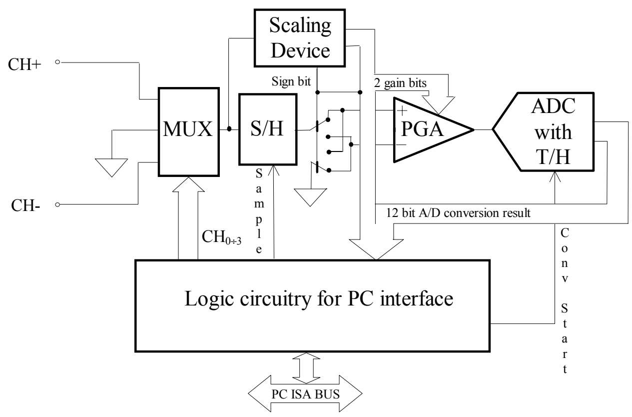

Fig.2 Sample auto-ranging architecture

Similar architectures are also presented in the literature [4] and [5]. If the gains are in the series 1, 2, 4, 8,… the system is entitled floating-point data acquisition. The name it is taken from the division operation which gives the result with float-point arithmetic:

$$U(i) = U_{ADC}(i) / G(i) \qquad (1)$$

The authors analyzed the advantages and the error improvement when various quantities are computed: mean value, RMS value, electrical power and FFT results ([1], [2], and [3]). It resulted an important improvement, without increasing the ADC resolution, no matter the desired quantity. The most advantageous situation occurs when the

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input signal has great shape factor, or for power measurements when cosφ is small.

2. SYSTEM DESCRIPTION

Starting from the principle presented in paragraph 1, the authors implemented the auto-ranging function by software. The hardware support is a DAQCard-AI-16E-4 with 12 bits of resolution and 16 single-ended channels, produced by National Instruments.

The auto-ranging function is the main idea of this paper. Because the board takes much time to be configured (using AI-Config), the gain can not be changed on line from one sample to the other. This problem was solved by using four channels, each one being setup with different gains as follows: Ch_0 – G=1, Ch_1 – G=2, Ch_2 – G=5, Ch_3 – G=10. Because of the board architecture (with the multiplexer located at the front end), samples from different channels read in one scan are delayed with one inter-channel delay. Thus the delay between first channel and channel k is:

$$\Delta t_k = t_{ID} \cdot k \qquad (2)$$

where t_{ID} is the inter-channel delay. National Instruments said that t_{ID} is 20µs for sample frequency under 10Khz, and 10us for higher frequencies. In fact, this value is 14us. To determine it, we measured the same signal on 4 channels with the same gain. Results on channels 1, 2, 3 have been translated using formula:

$$u'_j(t_k) = \frac{[u_j(t_k) - u_j(t_{k-1})]\cdot(T_s - j\cdot t_{ID})}{T_s} + u_j(t_{k-1})$$

(3)

where $j=0, 1, 2, 3$ is the channel number

$u_j(t_k)$ is the sample acquired at $t_k + j\cdot t_{ID}$

$u'_j(t_k)$ is the translated sample to t_k moment

T_s is the sampling period

t_{ID} is the inter-channel delay

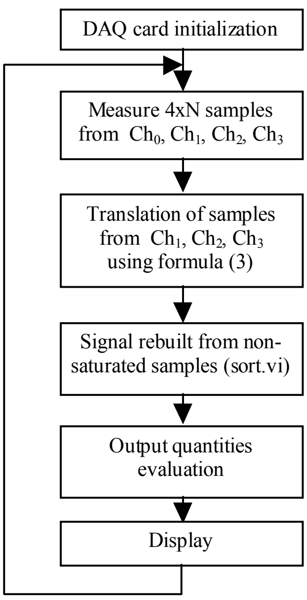

Fig. 3 The software logic diagram

In (3), t_{ID} was adjusted till the four acquired signals were overlaid. It occurs for 14us, value that is constant for every channel.

The software is built in LabView 5, and it follows the logic diagram shown in figure 3.

The signal reconstruction must replace samples smaller than 2.5V from Ch_0 with samples from channel Ch_1, samples smaller than 1V from Ch_1 with samples from Ch_2, and so on.. This operation is performed with a VI named *sort.vi*, its

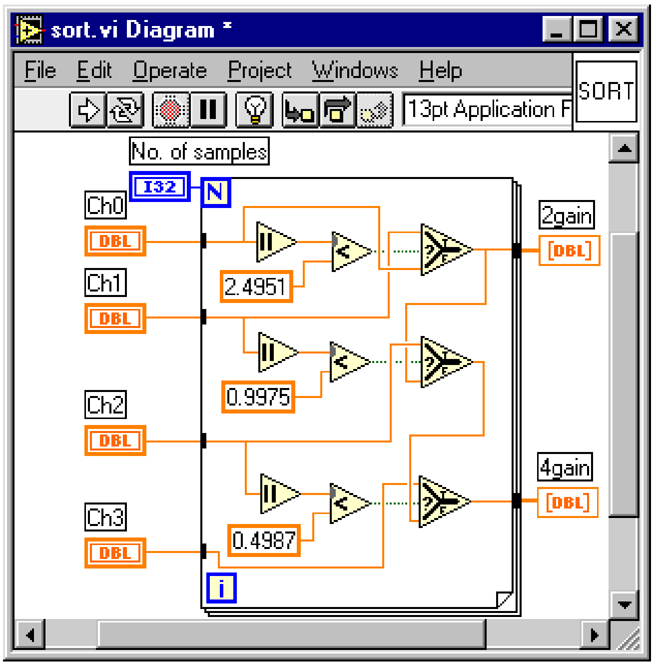

Fig.4. Logic diagram for *sort.vi*

diagram being presented in figure 4.

There are three edges where data from other channel has to be read: 2.5, 1.0 and 0.5 volts. In program, these values are 2.4951, 0.9975 and 0.4987, with 2 LSB smaller than the full-scale range. It is important that the bound to be compared with samples taken from the channel with the smaller gain to decide when "to change the gain". The explanation is that near to upper bound, the amplifier is not linear. In fact when the input signal (multiplied with the gain) is much greater than the upper bound, the PGA output goes quickly in saturation, but when the value is near the upper bound, PGA output goes slowly into saturation. Also, the first sample under the upper bound for each channel is not good, because the translated value is computed with one good sample and one, which represents the PGA saturation voltage.

The rest of the instrument makes the acquisition and computes the RMS and the mean values for the input signal. In order to analyze the influence of the number of gains, two and four channels are considered. In figure 4, *2gains* and *4gains* are the corresponding outputs. Measured data are saved in a file, in ASCII format, on the hard-disk in order to be processed in Excel.

Figure 5 illustrates the complete schematic diagram of the program.

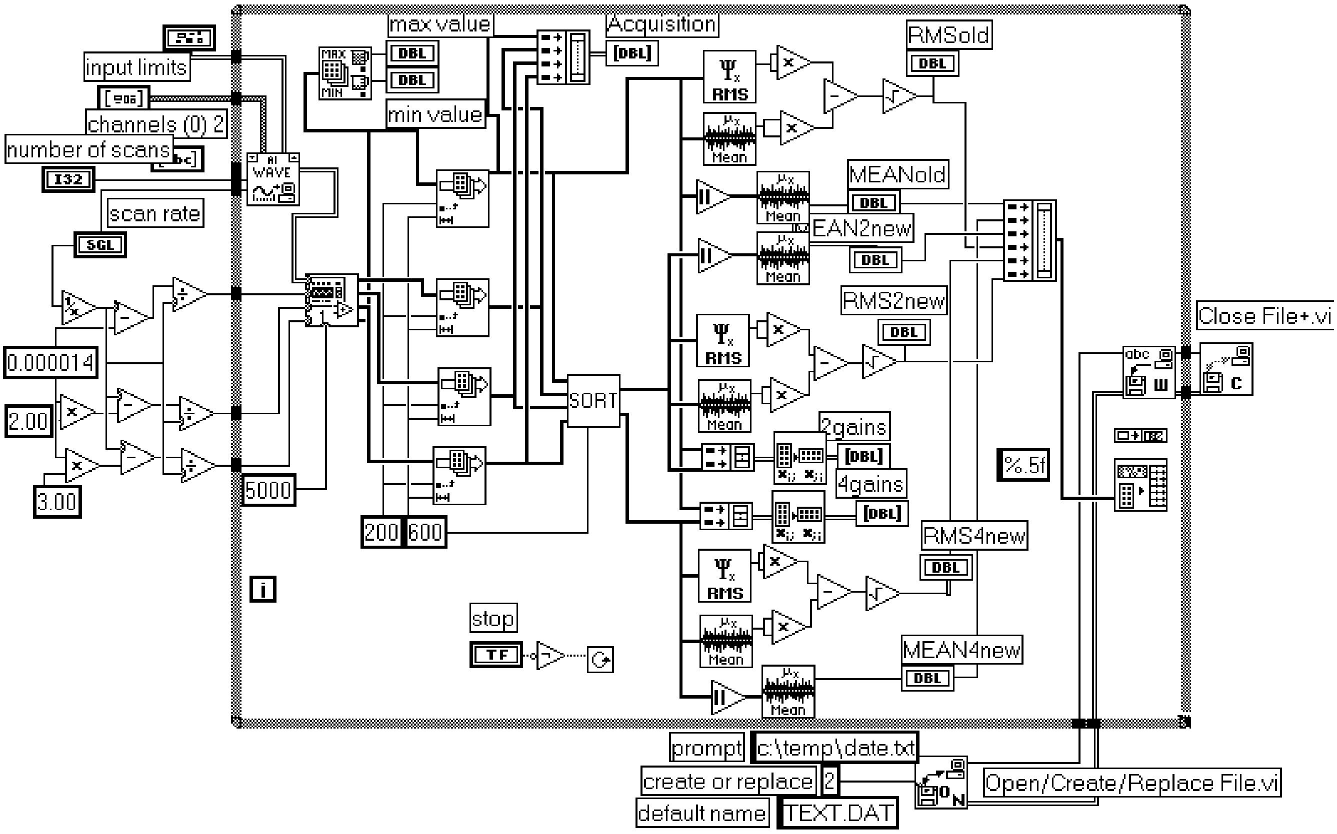

Fig.5 – Diagram of the sample auto-ranging data acquisition virtual instrument

3. RESULTS

Several experiments have been done in order to acquire the RMS and mean values using both the classic (single gain) and proposed (2 and 4 gains) algorithms. The measured values were saved and then passed to Excel in order to be processed. To obtain a greater accuracy, the computed RMS and mean values have been mediated over 25 periods of acquired signal. The results obtained are presented in Fig.6

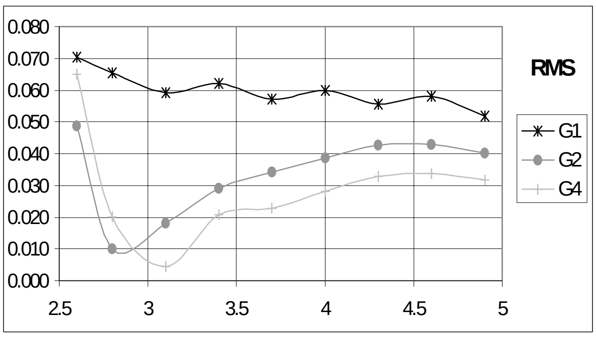

Fig. 6 – Relative error in evaluating RMS with single gain (G1) two gains (G2) and four gains (G4) algorithms

The curves are representing the relative error of the measured RMS values using as reference the indication of a Keithley 2000 multimeter (6 and 1/2 digits). As expected there are significant differences between the reference indications and the computed values, differences due mainly to the quantization error occurred in the AD conversion process. Using a sampling auto-ranging acquisition system, the absolute errors are always smaller. A major increasing in precision can be observed when the amplitude of the acquired signal is about 3 V, region in which it becomes three or even four times smaller than the error made with the single gain acquisition system. When the amplitude of the signal is approaching 5 V (maximum value admitted by the ADC) the differences between the algorithms are decreasing. This behavior is in good concordance with the theoretical results.

Comparing the RMS value evaluated by an acquisition system with two gains (G2) and a four gains (G4), a slightly

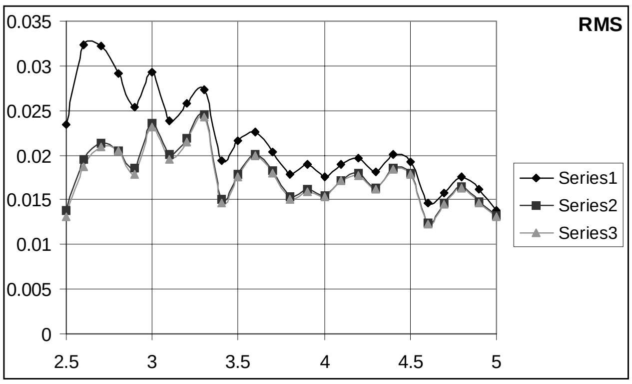

Fig. 7 – Theoretical relative error for RMS with single gain (e) two gains (enew2) and four gains (enew4) algorithms

different behavior than the theoretical one, shown in figure 7, may be observed. For amplitudes higher than 3V, the "four gain" algorithm gives, as expected, better results.

Figure 7 shows an insignificant improvement from 2 to four gains, because only quantization errors have been considered. In practice, it remains still significant due to other errors specific for a data acquisition system.

Similar analysis has been performed for mean value. Figures 8 and 9 are presenting theoretical and measured errors for this quantity.

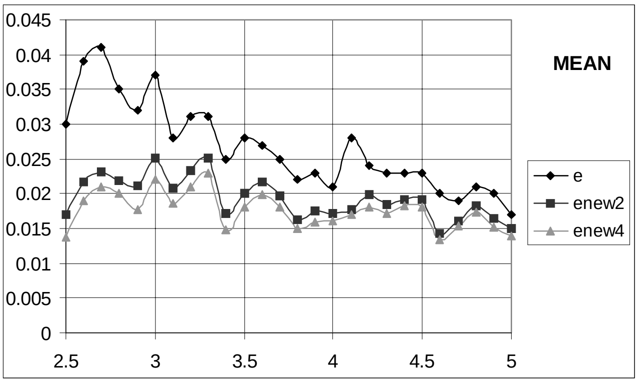

Fig. 8 Theoretical relative error for MEAN value with single gain (e) two gains (enew2) and four gains (enew4) algorithms

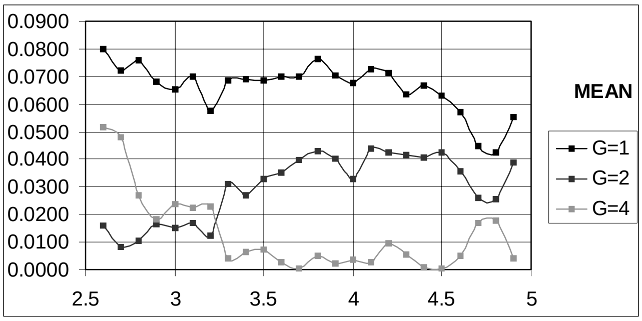

Fig. 9 Relative error for MEAN value with single gain (e) two gains (enew2) and four gains (enew4) algorithms

Measurement confirms theory. For 2 and 4 gains, the errors decrease when input amplitude approaching to 2.5V (the first bound). The improvement is much important than theory estimates.

Also, improvements for FFT have been studied. Because a reference measurement instrument was missing, only results for sinus wave are provided in the figure 10.

As it can be observed, the errors are almost the same, as for the RMS value.

Different from theoretical estimations, where only quantization errors has been considered, the measurement reveals an increase of the error near to the bound with the second scale (1÷2.5 V), no matter which is the quantity measured.

4. CONCLUSIONS

Using the facilities offered by the National Instruments data acquisition boards (different gains on different

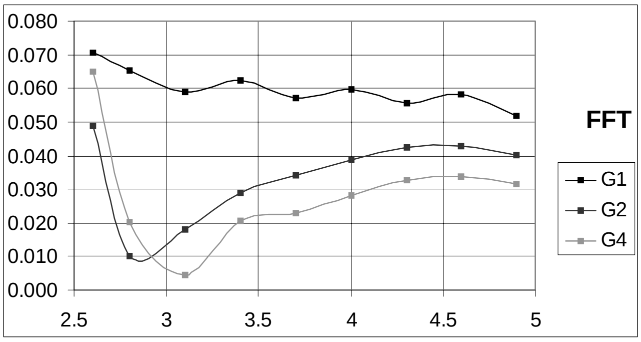

Fig. 10 Relative error for FFT

channels), authors used multiple channel acquisition of the same signal, with different gains, in order to obtain an acquisition with sample auto-ranging. The proposed solution uses two or four channel instead of one and software processing of the acquired signals. The system was tested with a 50 Hz sine wave, for RMS, Mean and FFT.

Result shows a significant improvement of the measuring error, starting from 2 times near full scale, and going up to 12 or 15 times around 3V. Different from simulation, where only quantization error has been considered, experiment shows that this method minimizes several other errors influences (non-linearity, offset, settling-error).

A cost-effective alternative is to use only 2 channels (two gains), instead of four, in order to increase the number of input channels and the maximum sampling frequency. The improvement remains significant, from 1.2 to 7 times.

The sinus wave is not the case where improvement is maximum. For different shapes (cardiac wave, waves with high amplitude and small mean value, etc), or when measuring electrical power or loses in coils, the improvement is more significant. In the previous papers, this aspect has been analyzed, starting with triangular wave, where the improvement is not so consistent, and going up to waves with the crest factor equals to 3, where the minimum improvement is between 4 and 5.

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