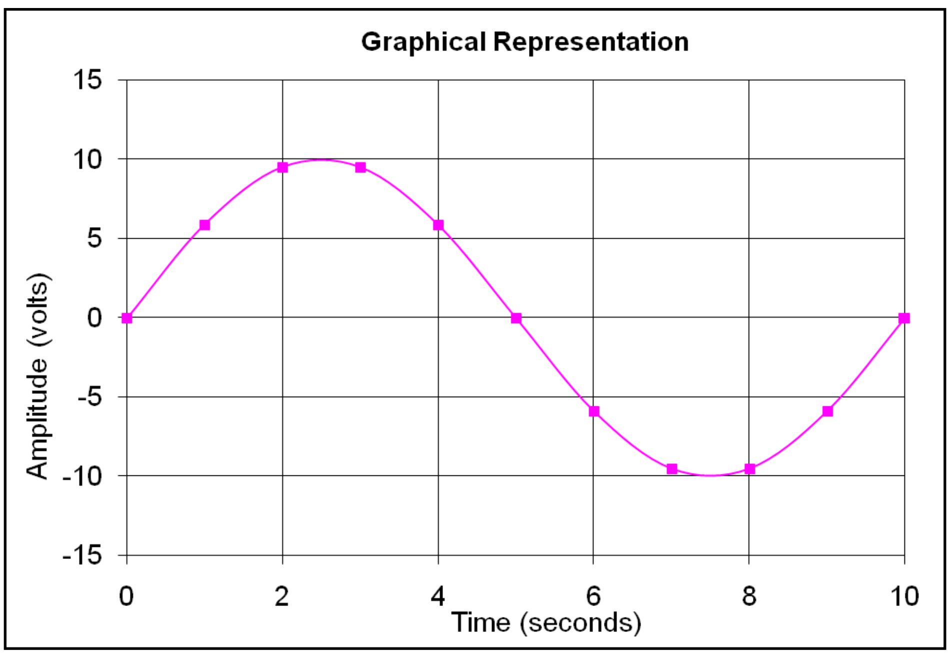
# Dynamic Signal Acquisition

## Aliasing

How rapidly do we need to sample a signal in order to accurately describe it? The answer depends on how fast the input signal is changing. The speed of a signal's fluctuations is referred to as the frequency content of the signal.

**Period** - The length of time of a single cycle of a fluctuating signal. In the signal below the period is 10 seconds.

**Frequency** - describes the time variation of a signal. It is the inverse of period and has units of 1/time. The signal plotted below has a frequency of 1 cycle every 10 seconds or 1/10 cycle/sec or 0.1 **Hertz (Hz)**.




If the sampling rate is too slow, aliasing will occur. That is, the signal will appear to have a frequency in the sampled data which is not the same as that of the original input signal. Aliasing always causes the sample data to have a frequency which is lower than the input signal frequency. Digital sampling always results in some level of distortion of the input signal (see Figure 7.2 b & c). Although the signal is distorted the period of the input signal can be reasonably ascertained and therefore the frequency can be reasonably approximated. If the signal is sampled to slowly then aliasing will occur resulting in a sampled wave form with a frequency that is lower than the original. In Figure 7.2d the period of the sampled wave form is 0.5 seconds which is significantly different than the 0.1 second period of the input wave form. The input signal has been aliased the apparent frequency of the sampled data is now 2 Hz where as the input signal has a frequency of 10 Hz.

The sampling theorem provides the guideline needed to avoid aliasing. The input signal must be sampled at least twice in shorted period of the signal. This is equivalent to saying the sampling frequency must twice the highest frequency in the input signal.

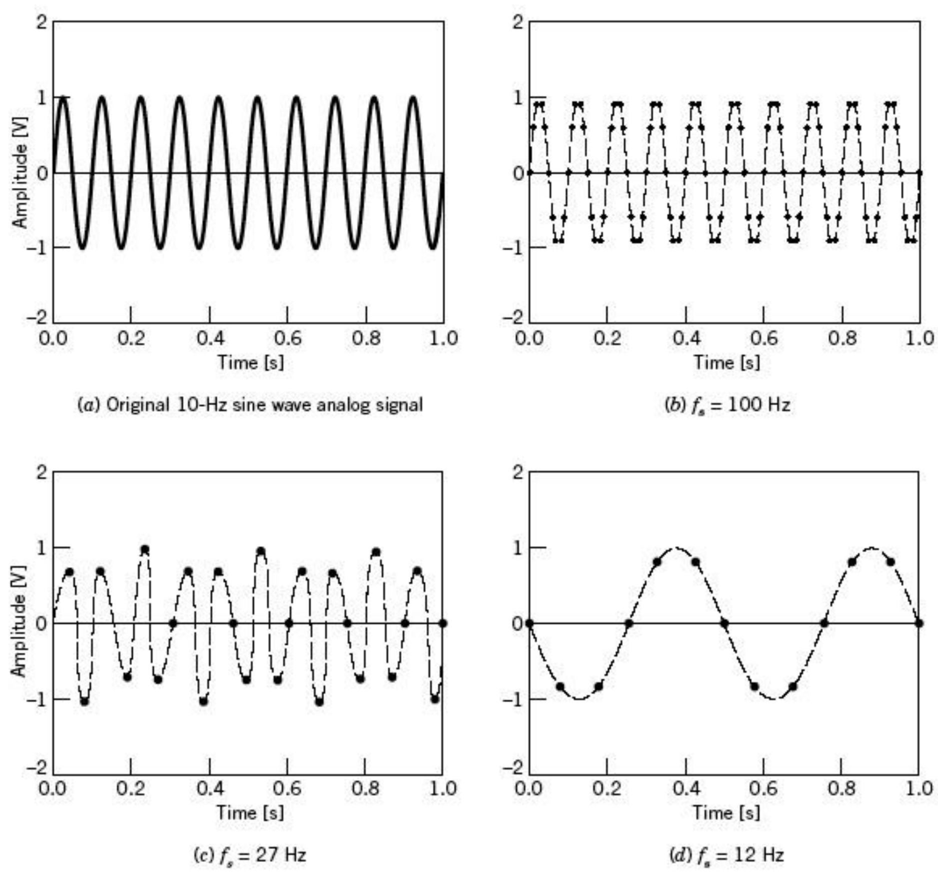



**Figure 7.2** The effect of sample rate on signal frequency and amplitude interpretation.

**Nyquist Frequency, $f_N$**: The maximum frequency component which can be represented in a digitally sampled data set.

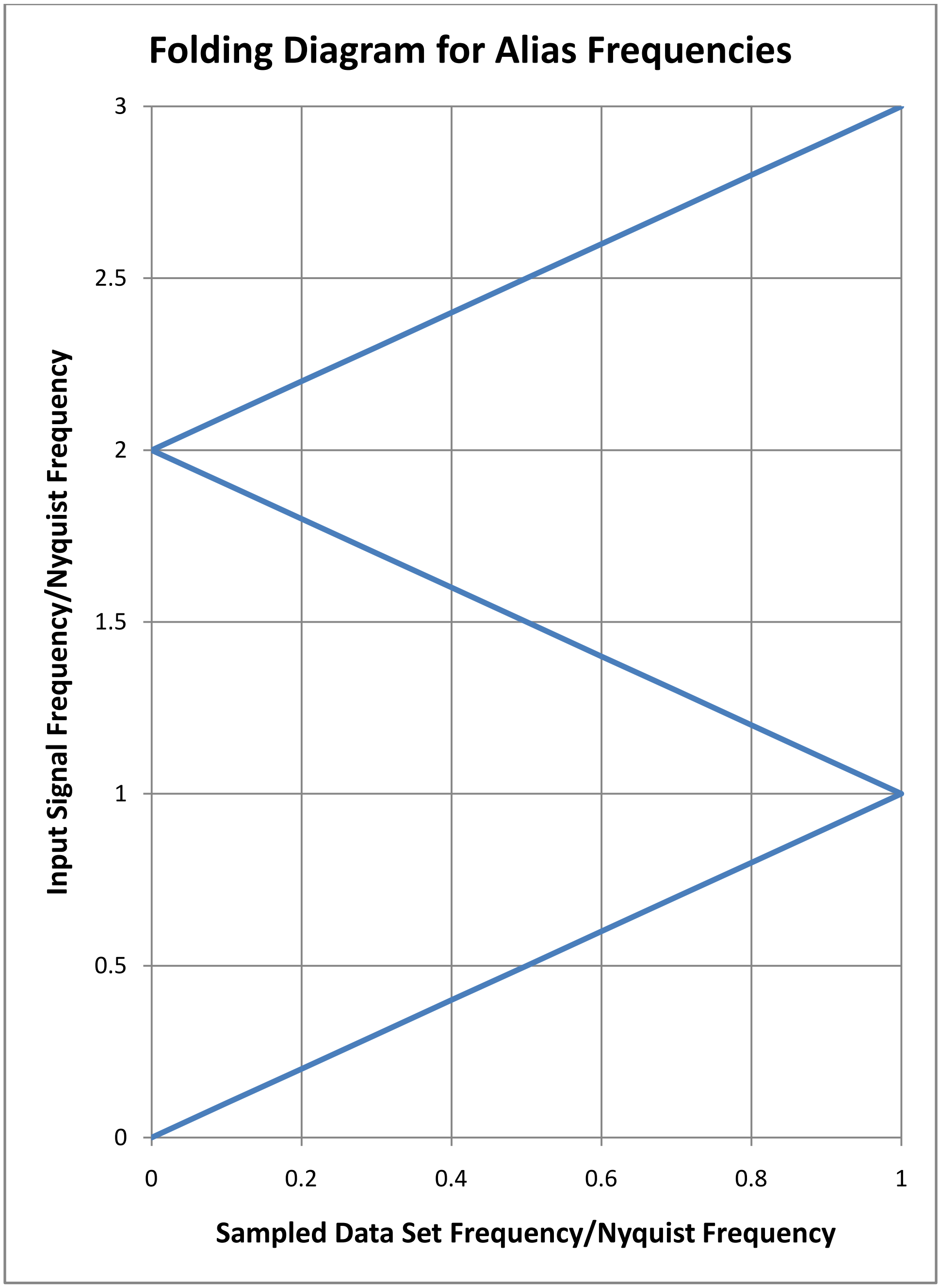
$$f_N = \frac{f_s}{2} = \frac{1}{2\delta t}$$

where $\delta t$ is the sampling interval. The **Nyquist Criteria** is satisfied if at least 2 data samples are acquired in the fastest cycle (shortest period, highest frequency) of the input signal. The Nyquist frequency is half the sampling rate. The Nyquist Criteria is satisfied if the sampling rate is twice the highest frequency in the input signal.

The A/D converters in the lab have a maximum sampling rate of 100,000 Hz with the software we are using. Thus we can determine the frequency of input signals up to about 50,000 Hz.

Review: The converters have a resolution of 12 bits, so the quantization error is 1 part in $2^{12}$ or 1 part in 4096 of full scale. The maximum Nyquist frequency of our ADC is 50 KHz. The gains range from 1 to 200.

The following figure may be easier to understand than “Figure 7.3 The folding diagram for alias frequencies” in the text book.




In the above diagram notice that the sampled data set frequency corresponds to the input signal frequency up to the Nyquist frequency. When the input frequency is above the Nyquist frequency aliasing occurs and the sampled data set frequency is less than the Nyquist frequency. This is called frequency folding. Remember the maximum data set frequency is the Nyquist frequency (½ the sampling frequency).

Homework: Take time to interact with the spreadsheet used to create the above diagram, AliasingFrequencyDiagram.xlsx, and the ADC Simulation spreadsheet.