

# Using PC Sound Blaster as a Digital Oscilloscope

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**Abstract** - Computer interfacing software has taken advantage of faster processing, development of user-friendly languages for integrating machines and computers. The purpose of this research work is to describe a powerful, widely applicable approach to interfacing that uses object-oriented programming and the concept of "virtual instruments" and to show how this approach can be used in the research laboratory and in practice. The proposed method can be used in engineering education and in web based distance learning.

**Key words** – Digital oscilloscopes, Virtual instruments, Education, Distance education.

## I. INTRODUCTION

In our days many functions, such as communication, data display, data output, human interface, memory, data processing in a measurement instruments system, and signal processing, are gradually taken in by improvement in the data-processing capability of a personal computer. Measurements systems can be customized comparatively free according to a user intention.

The development of powerful personal computers and workstations has transformed the way scientists work. Increased processing speed and available memory have led to the development of highly sophisticated programs that perform intricate calculations and handle large data sets.

Electronic signal processing can now often be replaced by digital computer processing. Windows and graphical user interfaces have made it possible for computers to perform multiple tasks simultaneously and have made it easier for scientists to analyze and display data as well as to write papers, manage references, and so on.

## II. DIGITAL OSCILLOSCOPES

Digital oscilloscopes and waveform digitizers sample signals using a fast analog-to-digital converter (ADC). At evenly spaced intervals, the ADC measures the voltage level and stores the digitized value in high-speed dedicated memory. The shorter the intervals, the faster the digitizing rate and the higher the signal frequency that can be recorded. The greater the resolution of the ADC, the better the sensitivity to small voltage changes. The more memory, the longer the recording time. What are the benefits of this digital technol-

ogy? Multiple signals associated with intermittent and infrequent events can be captured and analyzed instantly.

Complex problems can be quickly identified by viewing waveform data that precedes a failure condition (pre-trigger data). Captured waveforms can be expanded to reveal minute details such as fast glitches, overshoot on pulses, and noise. These captured waveforms can be analyzed in either the time or frequency domains.

Some oscilloscopes will:

- Monitor parameters such as amplitude fluctuations, timing jitter, risetime, etc., and display worst-case values.
- Provide histograms of parameter measurements to accurately identify important signal characteristics.
- Let you use the full screen as a signal viewing area.
- Allow signals to be saved or recalled from PC card devices such as portable hard drives, ATA Flash Cards, or IC memory cards.

## III. VIRTUAL DIGITAL OSCILLOSCOPES

There are a lot of types of digital oscilloscopes made especially for personal computers. All they utilize a PC interface card which has an analog-to-digital converter that manages the incoming analog waves and converts them into a digital sequence. Next, there is special software that manipulates the data and visualizes it on the screen. The most common types use the PCI bus and are quite cheaper compared with the traditional digitizers. The price of such an oscilloscope is about 650EUR while the price of one digital scope is in the range of 1500 to 2000 EUR and has no significant advantages compared with the PC based model. Contemporary PCI based oscilloscopes have 8, 12 and some of them even 16 bit ADC and have the sampling rate of 5GS/s which can guarantee very good measurement results.

One cheap solution to construct a PC based digital oscilloscope is to use as an analog-to-digital converter the PC sound blaster. From microphone or line-in input, a sound card receives a signal in its native format, a continuous analog signal of a sound wave that contains frequencies and volumes that are constantly changing. The sound card can handle more than one signal at a time, allowing us to record signals in stereo or to construct two channel oscilloscopes. It has two inputs one mono input for microphone with amplitude voltage range: 0 – 2mV and frequency range: 20Hz – 20kHz and one stereo input with amplitude voltage range: 0 – 500mV and frequency range: 20Hz – 20kHz. Here is one limitation of this kind of scope as it will work only with frequencies in the sound spectrum from 20Hz to 20kHz. The sound card uses the Sigma-Delta analog-to-digital converter. It is the same as the ADCs used for measurements purposes and has resolution of 16 or even 24 bits. Sigma-Delta Analog-Digital Converters (Σ-ADCs) have been known for nearly thirty years, but only recently has the technology existed to manufacture them as inexpensive monolithic integrated circuits. One of the most

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advantageous features of the sigma-delta architecture is the capability of noise shaping, a phenomenon by which much of the low-frequency noise is effectively pushed up to higher frequencies and out of the band of interest. As a result, the sigma-delta architecture has been very popular for designing low-bandwidth high-resolution ADCs for precision measurement. Also, since the input is sampled at a high “over-sampled” rate, unlike the other architectures described in this paper, the requirement for external anti-alias filtering is greatly relaxed.

Next the digital data is sent to the DSP of the sound card and after that goes to the CPU of the computer and can be manipulated by the application software. This software can be made easily with LabView (fig.1).

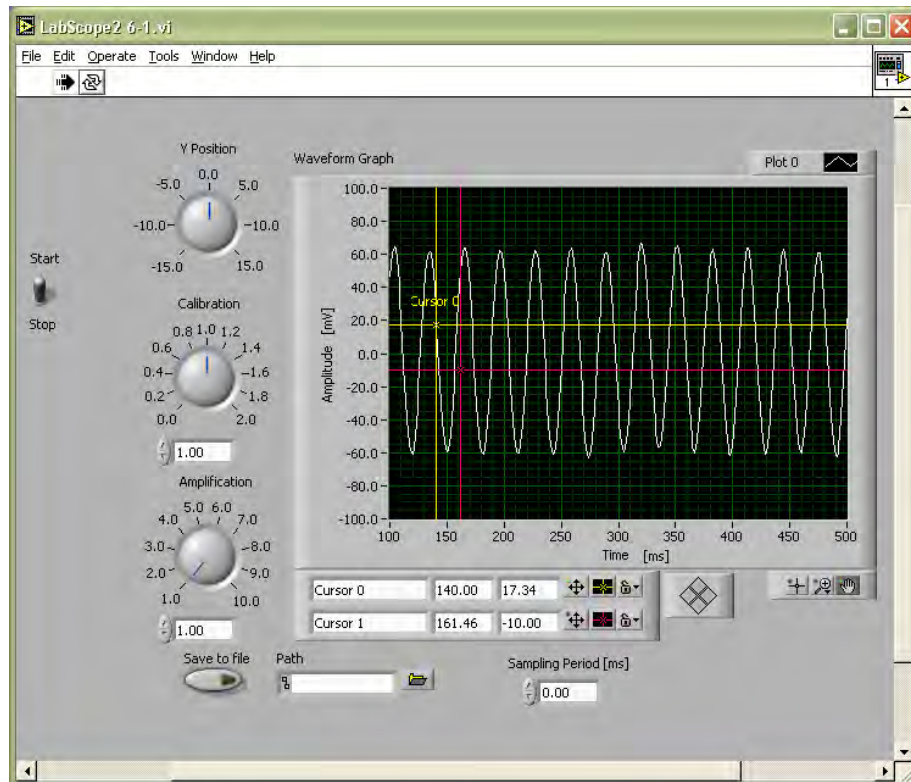


Fig. 1. Front panel

A block diagram of a simple program for this purpose is shown in fig. 2.

The initialization of the program starts with Sound Input Configuration module (fig. 2). Here it is configured which sound device will be used and the format of the incoming sound waves. “Device 0” means that the first sound card will be used. If there is a second sound card it will be addressed as “Device 1”. In this case it is used Sound Card 1. The format of the sound can be mono or stereo. Next is the bit rate it can be 8000, 11025, 22050 or 44100. We set it to be 11025. The last adjustment in this block is the bits per sample. They can be 8 or 16. In the LabScope we use 16 bits.

The next block of the program is Sound Input Start module. It prompts the sound input device to begin accumulating incoming data. If the device is running already, calling this block has no effect. Following the SI Start is a block which reads data from the sound input device and sends it for further processing. If data has arrived in the device buffer, it returns that data after buffering, otherwise it waits until data arrives. If for some reason, the unbuffered data is overwritten, no data is returned, and instead, an overwrite error is reported. After the sound is read the SI clear closes the sound input device associated with the task ID in and returns all the resources the device uses to the system.

From the SI Read block the signal is sent to the waveform graph. But before displaying it to the screen the signal passes through a calibration block to adjust it for proper visualization according to the type of the sound card used. This is necessary because different types of sound cards have different voltage levels of input and output signals. It is good practice before starting the first measurement to use a predefined cali-

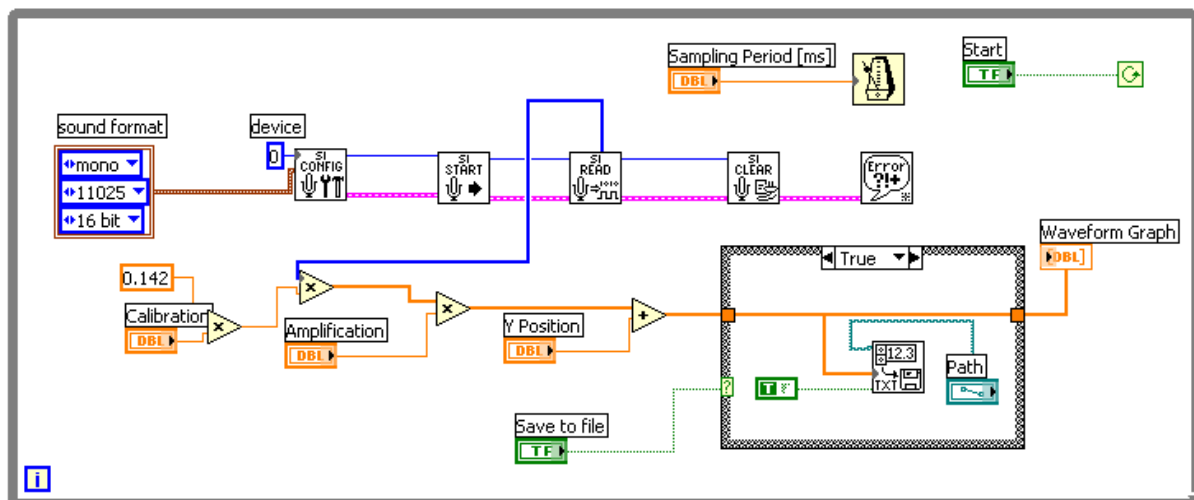


Fig. 2. Block diagram

brating signal to adjust the scope for the concrete sound card input and then we can be sure that the measured results are correct. Next there are some options for the signal: to be amplified from 1 up to 10 times depending on the position of the amplification knob 1 or/and to be added or subtracted 15 units in order to be better visualized. After this adjustment and calibration the data is passed to one cycle which checks the state of the “Save to file” button and if button is not pressed sends the data directly to the waveform graph. In case the button is pressed the data is sent to the waveform graph and to the file writing module. It converts a 2D or 1D array of single-precision numbers to a text string and writes the string to a new byte stream file or appends the string to an existing file, depending on the state of the “Append” input. If the “Append” input is true the new data is written at the end of the same file, otherwise each new group of data is saved in a separate file.

The functioning of the whole program is controlled by one While cycle which executes all of the above operations when the Start/Stop switch is in the true state. When the user turns off the main switch the While cycle receives False command and the operation of the LabScope program is terminated.

The limitation of the sound card is the input voltage range: 2mV for the mono input and 500mV for the stereo input. Range can be extended using the following very simple input buffer. It is used to protect the inputs of the PC sound card from the unwanted from the high voltage of the measured signal and thus it extends the voltage range to  $\pm 10V$ .

The principle of operation of the input buffer (fig. 3) is as follows:

By resistor of  $R1=1M\Omega$  we ensure high input impedance thus we will not disturb the functioning of the investigated circuit. The input impedance of  $1M\Omega$  is typical for most oscilloscopes. The capacitors  $C1$  and  $C2$  are used to reject the signals with frequencies lower than 20Hz and higher than 20kHz e.g. these signals that are outside the

range of the sound card. Their values are as follows:  $C1=10nF$  and  $C2=20pF$ .

The next part of the buffer is the input protection of 150V overvoltage (fig.3). The resistor  $R2$  limits the input current of the operational amplifier in the next stage of the circuit. That’s why it is necessary this resistor to be able to dissipate power of 0,5W. Its value of  $47k\Omega$  is most commonly used in the frequency range of 20Hz-20kHz.  $R2$  is connected in parallel with one capacitor of 100pF. This capacitor is called accelerating capacitor because it accelerates the high frequency components of the signals and thus straightens shape of the pulses.

The three diodes  $D1$ ,  $D2$  and  $D3$  are used for protection in case the input voltage of the operational amplifier is larger than 12V. If there are some peaks of the signal, which are higher than 12V the diode  $D1$  opens and sends the overvoltage signal back to +12V terminal. In case the signal is lower than -12V the diode  $D2$  opens and sends the signal to the ground.

Next we place one non-inverting amplifier (fig. 3) to made possible very low signal to be amplified 10 times. For amplifier we use the Operational amplifier TL082. It is connected with two resistors  $R4=27k\Omega$  and  $R5=3k\Omega$ . When the switch  $SW1$  is closed the amplification factor  $A$  becomes:

$$A = \frac{R4}{R5} + 1 = \frac{27}{3} + 1 = 10$$

When the switch  $SW1$  is open the circuit repeats the input signal without any amplification.

At the output of the amplifier we place one additional trimmer (fig. 3) to adjust more precisely the amplitude of the voltage that will be forwarded to the 2 mV sound card microphone input. It is realized with the one resistor of  $R6 = 300K$  and one trimmer of 1K for fine adjustment of the signal.

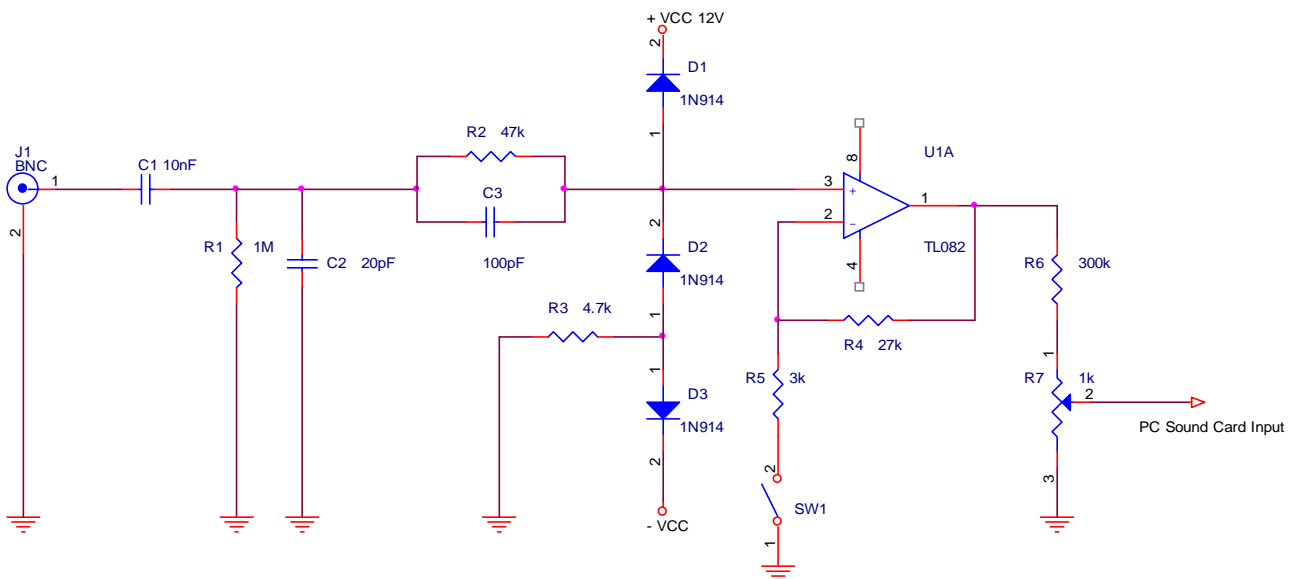


Fig. 3. Buffer circuit

#### IV. CONCLUSION

In the present project is created one virtual measuring system which can be used as a digital oscilloscope. It uses data acquisition techniques to send the results from the measurements to the personal computer. As a data acquisition board is used an ordinary PC sound card which makes the utilization of such kind of measuring instrument very cheap. That type of oscilloscope is suitable to be used for educational purposes and in research laboratories for low voltage measurements of AC signals.

Application software is worked out for Windows 95/98, NT and XP to handle the incoming data from the sound card microphone input, make it in the form suitable for visualization and display it on the PC screen.

It is designed an input buffer circuit which has two functions: first it is used to protect the input of the sound card from the incoming high voltage peak signals and its second function as an oscilloscope probe and has the ability to multiply 10 times the low voltage signals.

The realized digital oscilloscope has several advantages compared with the traditional analog scopes. On the first place is the possibility to store the data from the measured wave and to record it into a spreadsheet file and thus makes it possible the data to be further processed and analyzed by the standard programs for data analysis. There is no flickering of the waves on the screen. At every time the visualization can be paused in order to be taken some measurements and analysis.

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