Virtual Work Bench for Electronic Instrumentation Teaching

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Abstract—This article describes the design and implementation of a virtual laboratory, computer-generated for the theoretical and practical teaching of electronic instrumentation. The virtual laboratory is implemented in a windows framework for which the teacher or student requires no prior knowledge to manage the program. The Virtual Instrumentation Workbench for Instrumentation allows visualization of the time and frequency effects produced in analog to digital conversion and in digital signal processing as well as visualizing the effects of 16 different teaching topics. It incorporates ten different waveforms, eight windows algorithms, five digital filters, and five discrete/fast Fourier transform (DFT/FFT). The use of the virtual laboratory has allowed more than 500 students to understand the functioning of the new devices that are flooding the instrumentation market. Although it has been developed as an aid for electronic instrumentation teaching, it can also be used for the teaching of digital signal treatment.

I. INTRODUCTION

TRADITIONALLY, theoretical teaching of electronic instrumentation has been carried out in classrooms where the theoretical concepts cannot be verified by laboratory work. It is important in this practical laboratory work, to cover all the concepts taught in the classroom, but due to time limitations, it is difficult to create the conditions under which the comparative merits of the techniques employed can be assessed. Taking into account that in the last ten years, electronic instruments have increasingly come to incorporate digital devices, analog-to-digital converters, and the teaching of the techniques of sampling and digital signal processing have become fundamental to the study of electronic instrumentation.

Furthermore, the Valencia Polytechnic University's (U.P.V.) new study plans, must incorporate the reduction in theoretical teaching hours stipulated by the University Reform Law (L.R.U.) [1]. Under the new plan, topics related to digital signal processing are to be taught simultaneously. The teaching staff face a challenge, to ensure that students absorb the necessary knowledge and also understand the capabilities of the new instrumentation. The growing classroom use of computers linked to TV circuits, which allow students to see on-screen information, has brought a new teaching methodology to theoretical classes.

In this article a teaching tool is presented which allows the student to see immediately the advantages or disadvantages of signal digitalization techniques and their limitations. The Virtual Instrumentation Workbench for instrumentation (VIEW) has been used in the teaching of industrial and telecommunications engineers.

II. BACKGROUND

Personal computers have, since they became affordable, been seen as flexible and versatile teaching tools. Their efficiency has been studied in the last 20 years by authors such as Cañizares and Faur in [2]. Recently published applications cover the whole teaching process; theory, practical experience, exams, evaluation, as suggested by Bengu and Swart in [3], others have studied the use of PC's in certain contexts only, among them Mosterman *et al.* in [4], Oakley in [5], and Hollabaugh and Allen's virtual workbench in [6].

The VIEW was conceived originally as a tool for theoretical/practical teaching of analog-to-digital converters. A functional model was produced [7] with the intention of utilizing to best effect the teacher's time thus solving the above mentioned time problem. It has since been modified and improved and is now multidisciplinary and IBM-PC, Macintosh, and Workstation (UNIX) compatible.

III. USER INTERFACE

Fig. 1, shows the on-screen appearance of the VIEW. The left of the screen is for data entry and the right for graphical representation. The data entry boxes can be modified as often as necessary, the results of the operations selected appear with a double click on the virtual *DATA VALID* button situated in the control section. To finish a session, you must click on the virtual *STOP* button or press *ESC* key on the keyboard.

As can be seen in the block diagram, Fig. 2, data entry is divided into four sections; signal generation and sampling, device under test (D.U.T.), digital signal processing, and graphical representation and control. The capabilities are described below.

A. Signal Generation and Sampling

In this section, the basic conditions of the signal that will excite the D.U.T. are established. These conditions are divided into excitation and sampling. For excitation, the modifiable parameters are frequency (f_e), amplitude, phase, continuous component, and number of cycles (if *Mode Cycles* is selected). It is also possible to select from a total of nine different waveforms; sine, triangle, square, sawtooth, sinc, uniform white noise, Gaussian white noise, periodic noise, and ramp pattern. When a wave form which requires no specific data is chosen, the corresponding entry box automatically disappears from the scene.
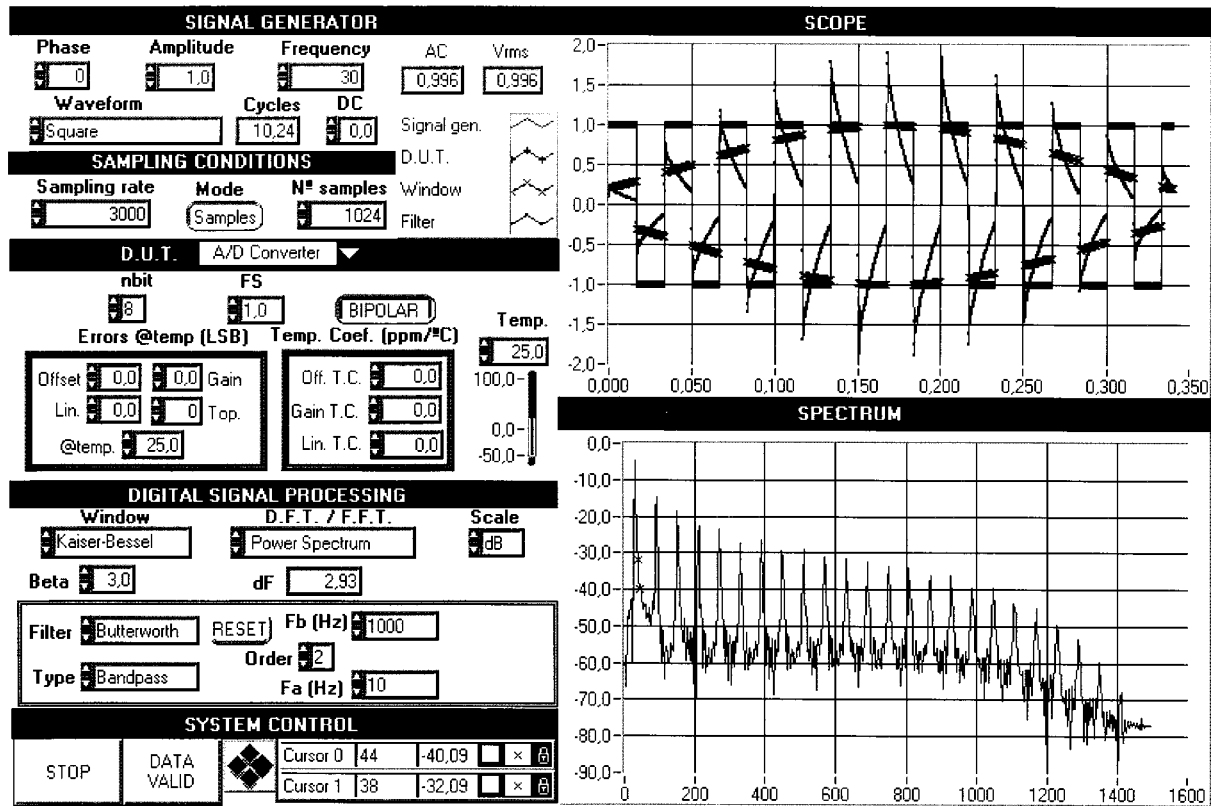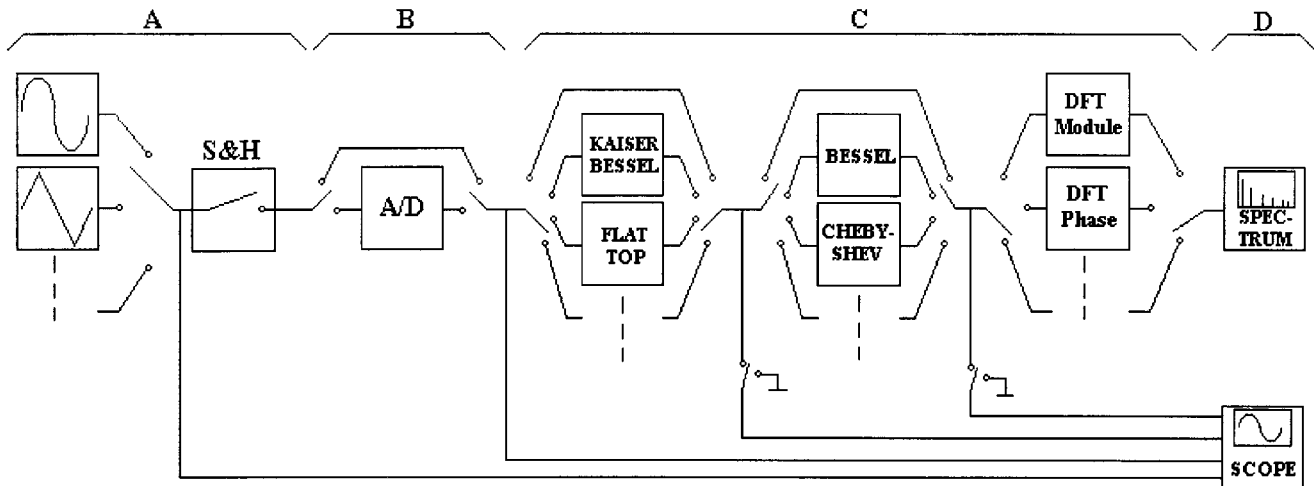
Fig. 1.   User Interface of VIEW.

Fig. 2.   Block diagram of VIEW.

With respect to sampling, it is possible to fix sampling speed (f_s) and to choose from two alternatives; Mode *Samples* and Mode *Cycles*. In first case the user fixes the number of samples, N, to be acquired and in the second case the user fixes the number of complete cycles, C, to be acquired. In both cases it is necessary to fulfill the condition

$$\frac{C}{f_e} = \frac{N}{f_s}$$

obviously, only three of the four variables can be fixed.

The maximum number of samples depends on the system's memory and is, theoretically, limited to $2^{31} - 1$. However, the digital processing of the signal and CPU time must be taken into account. For a number of samples of 256 k, the processor time of a machine with a Pentium® 166 MHz processor with 32 MB of RAM, is about 54 s, while with 2^{20} samples (1M), processing time is 16 min 42 s.

In this section the AC and RMS level of the signal is measured. The measured values of both indicators will be different when the signal contains a continuous component.

B. D.U.T.

In this section an analog-to-digital converter can be connected or bypassed. The variables that can be selected are the static parameters; number of bits (*nbit*), full scale (*FS*), unipolar or bipolar (*UNIPOLAR/BIPOLAR*), offset and gain error, linearity errors (*Lin.*) with different topologies (*Top.*) [7], all shown in LSB for a reference temperature (@*temp.*). It is therefore possible to select the temperature coefficient in ppm/°C, for each of above errors and the current working temperature (*Temp.*). All errors conform to the definitions established by Gordon in [8] and Tewksbury *et al.* in [9].

The VIEW has been designed in such a way that it can be easily extended with other D.U.T.'s. equally, there can be no D.U.T. and the excitation output be connected directly to the measuring device input, which allows assessment of the noise floor inherent in the measuring system.

C. Digital Signal Processing

The digital processing of the signal is carried out in three stages; Window, Filter, and Transform. The windows available in the VIEW are specifically designed for spectrum analysis applications, according to the observations of Harris in [10]. The window obeys the general expression $y[n] = x[n]w[n]$, where $x[n]$ are the input samples, $w[n]$ the windows coefficients and $y[n]$ the window output. There are eight windows algorithms available; Blackman, Blackman–Harris, Hamming, Hanning, Kaiser–Bessel, Cosine Tapered, Flat Top, and Triangle. Their mathematical expressions are shown in [10] and [11].

The next stage consists of filtering the signal at the exit of the window. All of the available filters are infinite impulse response (IIR) [11], which obey the direct general expression

$$y_n = \sum_{j=0}^{N_b-1} b_j x_{n-j} - \sum_{k=1}^{N_a-1} a_k y_{n-k}$$

where N_b are the direct coefficients (b_j) and N_a are the reverse coefficients (a_k). These coefficients depend on the type of filter chosen. The available filters are Bessel, Butterworth, Elliptic, Chebyshev, and Chebyshev inverse (II). Due to causes inherent in their functioning, the negative indexes of the above expression are considered null by defect. The consequence of this is a proportional transient at the order of the filter. If we wish that the filter considers the data of the last processing as the entry data of the present one it is necessary to select the functional mode *CONT*, otherwise select *RESET*. All the available filters feature all the filter modes; low-pass, high-pass, bandpass, and bandstop. It is therefore possible to select the order of the filter and its frequency characteristics.

The last stage of the processing is the DFT, which obeys the transformation [11]:

$$\mathbf{X}_k = \sum_{n=0}^{N-1} x[n]e^{-j\frac{2\pi nk}{N}} \qquad \text{for } k = 0, 1, 2, \ldots, N-1$$

if N is power of two, the VIEW process the FFT automatically as it optimizes memory and processing time. The limit of samples for the DFT is $2^{22} - 1$ ($4M - 1$), while for FFT is 2^{23} ($8M$). The available transform are DFT/FFT (module), DFT/FFT (phase), Power Spectrum (power of the respective harmonics, in watts, over 1Ω), Amplitude Spectrum (spectrum in V_{rms}) and Phase Spectrum (phase in deg).

D. Graphical Representation and Control

The graphical representation permits visualization of the signals of the VIEW, in the time dominion (oscilloscope) or in the frequency dominion (spectrum analyzer). The original waveform, the sample signal, the window output and the filter output are represented simultaneously on the oscilloscope. It is possible to annul the signals coming from the filter output and window, independently, for greater clarity of the graphic.

The spectral representation corresponds to the signal at the output of the digital filter, permitting switching between logarithmic or lineal vertical scale. Thus there are two cursors that can travel around the spectral representation independently with visualization of the frequency value and amplitude of their positions. The horizontal and vertical axes of graphics adjust automatically.

IV. TEACHING TOPICS

As has been mentioned above, with VIEW it is possible to visualize and verify concepts associated with the areas of electronic instrumentation and/or digital signal treatment. The following examples are a demonstration of the possibilities offered by the VIEW, contained in the two areas of knowledge mentioned:

DIGITAL SIGNAL TREATMENT Visualization of the frequency spectrum of ten different waveforms. Visualization of the sampled signal in time and frequency dominion, when the frequency of the signal is close to the Nyquist frequency. Visualization of the samples obtained by the window algorithms. Differences in temporal and frequency domain, between the different windows. Improvement of the spectrum following the window. Visualization of the temporal and frequency effects of the filtering. Visualization of frequency response of the digital filters selected in all the defined function modes. Limitation of the DFT/FFT.
ELECTRONIC INSTRUMENTATION Visualization of the transference function of an ideal A/D converter. Visualization of the transference function of A/D converters with offset, gain and/or linearity errors. Visualization of the lost codes in A/D converters. Visualization of the effects of temperature in real A/D converters. Temporal and spectral visualization of distortion by saturation in A/D converters. Improvement of the signal/noise ratio with the increase in the number of the A/D converter's bits. Measurement of the electrical power of a signal by the harmonics decomposition method.

V. CONCLUSION

The VIEW is a program that offers the following fundamental advantages; 1) It permits real-time viewing of the effects of sampling, A/D conversion and signal filtering. 2) It permits recording of the limitations of the DFT/FFT. 3) It is tolerant of possible data entry errors. 4) It has no instructions, therefore teachers and students require no prior knowledge of

its handling or of the digital signal treatment algorithms. 5) It can be compiled so as to function on IBM-PC (Windows 95), Macintosh or Workstation (UNIX) platforms. 6) The program is freeware (http://www.upv.es/view), allowing students to study and to experiment outside the university teaching laboratories. Thanks to the use of the VIEW, the problems explained in the introduction have been minimized.

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