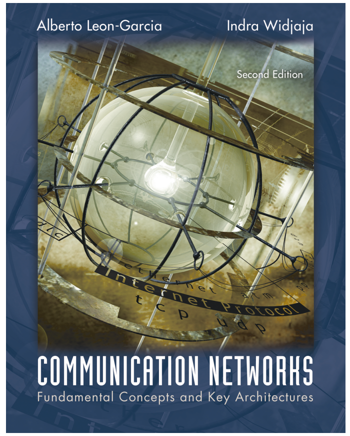
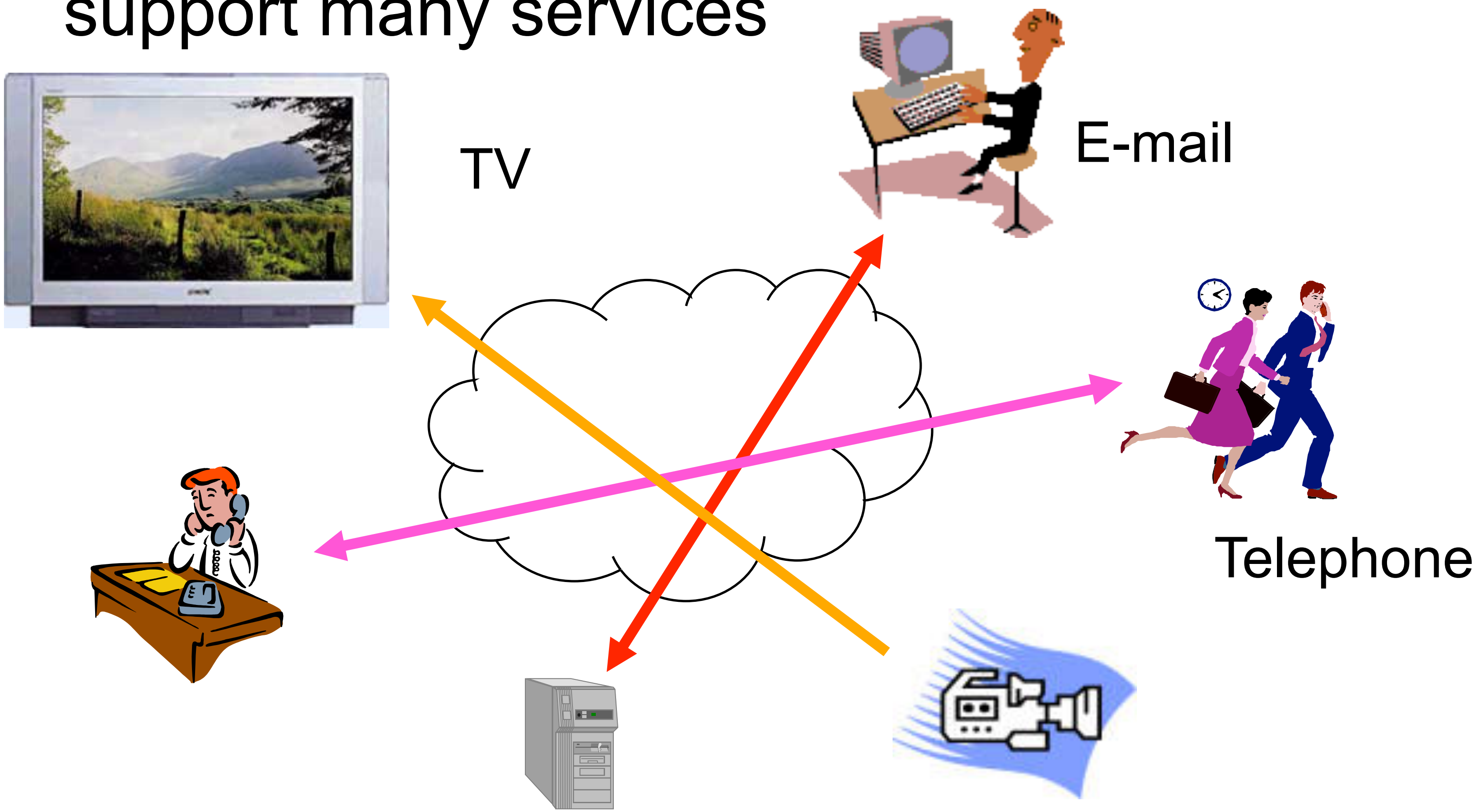
Chapter 3
Digital Transmission Fundamentals



Digital Representation of Information
Why Digital Communications?
Digital Representation of Analog Signals
Characterization of Communication Channels
Fundamental Limits in Digital Transmission
Line Coding
Modems and Digital Modulation
Properties of Media and Digital Transmission Systems
Error Detection and Correction

Digital Networks

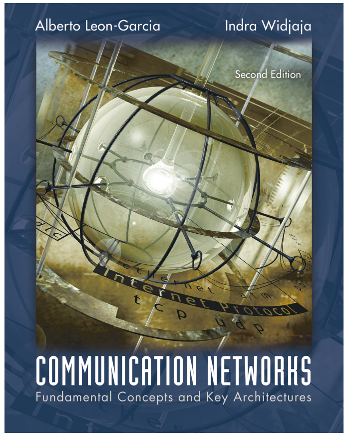
- Digital transmission enables networks to support many services



Questions of Interest

- How long will it take to transmit a message?
  - How many bits are in the message (text, image)?
  - How fast does the network/system transfer information?
- Can a network/system handle a voice (video) call?
  - How many bits/second does voice/video require? At what quality?
- How long will it take to transmit a message without errors?
  - How are errors introduced?
  - How are errors detected and corrected?
- What transmission speed is possible over radio, copper cables, fiber, infrared, …?

Chapter 3
Digital Transmission Fundamentals



Digital Representation of Information

Bits, numbers, information

- Bit: number with value 0 or 1
  - *n* bits: digital representation for 0, 1, … , 2^n
  - Byte or Octet, $n = 8$
  - Computer word, $n = 16$, 32, or 64
- *n* bits allows enumeration of 2^n possibilities
  - *n*-bit field in a header
  - *n*-bit representation of a voice sample
  - Message consisting of n bits
- *The number of bits required to represent a message is a measure of its information content*
  - More bits → More content

Block vs. Stream Information

Block

- Information that occurs in a single block
  - Text message
  - Data file
  - JPEG image
  - MPEG file
- Size = Bits / block

  or bytes/block
  - 1 kbyte = 2^{10} bytes
  - 1 Mbyte = 2^{20} bytes
  - 1 Gbyte = 2^{30} bytes

Stream

- Information that is produced & transmitted *continuously*
  - Real-time voice
  - Streaming video
- Bit rate = bits / second
  - 1 kbps = 10^3 bps
  - 1 Mbps = 10^6 bps
  - 1 Gbps = 10^9 bps

Transmission Delay

- L — number of bits in message
- R bps — speed of digital transmission system
- L/R — time to transmit the information
- t_{prop} — time for signal to propagate across medium
- d — distance in meters
- c — speed of light ($3\text{x}10^8$ m/s in vacuum)

$$Delay = t_{prop} + L/R = d/c + L/R \text{ seconds}$$

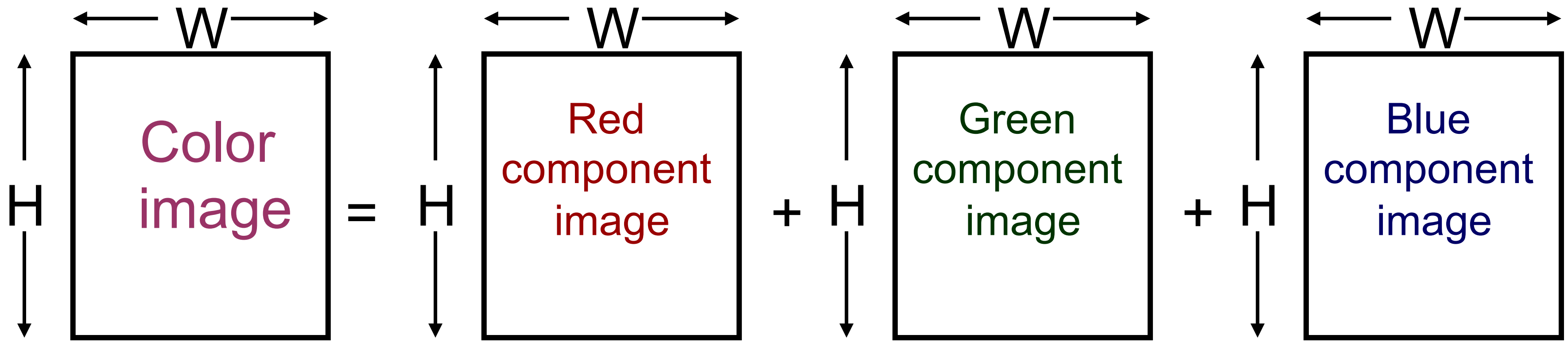
Use data compression to reduce L
Use higher speed modem to increase R
Place server closer to reduce d

Compression

- Information usually not represented efficiently
- Data compression algorithms
  - Represent the information using fewer bits
  - Noiseless: original information recovered exactly
    - E.g. `zip`, `compress`, GIF, fax
  - Noisy: recover information approximately
    - JPEG
    - Tradeoff: # bits vs. quality
- Compression Ratio

#bits (original file) / #bits (compressed file)

Color Image



Total bits = 3 $\times$ H $\times$ W pixels $\times$ B bits/pixel = 3HWB bits

Example: 8$\times$10 inch picture at 400 $\times$ 400 pixels per inch^2

400 $\times$ 400 $\times$ 8 $\times$ 10 = 12.8 million pixels

8 bits/pixel/color

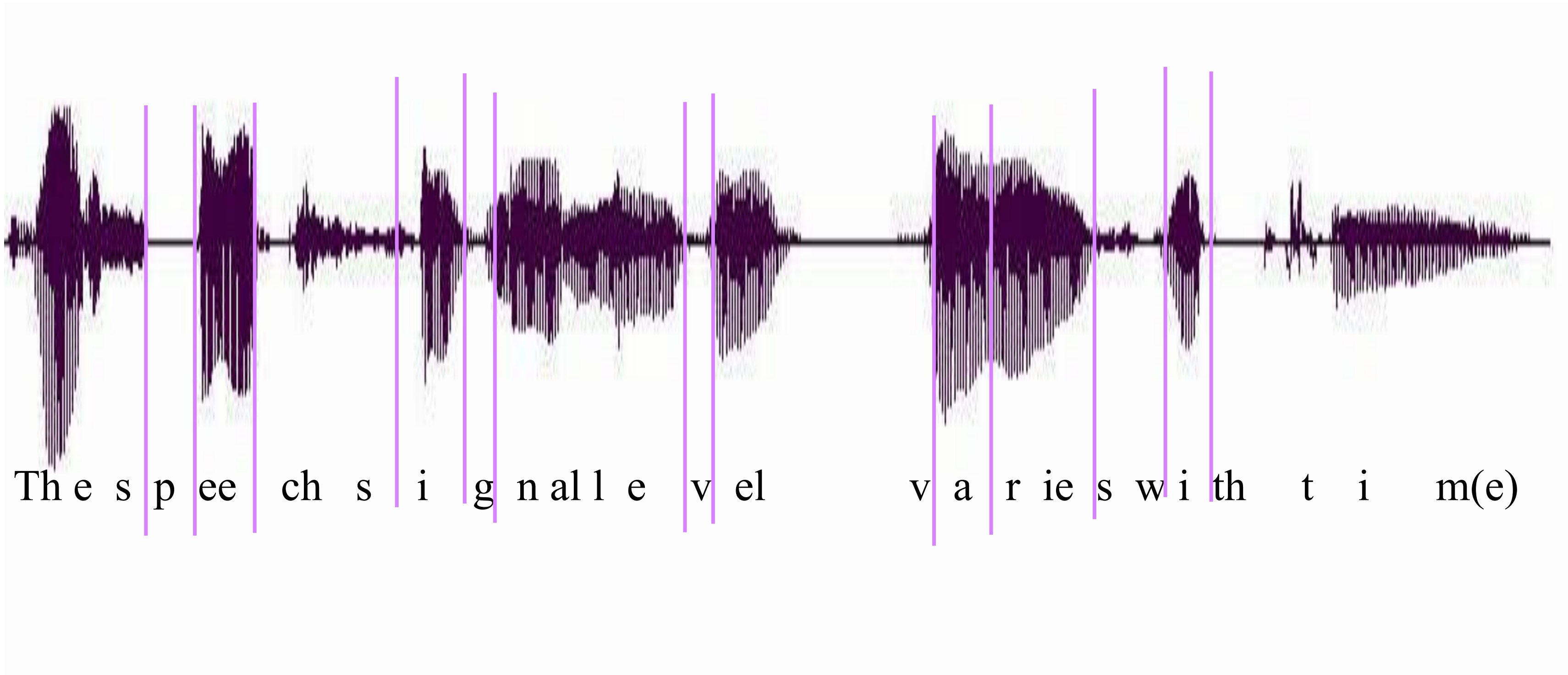
12.8 megapixels $\times$ 3 bytes/pixel = 38.4 megabytes

Examples of Block Information

Type	Method	Format	Original	Compressed (Ratio)
Text	Zip, compress	ASCII	Kbytes-Mbytes	(2-6)
Fax	CCITT Group 3	A4 page 200x100 pixels/ in^2	256 kbytes	5-54 kbytes (5-50)
Color Image	JPEG	8x10 in^2 photo 400^2 pixels/in^2	38.4 Mbytes	1-8 Mbytes (5-30)

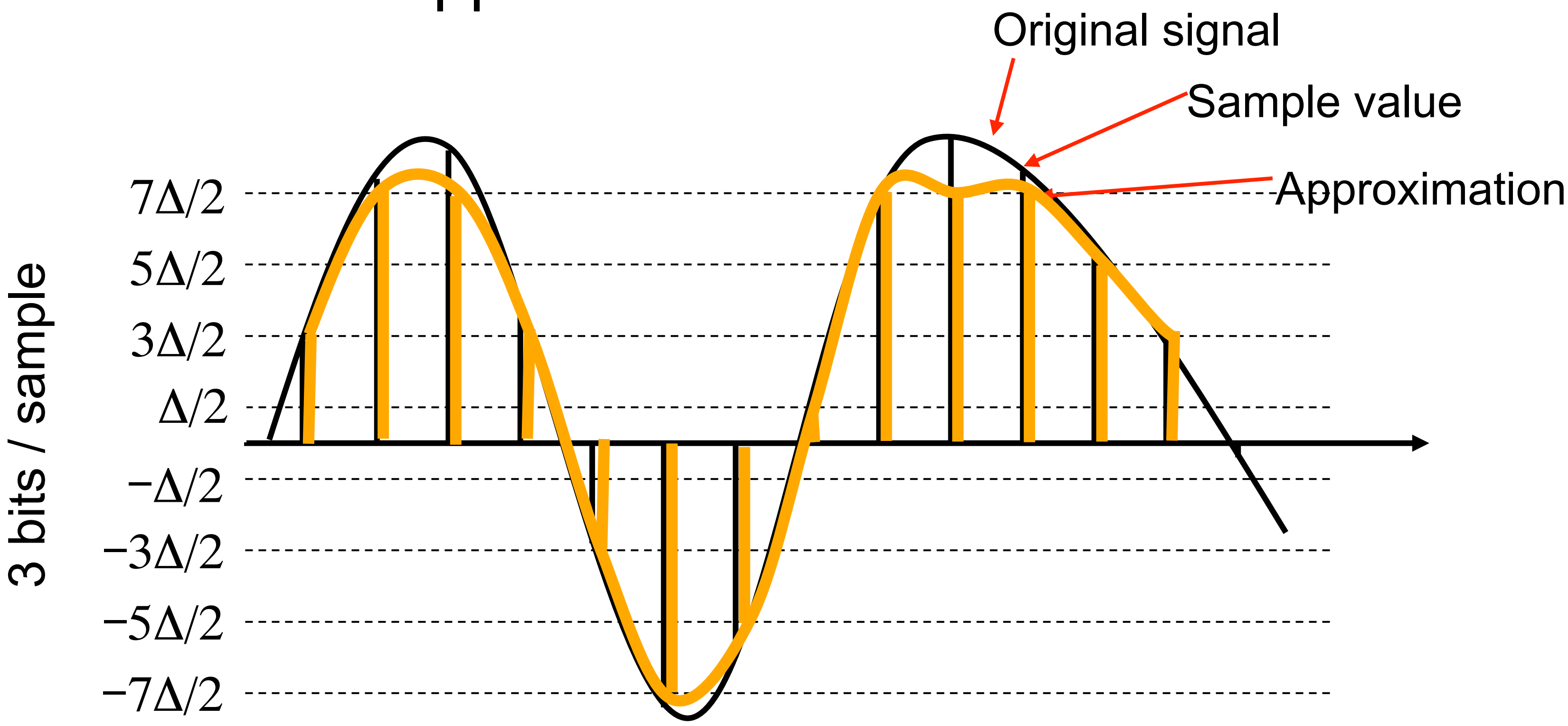
Stream Information

- A real-time voice signal must be digitized & transmitted as it is produced
- Analog signal level varies continuously in time



Digitization of Analog Signal

- Sample analog signal in time and amplitude
- Find closest approximation



R_s = Bit rate = # bits/sample x # samples/second

Bit Rate of Digitized Signal

- Bandwidth W_s Hertz: how fast the signal changes
  - Higher bandwidth → more frequent samples
  - Minimum sampling rate = 2 x W_s

- Representation accuracy: range of approximation error
  - Higher accuracy

  → smaller spacing between approximation values

  → more bits per sample

Example: Voice & Audio

Telephone voice

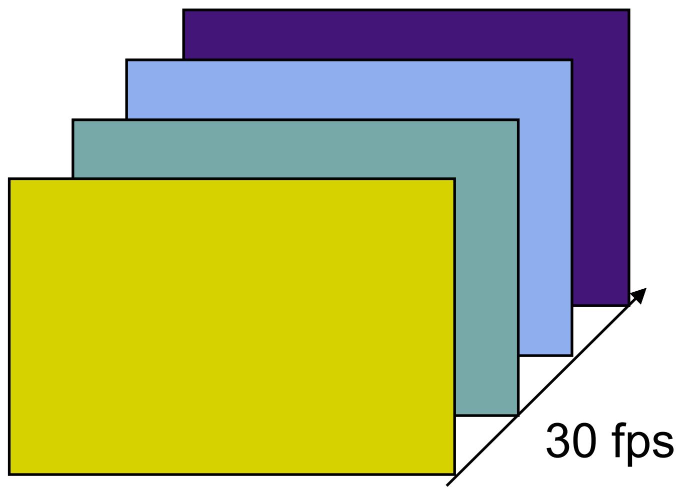
- W_s = 4 kHz → 8000 samples/sec
- 8 bits/sample
- R_s=8 x 8000 = 64 kbps
- Cellular phones use more powerful compression algorithms: 8-12 kbps

CD Audio

- W_s = 22 kHertz → 44000 samples/sec
- 16 bits/sample
- R_s=16 x 44000= 704 kbps per audio channel
- MP3 uses more powerful compression algorithms: 50 kbps per audio channel

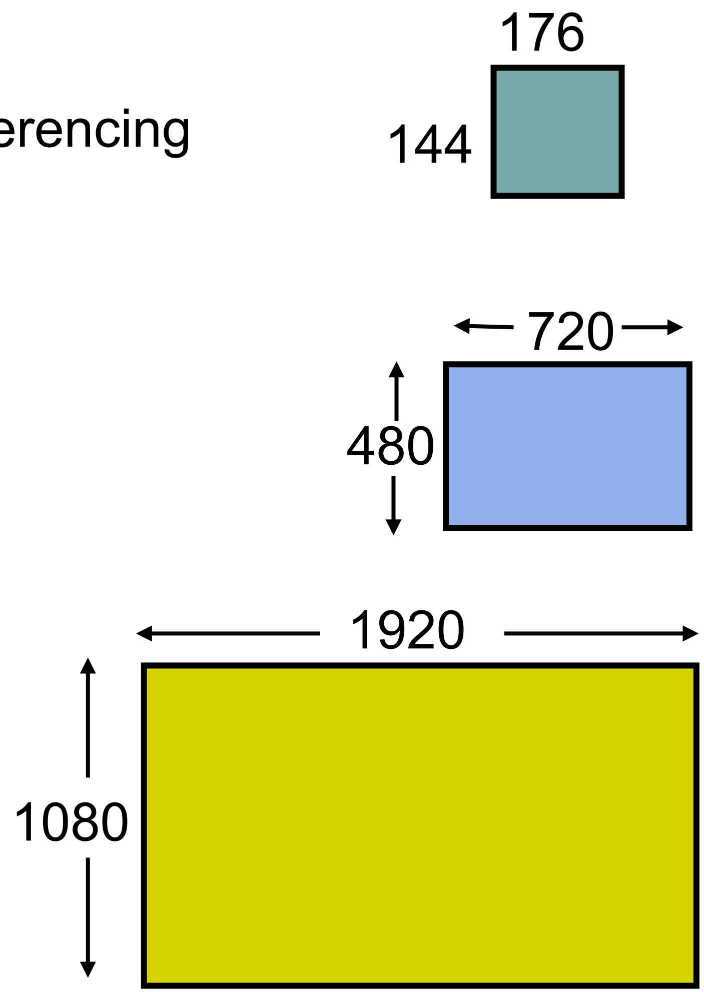
Video Signal

- Sequence of picture frames
  - Each picture digitized & compressed
- Frame repetition rate
  - 10-30-60 frames/second depending on quality
- Frame resolution
  - Small frames for videoconferencing
  - Standard frames for conventional broadcast TV
  - HDTV frames



Rate = M bits/pixel **x** (WxH) pixels/frame **x** F frames/second

Video Frames



QCIF videoconferencing

at 30 frames/sec =

760,000 pixels/sec

Broadcast TV

at 30 frames/sec =

10.4×10^6 pixels/sec

HDTV

at 30 frames/sec =

67×10^6 pixels/sec

Digital Video Signals

Type	Method	Format	Original	Compressed
Video Confer-ence	H.261	176x144 or 352x288 pix @10-30 fr/sec	2-36 Mbps	64-1544 kbps
Full Motion	MPEG 2	720x480 pix @30 fr/sec	249 Mbps	2-6 Mbps
HDTV	MPEG 2	1920x1080 @30 fr/sec	1.6 Gbps	19-38 Mbps

Transmission of Stream Information

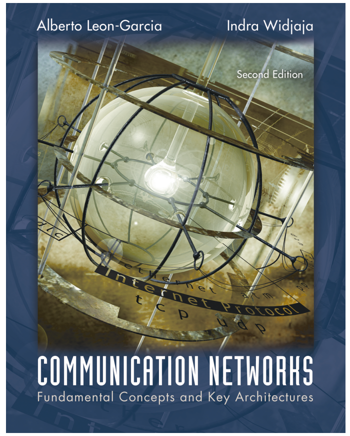
- Constant bit-rate
  - Signals such as digitized telephone voice produce a steady stream: e.g. 64 kbps
  - Network must support steady transfer of signal, e.g. 64 kbps circuit
- Variable bit-rate
  - Signals such as digitized video produce a stream that varies in bit rate, e.g. according to motion and detail in a scene
  - Network must support variable transfer rate of signal, e.g. packet switching or rate-smoothing with constant bit-rate circuit

Stream Service Quality Issues

Network Transmission Impairments

- Delay: Is information delivered in timely fashion?
- Jitter: Is information delivered in sufficiently smooth fashion?
- Loss: Is information delivered without loss? If loss occurs, is delivered signal quality acceptable?
- Applications & application layer protocols developed to deal with these impairments

Chapter 3
Communication Networks and Services



Why Digital Communications?

A Transmission System

Transmitter

- Converts information into *signal* suitable for transmission
- Injects energy into communications medium or channel
  - Telephone converts voice into electric current
  - Modem converts bits into tones

Receiver

- Receives energy from medium
- Converts received signal into form suitable for delivery to user
  - Telephone converts current into voice
  - Modem converts tones into bits

Transmission Impairments

Communication Channel

- Pair of copper wires
- Coaxial cable
- Radio
- Light in optical fiber
- Light in air
- Infrared

Transmission Impairments

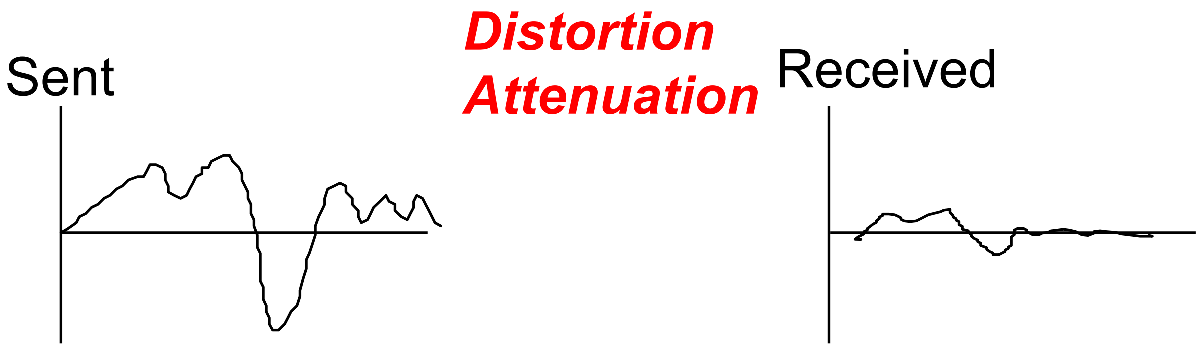
- Signal attenuation
- Signal distortion
- Spurious noise
- Interference from other signals

Analog Long-Distance Communications

- Each repeater attempts to restore analog signal to its original form
- Restoration is imperfect
  - Distortion is not completely eliminated
  - Noise & interference is only partially removed
- Signal quality decreases with # of repeaters
- Communications is distance-limited
- Still used in analog cable TV systems
- Analogy: Copy a song using a cassette recorder

Analog vs. Digital Transmission

Analog transmission: all details must be reproduced accurately



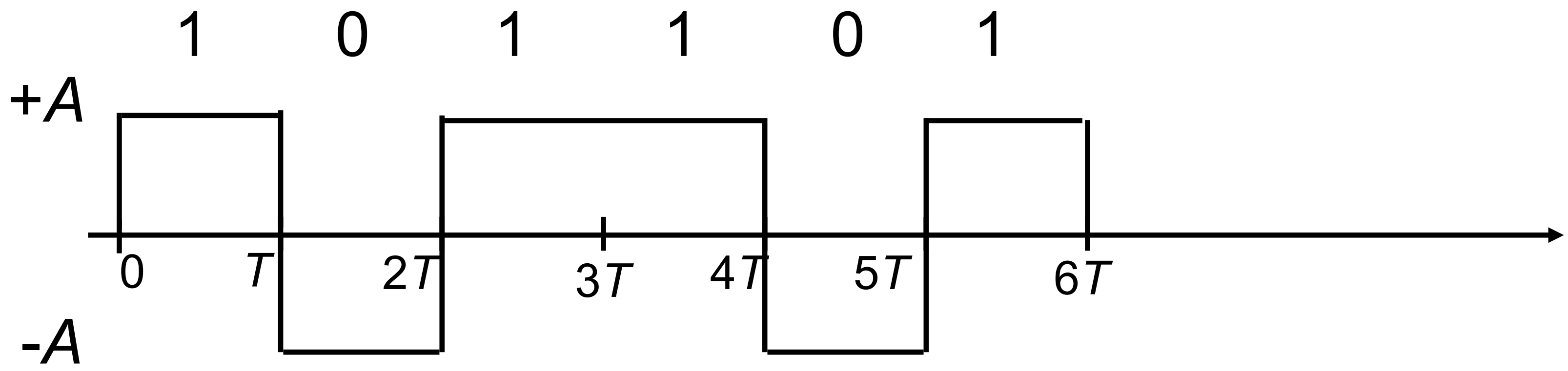
Digital transmission: only discrete levels need to be reproduced

Simple Receiver:
Was original pulse
positive or
negative?

Digital Long-Distance Communications

- Regenerator recovers original data sequence and retransmits on next segment
- Can design so error probability is very small
- Then each regeneration is like the first time!
- Analogy: copy an MP3 file
- Communications is possible over very long distances
- Digital systems vs. analog systems
  - Less power, longer distances, lower system cost
  - Monitoring, multiplexing, coding, encryption, protocols…

Digital Binary Signal



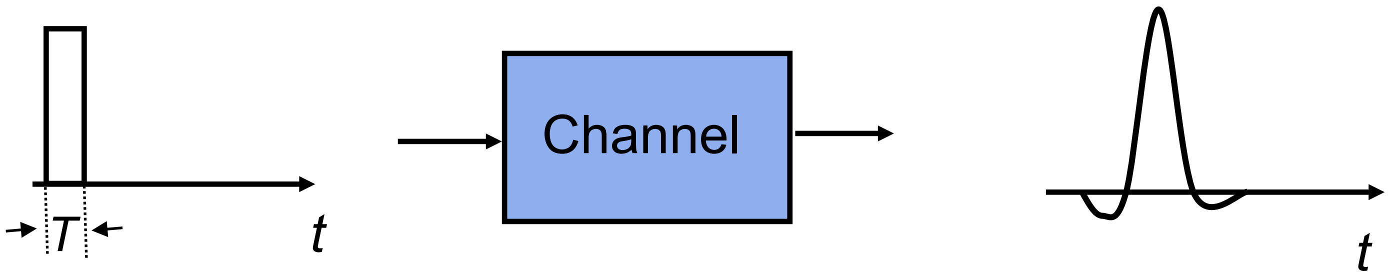
Bit rate = 1 bit / T seconds

For a given communications medium:

- How do we increase transmission speed?
- How do we achieve reliable communications?
- Are there limits to speed and reliability?

Pulse Transmission Rate

- Objective: Maximize pulse rate through a channel, that is, make *T* as small as possible



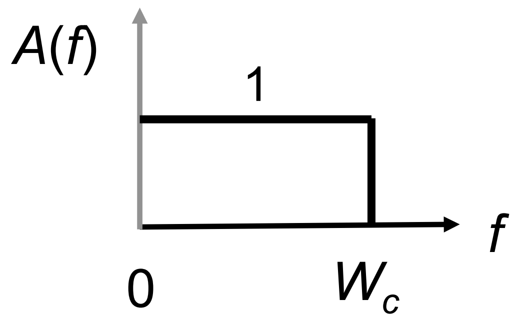
- If input is a narrow pulse, then typical output is a spread-out pulse with ringing
- Question: How frequently can these pulses be transmitted without interfering with each other?
- Answer: 2 x W_c pulses/second

where W_c is the bandwidth of the channel

Bandwidth of a Channel

$X(t) = a \cos(2\pi f t) \longrightarrow$ Channel $\longrightarrow Y(t) = A(f)\, a \cos(2\pi f t)$

- If input is sinusoid of frequency f, then
  - output is a sinusoid of same frequency f
  - Output is attenuated by an amount *A*(f) that depends on f
  - *A*(f)≈1, then input signal passes readily
  - *A*(f)≈0, then input signal is blocked
- Bandwidth W_c is range of frequencies passed by channel

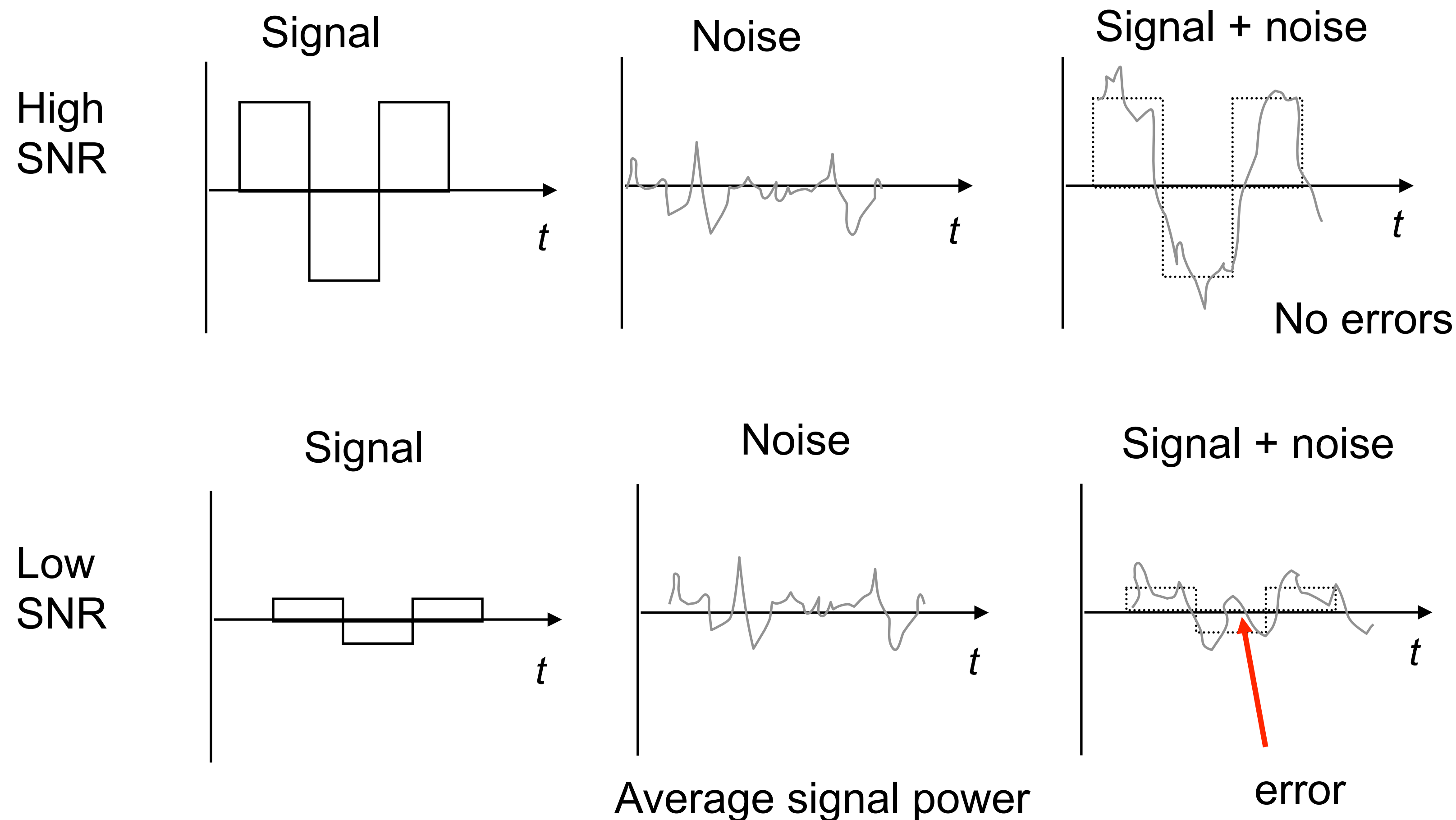


Ideal low-pass channel

Noise & Reliable Communications

- All physical systems have noise
  - Electrons always vibrate at non-zero temperature
  - Motion of electrons induces noise
- Presence of noise limits accuracy of measurement of received signal amplitude
- Errors occur if signal separation is comparable to noise level
- Bit Error Rate (BER) increases with decreasing signal-to-noise ratio
- Noise places a limit on how many amplitude levels can be used in pulse transmission

Signal-to-Noise Ratio



$$\text{SNR} = \frac{\text{Average signal power}}{\text{Average noise power}}$$

$$\text{SNR (dB)} = 10 \log_{10} \text{SNR}$$

Shannon Channel Capacity

$$C = W_c \log_2 (1 + SNR) \text{ bps}$$

- Arbitrarily reliable communications is possible if the transmission rate $R < C$.
- If $R > C$, then arbitrarily reliable communications is not possible.
- "Arbitrarily reliable" means the BER can be made arbitrarily small through sufficiently complex coding.
- C can be used as a measure of how close a system design is to the best achievable performance.
- Bandwidth W_c & SNR determine C

Example

- Find the Shannon channel capacity for a telephone channel with W_c = 3400 Hz and *SNR* = 10000

$$C = 3400 \log_2 (1 + 10000)$$
$$= 3400 \log_{10} (10001)/\log_{10} 2 = 45200 \text{ bps}$$

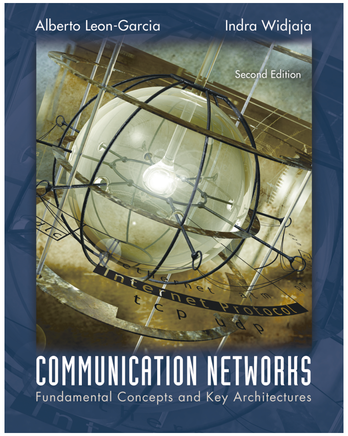
Note that *SNR* = 10000 corresponds to

$$SNR\ (\text{dB}) = 10 \log_{10}(10001) = 40 \text{ dB}$$

Bit Rates of Digital Transmission Systems

System	Bit Rate	Observations
Telephone twisted pair	33.6-56 kbps	4 kHz telephone channel
Ethernet twisted pair	10 Mbps, 100 Mbps	100 meters of unshielded twisted copper wire pair
Cable modem	500 kbps-4 Mbps	Shared CATV return channel
ADSL twisted pair	64-640 kbps in, 1.536-6.144 Mbps out	Coexists with analog telephone signal
2.4 GHz radio	2-11 Mbps	IEEE 802.11 wireless LAN
28 GHz radio	1.5-45 Mbps	5 km multipoint radio
Optical fiber	2.5-10 Gbps	1 wavelength
Optical fiber	>1600 Gbps	Many wavelengths

Chapter 3
Digital Transmission Fundamentals



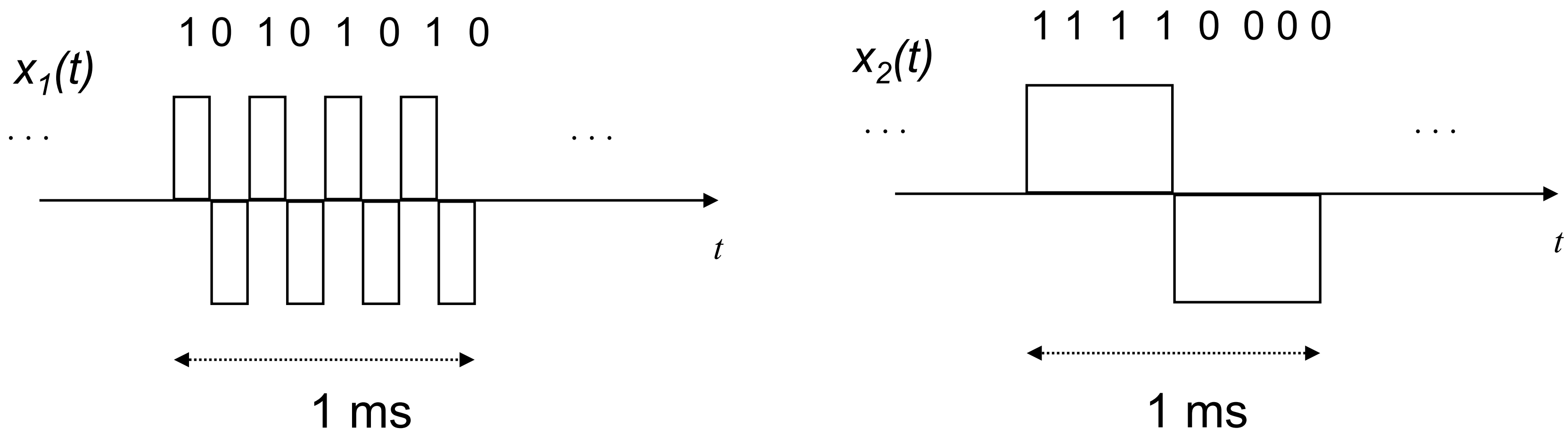
Digital Representation of Analog Signals

Digitization of Analog Signals

1. Sampling: obtain samples of *x(t)* at uniformly spaced time intervals
2. Quantization: map each sample into an approximation value of finite precision
   - Pulse Code Modulation: telephone speech
   - CD audio
3. Compression: to lower bit rate further, apply additional compression method
   - Differential coding: cellular telephone speech
   - Subband coding: MP3 audio
   - Compression discussed in Chapter 12

Sampling Rate and Bandwidth

- A signal that varies faster needs to be sampled more frequently
- *Bandwidth* measures how fast a signal varies



- What is the bandwidth of a signal?
- How is bandwidth related to sampling rate?

Periodic Signals

- A periodic signal with period T can be represented as sum of sinusoids using Fourier Series:

$$x(t) = a_0 + a_1\cos(2\pi f_0 t + \phi_1) \;+\; a_2\cos(2\pi 2 f_0 t + \phi_2) \;+ \ldots$$
$$+\; a_k\cos(2\pi k f_0 t + \phi_k) + \ldots$$

"DC"
long-term
average

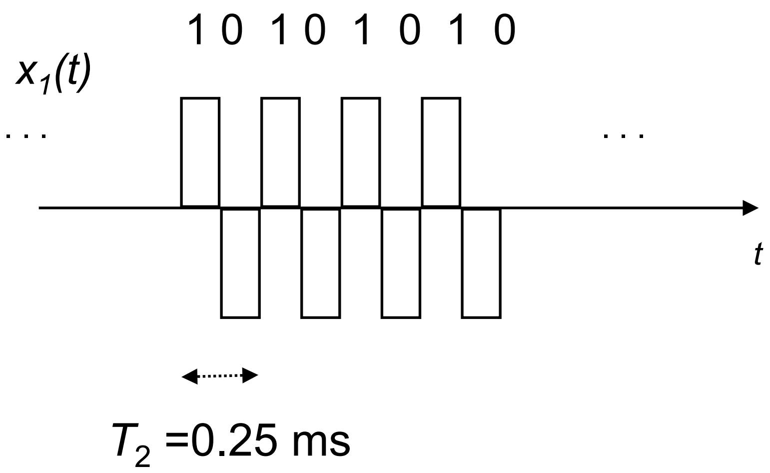
fundamental
frequency $f_0 = 1/T$
first harmonic

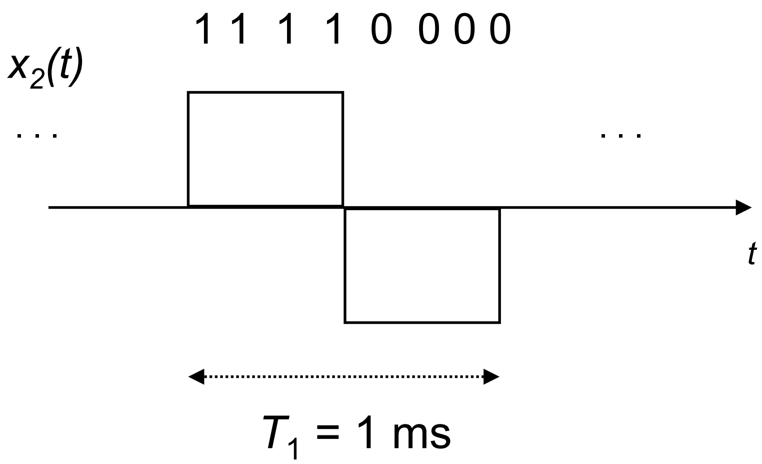
kth harmonic

•$|a_k|$ determines amount of power in kth harmonic

•Amplitude specturm $|a_0|$, $|a_1|$, $|a_2|$, …

Example Fourier Series





$$x_1(t) = 0 + \frac{4}{\pi}\cos(2\pi 4000t) + \frac{4}{3\pi}\cos(2\pi 3(4000)t) + \frac{4}{5\pi}\cos(2\pi 5(4000)t) + \ldots$$

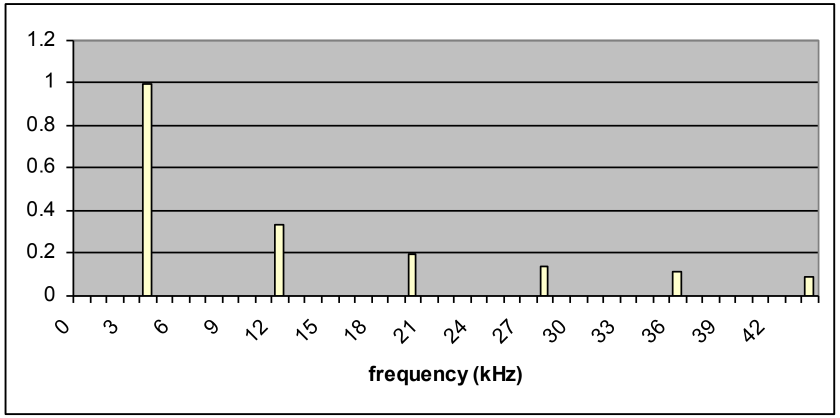
$$x_2(t) = 0 + \frac{4}{\pi}\cos(2\pi 1000t) + \frac{4}{3\pi}\cos(2\pi 3(1000)t) + \frac{4}{5\pi}\cos(2\pi 5(1000)t) + \ldots$$

Only odd harmonics have power

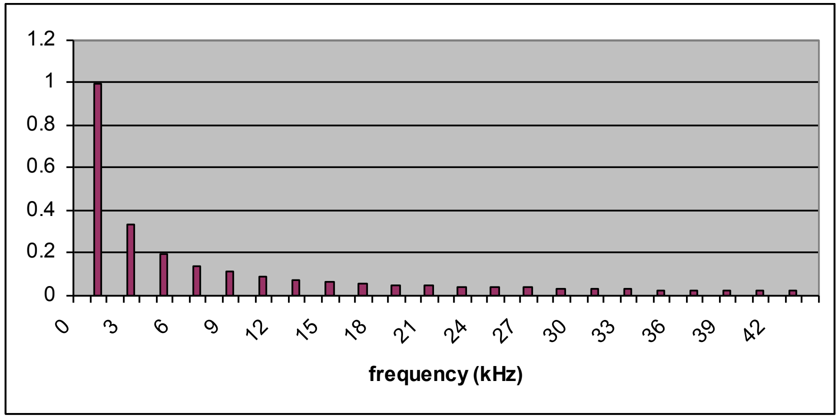
Spectra & Bandwidth

- Spectrum of a signal: magnitude of amplitudes as a function of frequency
- $x_1(t)$ varies faster in time & has more high frequency content than $x_2(t)$
- Bandwidth W_s is defined as range of frequencies where a signal has non-negligible power, e.g. range of band that contains 99% of total signal power

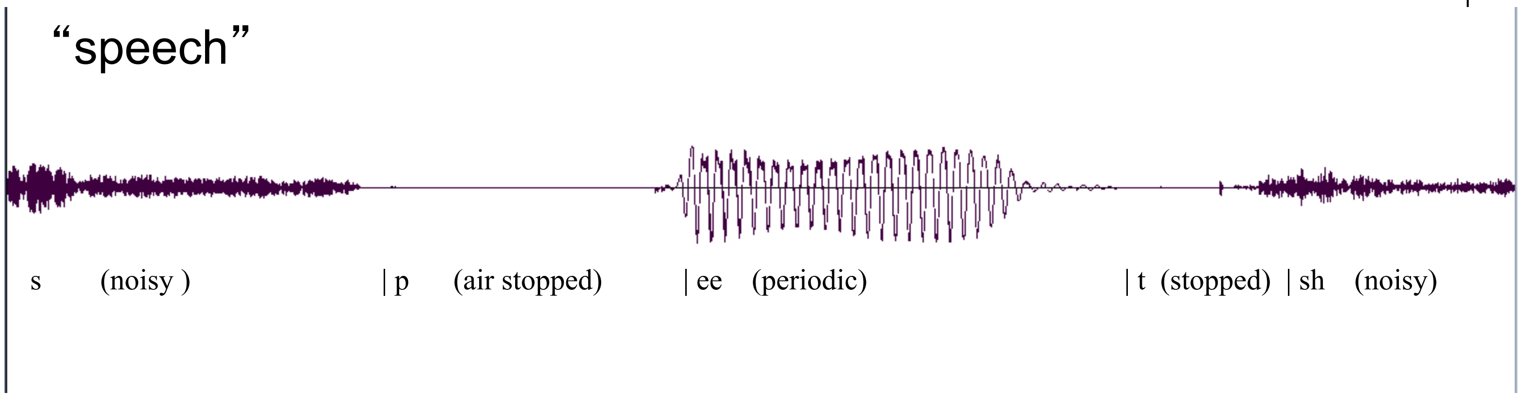
Spectrum of $x_1(t)$



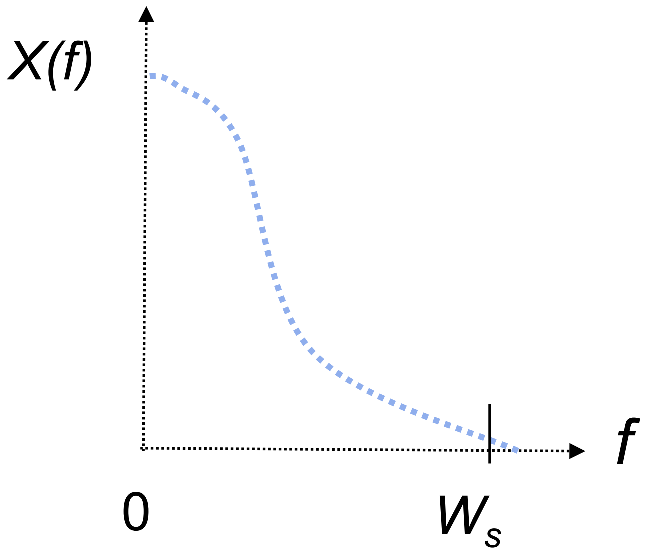
Spectrum of $x_2(t)$



Bandwidth of General Signals



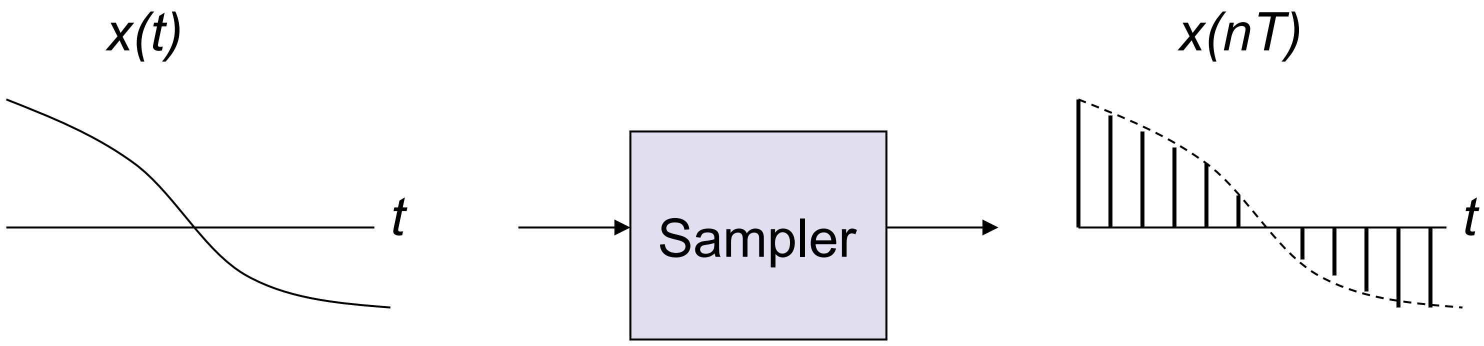
- Not all signals are periodic
  - E.g. voice signals varies according to sound
  - Vowels are periodic, “s” is noiselike
- Spectrum of long-term signal
  - Averages over many sounds, many speakers
  - Involves Fourier transform
- Telephone speech: 4 kHz
- CD Audio: 22 kHz



Sampling Theorem

Nyquist: Perfect reconstruction if sampling rate $1/T > 2W_s$

(a)



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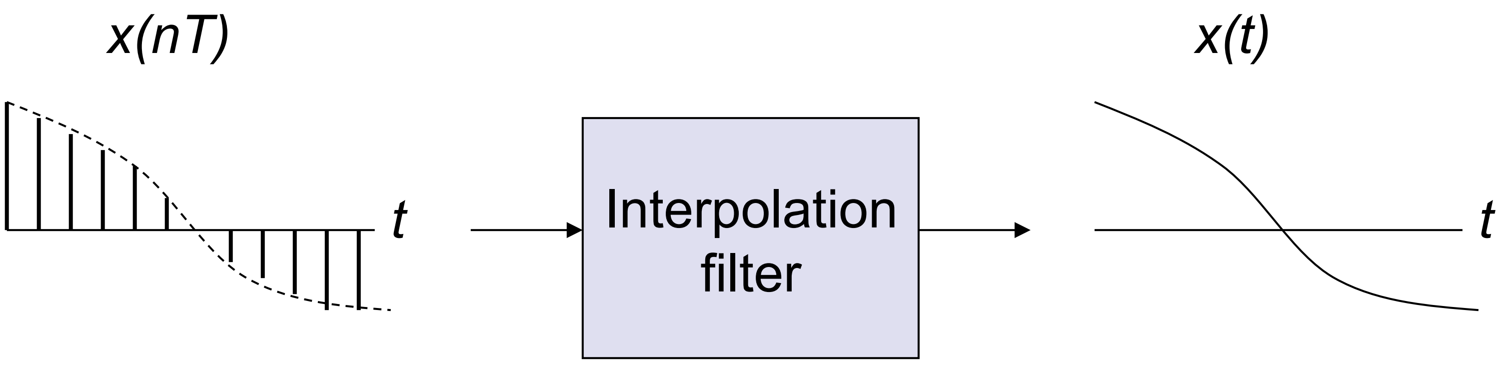
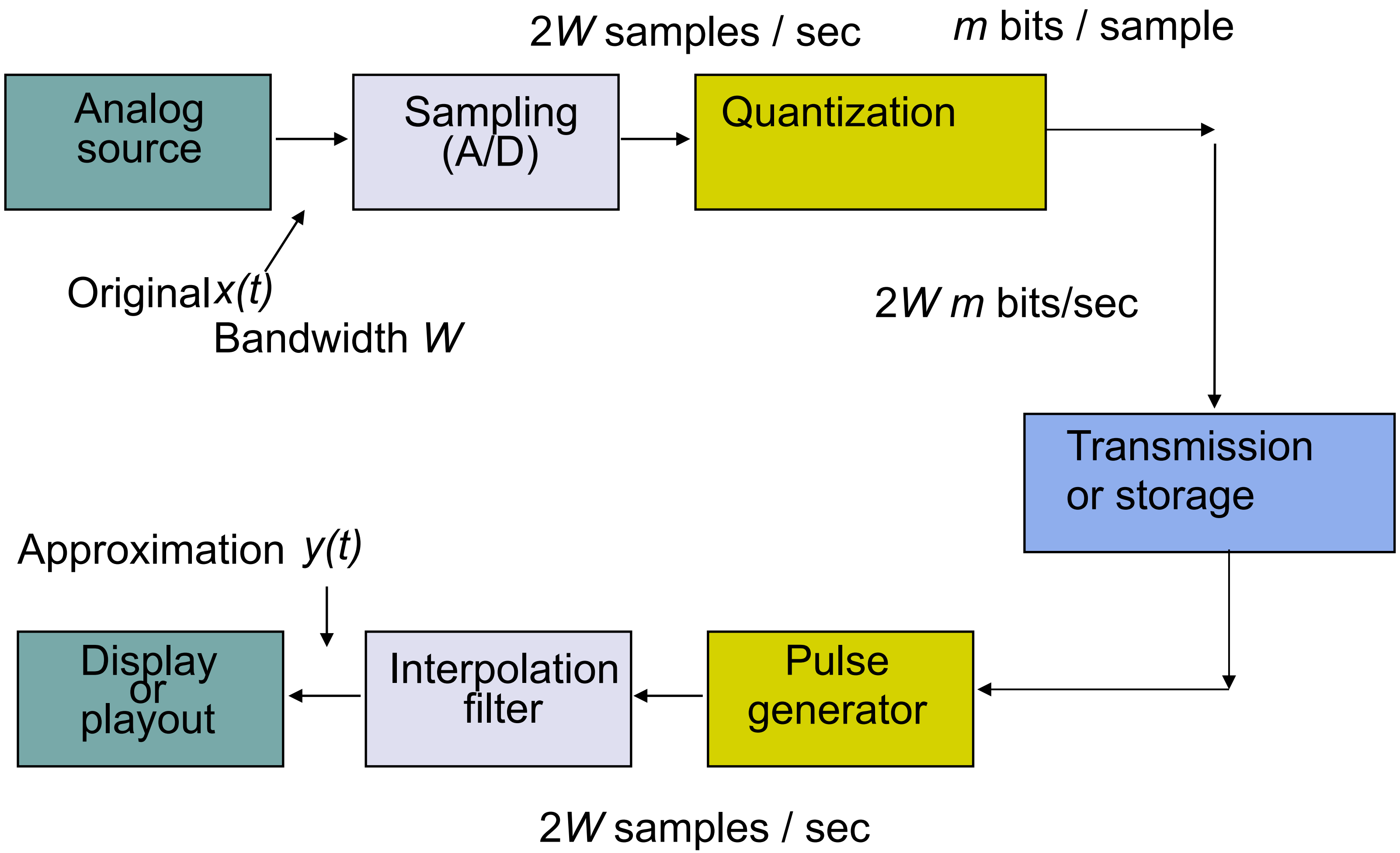


Digital Transmission of Analog Information



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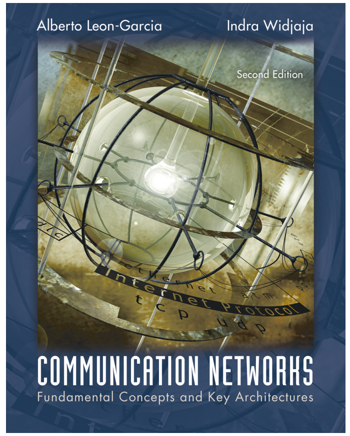
Digital Transmission Fundamentals



Characterization of Communication Channels

Communications Channels

- A *physical medium* is an inherent part of a communications system
  - Copper wires, radio medium, or optical fiber
- Communications system includes electronic or optical devices that are part of the path followed by a signal
  - Equalizers, amplifiers, signal conditioners
- By *communication channel* we refer to the combined end-to-end physical medium and attached devices
- Sometimes we use the term *filter* to refer to a channel especially in the context of a specific mathematical model for the channel

How good is a channel?

- Performance: What is the maximum reliable transmission speed?
  - Speed: Bit rate, *R* bps
  - Reliability: Bit error rate, BER=10^{-k}
  - Focus of this section
- Cost: What is the cost of alternatives at a given level of performance?
  - Wired vs. wireless?
  - Electronic vs. optical?
  - Standard A vs. standard B?

Communications Channel

Signal Bandwidth

- In order to transfer data faster, a signal has to vary more quickly.

Channel Bandwidth

- A channel or medium has an inherent limit on how fast the signals it passes can vary
- *Limits how tightly input pulses can be packed*

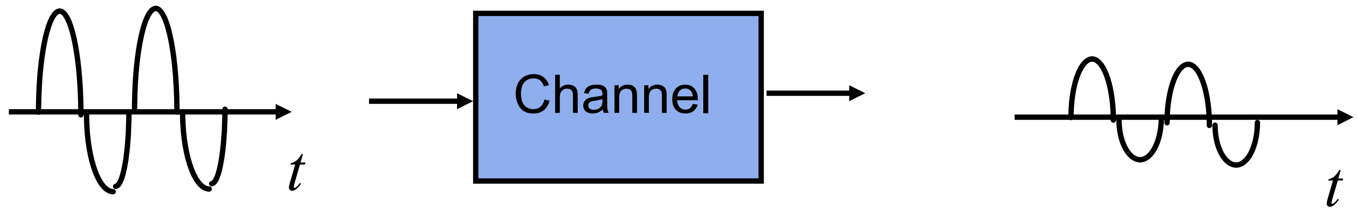
Transmission Impairments

- Signal attenuation
- Signal distortion
- Spurious noise
- Interference from other signals
- *Limits accuracy of measurements on received signal*

Frequency Domain Channel Characterization

$x(t) = A_{in} \cos 2\pi ft$

$y(t) = A_{out} \cos (2\pi ft + \varphi(f))$



$$A(f) = \frac{A_{out}}{A_{in}}$$

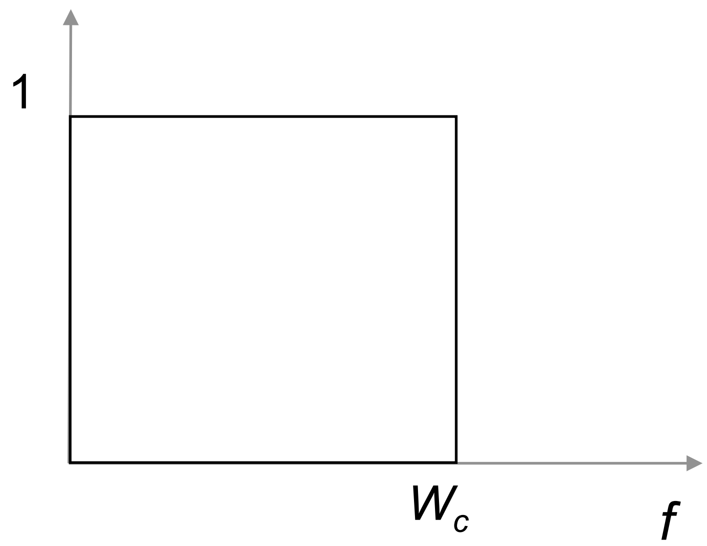
- Apply sinusoidal input at frequency *f*
  - Output is sinusoid at same frequency, but attenuated & phase-shifted
  - Measure amplitude of output sinusoid (of same frequency *f*)
  - Calculate amplitude response
    - $A(f)$ = ratio of output amplitude to input amplitude
  - If $A(f) \approx 1$, then input signal passes readily
  - If $A(f) \approx 0$, then input signal is blocked
- Bandwidth W_c is range of frequencies passed by channel

Ideal Low-Pass Filter

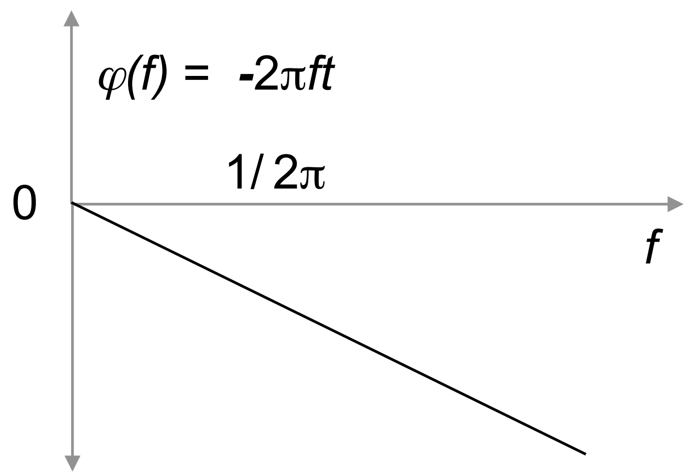
- Ideal filter: all sinusoids with frequency $f<W_c$ are passed without attenuation and delayed by τ seconds; sinusoids at other frequencies are blocked

$$y(t)=A_{in}\cos(2\pi ft - 2\pi f\tau) = A_{in}\cos(2\pi f(t-\tau)) = x(t-\tau)$$

Amplitude Response



Phase Response

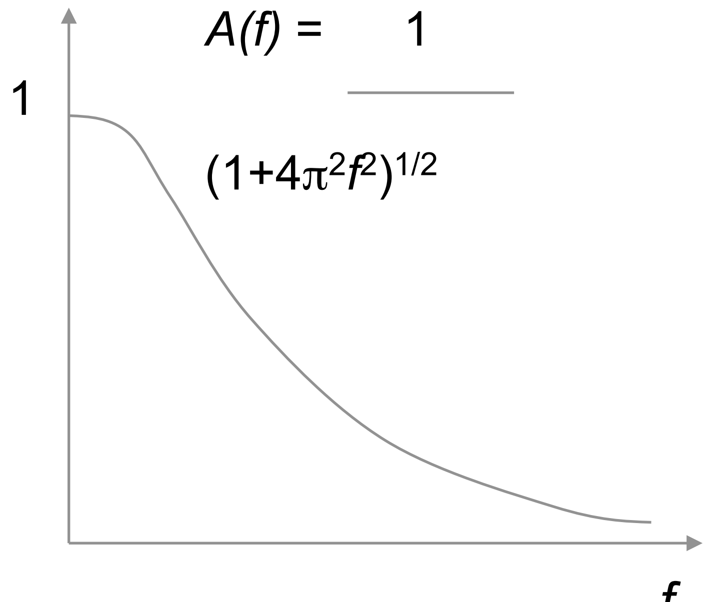


Example: Low-Pass Filter

- Simplest non-ideal circuit that provides low-pass filtering
  - Inputs at different frequencies are attenuated by different amounts
  - Inputs at different frequencies are delayed by different amounts

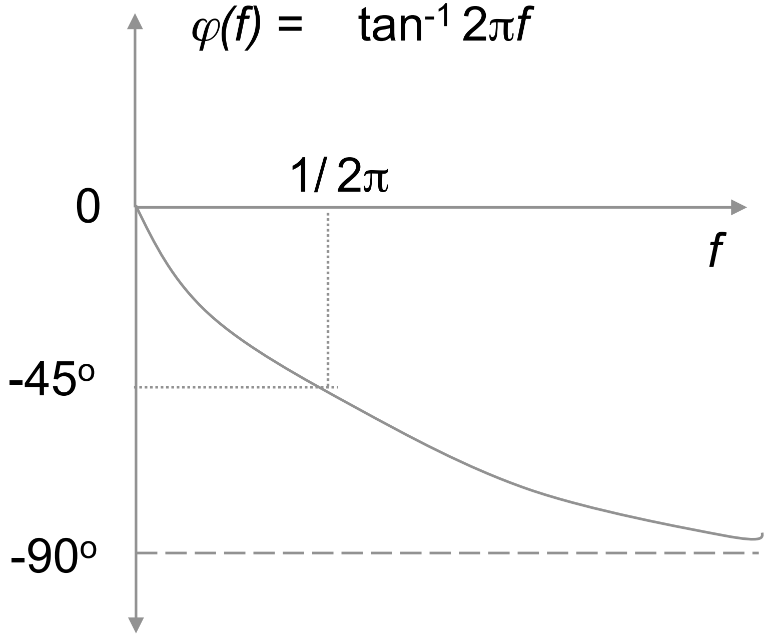
Amplitude Response

$$A(f) = \frac{1}{(1+4\pi^2 f^2)^{1/2}}$$

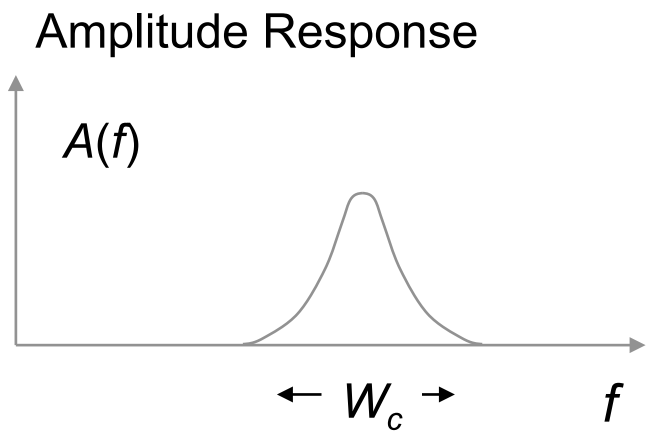


Phase Response

$$\varphi(f) = \tan^{-1} 2\pi f$$

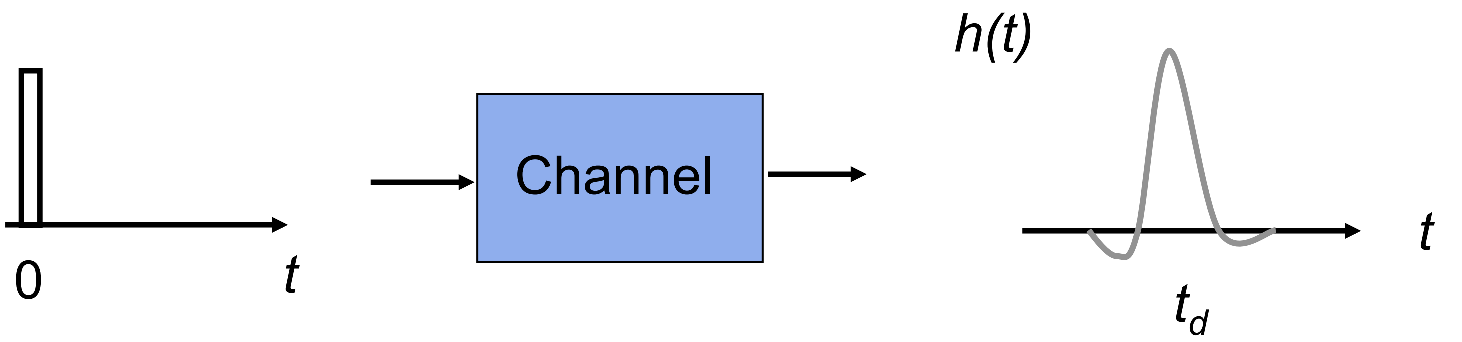


Example: Bandpass Channel



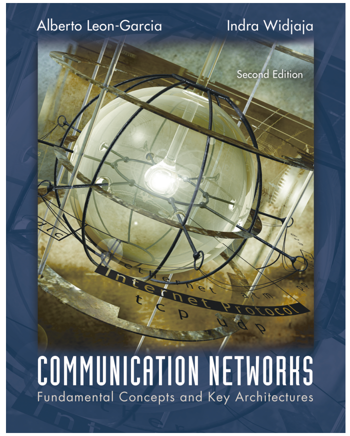
- Some channels pass signals within a band that excludes low frequencies
  - Telephone modems, radio systems, …
- *Channel bandwidth* is the width of the frequency band that passes non-negligible signal power

Time-domain Characterization



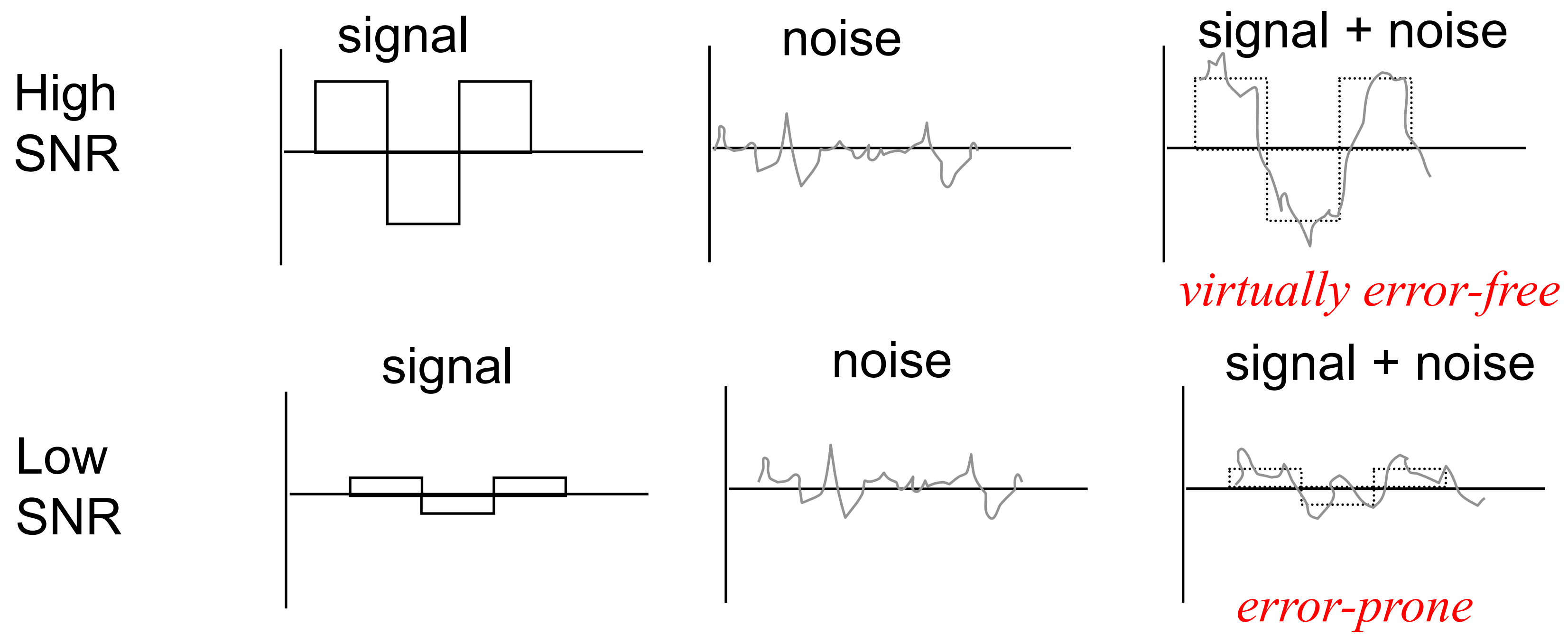
- Time-domain characterization of a channel requires finding the *impulse response h(t)*
- Apply a very narrow pulse to a channel and observe the channel output
  - *h(t)* typically a delayed pulse with ringing
- Interested in system designs with *h(t)* that can be packed closely without interfering with each other

Chapter 3
Digital Transmission Fundamentals



Fundamental Limits in Digital Transmission

Channel Noise affects Reliability



$$\text{SNR} = \frac{\text{Average Signal Power}}{\text{Average Noise Power}}$$

$$\text{SNR (dB)} = 10 \log_{10} \text{SNR}$$

Shannon Channel Capacity

- If transmitted power is limited, then as *M* increases spacing between levels decreases
- Presence of noise at receiver causes more frequent errors to occur as *M* is increased

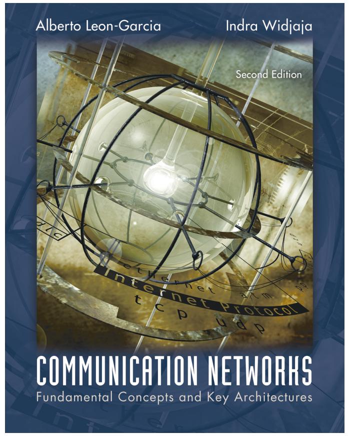
Shannon Channel Capacity:

The maximum reliable transmission rate over an ideal channel with bandwidth *W* Hz, with Gaussian distributed noise, and with SNR *S/N* is

$$C = W \log_2 (1 + S/N) \text{ bits per second}$$

- Reliable means error rate can be made arbitrarily small by proper coding

Chapter 3
Digital Transmission Fundamentals

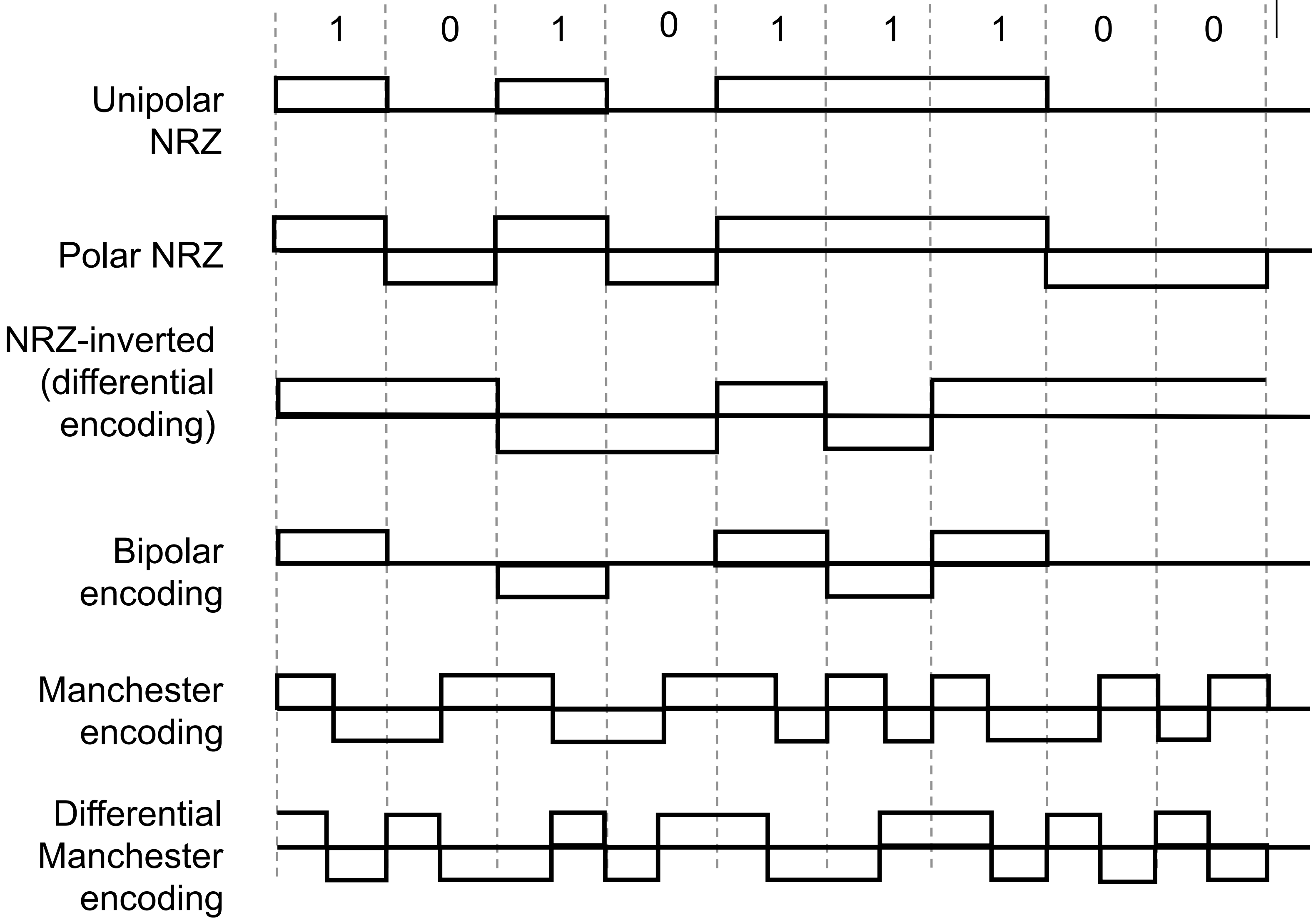


Line Coding

What is Line Coding?

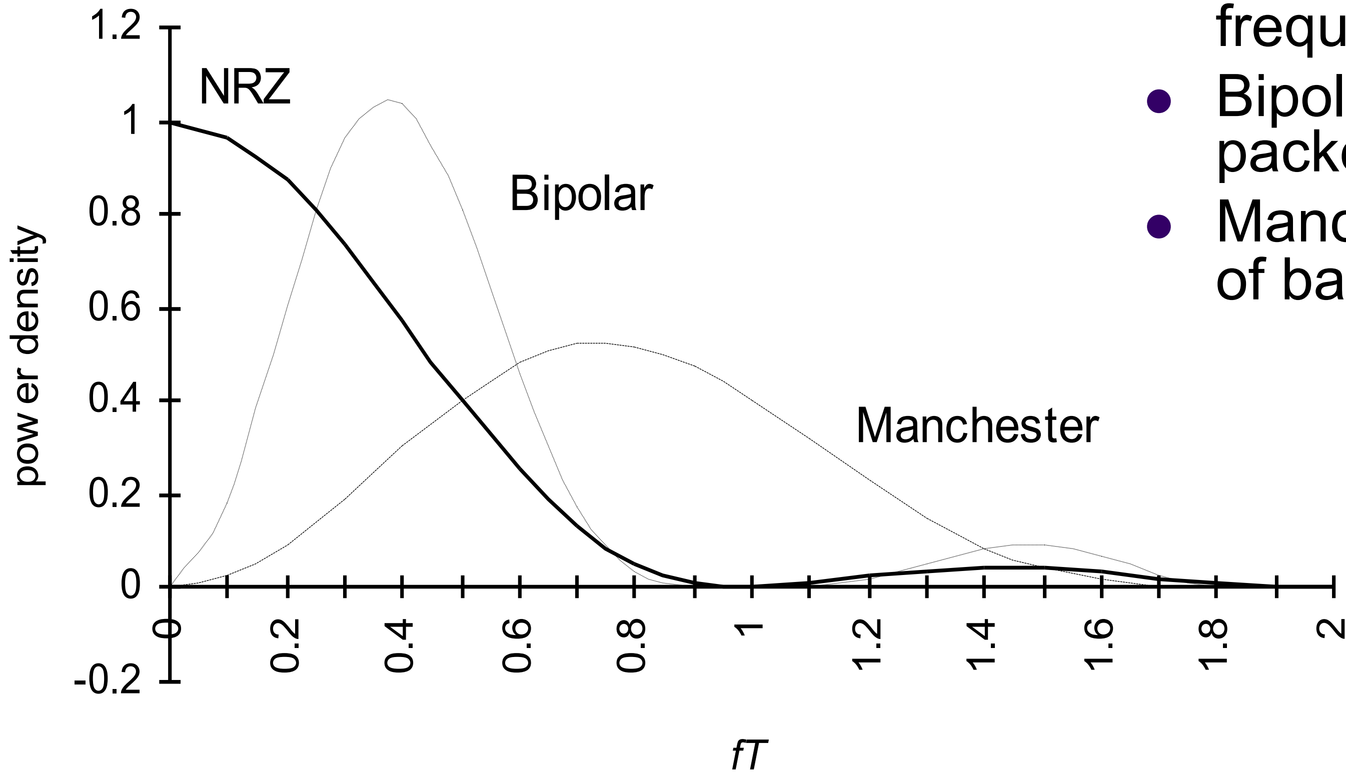
- Mapping of binary information sequence into the digital signal that enters the channel
  - Ex. “1” maps to +A square pulse; “0” to –A pulse
- Line code selected to meet system requirements:
  - *Transmitted power*: Power consumption = $
  - *Bit timing*: Transitions in signal help timing recovery
  - *Bandwidth efficiency*: Excessive transitions wastes bw
  - *Low frequency content*: Some channels block low frequencies
    - long periods of +A or of –A causes signal to “droop”
    - Waveform should not have low-frequency content
  - *Error detection*: Ability to detect errors helps
  - *Complexity/cost*: Is code implementable in chip at high speed?

Line coding examples



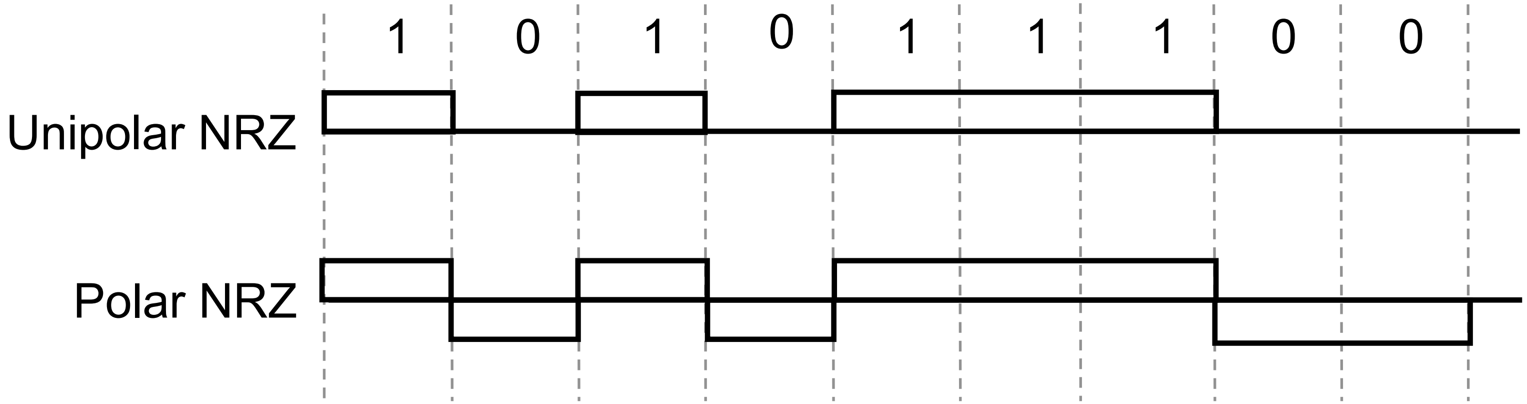
Spectrum of Line codes

- Assume 1s & 0s independent & equiprobable



- NRZ has high content at low frequencies
- Bipolar tightly packed around *T*/2
- Manchester wasteful of bandwidth

Unipolar & Polar Non-Return-to-Zero (NRZ)



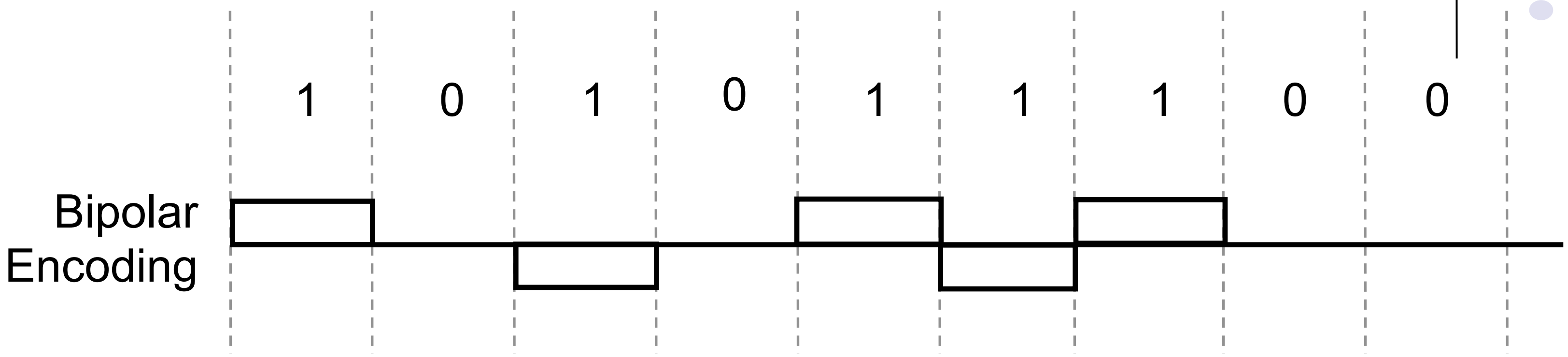
Unipolar NRZ

- "1" maps to +A pulse
- "0" maps to no pulse
- High Average Power
  $0.5*A^2 + 0.5*0^2 = A^2/2$
- Long strings of A or 0
  - Poor timing
  - Low-frequency content
- Simple

Polar NRZ

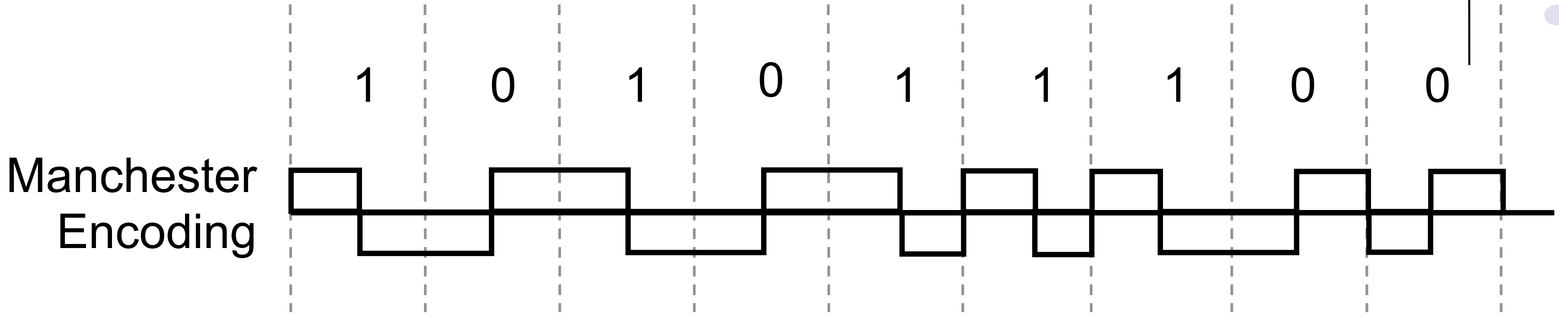
- "1" maps to +A/2 pulse
- "0" maps to –A/2 pulse
- Better Average Power
  $0.5*(A/2)^2 + 0.5*(-A/2)^2 = A^2/4$
- Long strings of +A/2 or –A/2
  - Poor timing
  - Low-frequency content
- Simple

Bipolar Code



- Three signal levels: {-A, 0, +A}
- "1" maps to +A or –A in alternation
- "0" maps to no pulse
  - Every +pulse matched by –pulse so little content at low frequencies
- String of 1s produces a square wave
  - Spectrum centered at $T/2$
- Long string of 0s causes receiver to lose synch
- Zero-substitution codes

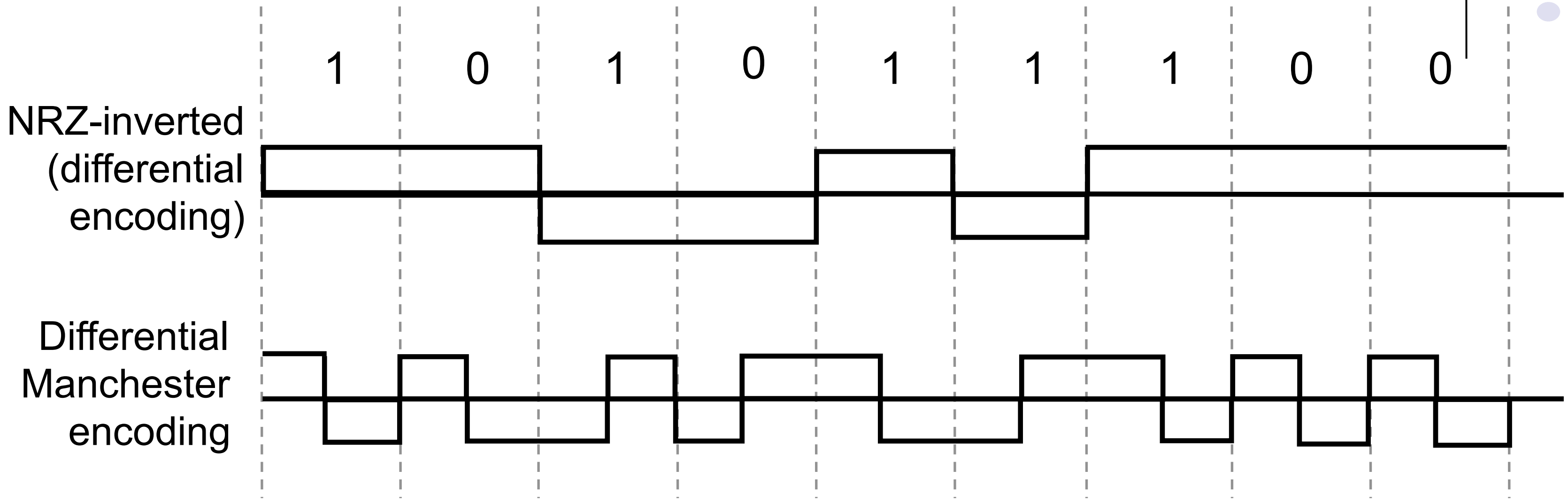
Manchester code & *m*B*n*B codes



- "1" maps into A/2 first *T*/2, -A/2 last T/2
- "0" maps into -A/2 first *T*/2, A/2 last *T*/2
- Every interval has transition in middle
  - Timing recovery easy
  - Uses double the minimum bandwidth
- Simple to implement
- Used in 10-Mbps Ethernet & other LAN standards

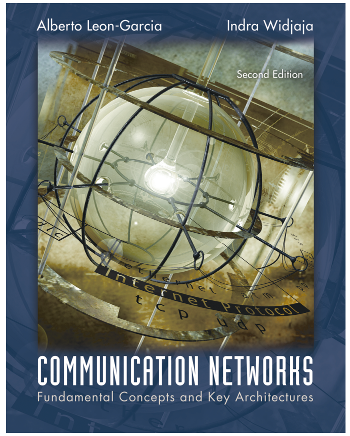
- *m*B*n*B line code
- Maps block of *m* bits into *n* bits
- Manchester code is 1B2B code
- 4B5B code used in FDDI LAN
- 8B10b code used in Gigabit Ethernet
- 64B66B code used in 10G Ethernet

Differential Coding



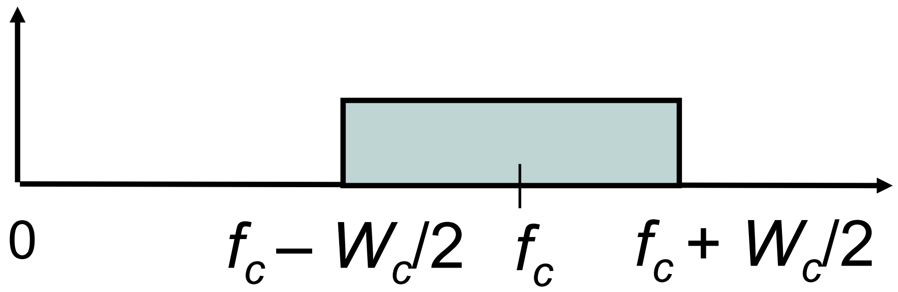
- Errors in some systems cause transposition in polarity, +A become –A and vice versa
  - All subsequent bits in Polar NRZ coding would be in error
- Differential line coding provides robustness to this type of error
- "1" mapped into transition in signal level
- "0" mapped into no transition in signal level
- Same spectrum as NRZ
- Errors occur in pairs
- Also used with Manchester coding

Chapter 3
Digital Transmission Fundamentals



Modems and Digital Modulation

Bandpass Channels

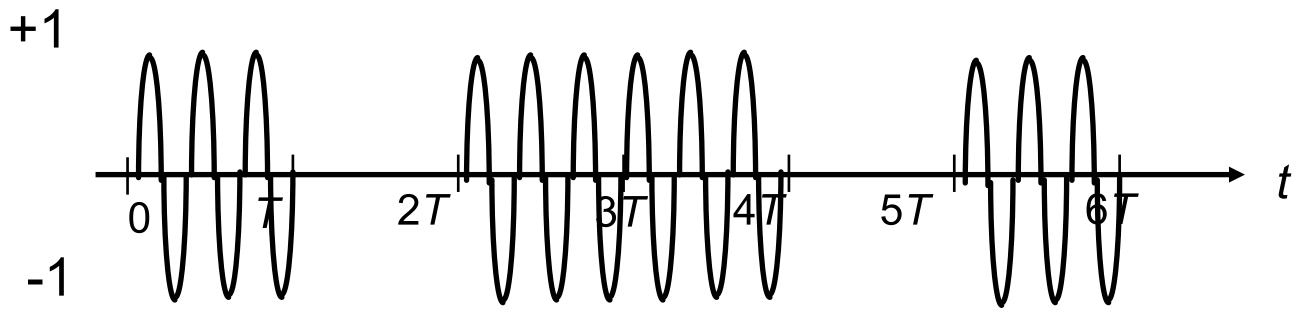


- Bandpass channels pass a range of frequencies around some center frequency f_c
  - Radio channels, telephone & DSL modems
- Digital modulators embed information into waveform with frequencies passed by bandpass channel
- Sinusoid of frequency f_c is centered in middle of bandpass channel
- Modulators embed information into a sinusoid

Amplitude Modulation and Frequency Modulation

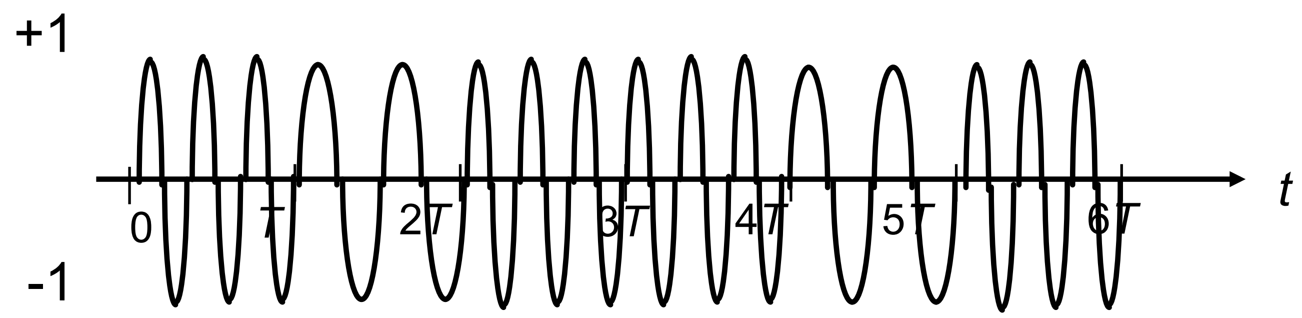
Information 1 0 1 1 0 1

Amplitude Shift Keying



Map bits into amplitude of sinusoid: "1" send sinusoid; "0" no sinusoid
Demodulator looks for signal vs. no signal

Frequency Shift Keying

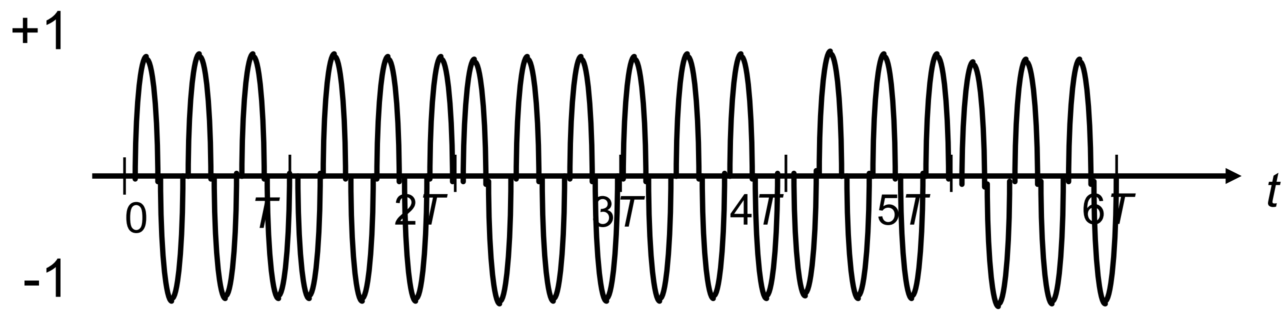


Map bits into frequency: "1" send frequency $f_c + \delta$; "0" send frequency $f_c - \delta$
Demodulator looks for power around $f_c + \delta$ or $f_c - \delta$

Phase Modulation

Information 1 0 1 1 0 1

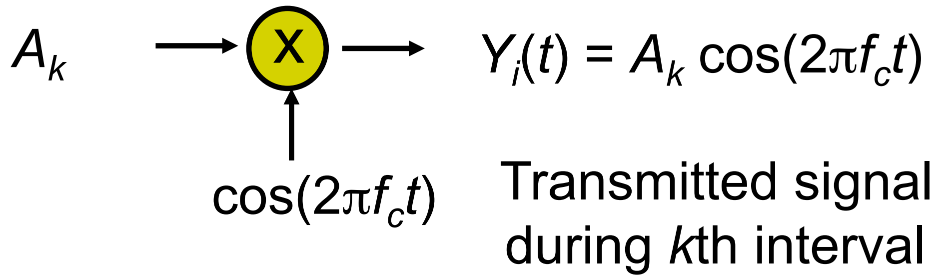
Phase Shift Keying



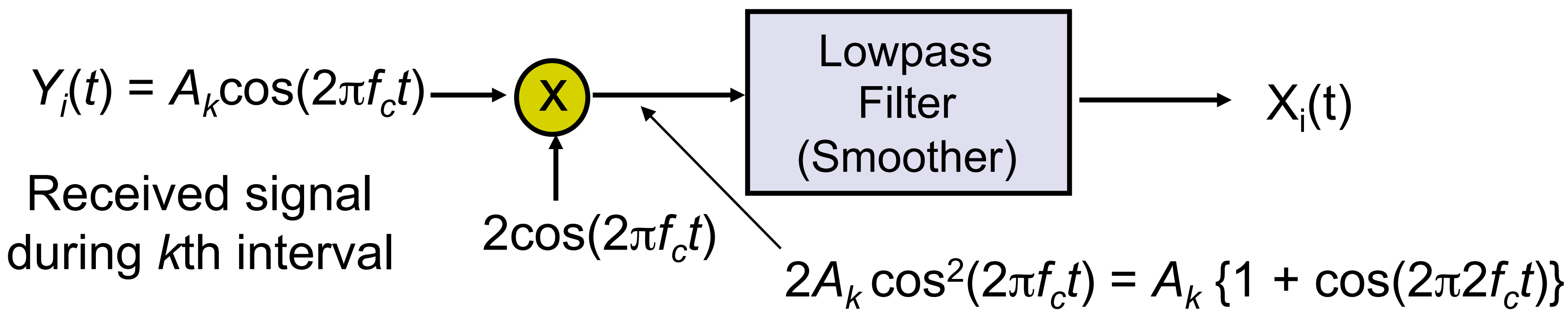
- Map bits into phase of sinusoid:
  - "1" send A cos($2\pi ft$) , i.e. phase is 0
  - "0" send A cos($2\pi ft+\pi$) , i.e. phase is π
- Equivalent to multiplying cos($2\pi ft$) by +A or -A
  - "1" send A cos($2\pi ft$) , i.e. multiply by 1
  - "0" send A cos($2\pi ft+\pi$) = - A cos($2\pi ft$) , i.e. multiply by -1
- We will focus on phase modulation

Modulator & Demodulator

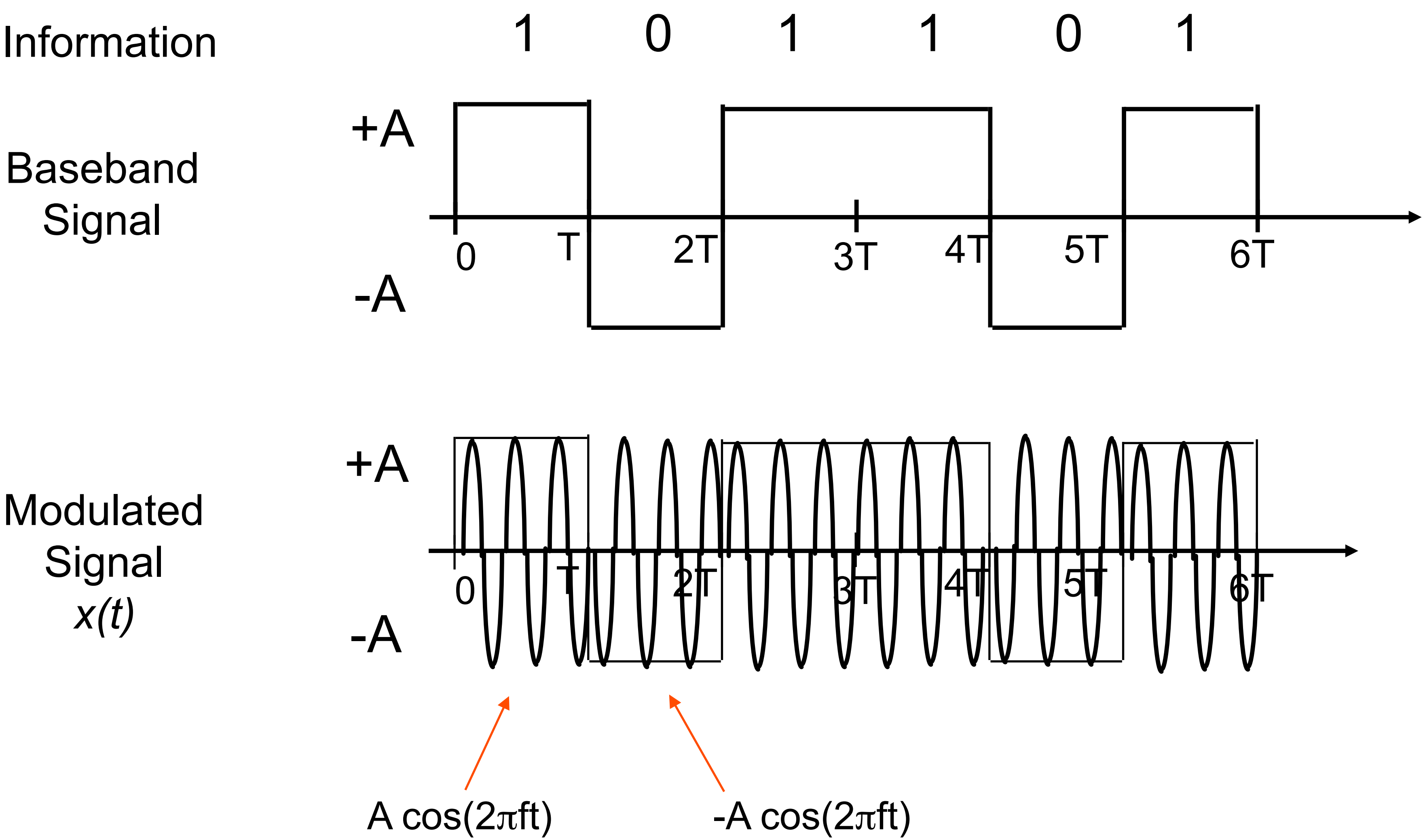
Modulate $\cos(2\pi f_c t)$ by multiplying by A_k for T seconds:



Demodulate (recover A_k) by multiplying by $2\cos(2\pi f_c t)$ for T seconds and lowpass filtering (smoothing):



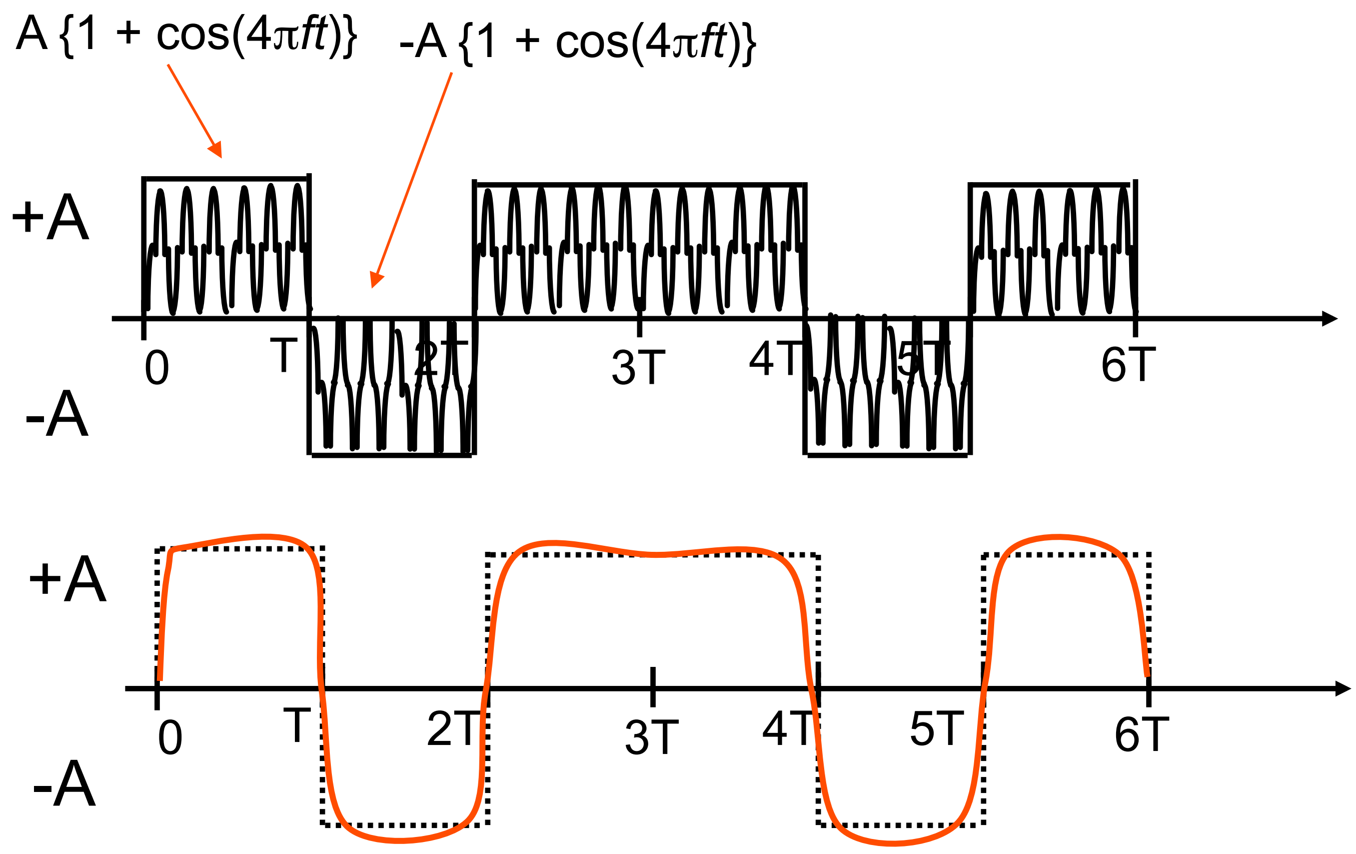
Example of Modulation



Example of Demodulation

After multiplication at receiver
$x(t)\cos(2\pi f_c t)$

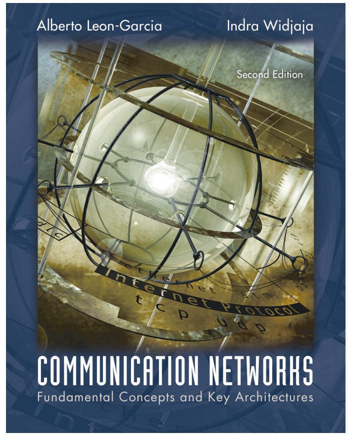
Baseband signal discernable after smoothing



Recovered Information

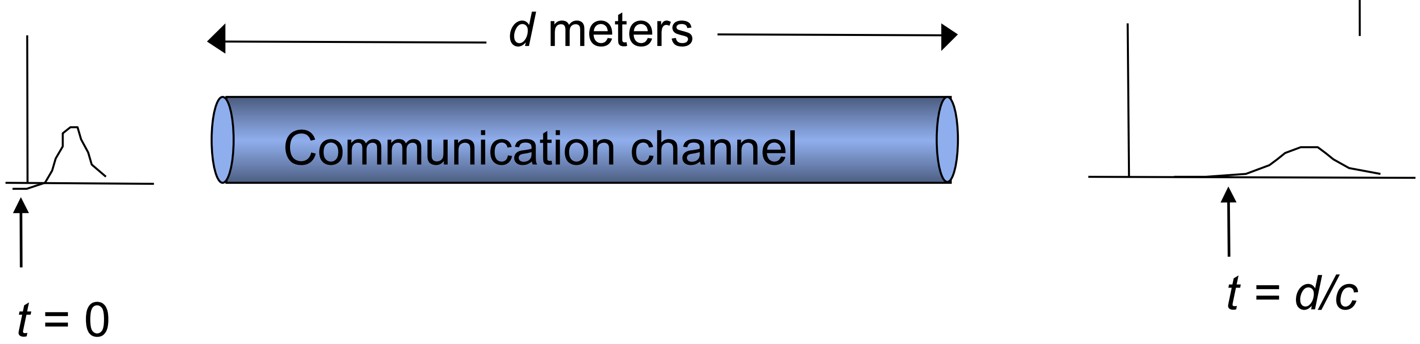
1 0 1 1 0 1

Chapter 3
Digital Transmission Fundamentals



Properties of Media and Digital Transmission Systems

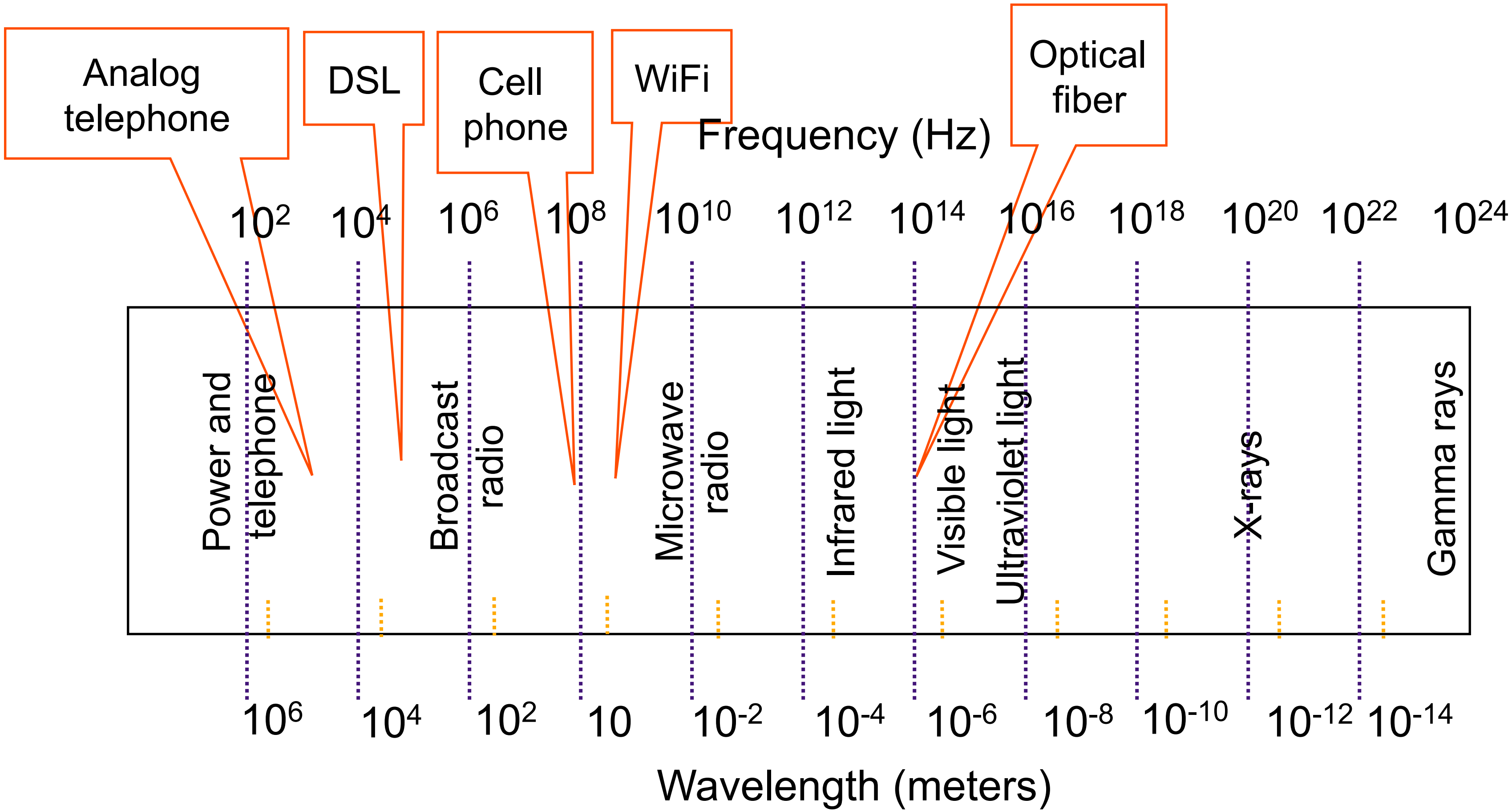
Fundamental Issues in Transmission Media



- Information bearing capacity
  - Amplitude response & bandwidth
    - dependence on distance
  - Susceptibility to noise & interference
    - Error rates & SNRs
- Propagation speed of signal
  - $c = 3 \times 10^8$ meters/second in vacuum
  - $\nu = c/\sqrt{\varepsilon}$ speed of light in medium where $\varepsilon > 1$ is the dielectric constant of the medium
  - $\nu = 2.3 \times 10^8$ m/sec in copper wire; $\nu = 2.0 \times 10^8$ m/sec in optical fiber

Communications systems & Electromagnetic Spectrum

- Frequency of communications signals



Wireless & Wired Media

Wireless Media

- Signal energy propagates in space, limited directionality
- Interference possible, so spectrum regulated
- Limited bandwidth
- Simple infrastructure: antennas & transmitters
- No physical connection between network & user
- Users can move

Wired Media

- Signal energy contained & guided within medium
- Spectrum can be re-used in separate media (wires or cables), more scalable
- Extremely high bandwidth
- Complex infrastructure: ducts, conduits, poles, right-of-way

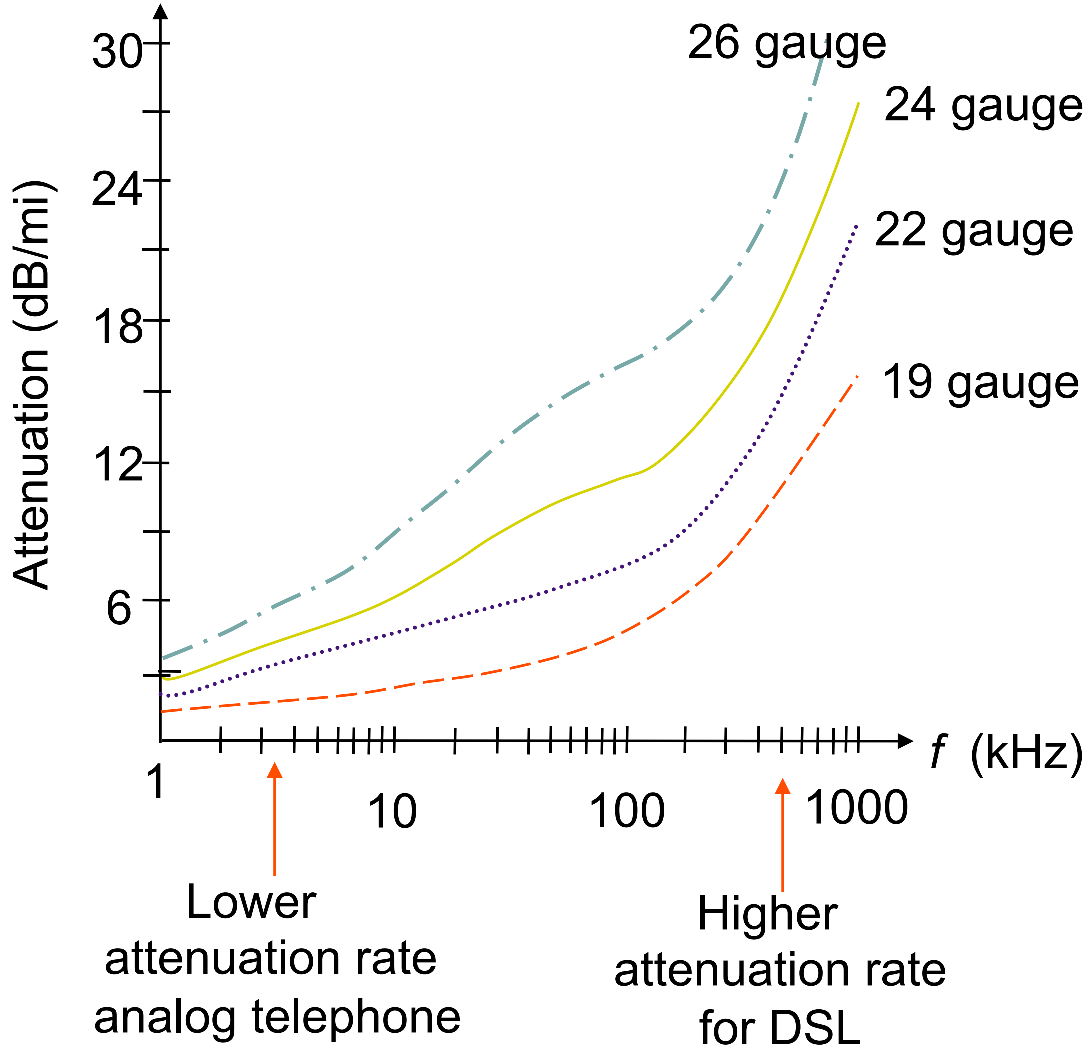
Attenuation

- Attenuation varies with media
  - Dependence on distance of central importance
- Wired media has exponential dependence
  - Received power at d meters proportional to 10^{-kd}
  - Attenuation in dB = $k\,d$, where k is dB/meter
- Wireless media has logarithmic dependence
  - Received power at d meters proportional to d^{-n}
  - Attenuation in dB = $n \log d$, where n is path loss exponent; $n=2$ in free space
  - Signal level maintained for much longer distances
  - Space communications possible

Twisted Pair

Twisted pair

- Two insulated copper wires arranged in a regular spiral pattern to minimize interference
- Various thicknesses, e.g. 0.016 inch (24 gauge)
- Low cost
- Telephone subscriber loop from customer to CO
- Old trunk plant connecting telephone COs
- Intra-building telephone from wiring closet to desktop
- In old installations, loading coils added to improve quality in 3 kHz band, but more attenuation at higher frequencies



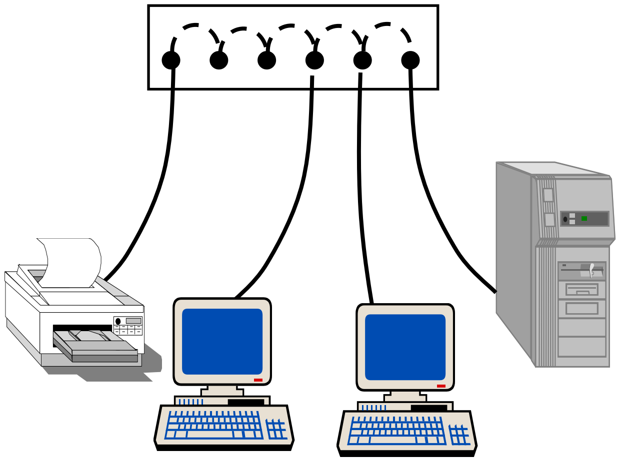
Twisted Pair Bit Rates

Table 3.5 Data rates of 24-gauge twisted pair

Standard	Data Rate	Distance
T-1	1.544 Mbps	18,000 feet, 5.5 km
DS2	6.312 Mbps	12,000 feet, 3.7 km
1/4 STS-1	12.960 Mbps	4500 feet, 1.4 km
1/2 STS-1	25.920 Mbps	3000 feet, 0.9 km
STS-1	51.840 Mbps	1000 feet, 300 m

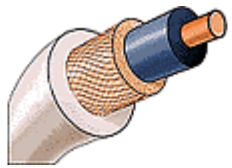
- Twisted pairs can provide high bit rates at short distances
- Asymmetric Digital Subscriber Loop (ADSL)
  - High-speed Internet Access
  - Lower 3 kHz for voice
  - Upper band for data
  - 64 kbps inbound
  - 640 kbps outbound
- Much higher rates possible at shorter distances
  - Strategy for telephone companies is to bring fiber close to home & then twisted pair
  - Higher-speed access + video

Ethernet LANs



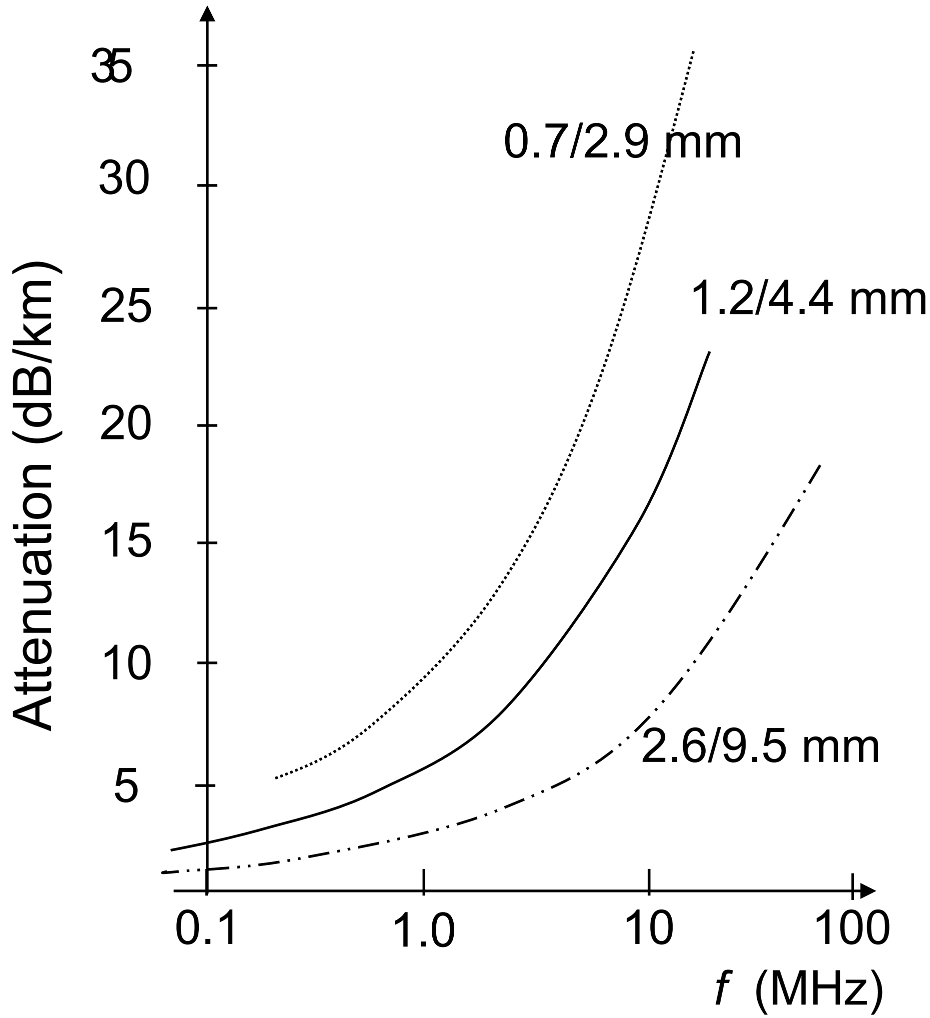
- Category 3 unshielded twisted pair (UTP): ordinary telephone wires
- Category 5 UTP: tighter twisting to improve signal quality
- Shielded twisted pair (STP): to minimize interference; costly
- 10BASE-T Ethernet
  - 10 Mbps, Baseband, Twisted pair
  - Two Cat3 pairs
  - Manchester coding, 100 meters
- 100BASE-T4 *Fast* Ethernet
  - 100 Mbps, Baseband, Twisted pair
  - Four Cat3 pairs
  - Three pairs for one direction at-a-time
  - 100/3 Mbps per pair;
  - 3B6T line code, 100 meters
- Cat5 & STP provide other options

Coaxial Cable

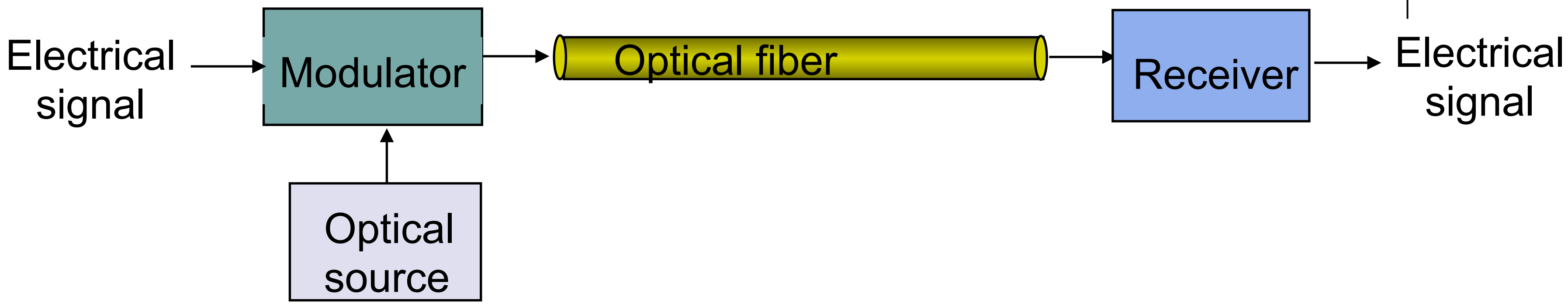


Twisted pair

- Cylindrical braided outer conductor surrounds insulated inner wire conductor
- High interference immunity
- Higher bandwidth than twisted pair
- Hundreds of MHz
- Cable TV distribution
- Long distance telephone transmission
- Original Ethernet LAN medium



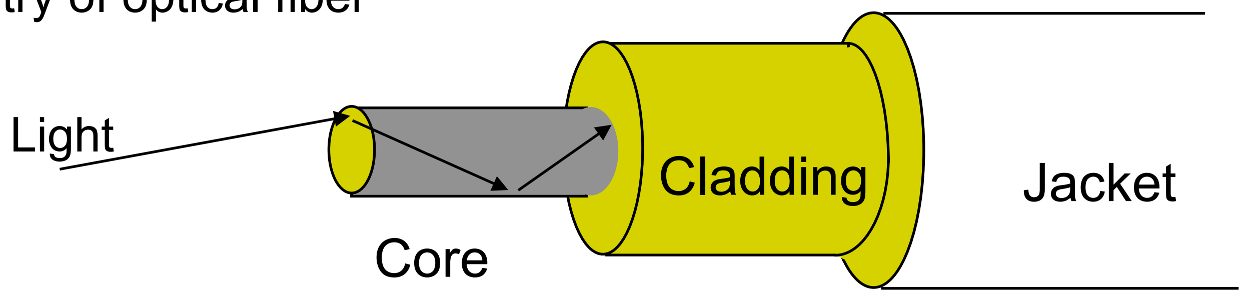
Optical Fiber



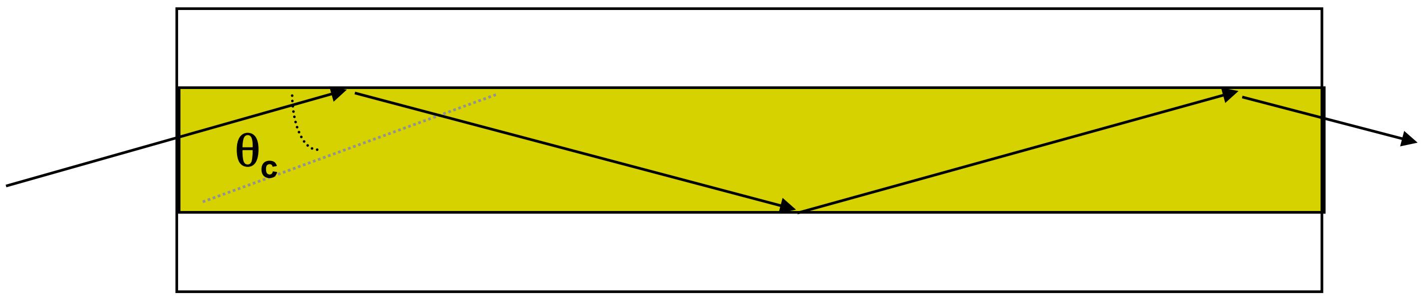
- Light sources (lasers, LEDs) generate pulses of light that are transmitted on optical fiber
  - Very long distances (>1000 km)
  - Very high speeds (>40 Gbps/wavelength)
  - Nearly error-free (BER of 10^{-15})
- Profound influence on network architecture
  - Dominates long distance transmission
  - Distance less of a cost factor in communications
  - Plentiful bandwidth for new services

Transmission in Optical Fiber

Geometry of optical fiber

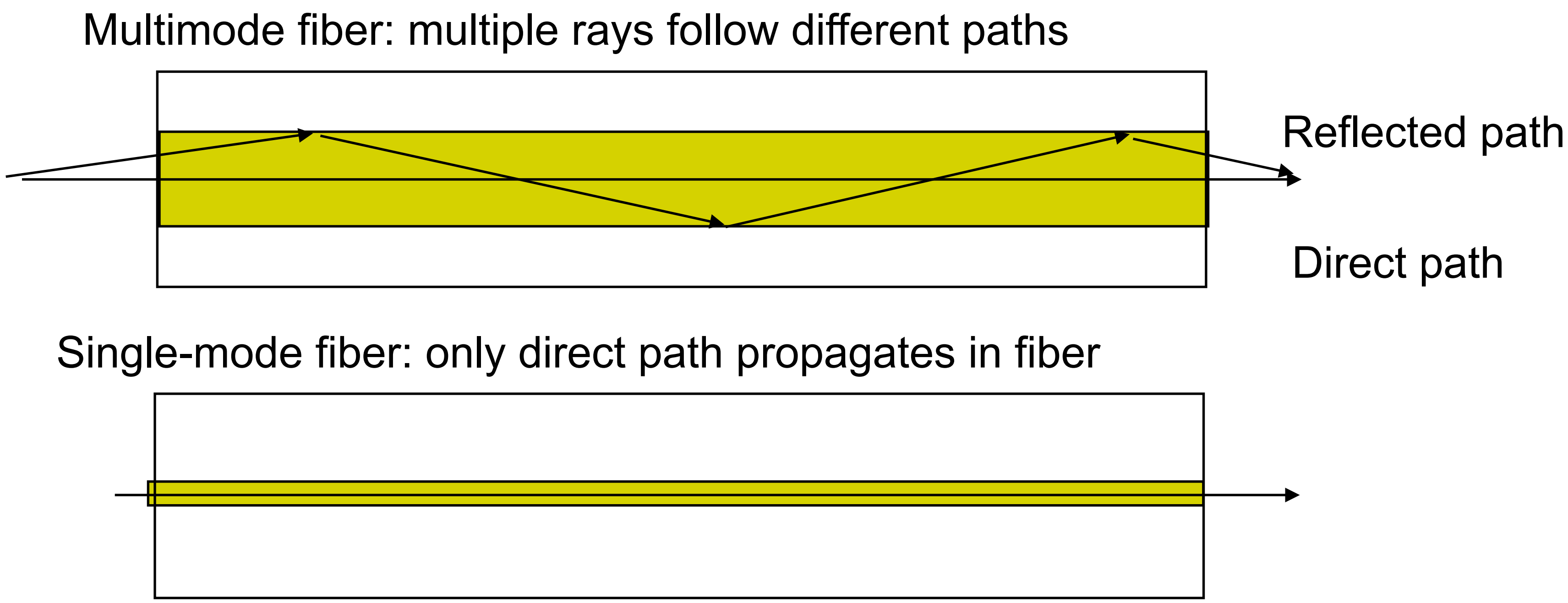


Total Internal Reflection in optical fiber



- Very fine glass cylindrical core surrounded by concentric layer of glass (cladding)
- Core has higher index of refraction than cladding
- Light rays incident at less than critical angle θ_c is completely reflected back into the core

Multimode & Single-mode Fiber

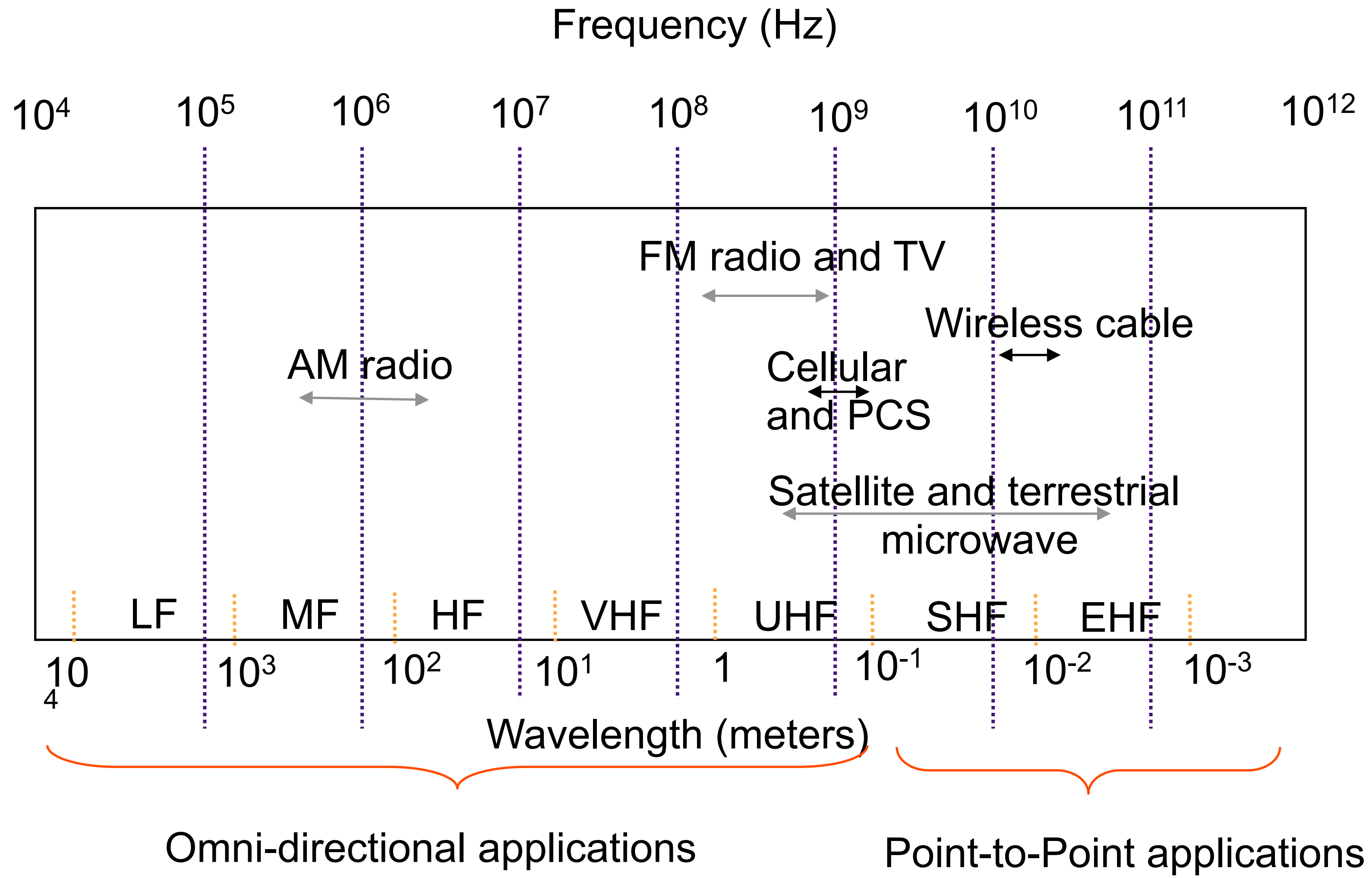


- Multimode: Thicker core, shorter reach
  - Rays on different paths interfere causing dispersion & limiting bit rate
- Single mode: Very thin core supports only one mode (path)
  - More expensive lasers, but achieves very high speeds

Radio Transmission

- Radio signals: antenna transmits sinusoidal signal (“carrier”) that radiates in air/space
- Information embedded in carrier signal using modulation, e.g. QAM
- Communications without tethering
  - Cellular phones, satellite transmissions, Wireless LANs
- Multipath propagation causes fading
- Interference from other users
- Spectrum regulated by national & international regulatory organizations

Radio Spectrum



Examples

Cellular Phone

- Allocated spectrum
- First generation:
  - 800, 900 MHz
  - Initially analog voice
- Second generation:
  - 1800-1900 MHz
  - Digital voice, messaging

Wireless LAN

- Unlicenced ISM spectrum
  - Industrial, Scientific, Medical
  - 902-928 MHz, 2.400-2.4835 GHz, 5.725-5.850 GHz
- IEEE 802.11 LAN standard
  - 11-54 Mbps

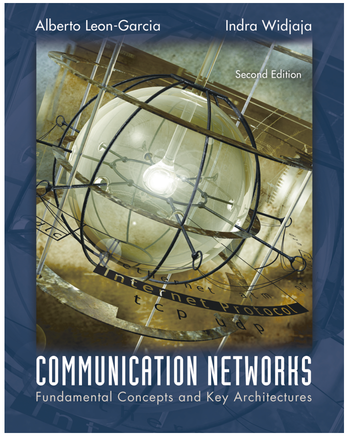
Point-to-Multipoint Systems

- Directional antennas at microwave frequencies
- High-speed digital communications between sites
- High-speed Internet Access Radio backbone links for rural areas

Satellite Communications

- Geostationary satellite @ 36000 km above equator
- Relays microwave signals from uplink frequency to downlink frequency
- Long distance telephone
- Satellite TV broadcast

Chapter 3
Digital Transmission Fundamentals



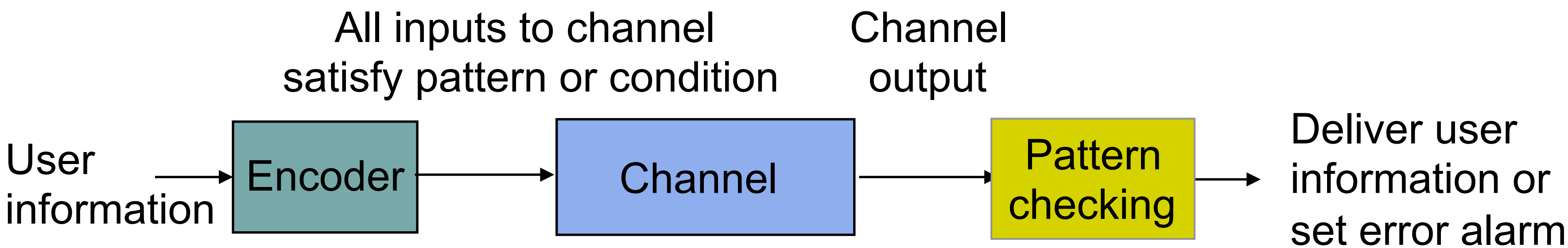
Error Detection and Correction

Error Control

- Digital transmission systems introduce errors
- Applications require certain reliability level
  - Data applications require error-free transfer
  - Voice & video applications tolerate some errors
- Error control used when transmission system does *not* meet application requirement
- Error control ensures a data stream is transmitted to a certain level of accuracy despite errors
- Two basic approaches:
  - Error ***detection*** & retransmission (ARQ)
  - Forward error ***correction*** (FEC)

Key Idea

- All transmitted data blocks ("codewords") satisfy a pattern
- If received block doesn't satisfy pattern, it is in error
- Redundancy: Only a subset of all possible blocks can be codewords
- Blindspot: when channel transforms a codeword into another codeword



Single Parity Check

- Append an overall parity check to *k* information bits

Info Bits: $b_1, b_2, b_3, \ldots, b_k$

Check Bit: $b_{k+1} = b_1 + b_2 + b_3 + \ldots + b_k$ modulo 2

Codeword: $(b_1, b_2, b_3, \ldots, b_{k,}, b_{k+!})$

- All codewords have even # of 1s
- Receiver checks to see if # of 1s is even
  - All error patterns that change an odd # of bits are detectable
  - All even-numbered patterns are undetectable
- Parity bit used in ASCII code

Example of Single Parity Code

- Information (7 bits): (0, 1, 0, 1, 1, 0, 0)
- Parity Bit: $b_8 = 0 + 1 + 0 + 1 + 1 + 0 = 1$
- Codeword (8 bits): (0, 1, 0, 1, 1, 0, 0, 1)

- If single error in bit 3 : (0, 1, 1, 1, 1, 0, 0, 1)
  - # of 1's =5, odd
  - Error detected

- If errors in bits 3 and 5: (0, 1, 1, 1, 0, 0, 0, 1)
  - # of 1's =4, even
  - Error not detected

Two-Dimensional Parity Check

- More parity bits to improve coverage
- Arrange information as columns
- Add single parity bit to each column
- Add a final "parity" column
- Used in early error control systems

$$
\begin{array}{ccccc|c}
1 & 0 & 0 & 1 & 0 & 0 \\
0 & 1 & 0 & 0 & 0 & 1 \\
1 & 0 & 0 & 1 & 0 & 0 \\
1 & 1 & 0 & 1 & 1 & 0 \\
\hline
1 & 0 & 0 & 1 & 1 & 1
\end{array}
$$

Last column consists of check bits for each row

Bottom row consists of check bit for each column

Error-detecting capability

1	0	0	1	0	0
0	0	0	0	0	1
1	0	0	1	0	0
1	1	0	1	1	0
1	0	0	1	1	1

One error

1	0	0	1	0	0
0	0	0	0	0	1
1	0	0	1	0	0
1	0	0	1	1	0
1	0	0	1	1	1

Two errors

1	0	0	1	0	0
0	0	0	1	0	1
1	0	0	1	0	0
1	0	0	1	1	0
1	0	0	1	1	1

Three errors

1	0	0	1	0	0
0	0	0	1	0	1
1	0	0	1	0	0
1	0	0	0	1	0
1	0	0	1	1	1

Four errors
(undetectable)

1, 2, or 3 errors can always be detected; Not all patterns >4 errors can be detected

Arrows indicate failed check bits

Other Error Detection Codes

- Many applications require very low error rate
- Need codes that detect the vast majority of errors
- Single parity check codes do not detect enough errors
- Two-dimensional codes require too many check bits
- The following error detecting codes used in practice:
  - Internet Check Sums
  - CRC Polynomial Codes

Polynomial Codes

- Polynomials instead of vectors for codewords
- Polynomial arithmetic instead of check sums
- Implemented using shift-register circuits
- Also called *cyclic redundancy check (CRC)* codes
- Most data communications standards use polynomial codes for error detection
- Polynomial codes also basis for powerful error-correction methods

Standard Generator Polynomials

CRC = cyclic redundancy check

- CRC-8:

  $= x^8 + x^2 + x + 1$ ATM

- CRC-16:

  $= x^{16} + x^{15} + x^2 + 1$ Bisync

  $= (x + 1)(x^{15} + x + 1)$

- CCITT-16:

  $= x^{16} + x^{12} + x^5 + 1$ HDLC, XMODEM, V.41

- CCITT-32: IEEE 802, DoD, V.42

  $= x^{32} + x^{26} + x^{23} + x^{22} + x^{16} + x^{12} + x^{11} + x^{10} + x^8 + x^7 + x^5 + x^4 + x^2 + x + 1$

Hamming Codes

- Class of *error-correcting* codes
- Capable of correcting all *single-error* patterns
- For each $m \geq 2$, there is a Hamming code of length $n = 2^m - 1$ with $n - k = m$ parity check bits

Redundancy

m	$n = 2^m - 1$	$k = n - m$	m/n
3	7	4	3/7
4	15	11	4/15
5	31	26	5/31
6	63	57	6/63