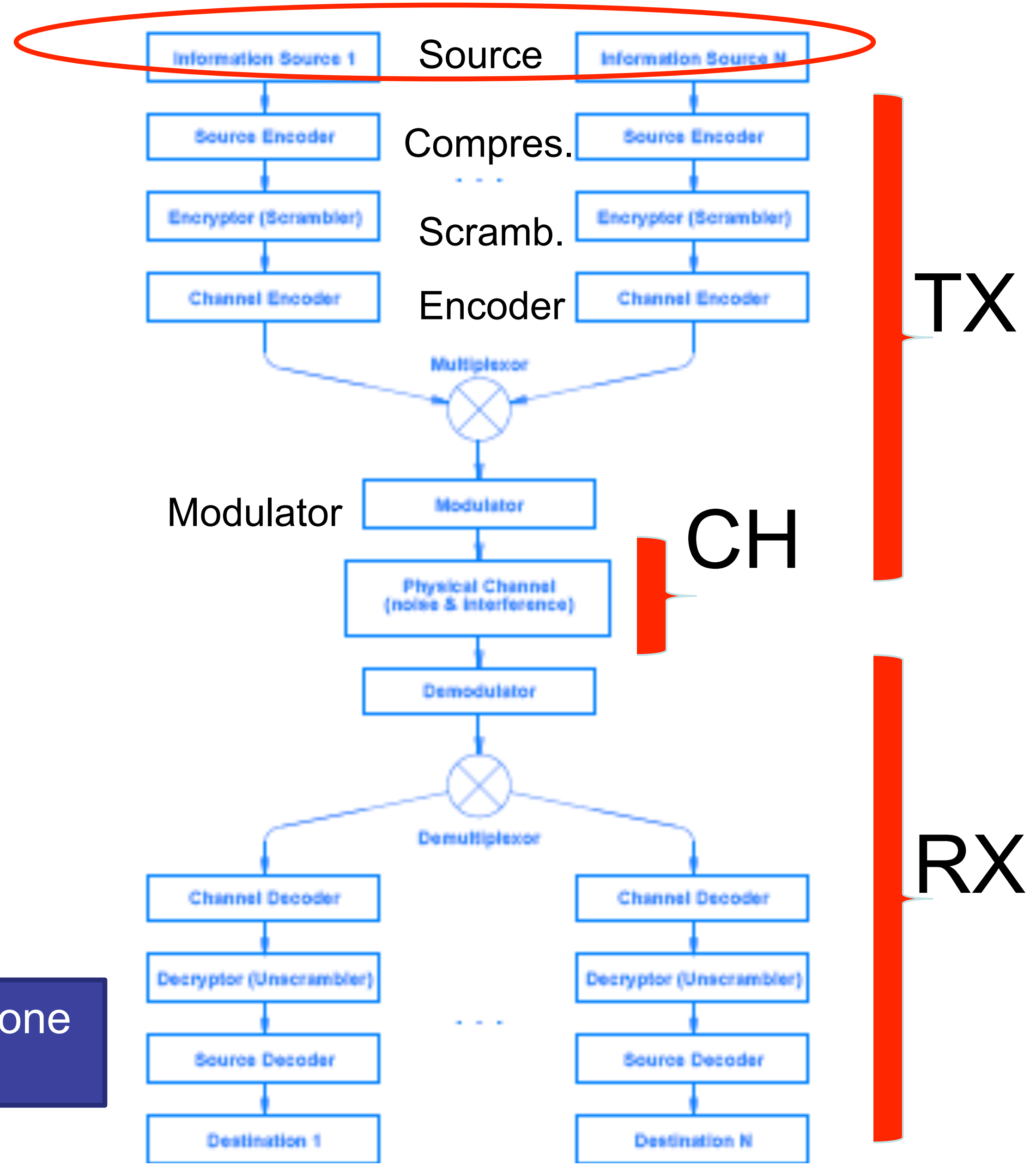
Chapter 6

Information Sources and Signals

Big Idea in Data Communications:

A conceptual framework for a data communications system. Multiple sources send to multiple destinations through an underlying physical channel



Each of the boxes corresponds to one subtopic of data communications:

The Subtopics of Data Communications

- Information Sources
  - The source of information can be either analog or digital
  - Important concepts include characteristics of signals, such as amplitude, frequency, and phase, and classification as either periodic or aperiodic
    - Conversion between analog and digital representations of information
- Source Encoder and Decoder
  - Once information has been digitized, digital representations can be transformed and converted
  - Concepts include data compression and consequences for communications

The Subtopics of Data Communications

- Encryptor and Decryptor
  - To protect information and keep it private, the information can be encrypted (i.e., scrambled) before transmission and decrypted upon reception
  - Concepts include cryptographic techniques and algorithms
- Channel Encoder and Decoder
  - Channel coding is used to detect and correct transmission errors
  - Topics include methods to detect and limit errors
  - Practical techniques like parity checking, checksums, and cyclic redundancy codes that are employed in computer networks
- Multiplexor and Demultiplexor
  - Multiplexing refers to the way information from multiple sources is combined for transmission across a shared medium
  - Concepts include techniques for simultaneous sharing as well as techniques that allow sources to take turns when using the medium

The Subtopics of Data Communications

- Modulator and Demodulator
  - Modulation refers to the way electromagnetic radiation is used to send information
  - Concepts include both analog and digital modulation schemes
  - Devices known as modems that perform the modulation and demodulation
- Physical Channel and Transmission
  - transmission media
  - transmission modes, such as serial and parallel
  - channel bandwidth
  - electrical noise and interference
  - channel capacity

Signals &
Signal Sources

Signal Characteristics

- Analog (continuous) or digital (discrete)
- Periodic or aperiodic
- Components of a periodic electromagnet wave signal
  - Amplitude (maximum signal strength) – e.g., in V
  - Frequency (rate at which the a periodic signal repeats itself) – expressed in Hz
  - Phase (measure of relative position in time within a single period) – in deg or radian (2π = 360 = 1 period)

Periodic:

$$S(t) = S(t + T)$$

$$S(t) = A\sin(2\pi ft + \varphi)$$

$$\varphi = phase$$

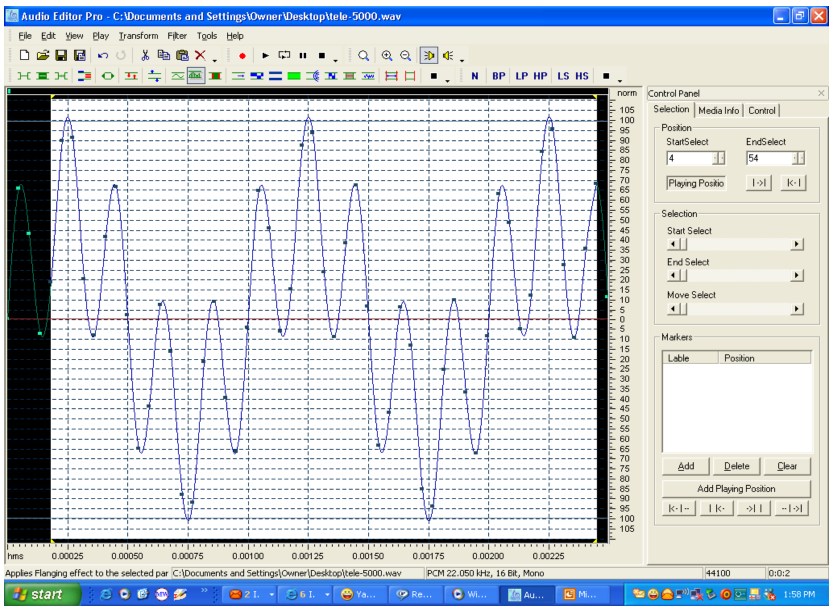
$$A = amplitude$$

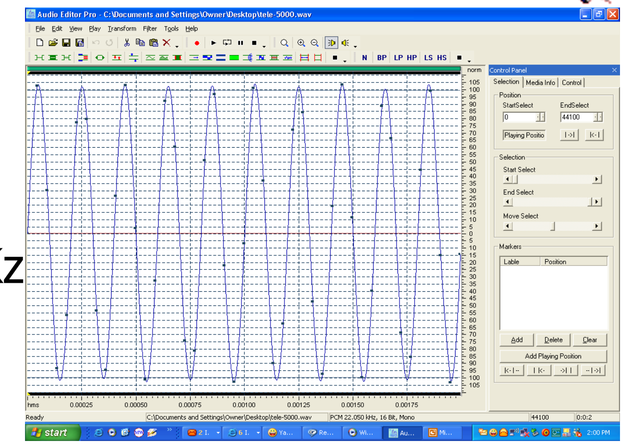
$$f = frequncy$$

$$T = period = 1/f$$

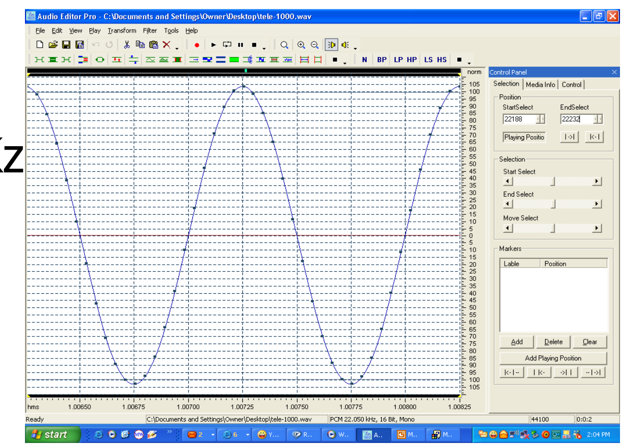
Sound Wave Examples

Each signal is represented by $x(t) = \sin(2\pi f.t)$





f = 5Kz



f = 1Kz

A dual tone signal with f1 and f2 is represented by
$x(t) = \sin(2\pi f1.t) + \sin(2\pi f2.t)$

Taylor Series

- Complex signals are often broken into simple pieces
- Signal requirements
  - Can be expressed into simpler problems
  - Is linear
  - The first few terms can approximate the signal

- Example: The Taylor series of a real or complex function *f* (*x*) is the power series

$$f(a)+\frac{f'(a)}{1!}(x-a)+\frac{f''(a)}{2!}(x-a)^2+\frac{f^{(3)}(a)}{3!}(x-a)^3+\cdots$$

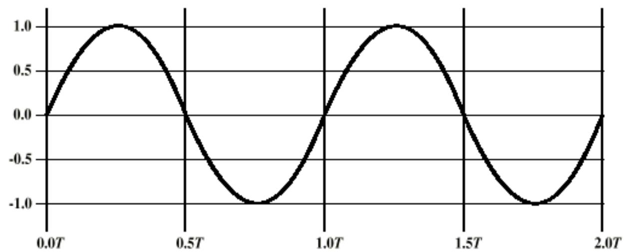
$$\sum_{n=0}^{\infty}\frac{f^{(n)}(a)}{n!}(x-a)^n$$

Signal Representation

- Fourier Representation: We can represent all complex signals as harmonic series of simpler signals
- Frequency components of the square wave with amplitude A can be expressed as

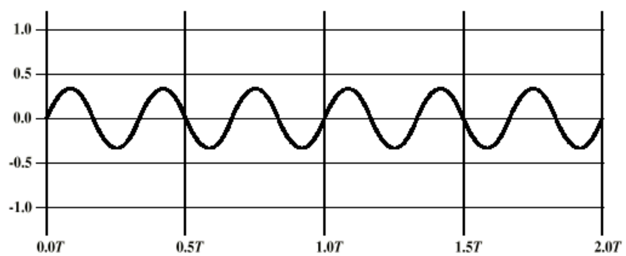
$$s(t) = A \times \frac{4}{\pi} \sum_{K=1/odd}^{\infty} \frac{\sin(2\pi kft)}{k}$$

Square Wave



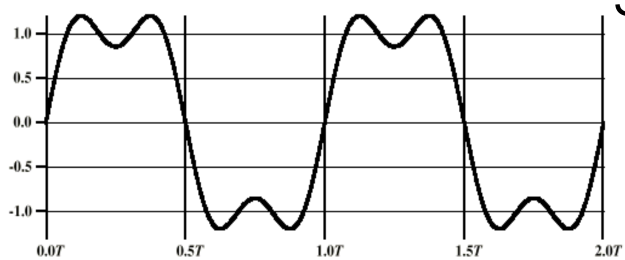
(a) sin (2 ft)

$S(t)=\sin(2\pi ft)$



(b) (1/3) sin (2 (3f)t)

$S(t)=1/3[\sin(2\pi(3f)t)]$



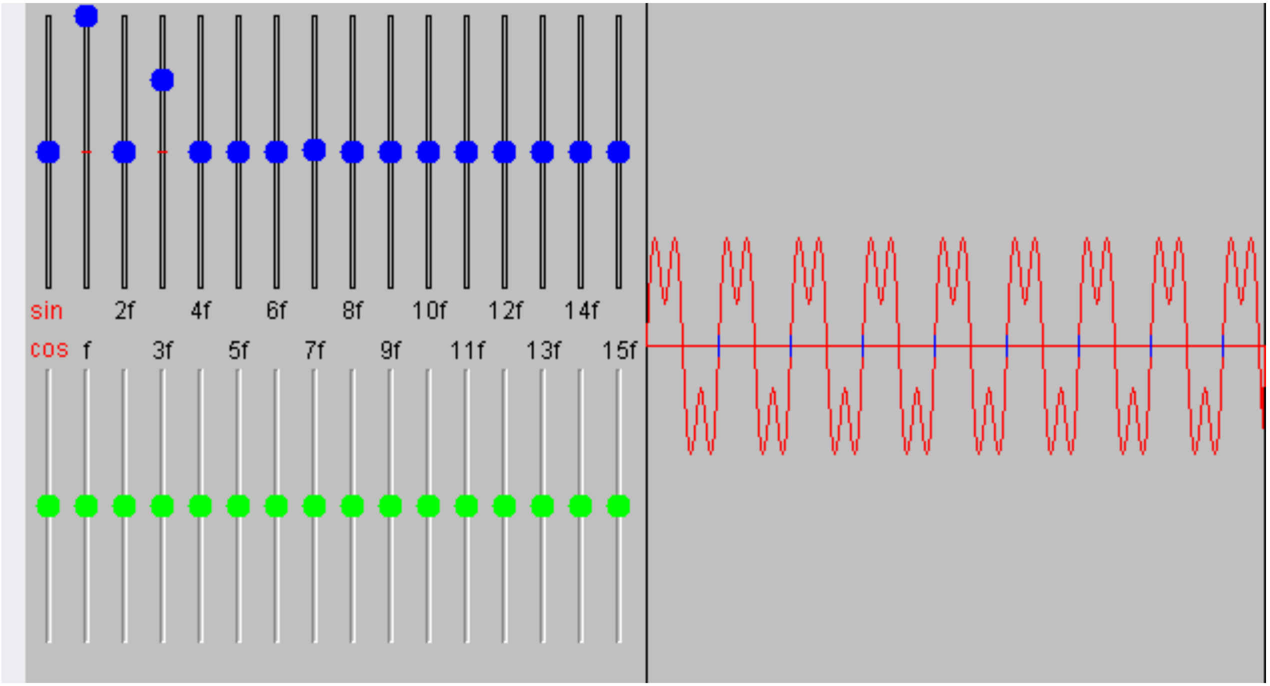
(c) (4/) [sin (2 ft) + (1/3) sin (2 (3f)t)]

$S(t)= 4/\pi\{\sin(2\pi ft) +1/3[\sin(2\pi(3f)t)]\}$

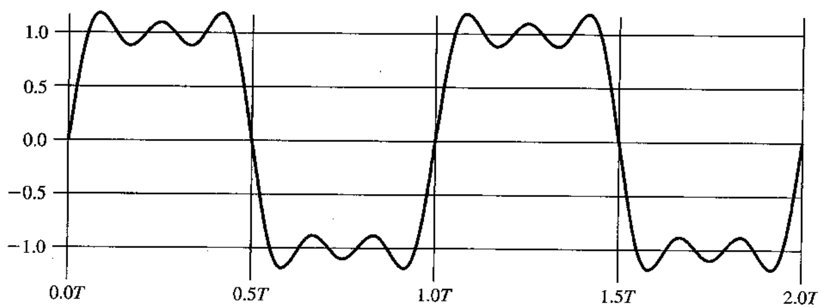
$$s(t) = A \times \frac{4}{\pi} \sum_{K=1/odd}^{\infty} \frac{\sin(2\pi kft)}{k}$$

Example

- http://www.phy.ntnu.edu.tw/ntnujava/index.php?topic=17

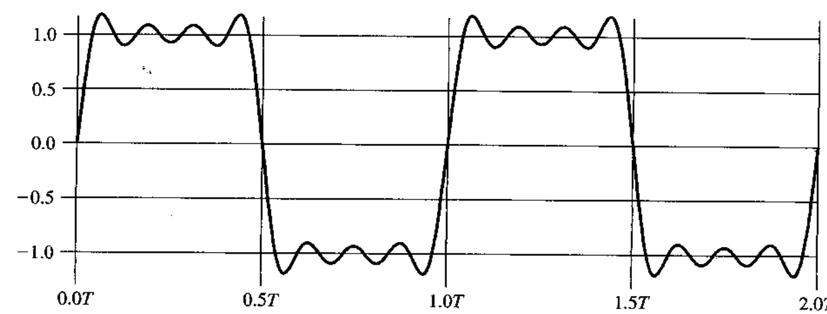


Square Wave



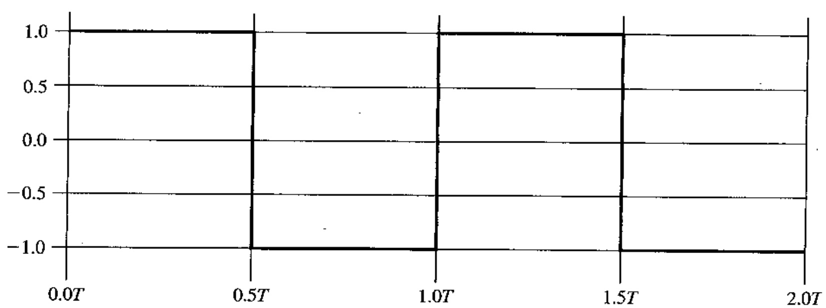
(a) $(4/\pi)\,[\sin(2\pi ft) + (1/3)\sin(2\pi(3f)t) + (1/5)\sin(2\pi(5f)t)]$

K=1,3,5



(b) $(4/\pi)\,[\sin(2\pi ft) + (1/3)\sin(2\pi(3f)t) + (1/5)\sin(2\pi(5f)t) + (1/7)\sin(2\pi(7f)t)]$

K=1,3,5, 7



(c) $(4/\pi)\,\Sigma\,(1/k)\sin(2\pi(kf)t)$, for k odd

K=1,3,5, 7, 9, …..

Frequency Components of Square Wave

$$s(t) = A \times \frac{4}{\pi} \sum_{K=1/odd}^{\infty} \frac{\sin(2\pi kft)}{k}$$

Fourier Expansion

Periodic Signals

- A Periodic signal/function can be approximated by a sum (possible infinite) sinusoidal signals.
- Consider a periodic signal with period T
- A periodic signal can be Real or Complex
- The fundamental frequency: ωo
- Example:
  - Prove that x(t) is periodic:

$$Periodic \Rightarrow x(t+nT) = x(t)$$
$$\mathrm{Re}\,al \rightarrow x(t) = \cos(\omega_o t + \theta)$$
$$Complex \rightarrow x(t) = Ae^{j\omega_o t}$$
$$\omega_o = 2\pi / T_o$$
$$T_o = 2\pi / \omega_o$$

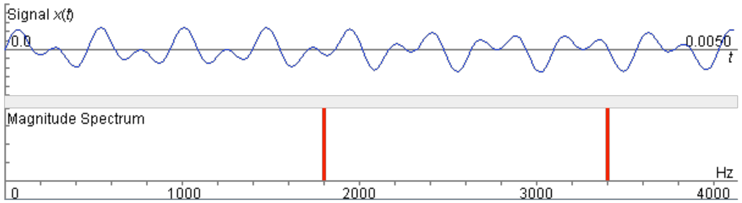
$$x(t) = \cos(\omega_o t + \theta)$$

Note: cos(A + B) = cosAcosB - sinAsinB

Frequency Spectrum

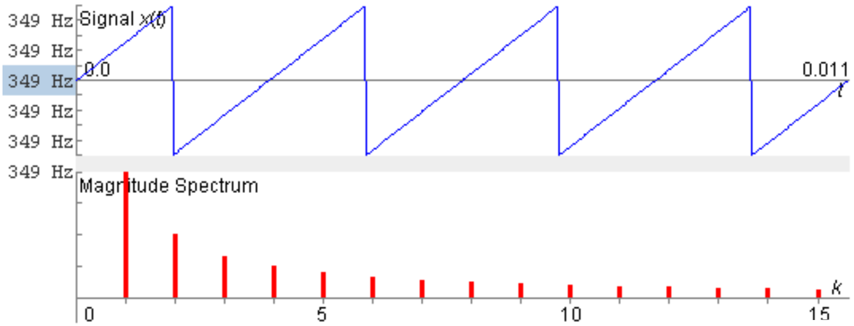
- We can plot the *frequency spectrum* or *line spectrum* of a signal
  - In Fourier Series k represent harmonics
  - Frequency spectrum is a graph that shows the amplitudes and/or phases of the Fourier Series coefficients *Ck*.
    - Amplitude spectrum **|Ck|=4A/k.pi**
    - The lines |Ck| are called line spectra because we indicate the values by lines

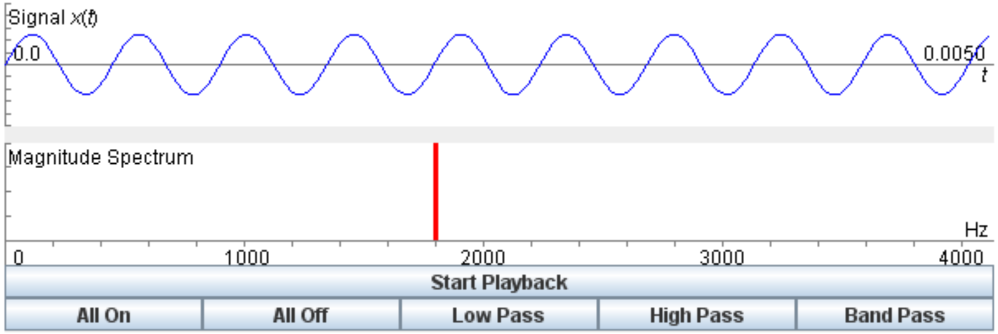
$$s(t) = A \times \frac{4}{\pi} \sum_{K=1/odd}^{\infty} \frac{\sin(2\pi kft)}{k}$$



Examples

- http://www.jhu.edu/~signals/listen-new/listen-newindex.htm



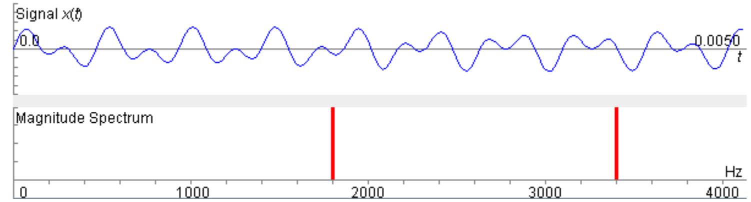


Periodic Signal Characteristics

- A signal can be made of many frequencies
  - All frequencies are multiple integer of the *fundamental frequency*
  - Spectrum of a signal identifies the range of frequencies the signal contains
  - Absolute bandwidth is defined as: Highest_Freq – Lowest_Freq
  - Bandwidth in general is defined as the frequency ranges where a signal has its most of energies
- Signal data rate
  - Information carrying capacity of a signal
  - Expressed in bits per second (bps)
  - Typically, the larger frequency → larger data rate

Example →

Periodic Signals



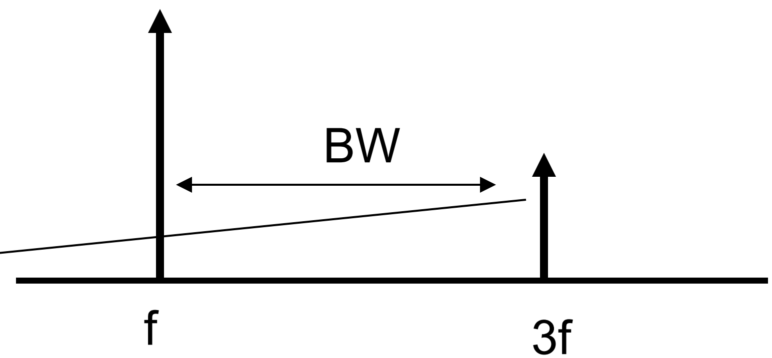
- Consider the following signal
  - Consists of two freq. component (f) and (3f) with BW = 2f

$$S(t) = (4/\pi)\sin(2\pi f t) + (4/3\pi)\sin(2\pi(3f)t)$$

$$Fundamental\_freq = f$$

$$Max\_freq = 3f$$

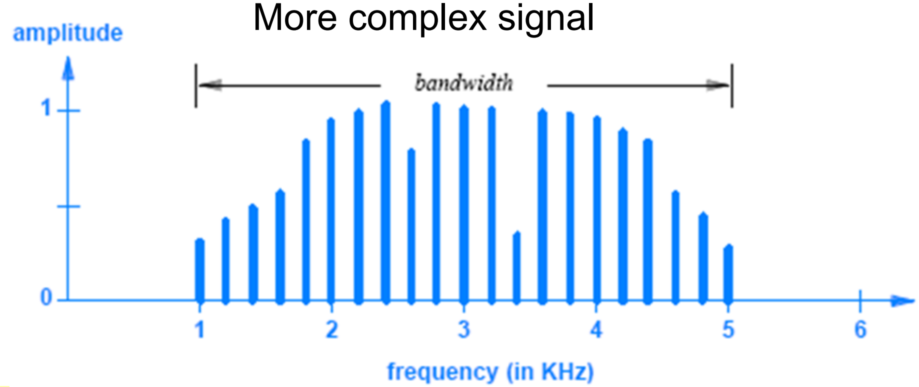
$$Abs\_BW = 3f - f = 2f$$



What is the Max amplitude of this component?

$$s(t) = A \times \frac{4}{\pi} \sum_{K=1/odd}^{\infty} \frac{\sin(2\pi k f t)}{k}$$

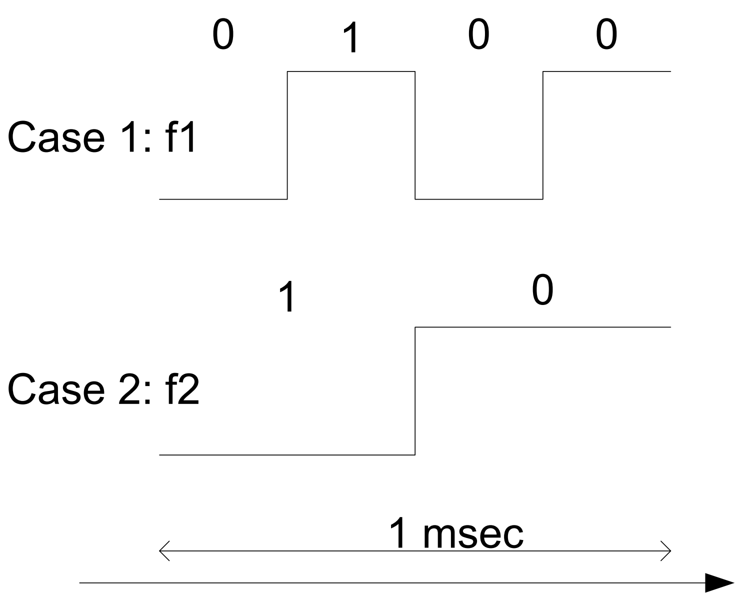
More complex signal



http://www.jhu.edu/~signals/listen-new/listen-newindex.htm

Data Rate & Frequency

- Example:
  - What is the data rate in case 1?
  - What is the data rate in case 2?
  - Which case has larger data rate? (sending more bits per unit of time)



- Case I data rate=one bit per (0.25msec)
  - → 4 Kbps
- Case II: f2 = 1 KHz → data rate=2Kbps
- Case 1 has higher data rate (bps)

Bandwidth and Data Rate

- Case 1:
  - Assume a signal has the following components: f, 3f, 5f ; f=10^6 cycles/sec
  - What is the Absolute BW (Hz)?
  - What is the period?
  - How often can we send a bit? (bit/sec)
  - What is the data rate? (bps)
  - Express the signal equation in time domain

BW=4MHz
T=1usec
1 bit every 0.5usec
Data rate=2*f=2bit/usec=2Mbps

$$s(t) = A \times \frac{4}{\pi} \sum_{K=1/odd}^{\infty} \frac{\sin(2\pi kft)}{k}$$

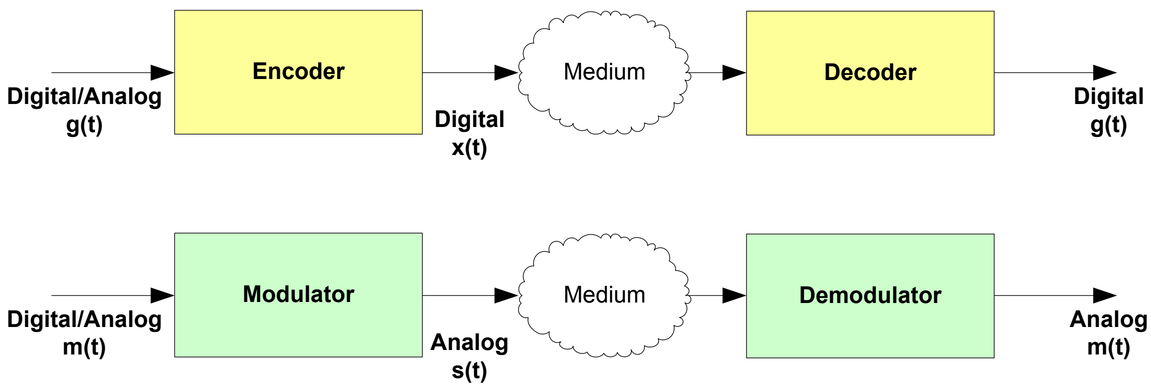
Bandwidth and Data Rate

- Case 2:
  - Assume a signal has the following components: f, 3f ; f=2x10^6 cycles/sec
  - What is the Absolute BW?
  - What is the period?
  - How often can we send a bit?
  - What is the data rate (bps)?
  - Express the signal equation in time domain

BW=4MHz
T=0.5 usec
1 bit every 0.25usec
Data rate=2*f=4bit/usec=4x10^6bps

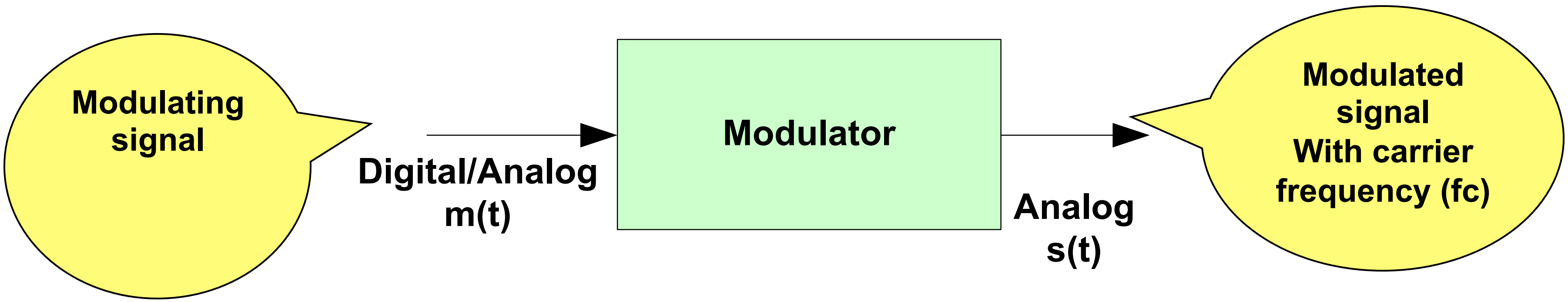
Remember: Greater BW → Faster system → Higher cost;
Greater BW → more potential distortion;

Typical Modulation and Coding



		Signal Transmitted	
		Digital	**Analog**
Original Data	**Digital**	**NRZ/ Multilevel/ Biphase**	**ASK** **FSK/BFSK/MFSK** **PSK/BPSK/MPSK**
	Analog	**PCM** **PAM** **DM**	**AM** **PM** **FM**

What is Modulation or Encoding?



Changing signal characteristics including
-Phase
-Amplitude
-Frequency

Depending on the medium, signal range, and data Properties different encoding techniques can be used

Reasons for Choosing Different Encoding/Modulation Techniques

- Digital data, digital signal
  - Less complex equipments
  - Less expensive than digital-to-analog modulation equipment
- Analog data, digital signal
  - Permits use of modern digital transmission and switching equipment
  - Requires conversion to analog prior to wireless transmission

Reasons for Choosing Encoding Techniques

- Digital data, analog signal
  - Some transmission media will only propagate analog signals
  - E.g., optical fiber and unguided media
- Analog data, analog signal
  - Analog data in electrical form can be transmitted easily and cheaply
  - Done with voice transmission over voice-grade lines

Used in Wireless!

Some Terms

- Unipolar - signal elements have the same sign
- Polar - One logic state represented by positive voltage
- Bit Period - Duration or length of a bit
- Modulation rate – baud rate
- Remember:
  - Modulation rate (baud) x $\log_2$ M = data rate (channel capacity)
  - M is the number of signal levels (symbols)
  - M = 2^L ; L is the number of bits used per symbol

Interpreting Digital Signals

- Receiver needs to know
  - timing of bits - when they start and end
  - signal levels
- Factors affecting signal interpretation
  - signal to noise ratio
  - data rate
  - bandwidth
  - encoding scheme – affects performance

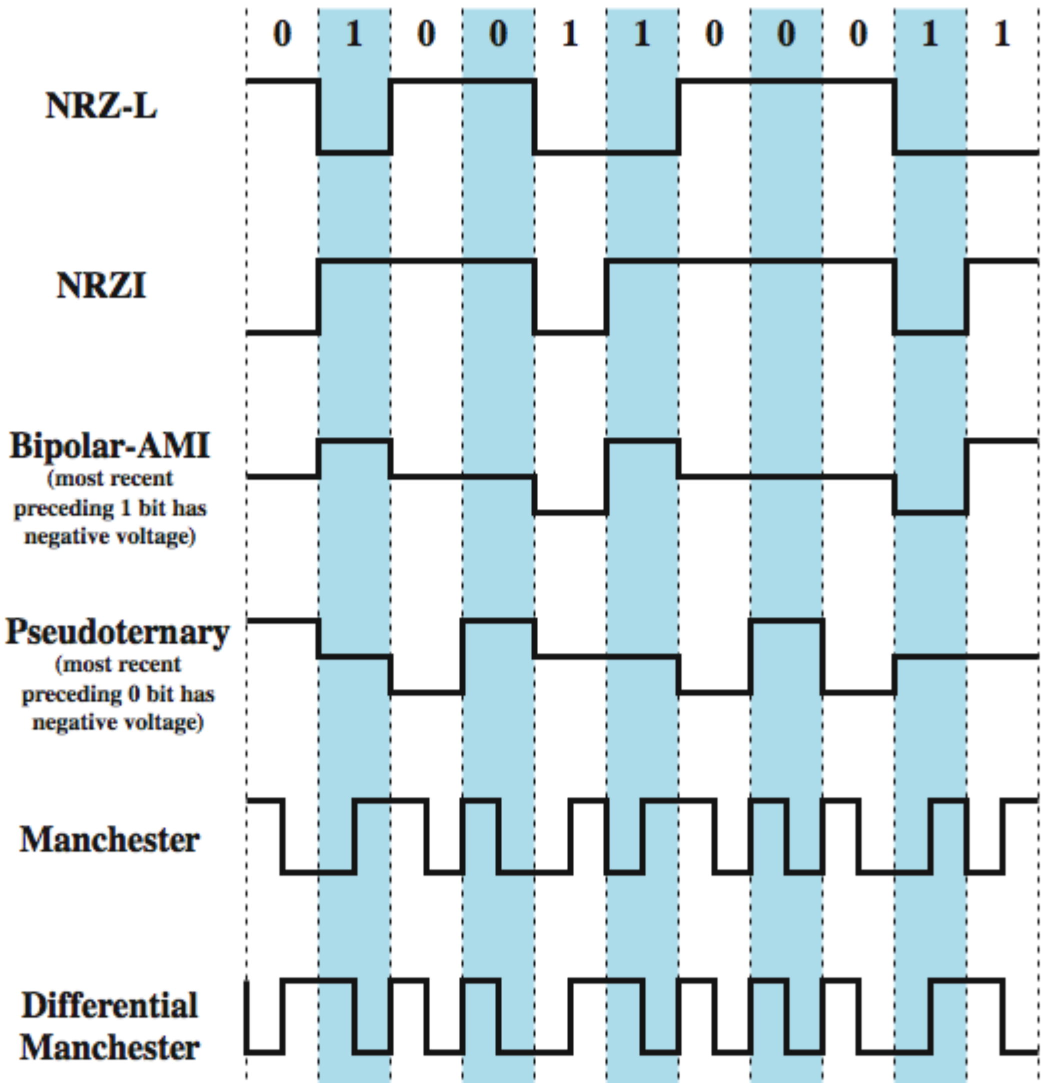
1- $R \propto BER$
2- $SNR \propto 1/BER$
3- $BW \propto BER$

- An increase in data rate <u>increases</u> bit error rate
- An increase in SNR <u>decreases</u> bit error rate
- An increase in bandwidth allows an <u>increase</u> in data rate

Signal Encoding Design Goals

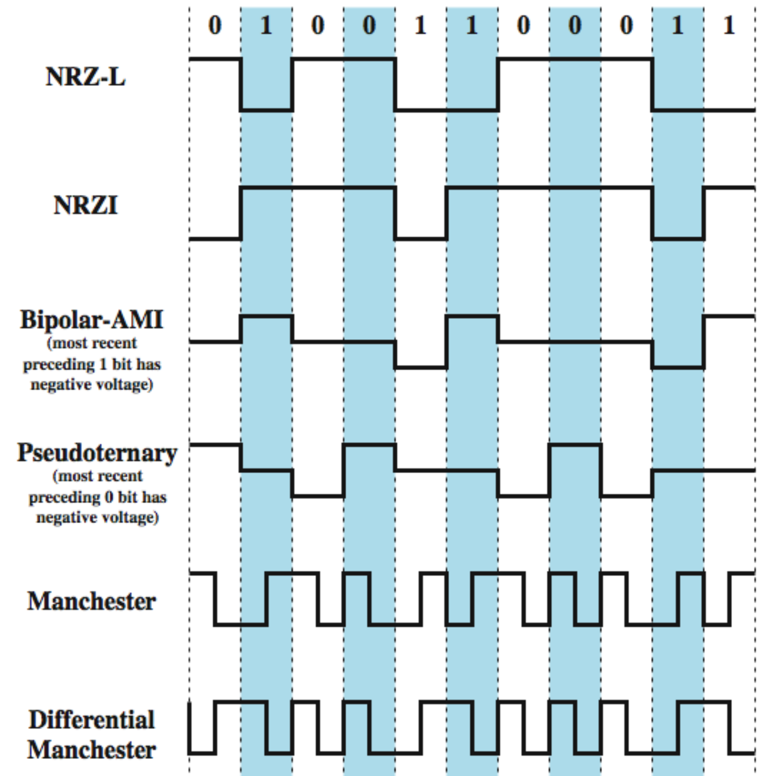
- No DC components
- No long sequence of zero-level line signals
- No reduction in data rate
- Error detection ability
- Low cost

Encoding Schemes (Line Coding Mechanisms)



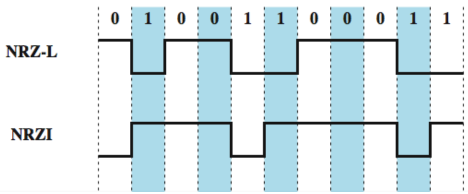
Nonreturn to Zero-Level (NRZ-L)

- two different voltages for 0 and 1 bits
  - 0= high level / 1 = low level



NRZI (Nonreturn to Zero – Invert on ones)

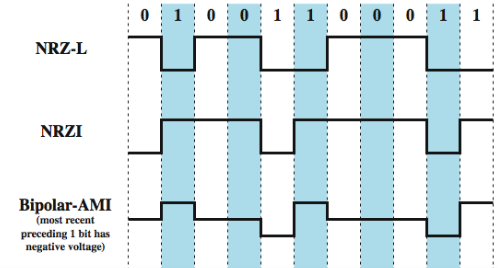
- Non-return to zero, inverted on ones
- constant voltage pulse for duration of bit
- data encoded as presence or absence of signal transition at the beginning of bit time
  - Data is based on transitions (low to high or high to low) – level change
  - Where there is a ONE → Transition occurs
  - Where there is a ZERO→ No transition occurs
- Advantages
  - data represented by changes rather than levels
  - more reliable detection of transition rather than level – when noise exists!



NRZI
Transition when we have a ONE
Otherwise→ no transition

NRZI is Differential encoding: information is transmitted based on changes between successive signal elements

Multilevel Binary
Bipolar-AMI

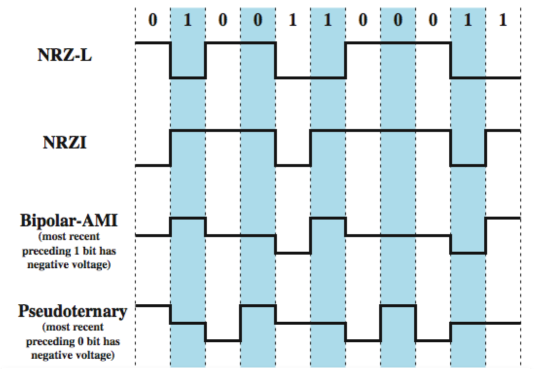


- AMI stands for alternate mark inversion
- Use more than two levels
- Bipolar-AMI
  - zero represented by no line signal
  - one represented by positive or negative pulse
  - One’s pulses alternate in polarity
  - no loss of sync if a long string of ones
    - long runs of zeros still a problem
  - no net dc component
  - lower bandwidth
  - easy error detection

Bipolar - AMI
0 → 0
1,1→ +,-

Multilevel Binary Pseudoternary

- one represented by absence of line signal
- zero represented by alternating positive and negative
- no advantage or disadvantage over bipolar-AMI
- each used in some applications



1 → 0
0,0 → +,-

Multilevel Binary Issues

- synchronization with long runs of 0's or 1's
  - can insert additional bits, e.g.,ISDN
  - scramble data
- not as efficient as NRZ
  - each signal element only represents one bit
    - receiver distinguishes between three levels: +A, -A, 0
  - In a 3 level system each signal element (representing $\log_2 3$ = 1.58 bits) bears ONE bit of information
    - In this case 1.58 bits represent 1 bit of information!
    - Requires approx. 3dB more signal power for same probability of bit error (lower S/N ratio)
  - **the bit error rate for NRZ codes**, at a given signal-to-noise ratio, **is significantly less than** that for multilevel binary

http://en.wikipedia.org/wiki/Spectral_efficiency

The Spectral Efficiency (bps/Hz) & Modulation Efficiency (bit/symbol)

- The modulation efficiency in bit/s is the gross bitrate (including any error-correcting code) divided by the bandwidth (= Bitrate/BW)
  - Used to compare performance of different digital modulations
- Normalized Spectral Efficiency: Number of bits that can be propagated through the BW for each Hz → The more the better
- Example: A transmission technique using one kilohertz of bandwidth to transmit 1,000 bits per second has a modulation efficiency of 1 (bit/s)/Hz (1000bps/1KHz=1)
- Example: A V.92 modem for the telephone network can transfer 56,000 bit/s downstream and 48,000 bit/s upstream over an analog telephone network. Due to filtering in the telephone exchange, the frequency range is limited to between 300 hertz and 3,400 hertz, corresponding to a bandwidth of 3,400 − 300 = 3,100 hertz.
  - The spectral efficiency or modulation efficiency is 56,000/3,100 = 18.1 (bit/s)/Hz downstream,
  - 48,000/3,100 = 15.5 (bit/s)/Hz upstream

Example

- For an 8-PSK (3 bits generating 8 symbols) system with bit rate of 24 kbps find:
  - Baud (modulation rate)
  - Minimum BW
  - BW Efficiency

Baud = capacity or data rate / number of bits =

= $24000/\log_2 8$ = 8000 baud

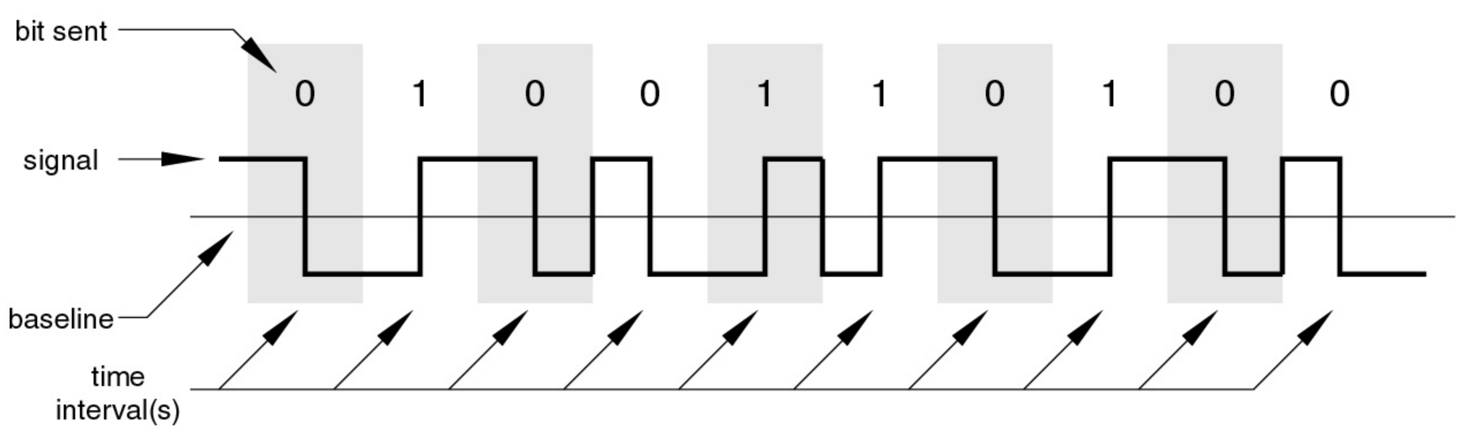
BW = Channel capacity x number of bits = 8000 Hz

BW Eff = Spectral Eff. = channel capacity / BW = 2400/8000 = 3

Manchester Encoding

- has transition in the middle of each bit period
- transition serves as clock and data
- low to high represents one (1= 0 to 1)
- high to low represents zero (0=1 to 0)
- used by IEEE 802.3 - IEEE standards defining the Physical Layer and Data Link Layer's media access control (MAC) sublayer of wired Ethernet

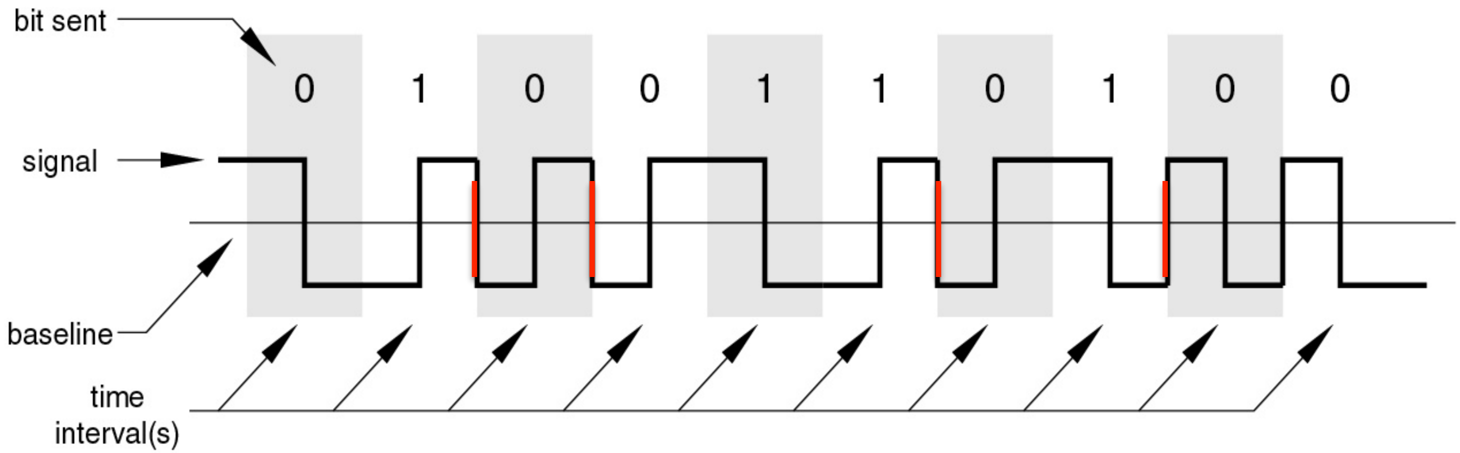
Manchester Encoding



Differential Manchester Encoding

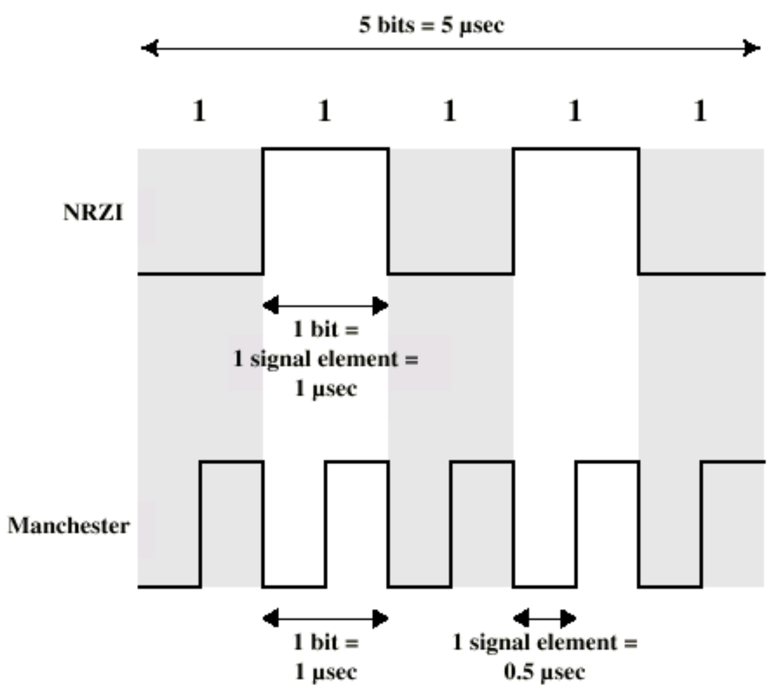
- Mid-bit transition is always there and represents clocking only
- transition at start of bit period representing 0
- no transition at start of bit period representing 1
  - this is a differential encoding scheme
- used by IEEE 802.5 - token ring LAN

Differential Manchester Encoding

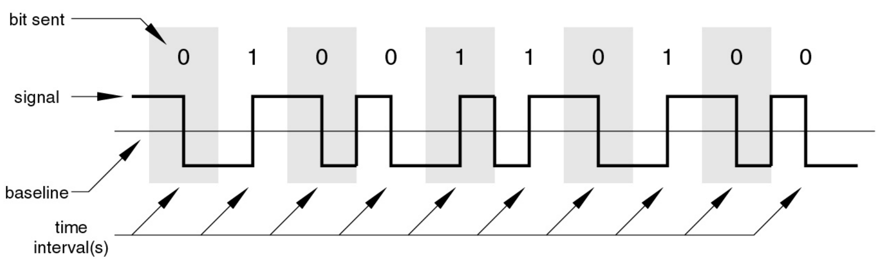


Biphase Pros and Cons

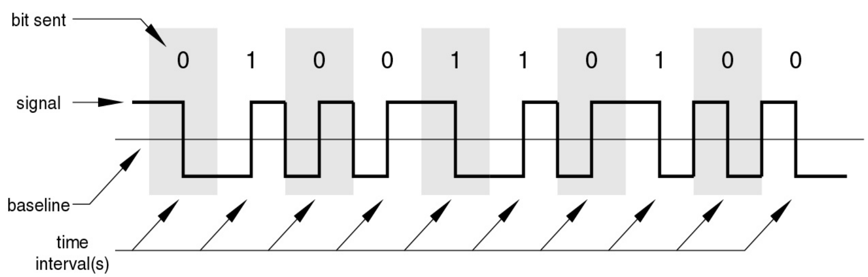
- Con
  - at least one transition per bit time and possibly two (if differential)
  - maximum modulation rate is twice NRZ
  - requires more bandwidth
- Pros
  - synchronization on mid bit transition (self clocking)
  - has no dc component
  - has error detection



Manchester Encoding



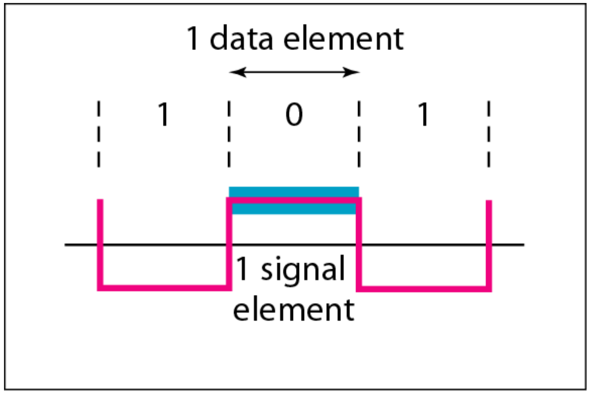
Differential Manchester Encoding



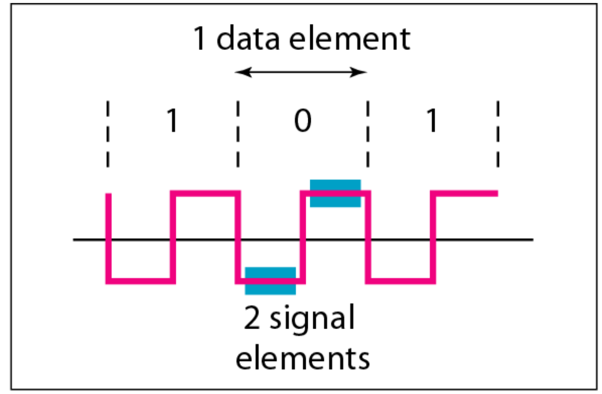
NRZ Pros & Cons

- Pros
  - easy to engineer
  - make good use of bandwidth
- Cons
  - dc component (too many ones or zeros; average is not zero))
  - lack of synchronization capability
- Commonly used for magnetic recording
- Not often used for signal transmission

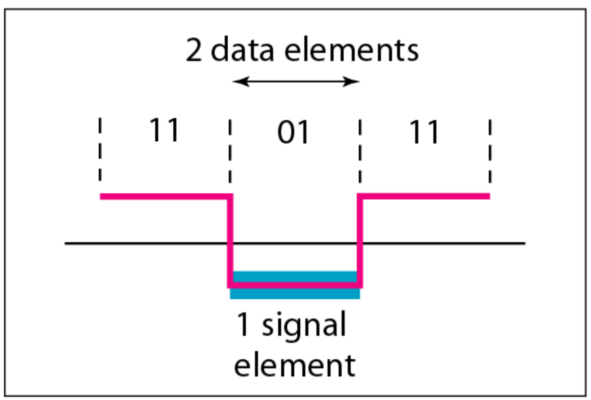
Signal Element and Data Element



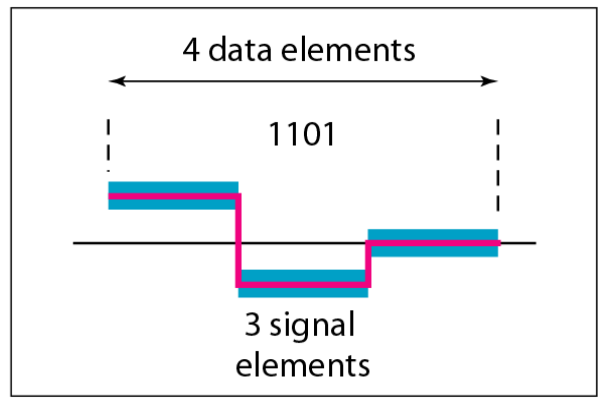
a. One data element per one signal element (r = 1)



b. One data element per two signal elements $\left(r = \frac{1}{2}\right)$



c. Two data elements per one signal element (r = 2)



d. Four data elements per three signal elements $\left(r = \frac{4}{3}\right)$

r = data_element / signal_element

Best case is c! With only one signal element we are sending 2 bits!

Data Rate and Signal Rate

- The data rate defines the number of bits sent per sec – bps.
  - It is often referred to the bit rate.
- The signal rate is the number of signal elements sent in a second and is measured in bauds .
  - It is also referred to as the modulation rate or baud rate
- **Goal** is to increase the data rate whilst reducing the baud rate

Data Rate and Baud Rate

- The baud or signal rate can be expressed as:

$$D = c \times R \times 1/r \quad (\text{ in bauds})$$

Where R is data rate

c is the case factor (worst, best & avg.)

r is the ratio between data element & signal element

The Goal is to increase the data rate whilst reducing the baud rate (c=const.): r = R/D
→ We want higher **r**

r = data_element / signal_element

Example

A signal is carrying data in which one data element is encoded as one signal element (r = 1). If the bit rate is 100 kbps, what is the average value of the baud rate if c is between 0 and 1 (c=1/2)?

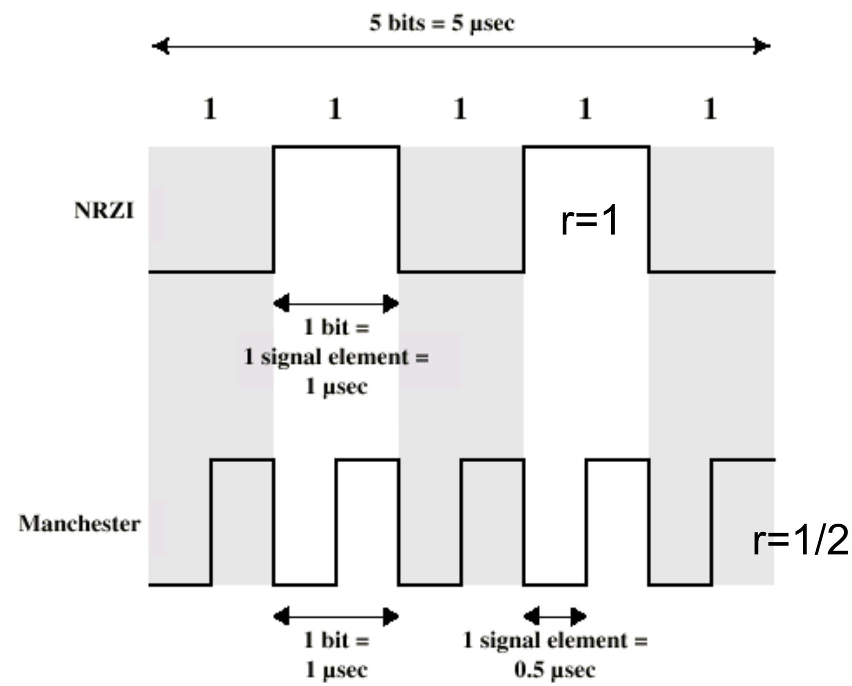
Solution

We assume that the average value of c is 1/2 . The baud rate is then

$$D = c \times R \times 1/r = 0.5 \times 1000 \times 1 = 50 \text{ kbaud}$$

Example

- Using NRZI, how do you represent 1 1 1 1 1?
- Assuming it takes 5usec to send 5 bits what is the duration of each bit?
- Assuming it takes 5usec to send 5 bits what is the duration of each signal element?
  - The signal will be 0 1 0 1 0 (toggling – starting with Zero as the initial state)
  - Each bit = 1 usec
  - Each signal element = 1 usec
- Using Manchester, how do you represent 1 1 1 1 1?
  - The signal will be 01 01 01 01 01 (toggling in the middle of each bit – starting with Zero as the initial state)
  - Each bit = 1 usec
  - Each signal element = 0.5 usec



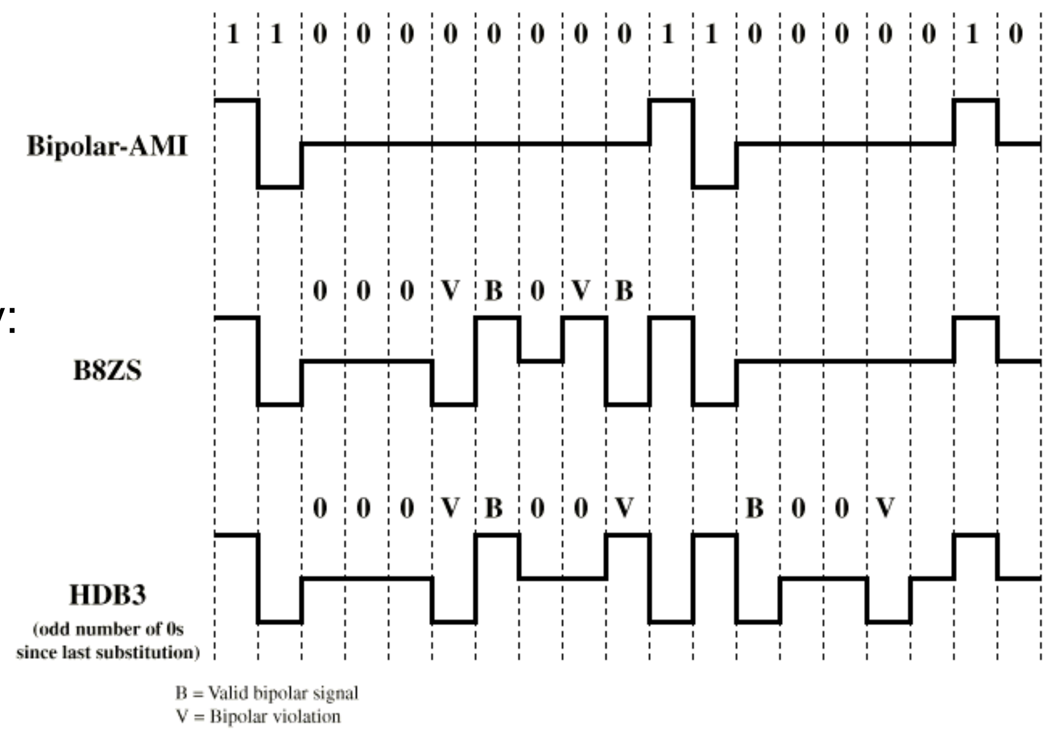
Note that in Bipolar
maximum modulation rate is twice NRZ
c=1; R=constant
D = R/r

NOT GOOD!
We want D to decrease R to increase!

Scrambling

- The objective is to avoid long sequences of zero level line signals and providing some type of error detection capability
- We compare two techniques:
  - B3ZS (bipolar 8-zero substitution)
  - HDB3 (High-density Bipolar-3 zeros)

B8ZS:
One octet of zero is replaced by:
000VB0VB
V = 1 code violation



Scrambling

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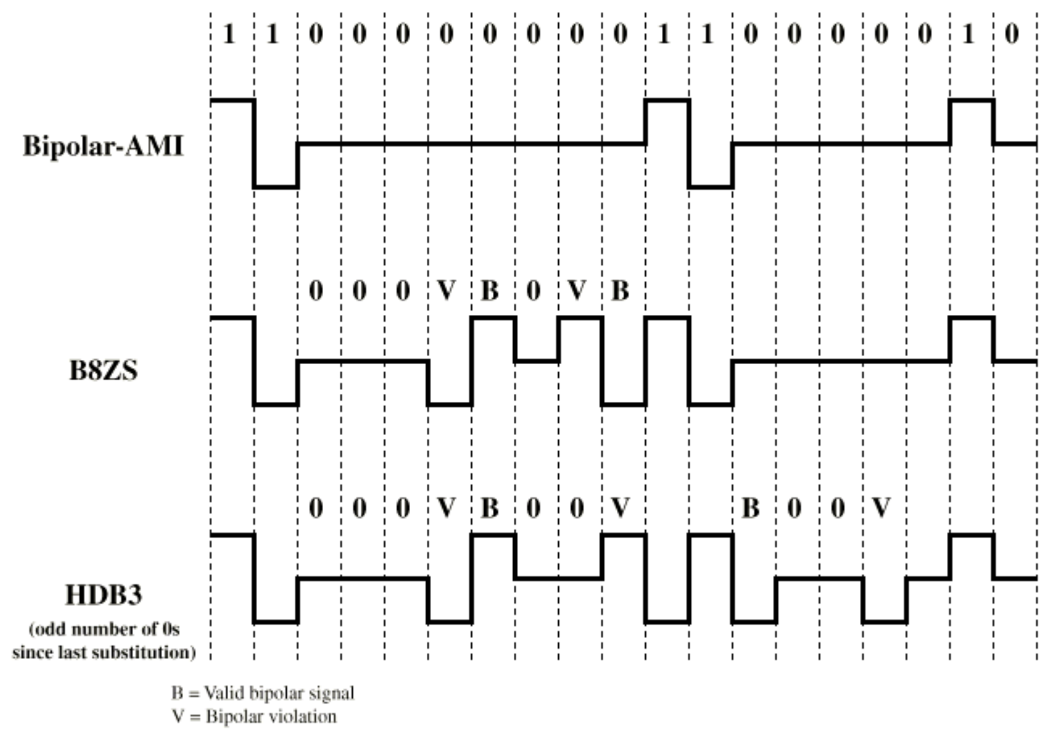


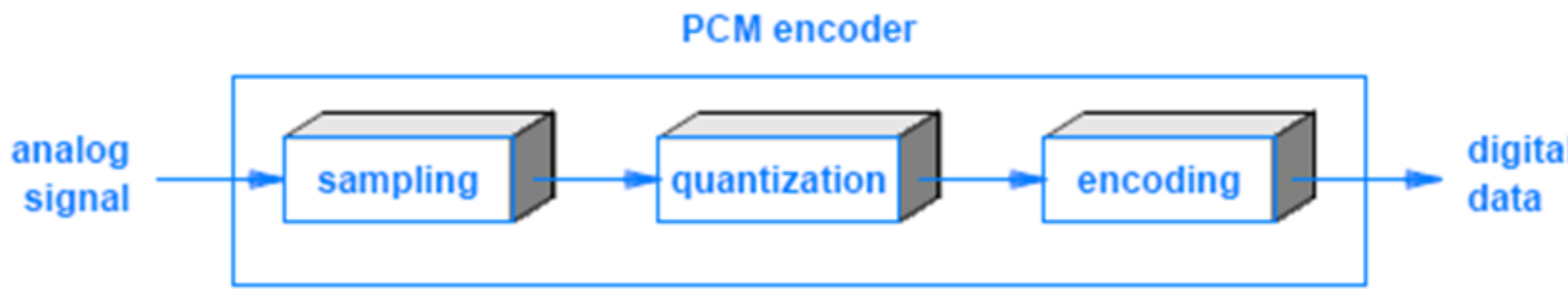
HDB3:
4 zeros are replaced by:
- 000V if the number of pulses (ones) since last substitution was ODD
- B00V if the number of pulses (ones) since last substitution was EVEN

V = 1code violation

Basic Encoding Techniques

- Analog data to digital signal
  - Pulse code modulation (PCM)
  - Delta modulation (DM)
- Basic process of digitizing analog data

Original Data \ Signal Transmitted	Digital	Analog
Digital		ASK FSK/BFSK/MFSK PSK/BPSK/MPSK
Analog	PCM DM	AM PM FM



The Nyquist Theorem and Sampling Rate

- An analog signal must be sampled in PCM or DM
- How frequently should an analog signal be sampled?
  - Taking too few samples (undersampling) means that the digital values only give a crude approximation of the original signal
  - Taking too many samples (oversampling) means that more digital data will be generated, which uses extra bandwidth
- A mathematician named Nyquist discovered the answer to the question of how much sampling is required:

$$\textit{sampling rate} = 2 \times f_{\max}$$

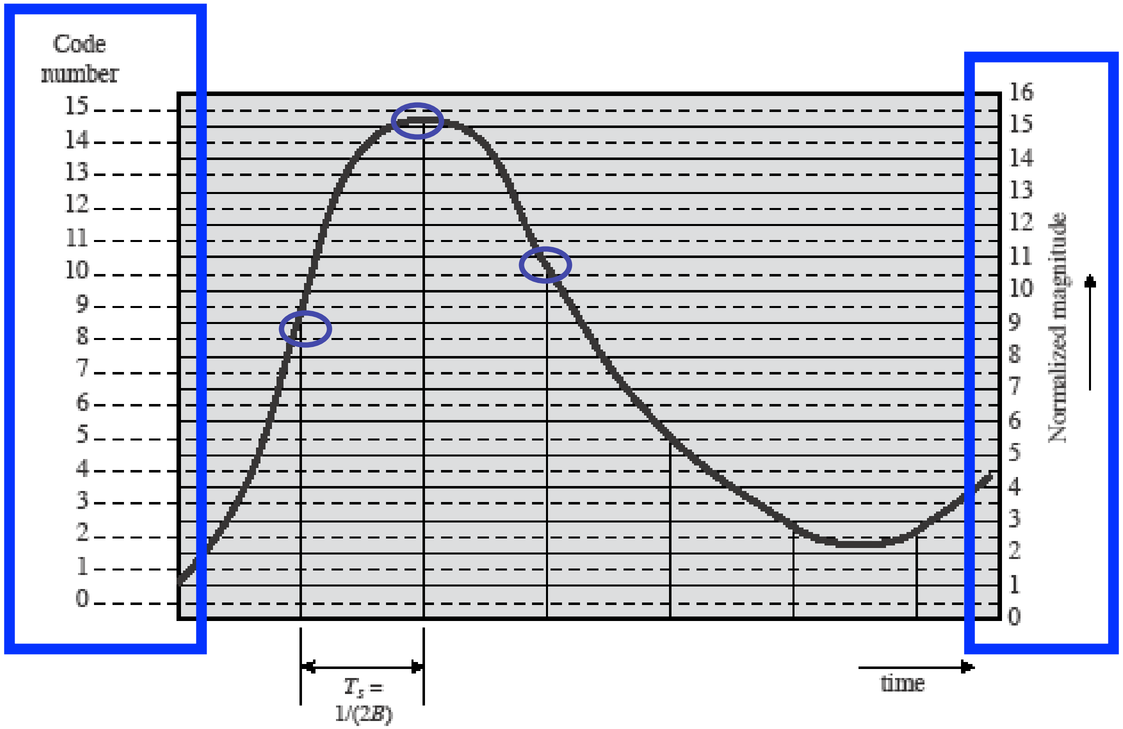
  - where f_{max} is the highest frequency in the composite signal
- Nyquist Theorem provides a practical solution to the problem:
  - sample a signal at least twice as fast as the highest frequency that must be preserved

Pulse Code Modulation

- Based on the sampling theorem
- Each analog sample is assigned a binary code
  - Analog samples are referred to as pulse amplitude modulation (PAM) samples
- The digital signal consists of block of *n* bits, where each *n*-bit number is the amplitude of a PCM pulse

Pulse Code Modulation

1- Sampling frequency (two times fmax)
2- Quantization levels (number of bits available)



PAM value	1.1	9.2	15.2	10.8	5.6	2.8	2.7
quantized code number	1	9	15	10	5	2	2
PCM code	0001	1001	1111	1010	0101	0010	0010

Pulse Code Modulation

- By quantizing the PAM pulse, original signal is only approximated
  - More quantization levels → more accurate signal approximation → more complex system
- Leads to quantizing noise
- Signal-to-noise ratio for quantizing noise
  - n being the number of bits used for quantization

$$\mathrm{SNR}_{\mathrm{dB}} = 20 \log 2^{n} + 1.76\,\mathrm{dB} = 6.02n + 1.76\,\mathrm{dB}$$

NOTE: each additional bit increases SNR by 6 dB, or a factor of 4

Signal to quantization noise ratio (SQNR)

The root mean square value of the sine wave signal

The error signal lies uniformly in the range [+/-1/2^b]; thus the root mean square value of the error signal

$$SQNR_{dB} = 20\log\left(\frac{\frac{1}{\sqrt{2}}}{\frac{1}{\sqrt{3}2^b}}\right)$$

$$= 20\log(2^b) + 20\log\left(\sqrt{\frac{3}{2}}\right)$$

$$= (6.02 \times b + 1.76)dB$$

NOTE: b=n being the number of bits used for quantization

Example:

- Assuming we use 7 bits to reconstruct the voice signal. Bandwidth of voice signal is 4KHz.
  - How may quantization levels can we create?
  - What is the sampling rate for the voice signal? (Nyquist Theorem)
  - What is the BW of the PCM-encoded digital signal? (bps)
  - What is the minimum frequency (Hz) required to carry the voice signal?
  - How much the S/N (in dB) will increase if we use 9 bits instead?

•2^7 = 128 levels
•Sampling rate: 2f = 8KHz (8000 samples / sec) ← according to the sampling theorem
•Each sample has 7 bits
•PCM BW = 8000 sample/sec x 7 bit/sample = 56 Kbit/sec → data rate
•Remember if rate of the signal is 2f then a signal with frequencies no greater than f is sufficient to carry the signal rate. → f=28 KHz. (this is in the absence of noise!!!)
•each additional bit increases SNR by 6 dB, or a factor of 4 → 12 dB.

Delta Modulation

- Analog input is approximated by staircase function
  - Moves up or down by one quantization level (δ) at each sampling interval
- Only the change of information is sent
  - only an increase or decrease of the signal amplitude from the previous sample is sent
  - a no-change condition causes the modulated signal to remain at the same 0 or 1 state of the previous sample

Delta Modulation

- Two important parameters
  - Size of step assigned to each binary digit (δ)
  - Sampling rate
- Accuracy improved by increasing sampling rate
  - However, this increases the data rate
- Advantage of DM over PCM is the simplicity of its implementation