ECE3614
Introduction to Communications Systems
Fall 2007

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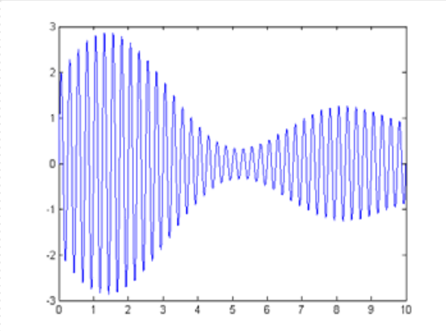
Lecture #21: Introduction to Digital Communications

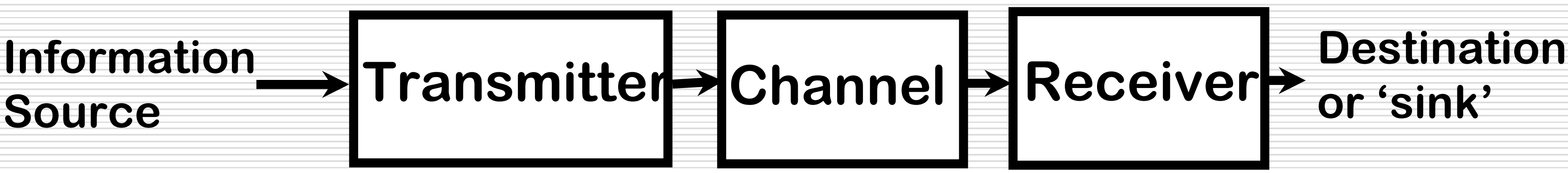


Overview

- To date we have assumed that the communication systems that we analyze are *analog*
- Today we will briefly introduce the concept of *digital communications.*
- Specifically we will examine digital communications of analog signals that requires
  - Sampling
  - Quantization
  - Waveform encoding
- We will examine sampling today and will examine quantization and waveform encoding in the next lecture
- Reading
  - 5.1 – 5.3

Communications

- **Definition:** Communications is the transfer of information at one time or location to another time or location



Can be analog or digital. Almost always baseband signal

Converts information into appropriate waveform
Can include analog-to-digital conversion, modulation and waveform coding. Can be baseband or bandpass

Transports and corrupts signal

Makes best guess as to what the transmitted signal was

Transmitting Information

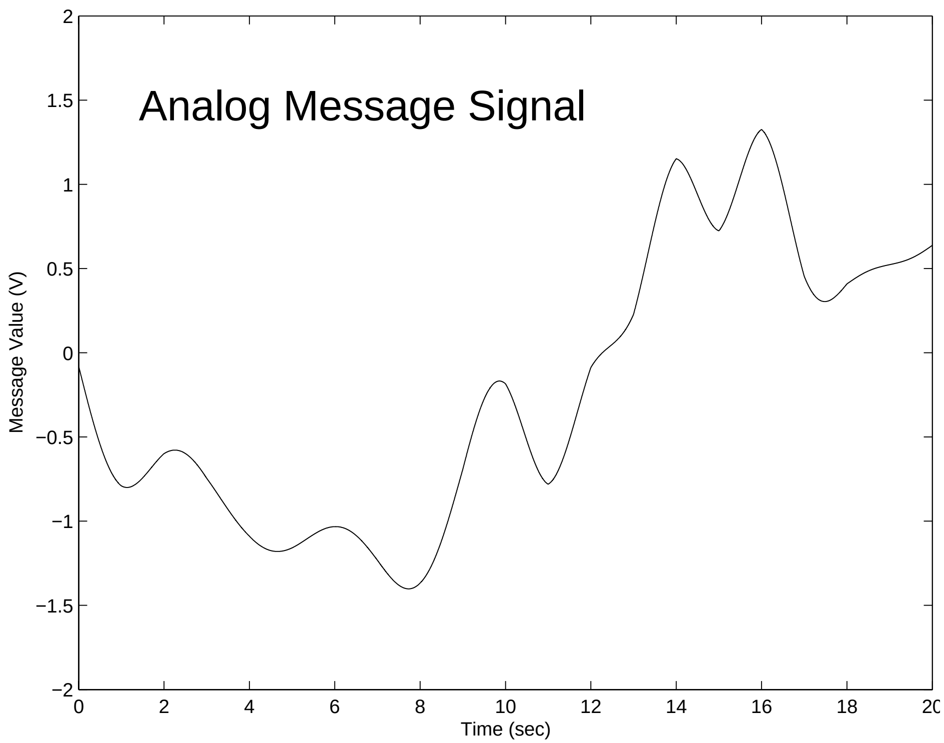
- The basic function of a communication system is to transfer the source information from source to sink
- The transmitter converts the message signal to a format suitable for transmission
- The message sent can be
  - Analog or digital
  - Baseband or bandpass
- So far in this class we have considered an analog message which modulates a sinusoidal carrier (bandpass)
- We will now focus on converting an analog signal to a digital signal for digital modulation of either a pulse stream (baseband) or a sinusoidal carrier (bandpass)

Analog vs. Digital

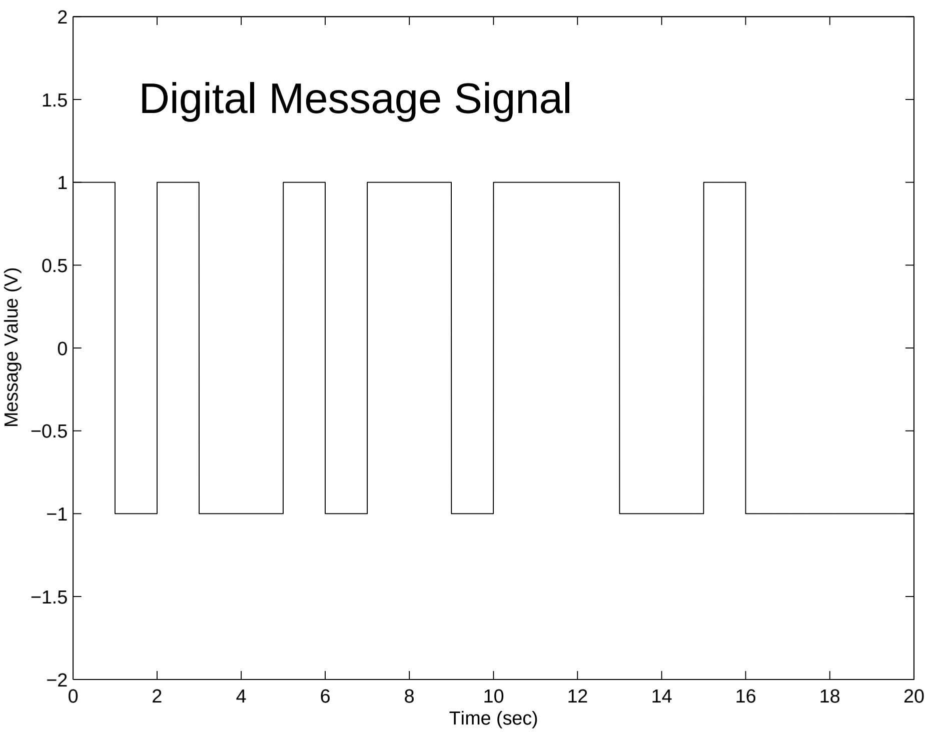
- Analog Communications
  - The message signal can take on an infinite number of possible values
  - Directly uses an analog information source as the message to be sent
- Digital Communications
  - The message signal must be one of a small number of discrete messages
  - Must convert analog signals into a sequence of discrete messages
  - If the number of possible messages is 2, the system is *binary* and the messages are termed binary digits or *bits*.

Example

- Analog system that transmits a voltage value between -2V and +2V.



- An example digital system transmits either +1V or -1V



Baseband vs Bandpass

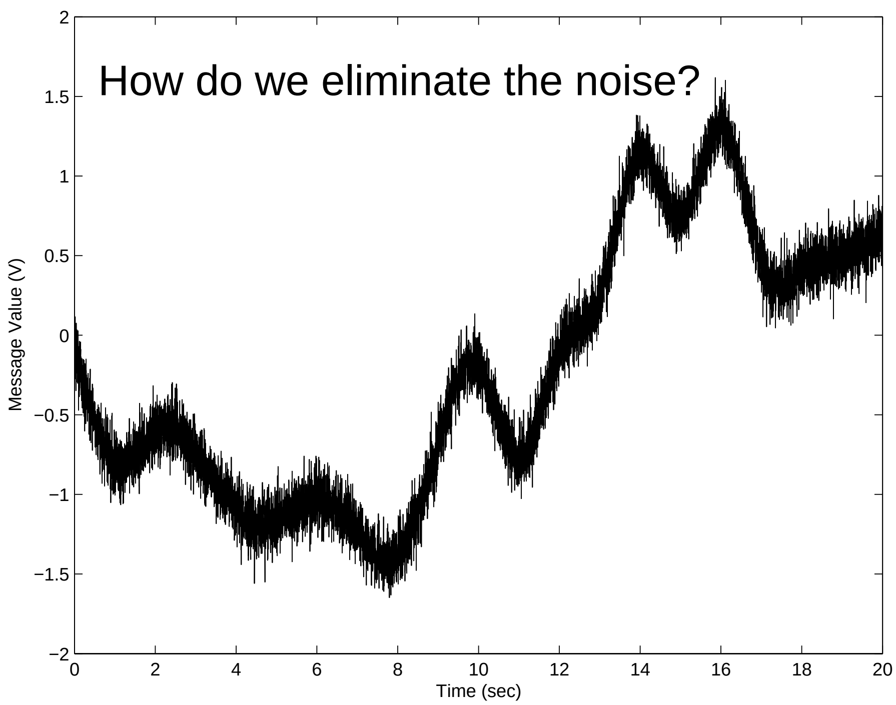
- The transmitter's job is to convert the information source into a waveform suitable for transmission.
- The resulting transmit signal can either be "baseband" or "bandpass"
  - Baseband - frequency content is primarily around DC
    - signal is typically a series of modulated pulses
  - Bandpass – frequency content is primarily around some center frequency $f_c >> 0$
    - signal is typically a modulated sinusoid

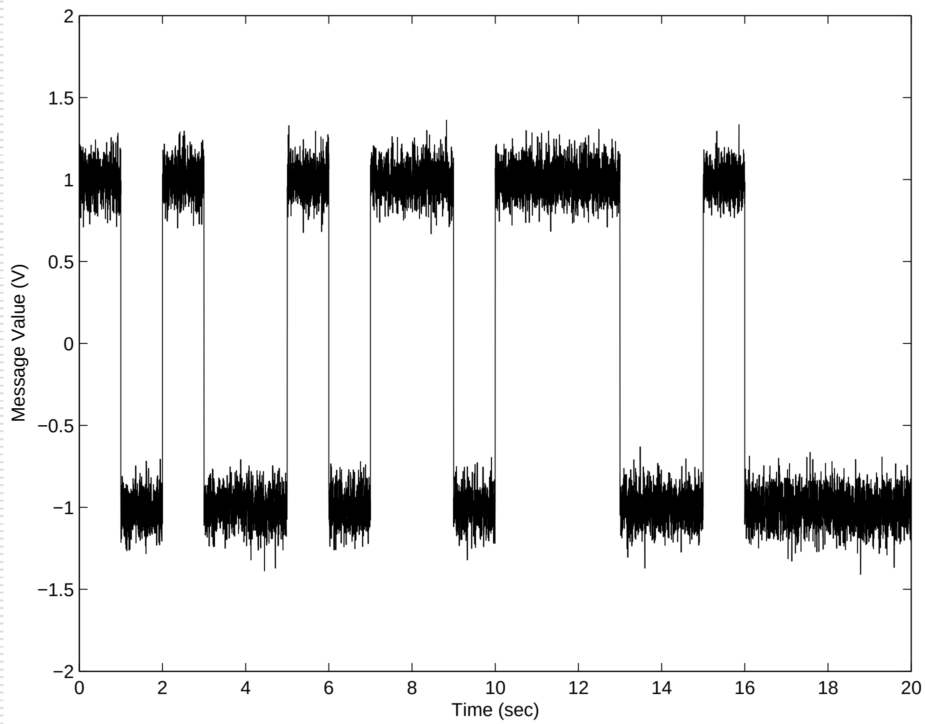
Why digital communications?

- Any noise introduces distortion to an analog signal. Since a digital receiver need only distinguish between a finite number of waveforms it is possible to recover digital information without corruption a large percentage time.
- Many signal processing techniques are available to improve system performance: source coding, channel (error-correction) coding, equalization, encryption
- Digital ICs are inexpensive to manufacture. A single chip can be mass produced at low cost, no mater how complex
- Digital communications allows integration of voice, video, and data on a single system
- Digital communications systems provide a more flexible tradeoff between bandwidth efficiency and energy efficiency than analog communications

Example Revisited

- When noise is added to the signal, all of the values are still valid
- When noise is added to the signal, the resulting values are not valid, thus we can correct

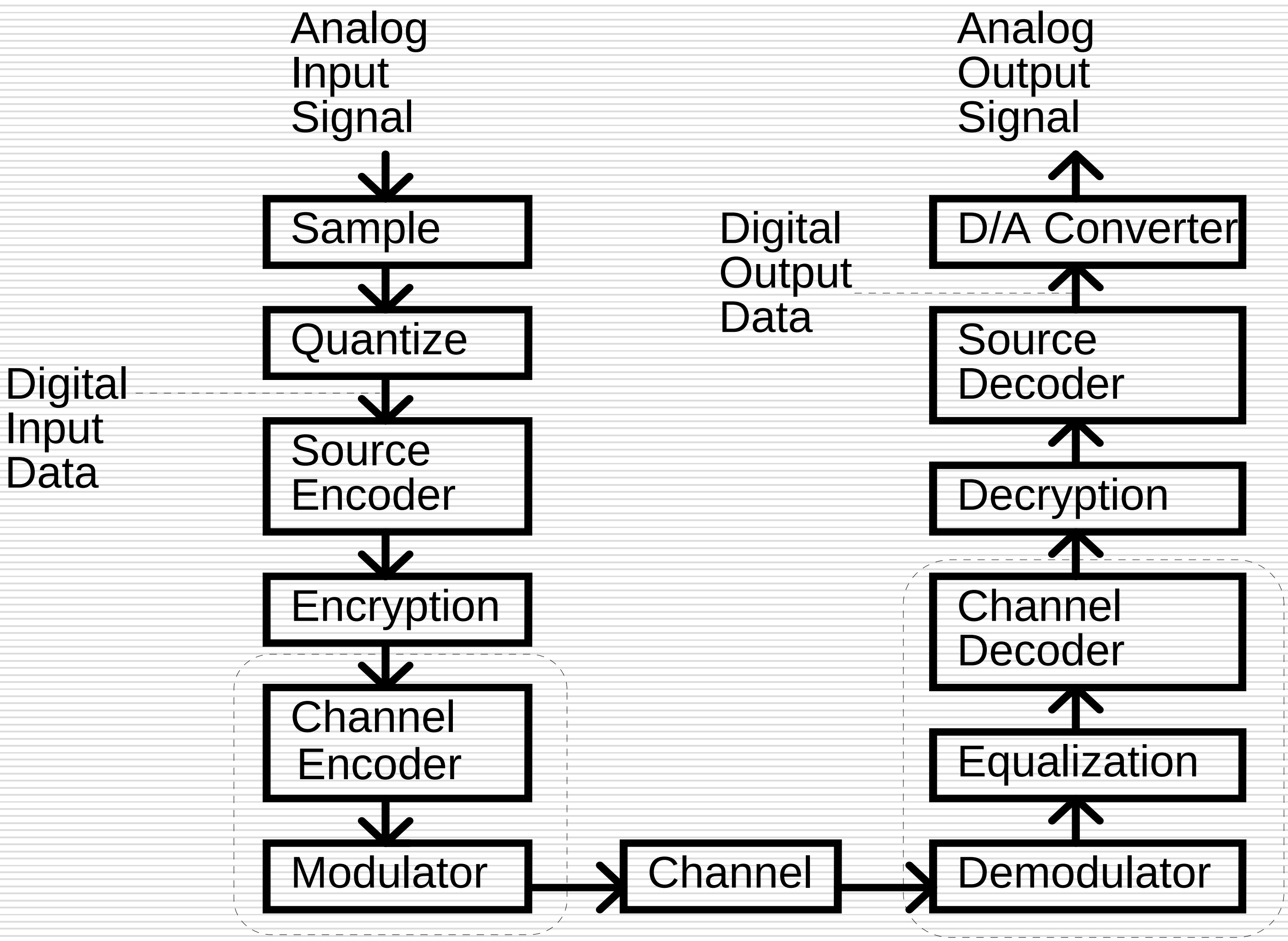




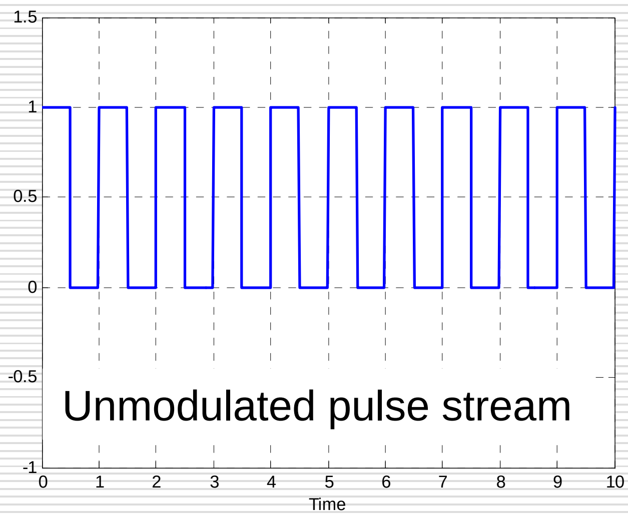
Limitation of Digital

- Analog system can naturally represent all of the message signal values
- Digital systems cannot represent all possible input values, thus, if the message is analog then all of the information can not be transmitted
- The process of converting an analog input signal to a digital signal is termed *analog-to-digital conversion* and is a lossy process

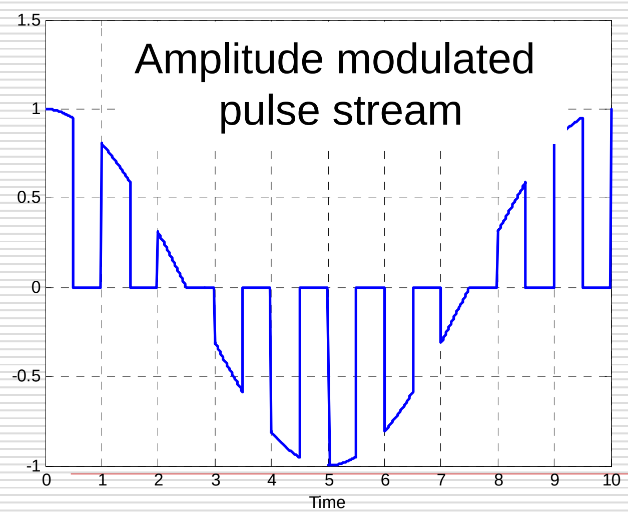
Block Diagram of Digital Communications System

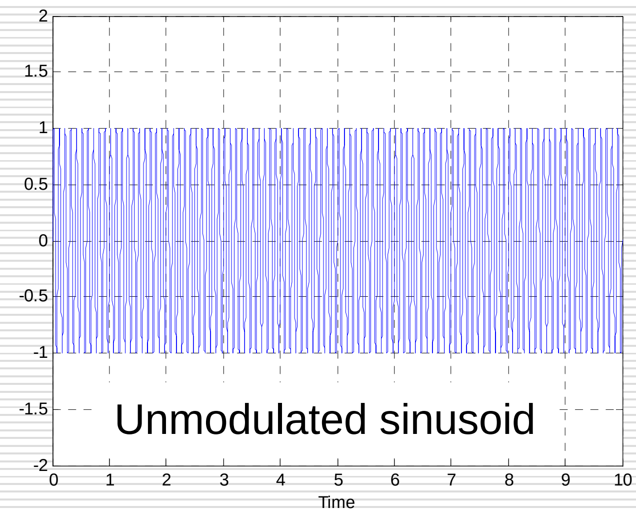


Modulation

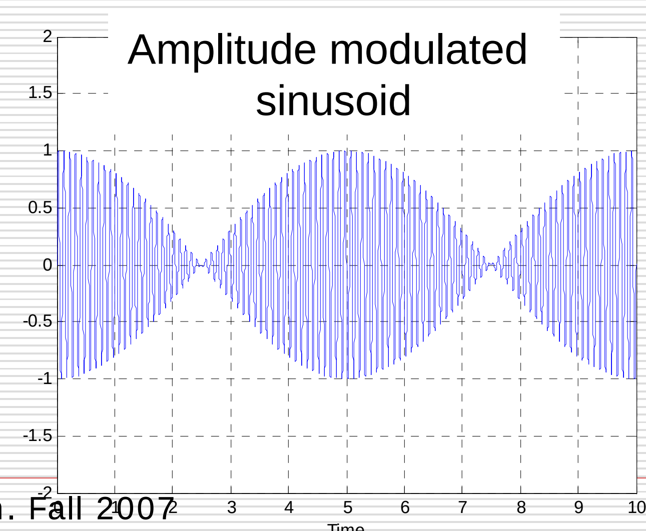


Baseband signal





Bandpass signal



Baseband Communications

- There are multiple baseband communications techniques that modulate a pulse stream using the message signal
  - Pulse Amplitude Modulation (PAM)
  - Pulse Width Modulation (PWM)
  - Pulse Position Modulation (PPM)
  - Pulse Code Modulation (PCM)

PAM, PWM, PPM: Pulse modulated by analog or digital signal

PCM: Requires conversion to digital signal prior to modulation

- However, PCM is the most common

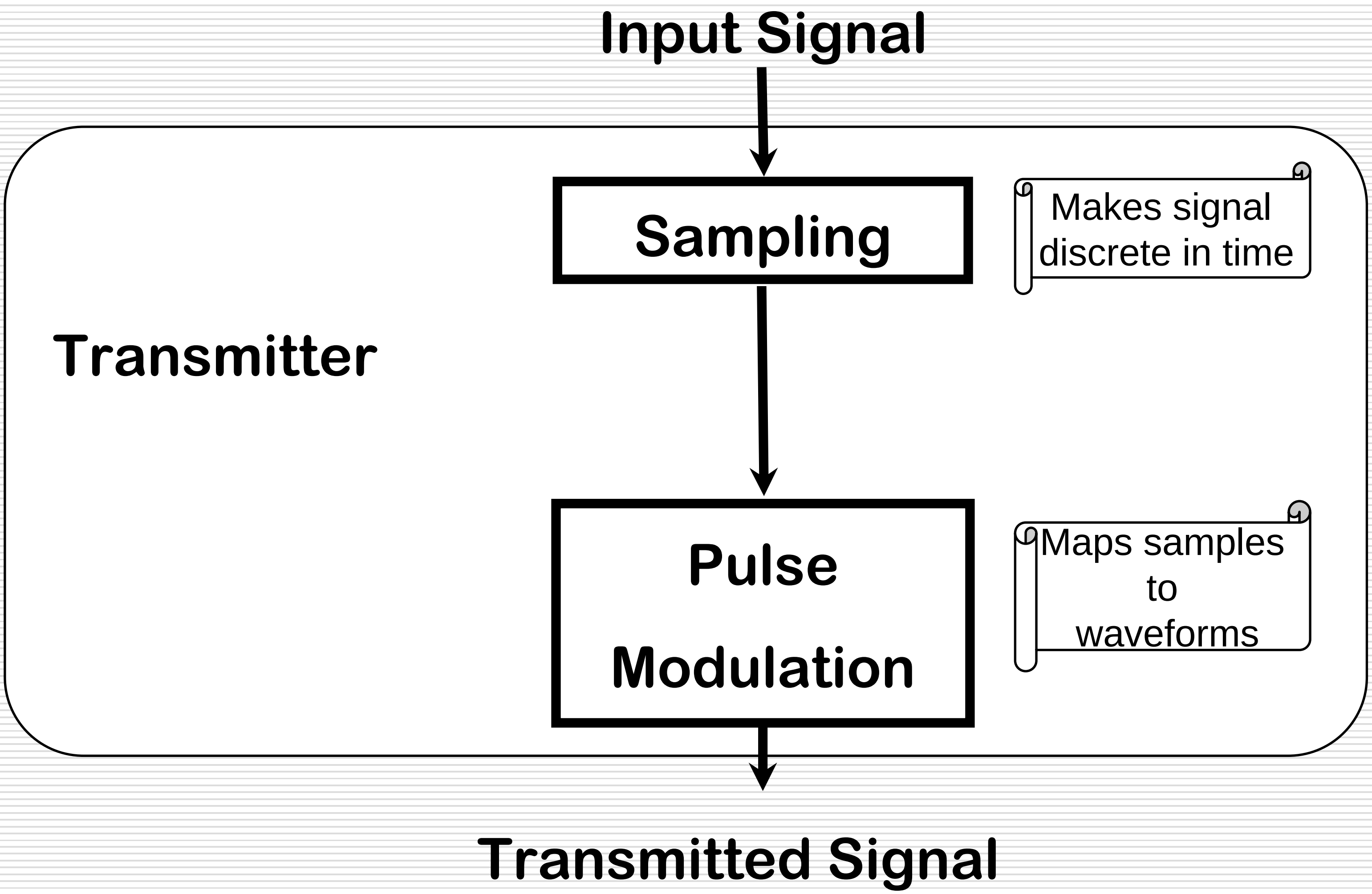
Analog Information

- Regardless of the type of system (analog or digital; bandpass or baseband) the original information source can be either analog or digital
- Traditional communication systems focused on the transfer of analog information
  - Examples: voice or video
- If the system uses continuous pulse modulation, the analog information signal must be sampled (made discrete in time)
- If the system is digital, the analog information signal must be sampled and quantized (made discrete in time and amplitude)

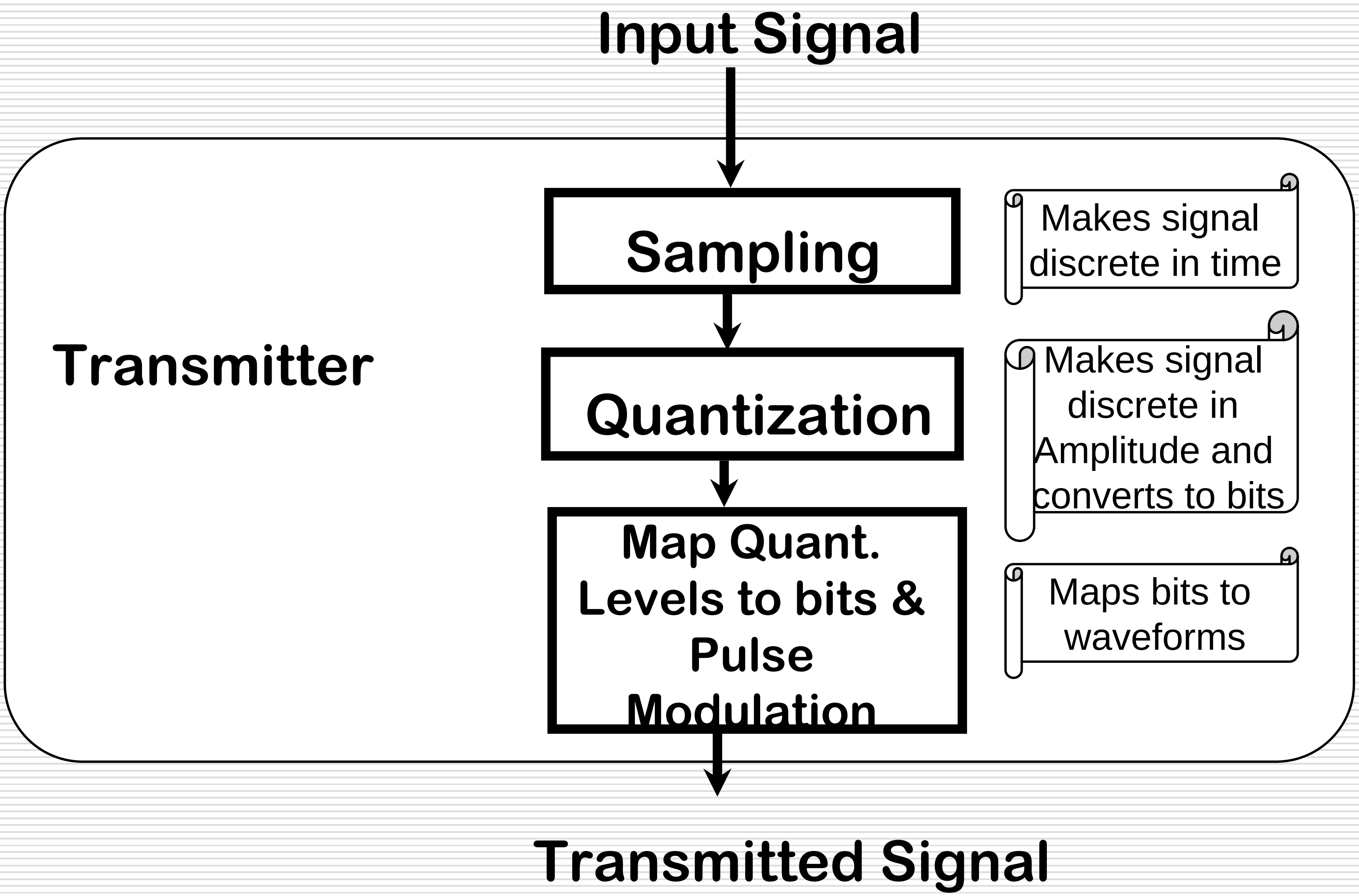
Digital Information

- With the rise of the internet, very often the 'source' of information is simply a computer which inherently uses digital information
- Such digital information fits naturally with a digital communication system
- No analog-to-digital conversion is necessary
- There may be conversion from binary to *M*-ary information within the digital communication system

Basic Structure of PAM / PPM / PWM



Basic Structure of PCM

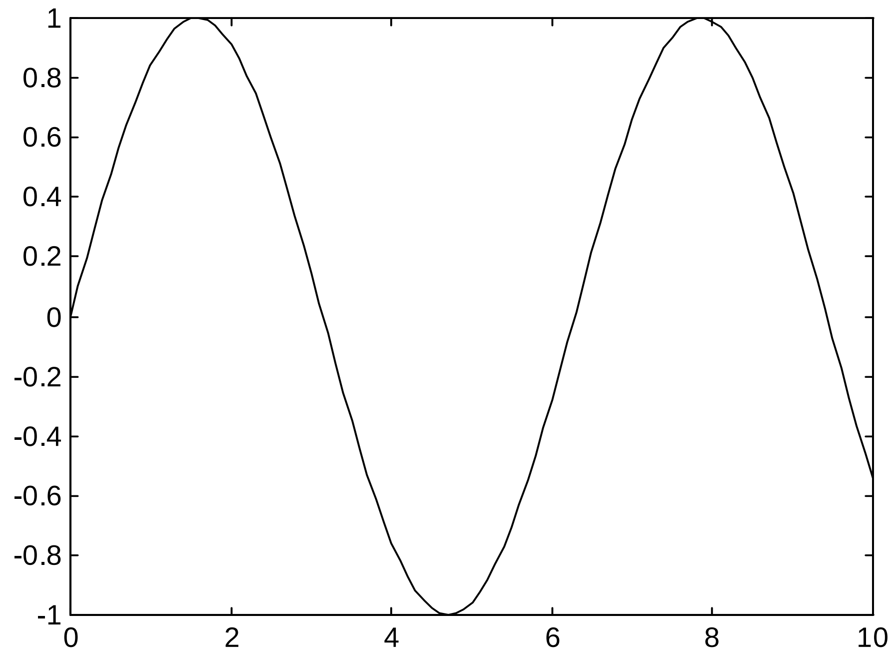


PAM / PWM / PPM vs PCM

- PAM/PWM/PPM are systems where the information signal is typically discrete in time but not necessarily in amplitude (thus not truly digital)
  - Infinite number of waveforms can be sent
  - Useful for time multiplexing multiple signals
  - Noise readily degrades information
  - Not particularly common
  - PAM is the first step in PCM, thus is useful for study of PCM
- PCM are systems where the information must be discrete in time and amplitude
  - Finite number of waveforms can be sent (i.e., digital)
  - Requires both sampling and quantization
  - Can be made more robust to noise
- Both require sampling, thus we study sampling first

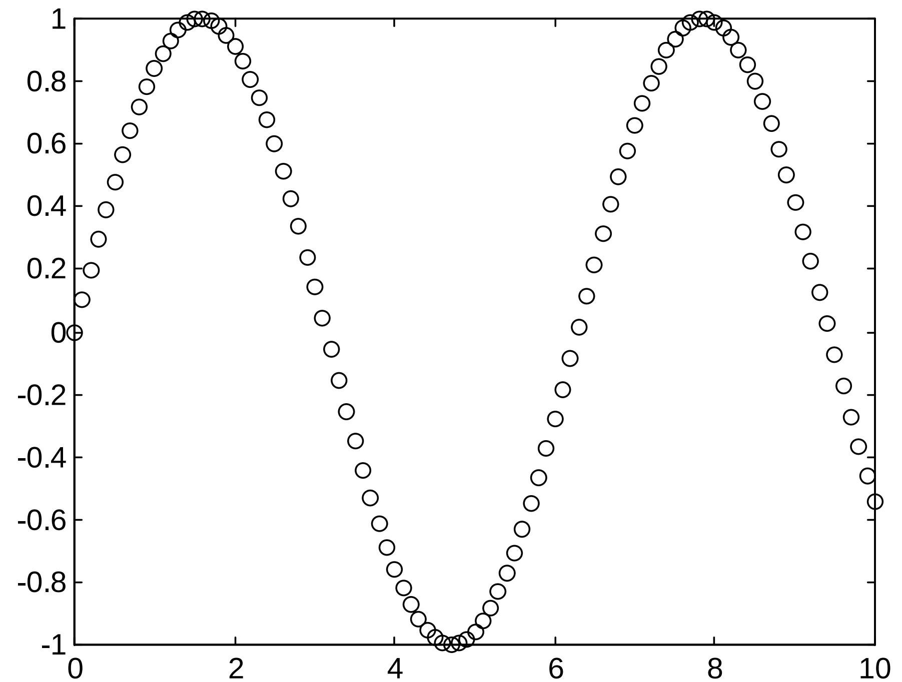
Digital Representation of Analog Signals

- Analog signals (e.g. voice,video) are continuous in time and amplitude:



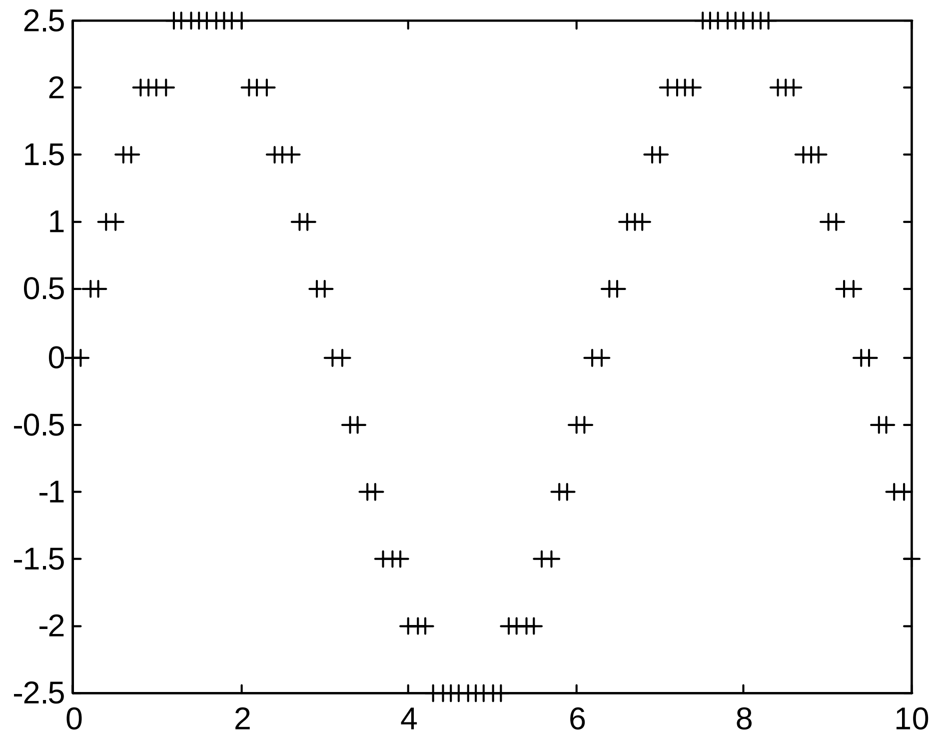
Digital Representation of Analog Signals

- <u>Sampling</u> analog signals makes them discrete in *time*:



Digital Representation of Analog Signals

- Quantization of sampled analog signals makes the samples discrete in amplitude:



- **The number of discrete amplitude levels is directly related to the number of bits we are willing to use to represent each sample. Thus, we trade-off bit rate and fidelity**

Digital Representation of Analog Signals

- If done properly, sampling introduces no distortion into the signal
- Quantization does introduce distortion
  - There is a tradeoff between distortion and bandwidth requirements
  - More bits per sample →less distortion
  - Fewer bits per sample → lower bandwidth requirements
- We will consider sampling in the next lecture.
- We will discuss quantization next class.

The Sampling Theorem

- We consider instantaneous sampling of a signal waveform ("ideal sampling" or "impulse sampling") which can be modeled as

$$w_s(t) = \underbrace{w(t)}_{\text{original signal}} \underbrace{\sum_{n=-\infty}^{\infty} \delta(t - nT_s)}_{\text{impulse train}}$$

$$= \sum_{n=-\infty}^{\infty} w(nT_s)\,\delta(t - nT_s)$$

- The train of impulse functions select sample values at regular intervals.
- How often do we have to sample to retrieve the original information? (*i.e.*, how small must T_s be?)

The Sampling Theorem (cont.)

$$w_s(t) = w(t)\sum_{n=-\infty}^{\infty}\delta(t - nT_s) = \sum_{n=-\infty}^{\infty} w(nT_s)\delta(t - nT_s)$$

- The train of impulse functions select sample values at regular intervals. Using a Fourier Series representation of the impulse train:

$$\sum_{n=-\infty}^{\infty}\delta(t - nT_s) = \frac{1}{T_s}\sum_{n=-\infty}^{\infty} e^{jn\omega_s t}, \omega_s = \frac{2\pi}{T_s}$$

- Rewriting, we have:

$$w_s(t) = w(t)\sum_{n=-\infty}^{\infty}\frac{1}{T_s}e^{jn\omega_s t}$$

The Sampling Theorem (cont.)

- Taking the Fourier Transform:

$$W_s(f) = \frac{1}{T_s} W(f) * F\left\{\sum_{n=-\infty}^{\infty} e^{jn\omega_s t}\right\}$$

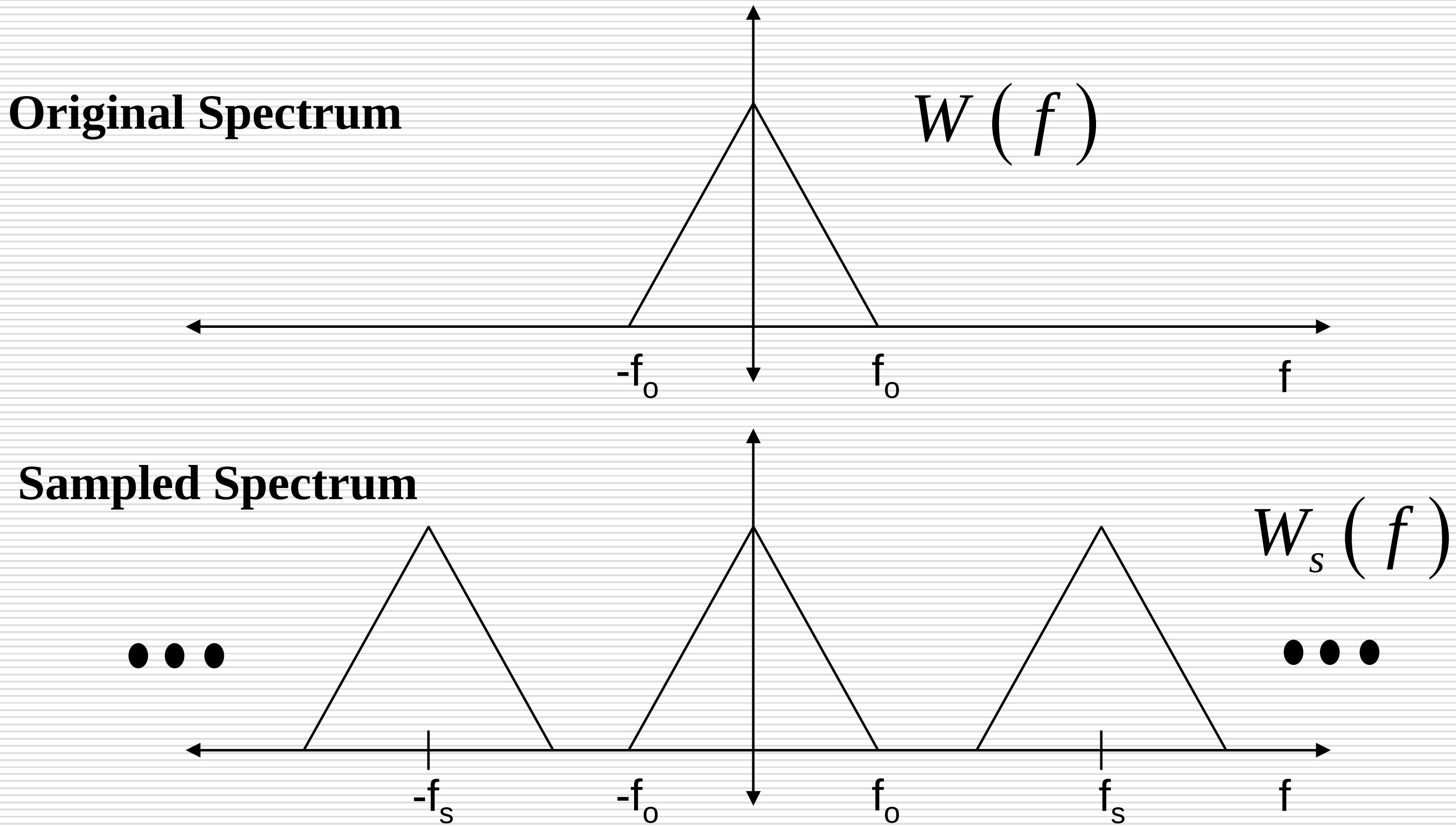
$$= \frac{1}{T_s} W(f) * \sum_{n=-\infty}^{\infty} F\left\{e^{jn\omega_s t}\right\}$$

$$W_S(f) = \frac{1}{T_S} W(f) * \sum_{n=-\infty}^{\infty} \delta(f - nf_S), \; f_S = \frac{\omega_S}{2\pi}$$

$$W_s(f) = \frac{1}{T_s} \sum_{n=-\infty}^{\infty} W(f - nf_s)$$

Note: This also follows from the fact that the Fourier Transform of an impulse train is simply an impulse train.

Sampling Theorem



Sampling Theorem

- Let *w(t)* be a band-limited signal with Fourier Transform:

$$W(f) = 0, \text{ for } |f| > B$$

- *X(t)* can be perfectly reconstructed from uniformly spaced samples, provided those samples are taken at a rate $f_s \geq 2B$
  - *2B* is called the Nyquist Rate
  - If $f_s < 2B$, aliasing results.
  - If signal is not strictly bandlimited, then it must be passed through lowpass filter before sampling to practically limit its bandwidth

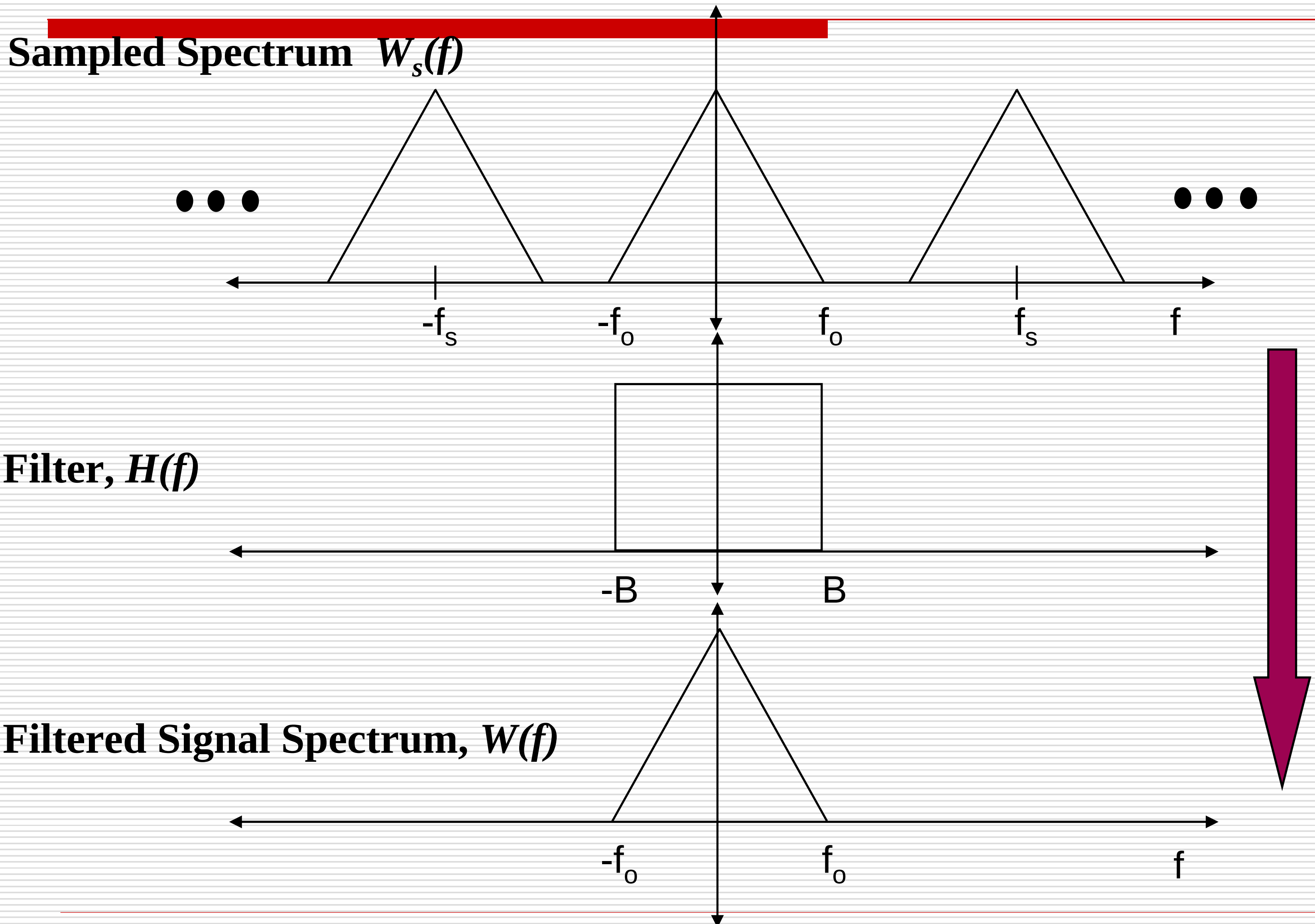
Recovering the Signal from Sampled Waveform

Sampled signal: $$W_S(f) = \frac{1}{T_S}\sum_{n=-\infty}^{\infty} W(f - nf_S)$$

- Apply lowpass filter to recover original signal

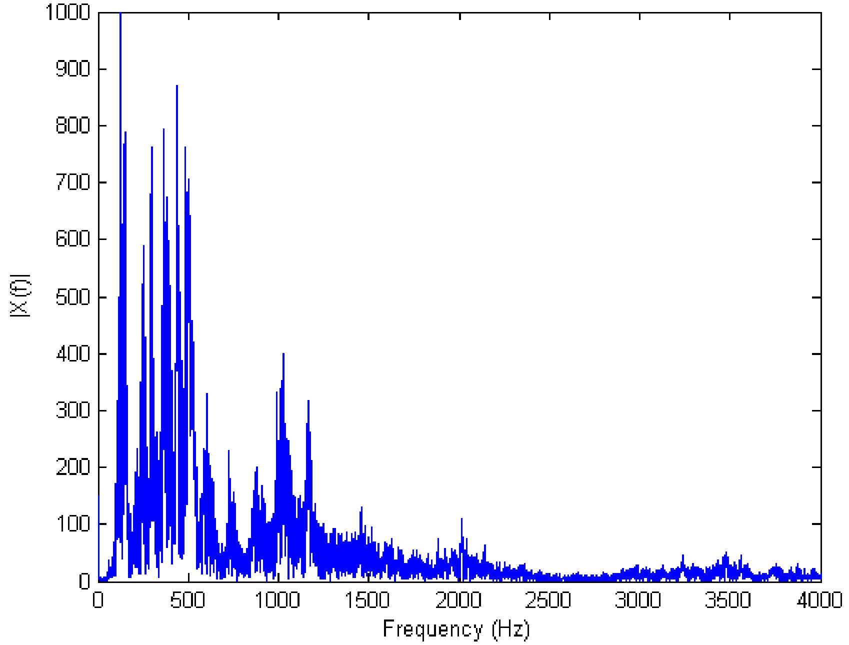
$$\begin{aligned} W(f) &= W_s(f)\Pi\left(\frac{f}{2B}\right) \\ &= \left(\frac{1}{T_s}\sum_{n=-\infty}^{\infty} W(f - nf_s)\right)\Pi\left(\frac{f}{2B}\right) \\ &= \frac{1}{T_s}W(f) \end{aligned}$$

Recovering the Original Signal

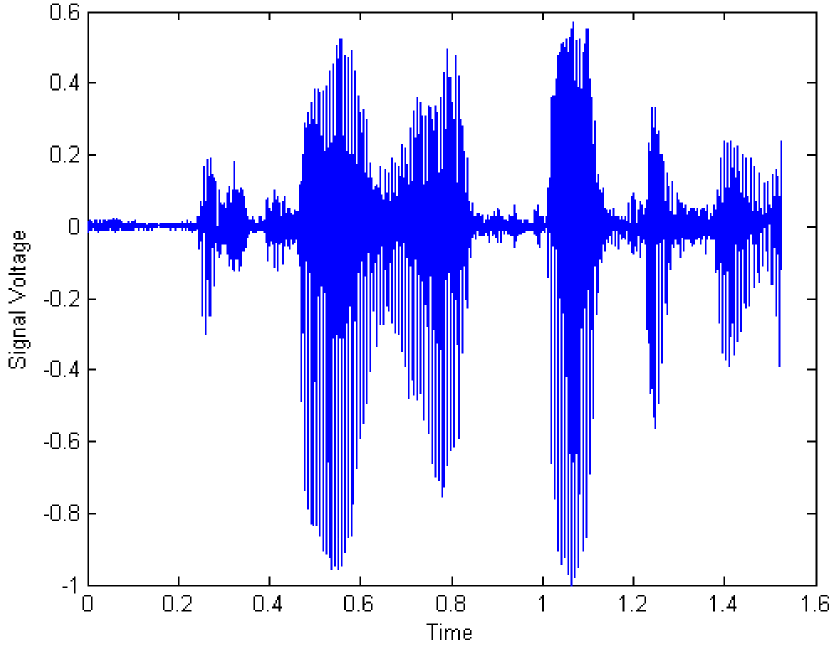


Example

Original Spectrum



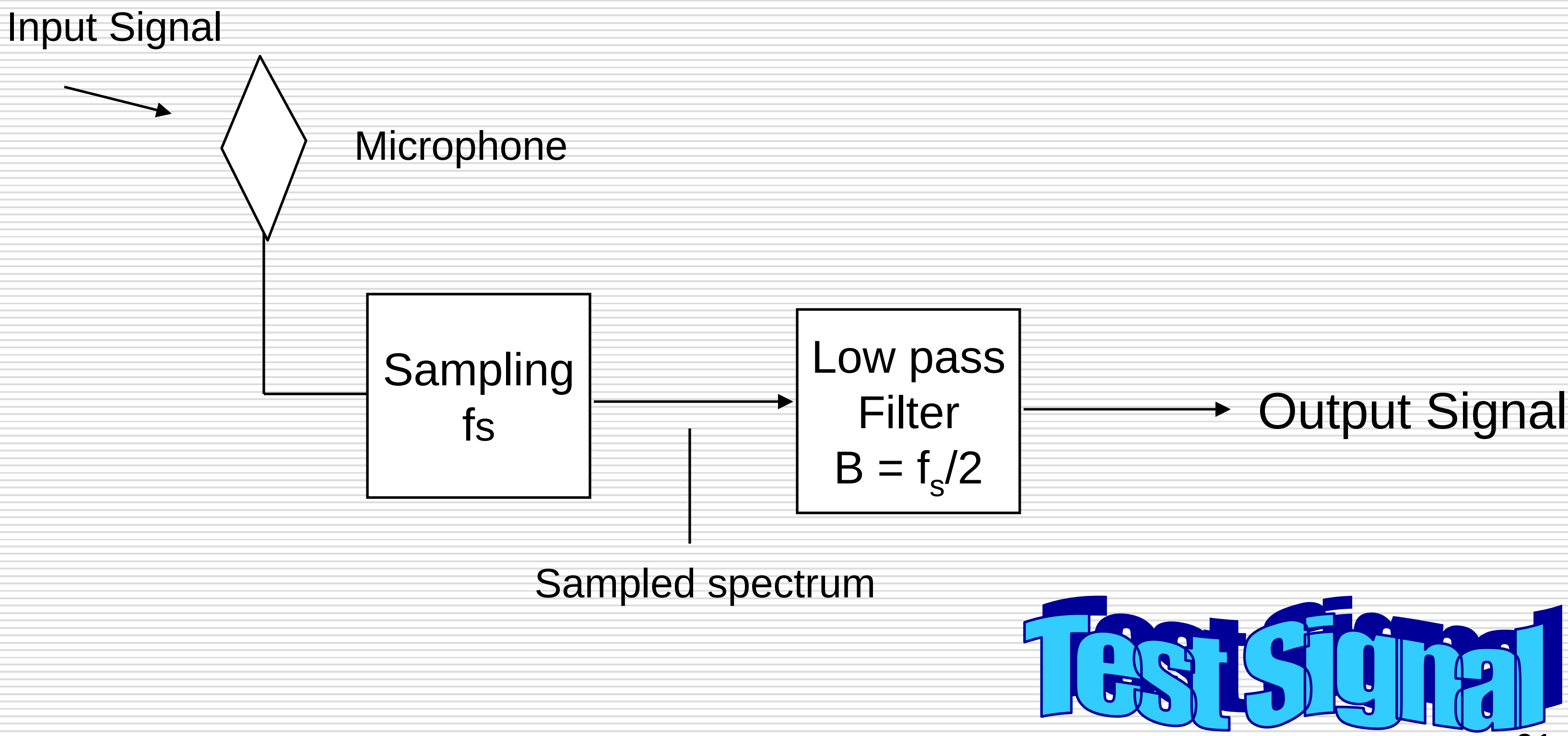
Time Signal



SamplingOriginal.wav

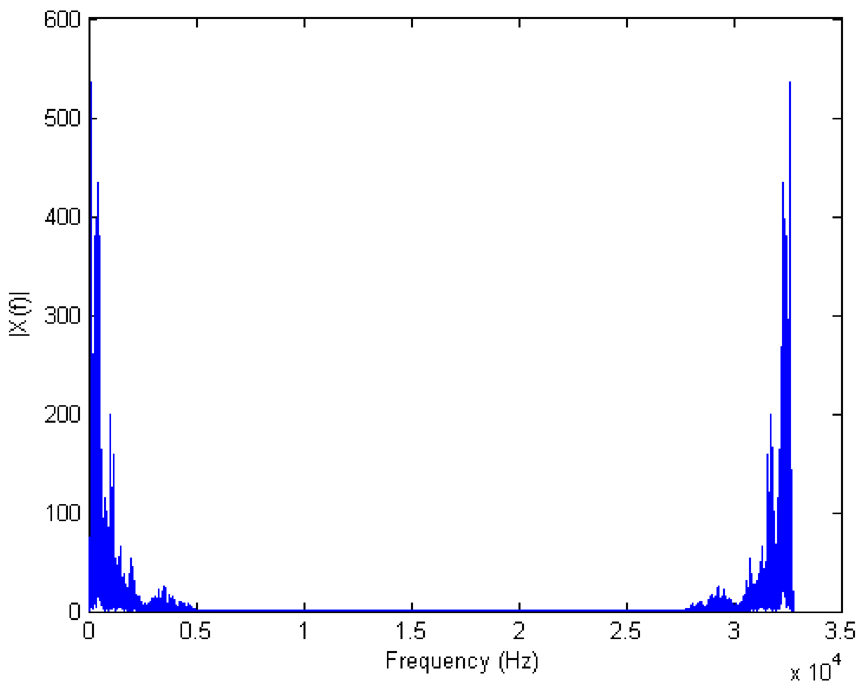
System

- Simple sampling and reconstruction

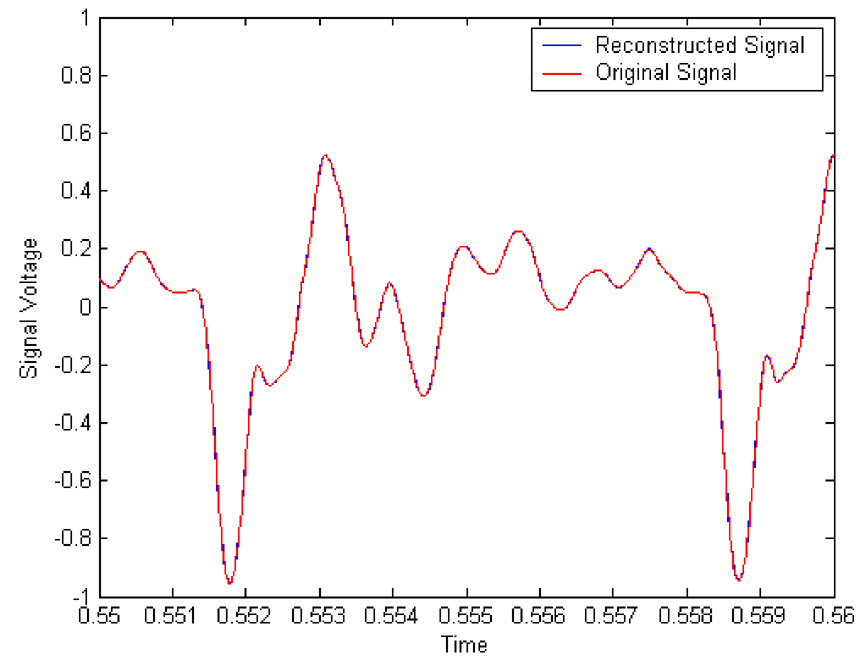


f_s = 32kHz

No Aliasing



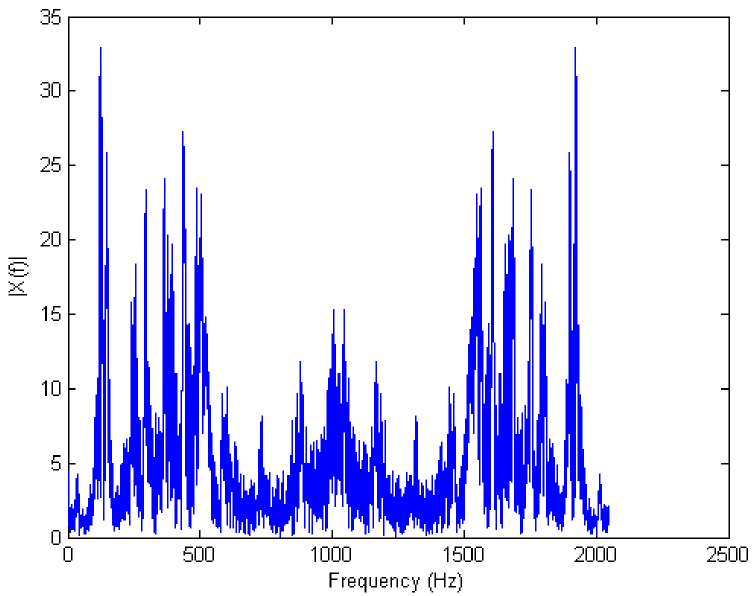
Perfect



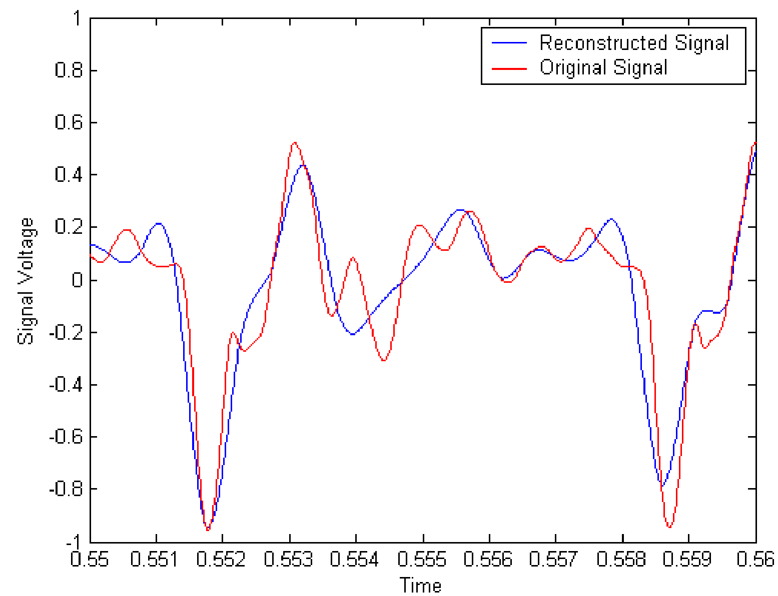
Sound Quality

f_s = 2kHz

Substantial Aliasing



Imperfect



Sound Quality

Practical Sampling Rates

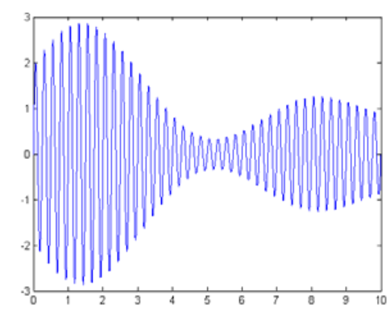
- Speech:
  - Telephone quality speech has a bandwidth of 4 kHz
  - Most digital telephone systems sample at 8000 samples/sec
- Audio:
  - The highest frequency the human ear can hear is approximately 15 kHz
  - CDs sample at rate 44,000 samples/sec
- Video:
  - The human eye requires samples at a rate of at least 20 frames/sec to achieve smooth motion

Summary

- Today we have introduced the concept of digital communications
- Digital communication systems are capable of transmitting analog information signals but require analog-to-digital conversion
  - Sampling
  - Quantization
  - Waveform mapping
- We focused on sampling first. Next class we will discuss quantization and waveform mapping.

Appendix

Alternate view of the Sampling Theorem



Another View of the Sampling Theorem

$$W(f) = W_s(f)\Pi\left(\frac{f}{2B}\right)$$

$$w(t) = w_s(t) * \Im^{-1}\left\{\Pi\left(\frac{f}{2B}\right)\right\}$$

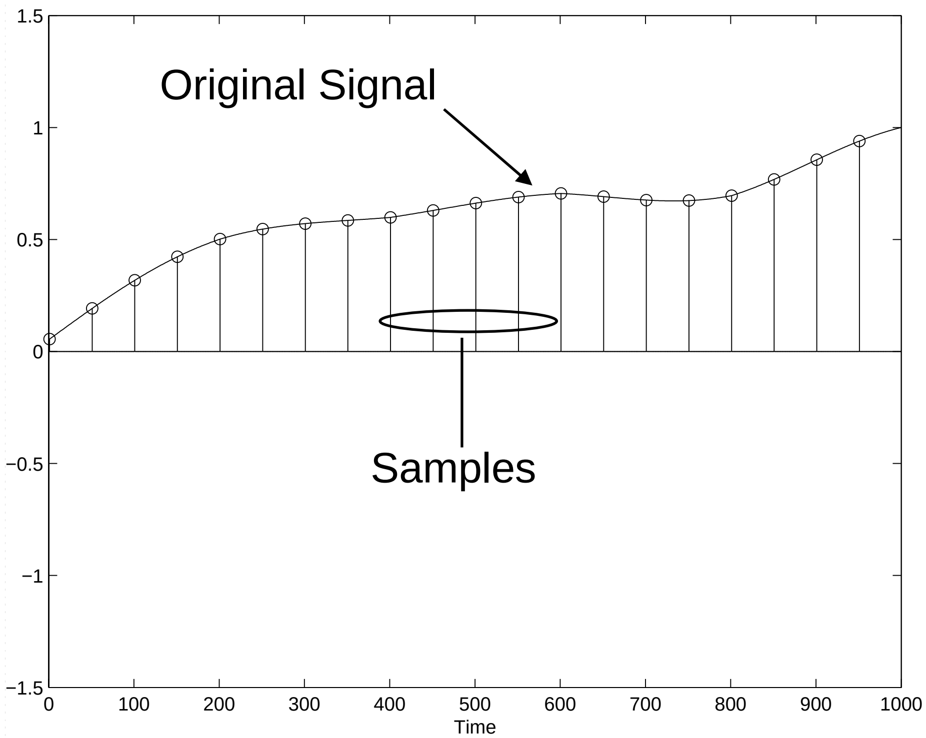
$$= w_s(t) * \mathrm{sinc}(2\mathrm{Bt})$$

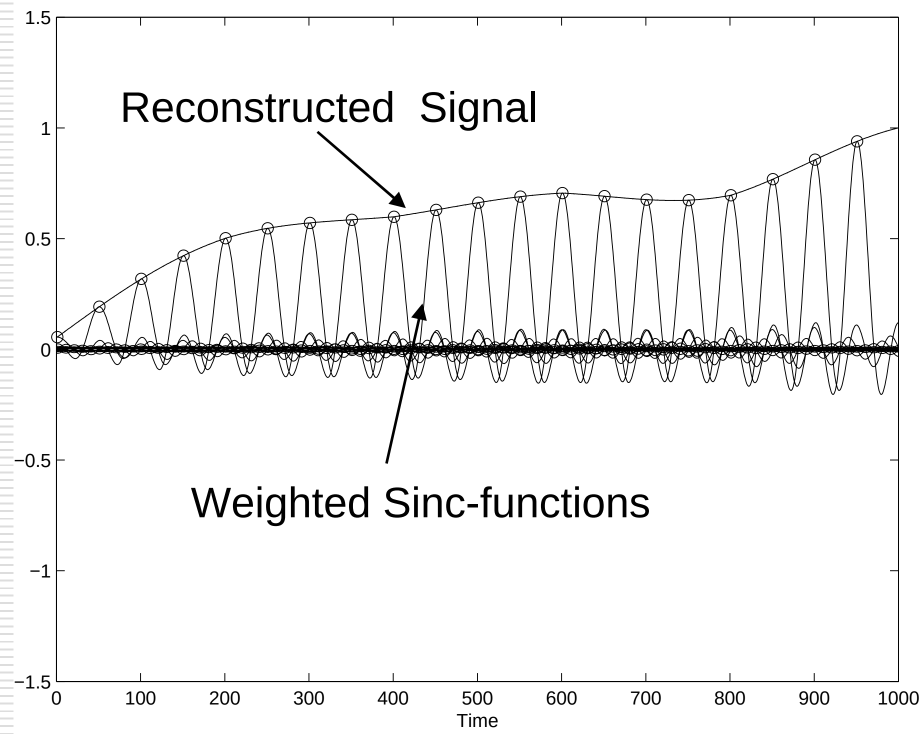
$$= \left(\sum_{n=-\infty}^{\infty} w(nT_s)\delta(t - nT_s)\right) * \mathrm{sinc}(2\mathrm{Bt})$$

$$= \sum_{n=-\infty}^{\infty} w(nT_s)\mathrm{sinc}(2Bt - n2BT_s)$$

$$= \sum_{n=-\infty}^{\infty} w(nT_s)\mathrm{sinc}\left(\frac{t}{T_s} - n\right)$$

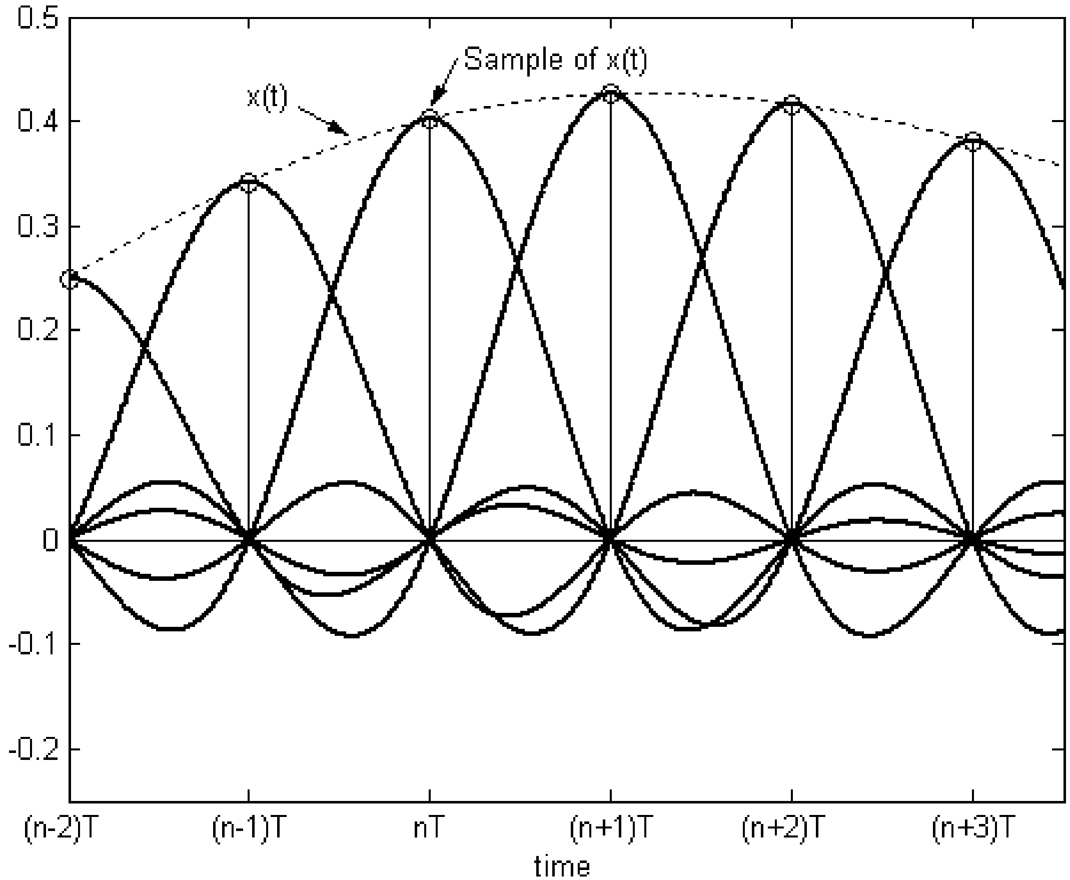
Time-Domain View of the Sampling Theorem





sin(x)/x is also referred to as the *Sampling Function*

Ideal Reconstruction



Sinc functions provide ideal reconstruction of values between samples