Communication Systems

Module 2

Analog to Digital Conversion

- **Analog-to-digital conversion** is (basically) a 2 step process:
  - Sampling
    - Convert from continuous-time analog signal $x_a(t)$ to discrete-time continuous value signal $x(n)$
  - Is obtained by taking the "samples" of $x_a(t)$ at discrete-time intervals, T_s

- **Quantization**
  - Convert from discrete-time continuous valued signal to discrete time discrete valued signal

Sampling

- **Sampling** is the processes of converting continuous-time analog signal, $x_a(t)$, into a discrete-time signal by taking the "samples" at discrete-time intervals
  - Sampling analog signals makes them discrete in time but still continuous valued
  - If done properly (***Nyquist theorem*** is satisfied), sampling does not introduce distortion
- **Sampled values:**
  - The value of the function at the sampling points
- **Sampling interval**:
  - The time that separates sampling points (interval b/w samples), T_s
  - If the signal is slowly varying, then fewer samples per second will be required than if the waveform is rapidly varying
  - So, the optimum sampling rate depends on the *maximum frequency* component present in the signal

Sampling

- **Sampling Rate (or sampling frequency f_s):**
  - The rate at which the signal is sampled, expressed as the number of samples per second (reciprocal of the sampling interval), $1/T_s = f_s$

- **Nyquist Sampling Theorem (or Nyquist Criterion):**
  - If the sampling is performed at a proper rate, no info is lost about the original signal and it can be properly reconstructed later on
  - Statement:

"If a signal is sampled at a rate greater than or equal to twice the max frequency component of the message signal, then the signal can be exactly reconstructed from the samples without any distortion"

$$f_s \geq 2 f_{\max}$$

Sampling Theorem - Example

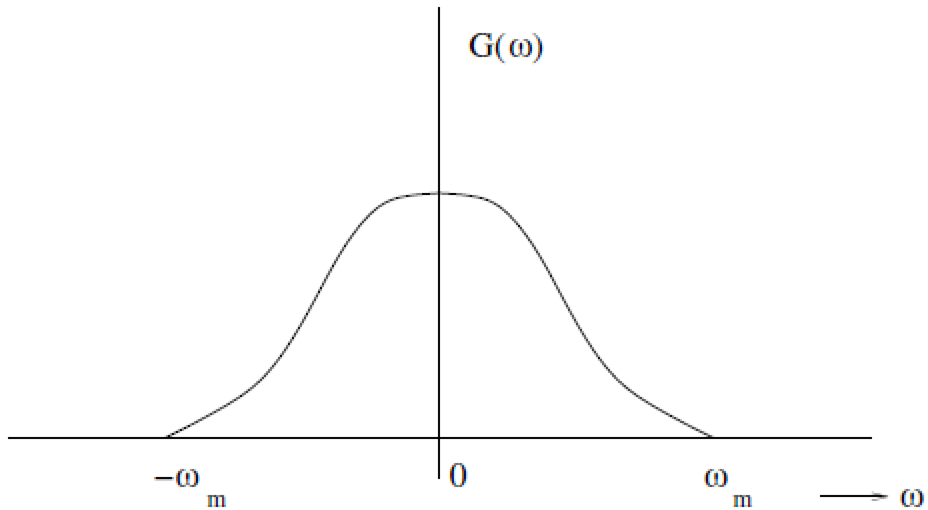


Figure 1: Spectrum of bandlimited signal g(t)

The maximum frequency component of g(t) is fm. To recover the signal g(t) exactly from its samples it has to be sampled at a rate fs 2fm.

The minimum required sampling rate fs = 2fm is called Nyquist rate.

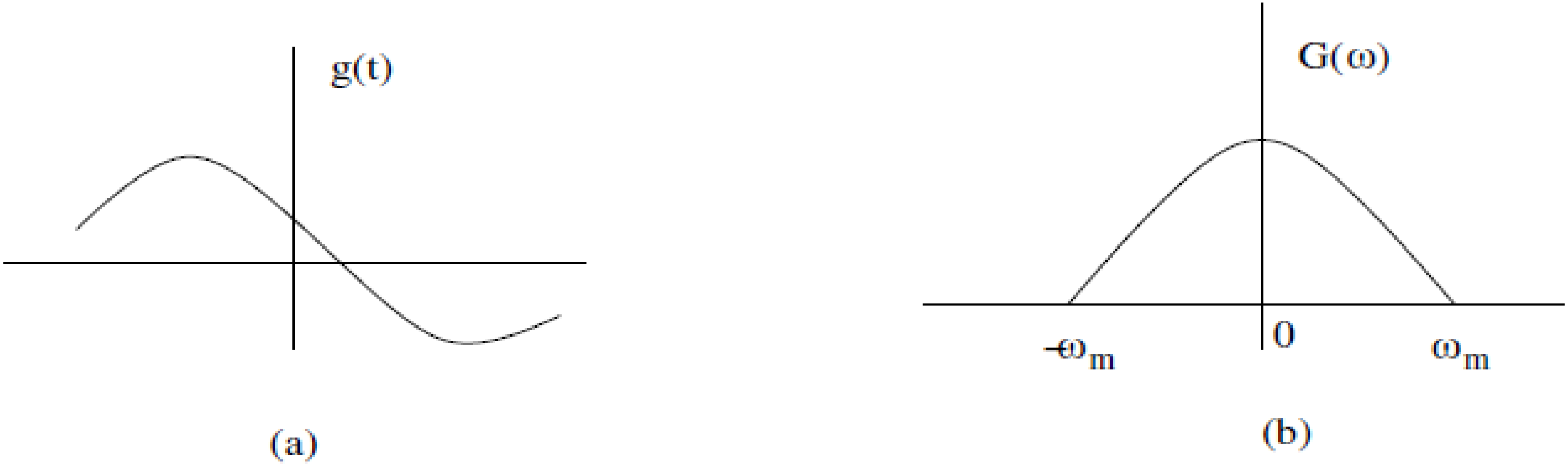


Figure 2: (a) Original signal $g(t)$ (b) Spectrum $G(\omega)$

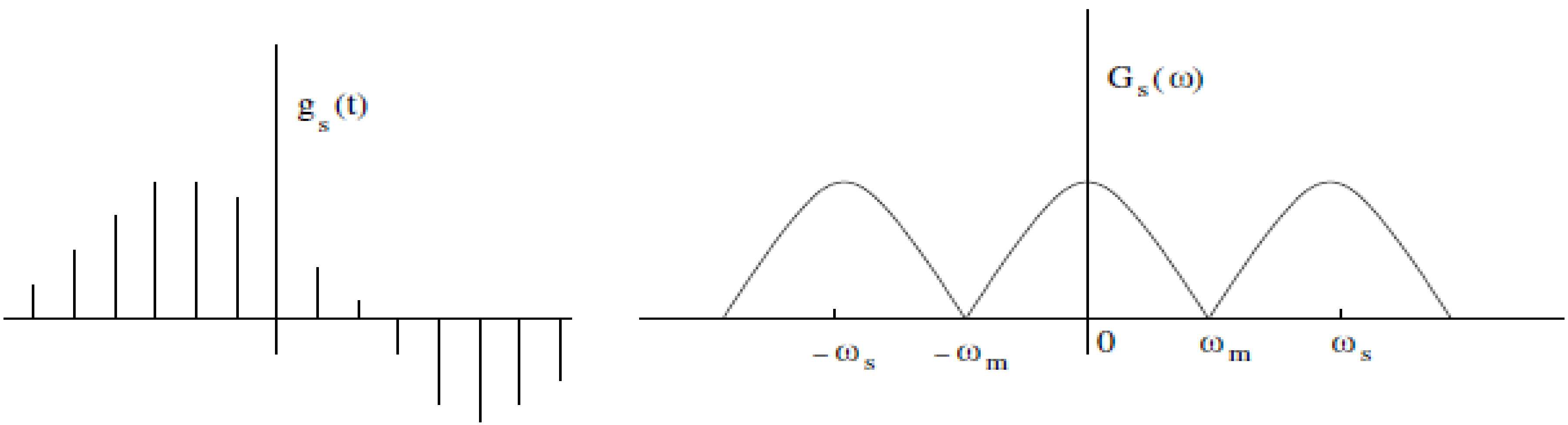


Figure 4: (a) sampled signal $g_s(t)$ (b) Spectrum $G_s(\omega)$

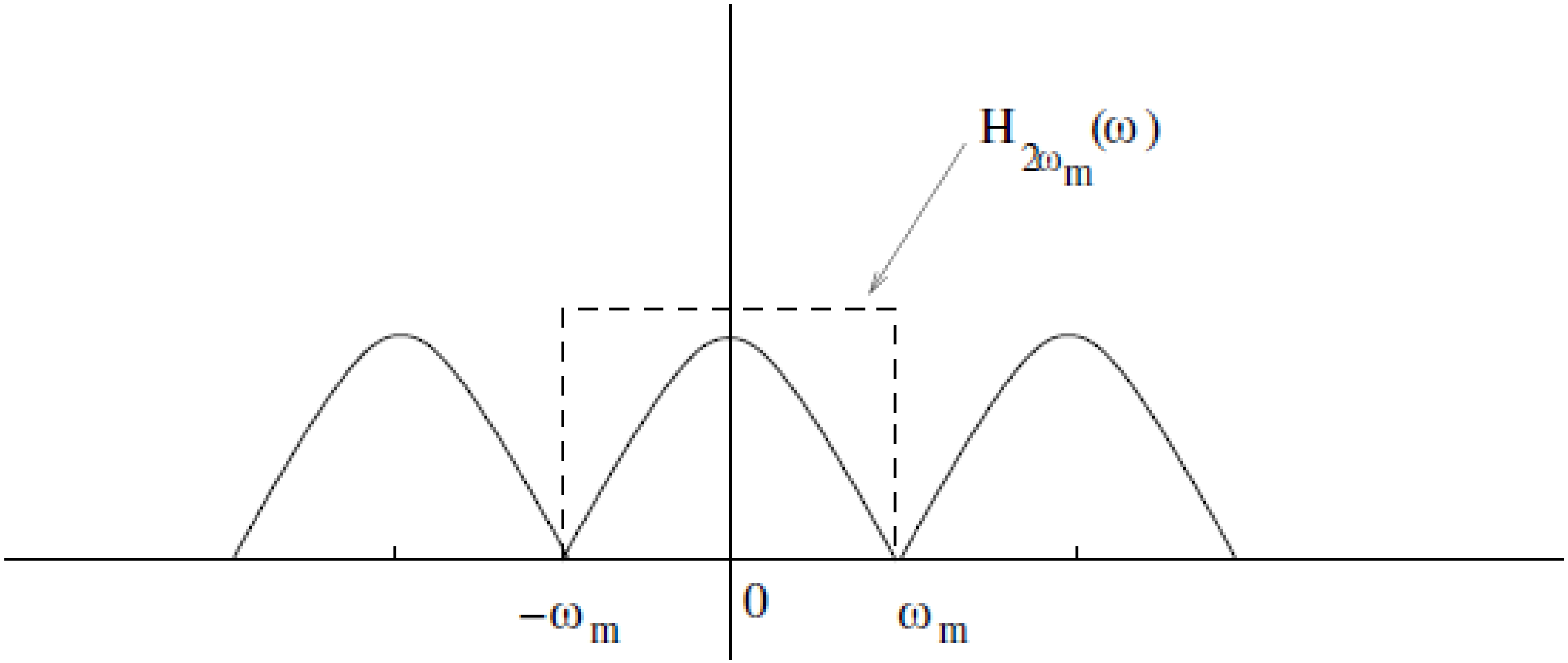


Figure 5: Recovery of signal by filtering with a filter of width $2\omega_m$

- Aliasing
  - Aliasing is a phenomenon where the high frequency components of the sampled signal interfere with each other because of inadequate sampling $\omega_s < 2\omega_m$.

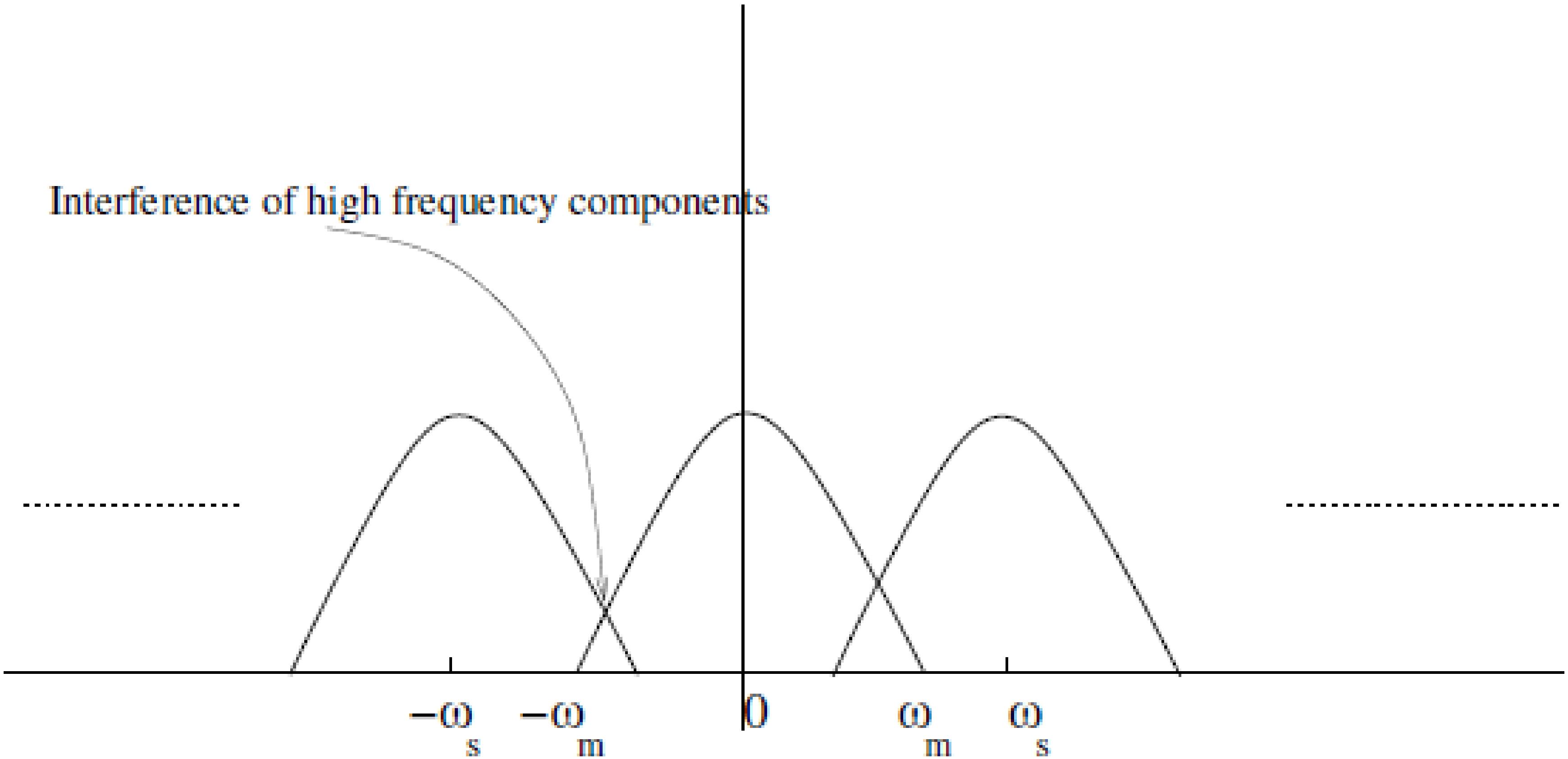


Figure 6: Aliasing due to inadequate sampling

- Oversampling
  - In practice signal are oversampled, where f_s is significantly higher than Nyquist rate to avoid aliasing.

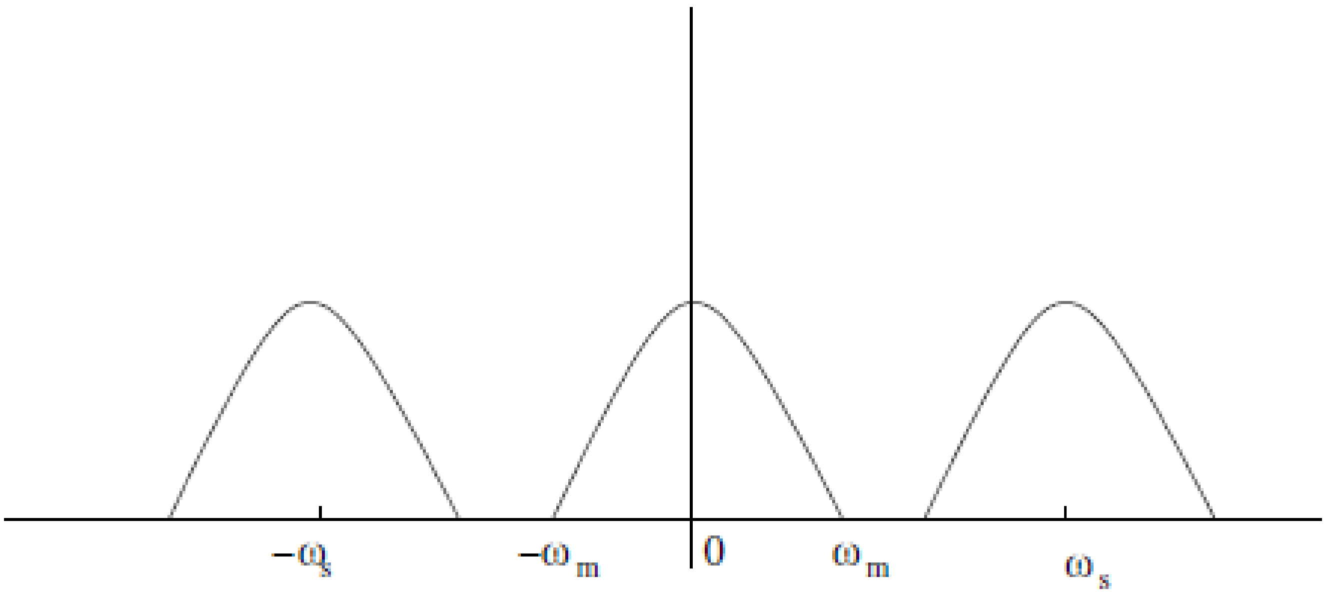


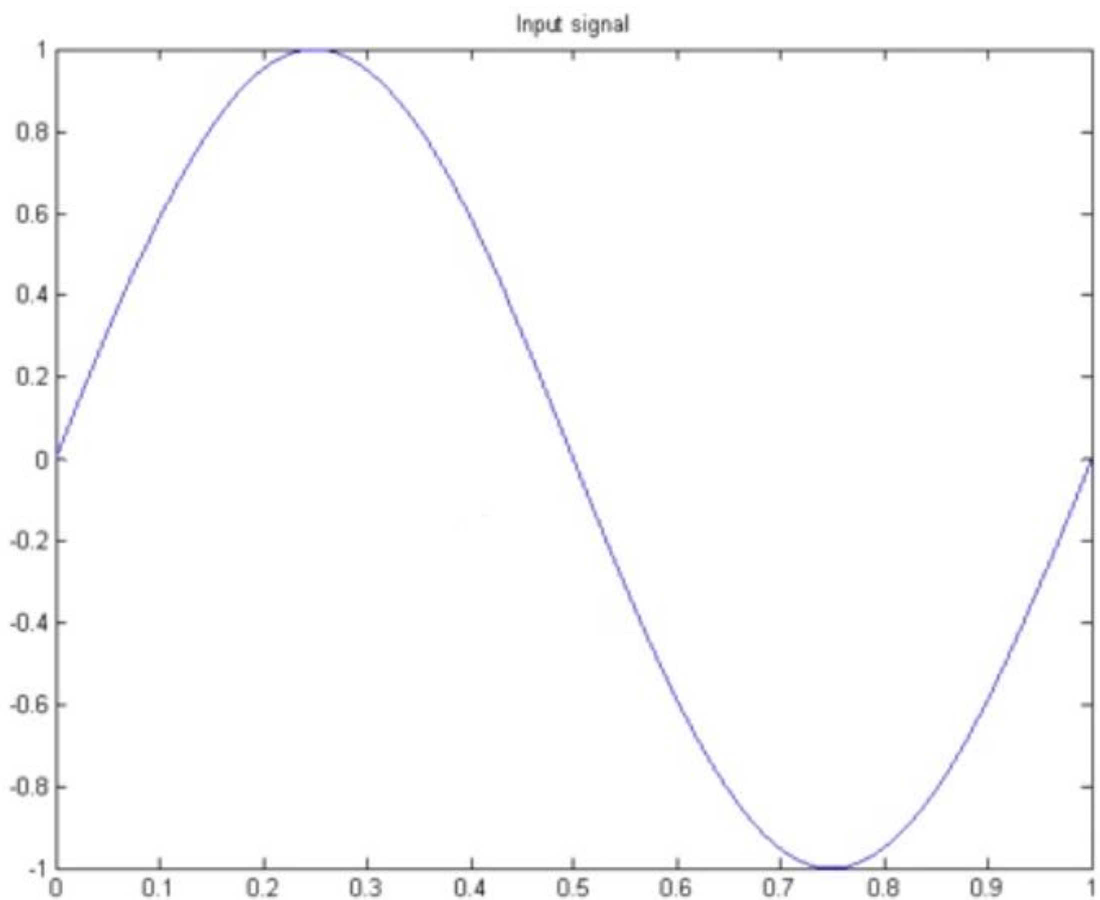
Figure 7: Oversampled signal-avoids aliasing

Quantization

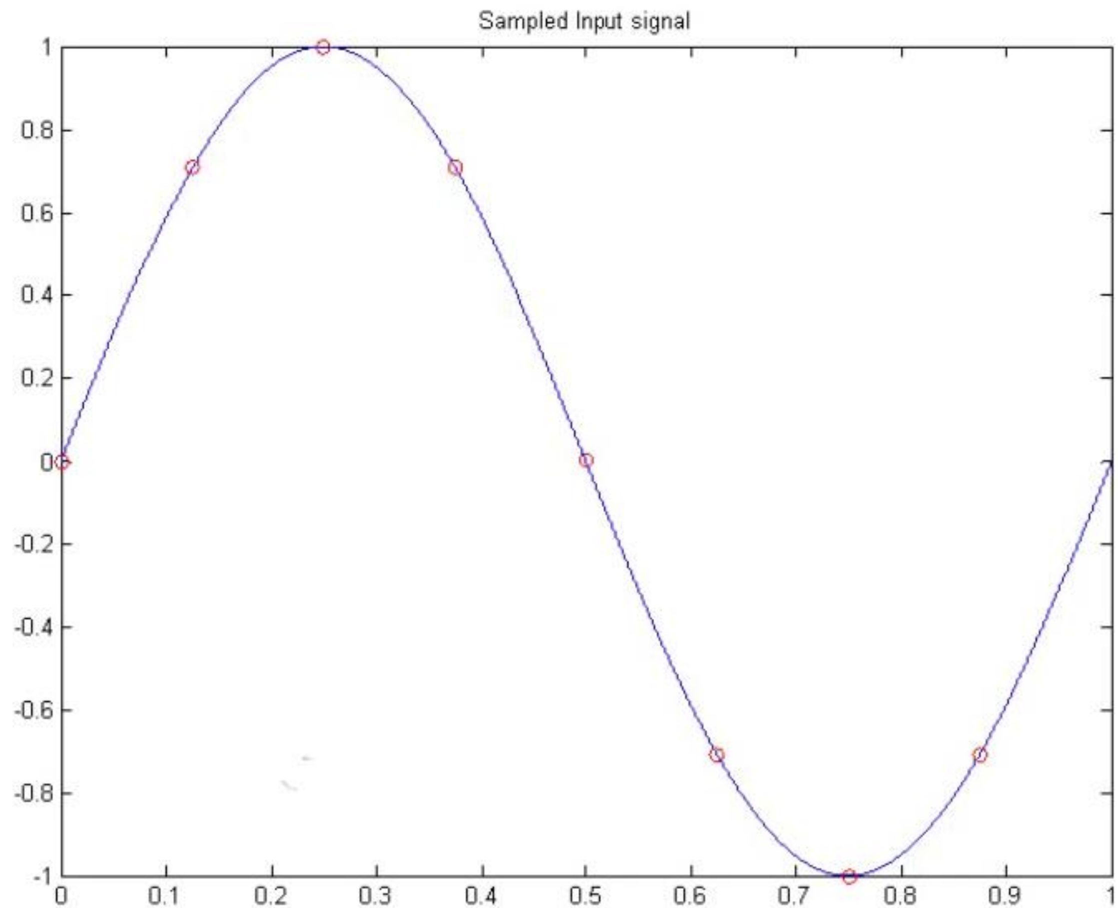
- Sampling results in a series of pulses of varying amplitude values ranging between two limits: a min and a max.
- The amplitude values are infinite between the two limits.
- We need to map the *infinite* amplitude values onto a finite set of known values.
- This is achieved by dividing the distance between min and max into L levels, each of height Δ.

$$\Delta = (\max - \min)/L$$

Input Signal: Sinusoidal Signal



Sampled Signal: 8 samples/sec



Quantization with 4 Levels