12th IMEKO TC4 International Symposium
Electrical Measurements and Instrumentation
September 25−27, 2002, Zagreb, Croatia

TESTING TOTAL HARMONIC DISTORTION FOR SAMPLE AUTO-RANGING SYSTEMS

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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 [1]. This is usually achieved with a supplementary hardware arrangement. The solution was proposed in [1] and consists in a numeric algorithm, developed in LabView, which replaces the hardware. The same signal is measured on many channels, set up with different gains, and the input signal is rebuild, choosing the samples measured with the maximum accuracy from all the channels. Sample translation must be performed due to the inter-channel delay. The main ideea in this paper is to test if the method does not affect the acquisition results because of different gains and delays of the amplifier. Other errors are introduced by the coarsely approximation when samples from different channels are translated to the reference. Following this, total harmonic distorsion and total harmonic distorsion & noise are evaluated for the system presented in [1], according to the IEEE standard 1057-1994 for digitizing waveform recorders. Comparison between simulation and experiment will be also presented in the extended paper.

Keywords: THD, samples, acquisition, virtual instrumentation.

1. INTRODUCTION

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 are fast and flexible solutions for a large variety of applications.

The authors proposed in [2], a different architecture for acquisition, which dynamically changes the gain, in order to measure every sample on the proper scale and to minimize the quantization error. The auto-ranging is performed before the acquisition using a flash ADC to determine the appropriate gain for measuring the current sample, during the previous conversion time. The sample auto-ranging system [2] is presented in fig. 1.

In [1], a simpler way to achive a sample auto-ranging acquisition, was proposed. The main idea is to measure the same signal on many channels, according to the number of gains are needed. For example if four gains are requested the system will use four input 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.

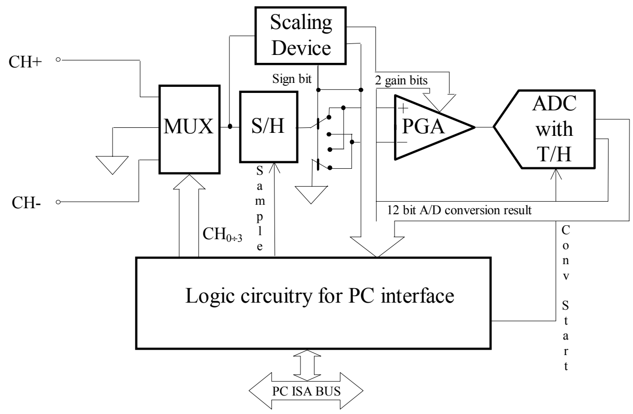

Fig. 1. Sample auto-ranging architecture

If the board is with sequential sampling between the samples with the same index from consecutive channels will appear a time delay. Between the first channel and channel *k*, the delay is:

$$\Delta t_k = t_{ID} \cdot k \qquad (1)$$

where t_{ID} is the inter-channel delay. Because every channel is delayed from the previous with the same t_{ID} due to the hardware arrangement, results on channels 1, 2, 3 must be translated using formula:

$$u'_j(t_k) = \frac{[u_j(t_k) - u_j(t_{k-1})] \cdot (T_s - j \cdot t_{ID})}{T_s} + u_j(t_{k-1}) \qquad (2)$$

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.

If the data acquisition board have simultaneous sampling channels, it will be no inter-channel delay, and no translation is needed. The errors have to be smaller in this case because the translation error does not count anymore.

Relative error improvement for different quantites (RMS and Mean) were also analysed in [1].

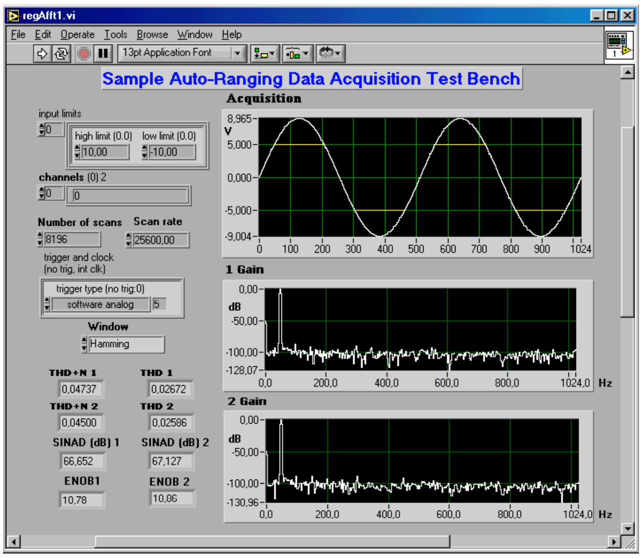

Fig. 2. THD VI's panel

2. THEORETICAL SUPPORT

For a pure sine wave input, the harmonic distortion is the output components at frequencies that are an integer multiple of the applied sine wave frequency. Their amplitudes are generally given as a decibel ratio with respect to the amplitude of the applied sine wave, and their frequencies are usually expressed as a multiple of the frequency of the applied sine wave. Total harmonic distortion is the root sum of squares of all harmonic distortion components including their aliases. Spurious components are persistent sine waves at frequencies other than those described as harmonic components. Total harmonic distortion is regularly expressed in percents [%] or in decibels [dB].

In order to obtain the THD, filtering and the computation of DFT must be performed for the record x_n:

$$X_m = \left| \sum_{i=0}^{M-1} x_i e^{-j2\pi i(m/M)} \right| \tag{3}$$

where $m = 0,1 \ldots M-1$. Any acquisition, due to imperfect synchronisation between the signal frequency and the sampling frequency will introduce in DFT's results components different from signal's harmonic one's. They are considered as noise introduced by the A/D conversion. Following this, there are two parameters which are relevant for the acquisition: THD and THD + Noise. Their computation can be performed using equations:

$$THD = \frac{\sqrt{H_2^2 + H_3^2 + \ldots + H_n^2}}{F} 100 \quad (\%) \tag{4}$$

and

$$THD + Noise = \frac{\sqrt{\sum PS}}{F} 100 \quad (\%) \tag{5}$$

where H_i are the signal's harmonics, F is the signal's fundamental and ΣPS is the signal's power spectrum.

The analysis of these parameters give us information about how accurate is the sample auto-ranging method, if the signal acquired is affected by the ample translations.

3. EXPERIMENTAL RESULTS

In order to achieve the purpose of the paper, a virtual instrument was built, as it is shown in fig. 2.

The signal is measured on four channel with different gains (channel 0 – gain 1, channel 1 – gain 2, channel 2 – gain 4 and channel 3 – gain 10). The samples are then processed in order to make the reconstruction of the original signal. Three data block were obtained: first is the signal read with 1 gain, the second with two gains and the last with 4 gains. A FFT is performed for each data block and finally THD and THD+Noise are computed and displayed on the panel.

Tests were performed using a Tektronix Waveform generator, by increasing the signal amplitude from near of the lower bound of the greatest scale (5V) to its higher bound (10V). The data acquisition system was a NI DAQCard AI-15E-4.

THD and THD+Noise versus signal's amplitude were obtained, like is shown in Fig. 3 and 4.

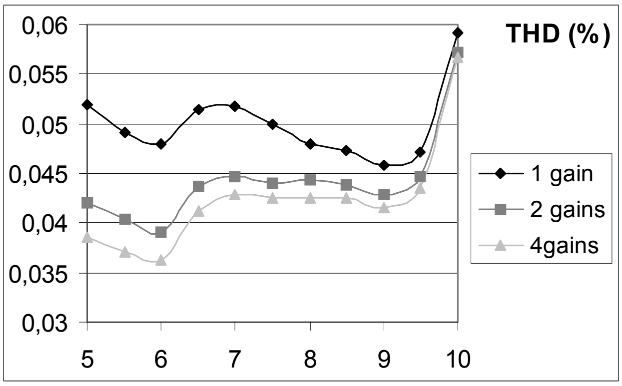

Fig. 3. THD versus signal amplitude for sequential sampling

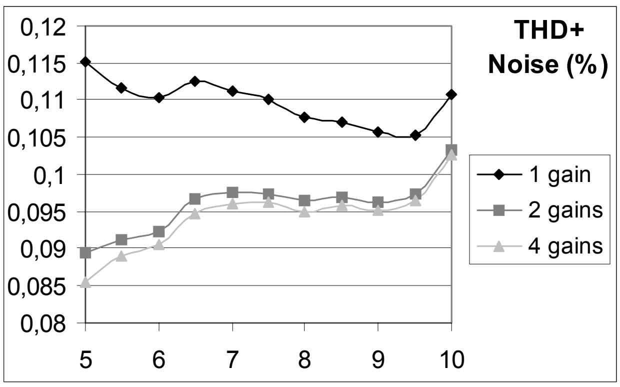

Fig. 4. THD+Noise versus signal amplitude for sequential sampling

The same experiment was performed for a simultaneous sampling data acquisition system (NI PCI-6111). Because the board has only two input channels, results for 1 gain and 2 gains are available in this case. Fig. 5 and 6 show the obtained experimental results.

As it can be seen and expected, a small improvement is achieved for THD. The curves have almost the same shape, the simultaneous sampling giving better results. The improvement is not so significant (from 0.05 to 0.04%), but from 3 a conclusion can be extracted: the improvement is more consistent when two gains are used, additional gains use being not justified.

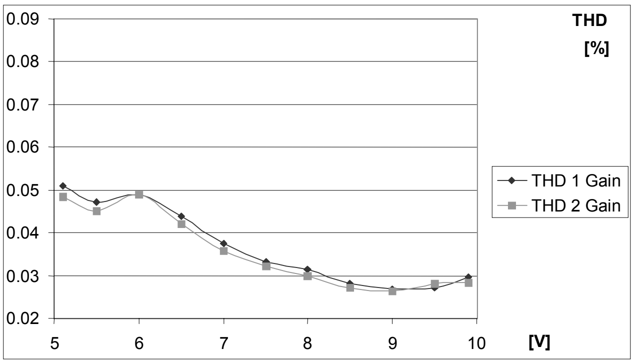

Fig. 5. THD versus signal amplitude for simultaneous sampling

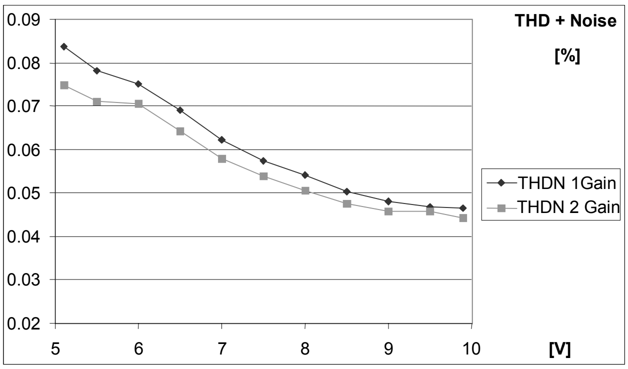

Fig. 6. THD+Noise versus signal amplitude for simultaneous sampling

For THD+Noise, the improvement is enough relevant, the use of two gains having as result the amelioration with 25% of the THD+Noise parameter, when the signal is going with the amplitude down to the lower bound of the highest scale.

In order to study if this is a resolution dependent, the same analysis was performed for an 8 bit digitizing system. A Tektronix oscilloscope TDS210 was used connected via GPIB to the computer. Also, the waveform generator was connected via GPIB, the VI giving directly the results for THD and THD+Noise: fig. 7 and 8. The TDS210 was set-up with 1V/div for the first channel and 0.5V for the second. The acquired waveforms are recorded via GPIB in the computer, and the original signal is reconstructed: samples smaller than 2.5V are taken from channel 2 and samples greater than 2.5V from the first channel. The THD analysis is performed for both, the signal acquired on channel 1 and the reconstructed signal.

As it can be observed from Fig. 7and 8, the sample auto-ranging method minimizes the noise of the acquired signal, without distorting the signal. For 2 gains, the THD+Noise curve is significantly under the curve for 1 gain. For THD the curves are closed, confirming that the auto-ranging method did not distort the acquired signal.

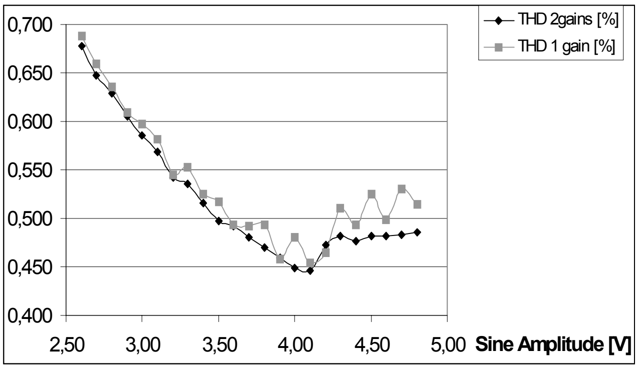

Fig. 7. THD versus signal amplitude for simultaneous sampling (8 bits)

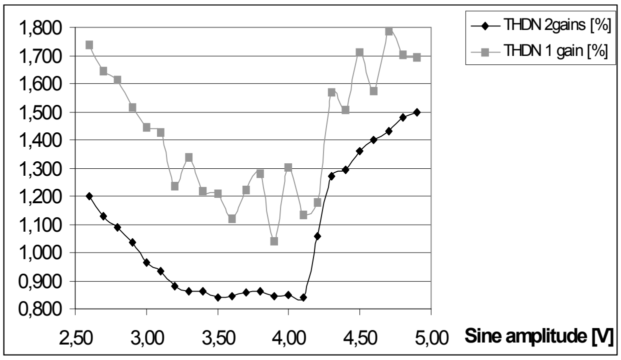

Fig. 8. THD+Noise versus signal amplitude for simultaneous sampling (8 bits)

4. CONCLUSIONS

The present paper consists in a study for THD and quantization noise of a finite resolution acquired signal using sample auto-ranging data acquisition. The analysis is based on data measured with 12 and 8 bits of resolution. Using more than 1 gain for measuring a sinusoidal signal provides an improvement in the acquired signal noise. The harmonic distortion is slowly improved, but the quantization noise is reduced down to 50%. Improvement results important when the signal's amplitude is near the lower bound of the most significant scale.

On the other hand, adding extra gains is not a business, because the improvement exponentially decreases, but the system resources will be overloaded. The cost-effective alternative is those with only 2 gains, or maximum 4.

Experimental results confirm simulations and theory, which said that an accuracy improvement has to be obtained. The proposed method for sample auto-ranging acquisition is better than the classic one. Significant improvement is achieved from 1 to 2 gains. Cost effective acquisition is offered by the proposed method.

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Acknowledgments

This work was financed and developed in the framework of the Leonardo da Vinci Pilot Project with the Contract number: RO/98/83550/PI.

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