# King Fahd University of Petroleum & Minerals Computer Engineering Dept

**COE 342 – Data and Computer Communications**

**Term 032**

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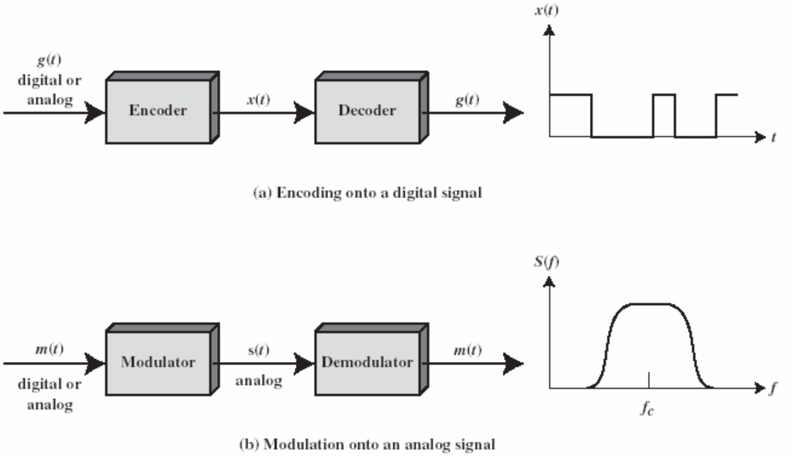
## Lecture Contents

1. Background
2. Digital Data, Digital Signals
3. Digital Data, Analog Signals
4. Analog Data, Digital Signals
5. Analog Data, Analog Signals
6. Spread Spectrum

# Background






# Background - Digital Signaling

- **Data source g(t)**
  - **Analog source – voice**
  - **Digital source – computer data (file)**
- **ENCODED – to match medium characteristics and optimize transmission – Result is x(t)**
- **Note that x(t) is digital (discrete voltage levels)**

## Background – Analog Signaling

- Data source m(t)
  - Analog source – voice
  - Digital source – computer data (file)
- MODULATED:
  - We need a carrier signal: continuous-time constant frequency signal (fc) {i.e. A $\cos(2\pi ft+\phi)$ or A$\sin(2\pi ft+\phi)$}
  - Frequency of carrier is chosen to match transmission characteristic of medium
  - Modulation: Encoding source data onto carrier:
    - Manipulating frequency – phase – Amplitude – or a combination of these elements
  - Process of encoding is chosen to optimize transmission

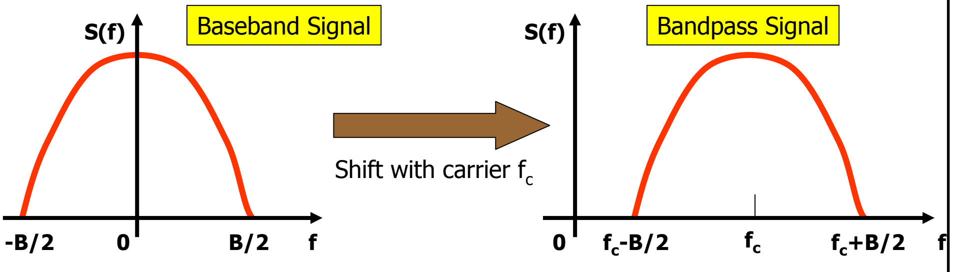


## Background – Analog Signaling (2)

- Note that s(t) is analog (continuous voltage levels)
- Bandwidth of s(t) is usually centered around fc
- s(t) is a *bandlimited* or *bandpass* signal:
  - Finite bandwidth at or around fc

## Background – Baseband vs. Bandpass Signals

- **Baseband Signal:**
  - **Spectrum not centered around non zero frequency**
  - **May have a DC component**
- **Bandpass Signal:**
  - **Does not have a DC component**
  - **Finite bandwidth around or at $f_c$**



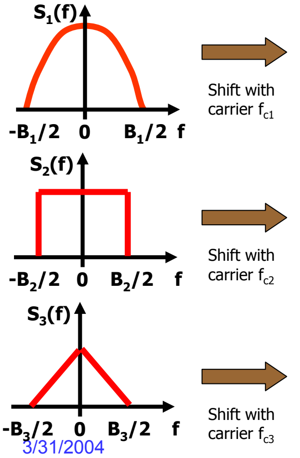



## Background

- **Digital Data, Digital Signaling:**
  - **Less complex/expensive than digital-to-analog modulation equipment**
- **Analog Data, Digital Signaling:**
  - **Conversion of analog data to digital allows the use of modern digital tx and switching technology**
- **Digital Data, Analog Signaling:**
  - **Some transmission media can *ONLY* propagate analog signals – such as fiber optics and unguided**
- **Analog Data, Analog Signaling:**
  - **Analog data can be transmitted as baseband signals cheaply**
  - **Shifting bandwidth of baseband signals to occupy another portion of spectrum – different signals share same medium using frequency division multiplexing**

# Frequency Division Multiplexing

- **Will be visited again in Chapter 8**

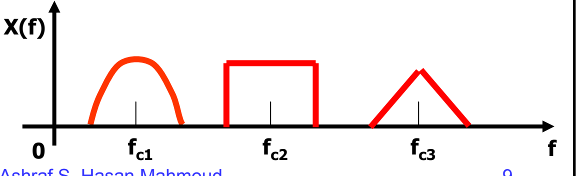



$x(t) = s_1(t) \text{ X } \cos(2\pi f_{c1}t) + s_2(t) \text{ X } \cos(2\pi f_{c2}t) + s_3(t) \text{ X } \cos(2\pi f_{c3}t)$

- x(t) is transmitted on the media

-The three spectra are not overlapping if $f_{c1}$, $f_{c2}$, and $f_{c3}$ are chosen appropriately

-Original composite signals $s_1(t)$, s2(t), and s3(t) can be recovered using bandpass filters with appropriate bandwidths centered at $f_{c1}$, $f_{c2}$, and $f_{c3}$, respectively.



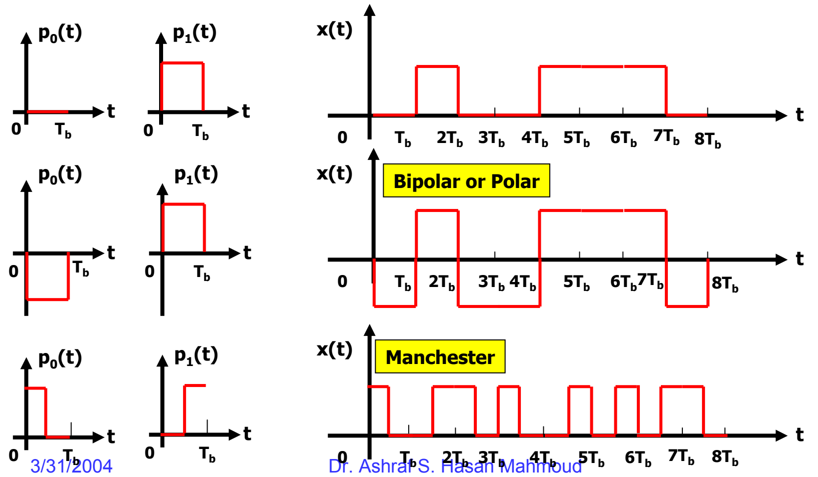



# Signal Elements or Pulses

- **Unit of transmission – repeated to form the overall signal**
- ***Shape* of pulse determines the bandwidth of the transmitted signal**
- **Digital data is mapped or encoded to the different pulses or units of transmission**

# Signal Elements or Pulses

**Definitions of Pulses** **Encoded Signal: 0 1 0 0 1 1 1 0**

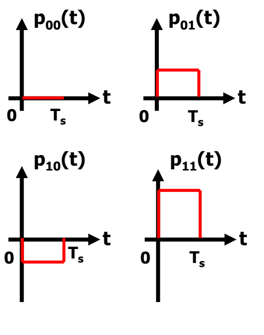


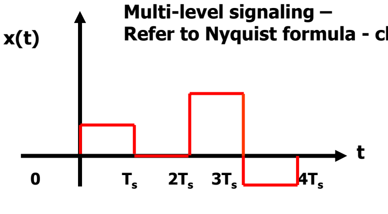

Examples of Digital Signaling



# Signal Elements or Pulses

**Pluses Definitions** **Encoded Signal: 0 1 0 0 1 1 1 0**



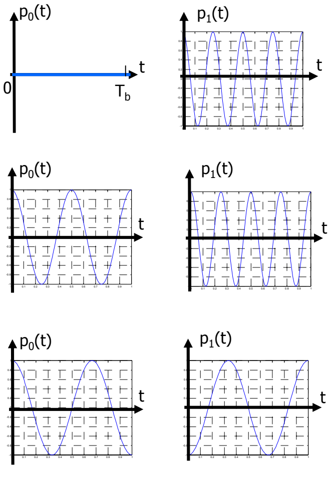




- Note that each symbol or pulse caries 2 bits
- Symbol duration is $T_s = 2T_b$
- Bit rate R equal to $1/T_b$
- Symbol rate or *baud rate* $R_s$ equal to $1/T_s$ ➔ $R = 2R_s$
- In general to encode n bits per pulse, you need $2^n$ pulses

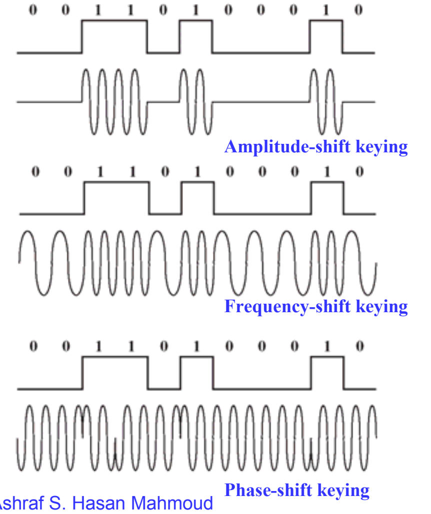
Example of Digital Signaling

# Signal Elements or Pulses

**Definitions of Pulses**




**Encoded Signal:**




**Example of Analog Signaling**



# Digital Data – Digital Signals

- **Digital signal: sequence of discrete, discontinuous voltage pulses**
- **Digital data (bits) are encoded (or mapped) into signal elements**
- **Baud-rate: number of signal elements per second**
- **Mark – Space = 1 – 0**
- **Communication Tasks – Receiver must have:**
  - **Transmission elements timings**
  - **Pulse voltage level (to know whether it is 0 or 1 for example) – Rxer samples at bit times to find voltage level**

# Key Data Transmission Terms

| Term | Units | Definition |
|---|---|---|
| Data element | Bits | A single binary one or zero |
| Data rate | Bits per second (bps) | The rate at which data elements are transmitted |
| Signal element | Digital: a voltage pulse of constant amplitude.<br>Analog: a pulse of constant frequency, phase, and amplitude. | That part of a signal that occupies the shortest interval of a signaling code |
| Signaling rate or modulation rate | Signal elements per second (baud) | The rate at which signal elements are transmitted |



# How to Overcome Impairments?

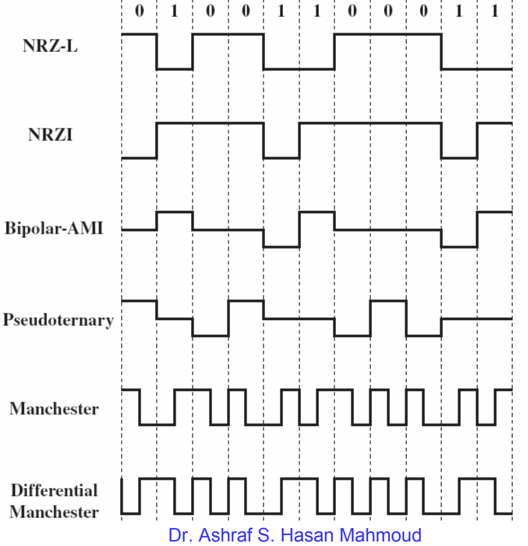
- Faults in detection of received signal register as BIT ERROR RATE at receiver – BER
  - A good communication channel has small or zero BER
- Factors:
  - SNR or Eb/No
  - Data bit rate
  - Channel/system bandwidth
  - Encoding of data bits into signal elements
- Encoding scheme also affects bandwidth of signal

## Digital Signal Encoding Formats

- **Nonreturn to Zero-Level (NRZ-L)**
  - **0 = high level**
  - **1 = low level**
- **Nonreturn to Zero Inverted (NRZI)**
  - **0 = no transition at beginning of interval**
  - **1 = transition at beginning of interval**
- **Bipolar-AMI**
  - **0 = no line signal**
  - **1 = +ve or –ve level; alternating successive ones**
- **Pseudoternary**
  - **0 = +ve or –ve level; alternating for successive ones**
  - **1 = no line signal**
- **Manchester**
  - **0 = transition from high to low in middle of interval**
  - **1 = transition from low to high in middle of interval**
- **Differential Manchester: Always transition in middle of interval**
  - **0 = transition at beginning of interval**
  - **1 = no transition at beginning of interval**
- **Bipolar with 8 Zeros Substitution (B8ZS): same as bipolar AMI, except that any string of 8 zeros is replaced by a string with two code violations**
- **High Density bipolar-3 Zeros (HDB3): same as bipolar AMI, except that any string of 8 zeros is replaced by a string with one code violation**



## Digital Signal Encoding Formats



## How to Evaluate Encoding Schemes?

- **Signal spectrum:** (remember this is dependent on pulse shape)
  - Lack of high frequency component ➔ lesser BW – signal does not required large BW - desirable
  - Lack of DC component - desirable
- **Clocking (Synchronization):**
  - Rxer needs to determine bit timing
    - Provide clock signal at receiver – EXPENSIVE
    - Derive clock signal from incoming signal
  - E.g. Differentiating a Manchester encoded signal results in the clock signal!



## How to Evaluate Encoding Schemes? (2)

- **Error detection:**
  - Capability built into physical layer encoding – e.g. for pseudoternary successive ones have opposite signs
  - More sophisticated error detection and correction codes are used (Chapter 7)
- **Signal interference and noise immunity**
  - Certain codes are superior than others in the presence of noise and interference (i.e. give lower BER for same SNR or Eb/No)
- **Cost and complexity:**
  - Not a major factor compared to the rest of factors
  - In general, the higher the data rate the more expensive the hardware is

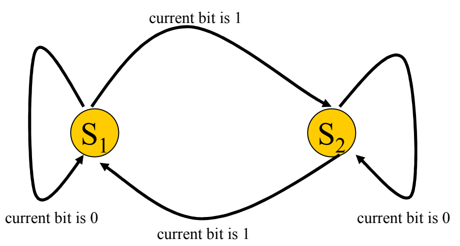
## Nonreturn to Zero (NRZ)

- **Nonreturn to Zero – Level (NRZ-L):**
  - **Binary 0 – constant +ve level**
  - **Binary 1 – constant -ve level**
- **Nonreturn to Zero – Invert on Ones (NRZI):**
  - **Binary 0 – no transition at beginning of bit interval**
  - **Binary 1 – transition at beginning of bit interval**
  - **NRZI is an example of *differential encoding*:**
    - **If bit is equal to 1, bit encoding is opposite to previous bit**
  - **Benefits of differential encoding**



## Nonreturn to Zero (NRZ) (2)

- **Differential encoding – Involved memory (similar to sequential circuit design)**
- **Best represented using a state machine**

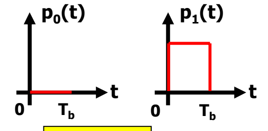



$S_1$: output constant –ve level for $T_b$
$S_2$: output constant +ve level for $T_b$

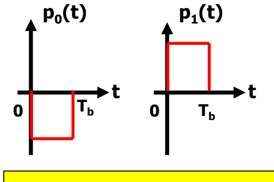
Nonreturn to Zero – Invert on Ones

- **NO MEMORY**

**For RZ or NRZ-L**




Unipolar




Nonreturn to Zero - Level

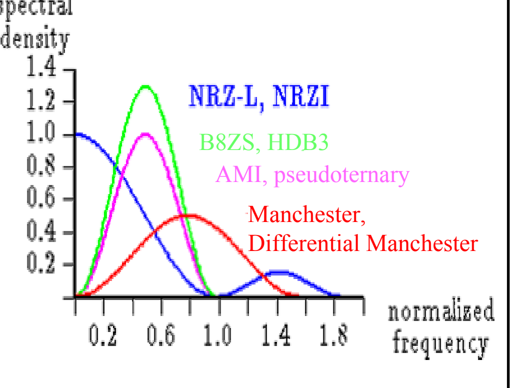
## Spectrum Characteristics of NRZ

- **Most of the energy in NRZ and NRZI signals is between DC and half of bit rate**
  - **For example: When R = 9600 b/s or Tb = 0.104 msec, most of energy of the signal is between 0 Hz and 4800 Hz**
- **Main limitations of NRZ:**
  1. **presence of DC component**
  2. **lack of synchronization capability**
  - **Consider the case of a long string of ones or zeros:**
    - **One constant voltage level for long duration ( >> $T_b$) – may cause drift in clock synchronization**
- **Applications:**
  - **Digital magnetic recording**
  - **Generally not used for signal transmission**



## Spectrum Characteristics of NRZ and Other Encoding Schemes

- **Note the x-axis: normalized frequency (f/R)**
  - **E.g. value equal to 1.2, means f = 1.2 R**
- **Schemes NRZ-L and NRZI have DC component**
- **Schemes B8ZS, HDB3, AMI, pseudoternary, Manchester and differential Manchester have no DC component**
- **NRZ-L, NRZI, B8ZS, HDB3, AMI, and pseudoternary have negligible energy beyond f = R**
- **B8ZS, HDB3, AMI, and pseudoternary have their energy concentrated around f = R/2**
- **Manchester and differential Manchester has significant energy concentration beyond f = R (because of the per bit transitions!)**



## Multilevel Binary – Bipolar - AMI

- Family of codes that uses more than two signal levels
- Bipolar-AMI:
  - Binary 0 – no signal level
  - Binary 1 – +ve or -ve level; alternating
- Advantages of Bipolar-AMI:
  - Synch: long string of 1s is not a problem – but a long string of 0s is
  - No net DC component
  - Smaller BW compared to NRZ
  - Alternating pulses – simple error detection (no two consecutive ones can have same polarity)



## Multilevel Binary – Pseudoternary

- Pseudoternary:
  - Binary 0 – +ve or -ve level; alternating
  - Binary 1 – no signal level
- Same advantages as bipolar-AMI
- To provide clock synch info:
  - Insert additional bits to force transition – used in ISDN for low bit rate connections – results in increased bit rate
  - Can not be used for already high bit rate connections – expensive
  - Use SCRAMBLING

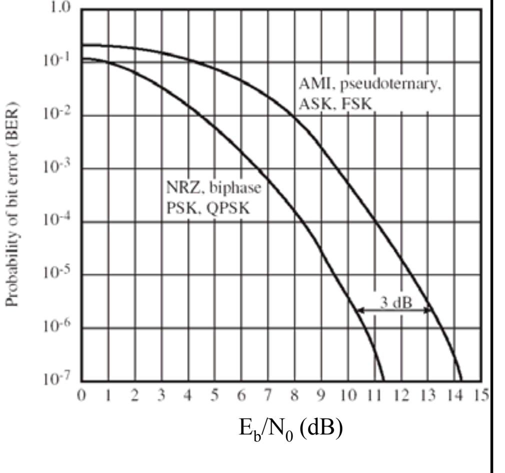
## NRZ V.S. Multilevel Binary

- **Spectrum:**
  - **NRZ has DC component**
  - **Multilevel binary does not have DC component – smaller bandwidth**
- **Synch:**
  - **NRZ: long strings of 1s AND 0s present a problem**
  - **Multilevel binary: long strings of 1s for bipolar-AMI or long strings of 0s for pseudoternary present a problem**



## NRZ V.S. Multilevel Binary (2)

- **Efficiency:**
  - **NRZ: two symbols – one for 0 and the other for 1 – i.e. $\log_2 2 = 1$ information bit per symbol**
  - **Multilevel binary: three symbols – one for 0 and two for 1 (or the reverse for pseudoternary) – i.e. $\log_2 3 = 1.58$ information bits per symbol**
  - **NRZ is more efficient – requires 3 dB less (1/2) signal power to give same BER as Multilevel**



## Biphase Encoding

- Manchester: transition at the middle of each bit
  - Binary 0 – high to low transition in the middle
  - Binary 1 – low to high transition in the middle
- Differential Manchester: transition at the middle of each bit
  - Binary 0 – transition at beginning of interval
  - Binary 1 – no transition at beginning of interval
- THERE IS ALWAYS a TRANSION at midbit – This provides the needed clock signal
- Biphase schemes require at least on transition per bit interval and sometimes two transitions per bit interval ➔ Generate signal with higher frequency components compared to NRZ for same rate!!



## Advantages of Biphase Encoding

- Synchronization:
  - There is a predictable transition during each bit time
  - To derive clock signal – differentiate biphase signal
  - Biphase = Self clocking codes
- No DC component
- Error Detection:
  - A transition must happen at mid bit – if not present ➔ ERROR
- Applications:
  - Manchester coding: IEEE 802.3 coaxial cable and TP CSMA/CD bus LANs
  - Differential Manchester: IEEE 802.5 token ring LANs on STP

## Modulation Rate

- Modulation (Baud) Rate - D: number of symbols or signal elements transmitted per second
- Data (or bit) Rate - R: number of bits transmitted per second
- D = R/b – where b is number of bits per symbol

Refer to slide number 12



## Transitions Per Bit Time

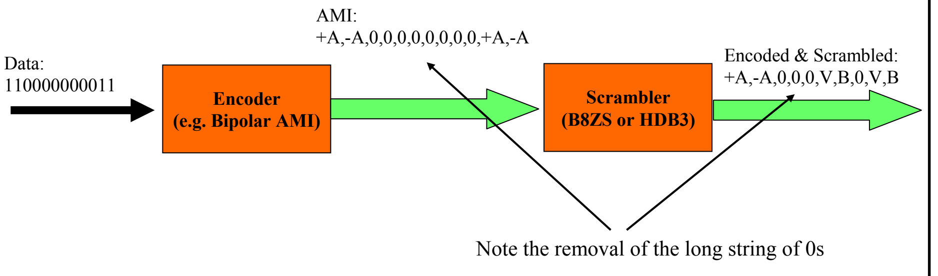
- The more transitions per bit time, the greater is the required bandwidth of the encoding scheme

| Encoding | Minimum | 10101010… | Maximum |
|---|---|---|---|
| NRZ-L | 0 (all 0s or 1s) | 1.0 | 1.0 |
| NRZI | 0 (all 0s) | 0.5 | 1.0 (all 1s) |
| Bipolar-AMI | 0 (all 0s) | 1.0 | 1.0 |
| Pseudoternary | 0 (all 1s) | 1.0 | 1.0 |
| Manchester | 1.0 (10101…) | 1.0 | 2.0 (all 0s or 1s) |
| Differential Manchester | 1.0 (all 1s) | 1.5 | 2.0 (all 0s) |

Note that Manchester and differential Manchester encoding have the maximum number of transitions per bit time – This is the reason, their spectrum have significant components for f/R greater than 1.0 (refer to slide 24)

# Scrambling Techniques

- **Want to achieve:**
  - **No DC component → media**
  - **No long sequence of zero-level signals → clocking/Synch**
  - **No reduction in data rate → capacity**
  - **Error-detection capability → reliability**






# Bipolar with 8-Zeros Substitution (B8ZS)

- **Substitution Rules:**
  - **If an octet of all zeros occurs and the last voltage pulse preceding the this octet was +ve, then the 8 zeros of the octet are encoded as 000+-0-+**
  - **If an octet of all zeros occurs and the last voltage pulse preceding the this octet was -ve, then the 8 zeros of the octet are encoded as 000-+0+-**
- **Cause two code violations (signal patterns that are not allowed in AMI)**
- **Unlikely to be caused by noise**
- **Recognized by receiver and interpreted as 8 zeros**

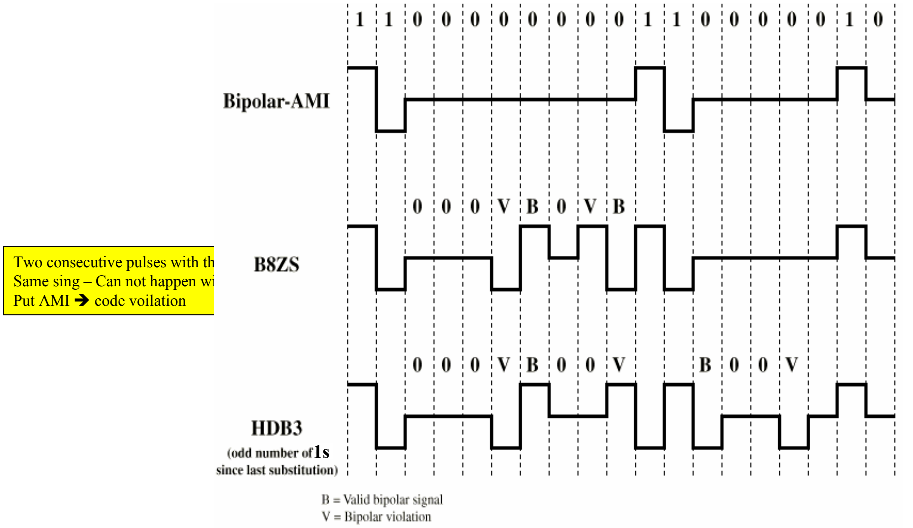
# High-Density Bipolar-3 Zeros (HDB3)

- Substitution Rules:
- Cause two code violations (signal patterns that are not allowed in AMI)
- Unlikely to be caused by noise
- Recognized by receiver and interpreted as 8 zeros

| Polarity of Preceding Pulse | Number of Bipolar Pulses (Ones) since Last Substitution | |
|---|---|---|
| | Odd | Even |
| - | 000- | +00+ |
| + | 000+ | -00- |



# B8ZS and HDB3



## Digital Data – Analog Signals

- Digital data (bits) transmitted using analog signals:
  - E.g. computer-modem-PSTN
- Subscriber–to–PSTN connection designed to carry analog (voice) signal from 300 Hz to 3400 Hz
- 56K Modem – encodes data and generates a signal occupying the same range for voice signals ➔ one line - one signal
- DSL Modem – encodes data and generates signal occupying higher range than that usually occupied by voice ➔ one line – two signals

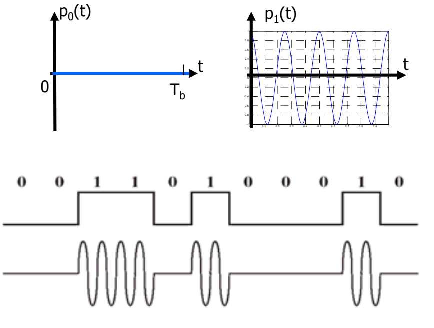


## Encoding Techniques – Amplitude Shift Keying (ASK)

- Analog pulses (signal elements) used are:

$$s(t) = \begin{cases} A\cos(2\pi f_c t) & bit = 1 \\ 0 & bit = 0 \end{cases}$$

- Spectrum of overall signal is centered around $f_c$
- Application: on voice-grade lines used up to 1200 bps

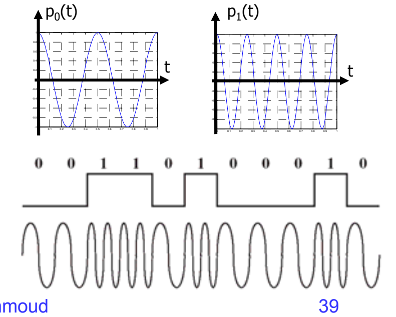


## Encoding Techniques – Frequency Shift Keying (FSK)

- **Analog pulses (signal elements) used are:**

$$s(t) = \begin{cases} A\cos(2\pi f_1 t) & bit = 1 \\ A\cos(2\pi f_2 t) & bit = 0 \end{cases}$$

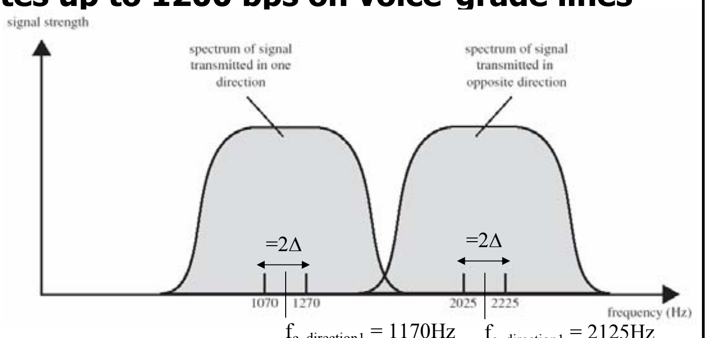
- **Spectrum of overall signal is centered around $f_1$ and $f_2$**






## Encoding Techniques – Frequency Shift Keying (FSK) (2)

- **Application: full duplex**
  - **Direction 1: f1 = 1070 Hz, f2 = 1270 Hz**
  - **Direction 2: f1 = 2025 Hz, f2 = 2225 Hz**
- **Less susceptible to errors (compared to ASK) – used for rates up to 1200 bps on voice-grade lines**
- **Also used for high frequency (3 to 30 MHz) radio transmission**
- **LANs – coaxial cables**



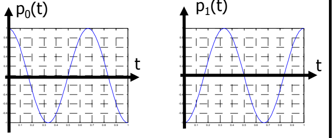
# Encoding Techniques – Phase Shift Keying (PSK)

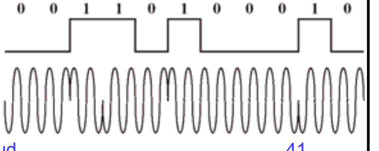
- **Analog pulses (signal elements) used are:**

$$s(t)=\begin{cases} A\cos(2\pi f_c t+\pi) & bit=1 \\ A\cos(2\pi f_c t) & bit=0 \end{cases}$$

- **Spectrum of overall signal is centered around $f_c$**

- **Example of 2-phase system**



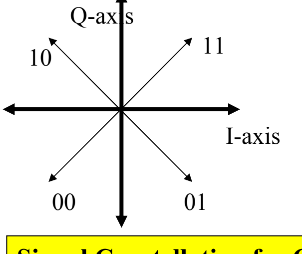



# Encoding Techniques – Quadrature Phase Shift Keying (QPSK)

- **Analog pulses (signal elements) used are:**

$$s(t)=\begin{cases} A\cos(2\pi f_c t+\pi/4) & bits=11 \\ A\cos(2\pi f_c t+3\pi/4) & bits=10 \\ A\cos(2\pi f_c t+5\pi/4) & bits=00 \\ A\cos(2\pi f_c t+7\pi/4) & bits=01 \end{cases}$$

- **Example of 4-phase system**
- **Each signal element carries 2 bits**
- **One can extend this scheme to obtain: 8PSK for example**
- **One can use ASK together with PSK to get more signal elements – e.g. 9600 kb/s modem uses 12 phase angles four of which have two amplitude values**

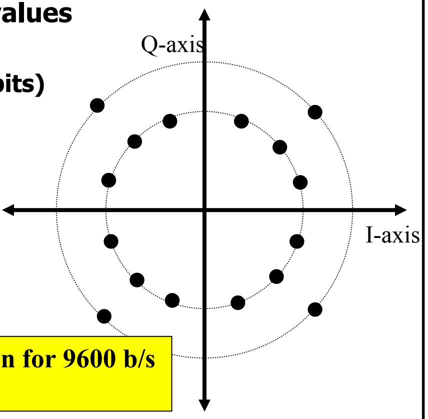



**Signal Constellation for QPSK**

# Encoding Techniques – Quadrature Phase Shift Keying (QPSK) (2)

- One can extend this scheme to obtain: 8PSK for example
- One can use ASK together with PSK to get more signal elements – e.g. 9600 kb/s modem uses 12 phase angles four of which have higher amplitude values
  - For this example: b = 4
  (i.e. every signal element carries 4 bits)

- In general:
  $D = R/b = \log_2 L$
  where D: modulation rate or baud rate
  R: data rate, bps
  L: # of signal levels
  b: # of bits per signal element




Signal Constellation for 9600 b/s modem standard



# Performance - Bandwidth

- Signal (ASK, PSK, FSK, etc) BW depend on:
  - Definition of BW
  - Filtering technique
- r – depends on filtering technique (0<r<1)
- For FSK: $\Delta f = f_2 - f_c = f_c - f_1$

| Encoding Scheme | BW (Signal Spectrum) |
|---|---|
| ASK | $B_T=(1+r)R$ |
| PSK | $B_T=(1+r)R$ |
| FSK | $B_T=2\Delta f + (1+r)R$ |

- For multi-level PSK
  $B_T=(1+r)D = (1+r)R/b = (1+r)/\log_2 L \text{ X } R$
- $R/B_T$ = data rate to transmission bandwidth → ***Bandwidth Efficiency***
  - The higher this number the more efficient the scheme is (i.e. less number of Hzs is required to transmit the bits)

## Performance – Bit Error Rate (BER)

This formula is valid for D/R = b = 1 (1 symbol = 1 bit)

- In presence of noise and in terms of BER: PSK and QPSK are 3 dB better than ASK And FSK
- Recall that Eb/No is equal to

$$\frac{E_b}{N_0} = \frac{S/R}{N/B_T} = \frac{S}{N} \times \frac{B_T}{R} = SNR \times \frac{1}{\text{BW efficiency}}$$

- Hence, one can decrease BER (i.e. increase $E_b/N_0$) by either increasing SNR, increasing the transmission bandwidth ($B_T$), or reducing the data rate (R)
- For multi-level signaling – Replace R with D



## Performance – Example

- What is the bandwidth efficiency for FSK, ASK, PSK, and QPSK for a BER of $10^{-7}$ on a channel with SNR = 12 dB
- *Solution*

Bandwidth efficiency = $R/B_T$

$E_b/N_0$ = SNR / $(R/B_T)$ or $(Eb/N0)_{dB} = SNR_{dB} - (R/B_T)_{dB}$

Therefore, $(R/B_T)_{dB} = SNR_{dB} - (Eb/N0)_{dB} = 12 - (Eb/N0)_{dB}$

Using the BER curves (slide 28):

<u>(for ASK & FSK)</u> BER = $10^{-7}$ → $(E_b/N_0)_{dB}$ = 14.2 dB

Hence, $(R/B_T)_{dB}$ = 12 – 14.2 = -2.2 dB, or $R/B_T$ = 0.6

<u>(for PSK)</u> BER = $10^{-7}$ → $(E_b/N_0)_{dB}$ = 11.2 dB

Hence, $(R/B_T)_{dB}$ = 12 – 11.2 = 0.8 dB, or $R/B_T$ = 1.2

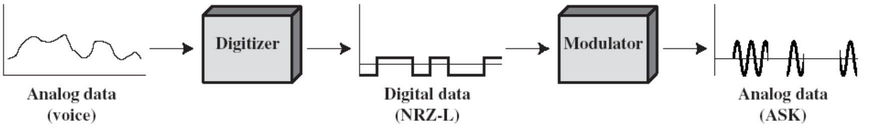
<u>(for QPSK)</u> same curve as PSK → $(E_b/N_0)_{dB}$ = 11.2 dB

Hence, $(D/B_T)_{dB}$ = 12 – 11.2 = 0.8 dB, or $D/B_T$ = 1.2 and $R/B_T$ = 2.4 (since D = R/2 for QPSK)

BW Efficiency: 0.6 → 1.2 → 2.4

## Analog Data – Digital Signal

- Analog Data is "Digitized" i.e converted to digital
- Once in digital form:
  - Use Digital signaling (NRZ-L, etc)
  - Use Analog Signaling (ASK, FSK, etc) – Shown in figure below
- CODEC: Device for converting analog data to digital for transmission – and for recovering original analog data






## CODEC Procedures

- Two main procedures are used in CODECs:
  1. Pulse Code Modulation (PCM)
  2. Delta Modulation

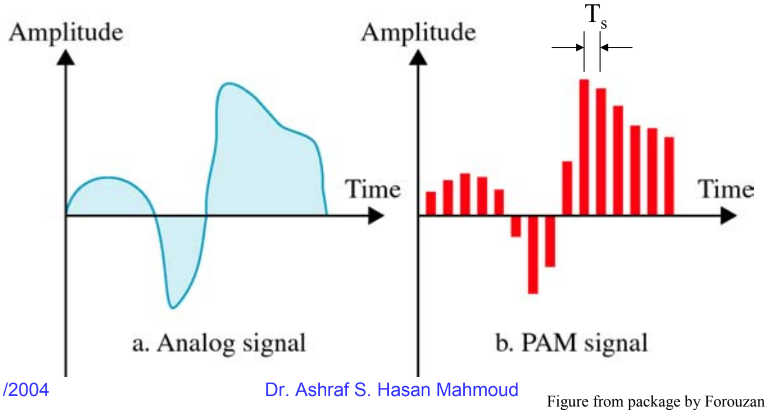
# Pulse Code Modulation (PCM)

- A scheme for digitizing ANALOG data
- For flash animation of PCM procedure click HERE
- Procedure:
  - SAMPLING: Analog signal is sampled (The rate of sampling SHOULD BE greater than twice the highest frequency – refer to the sampling theorem) ➔ Result: Analog Samples
  - QUANTIZATION: Analog samples are mapped to discrete levels and each level is given a binary code ➔Result: binary word for each sample
    - Example: if we decide to use 256 discrete levels, then every level will have 8-bit word – correspondingly, every analog sample will be translated into 8 bits



# Pulse Amplitude Modulation (PAM)

- Sampling Frequency, $f_s = 2Xf_m$
- Sampling Time, $Ts = 1/f_s = 1/(2Xf_m)$




Pulse Amplitude Modulation (PAM)

Figure from package by Forouzan

# Quantization

- Analog samples are ROUNDED to DISCRETE levels (finite number of levels)
- For N bits per word ➔ we have $2^N$ levels

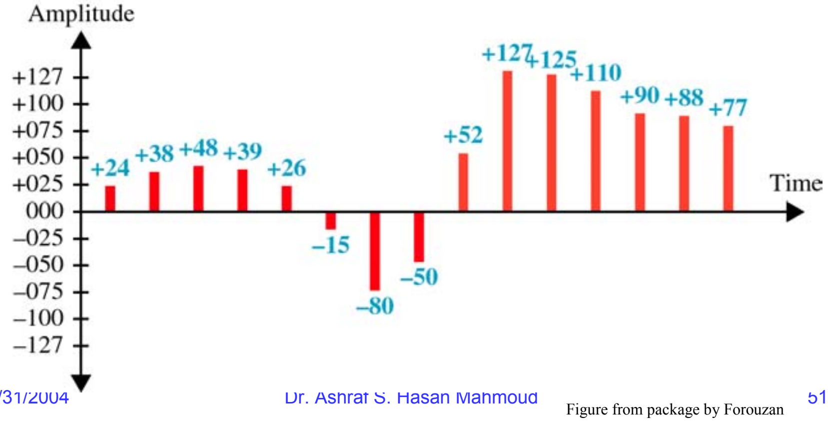





Figure from package by Forouzan

# Quantization – PCM

- N-bit word is then generated for every sample

| | | | | | |
|---|---|---|---|---|---|
| +024 | 00011000 | –015 | 10001111 | +125 | 01111101 |
| +038 | 00100110 | –080 | 11010000 | +110 | 01101110 |
| +048 | 00110000 | –050 | 10110010 | +090 | 01011010 |
| +039 | 00100111 | +052 | 00110110 | +088 | 01011000 |
| +026 | 00011010 | +127 | 01111111 | +077 | 01001101 |

Sign bit
+ is 0 – is 1

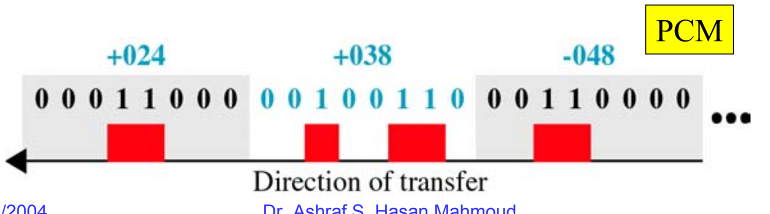





Figure from package by Forouzan

## PCM – overall picture

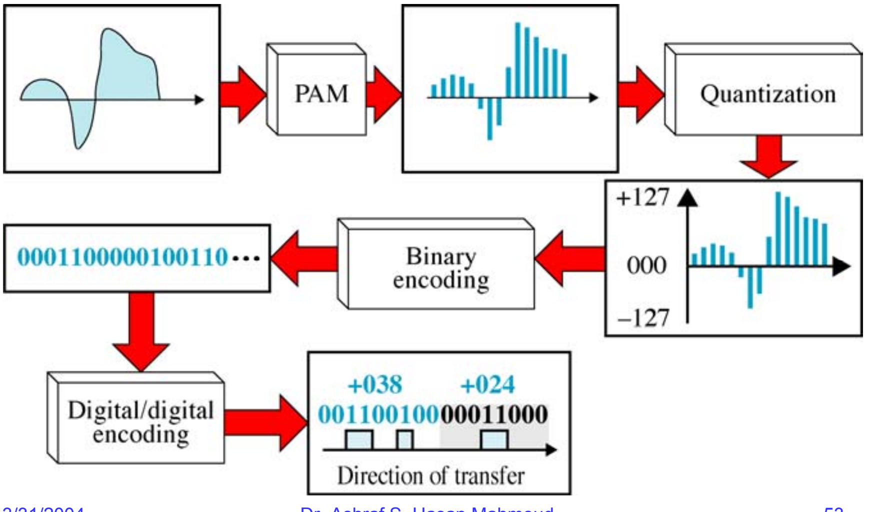



Figure from package by Forouzan



## Pulse Code Modulation - SNR

- SNR is given by

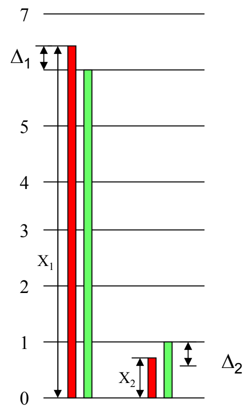
$$SNR = 6.02n + 1.76 \text{ dB}$$

Where n is the number of bits per word/sample

- Assumes uniform distribution of signal level
- Errors: difference between quantized samples and original analog samples → QUANTIZATION NOISE
- Thermal noise is NOT accounted for
- NOTE:
  - As number of bits is increased (less rounding errors), SNR increases by 6 dB every extra bit

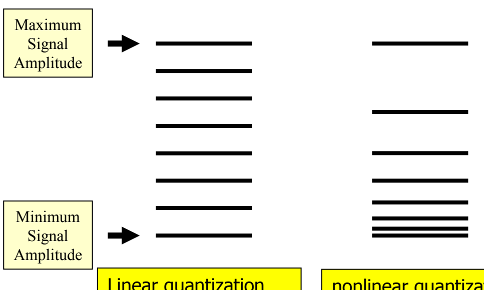
# Pulse Code Modulation – Linear vs. Nonlinear Encoding

- Linear quantization: equally spaced levels ➔ magnitude of quantization error is same for large amplitude samples and small amplitude samples >> low signal levels are more affected by quantization errors
- Solution: to increase "resolution" in the low signal level region ➔
  - increase total number of levels OR
  - use companding function before quantization
- For shown figure:
  - Relative error for $X_2$ is much greater than that for $X_1$
  - Relative error is equal to $\Delta$ (quantization error) divided by original signal level



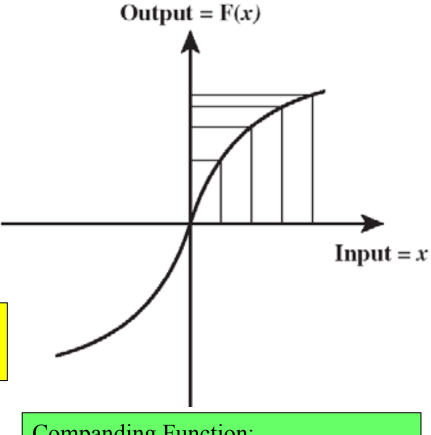



# Pulse Code Modulation – Linear vs. Nonlinear Encoding (2)




Linear quantization (signal samples occur in all regions equally)

nonlinear quantization (signal samples occur mostly in low levels)




Companding Function: Amplifies details of F(x) for low x and attenuates details of F(x) for high x

## Example: Problem 5-20

5-19: Consider an audio signal with spectral components in the range of 300 to 3000 Hz. Assuming a sampling rate of 7000 samples per second will be used to generate the PCM signal.

a) For SNR = 30 dB, what is the number of uniform quantization levels needed?

b) What data rate is required?



## Example: Problem 5-20 - Solution

a) $(SNR)_{dB} = 6.02\ n + 1.76 = 30$ dB

$n = (30 - 1.76)/6.02 = 4.69$

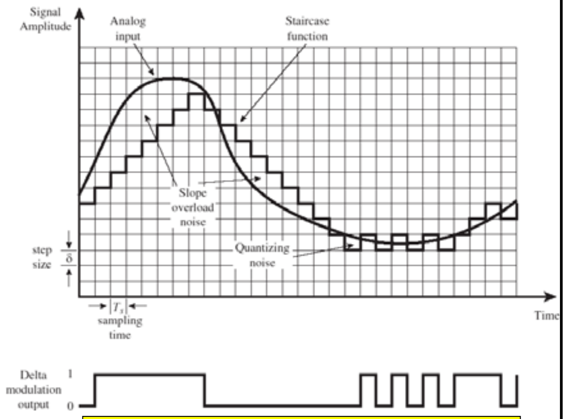
Rounded off, n = 5 bits

This yields $2^5 = 32$ quantization levels

b) R = 7000 samples/s X 5 bits/sample = 35 Kbps

# Sigma-Delta Modulation

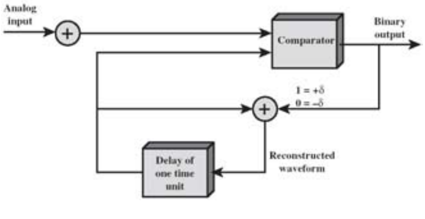
- Approximates the signal by a staircase function that moves up or down one quantization level ($\delta$) every step (sampling time $T_s$)
- Transition up or down occurs at sampling instant
- <u>Slope overload</u>: function increasing/decreasing at a rate faster than $\delta/T_s$ – staircase function can not catch up with original signal
- We still have <u>quantization noise</u> – rounding of original signal level



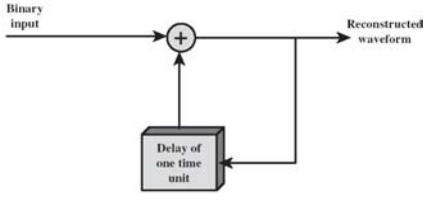

**<u>At the sample time</u>: The output of the modulator is just ONE bit (up or down) - compare this with PCM (n bits per sample)**



# Sigma-Delta Modulation( 2)




Transmission

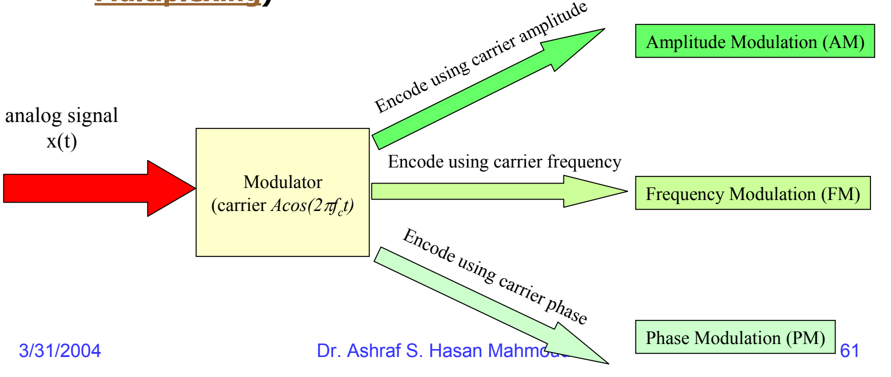



Reception

- Ways to improve over Delta Modulation:
  - Adaptive step delta modulation

# Analog Data – Analog Signals

- **Two principle reasons for analog modulation of analog signals:**
  - **High frequency may be more effective for transmission**
  - **Use of FDM (refer to slide 9: Frequency Division Multiplexing)**






# Amplitude Modulation (AM)

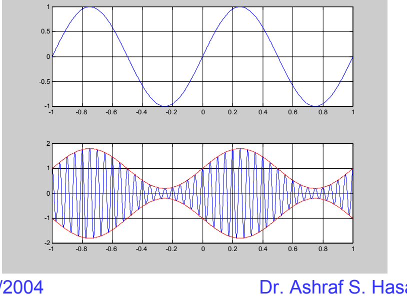
- **Simplest form of modulation:**

$$s(t) = [1 + n_a\, x(t)] \cos(2\pi f_c t)$$

**where $\cos(2\pi f_c t)$ is the carrier**

**$x(t)$ is the input signal (carrying data)**

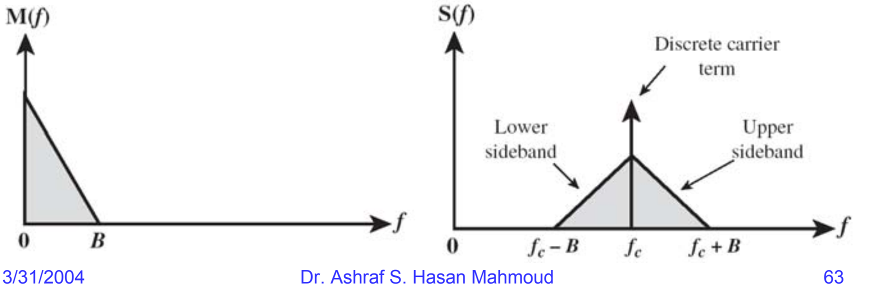
**$n_a$ is modulation index (control parameter)**



Information is "the envelope" of the overall signal s(t)
To preserve the envelope $n_a$ should be < 1; for $n_a$ > 1,
The envelope crosses the x-axis (info is lost)

## Bandwidth of AM signal

- S(t) has a double sided spectrum function centered around fc – in addition to the carrier itself → Double sideband transmitted carrier (DSBTC)
- If B is the bandwidth of x(t), the required transmission bandwidth for the AM signal is $B_T$ = 2B






## Power of AM signal

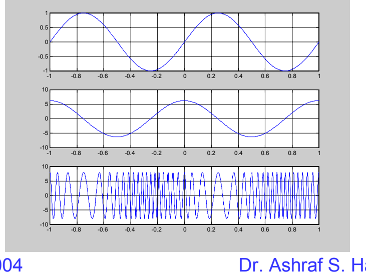
- Total signal power: $P_t = P_c(1+n_a^2/2)$, where $P_c$ is the transmitted power in carrier
- s(t) contains extra info: the carrier itself – removal of carrier (i.e s(t) = $m(t)\cos(2\pi f_c t)$) is referred to as double sideband suppressed carrier (DSBSC)
- DSBSC signal has same BW as DSBTC
- Carrier info is useful in helping receiver lock to exact frequency and phase of carrier

# Frequency Modulation (FM)

- Simplest form of modulation:

$$s(t) = A_c \cos(2\pi f_c t + \phi(t))$$
$$\phi'(t) = n_f\, m(t)$$

where $A_c/f_c$ are the amplitude/frequency of carrier
$m(t)$ is the input signal (carrying data)
$n_f$ is frequency modulation index (control parameter)



Instantaneous frequency of s(t), $f_i(t)$, is equal to $f_c + n_f m(t)/(2\pi)$



# Bandwidth/Power of FM signal

- If B is the bandwidth of m(t), the required transmission bandwidth for the FM signal is

$$B_T = 2(1+\beta)B$$
$$\beta = \Delta F/B = n_f A_m/(2\pi B)$$

- $\Delta F$ is the peak deviation around fc, $A_m$ is the maximum amplitude of m(t). Note $\Delta F = n_f A_m/(2\pi)$
- Note $B_T$ for PM signal is greater than that of AM signal
- Power of FM signal: $A_c^2/2$

## Phase Modulation (PM)

- **Simplest form of modulation:**

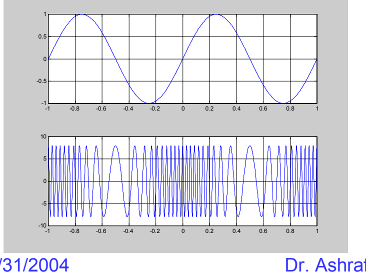
$$s(t) = A_c \cos(2\pi f_c t + \phi(t))$$

$$\phi(t) = n_p\, m(t)$$

**where $A_c/f_c$ are the amplitude/frequency of carrier**

**$m(t)$ is the input signal (carrying data)**

**$n_p$ is phase modulation index (control parameter)**




Instantaneous phase of $s(t)$, $\phi(t)$, is equal to $n_p m(t)$



## Bandwidth/Power of PM signal

- **If B is the bandwidth of m(t), the required transmission bandwidth for the FM signal is**

$$B_T = 2(1+\beta)B$$

$$\beta = n_p A_m$$

- **Note $B_T$ for PM signal is greater than that of AM signal**
- **Power of PM signal: $A_c^2/2$**

# Example: Problem 5-24

- Consider the angle modulation signal

$$s(t) = 10 \cos(10^8\pi t + 5\sin(2\pi(10^3)t)]$$

Find the maximum phase deviation and the maximum frequency deviation

**Solution:**

$s(t) = A_c \cos[2\pi f_c t + \phi(t)] = 10 \cos [(10^8)\pi t + 5 \sin(2\pi(10^3)t)]$

Therefore,

$$\phi(t) = 5 \sin 2\pi(10^3)t,$$

and the maximum phase deviation is 5 radians.

For frequency deviation, recognize that the change in frequency is determined by the derivative of the phase:

$$\phi'(t) = 5\ (2\pi)\ (10^3) \cos [2\pi(10^3)t]$$

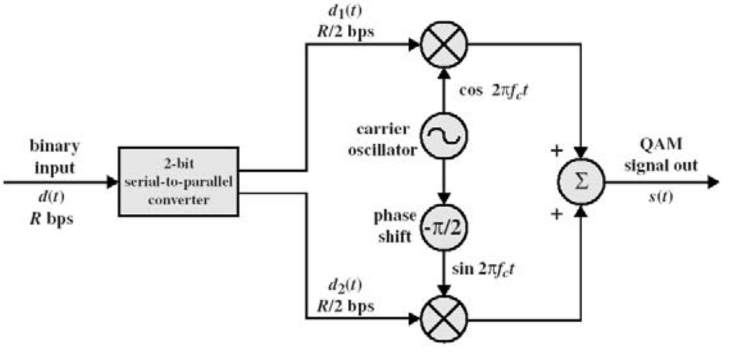
which yields a frequency deviation of $\Delta f = 1/(2\pi)[\ 5\ (2\pi)\ (10^3)] = 5$ kHz



# Quadrature Amplitude Modulation (QAM)

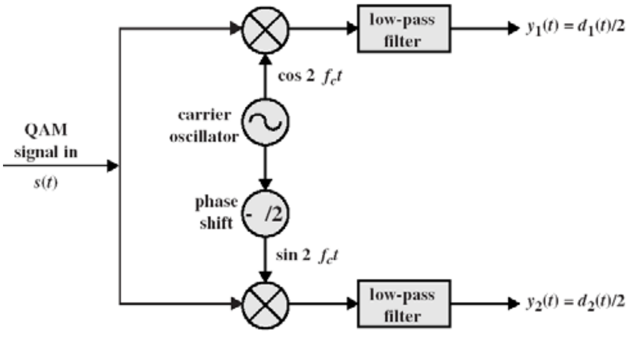
- QAM

$$s(t) = d_1(t) \cos(2\pi f_c t) + d_2(t) \sin(2\pi f_c t)$$



# Example: Problem 5-11

**5-10. The figure below shows the QAM demodulator corresponding to the to the QAM modulator shown in previous slide. Show that this arrangement DOES recover the two signals d1(t) and d2(t), which can be combined to recover the original signal.**






# Example: Problem 5-11 - Solution

**<u>Solution</u>:**

$$s(t) = d1(t)\cos(\omega_c t) + d2(t)\sin(\omega_c t)$$

**Use the following identities:**

$$\cos(2\alpha) = 2\cos^2(\alpha) - 1; \sin^2(\alpha) = 2\sin(\alpha)\cos(\alpha)$$

**For upper branch:**

$$s(t) \text{ X } \cos(\omega_c t) = d1(t)\cos(2\omega_c t) + d2(t)\sin(\omega_c t)\cos(\omega_c t)$$
$$= (1/2)d1(t) + (1/2)d1(t)\cos(2\omega_c t) + (1/2)d2(t)\sin(2\omega_c t)$$

**Use the following identities:**

$$\cos(2\alpha) = 1 - 2\sin^2(\alpha); \sin^2(\alpha) = 2\sin(\alpha)\cos(\alpha)$$

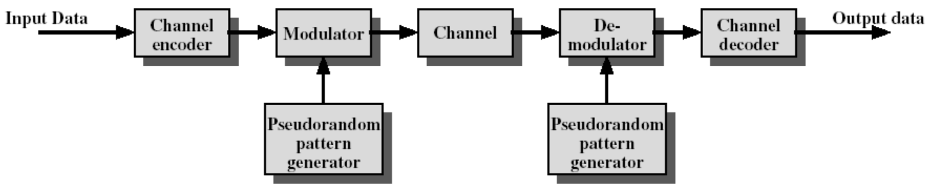
**For lower branch:**

$$s(t) \text{ X } \sin(\omega_c t) = d1(t)\cos(\omega_c t)\sin(\omega_c t) + d2(t)\sin(2\omega_c t)$$
$$= (1/2)d1(t)\sin(2\omega_c t) + (1/2)d2(t) - (1/2)d2(t)\cos(2\omega_c t)$$

**All terms at $2\omega_c$ are filtered out by the low-pass filter, yielding:**

$$y1(t) = (1/2)d1(t); \quad y2(t) = (1/2)d2(t)$$

# Spread Spectrum

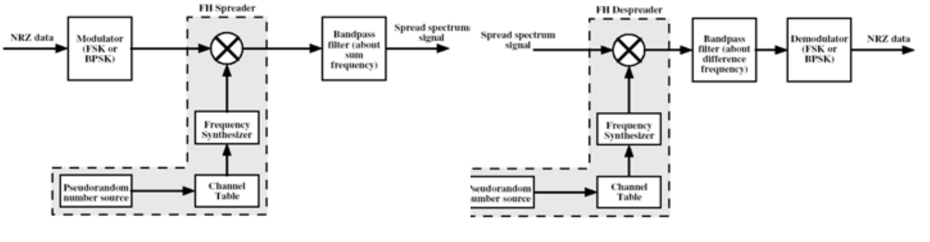



- **Key Characteristics:**
  - **Input data is encoded ➔ NARROWBAND analog signal**
  - **Further modulated using Pseudorandom numbers**
    - **This results in BROADBAND signal**
  - **At receiver – exact same pseudorandom number are generated and used to demodulate signal**
  - **The resulting narrowband signal is then decoded to retrieve original data**



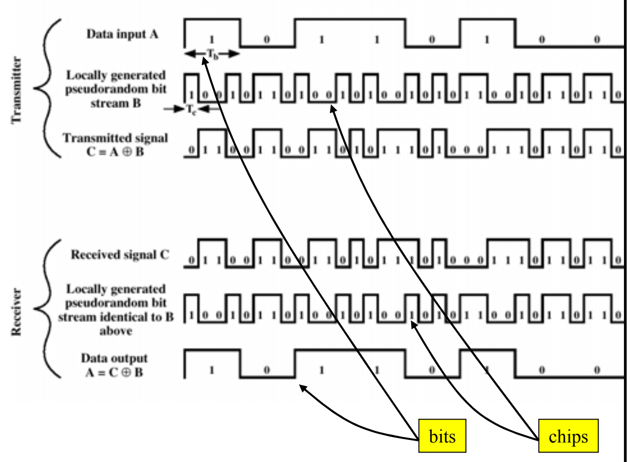
# Spread Spectrum – Frequency Hopping

- **The signal is transmitted over a *seemingly* random series of frequencies: jumping from one frequency to the next ➔ HOPPING**
- **A receiver, hopping between frequencies and in SYNCHRONIZATION with transmitter, is able to demodulate and decode the original message**



# Spread Spectrum – Direct Sequence

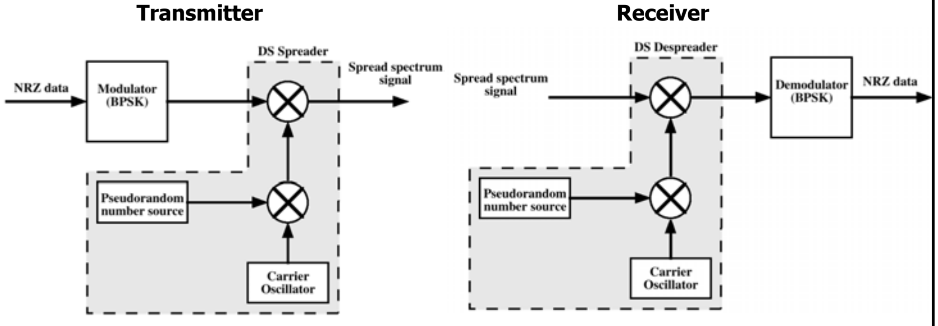
- Data input A is "spread" or multiplied by pseudorandom chip stream B ➔ $T_b/T_c$ chips per bit
- $R_c = 1/T_c$, $R = 1/T_b$
- SPREADING GAIN (G) = $T_b/T_c$ or $R_c/R$
- Intended receiver using SAME pseudorandom chips to recover original data
- Original data: A
- Transmitter forms: C = A XOR B
- Receiver forms: C XOR B = (A XOR B) XOR B = A XOR 0 = A ➔ which is the original data

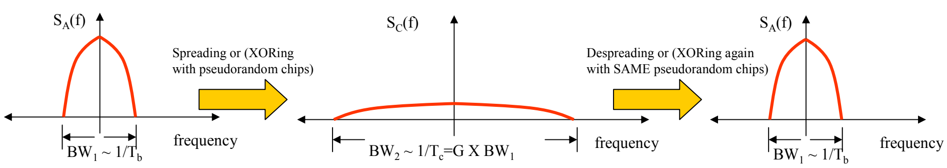



Remember (from COE-200) that B XOR B = 0, and A XOR 0 = A



# Spread Spectrum – Direct Sequence (2)



# Textbook Problems of INTEREST

- **Textbook Problems list: 5-4, 5-5, 5-6, 5-7, 5-8, 5-9, 5-11$^{s}$, 5-12, 5-13, 5-14, 5-16, 5-19, 5-20$^{s}$, 5-21, 5-22, 5-23, 5-24$^{s}$, 5-25, and 5-26**