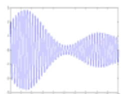
# ECE3614
# Introduction to Communications Systems
# Fall 2007

Instructor: Dr. R. Michael Buehrer

Lecture #21: Introduction to Digital Communications





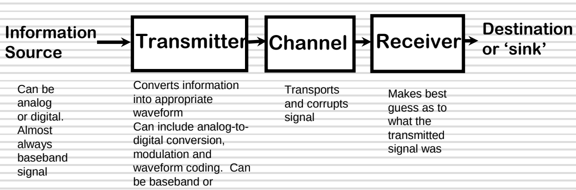
## Overview

- 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



## 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)



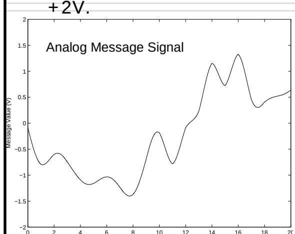
## Analog vs. Digital

- 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*.

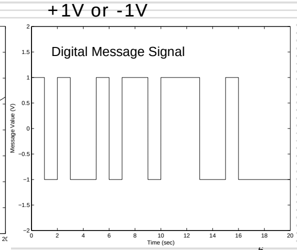


## Example

- 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



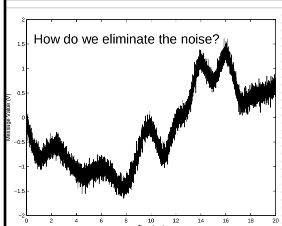
# Why digital communications?

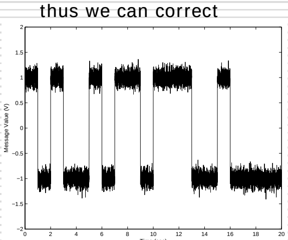
- Any noise introduces distortion to an analog signal. Since a digital receiver need only distinguish between a finite number of waveforms it is possible to recover digital information without corruption a large percentage time.
- Many signal processing techniques are available to improve system performance: 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



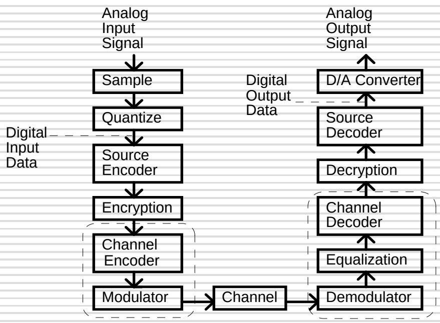



## Limitation of Digital

- 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

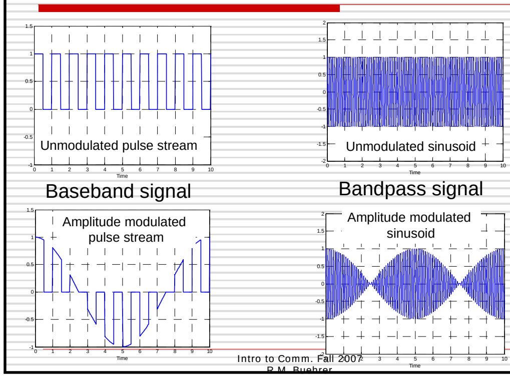


## Block Diagram of Digital Communications System






## Modulation



## Baseband 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)

Pulse modulated by analog or digital signal (PAM, PWM, PPM)

Requires conversion to digital signal prior to modulation (PCM)

- However, PCM is the most common



## Analog Information

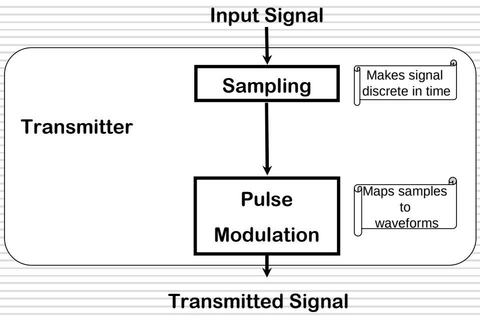
- Regardless of the type of system (analog or digital; bandpass or baseband) the original information source can be either analog or digital
- Traditional communication systems focused on the transfer of analog information
  - Examples: voice or video
- If the system uses continuous pulse modulation, 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

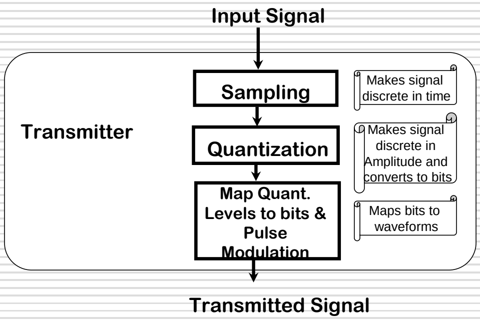
- With the rise of the internet, very often the ‘source’ of information is simply a computer which inherently uses digital information
- Such digital information fits naturally with a digital communication system
- No analog-to-digital conversion is necessary
- There may be conversion from binary to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

## Basic Structure of PAM / PPM / PWM






## Basic Structure of PCM



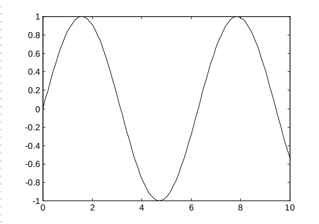



## PAM / PWM / PPM vs PCM

- 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

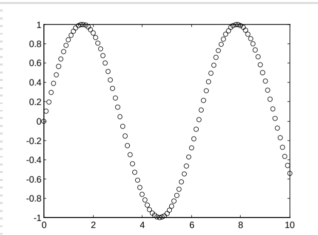
- Analog signals (e.g. voice,video) are continuous in time and amplitude:






## Digital Representation of Analog Signals

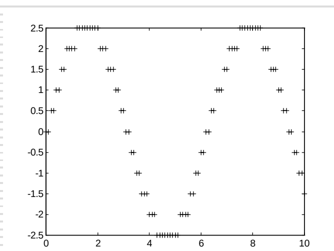
- Sampling analog signals makes them discrete in *time*:






## 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:

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

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

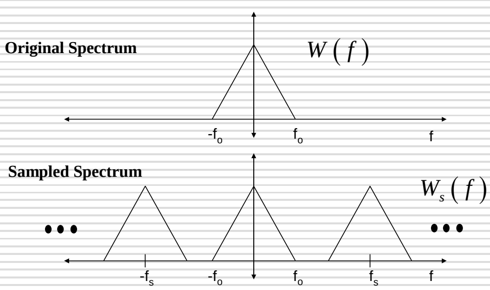
$$W_s(f) = \frac{1}{T_s} W(f) * \sum_{n=-\infty}^{\infty} \delta(f - nf_s), \quad f_s = \frac{\omega_s}{2\pi}$$

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

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

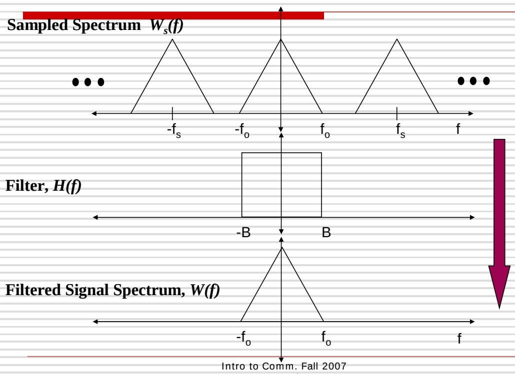
- Apply lowpass filter to recover original signal

$W_S(f)$ → **Ideal Lowpass Filter w/bandwidth B** → $W(f)$

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



## Recovering the Original Signal

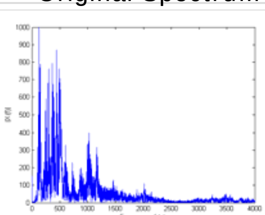


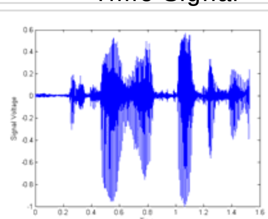



## Example

Original Spectrum

Time Signal

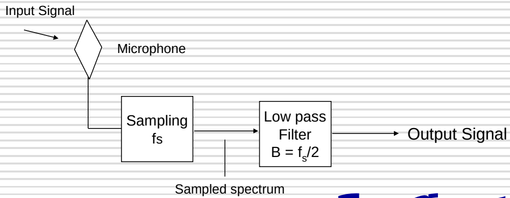



SamplingOriginal.wav

## System

- Simple sampling and reconstruction

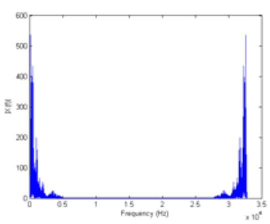



Test Signal

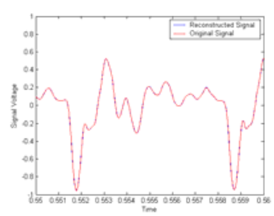


## $f_s$ = 32kHz

No Aliasing




Perfect

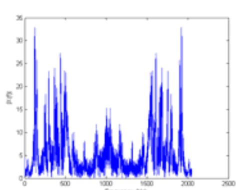



Sound Quality

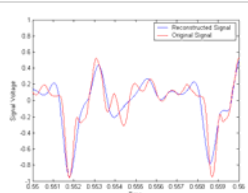


## $f_s$ = 2kHz

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



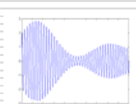
## Summary

- 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

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

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

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

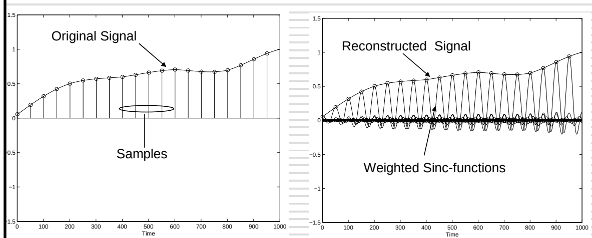
$$= \left(\sum_{n=-\infty}^{\infty} w(nT_s)\delta(t-nT_s)\right) * \text{sinc}(2\text{Bt})$$

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

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



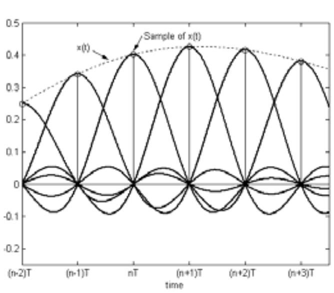
## Time-Domain View of the Sampling Theorem




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



## Ideal Reconstruction




Sinc functions provide ideal reconstruction of values between samples