EC 745 Telecommunication Networks

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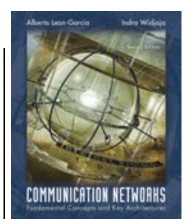
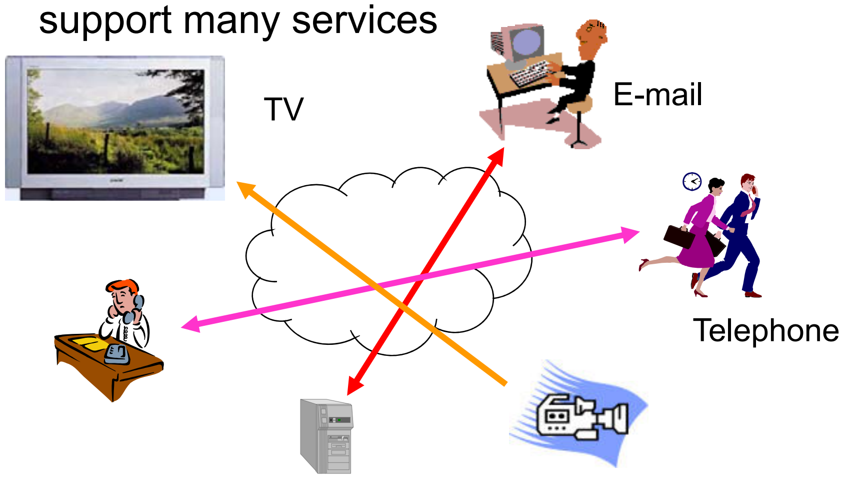
Chapter 3 Digital Transmission Fundamentals



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

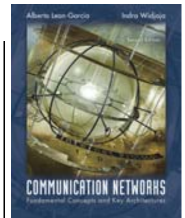
- Digital transmission enables networks to support many services



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 (3×10^8 m/s in vacuum)

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

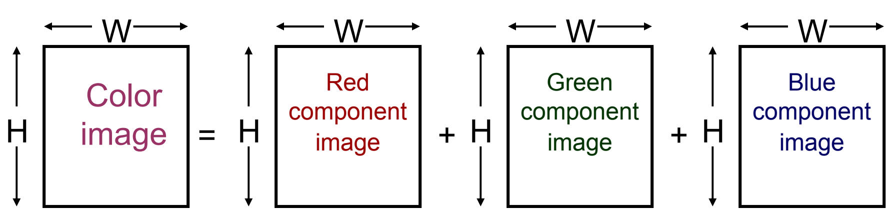
Use data compression to reduce L
Use higher speed modem to increase R
Place server closer to reduce d

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

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

8 bits/pixel/color

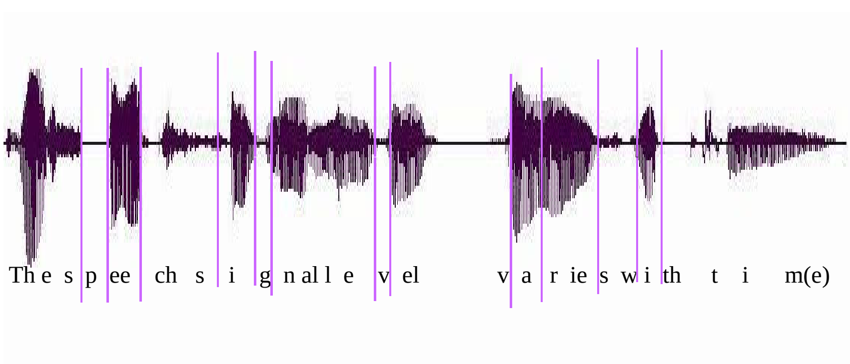
12.8 megapixels $\times$ 3 bytes/pixel = 38.4 megabytes

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)

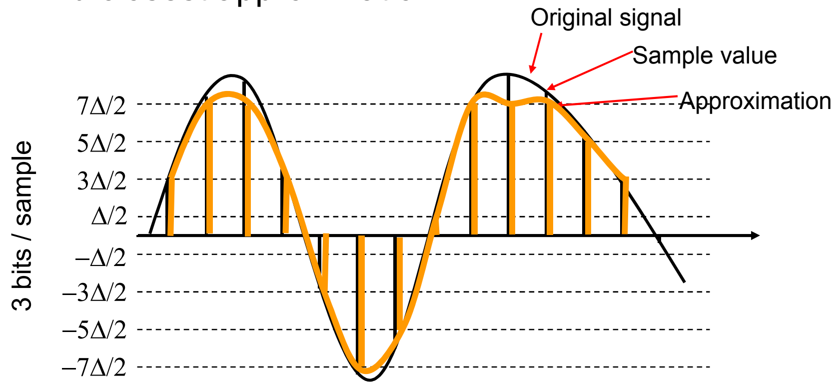
Stream Information

- 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

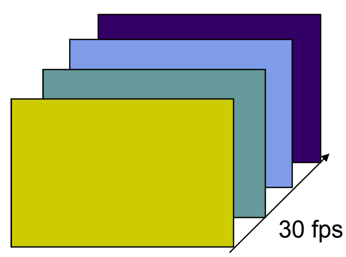
- W_s = 4 kHz → 8000 samples/sec
- 8 bits/sample
- R_s=8 x 8000 = 64 kbps
- Cellular phones use more powerful compression algorithms: 8-12 kbps

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



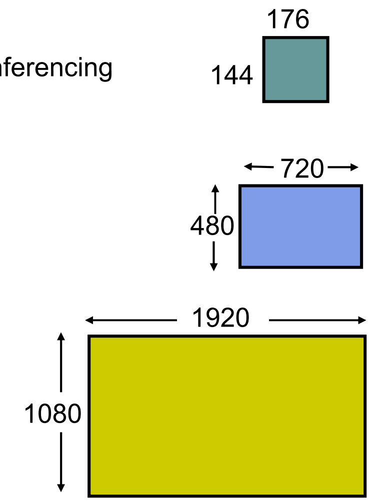
Rate = M bits/pixel **x** (WxH) pixels/frame **x** F frames/second

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

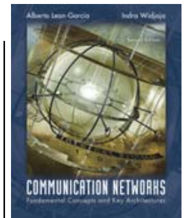
- Constant bit-rate
  - Signals such as digitized telephone voice produce a steady stream: e.g. 64 kbps
  - Network must support steady transfer of signal, e.g. 64 kbps circuit
- Variable bit-rate
  - Signals such as digitized video produce a stream that varies in bit rate, e.g. according to motion and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

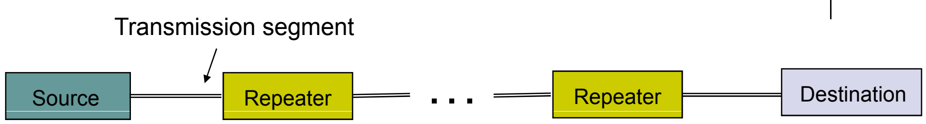
Communication Channel

- 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

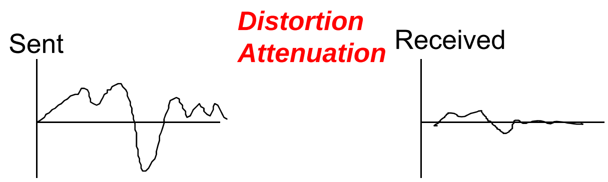
Analog Long-Distance Communications



- 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

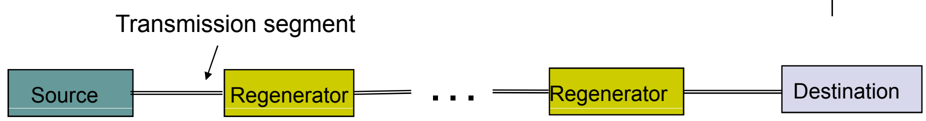
Analog transmission: all details must be reproduced accurately



Digital transmission: only discrete levels need to be reproduced

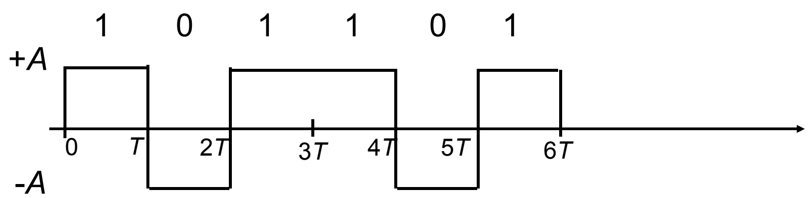
Simple Receiver: Was original pulse positive or negative?

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



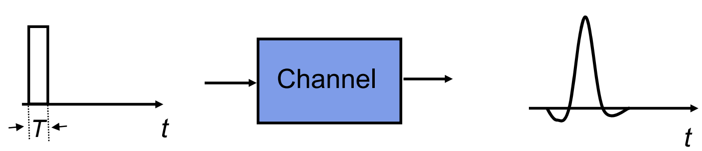
Bit rate = 1 bit / T seconds

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

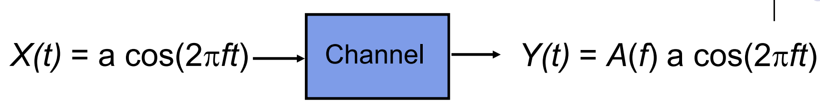
- Objective: Maximize pulse rate through a channel, that is, make T as small as possible



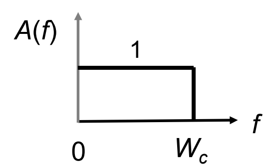
- If input is a narrow pulse, then typical output is a spread-out pulse with ringing
- Question: How frequently can these pulses be transmitted without interfering with each other?
- Answer: 2 x W_c pulses/second

where W_c is the bandwidth of the channel

Bandwidth of a Channel



- 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

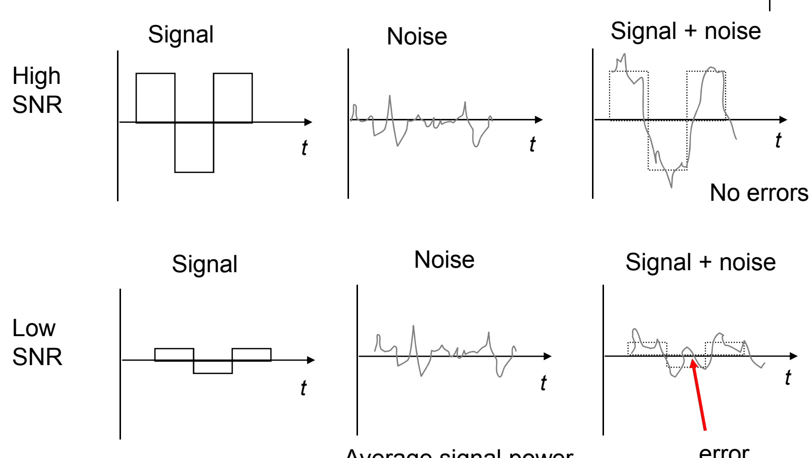
Bit Rate = *m* bits/pulse x 2W_c pulses/sec = 2*m*W_c bps

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

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

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

Note that SNR = 10000 corresponds to
SNR (dB) = $10 \log_{10}(10001) = 40$ dB

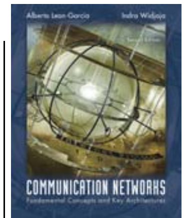
Bit Rates of Digital Transmission Systems

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

Examples of Channels

Channel	Bandwidth	Bit Rates
Telephone voice channel	3 kHz	33 kbps
Copper pair	1 MHz	1-6 Mbps
Coaxial cable	500 MHz (6 MHz channels)	30 Mbps/ channel
5 GHz radio (IEEE 802.11)	300 MHz (11 channels)	54 Mbps / channel
Optical fiber	Many TeraHertz	40 Gbps / wavelength

Chapter 3
Digital Transmission Fundamentals



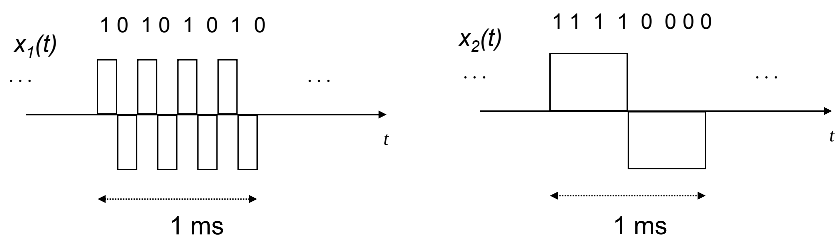
Digital Representation of Analog Signals

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:

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

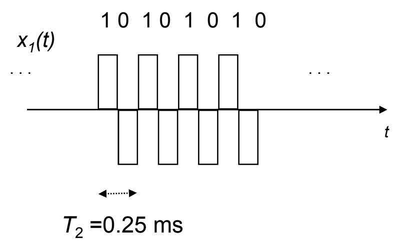
"DC" long-term average (a_0)

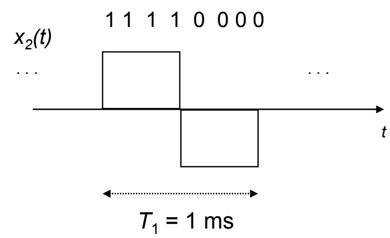
fundamental frequency $f_0 = 1/T$ first harmonic ($a_1\cos(2\pi f_0 t + \phi_1)$)

kth harmonic ($a_k\cos(2\pi k f_0 t + \phi_k)$)

- $|a_k|$ determines amount of power in kth harmonic
- Amplitude specturm $|a_0|, |a_1|, |a_2|, \ldots$

Example Fourier Series





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

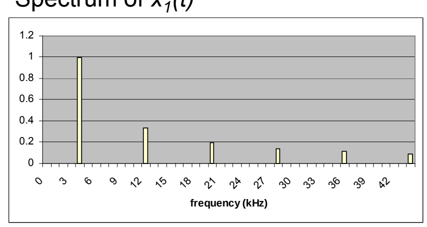
$$x_2(t) = 0 + \frac{4}{\pi}\cos(2\pi 1000t) + \frac{4}{3\pi}\cos(2\pi 3(1000)t) + \frac{4}{5\pi}\cos(2\pi 5(1000)t) + \ldots$$

Only odd harmonics have power

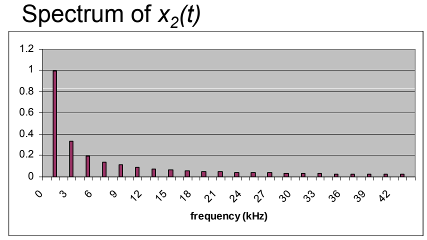
Spectra & Bandwidth

- 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

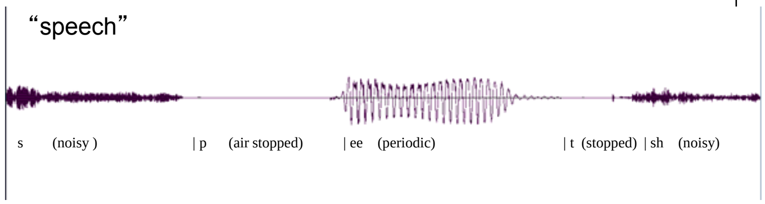
Spectrum of *$x_1(t)$*



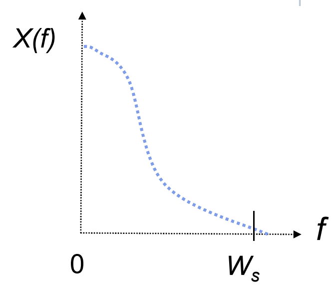
Spectrum of *$x_2(t)$*



Bandwidth of General Signals

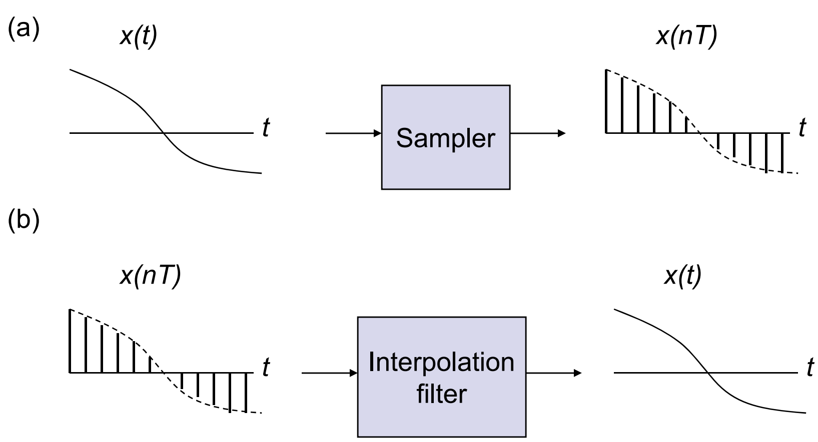


- Not all signals are periodic
  - E.g. voice signals varies according to sound
  - Vowels are periodic, "s" is noiselike
- Spectrum of long-term signal
  - Averages over many sounds, many speakers
  - Involves Fourier transform
- Telephone speech: 4 kHz
- CD Audio: 22 kHz

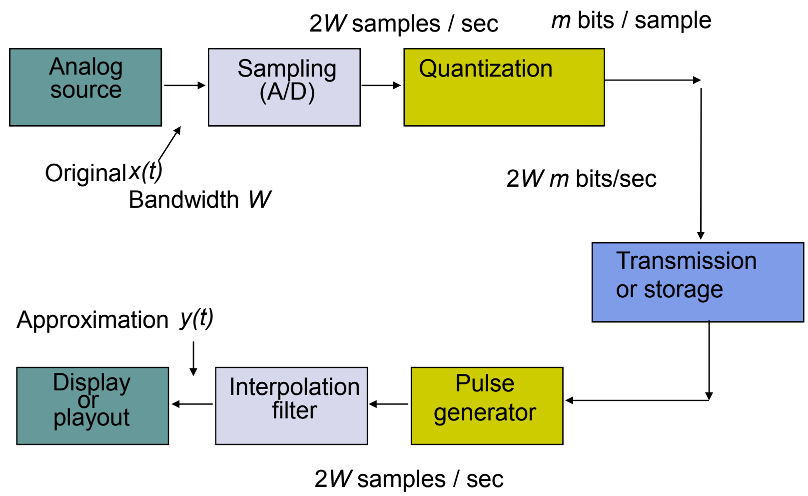


Sampling Theorem

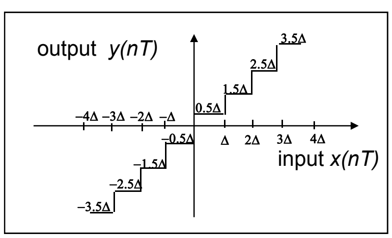
Nyquist: Perfect reconstruction if sampling rate $1/T > 2W_s$



Digital Transmission of Analog Information

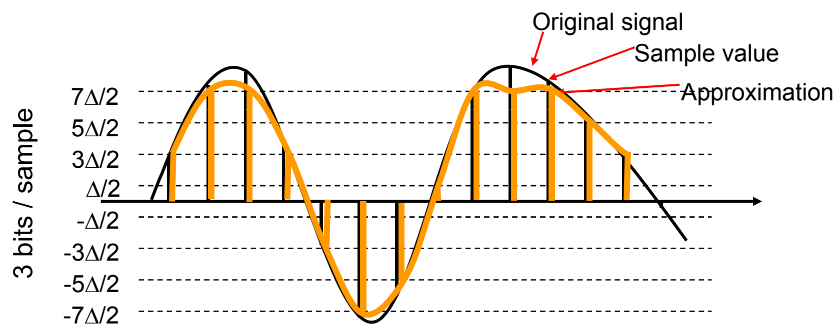


Quantization of Analog Samples



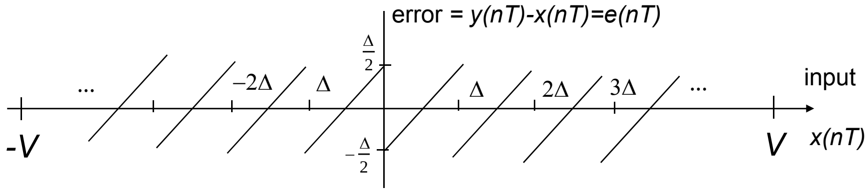
Quantizer maps input into closest of 2^m representation values

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



Quantizer Performance

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



If the number of levels M is large, then the error is approximately uniformly distributed between (-Δ/2, Δ2)

Average Noise Power = Mean Square Error:

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

Quantizer Performance

Figure of Merit:

Signal-to-Noise Ratio = Avg signal power / Avg 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 = $10 \log_{10} \sigma_x^2/\sigma_e^2 = 6 + 10 \log_{10} 3\sigma_x^2/V^2$

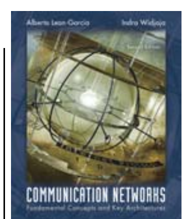
SNR db = 6*m* - 7.27 dB for $V/\sigma_x = 4$.

Example: Telephone Speech

W = 4KHz, 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 = 6*m* – 7
$\Rightarrow$ *m* = 8 bits/sample

PCM ("Pulse Code Modulation") Telephone Speech:
Bit rate= 8000 x 8 bits/sec= 64 kbps

Chapter 3
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?

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*

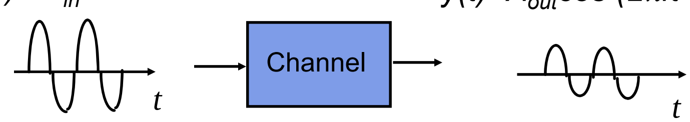
Transmission Impairments

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

Frequency Domain Channel Characterization

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

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



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

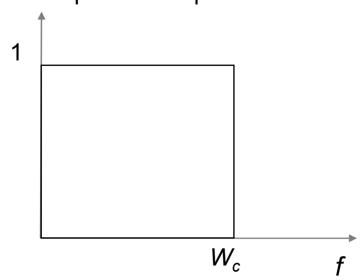
- Apply sinusoidal input at frequency *f*
  - Output is sinusoid at same frequency, but attenuated & phase-shifted
  - Measure amplitude of output sinusoid (of same frequency *f*)
  - Calculate amplitude response
    - $A(f)$ = ratio of output amplitude to input amplitude
  - If $A(f) \approx 1$, then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

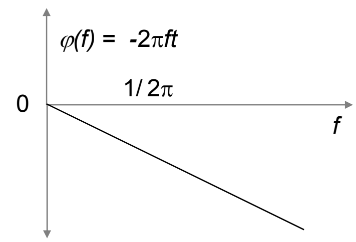
- Ideal filter: all sinusoids with frequency $f<W_c$ are passed without attenuation and delayed by τ seconds; sinusoids at other frequencies are blocked

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

Amplitude Response

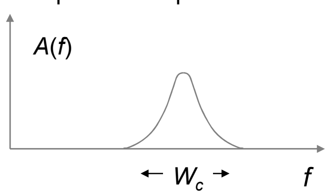


Phase Response



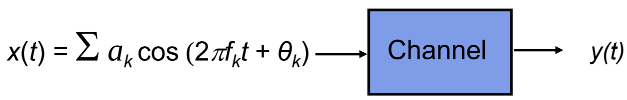
Example: Bandpass Channel

Amplitude Response



- 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

Channel Distortion

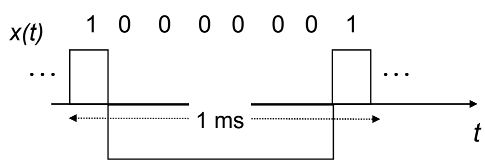


- Let *x(t)* corresponds to a digital signal bearing data information
- How well does *y(t)* follow *x(t)*?

$$y(t) = \sum A(f_k)\, a_k \cos(2\pi f_k t + \theta_k + \Phi(f_k))$$

- Channel has two effects:
  - If amplitude response is not flat, then different frequency components of *x(t)* will be transferred by different amounts
  - If phase response is not flat, then different frequency components of *x(t)* will be delayed by different amounts
- In either case, the shape of *x(t)* is altered

Example: Amplitude Distortion

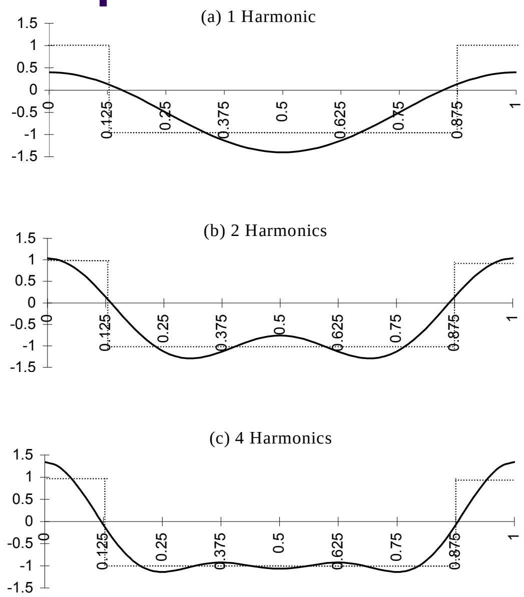


- Let *x(t)* input to ideal lowpass filter that has zero delay and W_c = 1.5 kHz, 2.5 kHz, or 4.5 kHz

$$x(t) = -0.5 + \frac{4}{\pi}\sin(\frac{\pi}{4})\cos(2\pi 1000t) + \frac{4}{\pi}\sin(\frac{2\pi}{4})\cos(2\pi 2000t) + \frac{4}{\pi}\sin(\frac{3\pi}{4})\cos(2\pi 3000t) + \ldots$$

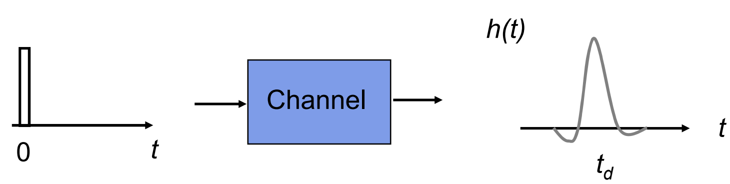
- W_c = 1.5 kHz passes only the first two terms
- W_c = 2.5 kHz passes the first three terms
- W_c = 4.5 kHz passes the first five terms

Amplitude Distortion



- As the channel bandwidth increases, the output of the channel resembles the input more closely

Time-domain Characterization

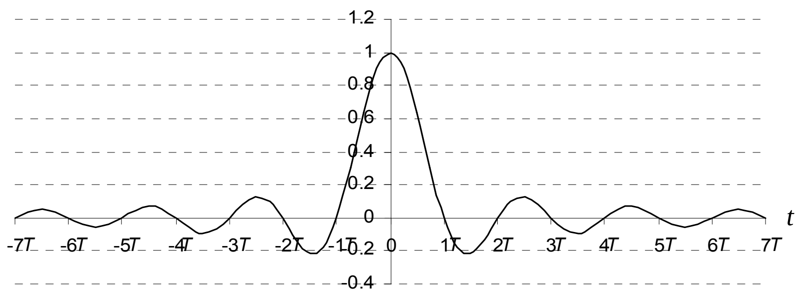


- Time-domain characterization of a channel requires finding the *impulse response h(t)*
- Apply a very narrow pulse to a channel and observe the channel output
  - *h(t)* typically a delayed pulse with ringing
- Interested in system designs with *h(t)* that can be packed closely without interfering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

Nyquist Pulse with Zero Intersymbol Interference

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

$$s(t) = \sin(2\pi W_c t)/2\pi W_c t$$



- $s(t)$ has zero crossings at $t = kT$, $k = \pm 1, \pm 2, \ldots$
- Pulses can be packed every T seconds with *zero interference*

Example of composite waveform

Three Nyquist pulses shown separately

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

Composite waveform

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

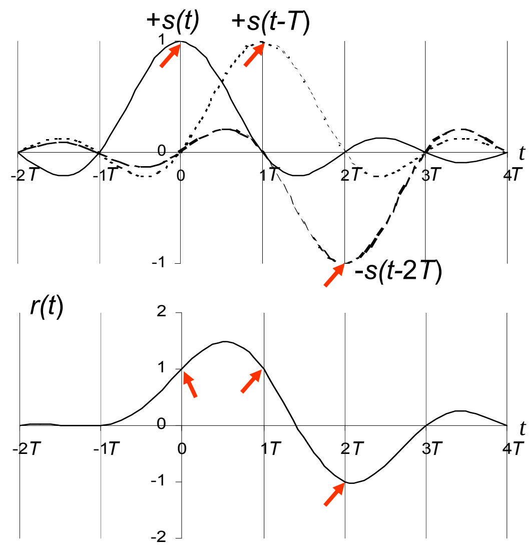
Samples at kT

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

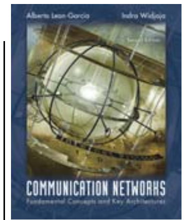
$r(T)=s(T)+s(0)-s(-T)=+1$

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

Zero ISI at sampling times kT



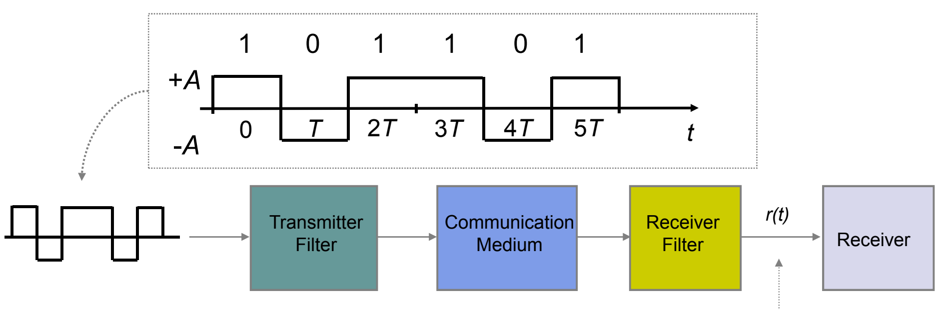
Chapter 3
Digital Transmission Fundamentals



Fundamental Limits in Digital Transmission

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 *s(t)* is a Nyquist pulse, then *r(t)* has zero intersymbol interference (ISI) when sampled at multiples of *T*



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

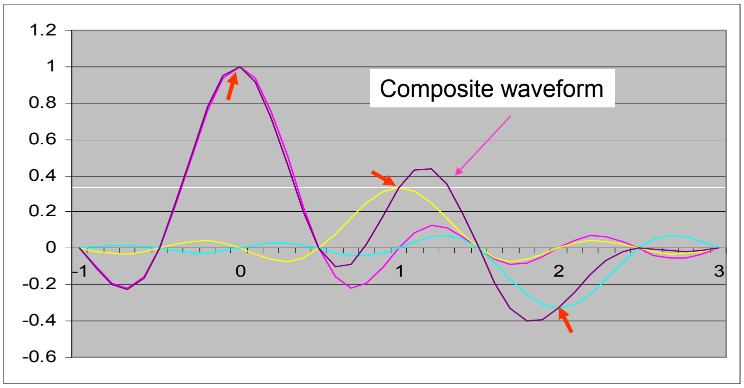
- With $M = 2^m$ signal levels, each pulse carries m bits

  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.*

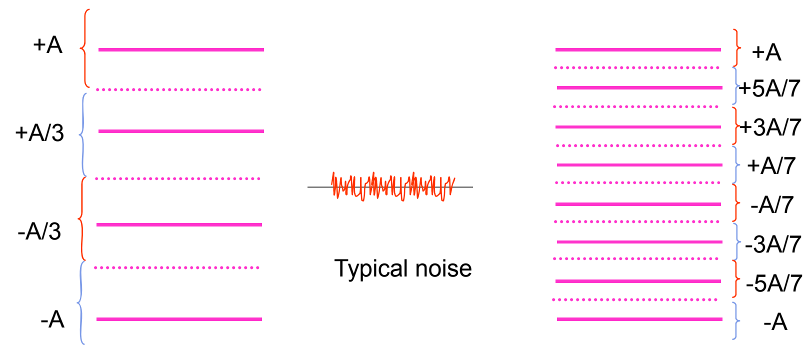
Example of Multilevel Signaling

- 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



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

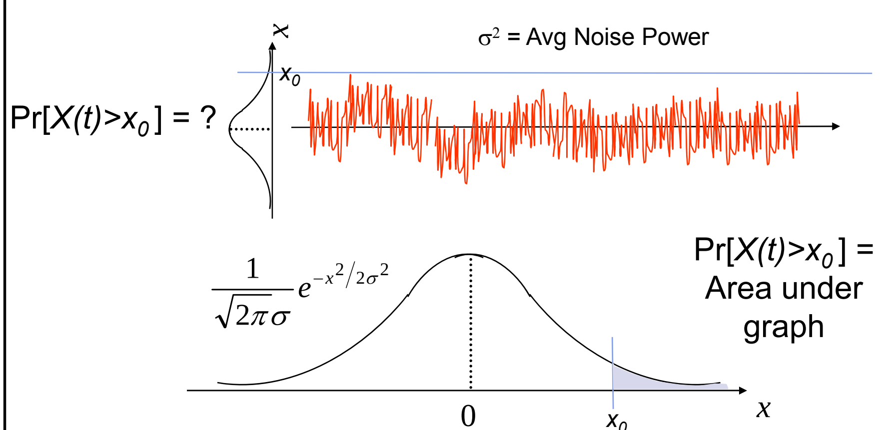


Four signal levels

Eight signal levels

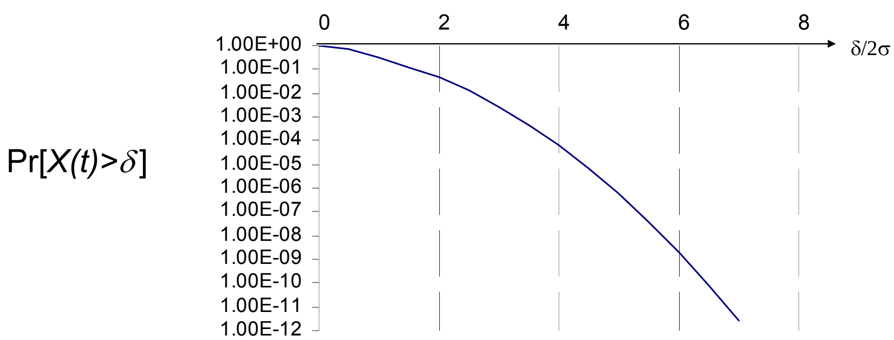
Noise distribution

- 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

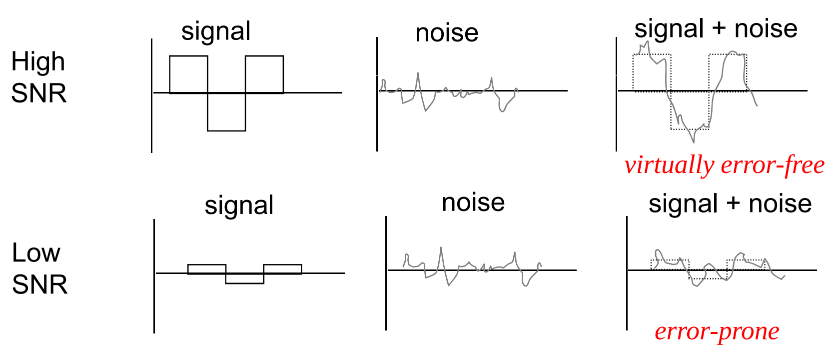


Probability of Error

- Error occurs if noise value exceeds certain magnitude
- Prob. 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



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

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

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

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

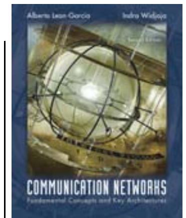
  Bit rate = 2*3000 pulses/sec * 3 bits/pulse = 18 kbps

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

  Implies $S/N = 100$
- Shannon Channel Capacity is then

  $C = 3000 \log (1 + 100) = 19,963$ bits/second

Chapter 3
Digital Transmission Fundamentals

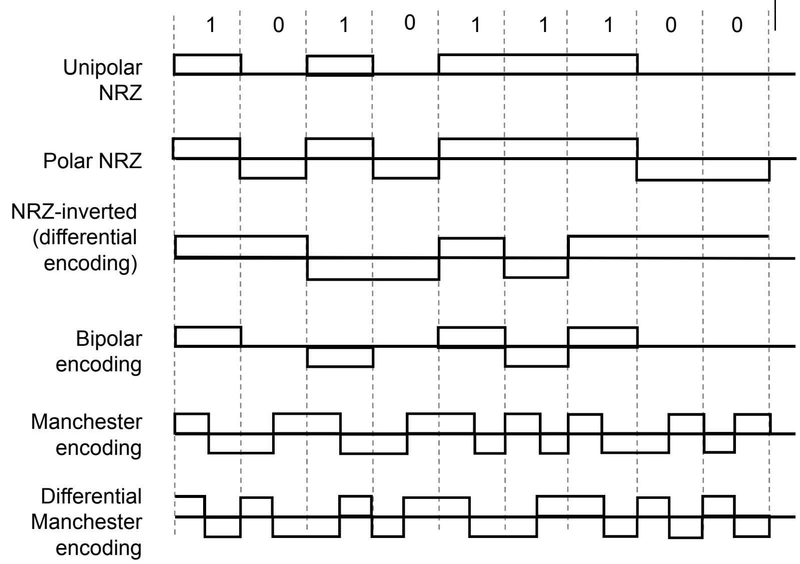


Line Coding

What is Line Coding?

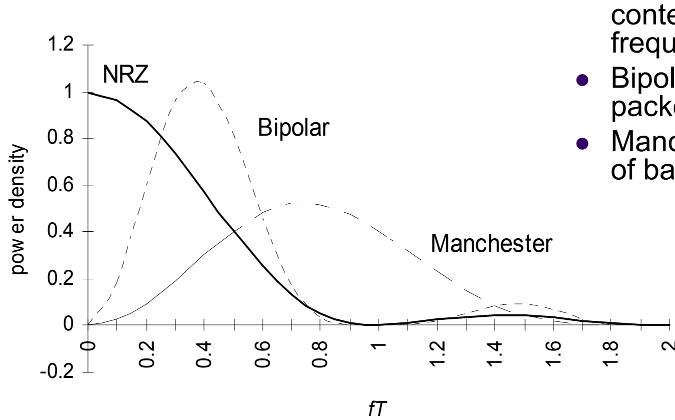
- Mapping of binary information sequence into the digital signal that enters the channel
  - Ex. “1” maps to +A square pulse; “0” to –A pulse
- Line code selected to meet system requirements:
  - *Transmitted power*: Power consumption = $
  - *Bit timing*: Transitions in signal help timing recovery
  - *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



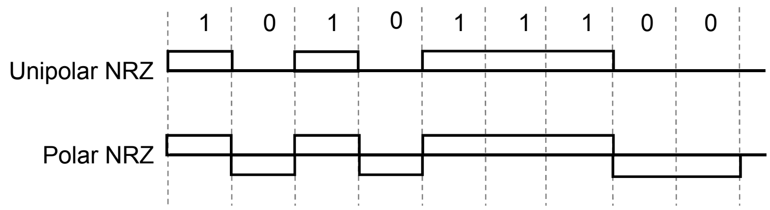
Spectrum of Line codes

- 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)



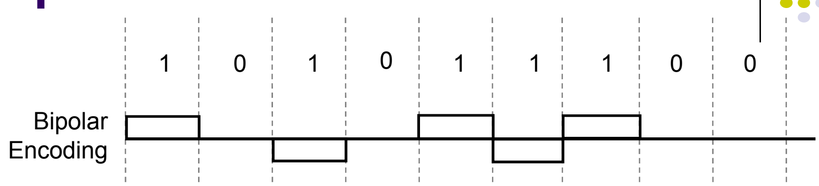
Unipolar NRZ

- "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

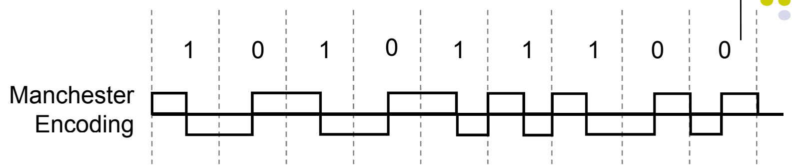
- "1" maps to +A/2 pulse
- "0" maps to –A/2 pulse
- Better Average Power
  $0.5*(A/2)^2 + 0.5*(-A/2)^2 = A^2/4$
- Long strings of +A/2 or –A/2
  - Poor timing
  - Low-frequency content
- Simple

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

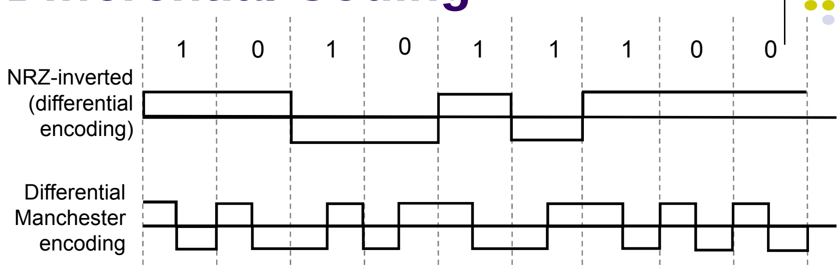
Manchester code & *m*B*n*B codes



- "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