

Digital Data Acquisition

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Digital electronic instruments have come to dominate many types of information systems, including the acquisition of experimental data. The digital representation of an analog signal offers several advantages: a digital signal can be stored in volatile (RAM) or permanent (magnetic) memory, it can be reproduced error-free, and it can be imported into a computer for manipulation and analysis.

There are two broad types of digital data acquisition systems commonly in use. The first consists of a computer, such as a PC or Macintosh, with a specialized plug-in board that performs analog-to-digital (A/D) and digital-to-analog (D/A) conversion. With the proper software to control the A/D board, the computer can act very much like a multi-channel oscilloscope, function generator, multimeter, or other dedicated instrument. The second type of digital measurement system consists of a computer with an interface board that permits it to communicate with 'real' external digital instruments, such as digital oscilloscopes, function generators, etc. The interface, known as GPIB (General-Purpose Interface Bus) or HPIB or IEEE 488, is a standard, similar in function to a serial (RS-232) interface that is commonly used to control printers and other computer peripherals. Both of these types of data acquisition systems are available in the ITLL. In this experiment, we will use computer-controlled GPIB instruments.

Figure 1 shows a schematic of the digital representation of an analog signal, in this case a 0.10 Hz sine wave. The time-varying continuous analog voltage is 'sampled' and converted into discrete values as a function of time. Typically a signal is sampled as a function of time. The **sample rate** is defined as $f_s = 1/\delta t$, where δt is the time between samples. The total length of time stored in memory is $(N-1) \times \delta t$ where N is the **number of samples**. The total number of samples taken may be set by the memory of the instrument (2,000 for a typical digital oscilloscope), or may be limited only by the available RAM in the computer with an A/D board.

How well the digital signal represents the original analog signal depends on the sample rate and number of samples taken. The faster the sample rate, the more closely the analog waveform may be described. If the sample rate is too low, errors may be experienced and the nature of the waveform will be lost. Figure 2 shows what happens to the digital representation of a 10 Hz sine wave when the sample rate is: (b) $f_s = 100$ samples/sec, (c) $f_s = 27$ samples/sec, (d) $f_s = 12$ samples/sec. As the sample rate decreases, the amount of information per unit time describing the signal decreases. In Figure 2 (b) and (c), the 10 Hz frequency content of the original signal can be discerned; however, at the slower rate, in (c), the representation of amplitude is distorted. At the even lower sample frequency, as in (d), the apparent frequency of the signal is highly distorted. This is called aliasing of

the signal. The way to avoid aliasing is to make sure that the sample rate f_s is at least twice the maximum frequency component of the analog signal:

$$f_{\text{sample}} \geq 2f_{\text{max}} \quad (1)$$

The minimum sample rate is known as the **Nyquist frequency**. In the case that $f_s < 2f_{\text{max}} < 2f_s$, the lowest image frequencies will be at $f_s - f_{\text{max}}$, the lowest image frequency in a sense *masquarades* as the sinusoid that was sampled and is called an **alias** of the sinusoid that was actually sampled, albeit inadequately sampled.

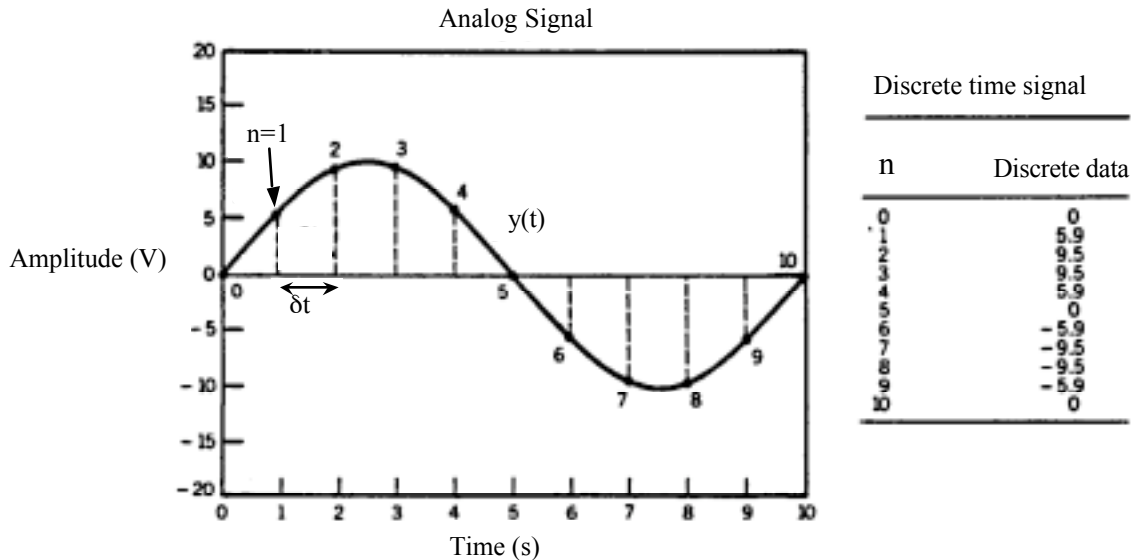


FIGURE 1. Analog signal and digital representation of a 0.10 Hz sine wave. [1]

An alternate method of avoiding aliasing is to filter the signal before sampling. This way higher frequency signals, like noise, is removed prior to sampling and thus is not aliased into the desired signal.

The digital representation of the analog signal, shown in Figure 1, is discrete in amplitude as well as time. The increment in time is the sample rate; the increment in amplitude is the resolution of the A/D converter. The voltage resolution of an A/D converter is given by

$$\epsilon_V = \frac{\Delta V_{fs}}{(2^n - 1)} \quad (2)$$

Where ΔV_{fs} is the full-scale voltage range and n is the number of bits in the A/D converter. Typical A/D converters have 8, 12, or 16 bits, corresponding to a division of ΔV_{fs} into $2^8=256$, $2^{12}=4096$, and $2^{16}=65,536$ increments. For example, an 8-bit converter with a -10 to +10 V range has a voltage resolution of 78 mV and a 16-bit converter with the same

voltage range has a resolution of 0.3 mV. In other words, any voltage value cannot be specified any more precisely than the resolution of the A/D converter. The HP digital storage oscilloscope used in this experiment has an 8-bit A/D converter. This low resolution is selected so that the A/D converter can operate at very high speeds (up to 20 Msamples/s). In contrast, music CDs are 16 bit in order to achieve high fidelity. This is possible because D/A converters in CD players operate relatively slowly ($f_s < 44$ ksamples/s) which is adequate for audio frequencies ($f_{\max} < 20$ kHz).

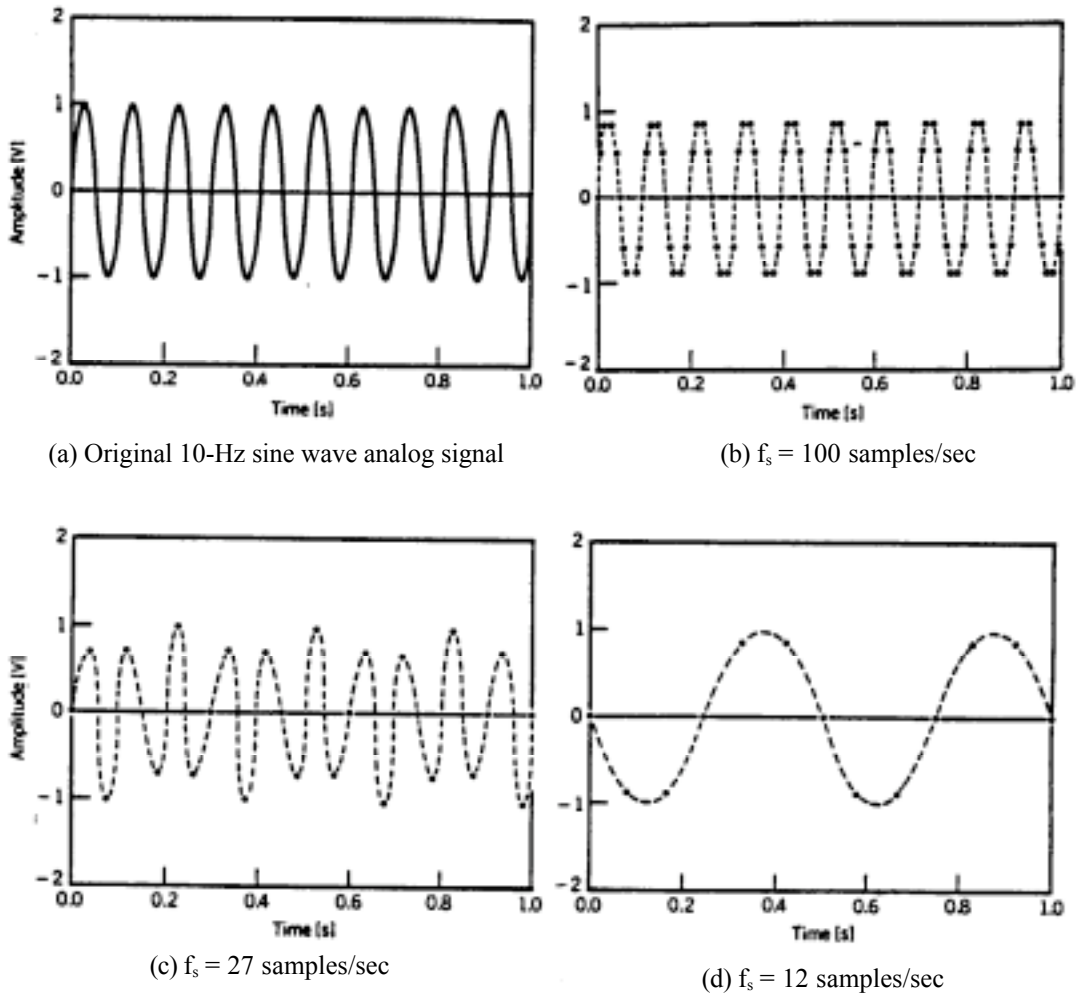


FIGURE 2. Effect of sample rate on digital signal amplitude and frequency. [1]

References

- Holman, *Experimental Methods for Engineers*, 7th Edition, McGraw-Hill, New York, 2001, A/D conversion, aliasing 14.5 p. 588.
- Figliola and D.E. Beasley, *Theory and Design for Mechanical Measurements*, Wiley, New York, 1991, p. 225.

- Wheeler, A.J. and Ganji, A.R., *Introduction to Engineering Experimentation*, Prentice-Hall, 1996, Chapters 4 and 5. (This is a short intro to A/D concepts)