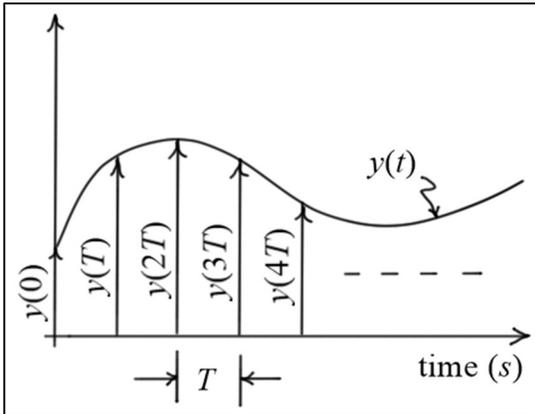


Introductory Motion and Control Sampled Data

When a *discrete compensator* is used to control a *continuous system*, some of the signals are *continuous functions* of time, and some are *sampled* at a discrete set of times. *Sampled data* may be generated directly by a *digital computation* or by *sampling a continuous signal* using an *analog-to-digital* converter. In either case, a continuous function is represented by a discrete set of values as shown below.



Sampled Data for a Continuous Function

$$t = [0, T, 2T, \dots, NT]$$

$$y(kT) = [y(0), y(T), y(2T), \dots, y(NT)]$$

Two important characteristics of sampled data are the *sample period* T and the *resolution*.

The *raw speed of the converter* and the *number of channels* determine the *smallest sample period* a converter supports. In some systems, each channel of data has its own *dedicated* converter, while in others a single converter is *shared* by multiple channels by using a multiplexer. The *resolution* is determined by the *number of bits* used in the conversion process.

It is important to sample using a *sufficiently small period* and sufficient resolution to accurately represent the original signal. If a signal is sampled at a rate of $1/T$ samples per second, then the *highest frequency captured* by the sampled data is

$$\omega_{\max} = \frac{1}{2} \omega_s = \frac{1}{2} \left(\frac{2\pi}{T} \right) \text{ (rad/sec)} \dots \text{ known as the } \mathbf{Nyquist \textit{frequency}}$$

Aliasing will occur if there is *significant energy* in the original signal above the Nyquist frequency. The signal content above the Nyquist frequency will be *erroneously interpreted* to be at lower frequencies.

Thus, if *aliasing* occurs, the *response at frequencies above the Nyquist frequency will be missed*, and the *response at lower frequencies will be distorted!*

Anti-aliasing (low pass) filters may be used to reduce the frequency content in the original signal above the Nyquist frequency, but *phase effects* will be noticeable at frequencies a *decade* below the *cut-off frequency* of the filter. These *phase effects* make the system appear to respond *slower* than it does.

It is also important to use the **full** voltage **range** of the *ADC* and *DAC* converters and use enough bits to provide a desirable **resolution**. The **quantization error** for a converter using “*n*” bits in the conversion process is

$$e_Q = \frac{V_{\max} - V_{\min}}{2^n - 1}$$

Here, $V_{\max} - V_{\min}$ represents the **maximum** voltage **range** of the converter.