2. Data transmission
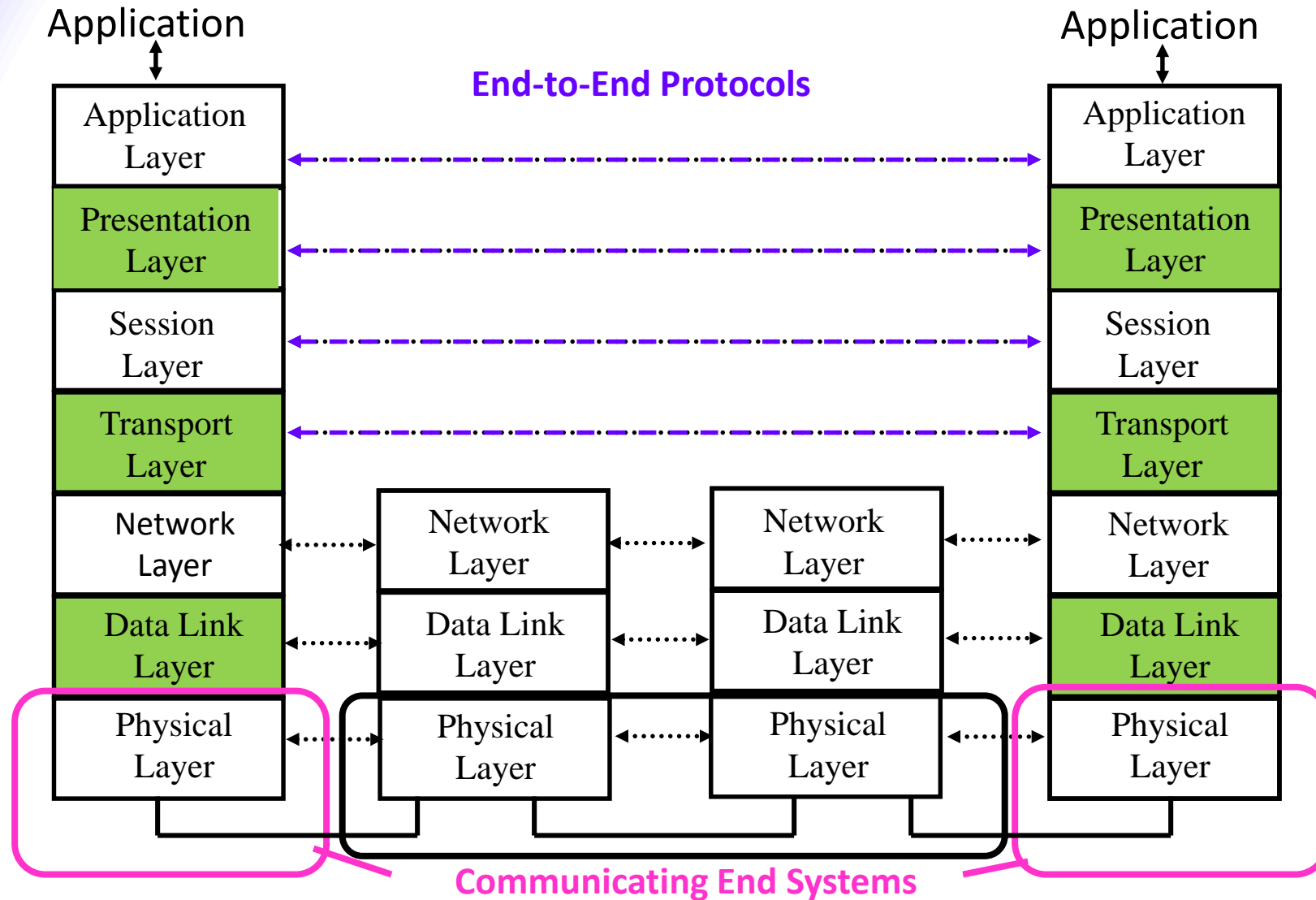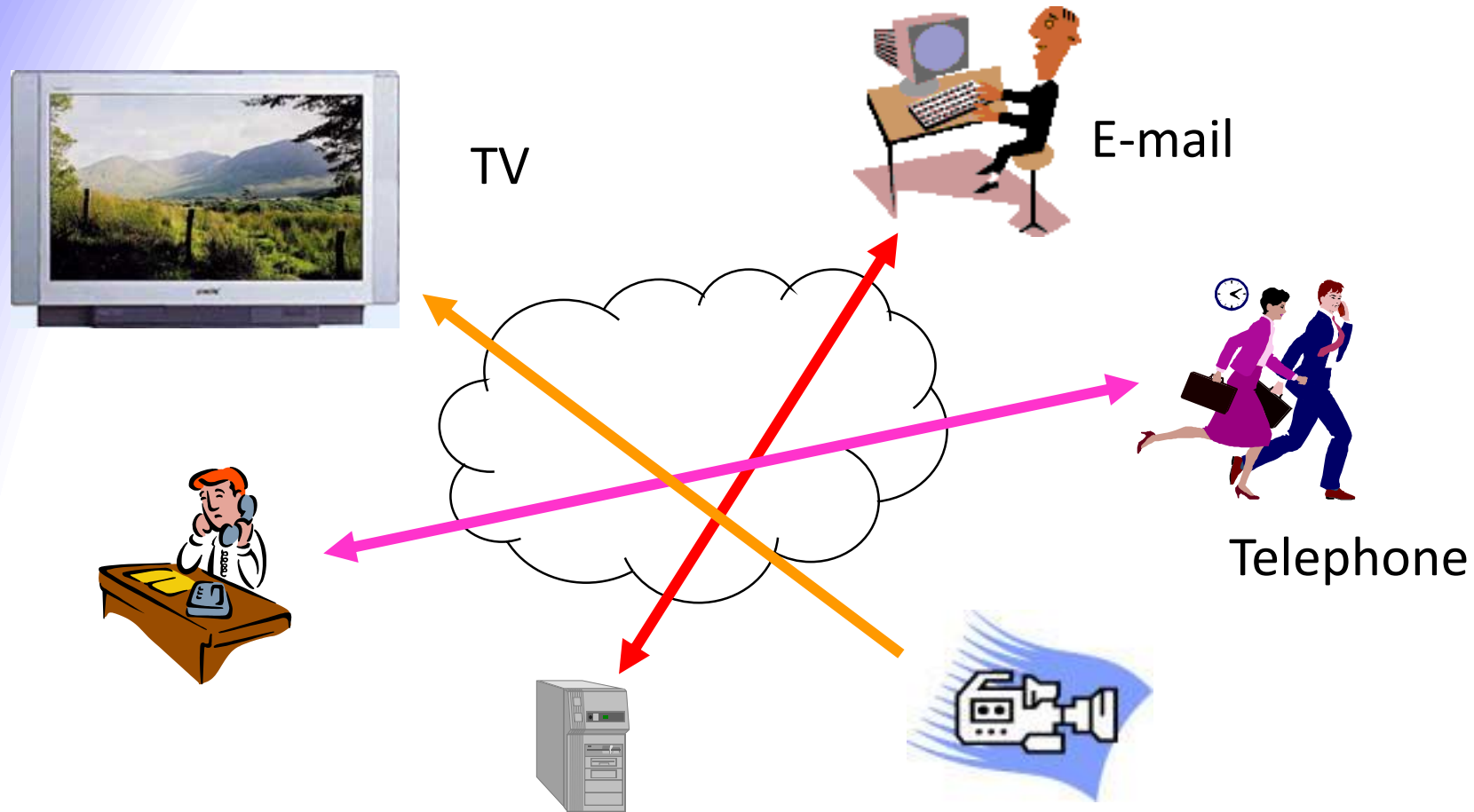
* digital representation
* digital and analog transmission
* asynchronous and synchronous communications
* transmission media

7-Layer OSI Reference Model

Application

Application

End-to-End Protocols

Application Layer	Application Layer
Presentation Layer	Presentation Layer
Session Layer	Session Layer
Transport Layer	Transport Layer
Network Layer	Network Layer	Network Layer	Network Layer
Data Link Layer	Data Link Layer	Data Link Layer	Data Link Layer
Physical Layer	Physical Layer	Physical Layer	Physical Layer

Communicating End Systems

Digital transmission enables networks to support many services

TV

E-mail

Telephone

2.1 Digital representation

- Bit: number with value 0 or 1
  - n bits: digital representation for $0, 1, \ldots, 2^n$
  - 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 $\rightarrow$ more content

Block

- Information that occurs in a single block
  - text message
  - data file
  - JPEG image

- 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
  - MP3 audio
  - streaming video (H.261, MPEG-2)

- Bit rate = bits / second (bps)
  - 1 Kbps $= 10^{3}$ bps
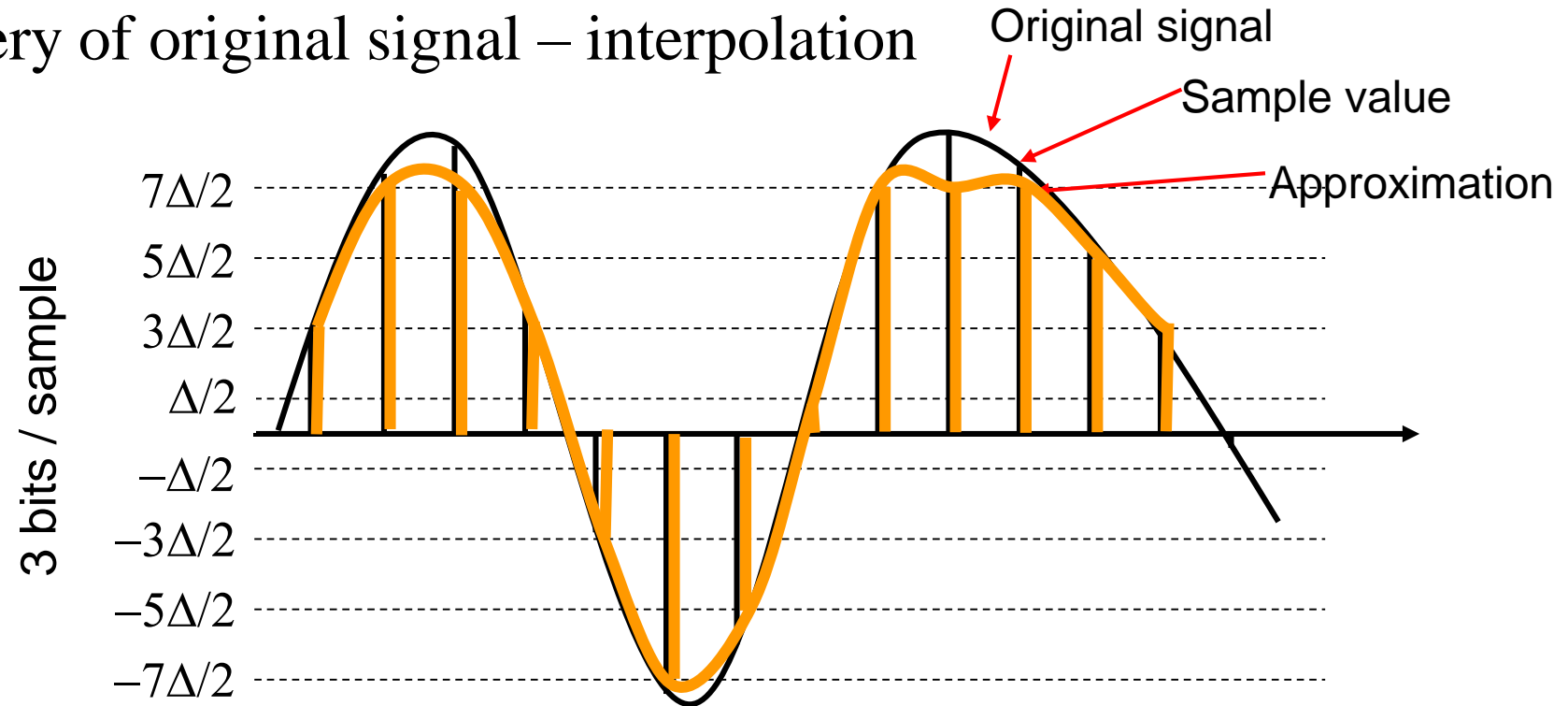 - 1 Mbps $= 10^{6}$ bps
  - 1 Gbps $= 10^{9}$ bps

Stream information

- a real-time voice signal must be digitized & transmitted as it is produced

- analog signal level varies continuously in time

The e s p ee ch s i g n al l e v el   v a r ie s w i th   t   i   m(e)
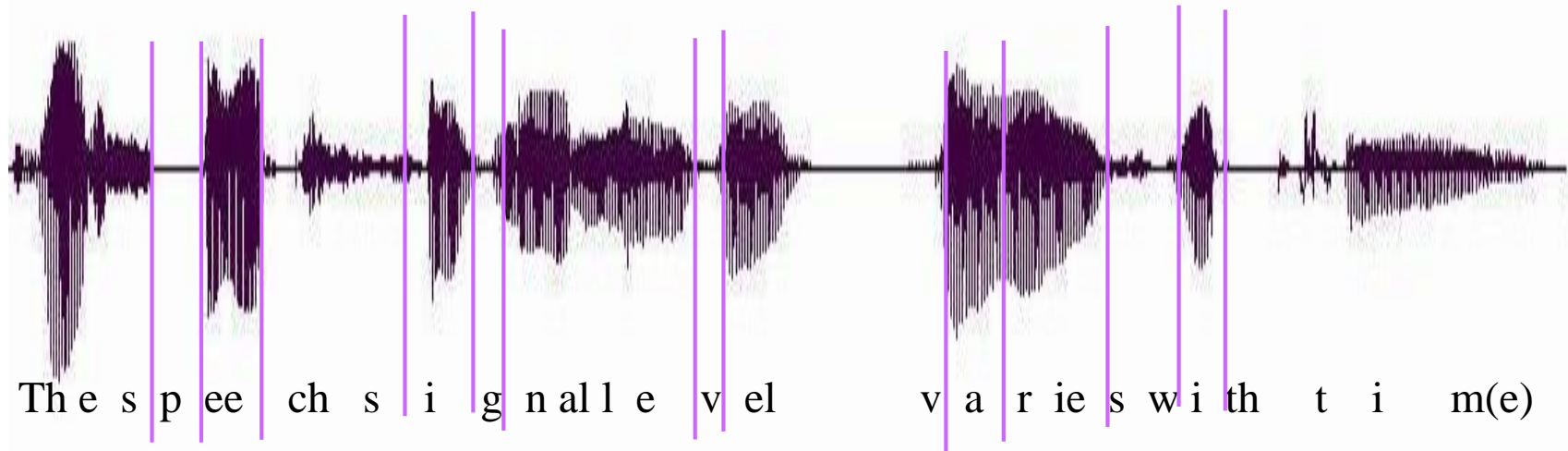
Digitization of analog signal

- sample analog signal in time
- find closest approximation – quantization
- recovery of original signal – interpolation

$$R_s = \text{bit rate} = \text{\# bits/sample x \# samples/second}$$
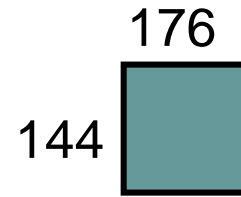
Video signal

- sequence of picture frames
  - each picture digitized & compressed
- frame repetition rate
  - 10-30-60 frames/second (fps), depending on quality
- frame resolution
  - small frames for videoconferencing
  - standard frames for conventional broadcast TV
  - HDTV frames

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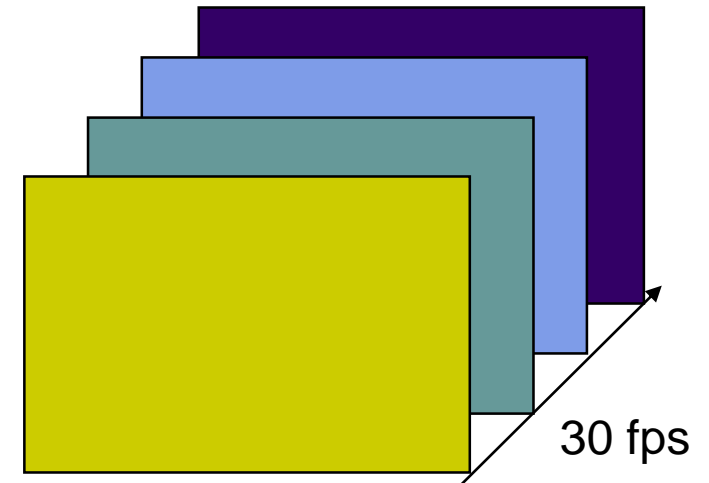
Rate = M bits/pixel **x** (WxH) pixels/frame **x** F frames/second

Video frames

QCIF videoconferencing

176

144

at 30 frames/sec =

760,000 pixels/sec

Broadcast TV

←— 720 —→

480
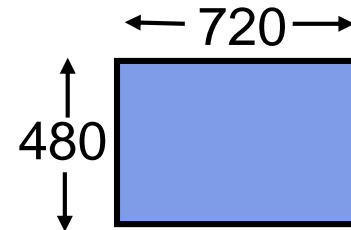
at 30 frames/sec =

10.4×10^6 pixels/sec

HDTV

←——— 1920 ———→

1080
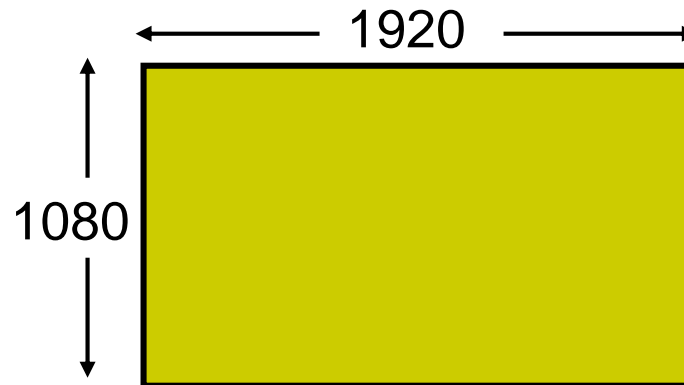
at 30 frames/sec =

67×10^6 pixels/sec

Digital video signals

Type	Method	Format	Original	Compressed
Video Conference	H.261	176x144 or 352x288 pix @10-30 fps	2-36 Mbps	64-1544 Kbps
Full Motion	MPEG-2	720x480 pix @30 fps	249 Mbps	2-6 Mbps
HDTV	MPEG-2	1920x1080 @30 fps	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

2.2 Digital and analog transmission

Transmitter

Communication channel

Receiver

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

Transmitter | **Transmitted Signal** | **Received Signal** | Receiver

Communication channel

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

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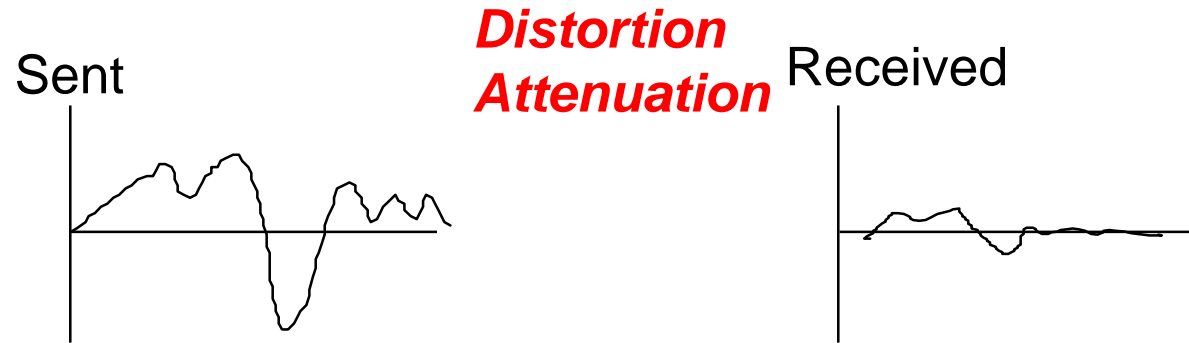- Spurious noise
- Interference from other signals

Analog long-distance communication

Transmission segment

```
┌──────────┐      ┌──────────┐          ┌──────────┐      ┌──────────────┐
│  Source  │──────│ Repeater │── . . . ═│ Repeater │──────│ Destination  │
└──────────┘      └──────────┘          └──────────┘      └──────────────┘
```
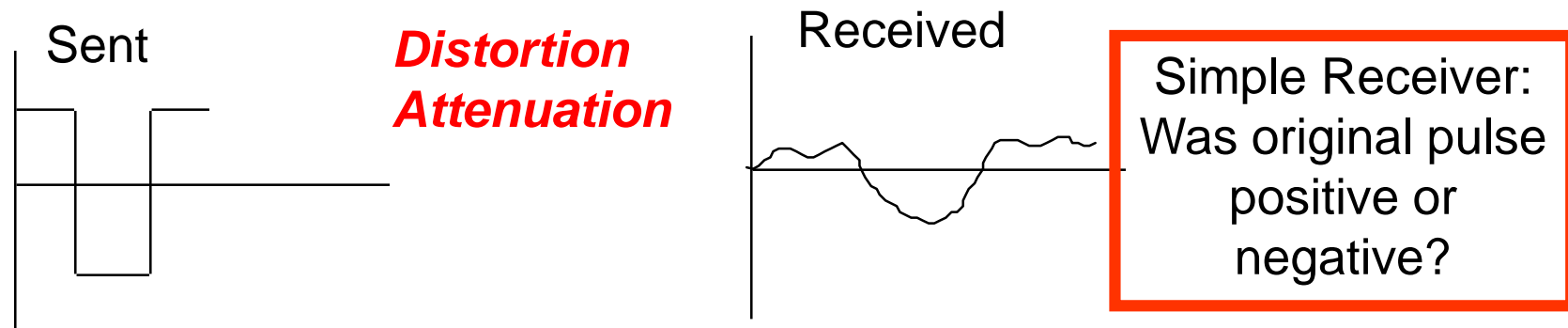
- Each repeater attempts to restore analog signal to its original form
- Restoration is imperfect
  - Distortion is not completely eliminated
  - Noise & interference is only partially removed
- Signal quality decreases with number of repeaters
- Communication is distance-limited
- Analogy: copy a song using a cassette recorder
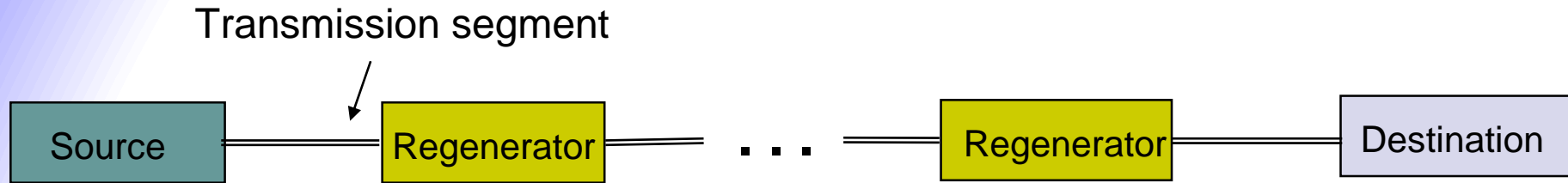
Analog vs. digital transmission

Analog transmission: all details must be reproduced accurately

Sent

Distortion Attenuation Received

Digital transmission: only discrete levels need to be reproduced

Sent

Distortion Attenuation Received

Simple Receiver: Was original pulse positive or negative?

Digital long-distance communication

Transmission segment

```
[Source] —— [Regenerator] —— . . . —— [Regenerator] —— [Destination]
```

- Regenerator recovers original data sequence and retransmits on next segment
- Each regeneration is like the first time!
- Can design a transmission system with very low bit error rate
- Analogy: copy an MP3 file
- Communication is possible over very long distances
- Digital systems vs. analog systems
  - Less power, longer distances, lower system cost
  - Monitoring, multiplexing, coding, encryption, protocols…

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 (PCM): telephone speech
   - CD audio
3. Compression: to lower bit rate further, apply additional compression method
   - Differential coding: cellular telephone speech
   - Subband coding: MP3 audio

Sampling rate and bandwidth

- A signal that varies faster needs to be sampled more frequently

- *Bandwidth* measures how fast a signal varies
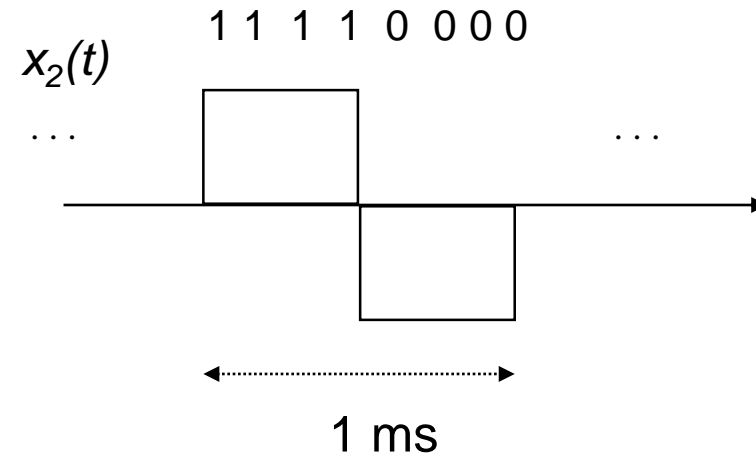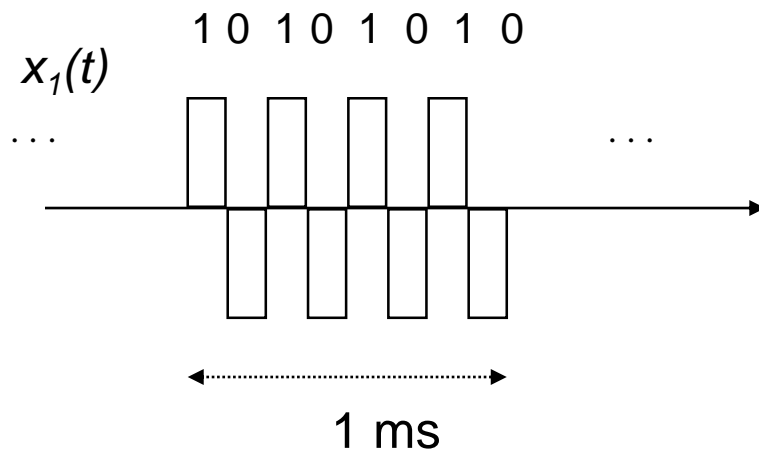
- What is the bandwidth of a signal?
- How is bandwidth related to sampling rate?

Periodic signals

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

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

"DC"
long-term
average

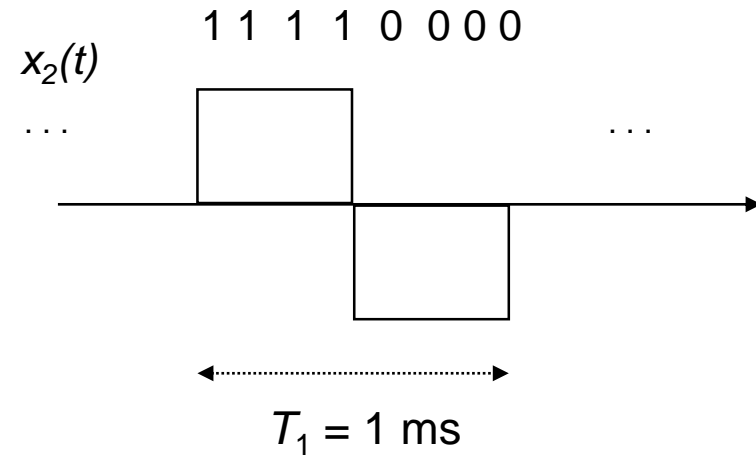fundamental
frequency $f_0 = 1/T$
first harmonic

*k*th harmonic

- $|a_k|$ determines amount of power in *k*th harmonic

- Amplitude spectrum $|a_0|$, $|a_1|$, $|a_2|$, …

Example - Fourier series

1 0 1 0 1 0 1 0

$x_1(t)$

...   ...

t

$T_2 = 0.25$ ms

1 1 1 1 0 0 0 0

$x_2(t)$

...   ...

t

$T_1 = 1$ ms
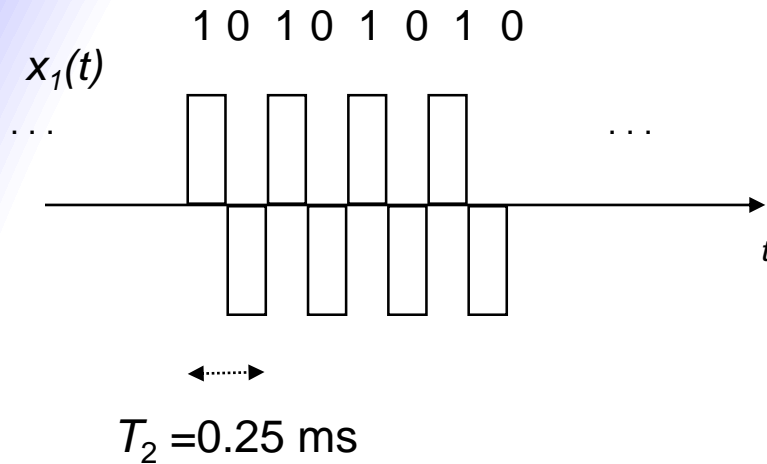
$$x_1(t) = 0 + \frac{4}{\pi}\cos(2\pi 4000t)$$

$$+ \frac{4}{3\pi}\cos(2\pi 3(4000)t)$$

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

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

$$+ \frac{4}{3\pi}\cos(2\pi 3(1000)t)$$

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

Only odd harmonics have power

Spectrum and bandwidth

- Spectrum of a signal: magnitude of amplitudes as a function of frequency

- $x_1(t)$ varies faster in time and 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
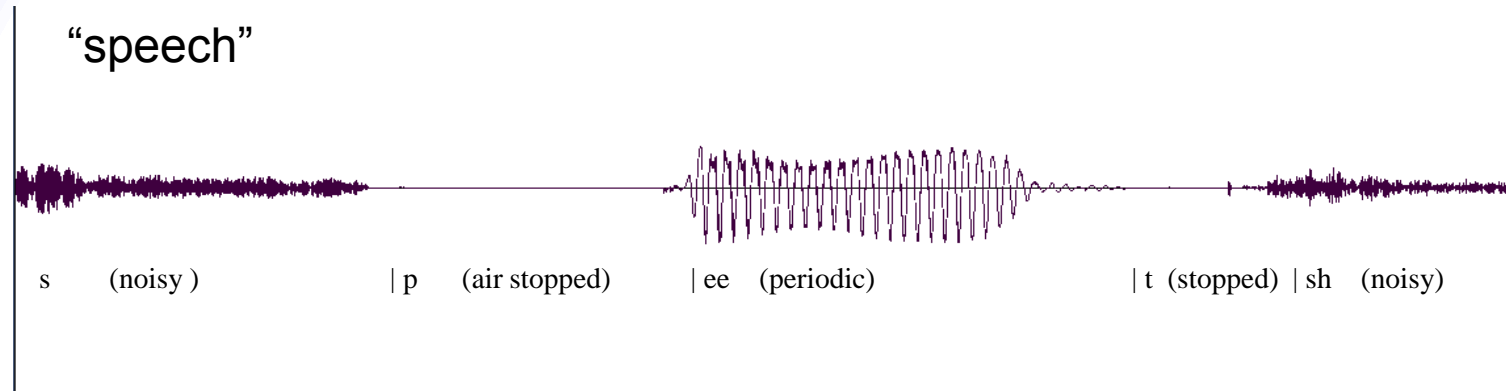
Spectrum of $x_1(t)$
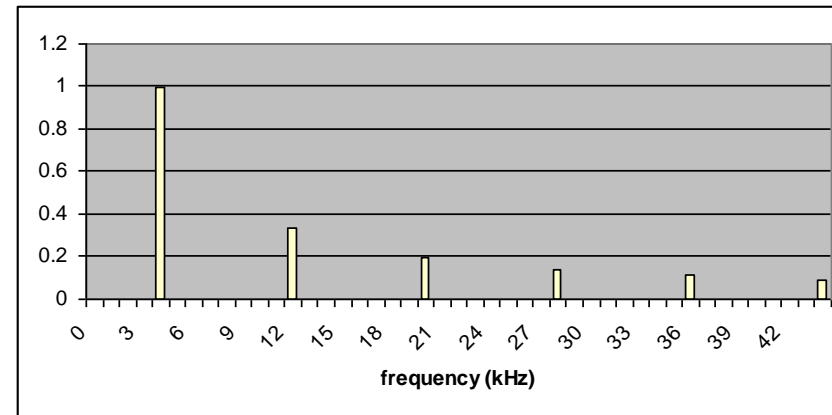
Spectrum of $x_2(t)$
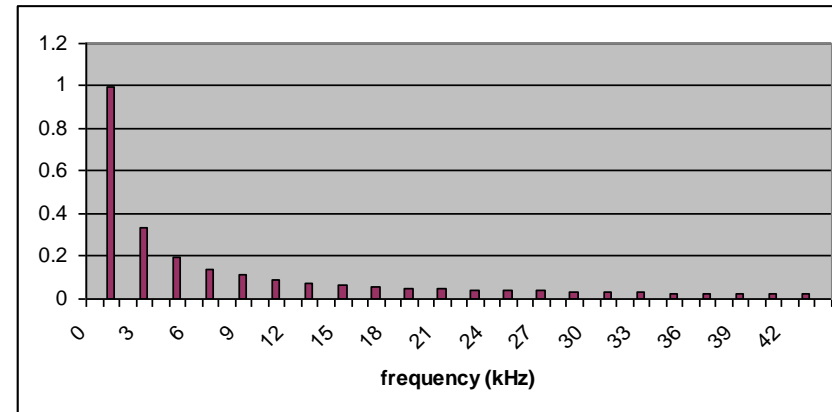
Bandwidth of general signals

"speech"

s   (noisy) | p   (air stopped) | ee   (periodic) | t (stopped) | sh   (noisy)
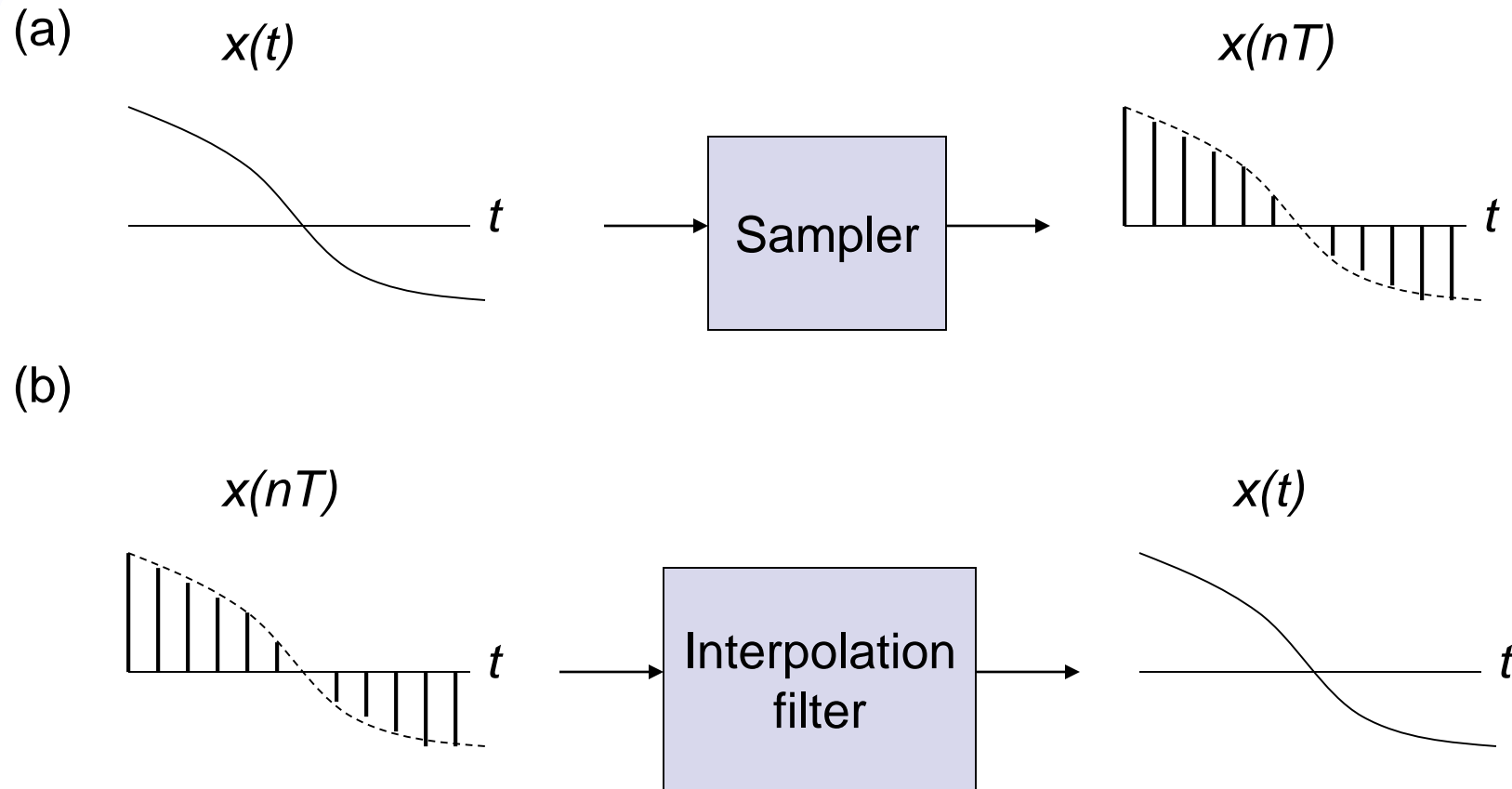
- Not all signals are periodic
  - e.g. voice signal varies according to sound
  - vowels are periodic, "s" is noise-like
- Spectrum of long-term signal
  - averages over many sounds, many speakers
  - involves Fourier transform
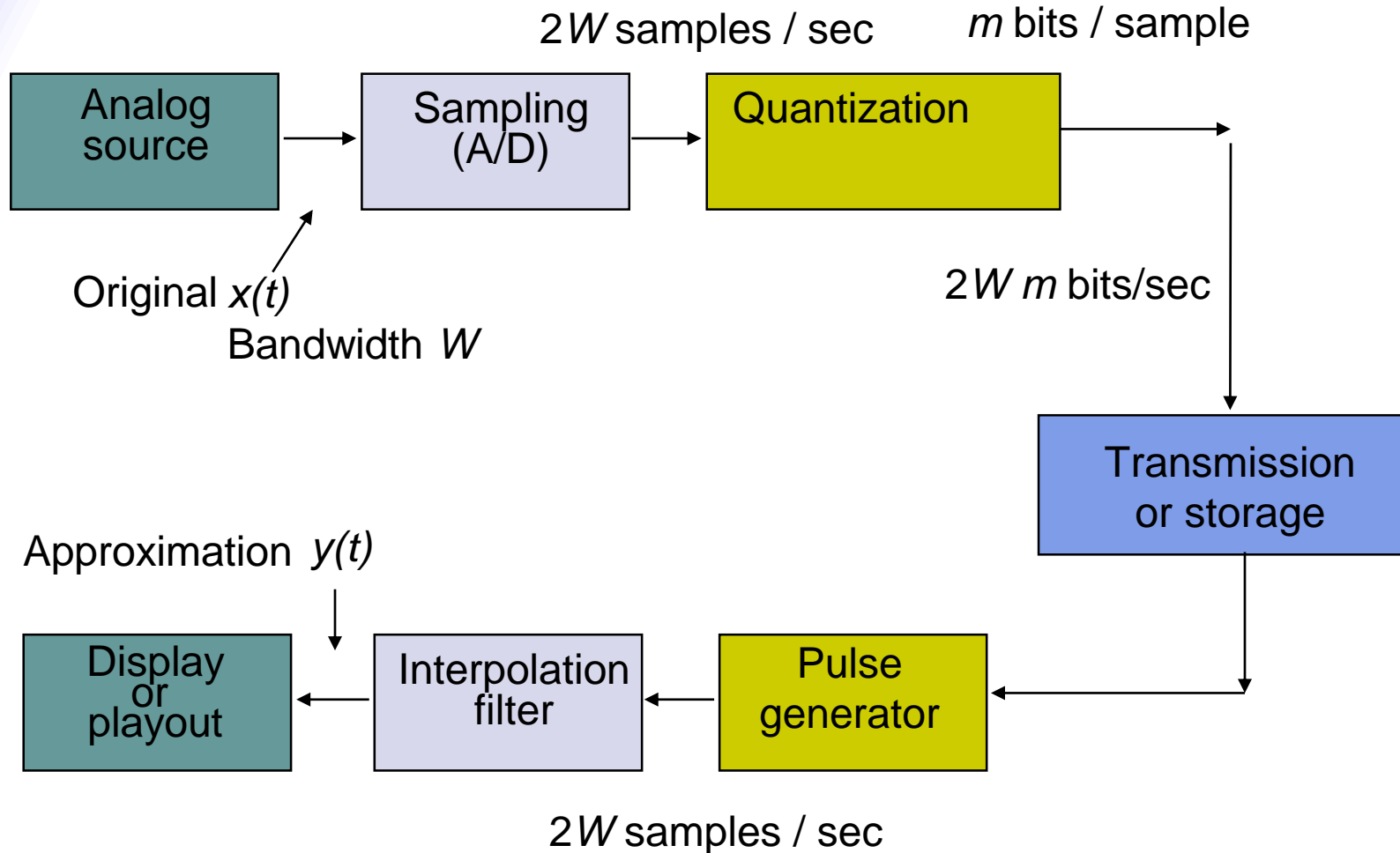- Telephone speech: 4 kHz
- CD Audio: 22 kHz

$X(f)$

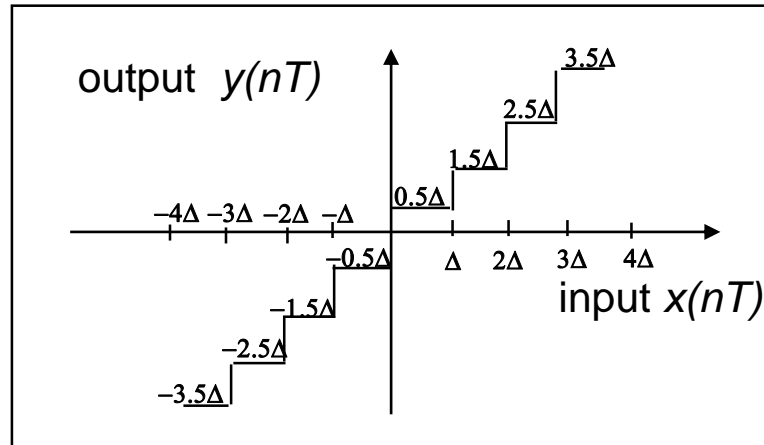0 W_s f
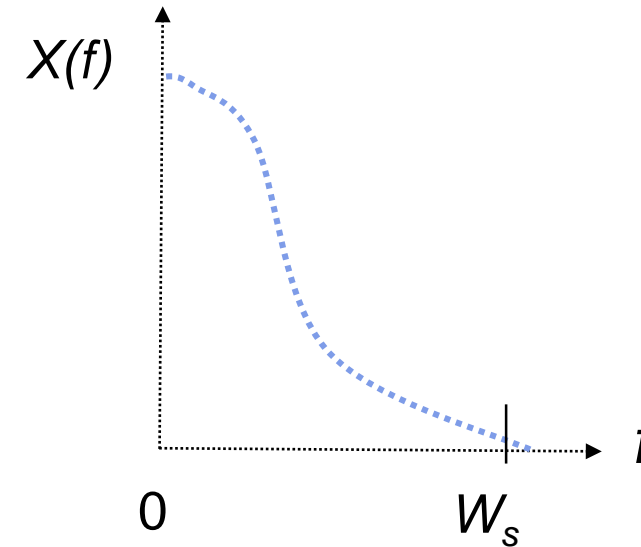
Sampling theorem

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

(a)

$x(t)$

t

$\rightarrow$ Sampler $\rightarrow$

$x(nT)$

t

(b)

$x(nT)$

t

$\rightarrow$ Interpolation filter $\rightarrow$

$x(t)$

t

Digital transmission of analog information

2W samples / sec m bits / sample

| Analog source | | Sampling (A/D) | | Quantization |

Original $x(t)$

Bandwidth W

2W m bits/sec

Transmission or storage

Approximation $y(t)$

| Display or playout | | Interpolation filter | | Pulse generator |

2W samples / sec

Quantization of analog samples

output *y(nT)*

3.5Δ
2.5Δ
1.5Δ
0.5Δ
−4Δ −3Δ −2Δ −Δ
−0.5Δ
Δ 2Δ 3Δ 4Δ
input *x(nT)*
−1.5Δ
−2.5Δ
−3.5Δ

Quantizer maps input
into closest of 2^m
representation values

Quantization error:
"noise" = $x(nT) - y(nT)$

Original signal
Sample value
Approximation

3 bits / sample

7Δ/2
5Δ/2
3Δ/2
Δ/2
-Δ/2
-3Δ/2
-5Δ/2
-7Δ/2
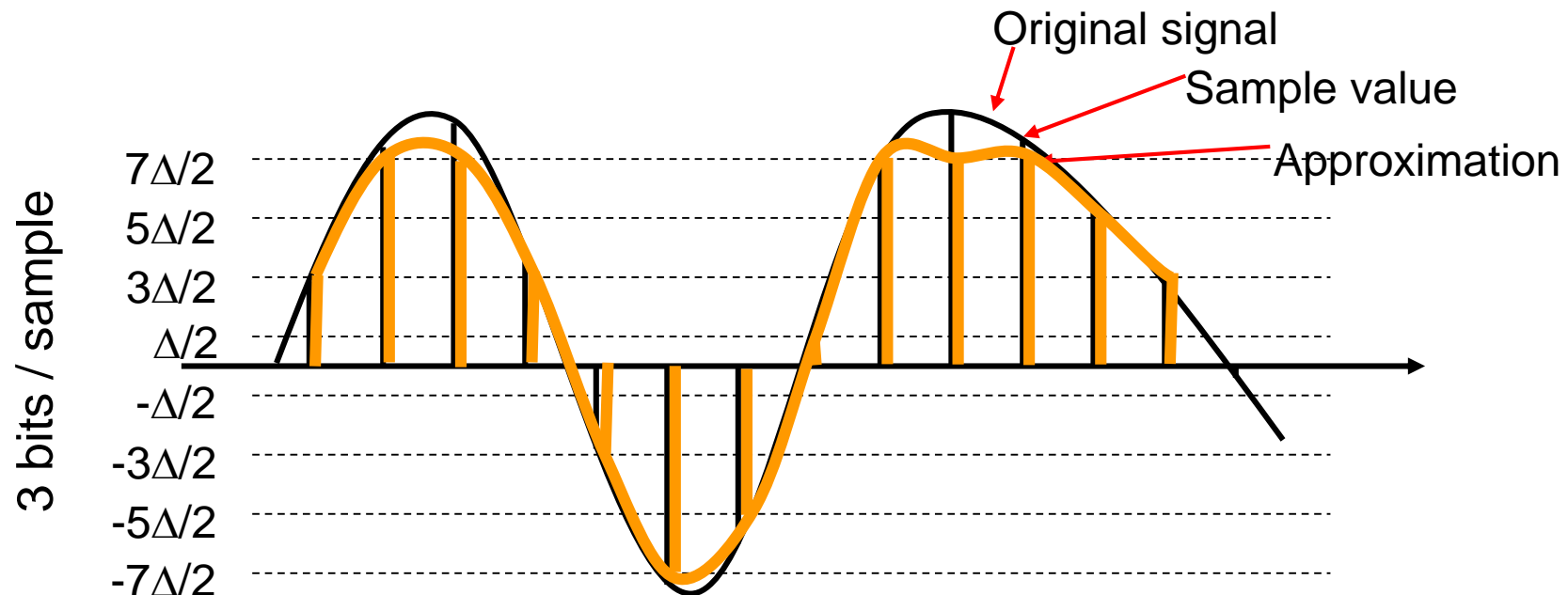
Example: voice and audio

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 KHz → 44,000 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

Quantizer performance

$M = 2^m$ levels,   Dynamic range $(-V, V)$ $\Delta = 2V/M$

error $= x(nT)-y(nT)=e(nT)$

$\frac{\Delta}{2}$

... -2Δ Δ Δ 2Δ 3Δ ...   input

$-V$ $-\frac{\Delta}{2}$ V $x(nT)$
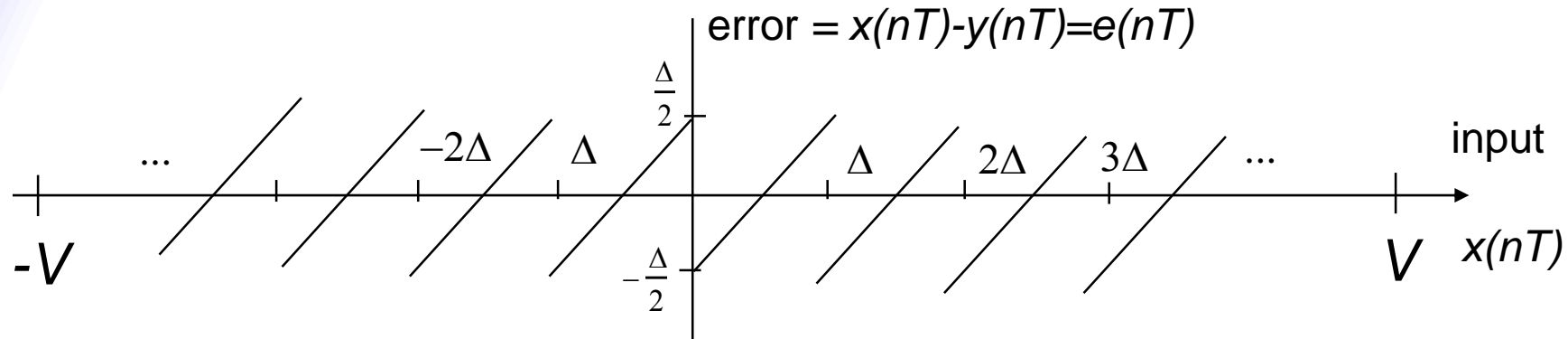
If the number of levels M is large, then the error is approximately uniformly distributed between $(-\Delta/2, \Delta 2)$

Average Noise Power = Mean Square Error:

$$\sigma_e^2 = \int_{-\frac{\Delta}{2}}^{\frac{\Delta}{2}} x^2 \, \frac{1}{\Delta} \, dx = \frac{\Delta^2}{12}$$

Figure of merit:

Signal-to-Noise Ratio (SNR) = average signal power / average noise power

Let σ_x^2 be the signal power, then

$$SNR = \frac{\sigma_x^2}{\Delta^2/12} = \frac{12\sigma_x^2}{4V^2/M^2} = 3\left(\frac{\sigma_x}{V}\right)^2 M^2 = 3\left(\frac{\sigma_x}{V}\right)^2 2^{2m}$$

The ratio $V/\sigma_x \approx 4$

The SNR is usually stated in decibels:

SNR dB $= 10 \log_{10} \sigma_x^2/\sigma_e^2 = 6m + 10 \log_{10} 3\sigma_x^2/V^2$

SNR dB $= 6m - 7.27$ dB   for $V/\sigma_x = 4$

Example: telephone speech

$W = 4$ KHz, so Nyquist sampling theorem

$\Rightarrow$ 2W = 8000 samples/second

Suppose error requirement $=$ 1% error

SNR = 10 log(1/.01)^2 = 40 dB

Assume $V/\sigma_x = 4$, then

40 dB = $6m - 7$

$\Rightarrow$ $m = 8$ bits/sample

PCM telephone speech:

Bit rate= 8000 x 8 bits/sec= 64 Kbps

Communication channel

- 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

Transmitter   **Transmitted
Signal**

**Received
Signal**   Receiver

Communication 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*

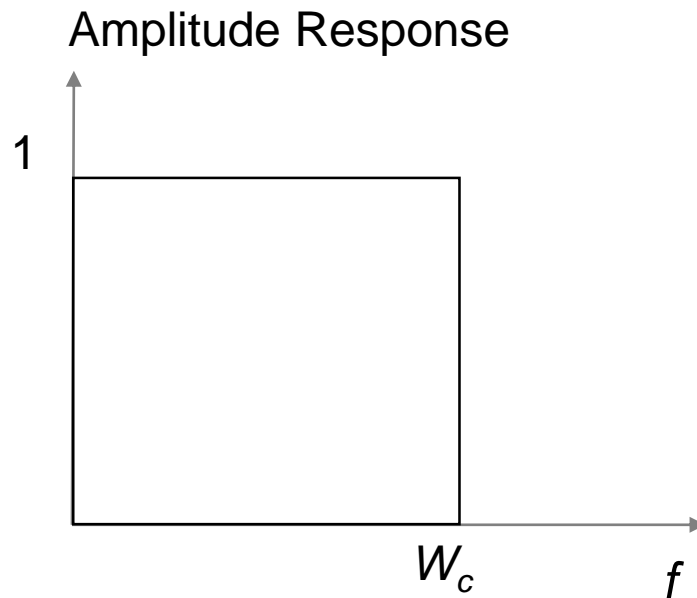# Frequency domain channel characterization

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

Channel

$$A(f) = \frac{A_{out}}{A_{in}}$$
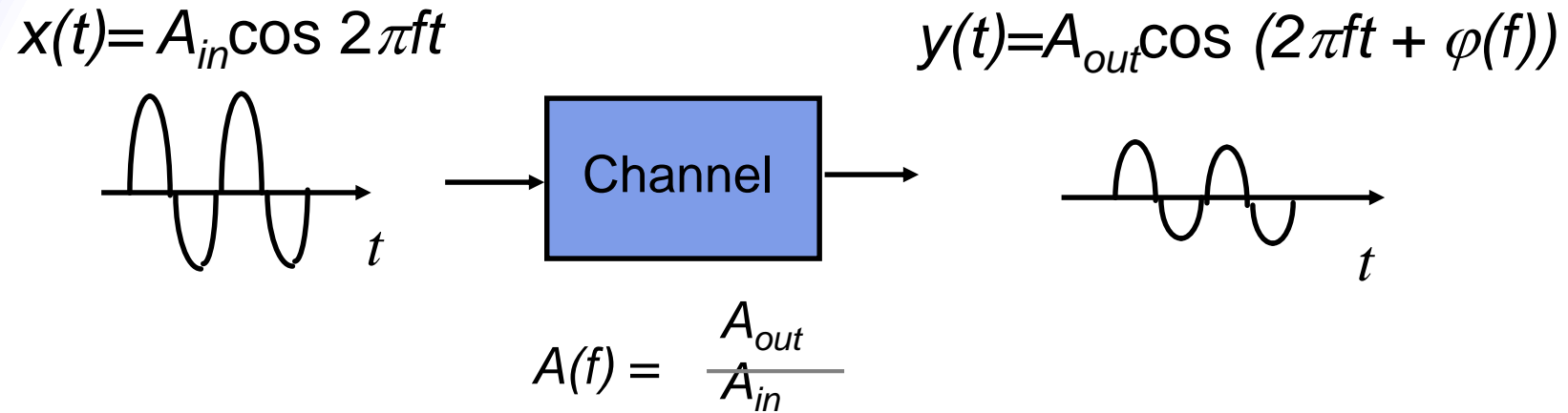
- Apply sinusoidal input at frequency f
  - Output is sinusoid at same frequency, but attenuated & phase-shifted
  - Measure amplitude of output sinusoid (of same frequency f)
  - Calculate amplitude response
    - $A(f)$ = ratio of output amplitude to input amplitude
  - If $A(f) \approx 1$, then input 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

$\varphi(f) = -2\pi ft$

1

$1/2\pi$

0

W_c

f

f
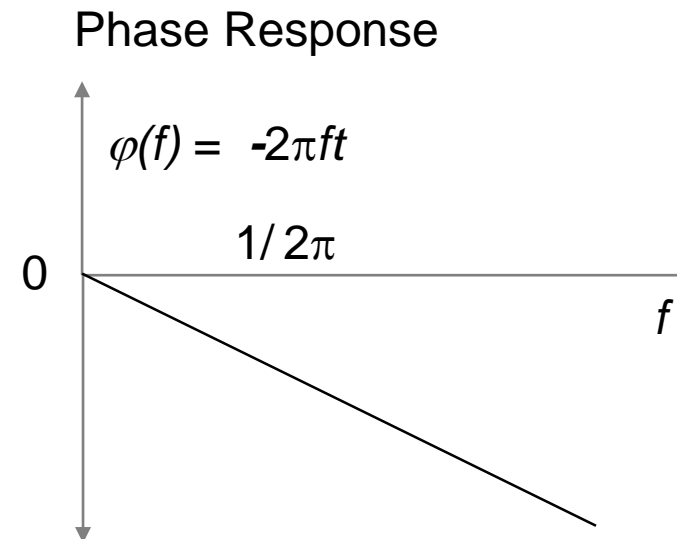
Time-domain characterization
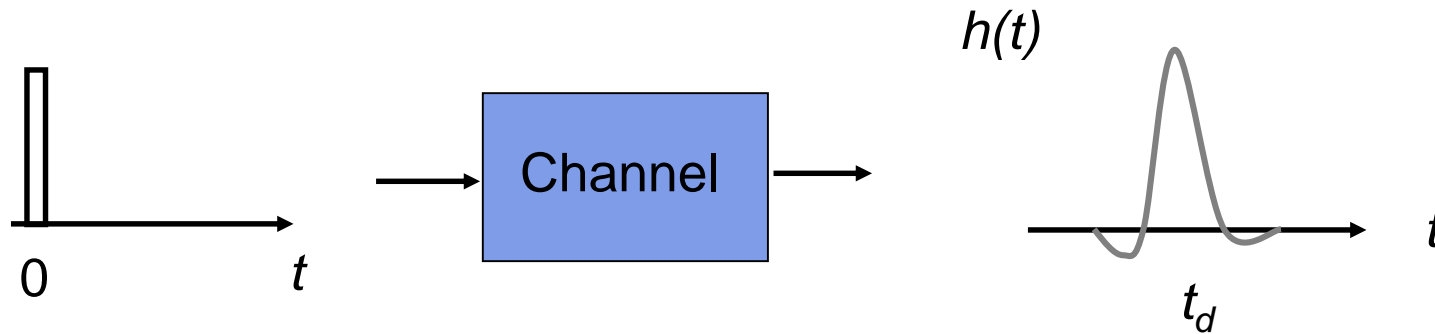
- Time-domain characterization of a channel requires finding the *impulse response h(t)*

- 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

Nyquist pulse with zero InterSymbol Interference (ISI)

- For channel with ideal low pass amplitude response of bandwidth W_c, the impulse response is a Nyquist pulse $h(t)=s(t-\tau)$, where $T=1/2W_c$, and

$$s(t) = \sin(2\pi W_c\, t)/2\pi W_c t$$
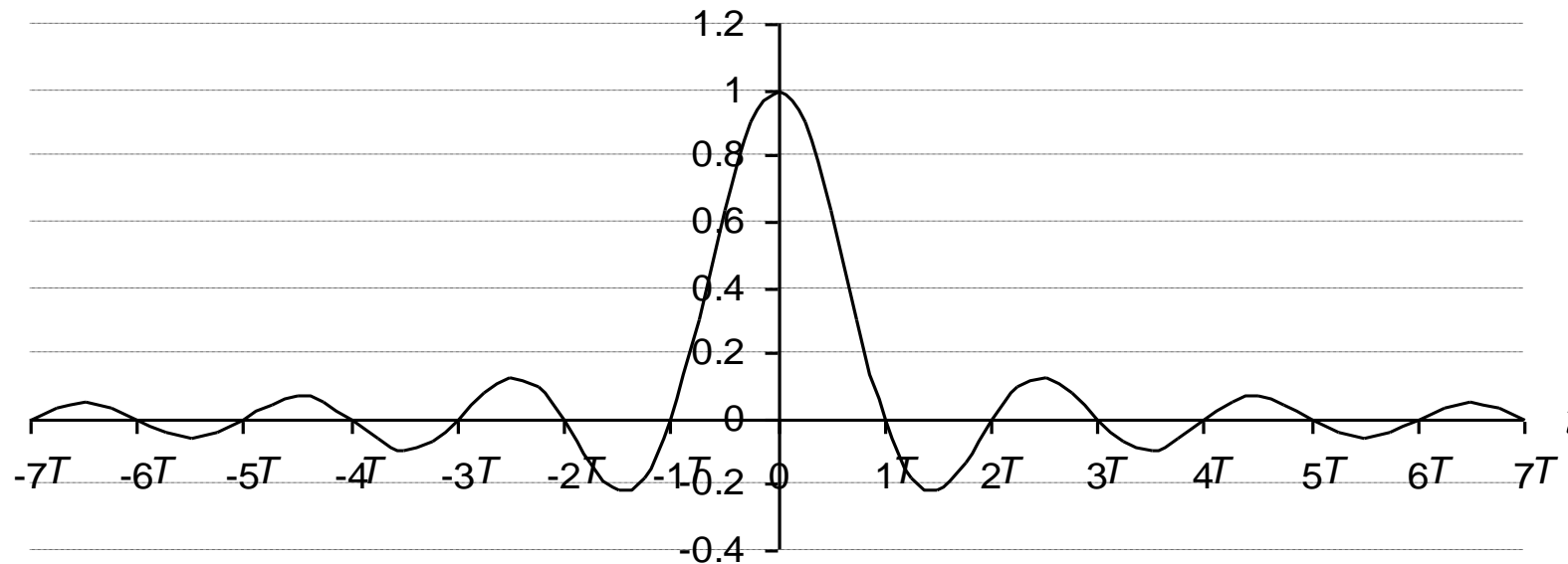
- $s(t)$ has zero crossings at $t = kT,\ k = \pm1, \pm2, \ldots$
- Pulses can be packed every T seconds with *zero interference*

Example - composite waveform

Three Nyquist pulses
shown separately

- $+ s(t)$
- $+ s(t-T)$
- $- s(t-2T)$

Composite waveform

$r(t) = s(t)+s(t-T)-s(t-2T)$
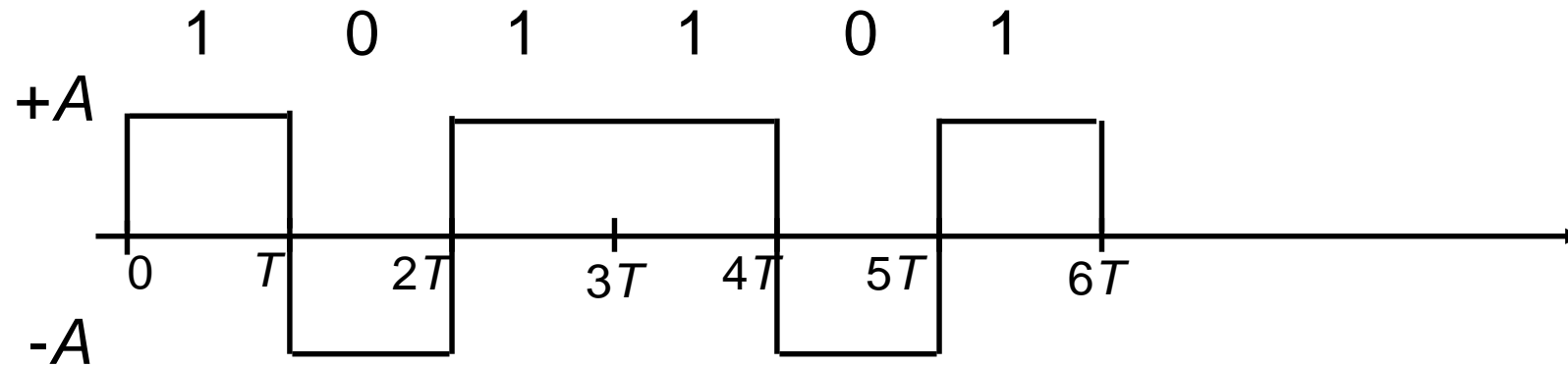
Samples at kT

$r(0)=s(0)+s(-T)-s(-2T)=+1$

$r(T)=s(T)+s(0)-s(-T)=+1$

$r(2T)=s(2T)+s(T)-s(0)=-1$

**Zero ISI at sampling
times kT**
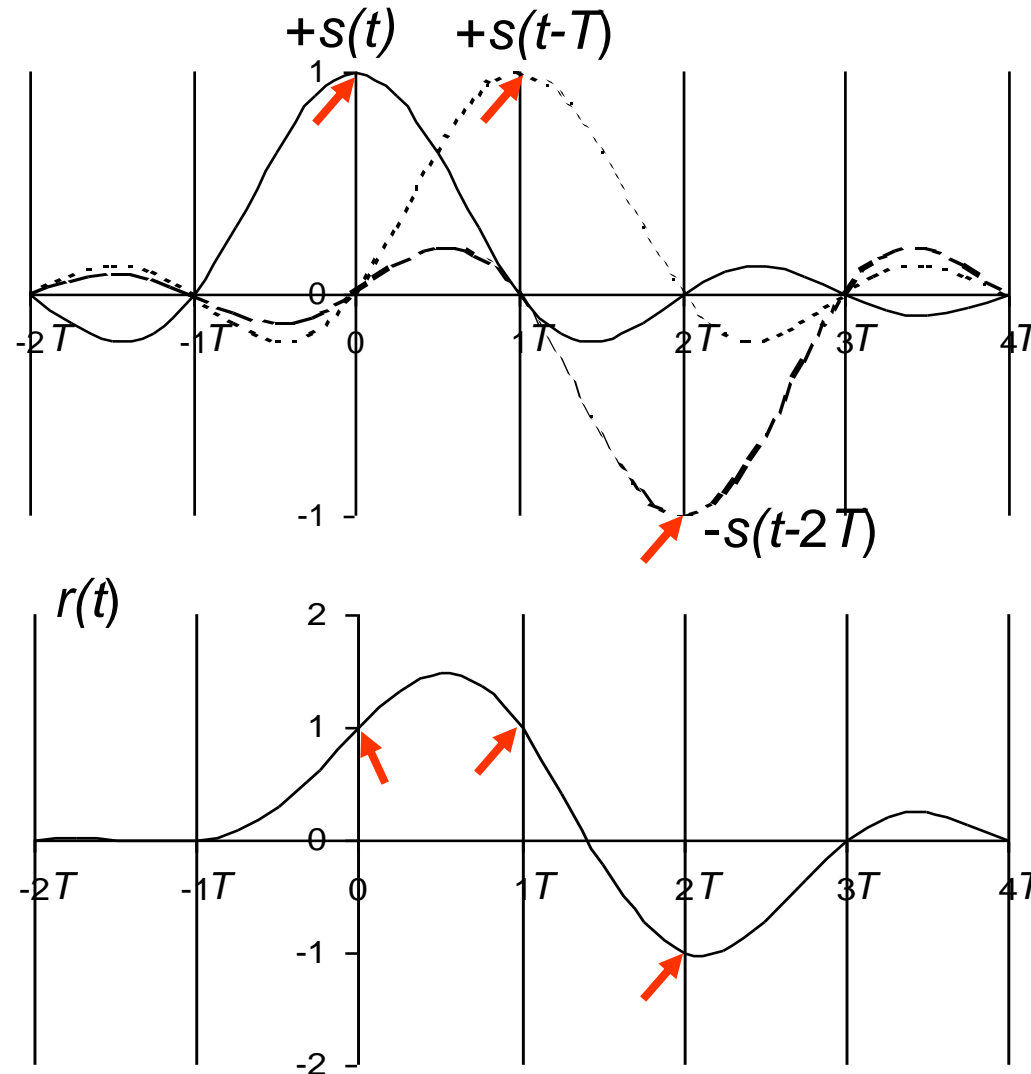
Digital binary signal

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?

Signaling with Nyquist pulses

- *p(t)* pulse at receiver in response to a single input pulse (takes into account pulse shape at input, transmitter & receiver filters, and communications medium)
- *r(t)* waveform that appears in response to sequence of pulses
- If *p(t)* is a Nyquist pulse, then *r(t)* has zero intersymbol interference (ISI) when sampled at multiples of *T*
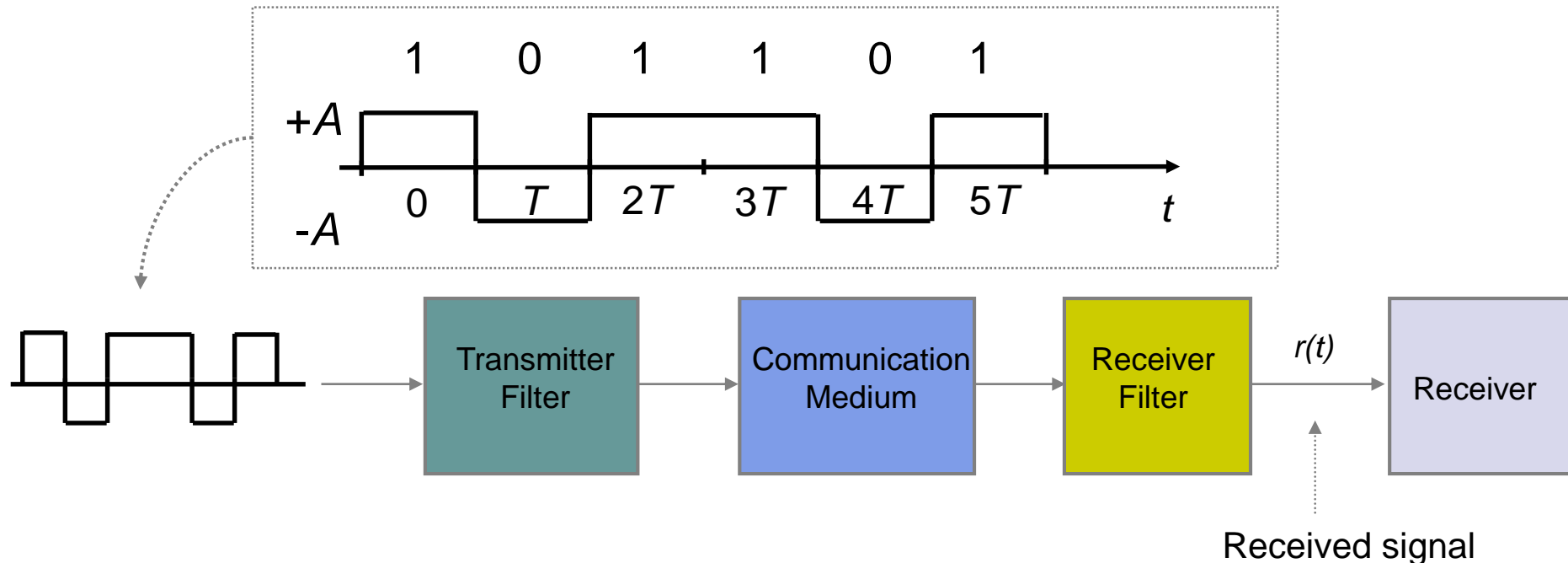
Pulse transmission rate

- Objective: maximize pulse rate through a channel, that is, make T as small as possible
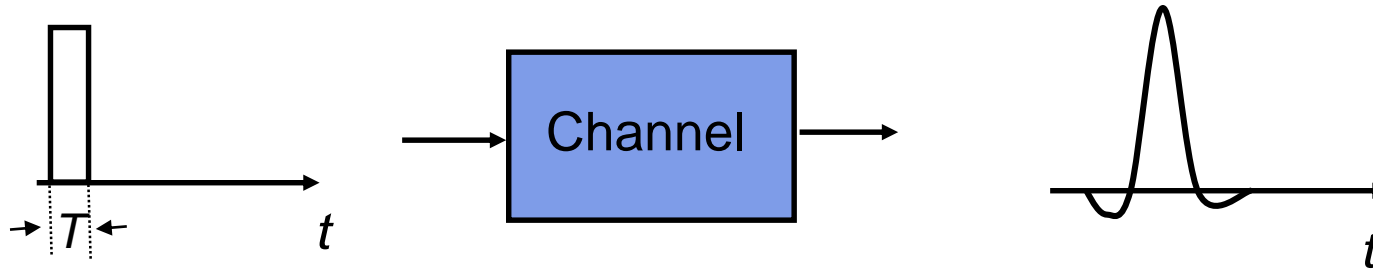
- If input is a narrow pulse, then typical output is a spread-out pulse with ringing

- 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

Multilevel signaling

- Nyquist pulses achieve the maximum signalling rate with zero ISI,
  $2W_c$ pulses per second or
  $2W_c$ pulses / W_c Hz = 2 pulses / Hz

- With two signal levels, each pulse carries one bit of information
  Bit rate = $2W_c$ bits/second

- With $M = 2^m$ signal levels, each pulse carries m bits
  Bit rate = $2W_c$ pulses/sec. * m bits/pulse = $2W_c\,m$ bps

- *Bit rate can be increased by increasing number of levels*

- *r(t) includes additive noise, that limits number of levels that can be used reliably.*

Example

- Four levels {-1, -1/3, 1/3, +1} for {00,01,10,11}
- Waveform for 11,10,01 sends +1, +1/3, -1/3
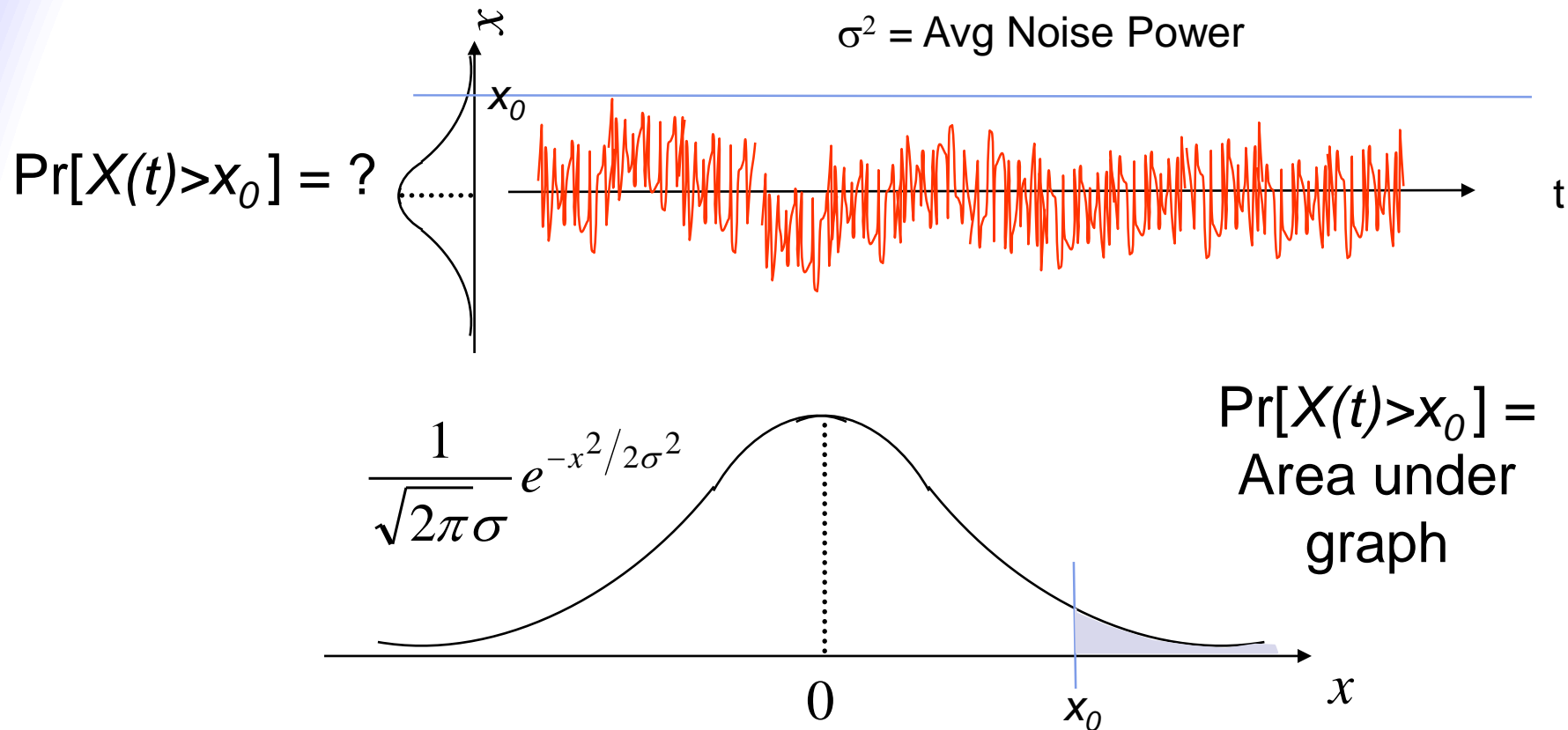- Zero ISI at sampling instants

Composite waveform
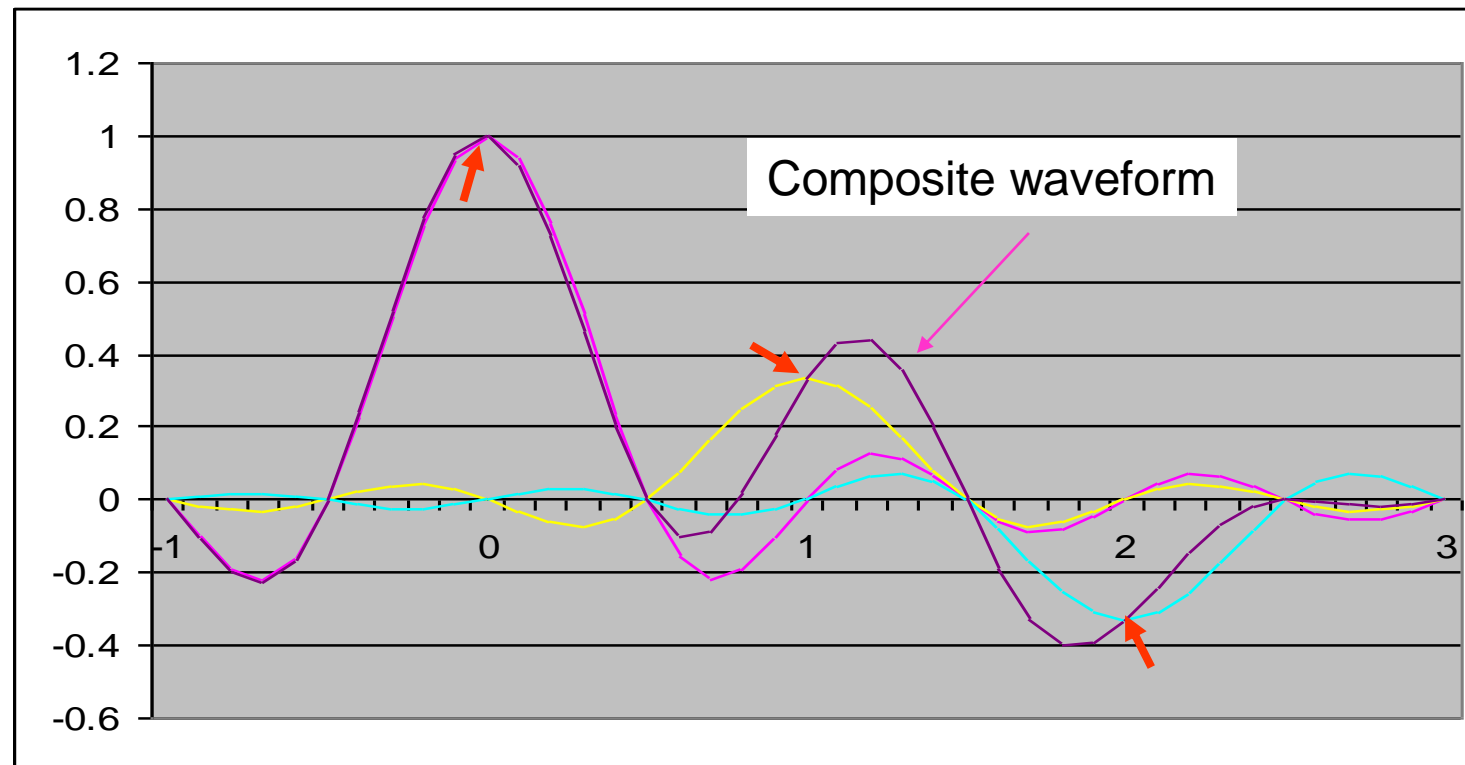
Noise limits accuracy

- Receiver makes decision based on transmitted pulse level + noise
- Error rate depends on relative value of noise amplitude and spacing between signal levels
- Large (positive or negative) noise values can cause wrong decision
- Noise level below impacts 8-level signaling more than 4-level signaling

Four signal levels

Typical noise

Eight signal levels
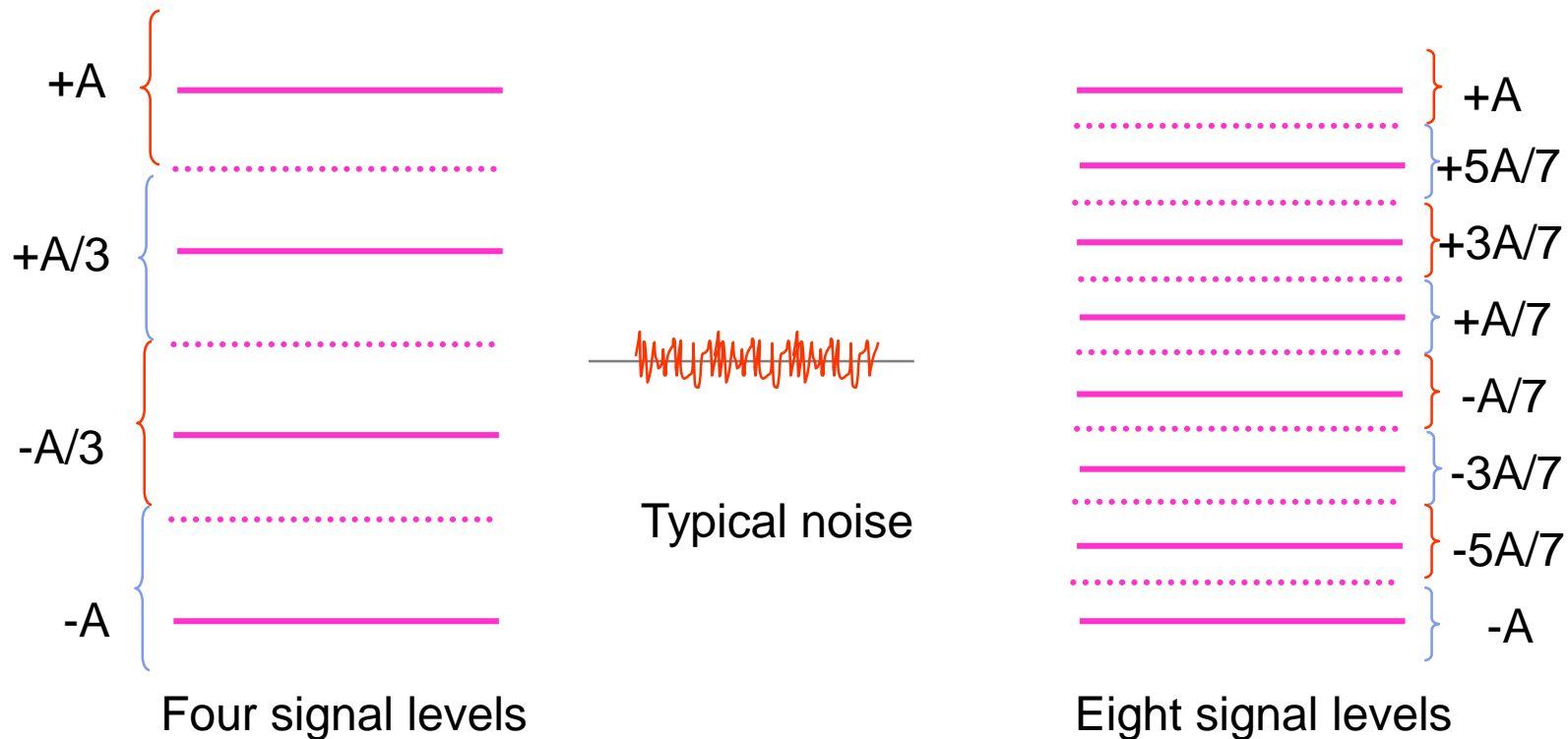
Noise distribution

- Noise is characterized by probability density of amplitude samples
- Likelihood that certain amplitude occurs
- Thermal electronic noise is inevitable (due to vibrations of electrons)
- Noise distribution is Gaussian (bell-shaped) as below

σ^2 = Avg Noise Power

x

x_0

$Pr[X(t)>x_0] = ?$

t

$\frac{1}{\sqrt{2\pi}\sigma}e^{-x^2/2\sigma^2}$

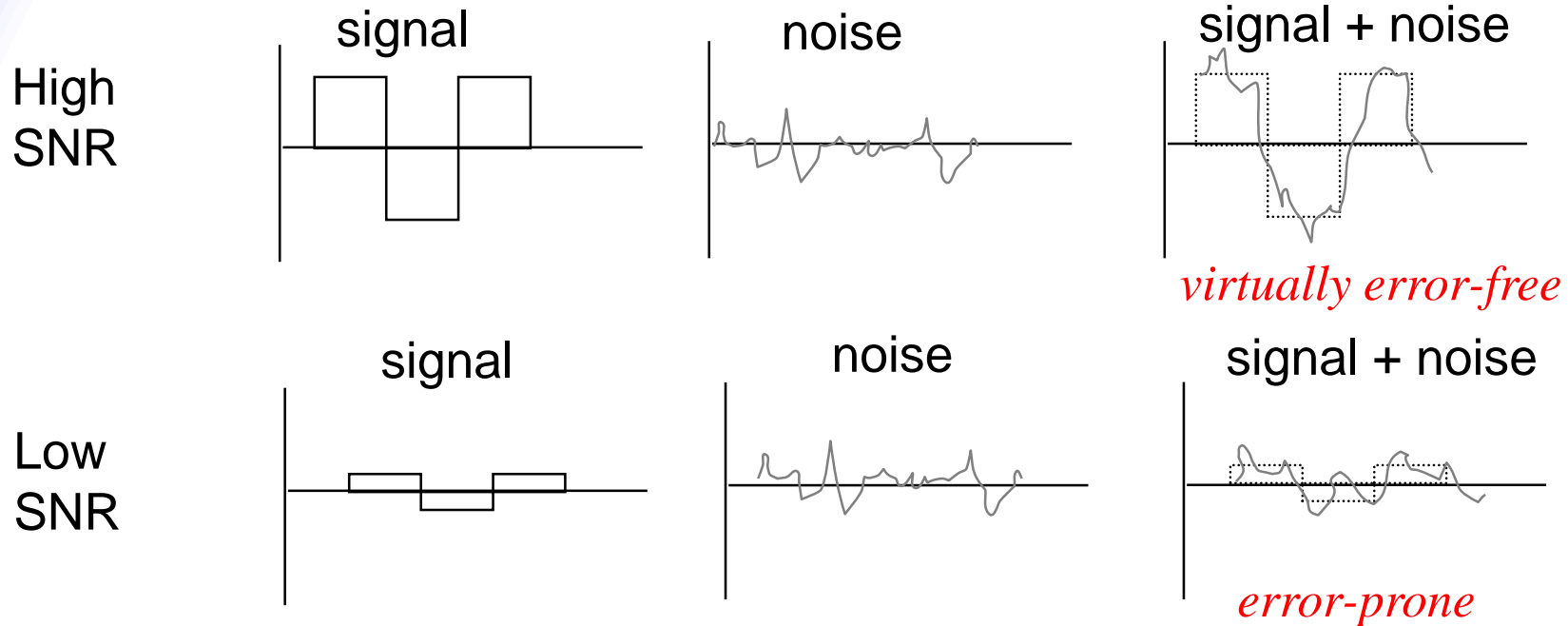$Pr[X(t)>x_0] =$ Area under graph

0

x_0

x

Probability of error

- Error occurs if noise value exceeds certain magnitude
- Probability of large values drops quickly with Gaussian noise
- Target probability of error achieved by designing system so separation between signal levels is appropriate relative to average noise power
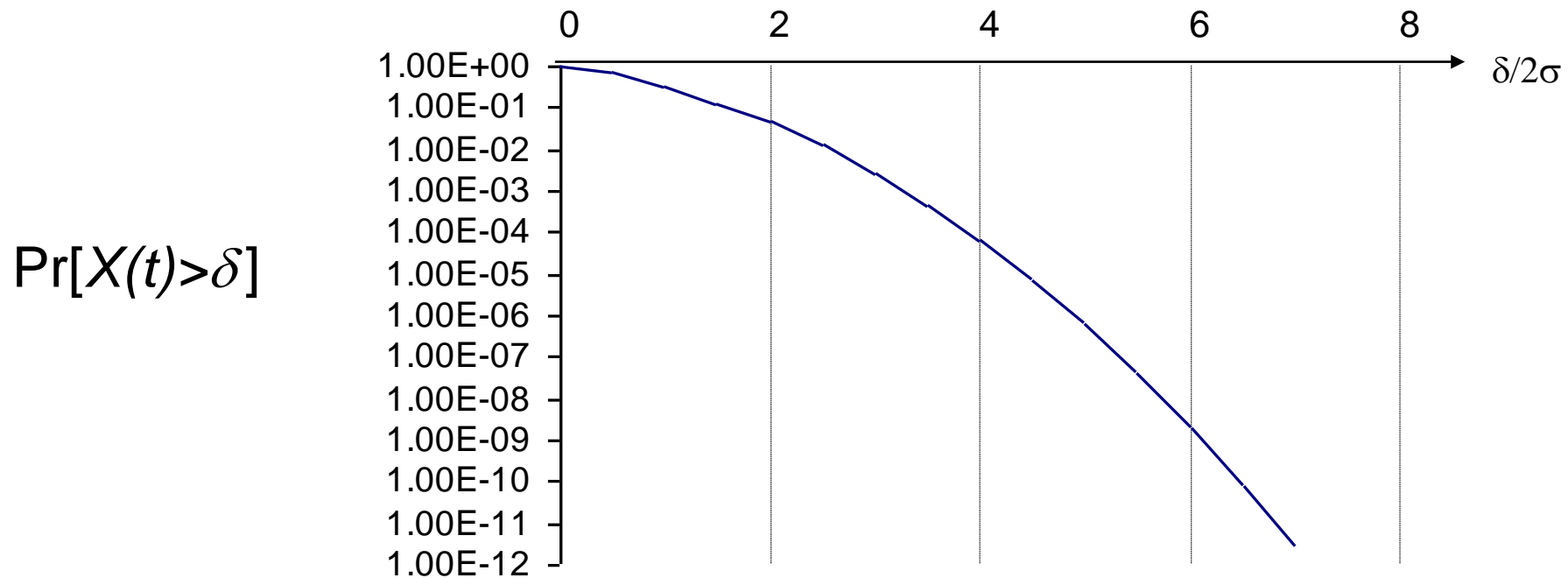
Channel noise affects reliability

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

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

- 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

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

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

Example

- Consider a 3 KHz channel with 8-level signaling. Compare bit rate to channel capacity at 20 dB SNR

- 3KHz telephone channel with 8 level signaling

  Bit rate = 2*3000 pulses/sec * 3 bits/pulse = 18 Kbps

- 20 dB SNR means $10 \log_{10} S/N = 20$

  Implies $S/N = 100$

- Shannon Channel Capacity is then

  $C = 3000 \log (1 + 100) = 19{,}963$ bps

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

Line coding

- Mapping of binary information sequence into the digital signal that enters the channel
  - Example - "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 bandwidth
  - *Low frequency content*: Some channels block low frequencies
    - Waveform should not have low-frequency content
  - *Error detection*: Ability to detect errors helps
  - *Complexity/cost*: Is code implementable in chip at high speed?
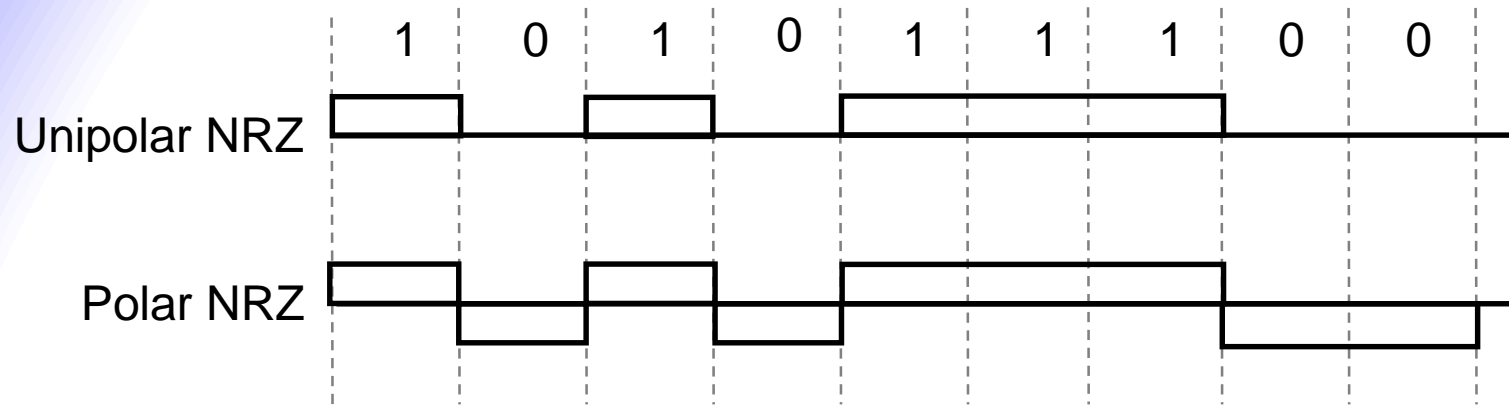
Examples
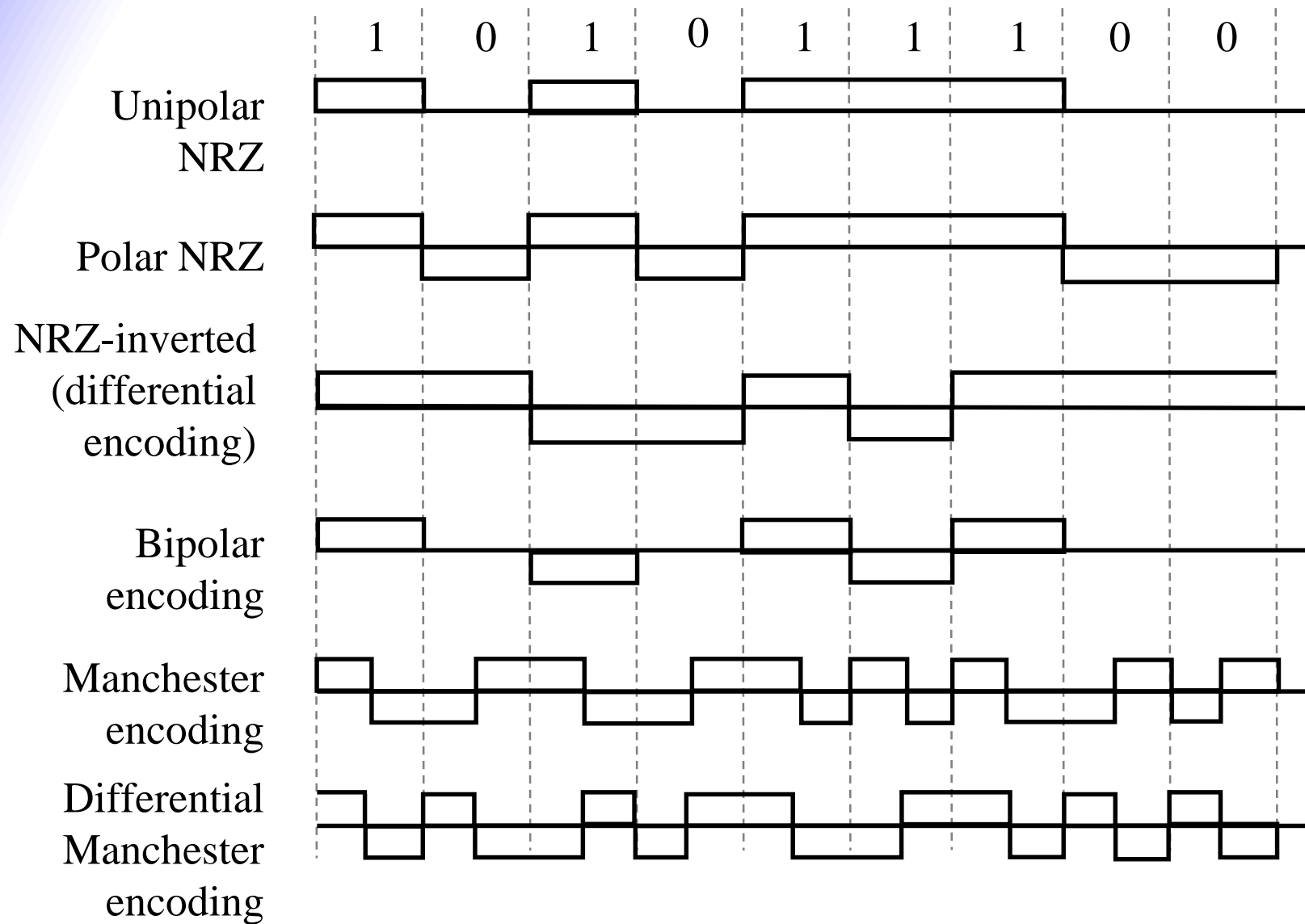
Spectrum of line codes

- Assume 1s & 0s independent & equiprobable
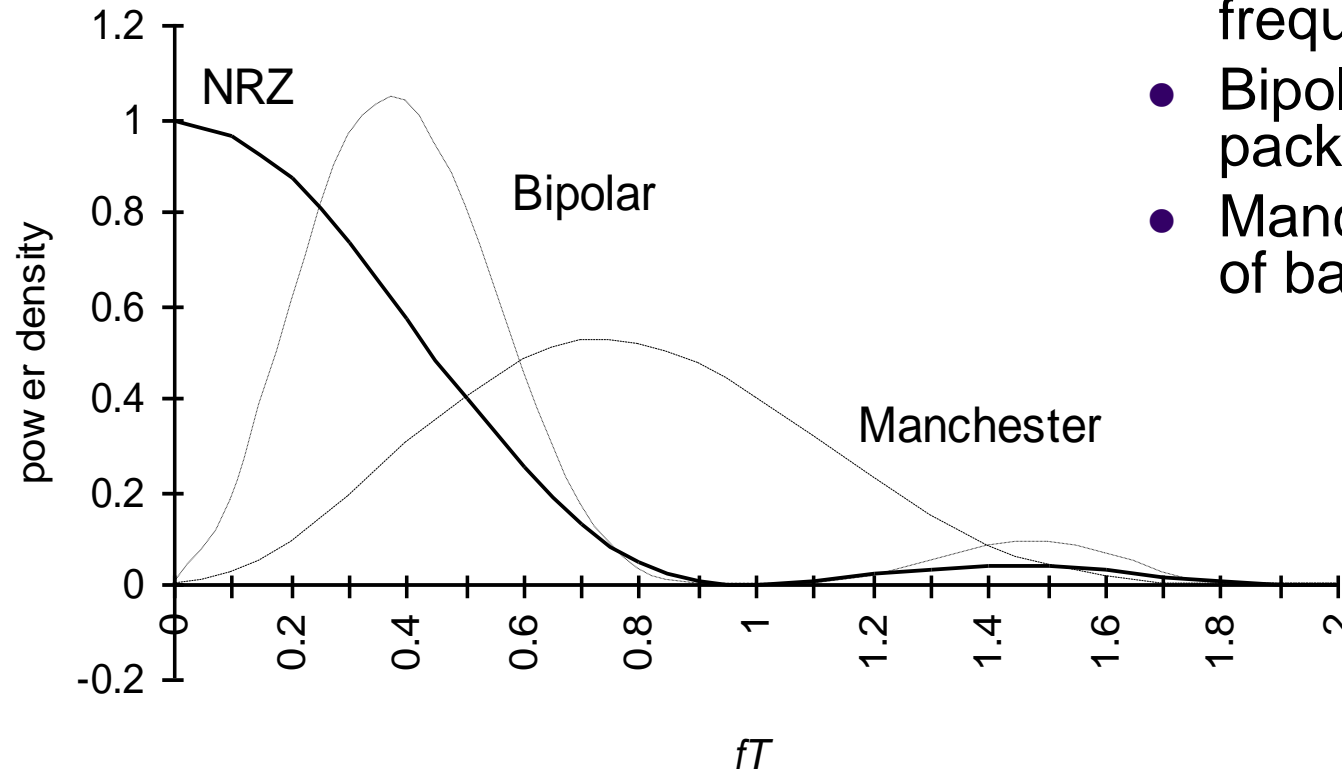
- NRZ has high content at low frequencies
- Bipolar tightly packed around $T/2$
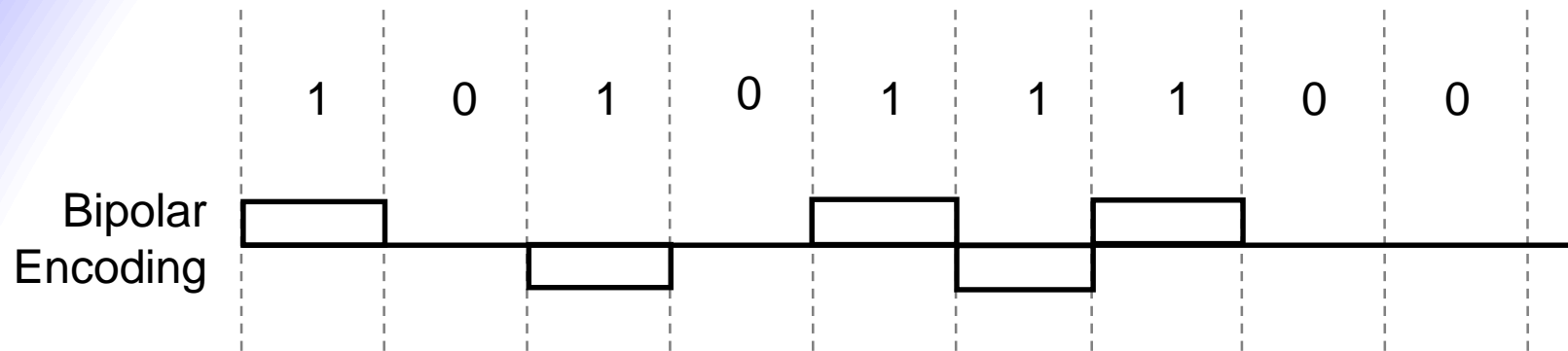- Manchester wasteful of bandwidth

Unipolar 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

	1	0	1	0	1	1	1	0	0

Bipolar
Encoding

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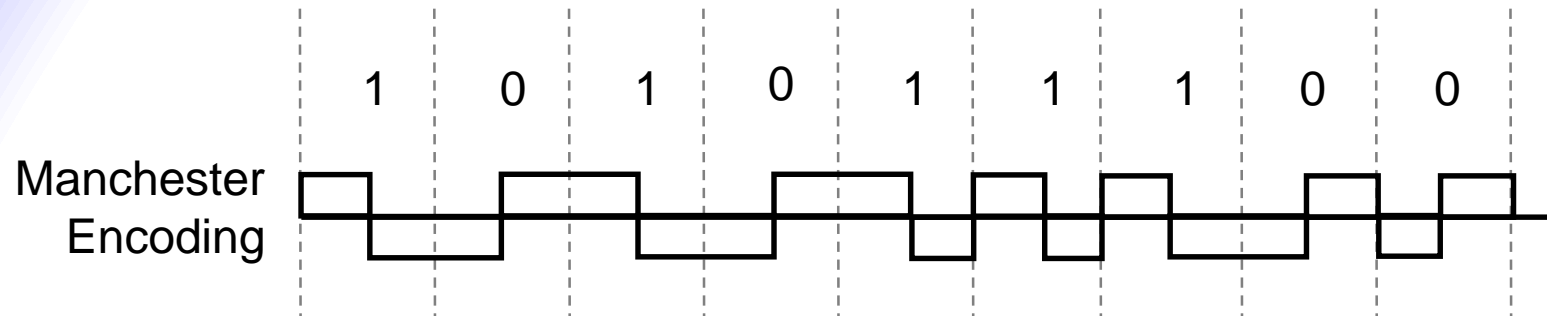
- "1" maps into A/2 first $T/2$, -A/2 last T/2
- "0" maps into -A/2 first $T/2$, A/2 last $T/2$
- Every interval has transition in middle
  - Timing recovery easy
  - Uses double the minimum bandwidth
- Simple to implement
- Used in 10-Mbps Ethernet & other LAN standards

- mBnB 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
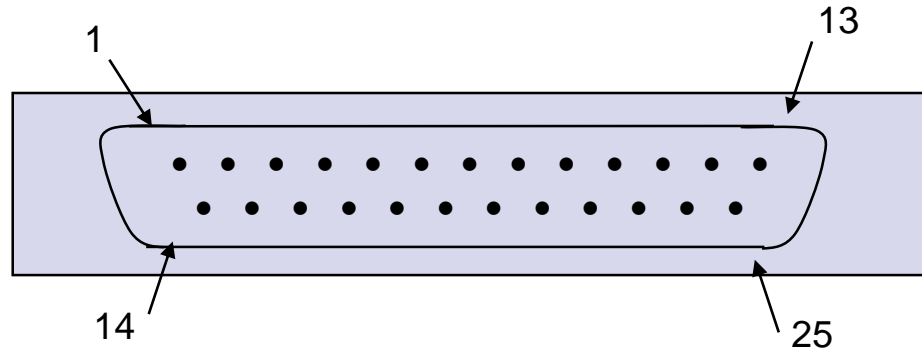- 64B66B code used in 10G Ethernet
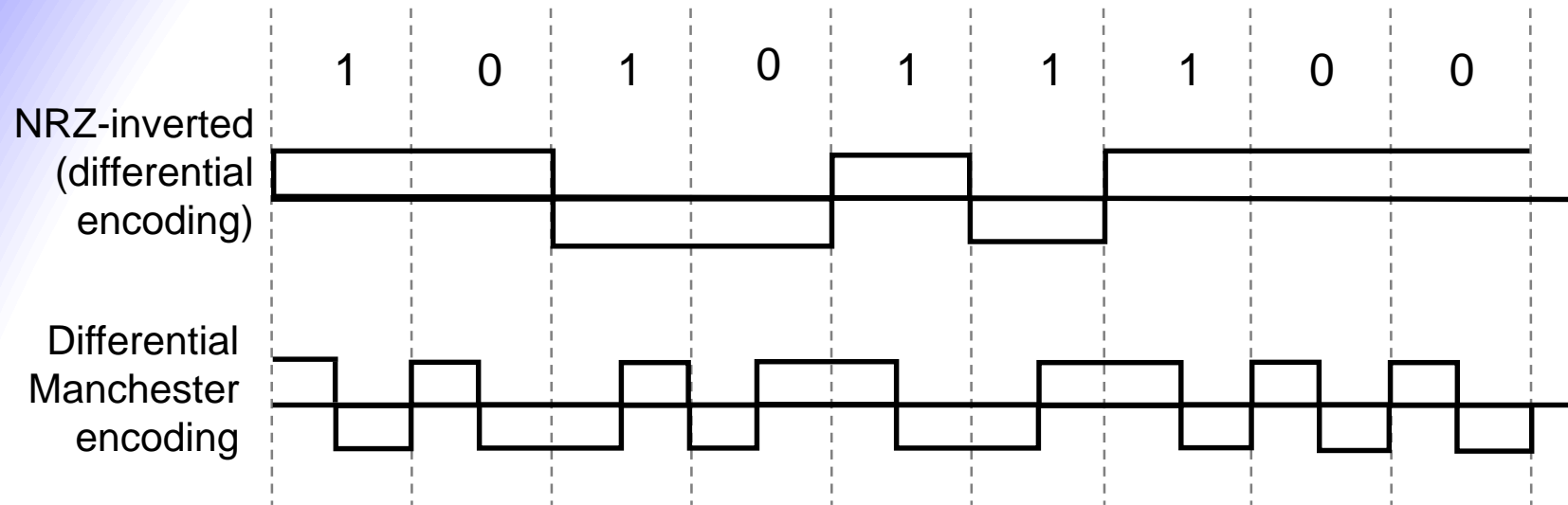
- Errors in some systems cause transposition in polarity, +A become –A and vice versa
  - All subsequent bits in Polar NRZ coding would be in error
- Differential line coding provides robustness to this type of error
- "1" mapped into transition in signal level
- "0" mapped into no transition in signal level
- Same spectrum as NRZ
- Errors occur in pairs
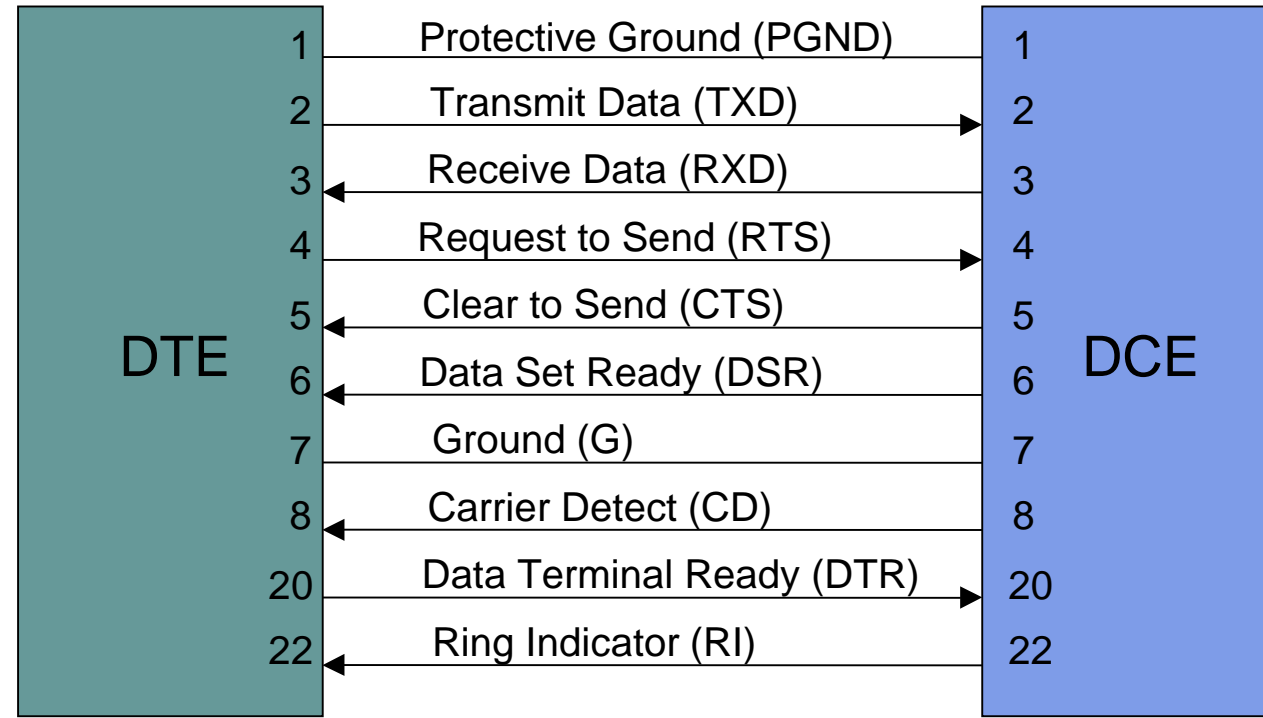- Also used with Manchester coding

2.3   Asynchronous and synchronous communications

- Recommended Standard (RS) 232
- serial line interface between computer and modem or similar device
- Data Terminal Equipment (DTE): computer
- Data Communications Equipment (DCE): modem
- mechanical and electrical specification

(a)

(b)

| DTE		DCE
1	Protective Ground (PGND)	1
2	Transmit Data (TXD) →	2
3	← Receive Data (RXD)	3
4	Request to Send (RTS) →	4
5	← Clear to Send (CTS)	5
6	← Data Set Ready (DSR)	6
7	Ground (G)	7
8	← Carrier Detect (CD)	8
20	Data Terminal Ready (DTR) →	20
22	← Ring Indicator (RI)	22

Synchronization

- Synchronization of clocks in transmitters and receivers.
    - clock drift causes a loss of synchronization
- Example: assume '1' and '0' are represented by V volts and 0 volts respectively
    - Correct reception
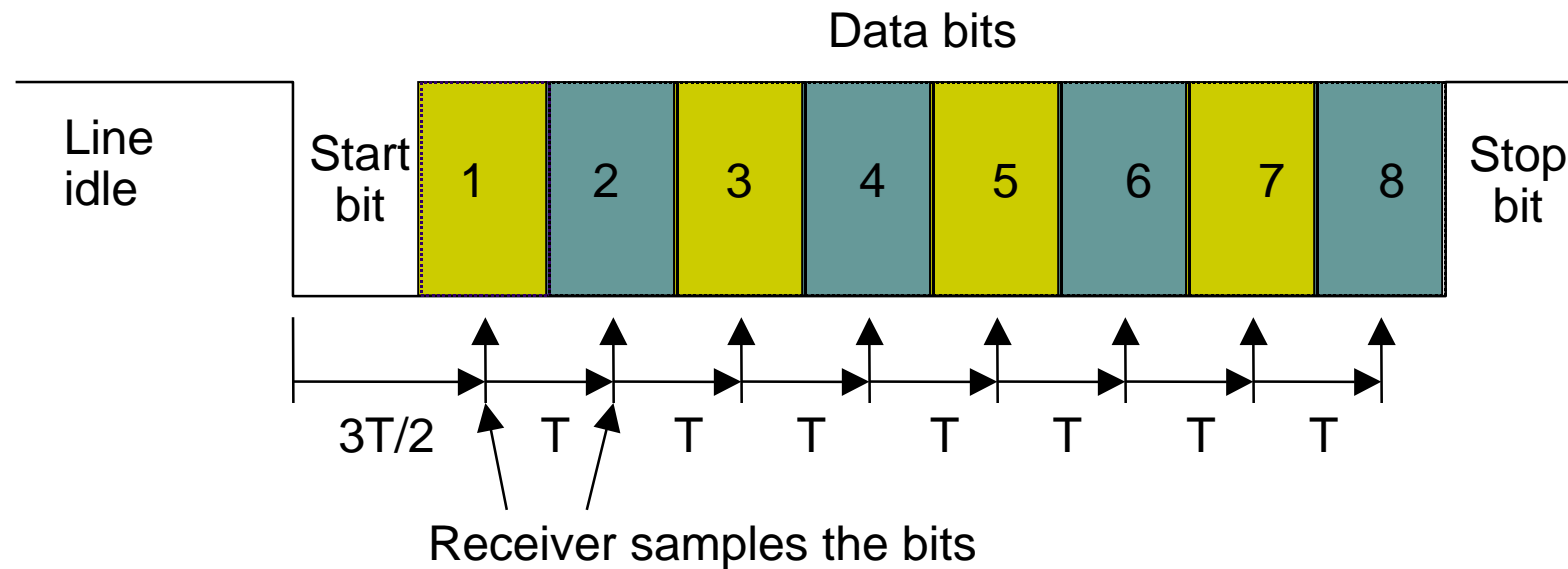   - Incorrect reception due to incorrect clock (slower clock)

1 0 1 1 0 1 0 0 1 0 0

Data

T

S

Clock

1 0 1 1 1 0 0 1 0 0 0

Data

T

S'

Clock
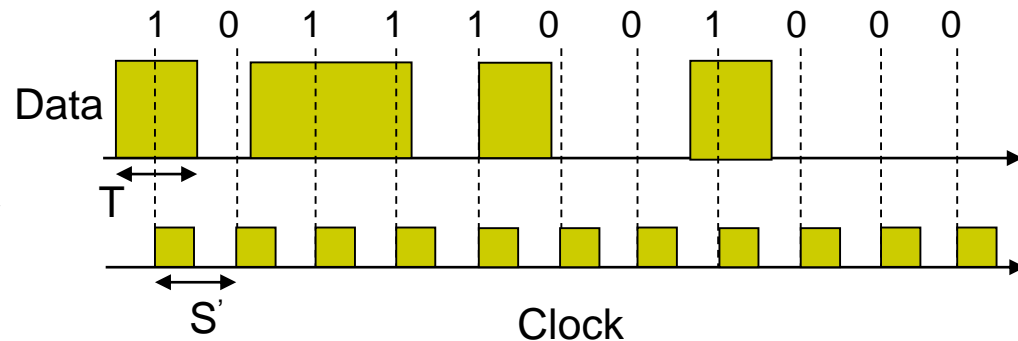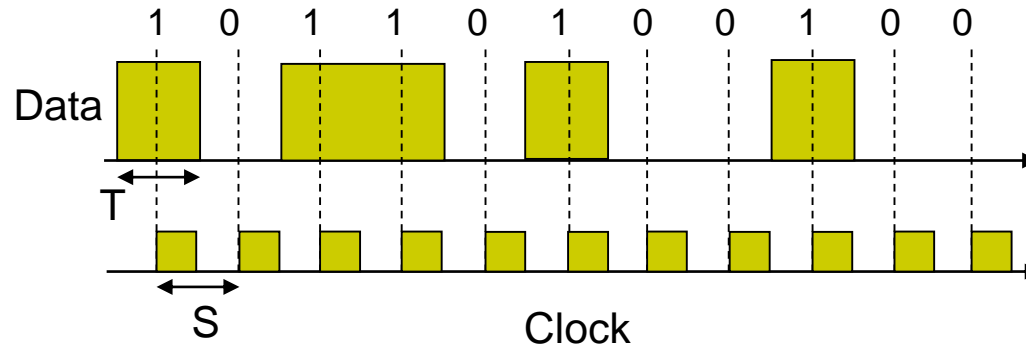
- Incorrect reception
- How to avoid a loss of synchronization?
  - Asynchronous transmission
 - Synchronous transmission
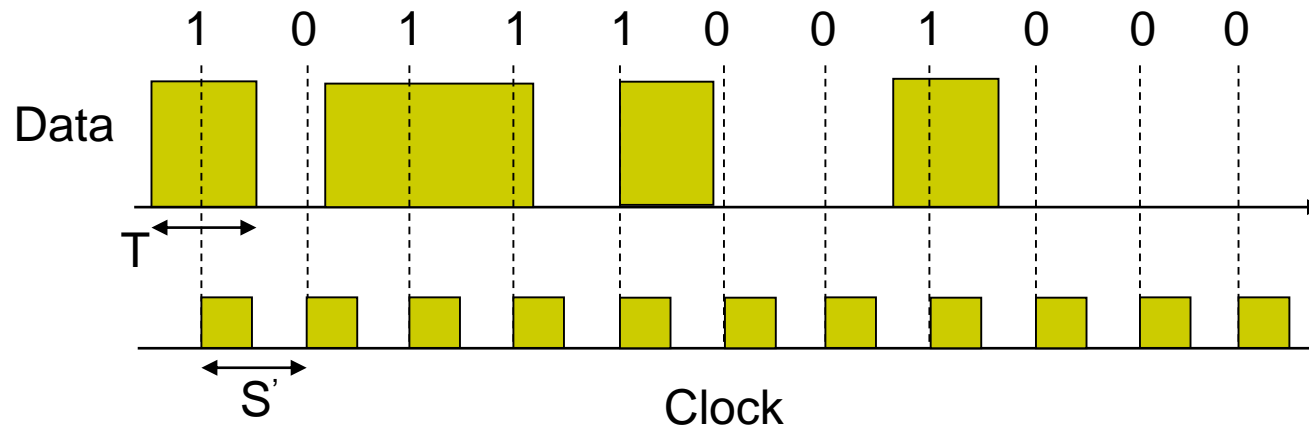
Asynchronous transmission

- Avoids synchronization loss by specifying a short maximum length for the bit sequences and resetting the clock in the beginning of each bit sequence.

Data bits

| Line idle | Start bit | 1 | 2 | 3 | 4 | 5 | 6 | 7 | 8 | Stop bit |

3T/2   T   T   T   T   T   T   T
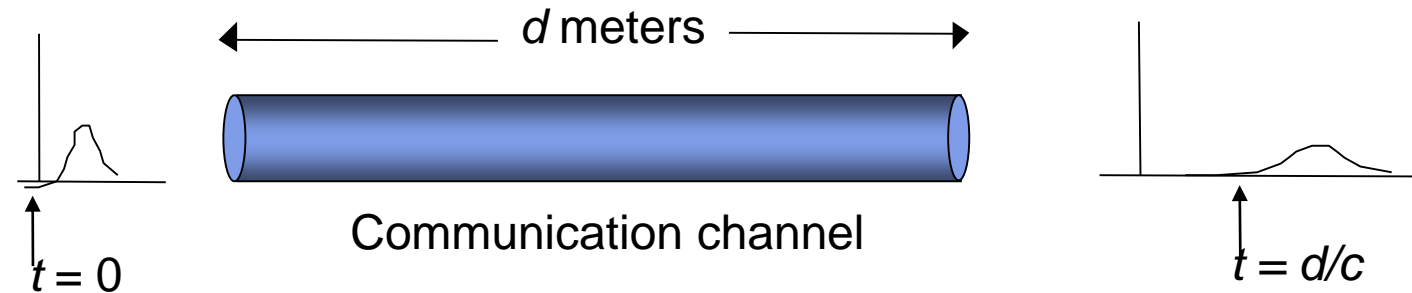
Receiver samples the bits

Synchronous transmission

- Sequence contains data + clock information (line coding)
  - i.e. Manchester encoding, self-synchronizing codes, is used.
- R transition for *R* bits per second transmission

2.4   Transmission media

d meters

Communication channel

$t = 0$ $t = d/c$
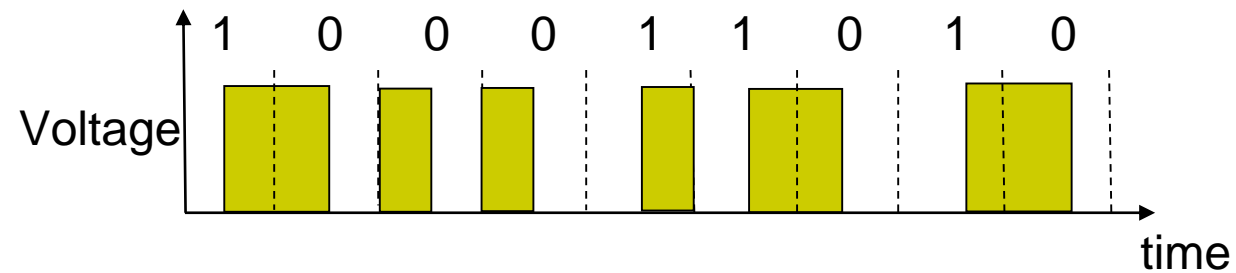
- Information bearing capacity
  - Amplitude response & bandwidth
    - dependence on distance
  - Susceptibility to noise & interference
    - Error rates & SNRs
- Propagation speed of signal
  - $c = 3 \times 10^8$ meters/second in vacuum
  - $n = c/\sqrt{e}$ speed of light in medium where e>1 is the dielectric constant of the medium
  - $n = 2.3 \times 10^8$ m/sec in copper wire; $n = 2.0 \times 10^8$ m/sec in optical fiber

frequency of communication signals
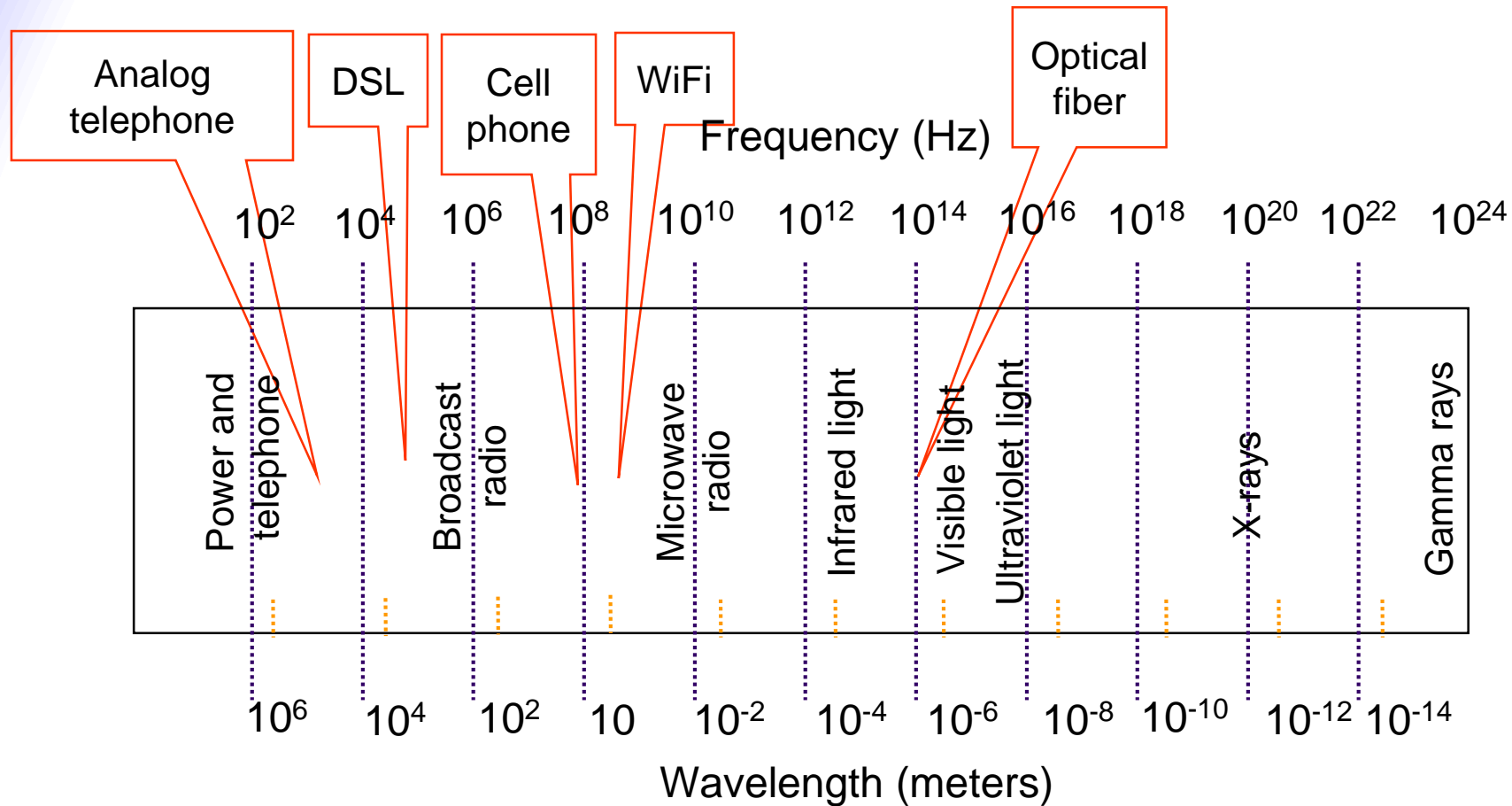
Wireless Media

- 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 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

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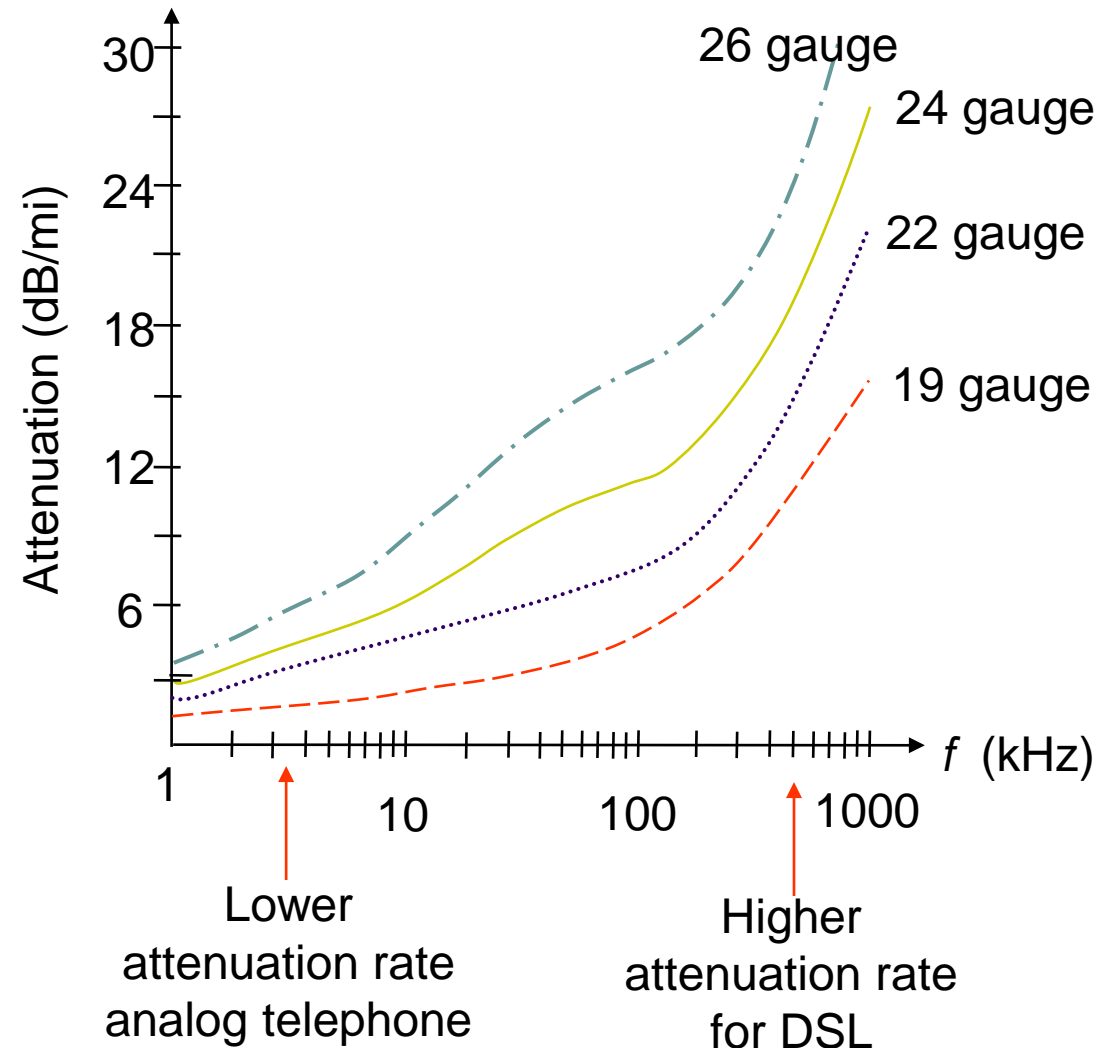
| Standard	Data Rate	Distance
T-1	1.544 Mbps	18,000 feet, 5.5 km
DS2	6.312 Mbps	12,000 feet, 3.7 km
1/4 STS-1	12.960 Mbps	4500 feet, 1.4 km
1/2 STS-1	25.920 Mbps	3000 feet, 0.9 km
STS-1	51.840 Mbps	1000 feet, 300 m

- 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 LAN
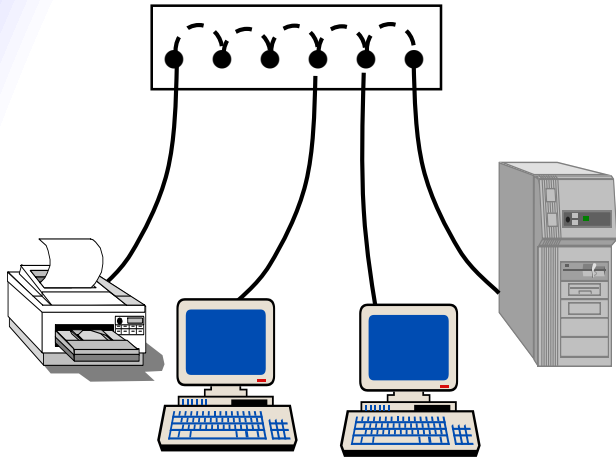
- Category 3 unshielded twisted pair (UTP): ordinary telephone wires
- Category 5 UTP: tighter twisting to improve signal quality
- Shielded twisted pair (STP): to minimize interference; costly
- 10BASE-T Ethernet
  - 10 Mbps, Baseband, Twisted pair
  - Two Cat3 pairs
  - Manchester coding, 100 meters
- 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
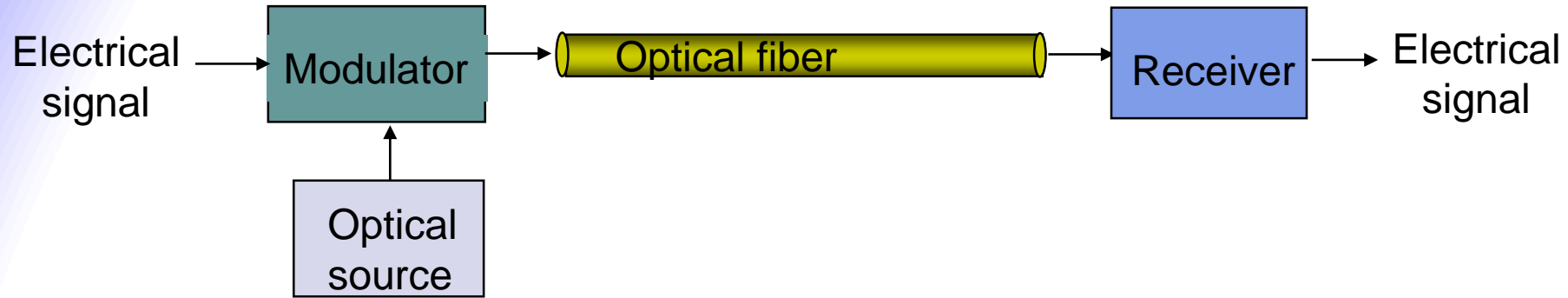- Cat5 & STP provide other options

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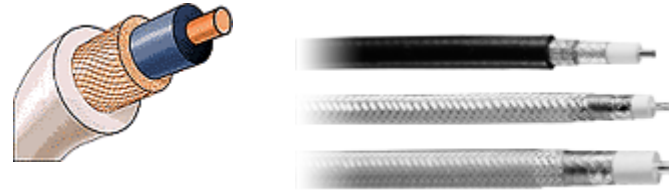
- Cylindrical braided outer conductor surrounds insulated inner wire conductor
- High interference immunity
- Higher bandwidth than twisted pair
- Hundreds of MHz
- Cable TV distribution
- Long distance telephone transmission
- Original Ethernet LAN medium
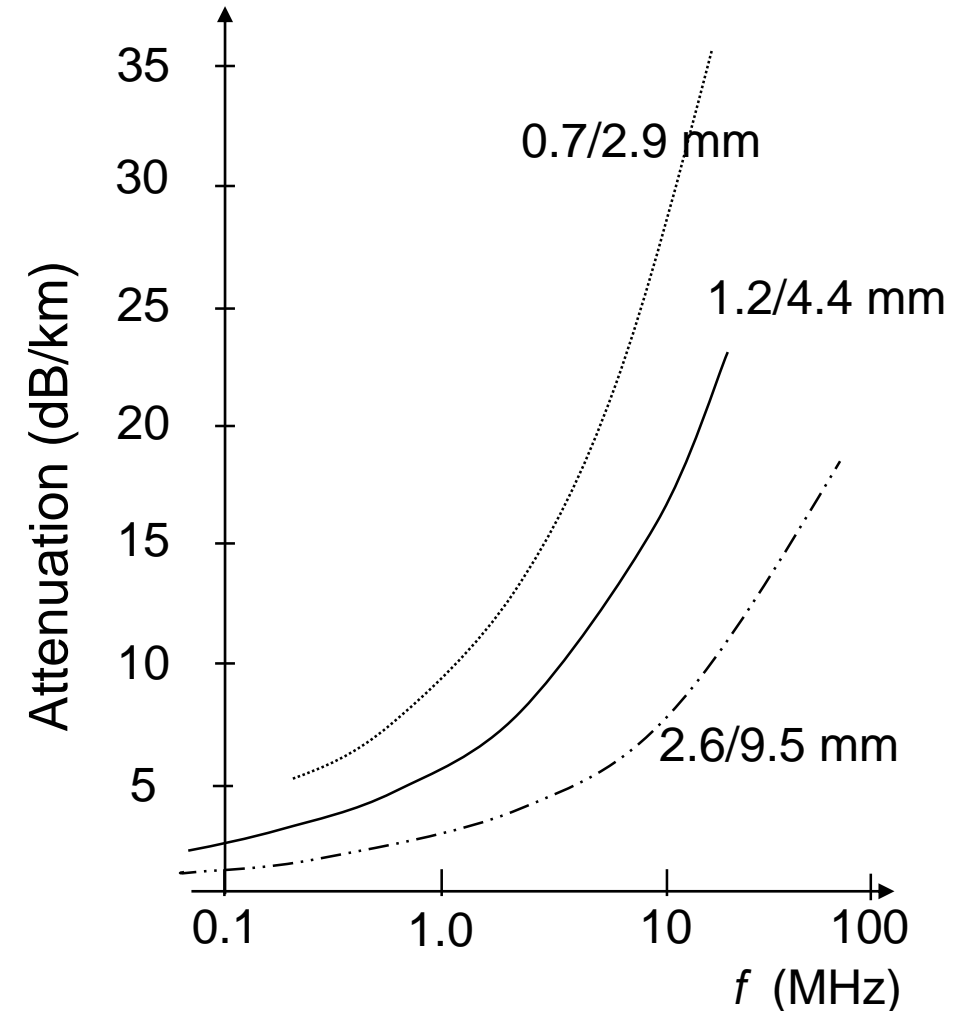
Optical fiber


- 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

Geometry of optical fiber

Light

Cladding

Jacket
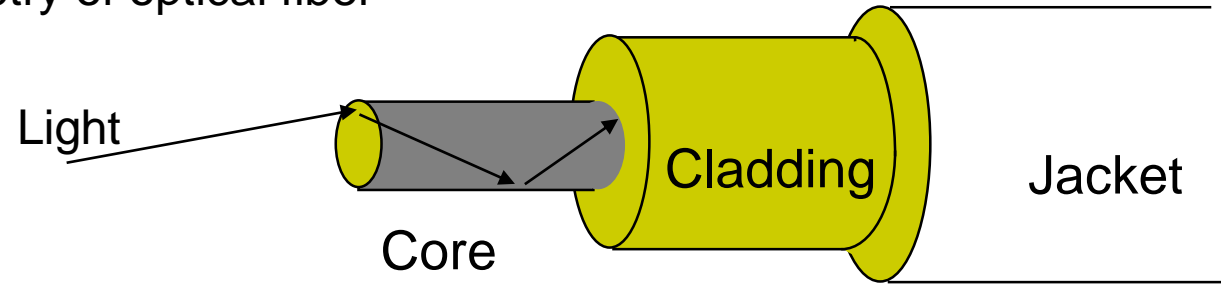
Core

Total Internal Reflection in optical fiber

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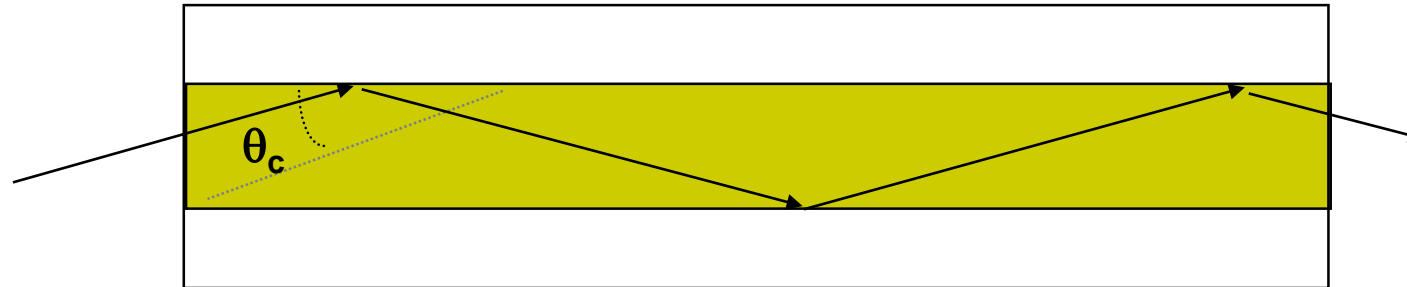
- Very fine glass cylindrical core surrounded by concentric layer of glass (cladding)
- Core has higher index of refraction than cladding
- Light rays incident at less than critical angle θ_c is completely reflected back into the core

Multimode fiber: multiple rays follow different paths

Reflected path
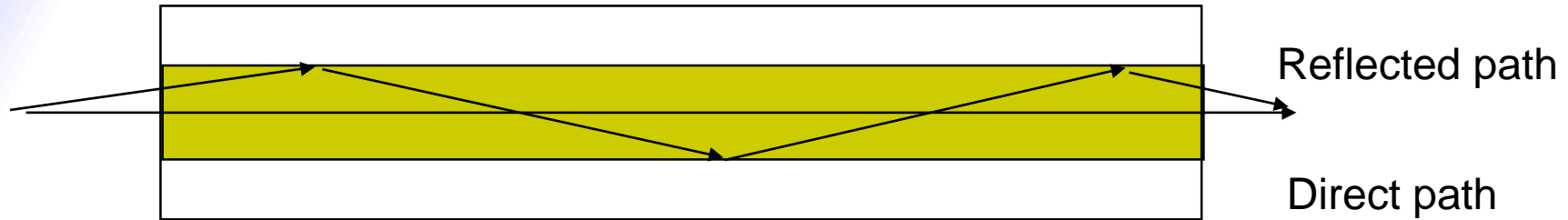
Direct path

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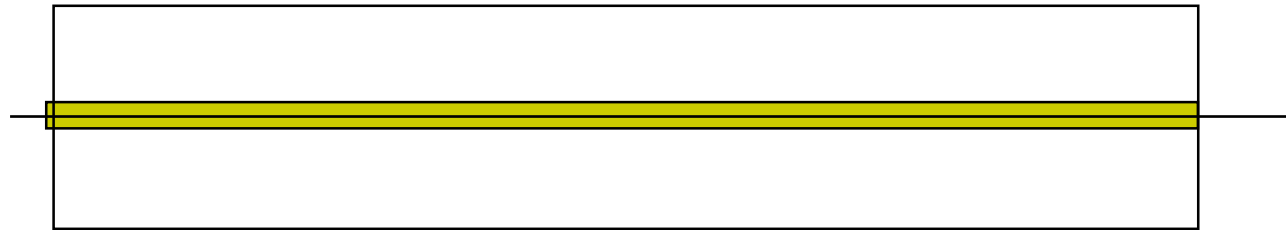
- Multimode: thicker core, shorter reach
  - Rays on different paths interfere causing dispersion & limiting bit rate
- Single mode: very thin core supports only one mode (path)
  - More expensive lasers, but achieves very high speeds

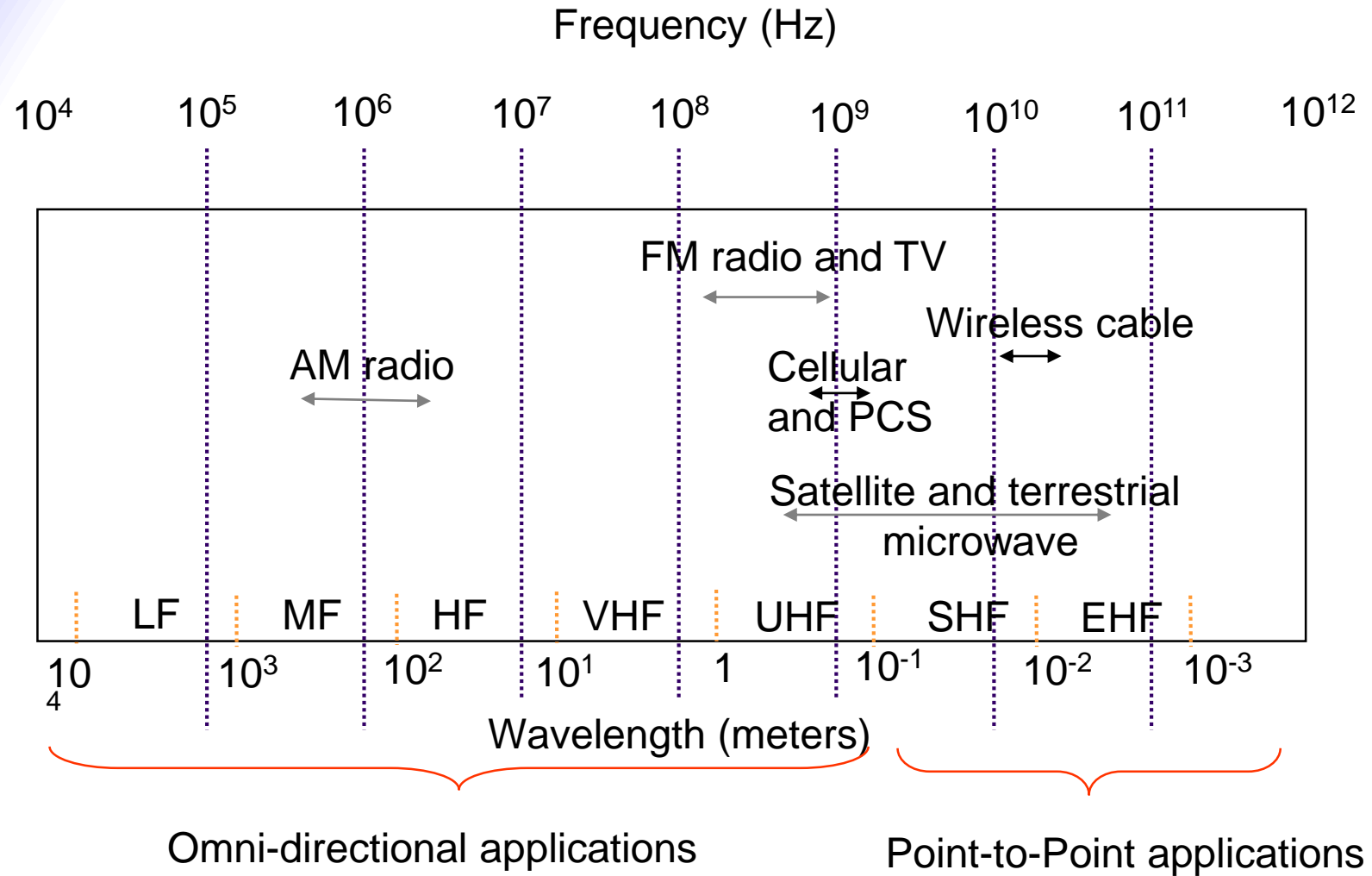# Radio transmission

- 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

Frequency (Hz)

10^4 10^5 10^6 10^7 10^8 10^9 10^{10} 10^{11} 10^{12}

FM radio and TV

Wireless cable

AM radio

Cellular
and PCS

Satellite and terrestrial
microwave

LF   MF   HF   VHF   UHF   SHF   EHF

10^4 10^3 10^2 10^1 1 10^{-1} 10^{-2} 10^{-3}

Wavelength (meters)

Omni-directional applications   Point-to-Point applications

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

Satellite Communications

- Geostationary satellite @ 36,000 km above equator
- Relays microwave signals from uplink frequency to downlink frequency
- Long distance telephone
- Satellite TV broadcast

Chapter Summary

♦ digital representation of information
♦ digital transmission – digitization of analog signal, characterization of communication channel, limitations, line coding
♦ asynchronous and synchronous communications
♦ transmission media

Reference

Chapter 3, Communication
Networks: Fundamental Concepts
and Key Architectures

Alberto Leon-Garcia   Indra Widjaja

Second Edition

COMMUNICATION NETWORKS
Fundamental Concepts and Key Architectures