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Experimental data frequency measurement with a PC

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Personal computers are ubiquitous in teaching laboratories and with only a small investment in time and equipment are capable of many types of data collection.[1] The following note describes a procedure for measuring analog or digital signal frequencies using only the parallel port of a standard IBM-PC computer and simple external circuits. It can be employed to measure signals of long or short duration and has been used to measure frequencies from 20 Hz to over 10 kHz, including the Doppler shift of a moving sound source as described below. The hardware and software components needed for the frequency measurement are few and reasonably straightforward to construct. When measuring an analog signal, simple hardware (Fig. 1) may be used to convert a signal's amplitude to a TTL-level binary value. For signals of short duration, automatic triggers such as photogate circuits are necessary to start and stop sampling of the signal. The binary signal is sampled via the parallel port at regular intervals determined by the computer system clock. When sampling is completed, the collected samples are transformed from the time to frequency domain using a fast Fourier[2] transform. The details of the hardware and software are described below as used by the authors for the Doppler shift measurement.

I. HARDWARE: CONVERSION AND TRIGGERING

Acting as a one bit analog to digital converter, the circuit (Fig. 1) amplifies and converts the amplitude of the microphone signal from analog to a binary, TTL-level value. The circuit output can connect directly to any of the parallel port status lines (pin 11–13, 15) for the computer to sample the digitized signal. Generally, samples must be taken at twice the signal's highest frequency with care that sampling occurs during both the high and low levels of the signal's highest frequency. Ideally, the signal's high-to-low value ratio will be 1:1 (equal) since the same signal with a high-to-low value ratio of 1:3 would require a higher sample rate (at least four times the signal's highest frequency). A potentiometer can be used to adjust the ratio (Fig. 1).

Triggering when sampling starts or stops is necessary when the signal is of short duration; for the Doppler shift, sampling should only occur when the sound source is within microphone range. Photogates are used to mark these range boundaries. The photogate output is amplified and converted to TTL-level and connected directly to the computer parallel port Acknowledge line (pin 10). The circuit output should normally be high, pulsing low when an object enters a photogate. The low output serves to trigger an interrupt to the computer at the moment sample taking should start and later, when it should stop.

II. SOFTWARE: SAMPLING AND TRANSFORMATION

In the measurement of Doppler sound shift, software does three functions: (1) starting and stopping sampling of the signal, (2) sampling at regular time intervals, (3) fast Fourier transform of samples from time to frequency domain to determine the frequency spectrum of the signal. The details of each are:

(1) Sampling starts and stops on a high-to-low transition of the photogate output (attached to the Acknowledge line) with the transition causing an interrupt on the computer. As the sound source enters the first photogate, sampling begins and a timer starts and continues until the

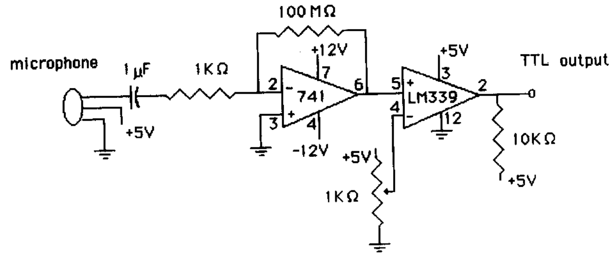

Fig. 1. Circuit of one bit analog to digital converter.

source enters the second photogate or all samples have been taken.

(2) The sample rate is controlled by using interrupts generated by the computer system clock to time reading of the parallel port. As the sample rate must be double the signal's maximum frequency, a 5-kHz signal must be sampled at 10 kHz or at 100-ms intervals. With limited time between clock interrupts to do all sampling related operations, the number of operations must be kept to a minimum, avoiding even the overhead of procedure calls. The critical operations done at each interrupt are to: (a) read the parallel status port and store the sampled bit, (b) increment an elapsed time counter to measure time between sampling start and stop, and (c) clear the interrupt controller to allow the next interrupt to be processed. The sample rate is mainly limited by the computer system execution rate and the number of instructions executed to service each interrupt; a 16-MHz IBM PS/2 is capable of sampling at over 20 kHz, fast enough to measure a 10-kHz signal. The Doppler shift example uses the elapsed sampling time from operation (b) to determine the velocity of the sound source, allowing verification of the measured Doppler shift.

(3) The collected data are transformed from the time to the frequency domain using a Fourier transform routine.[2] Since only a one bit value is read from the parallel port for each sample and immediately stored as read, after sampling is completed the value must be converted to a real number format for the Fourier transform. Fast Fourier transform routines generally require that the time domain samples be stored in array form with a power of 2 number of elements. When fewer samples are taken, the remainder of the array can be filled with zeros.

As a final note about this example, a single frequency piezoelectric sound generator commonly found in alarms was used as a reliable, if irritating, sound source. An object oriented Turbo Pascal or C + + version of the software is available through the authors by disk exchange or electronic mail to "rwisman@ucs.indiana.edu" or "forinas@iubacs.indiana.edu".

ACKNOWLEDGMENT

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[1]R. Wisman and K. Forinash, "Discount interfacing with the IBM Parallel Printer Port," Am. J. Phys. **57**, 561–562 (1989).

[2]W. Press, B. Flannery, S. Teukolsky, and W. Vetterling, *Numerical Recipes* (Cambridge U. P., Cambridge 1986), pp. 381–453.

Photographing helium–neon laser light

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Red helium–neon lasers are commonly used as light sources in undergraduate laboratory experiments. It frequently happens that one wants to record or make measurements from, for example, a diffraction pattern or an image produced using such a laser. If the pattern is large enough, a photodiode or similar detector can be scanned through the pattern and the detector output recorded as a function of position. On the other hand, smaller patterns are probably best recorded photographically, and any required measurements can then be made from the photograph. Usually this involves exposing photographic film, which must then be processed, washed and dried, and then making a positive print from the negative. This can take some time and requires a wet darkroom for processing the film. An alternative is to record the pattern directly on photographic paper, which can then be processed in a darkroom, or with a processing machine. The problem with this is that most photographic papers are, by design, insensitive to red light. This means that long exposure times are required, and even then the quality of the prints is not high.

The purpose of this note is to point out a type of photographic paper that is sensitive to red light and is therefore very useful for recording patterns and images in helium–neon laser light. Kodak Panalure II paper (soon to be superceded by Panalure Select, which has similar properties) is intended for making black and white prints from color negatives, and thus has a much broader spectral sensitivity than the usual black and white papers. It can be processed with the usual chemicals in a darkroom, or, more conveniently, with a processing machine right in the lab. Using this paper, high quality photographs of images and diffraction patterns formed with helium–neon laser light are easy to obtain, using exposure times of, typically, 1 to 30 s. By way of comparison, exposure times of several minutes with normal papers produce much less satisfactory results.