Digital Communications & Physical Layer

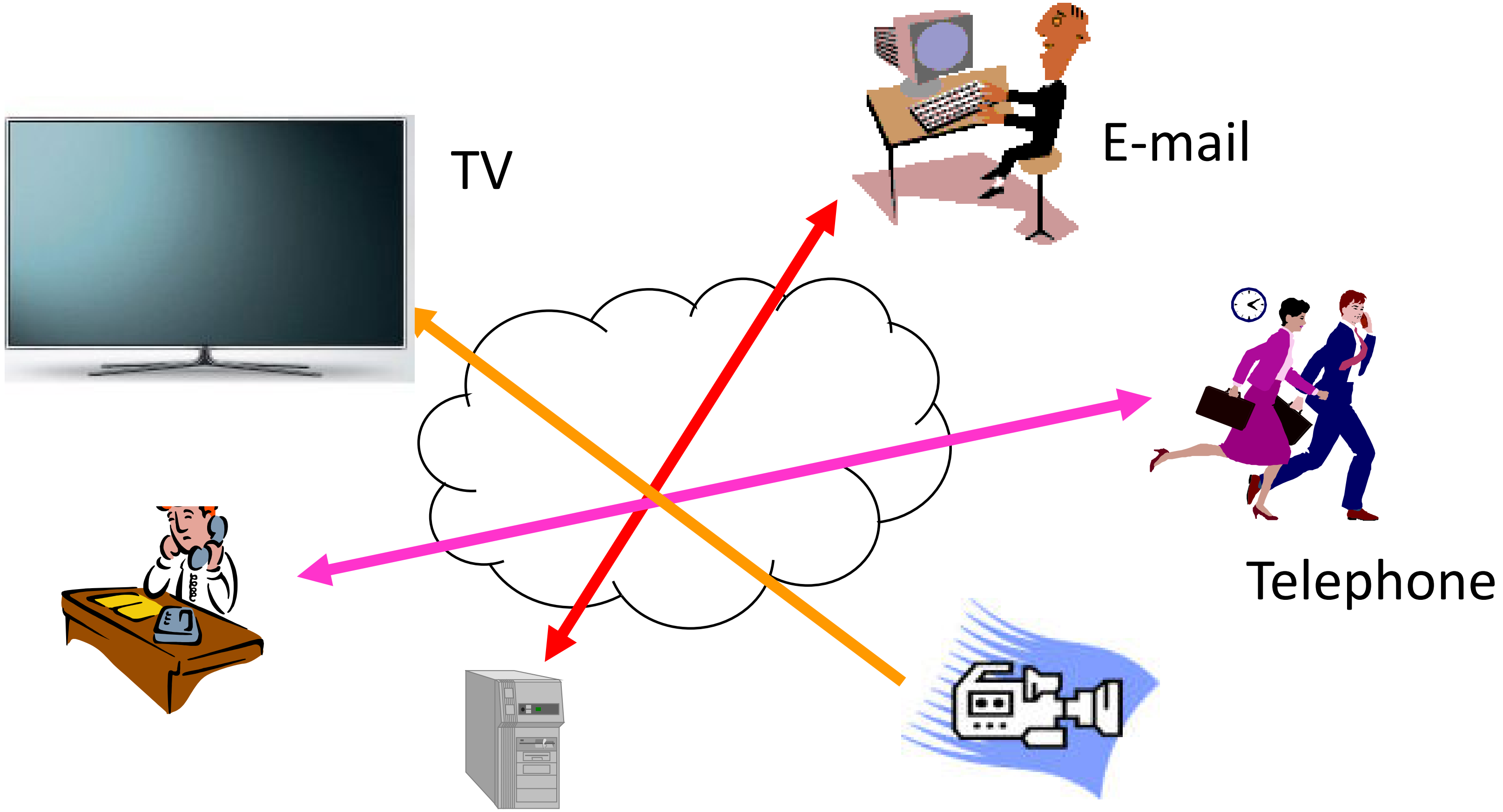
A/PROF. DUY NGO

Learning Objectives

- Digital communication
- Information properties
- Channel Capacity
- Line Coding

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, …?

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
  - More bits → better quality
  - More bits → high cost of transmission

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

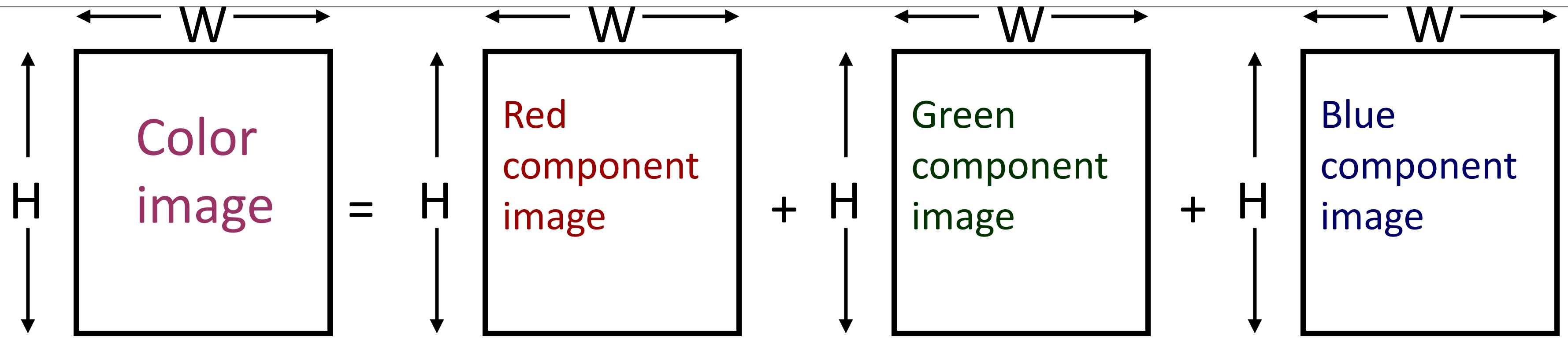
Stream

- 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

Data Rate Units

Unit	Value in Bytes	Value in Bits	Equal to
Zettabyte	2^{70} = 1.180596E21	9.444733E21	1024 Exabytes
Exabyte	2^{60} = 1.152921E18	9.223372E18	1024 Petabytes
Petabyte	2^{50}= 1.125899E15	9.0071993E15	1024 Terabytes
Terabyte	2^{40}= 1.0995116E12	8.796093E12	1024 Gigabytes
Gigabyte	2^{30}= 1.0737418E09	8.5899344E09	1024 Megabytes
Megabyte	2^{20}= 1.048576E06	8.388608E06	1024 Kilobytes
Kilobyte	2^{10}= 1.024E03	8.192E03	1024 Bytes

Color Image



Total bits = 3 × H × W pixels × B bits/pixel = 3HWB bits

Example: 8×10 inch picture at 400 × 400 pixels per inch^2

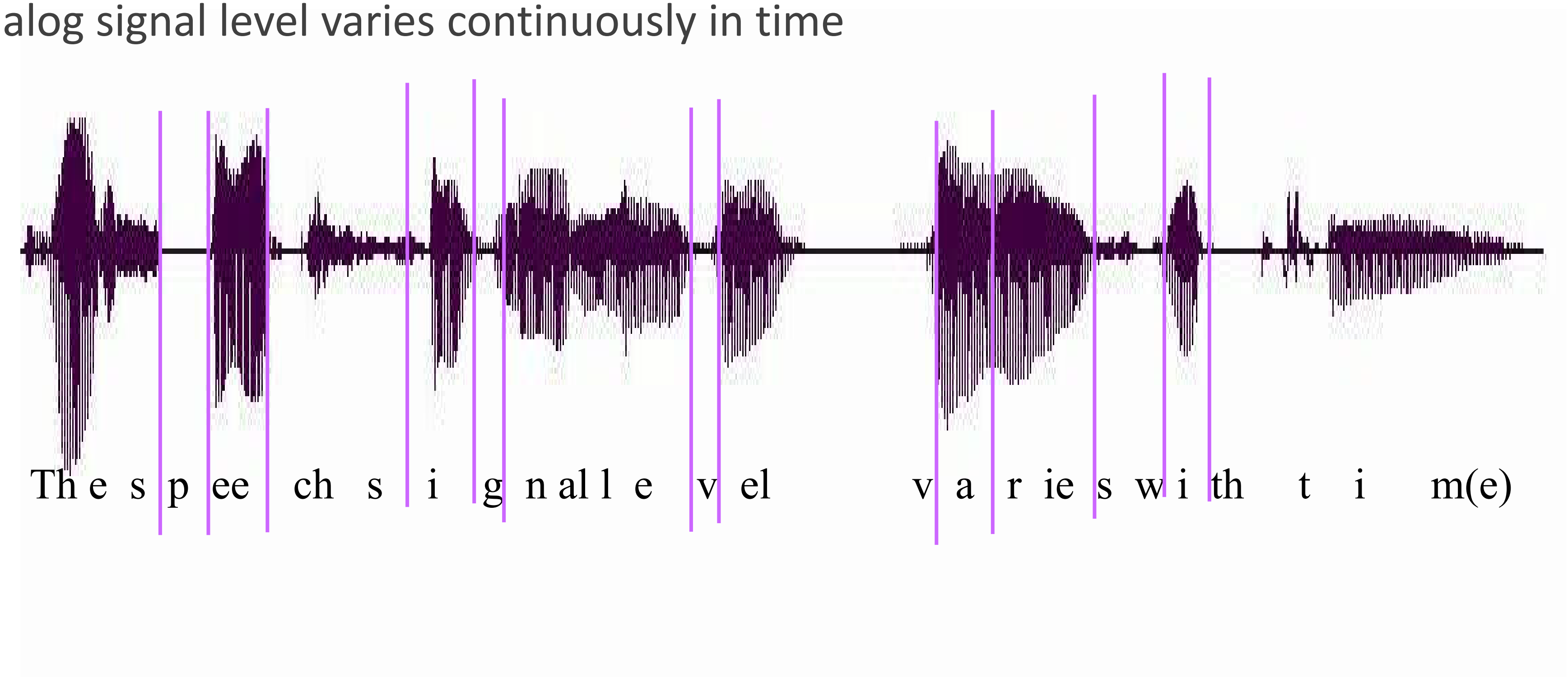
400 × 400 × 8 × 10 = 12.8 million pixels

8 bits/pixel/color

12.8 megapixels × 3 bytes/pixel = 38.4 megabytes

Stream Information

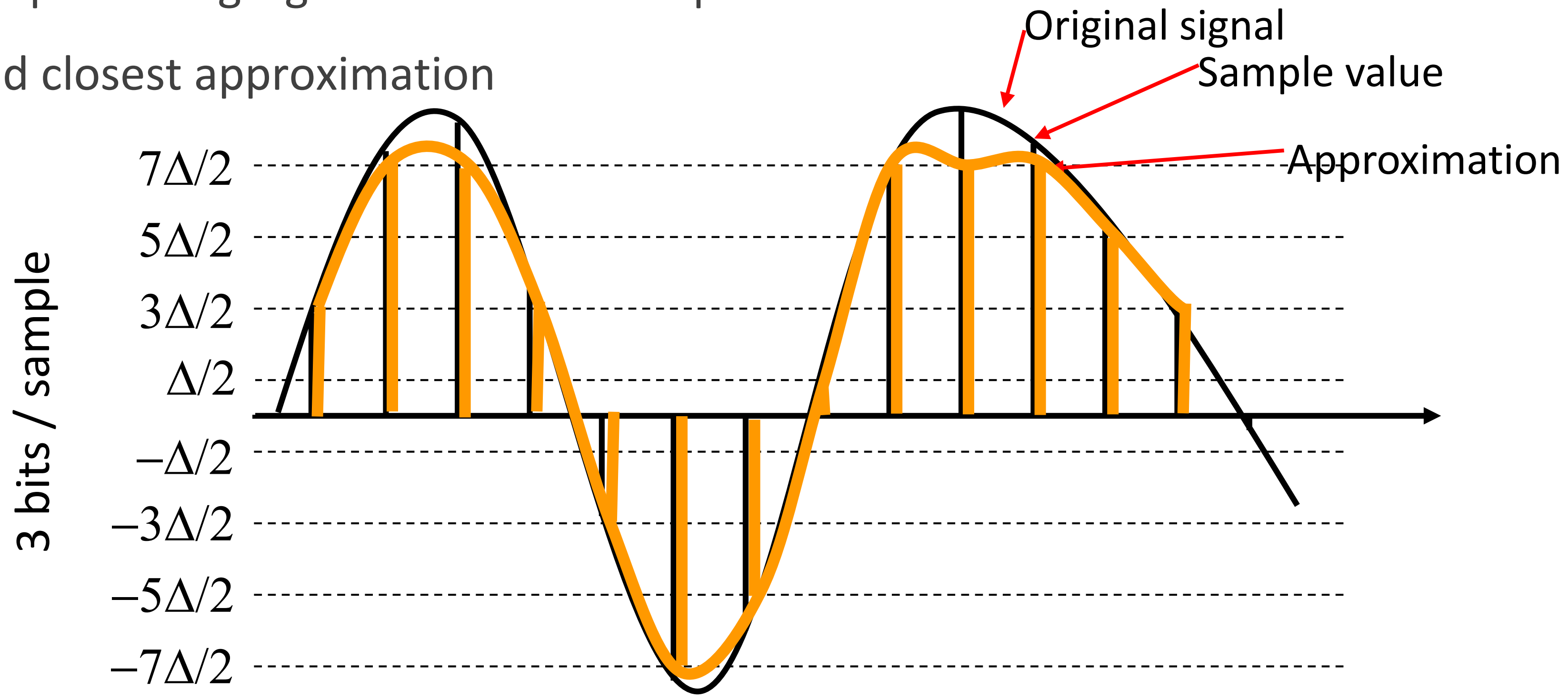
- A real-time voice signal must be digitized & transmitted as it is produced
- Analog signal level varies continuously in time



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 less quantization error
  - → Smaller spacing between approximation values
  - → More bits per sample
  - → Requires higher transmission rate
  - → Higher sampling rate could take advantage of block transmission technique

Example: Voice & Audio

Telephone voice

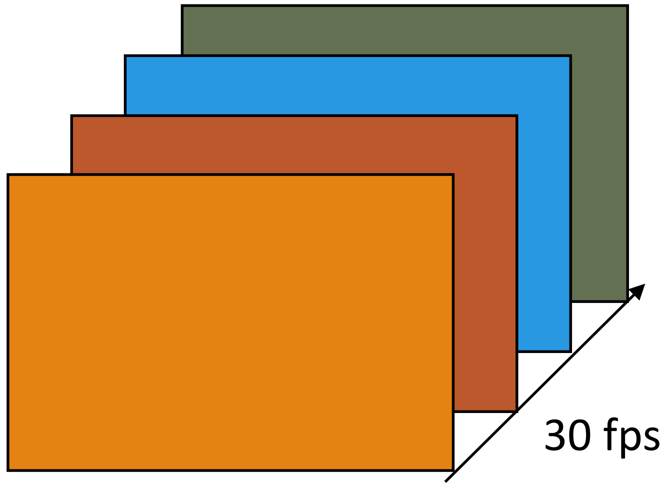
- W_s = 4 kHz → 8000 samples/sec
- 8 bits/sample
- R_s=8 x 8000 = 64 kbps
- Cellular phones use more powerful compression algorithms: 8-12 Kbps

CD Audio

- W_s = 22 KHz → 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

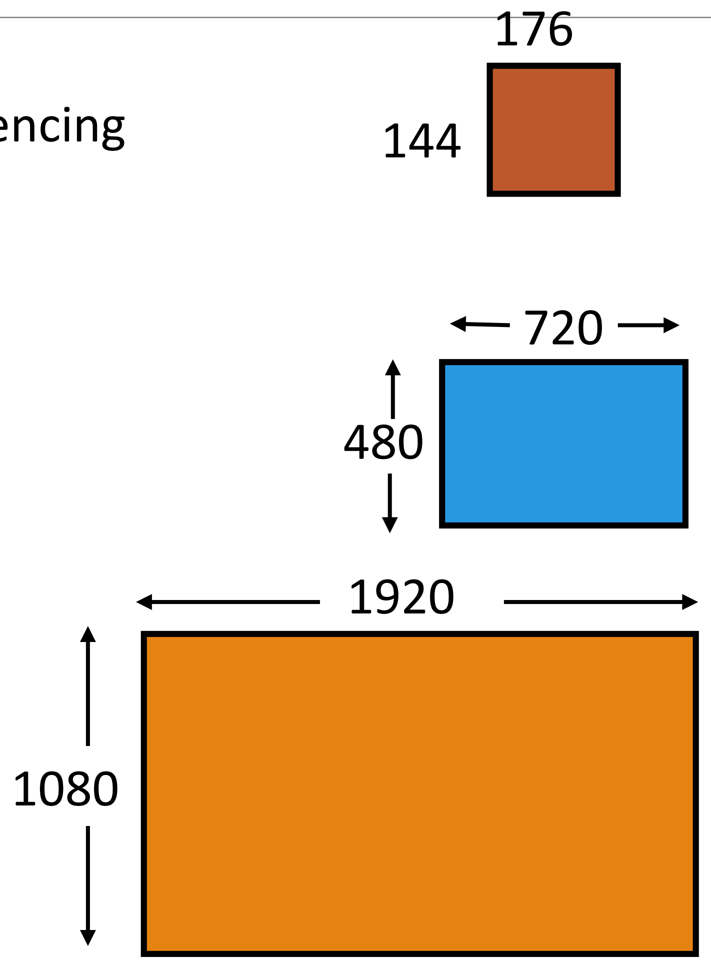
Video Signal

- 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 (ITU)	176x144 or 352x288 pix @10-30 fr/sec	2-36 Mbps	64-1544 kbps
Full Motion	MPEG2	720x480 pix @30 fr/sec	249 Mbps	2-4 Mbps
HDTV	MPEG2	1920x1080 @30 fr/sec	1.6 Gbps	6-10 Mbps
4K video	H.265 (ITU)	3840x2160 @60 fps	11.9 Gbs	16-25 Mbps

ITU: International Telecommunications Union

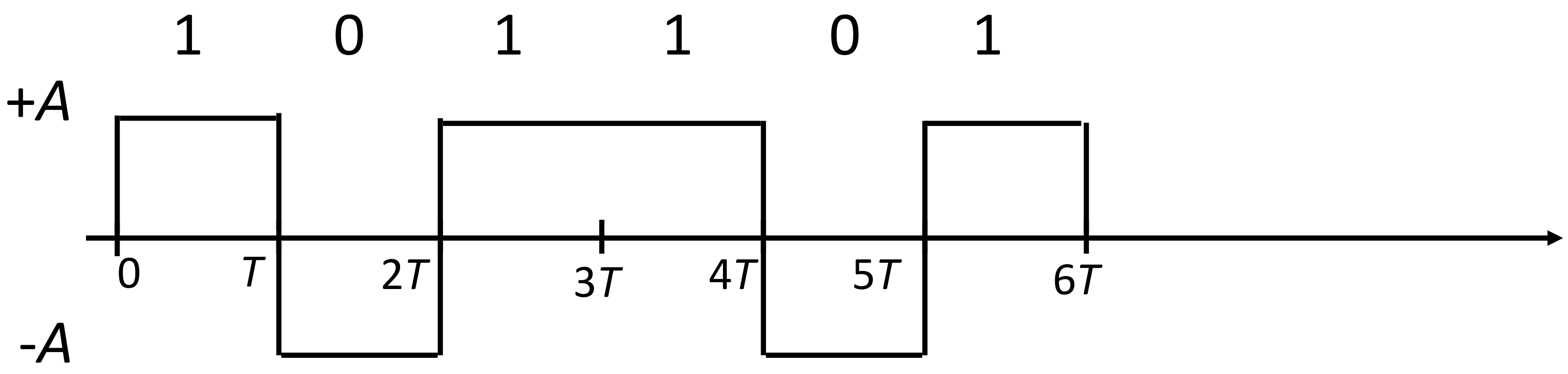
Transmission of Stream Information

- Constant bit-rate (CBR)
  - 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
  - Many video coders produce CBR traffic
- Variable bit-rate (VBR)
  - 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
  - Most of the data sources produces VBR traffic.
  - Packet switched networks can support vbr traffic more efficiently
  - VBR traffic applications are on the rise

Stream Service Quality Issues

- Network transmission impairments
- Delay: is information delivered in timely fashion? in a network we will consider end to end delay as a quality of a link. the end to end delay will consist of a number of components.
- Jitter: is information delivered in sufficiently smooth fashion? particularly important for real time traffic.
- Loss: is information delivered without loss? if loss occurs, is delivered signal quality acceptable? some applications such as voice connections can accept certain amount of loss whereas data connections can't accept any loss.
- Applications & application layer protocols developed to deal with these impairments

Digital Binary Signal

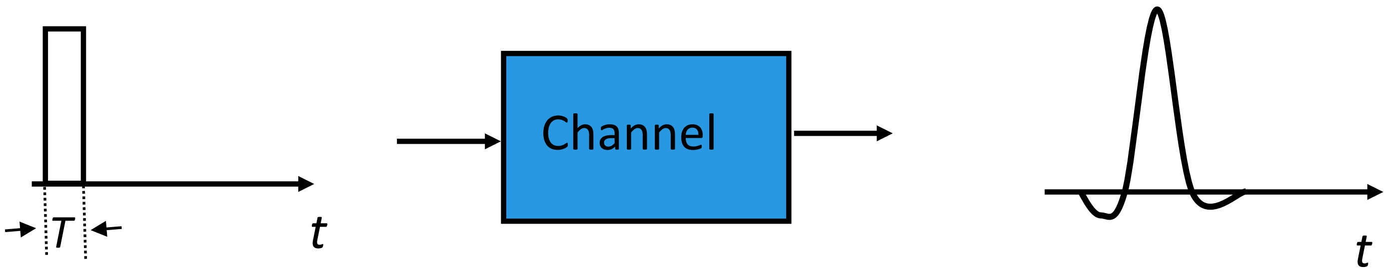


Bit rate = 1 bit / T seconds

- 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

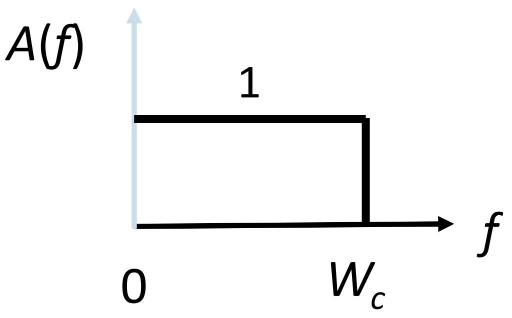


- If input is a narrow pulse, then typical output is a spread-out pulse with ringing
- Question: How frequently can these pulses be transmitted without interfering with each other?
- Answer: 2 x W_c pulses/second
- where W_c is the bandwidth of the channel

Bandwidth of a Channel

$$X(t) = a\cos(2\pi ft) \longrightarrow \boxed{\text{Channel}} \longrightarrow Y(t) = A(f)\, a\cos(2\pi ft)$$

- 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

Multilevel Pulse Transmission

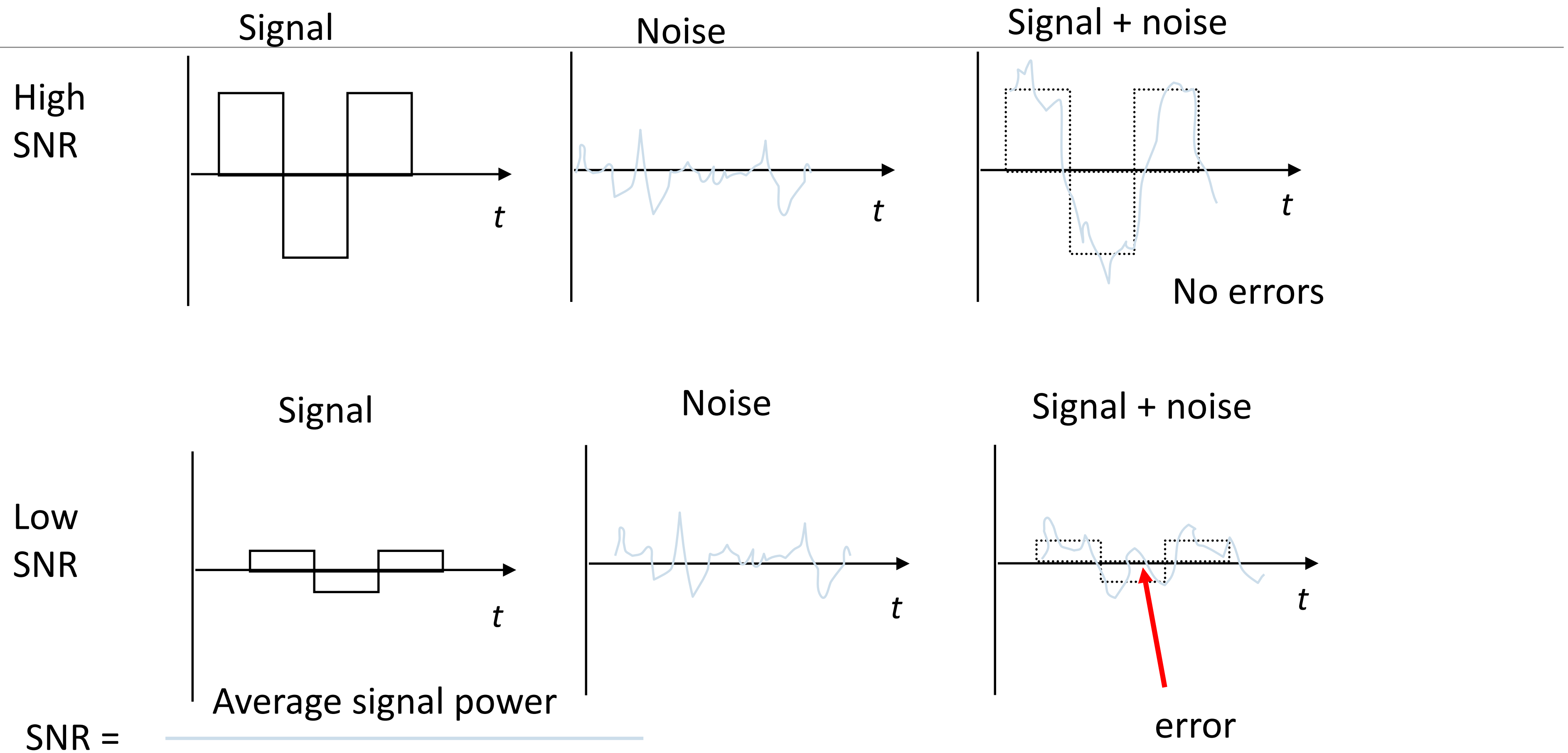
- Assume channel of bandwidth W_c, and transmit 2 W_c pulses/sec (without interference)
- If pulses amplitudes are either -A or +A, then each pulse conveys 1 bit, so
- **bit rate = 1 bit/pulse x 2w_c pulses/sec = 2w_c bps**
- If amplitudes are from {-A, -A/3, +A/3, +A}, then bit rate is 2 x 2w_c bps
- By going to $M = 2^m$ amplitude levels, we achieve
- **Bit rate = m bits/pulse x 2w_c pulses/sec = 2mw_c bps**

In the absence of noise, the bit rate can be increased without limit by increasing m

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
- Noise could be generated from many sources, particularly transmission noise more important to consider
- Interference also produces similar effect as the noise which is generated from other systems

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

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

- C is the channel data rate in bps, W_c is the channel bandwidth in Hz
- 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 (Bit Error Rate) 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.

Example

Find the shannon channel capacity for a telephone channel with W_c = 10 MHz and *SNR* = 10000 (40 dB)

C = 10,000000 $\log_2$ (1 + 10000)

= 10,000000 $\log_{10}$ (10001)/$\log_{10}$2 = 132.87 Mbits/sec

Note that *SNR* = 10000 corresponds to

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

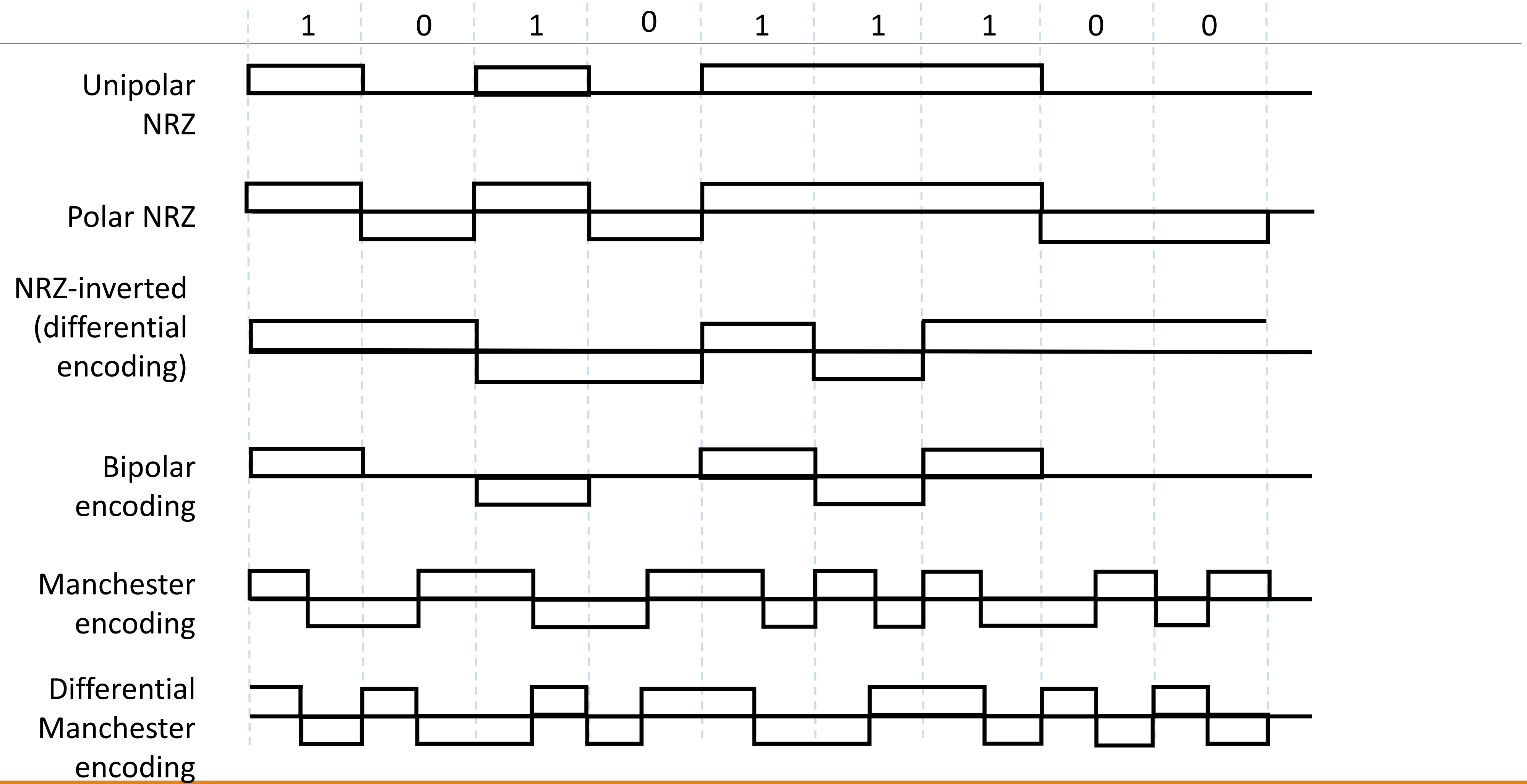
Digital Transmission Systems

System	Bit rate	Bandwidth	Technology
Voice band Copper	33.6 – 56 kbs (old telephone) 16~28 kbs (VoIP/NBN)	4 KHz	Modulation+ Coding
Ethernet	Cat5: 10/100 Mbs Cat5e: ~1 Gbs Cat6: ~ 10 Gbs	Cat5: 100 MHz Cat5e: 100 MHz Cat6: 250 MHz	Line coding: Manchester code
ADSL/ADSL2 Twisted pair	1.3/8.0 Mbs, 1.3/12 Mbs,	1/2 MHz, full duplex	ITU-T G.992.1, ITU 992.3, DMT
VDSL/VDSL2	3/55 Mbs, 100/100 Mbs	10 MHz, Duplexed	ITU-T G.993.2, DMT
G.Fast	~100m : 500 Mbs ~250m: 150 Mbs	106/212 MHz, Time Division Duplex	ITU-TG.9700/9701, DMT
Optical Fibre	2.5-10 Gbs	100 – 150 nm	Multi/Single mode fibre
Optical Fibre	>100 Tbs	As above	WDM, Single mode fibre
IEEE802.11ac	6~700 Mbits/sec	20 -160 MHz	Wireless

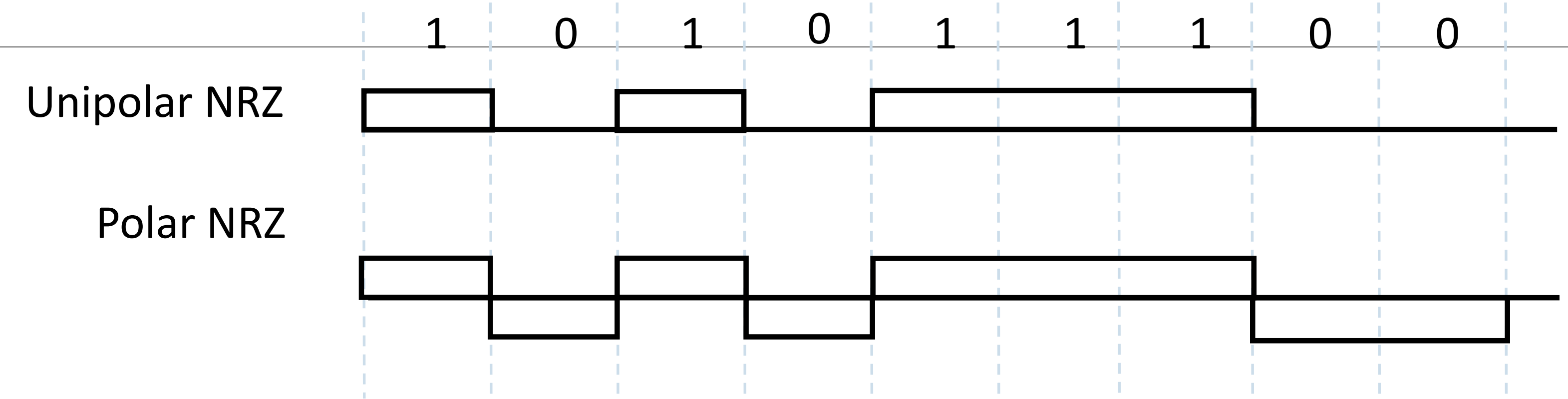
Line coding technique

- Line coding techniques are used for converting binary information sequence into a digital signal in a digital communication system. Line coding is an integral part of the NIC (Network Interface Card)
- 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



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

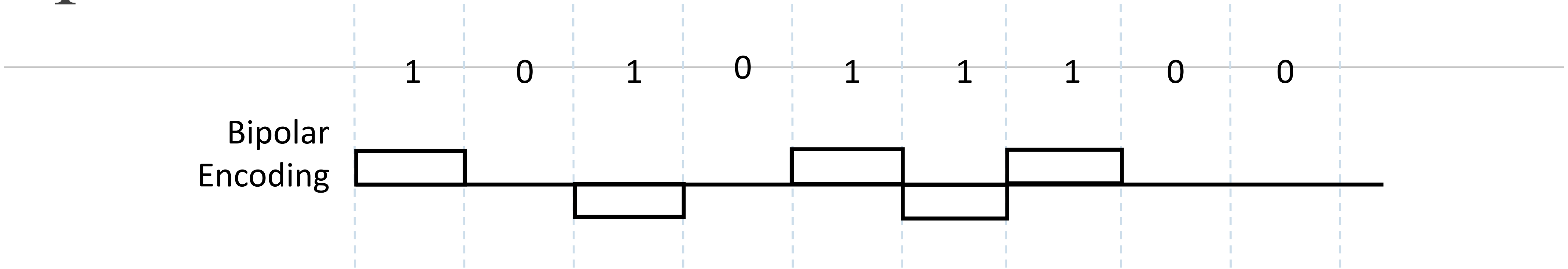
$0.5*(A/2)^2 + 0.5*(-A/2)^2 = A^2/4$

Long strings of +A/2 or –A/2

- Poor timing
- Low-frequency content

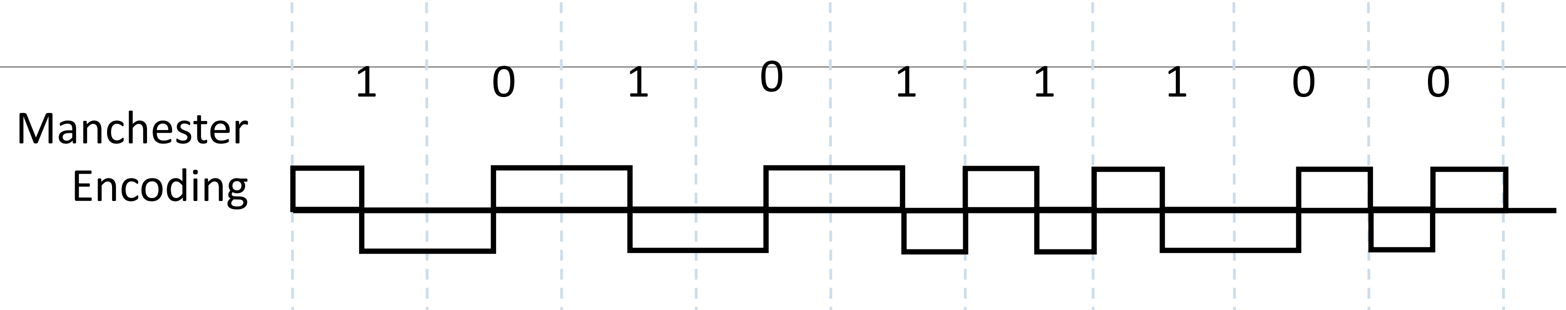
Simple

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

dB, dBm and Watt calculations

- dB is a relative unit, $dB = 10log_{10}\left(\frac{A}{B}\right)$
- A very commonly used unit to represent transmit and receive powers is dBm, which value is relative to 1 mW (milli-watt), 1 mW = 1W/1000
- If a transmitter sends 100 mW signal, the transmitter power in dBm is $10log_{10}\left(\frac{100\times10^{-3}}{1\times10^{-3}}\right) = 20\ dBm$
- If a received signal is 10 dBm then power is Watt is ? 10 mW?
- dB and dBm relationships: A(dB)+B(dB) = C (dB), A(dBm) +B(dB) = C dBm, A(dBm) – B(dB) = C(dBm)
- Example: Transmission power P_{TX} 10 dBm, transmission loss L 10 dB, received power $P_R = P_{Tx} - L = 10 - 10 = 0\ dBm$