Data Communication
Chapter 10
Data Encoding & Modulation

Himal Acharya
Course Instructor

Outline

- Encoding Vs Modulation
- Encoding of Digital Data as Digital Signals & Its Techniques, Amplitude, Frequency, and Phase Shift Keying. Pulse code and Delta Modulation. Analog Modulation (Amplitude, Frequency, and Phase Modulation)
- Multilevel Modulation, Differential PSK, QPSK ,QPSK Modem, Higher-Data Rate Modems

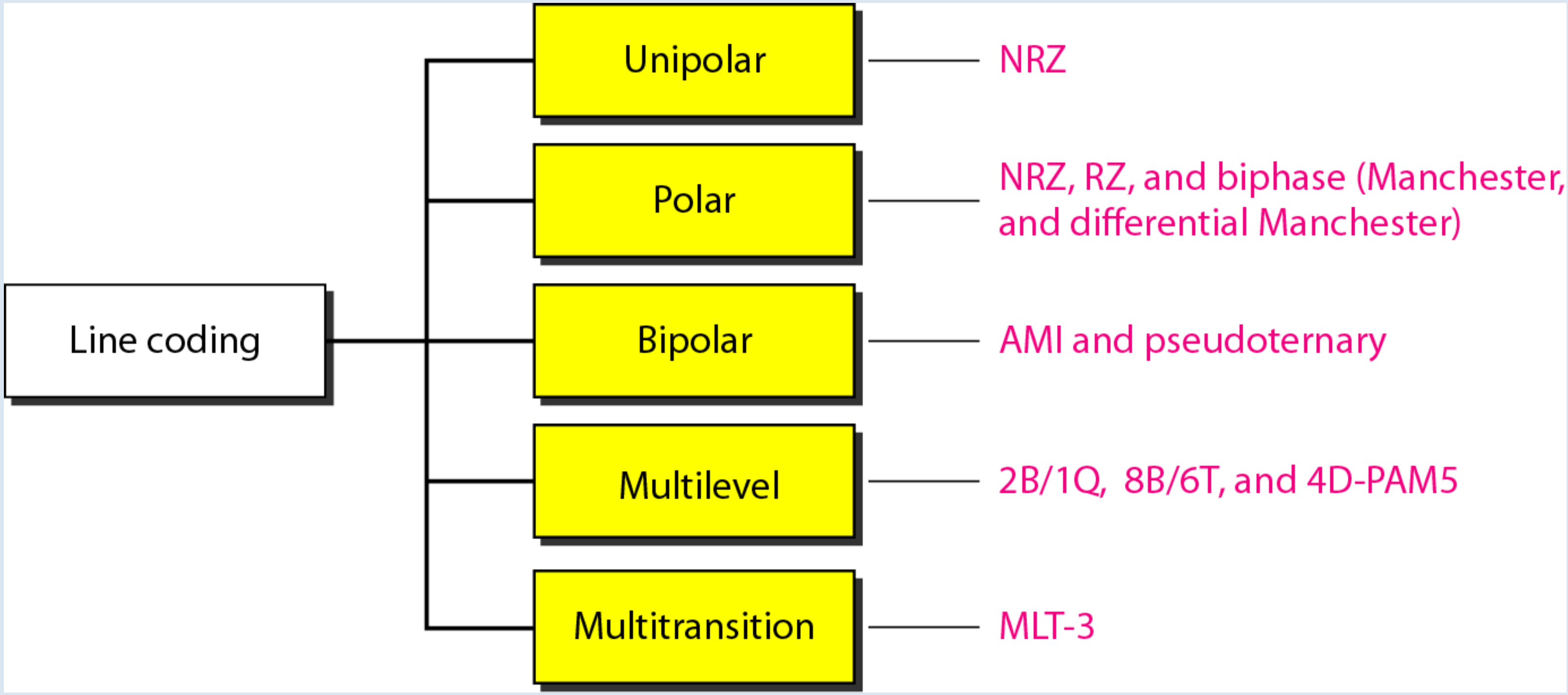
Encoding

- It is the technique of assigning binary value to a signal.
- In case of digital data, encoding is directly done while for analog data, sampling and quantization are necessary before encoding.

Line Coding (Digital to Digital Conversion)

- Line coding is the method used for converting a binary information sequence into a digital signal in a digital communication system.
- The last signal-processing operation in the transmitter is that of line coding, the purpose of which is to represent/convert sequence of bits by/into a sequence of (electrical) pulses.
- The selection of a line coding technique involves several considerations.
- Some line coding methods have built-in-error detecting capabilities, other methods have better immunity to noise and interference.
- Finally, the complexity and the cost of the line code implementations are always factors in the selection for a given application.

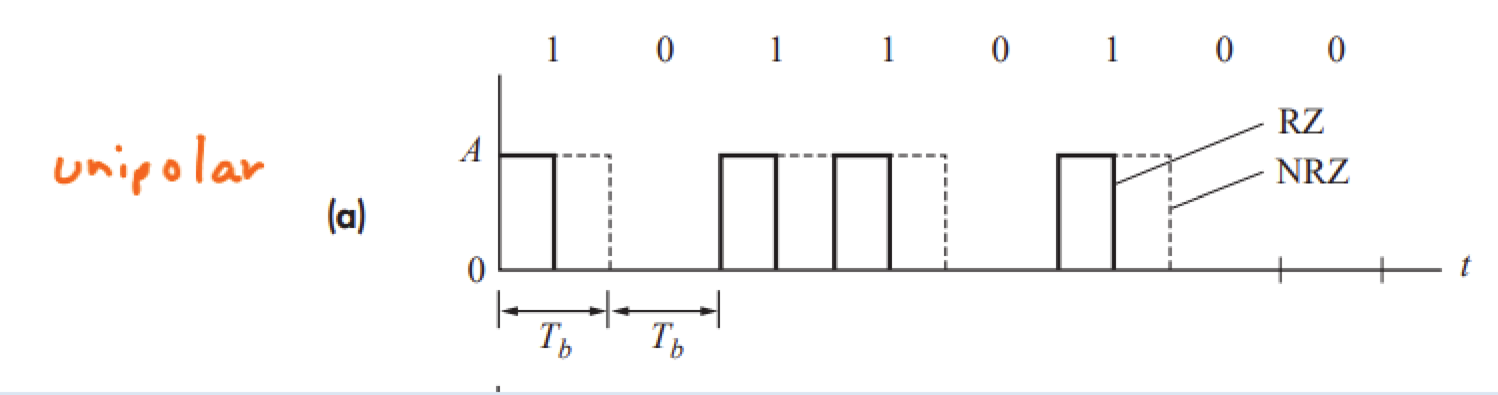
Figure *Line coding schemes*



Types of Line Coding

a) Unipolar Signaling

- On-off waveform
- The simplest scheme is the unipolar nonreturn-to-zero (NRZ) encoding in which a binary 1 is transmitted by sending +A voltage level, and 0 is transmitted by sending a 0 voltage. It has "on" pulses for full bit duration T_b.
- In the unipolar return-to-zero (RZ) format, the pulse duration is smaller than T_b after which the signal returns to the zero level.



Unipolar Signaling

If binary 0s and 1s both occur with probability ½, then the average transmitted power for this line code is $(1/2)A^2 + (1/2)0^2 = A^2/2$.

Advantage

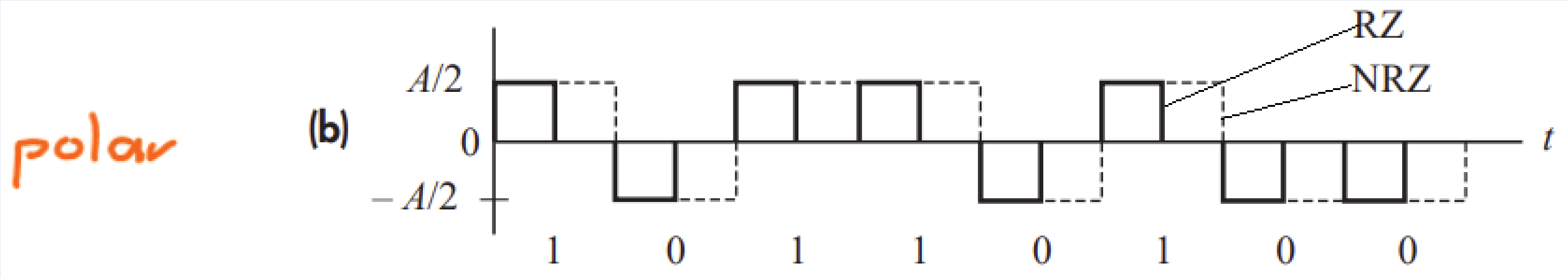
- Simple to implement

Disadvantage

- Presence of DC level
- Contains low frequency components causes 'Signal Droop'
- Doesn't have any error correction capability.

b) Polar signaling

- The polar NRZ (NRZ- L)encoding method that maps a binary 1 to +A/2 and binary 0 to – A/2.
- Polar return to zero RZ format : Pulse duration is smaller than T_b after which the signal return to the zero level.



Polar Signaling

Polar NRZ encoding is more efficient than unipolar NRZ in terms of average transmitted power. Its average power is given by $(1/2)(+A/2)^2 + (1/2)(-A/2)^2 = A^2/4$.

Advantage

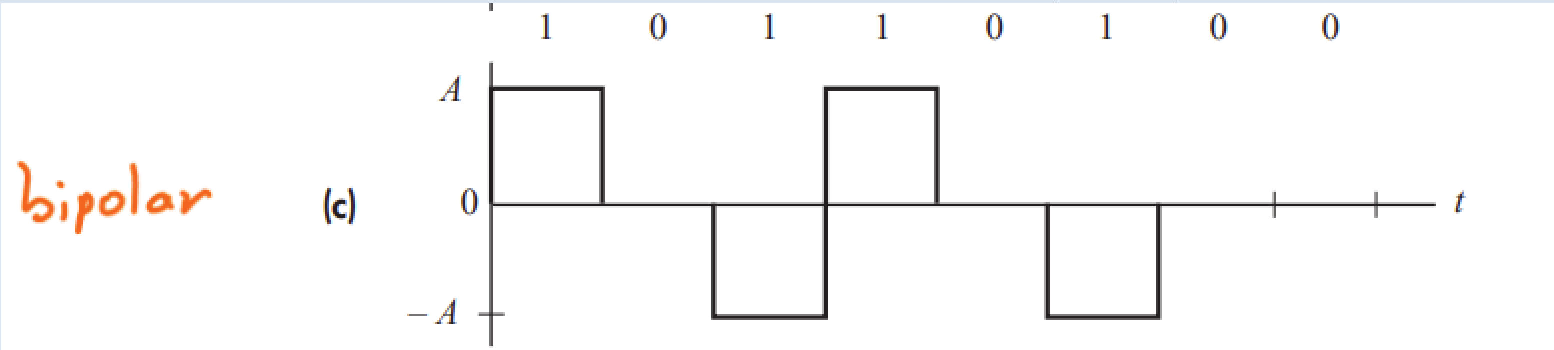
- Simple to implement
- No DC component if message contains 1s and 0s in equal proportion

Disadvantage

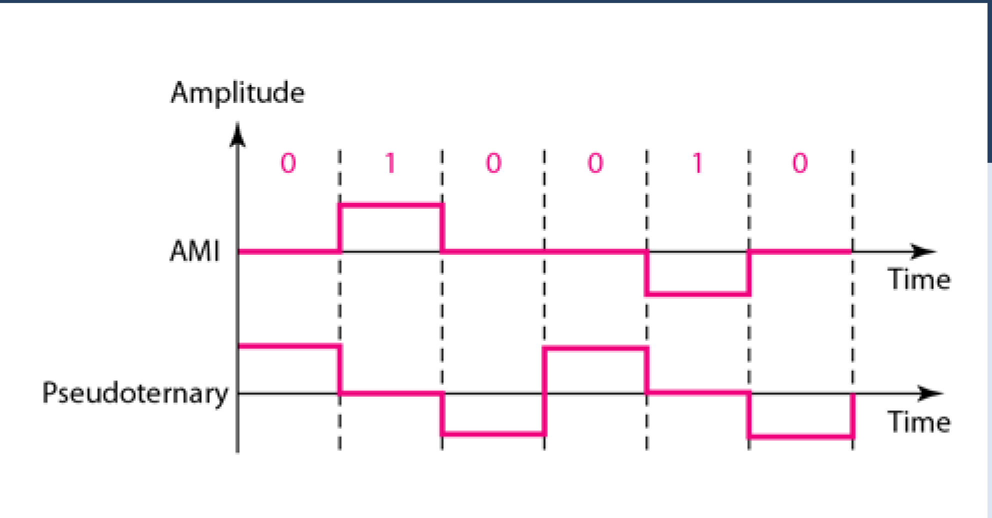
- Contains low frequency component
- Doesn’t have any error correction capability
- Doesn’t posses any clocking component for ease of synchronization

c) Bipolar Signaling

- In this method binary 0s are mapped into 0 voltage, thus making no contributions to the digital signals
- Consecutive 1s are alternately mapped into +A/2 and –A/2 (or +A to -A).
- Use three amplitude levels
-



Bipolar Signaling



Advantage:

- Power spectrum of the transmitted signal has no dc component and relatively insignificant low-frequency components for the case when symbols 1 and 0 occur with equal probability
- Posses single error detection capability

Disadvantage

Doesn’t posses any clocking component for ease of synchronization

d) High Density Bipolar Order n Encoding (HDBn)

- An enhancement of Bipolar signaling technique (i.e. relies on the transmission of both positive and negative pulses).
- It is based on Alternate Mark Inversion (AMI), but extends this by inserting violation codes whenever there is a run of (n+1) or more 0’s.
- This and similar (more complex) codes have replaced AMI in modern distribution networks

HDB3 Encoding

- AMI based and extends this by inserting violation codes whenever there is a run of 4 or more 0’s

HDB3 Encoding Rules

- In HDB3, a string of 4 consecutive zeros are replaced by either 000V or B00V.
- ‘B’ conforms to the AMI rule.
- ‘V’ is a violation of the AMI

The substitution is chosen according to the following rules.

a) If the number of nonzero pulses after the last substitution is odd, the substitution pattern will be 000V, which makes the total number of nonzero pulses even.

b) If the number of nonzero pulses after the last substitution is even, the substitution pattern will be BOOV, which makes the total number of nonzero pulses even. (if there is no 1 then assume no. of 1s is even.)

HDB3 Encoding

Example 1:(Figure in classroom board)

- The pattern of bits

" 1 0 0 0 0 1 1 0 " encoded in HDB3 is

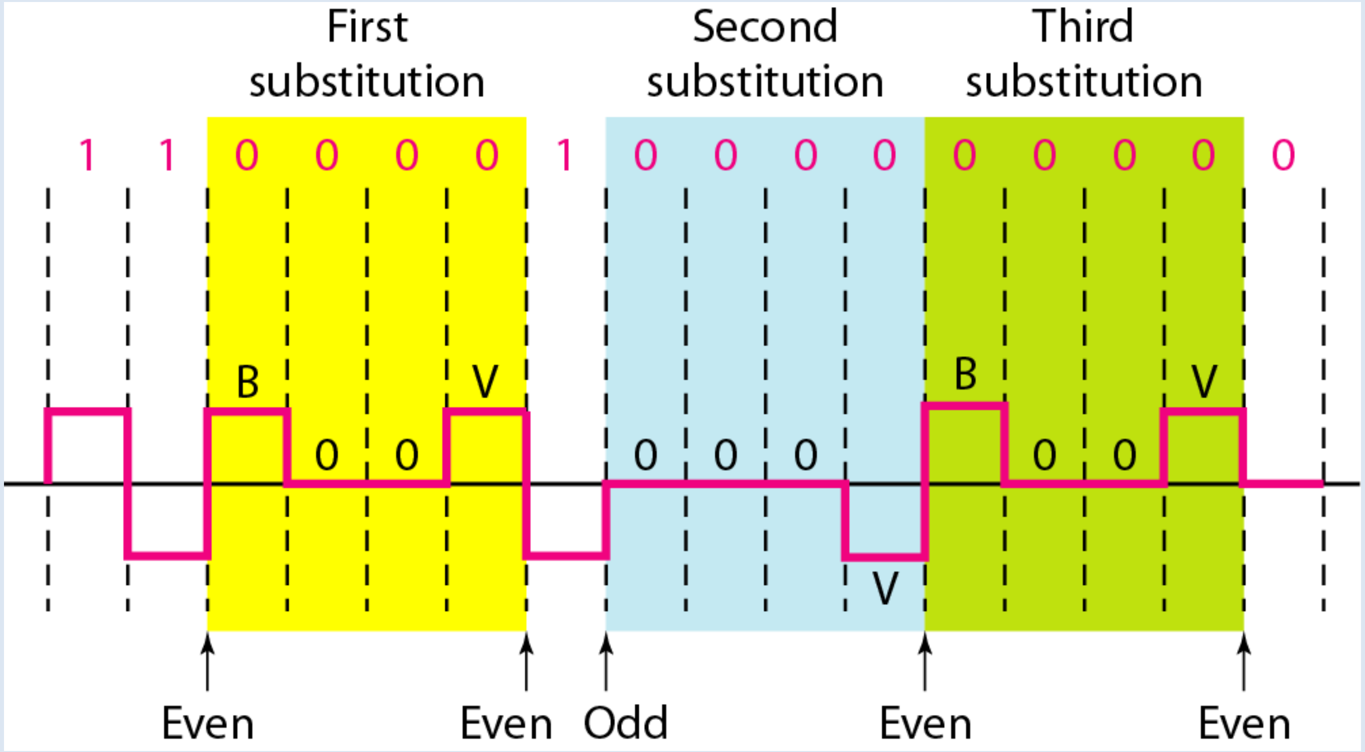
" + 0 0 0 V - + 0 "

(the corresponding encoding using AMI is " + 0 0 0 0 - + ")

Example 2 (Figure in classroom board)

Pattern	1	0	1	0	0	0	0	0	1	1	0	0	0	0	1	1	0	0	0	0	0	0
HDB3	1	0	1	B	0	0	V	0	1	1	B	0	0	V	1	1	B	0	0	V	0	0
	+	0	-	+	0	0	+	0	-	+	-	0	0	-	+	-	+	0	0	+	0	0
AMI	1 +	0	1 -	0	0	0	0	0	1 +	1 -	0	0	0	0	1 +	1 -	0	0	0	0	0	0

Figure *Different situations in HDB3 scrambling technique*



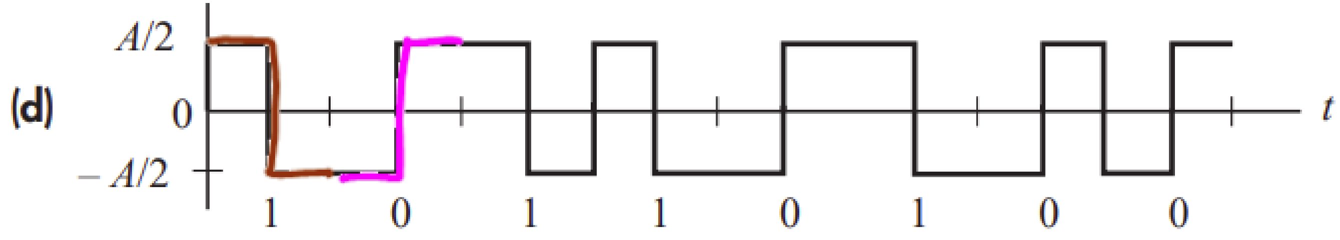
e) Split-phase (Manchester) Signaling : Biphase

- In this method, 1s are represented with a positive half-interval pulse followed by a negative half-interval pulse, and vice versa for the representation of 0s.

(In clear view): A binary 1 is denoted by a transition from +A/2 to –A/2 in the middle of the bit time interval, and a binary 0 by a transition from –A/2 to +A/2.

- Also called twinned binary

Manchester



Manchester Signaling

Advantage

- Guarantee zero DC component regardless of the message sequence.
- Self-synchronizing

Disadvantage

- Doesn’t have error detection capability
- The pulse rate is essentially double that of binary encoding, and this factor results in small content low frequencies.

Application:

Ethernet and token-ring LAN Standards

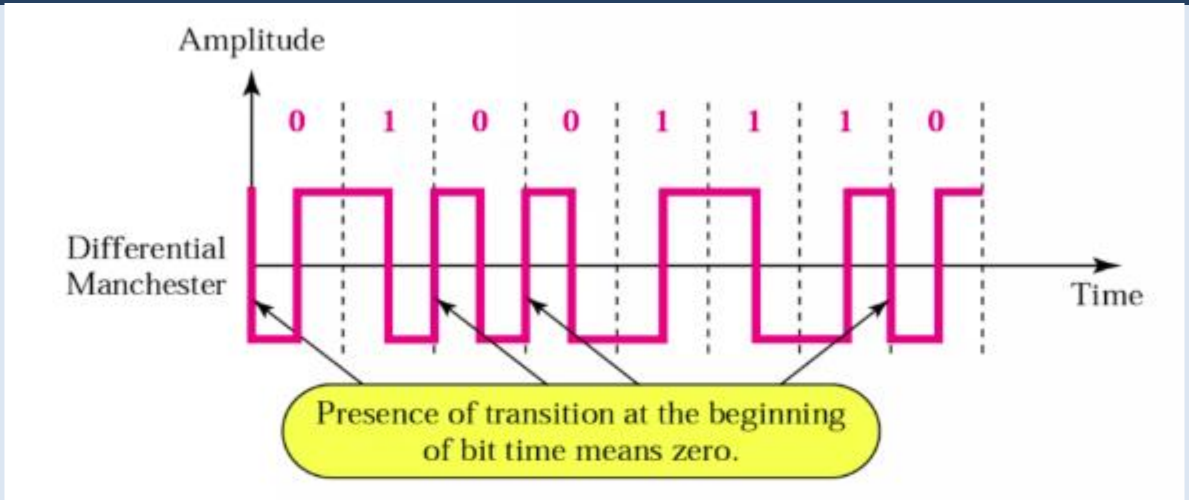
f) Differential Manchester Encoding

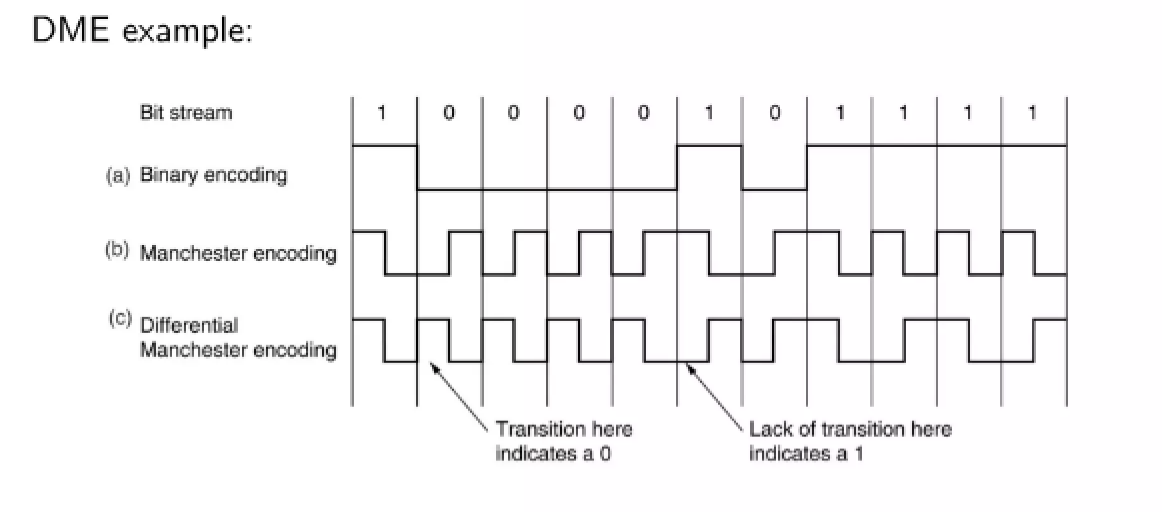
- Used in Token-ring networks, which retains the transition in the middle of every bit frame, but the binary sequence is mapped into the presence or absence of transitions in the beginning of the bit intervals.

- In this type of encoding, a binary 0 is marked by a transition at the beginning of an interval, whereas a 1 is marked by the absence of a transition.

(Figure in classroom board)

Differential Manchester Encoding- Another Example

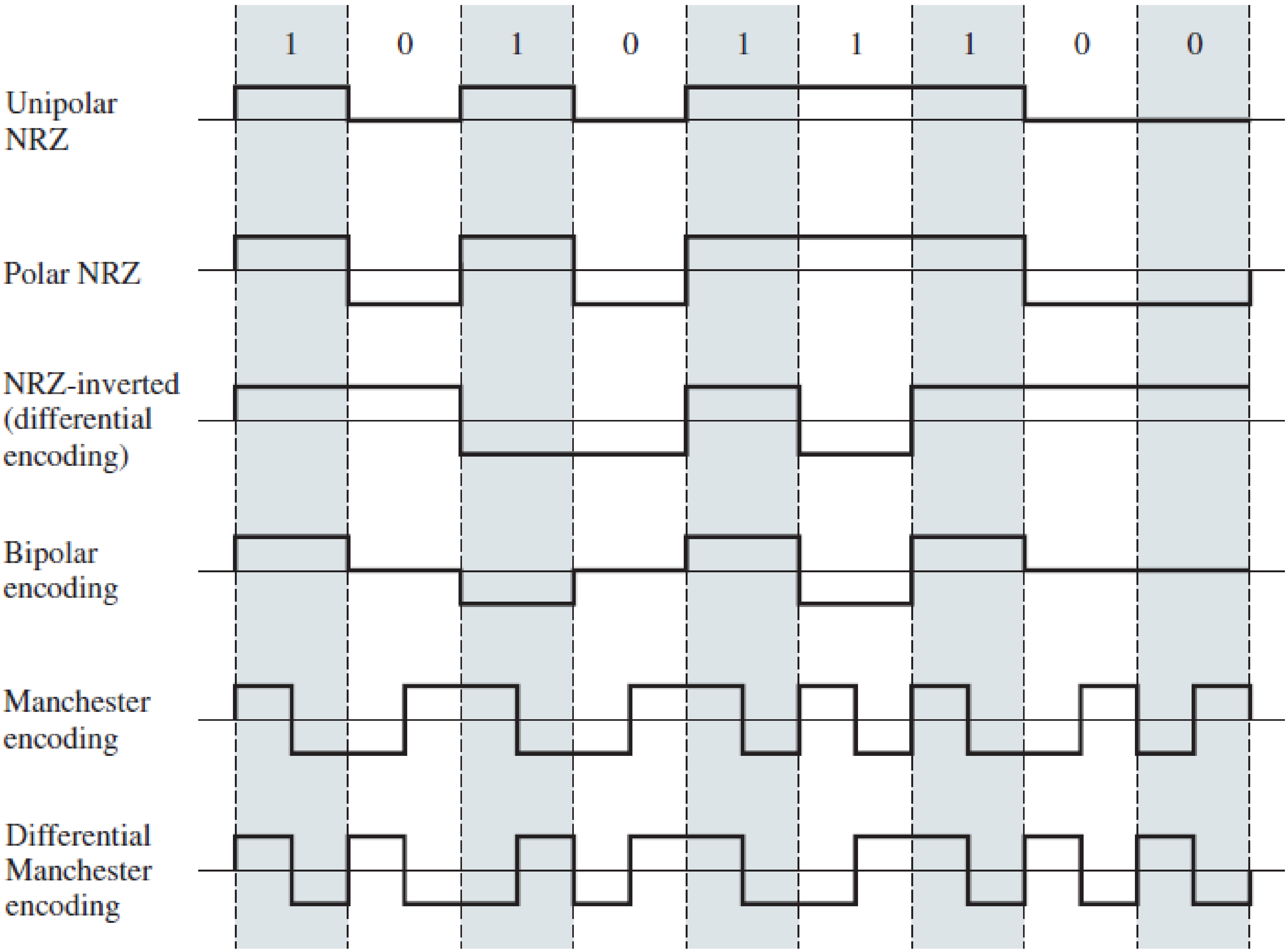




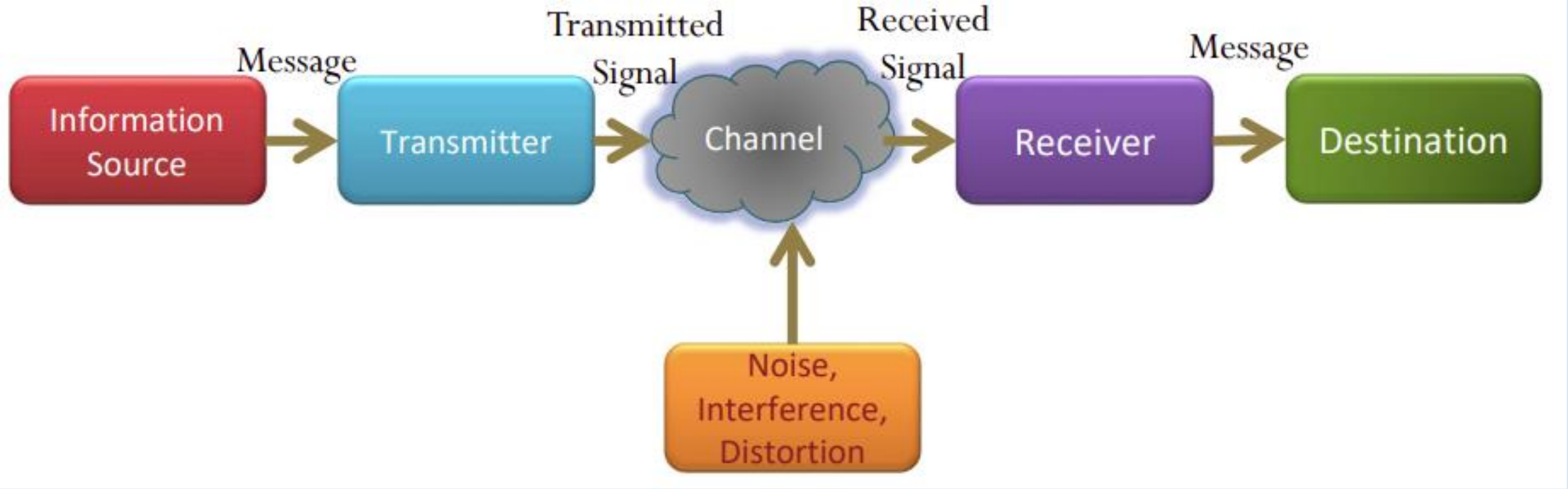
g) NRZ-inverted (NRZ-I)

- Starting at a given level, the sequence of bits determine the subsequent transitions at the beginning of each interval.
- Two-level signal has a transition at a boundary if the next bit we are going to transmit is a logical 1, and doesn’t have a transition if the next bit that we are going to transmit is a logical 0.

(Figure in classroom board)


1
0
1
0
1
1
1
0
0
Unipolar NRZ
Polar NRZ
NRZ-inverted (differential encoding)
Bipolar encoding
Manchester encoding
Differential Manchester encoding

Basic Elements of Communication



Modulation

- Baseband – band of frequencies of the signal delivered by the source.
- Telephony- Audio Band 300 Hz to 3.4 kHz
- Modulation- Baseband signal is used to modify some parameter of a radio-frequency RF signal.

Carrier Sinusoid of higher frequency

Modulating Signal Message Signal

- Fundamental Goal: Produce an information-bearing modulated wave whose properties are best suited to given communication task
- The part of the system that performs this task is called modulator.
- More general definition: Modulation is systematic alteration of one waveform, called the carrier according to characteristics of another waveform, the modulating signal or message.

Benefits and applications

a) Reasonable Antenna Size:

- For effective radiation of power over a radio link, the antenna size must be on the order of the wavelength of the signal to be radiated.
- "Too low frequency " = " Too large antenna"
- Audio signal frequencies are so low (wavelengths are so large) that impracticably large antennas will be required for radiation.

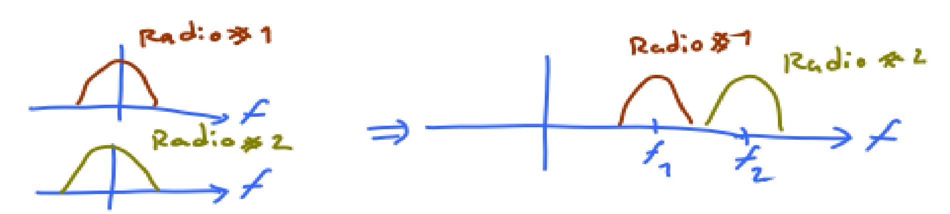
❖ Shifting the spectrum to a higher frequency (a smaller wavelength) by modulation solves the problem.

$$c = f\lambda \Rightarrow \lambda = \frac{c}{f} = \frac{3\times10^8}{?}$$

$$f = 3\ \text{kHz} \Rightarrow \lambda = \frac{3\times10^8}{3\times10^3} = 10^5\ \text{m} = 100\ \text{km}$$

$$f = 3\ \text{GHz} \Rightarrow \lambda = \frac{3\times10^8}{3\times10^9} = 0.1\ \text{m} = 10\ \text{cm}$$

$$f = 60\ \text{GHz} \Rightarrow \lambda = \frac{3\times10^8}{60\times10^9} = 5\ \text{mm}$$



b) Multiplexing

- If several signals (for example, all radio stations), each occupying the same frequency band, are transmitted simultaneously over the same transmission medium, interference occurs.
  - ❖ Difficult to separate or retrieve them at a receiver.
  - ❖ One solution is to use modulation whereby each radio station is assigned a distinct carrier frequency.
    - ➢ Each station transmits a modulated signal, thus shifting the signal spectrum to its allocated band, which is not occupied by any other station.
    - ➢ When you tune a radio or television set to a particular station, you are selecting one of many signals being received at that time.
    - ➢ Since each station has a different assigned carrier frequency, the desired signal can be separated from the others by filtering.

Types of Modulation

a) Analog Modulation (Analog to Analog Conversion)

Amplitude Modulation AM

Frequency Modulation FM

Phase Modulation PM

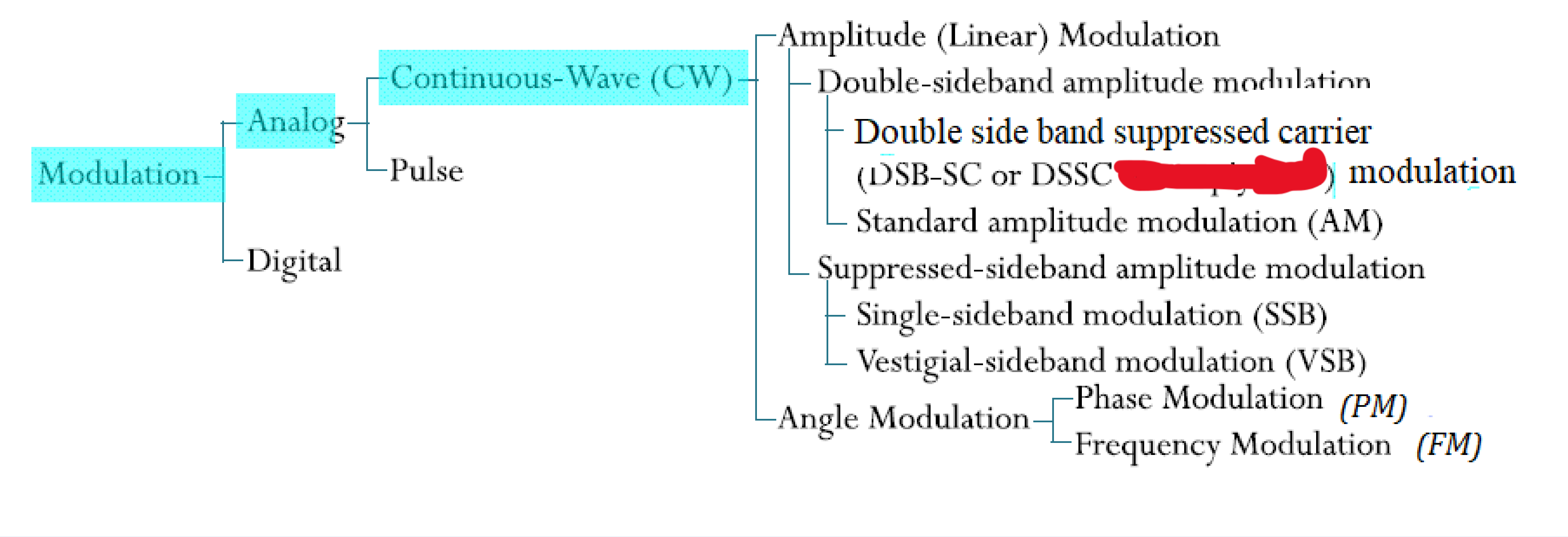
b) Digital Modulation

i) Transition from Analog to Digital Signaling

- Pulse Code Modulation PCM
- Delta Modulation DM

ii) Shift Keying (Digital to Analog Conversion)

- Amplitude Shift Keying ASK
- Frequency Shift Keying FSK
- Phase Shift Keying PSK


Modulation
Analog
Digital
Continuous-Wave (CW)
Pulse
Amplitude (Linear) Modulation
Double-sideband amplitude modulation
Double side band suppressed carrier
(DSB-SC or DSSC) modulation
Standard amplitude modulation (AM)
Suppressed-sideband amplitude modulation
Single-sideband modulation (SSB)
Vestigial-sideband modulation (VSB)
Angle Modulation
Phase Modulation (PM)
Frequency Modulation (FM)

Some Definitions

- $g(t) \equiv g(t)\cos(2\pi f_c t)$, $g(t) \equiv g(t)\cos(2\pi f_c t + \ \varphi)$
- The sinusoidal signals $\cos(2\pi f_c t)$ and $\cos(2\pi f_c t + \ \varphi)$ are called carrier signals and f_c is called carrier frequency.
- Has amplitude A_c and expression of carrier signal is $A_c \cos(2\pi f_c t + \ \varphi)$
- We use m(t) to denote baseband signal and band-limited to *B*; that is |M(f)| = 0 for |f| >B. Usually call it the message or modulating signal.

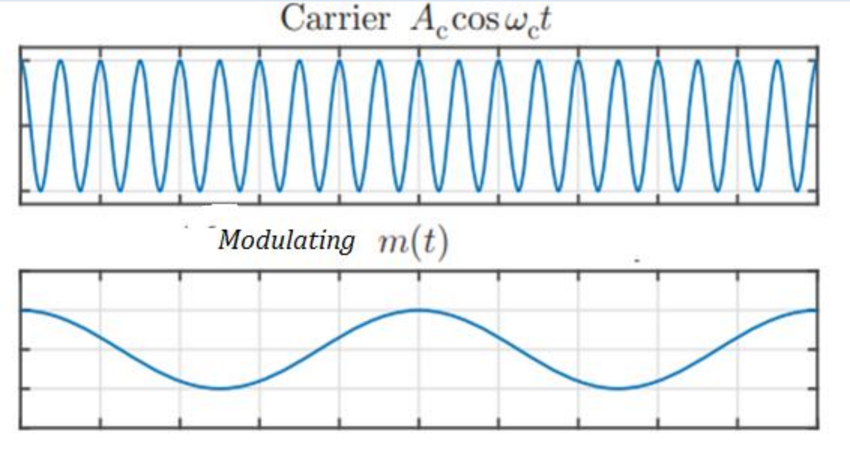
- A sinusoidal carrier signal $A_c \cos(2\pi f_c t + \varphi)$ has three basic parameters: amplitude, frequency, and phase.
- Varying these parameters in proportion to the baseband signal results in amplitude modulation (AM), frequency modulation (FM), and phase modulation(PM) respectively.
- These techniques are called continuous-wave modulation.

Amplitude Modulation

- Amplitude modulation (AM) is an analog and linear modulation process
- Amplitude modulation is characterized by the fact that the amplitude A_c of the carrier$A_c \cos(2\pi f_c t + \varphi)$ is varied in proportion to the baseband (message) signal m(t).
- Because the amplitude is linearly related to the message signal, this technique is also called linear modulation.
- An amplitude-modulated (AM) wave , as a function of time as follows.

$$s(t) = A_c[\, 1 + k_a \; m(t) \,] \cos(2\pi f_c t)$$

- k_a is amplitude sensitivity of the modulator
- We observe that the envelope of s(t) has essentially the same shape as the baseband signal m(t).



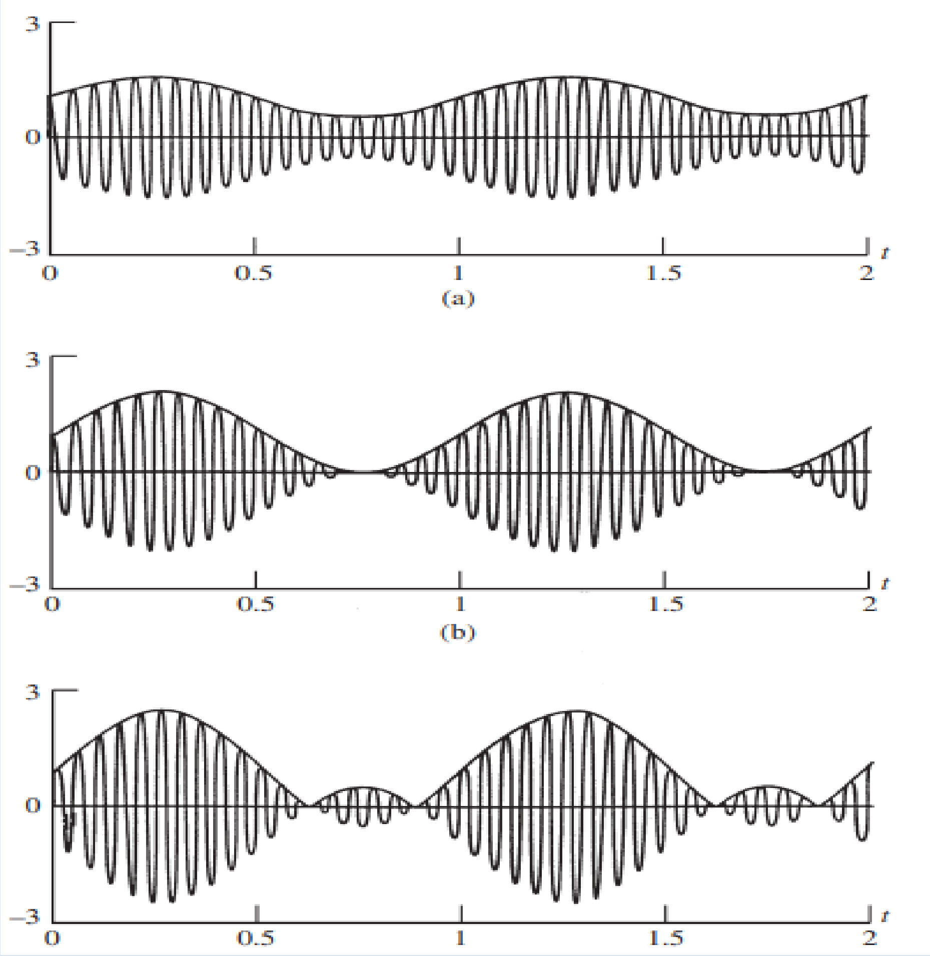
- $| k_a$ m(t) | < 1

Undermodulation

- $| k_a$ m(t) | =1

100 % modulation

- $| k_a$ m(t) | > 1
- Overmodulation,
- Phase reversal

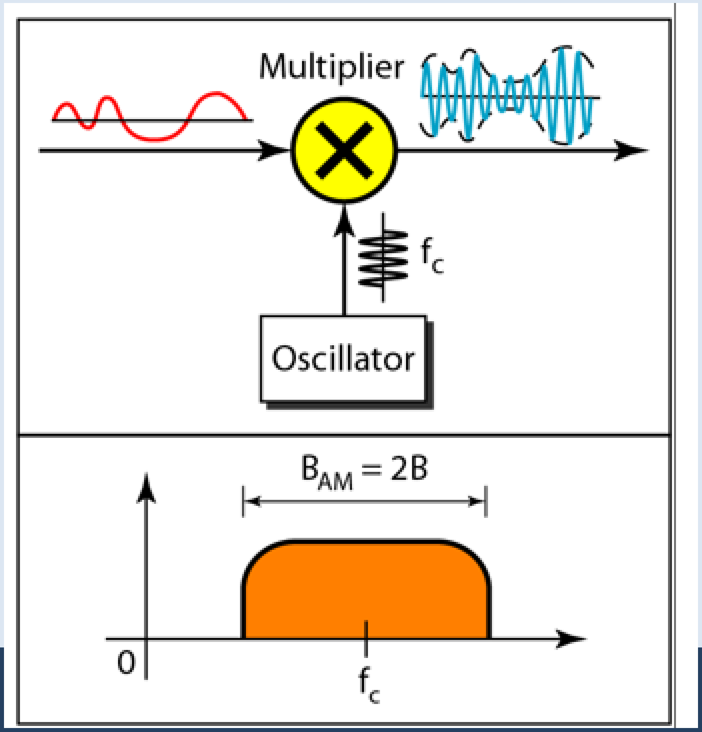


- Carrier frequency f_c is much greater than the highest frequency component B (or f_m) of the message signal m(t), $f_c >> f_m$.

B is the message bandwidth. If this condition is not satisfied, an envelope cannot be visualized satisfactorily.

- The modulating signal is the envelope of the carrier.
- The required bandwidth is 2B for

Standard AM, B for SSB.



- AM involves the variation of the carrier signal's amplitude in direct proportion to the modulating signal m(t).
- AM is simple to implement and can be accomplished inexpensively with a small number of components; but AM has a low power efficiency (ratio of power in the message signal relative to the total transmitted power) and is very susceptible to noise and interference.

Some Formula for standard AM:

Modulation index $\mu = \frac{A_m}{A_C}$

Bandwidth = $2f_m$ (for DSB/ standard AM); this is different for other AM techniques

Transmitted Power P_t of single tone AM wave is the sum of carrier wave power P_c and side band P_s. $P_t = P_c + P_s$; $P_c = \frac{A_c^2}{2}$; $P_s = \frac{A_m^2}{4}$; $P_t = P_c \left(1 + \frac{(\mu)^2}{2}\right)$

$$P_t = \frac{A_c^2}{2}\left(1 + \frac{\left({A_m}/{A_c}\right)^2}{2}\right)$$

Transmission Efficiency of Single Tone AM $\boldsymbol{\eta} = \frac{\mu^2}{\mu^2+2} x100\%$

Example: An audio signal of $5Sin500\pi t$ is used for AM with a carrier of $25Sin100000\pi t$. Calculate:

i) Modulation index

ii) Required Bandwidth

iii) Total Power

iv) Efficiency of AM

Solution: Audio (Message) signal $m(t) = 5Sin500\pi t$. Comparing with standard equation $\mathrm{m(t)} = A_m Sin2\pi f_m t$.

Message Amplitude A_m = 5 V; Modulating frequency f_m= 250 Hz

Carrier signal c$(t) = 25Sin100000\pi t$. Comparing with standard equation $\mathrm{c(t)} = A_c Sin2\pi f_c t$.

Carrier Amplitude A_c = 25 V; Carrier frequency f_c= 50000 Hz

i) Modulation index $\mu = \frac{A_m}{A_C}$ = 5/ 25 = 0.2 = 20%

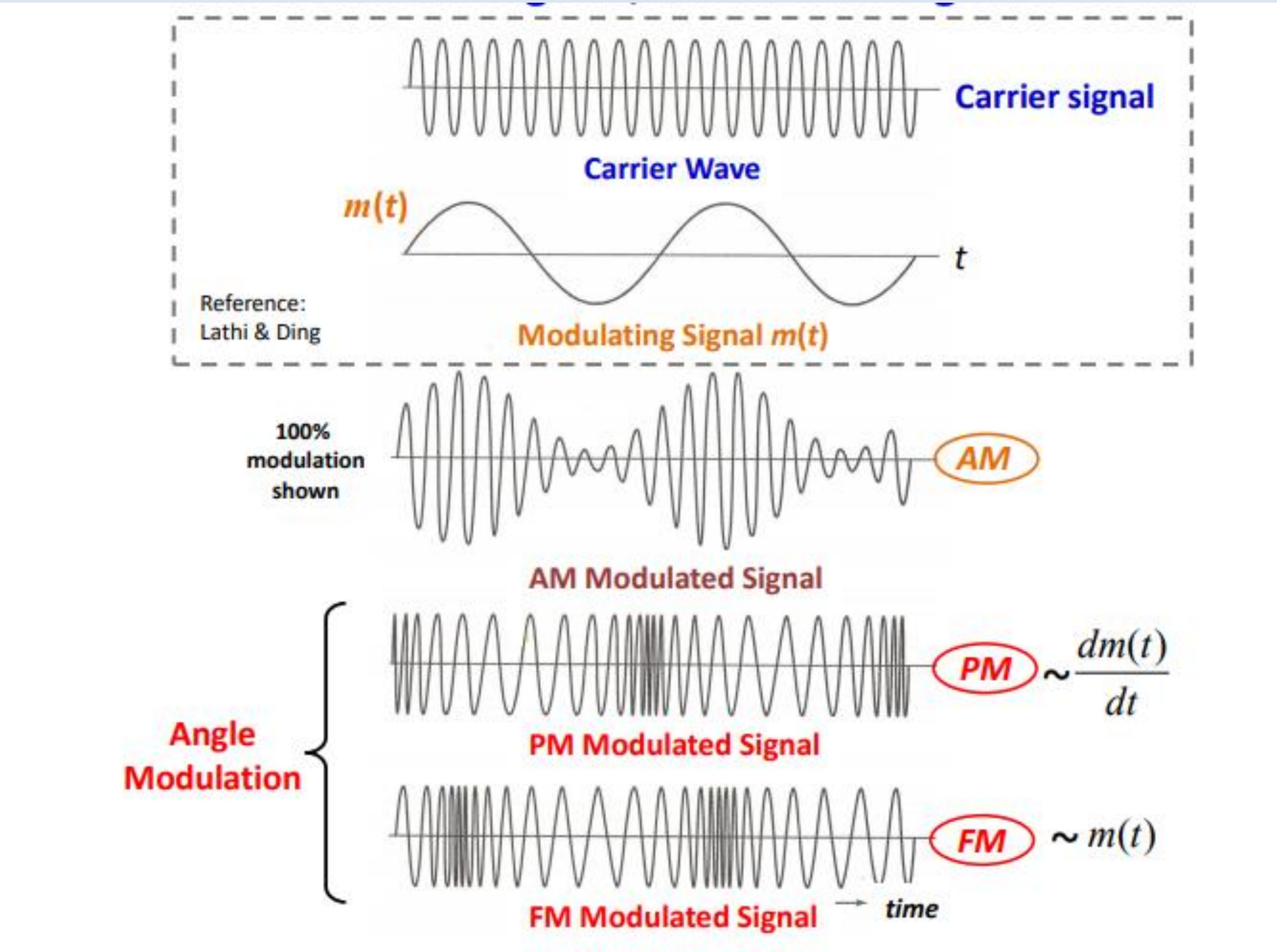
ii) Required Bandwidth = $2f_m = 2*250$ = 500 Hz

iii) Total Power $P_t = \frac{A_c^2}{2}\left(1 + \frac{\left({A_m}/{A_c}\right)^2}{2}\right) = \frac{25^2}{2}\left(1 + \frac{0.2^2}{2}\right)$ = 318.75 W

iv) Efficiency of AM $\boldsymbol{\eta} = \frac{\mu^2}{\mu^2+2}$x100% = $\frac{0.2^2}{0.2^2+2} x100\% = 1.96\%$

Angle Modulation

- FM
- PM



Advantages of Angle Modulation

- Angle modulation is very efficient in rejecting interference (i.e., it minimizes the effect of amplitude noise on the signal transmission).
- Angle modulation allows for more efficient use of transmitter power.
- Angle modulation can handle a greater dynamic range in the modulating signal without distortion (as would occur in AM).

- The general angle modulated signal is given by:

$$s(t) = A_c cos(\theta(t))$$

θ (t) is generalized angle.

$$s(t) = A_c cos(w_c t + \theta_0)$$

Differentiating θ (t) with respect to time ,

$$w_c = \frac{\mathrm{d}\theta(\mathrm{t})}{\mathrm{d}t}$$

- The term $\frac{d\theta}{dt}$ varies with time and depends upon the carrier frequency at that instant. So called instantaneous angular frequency.
- Instantaneous angular frequency (w_i) = $\frac{d\theta_i(t)}{dt}$

Or w_i dt = d $\theta_i(t)$

Integrating both sides w.r.t. to t,

$\theta_i(t)$ = $\int_{-\infty}^{t} w_i(t)dt$ $\quad \therefore \int_{-\infty}^{t} w_i(t)dt = \theta_i(t)$

Since θ(t) changes with instantaneous value of message signal i.e. angle is changed.

Frequency Modulation FM

In frequency modulation the instantaneous angular frequency w_i varies linearly with the modulating signal m(t).

$$w_i(t) = w_C + k_f m(t)$$

Integrating wr to t,

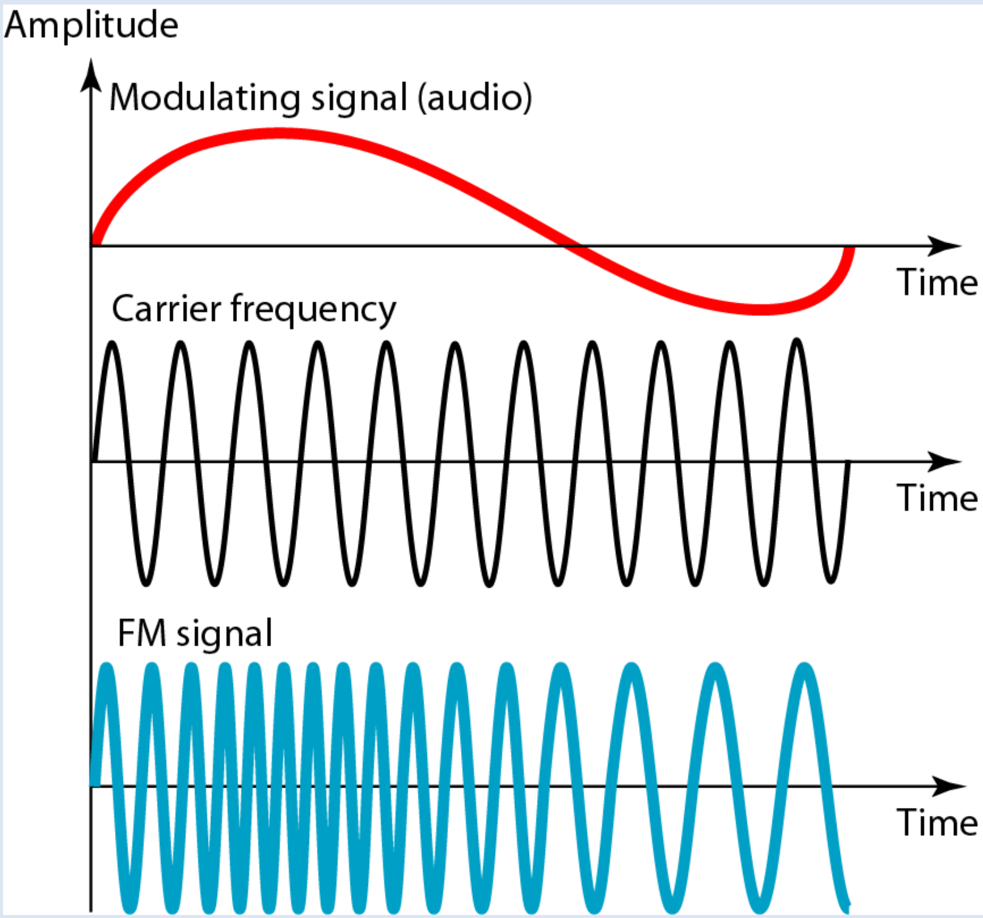
$$\int w_i(t)dt = \int [w_C + k_f m(t)]dt$$

$$\theta_i(t) = w_C t + k_f \int m(t)dt$$

K_f is frequency deviation (sensitivity constant) Unit radian volt

$$s_{FM}(t) = A_C Cos(w_C t + k_f \int m(t)dt)$$

Figure *Frequency modulation; B=Message Bandwidth, b =Modulation Index*



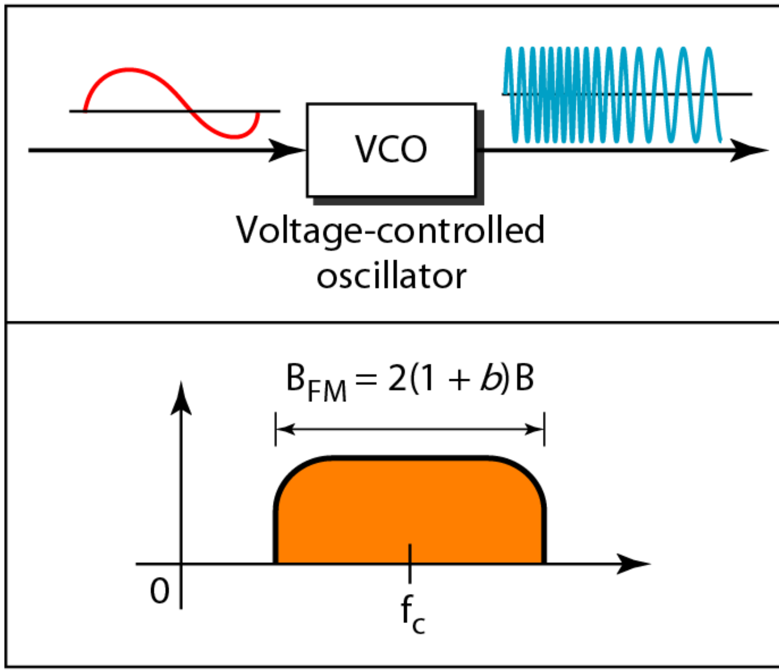
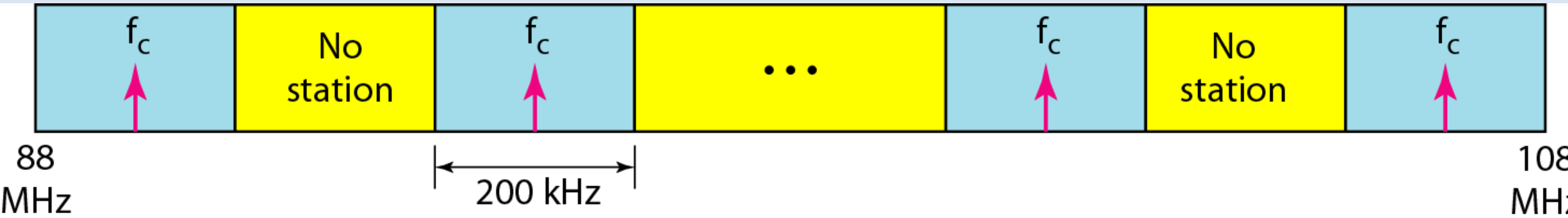
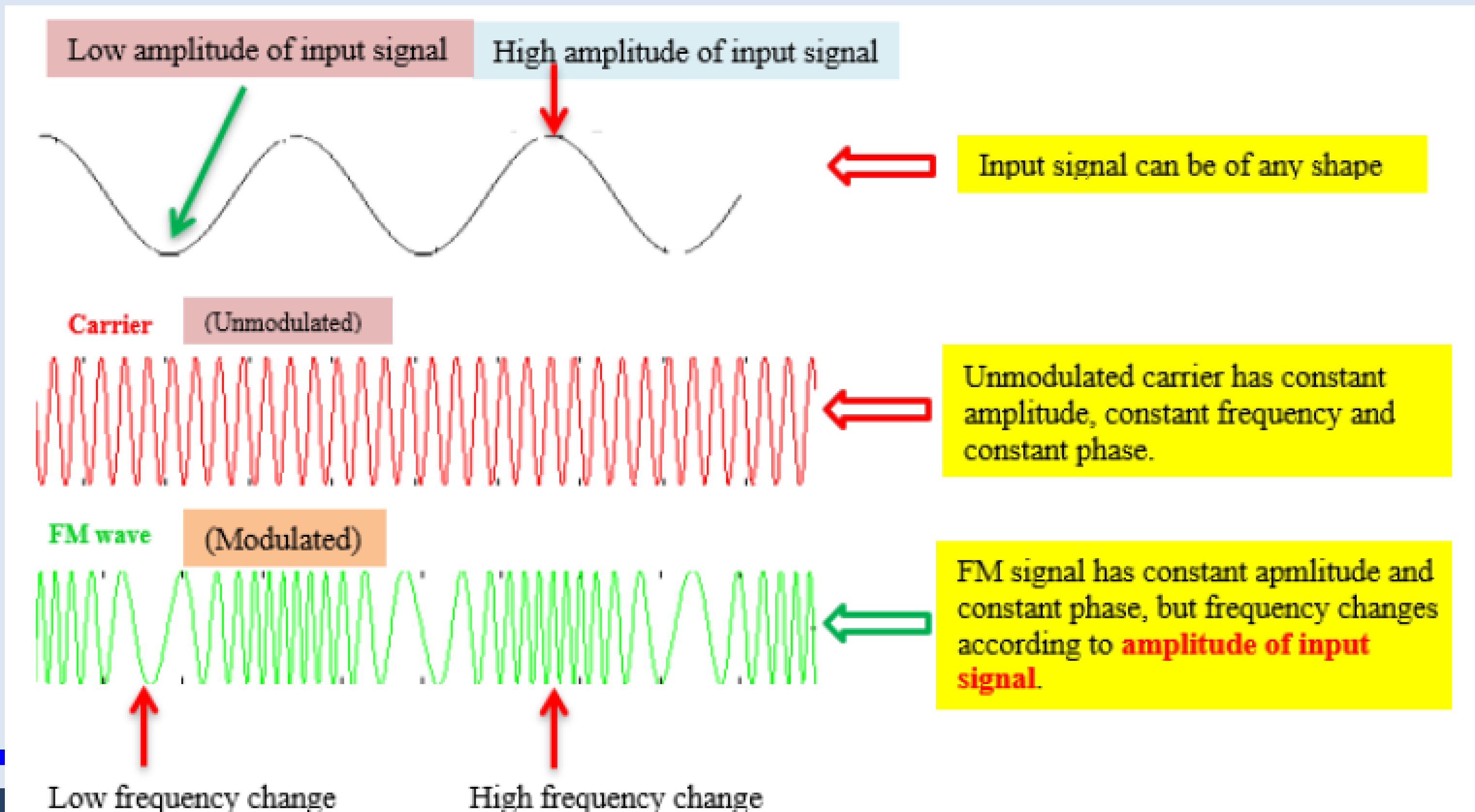


Figure *FM band allocation*





Phase Modulation (PM)

- The modulating signal only changes the phase of the carrier signal.
- The phase change manifests itself as a frequency change but the instantaneous frequency change is proportional to the derivative of the amplitude.
- The bandwidth is higher than for AM.

Phase Modulation PM

- The general angle modulated signal is given by:

$$s(t) = A_c cos(\theta(t))$$

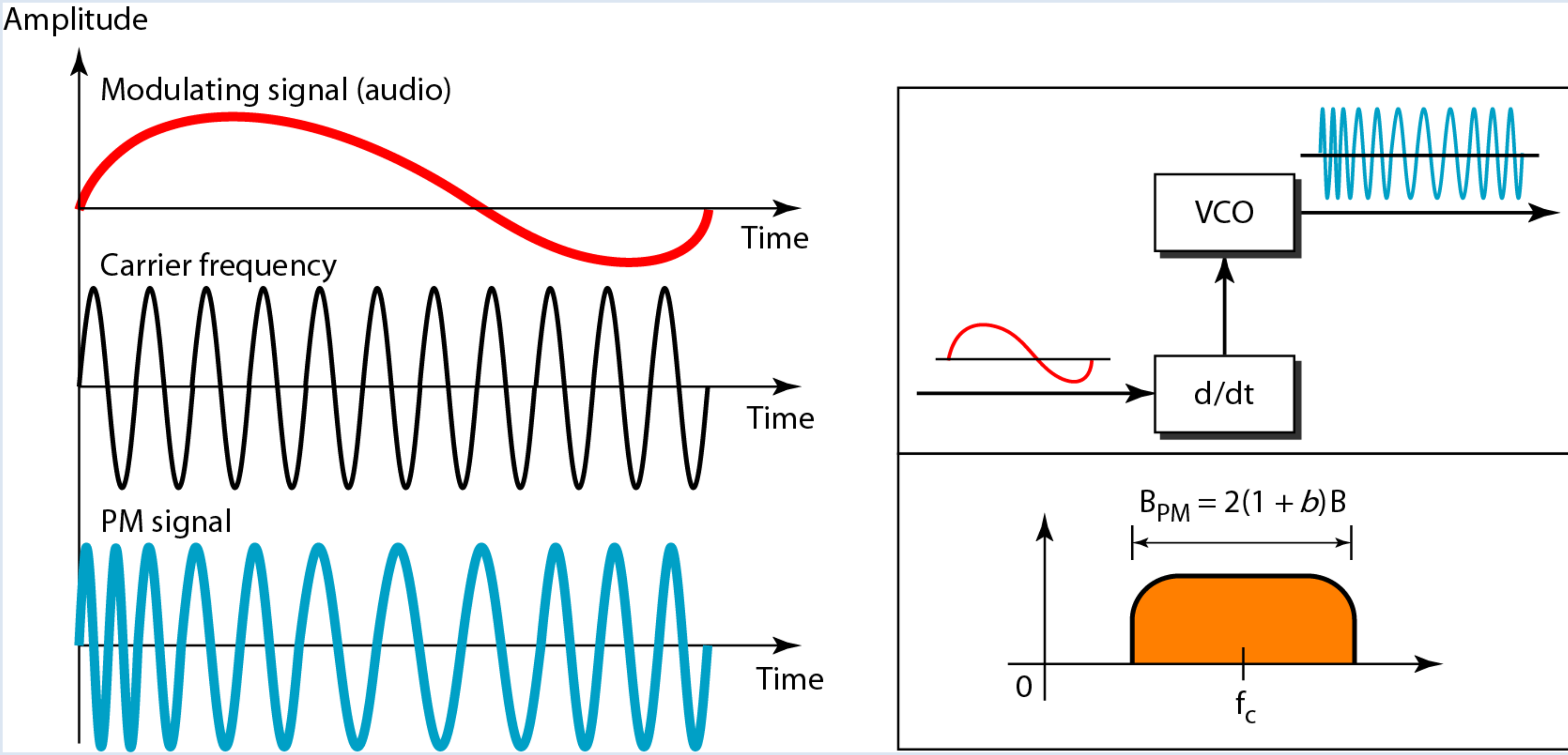
θ (t) is generalized angle.

$$s(t) = A_c cos(w_c t + \theta_0)$$

where for phase modulation $\theta_o = k_p m(t)$

$$s_{PM}(t) = A_c cos(w_c t + k_p m(t))$$

Figure *Phase modulation*



Note

The total bandwidth required for PM can be determined from the bandwidth and maximum amplitude of the modulating signal:

$B_{PM} = 2(1 + \beta)B$.

Where $\beta = 2$ most often.

In previous slide, b = β

Comparing Frequency Modulation to Phase Modulation

No.	Frequency Modulation (FM)	Phase Modulation (PM)
1	Frequency deviation is proportional to modulating signal $m(t)$	Phase deviation is proportional to modulating signal $m(t)$
2	Noise immunity is superior to PM (and of course AM)	Noise immunity better than AM, but not FM
3	Signal-to-noise ratio (SNR) is better than PM (and of course AM)	Signal-to-noise ratio (SNR) is not quite as good as with FM
4	FM is widely used for commercial broadcast radio (88 MHz to 108 MHz)	PM is primarily used for mobile radio services
5	Modulation index is proportional to modulating signal $m(t)$ as well as the modulating frequency f_m	Modulation index is proportional to modulating signal $m(t)$

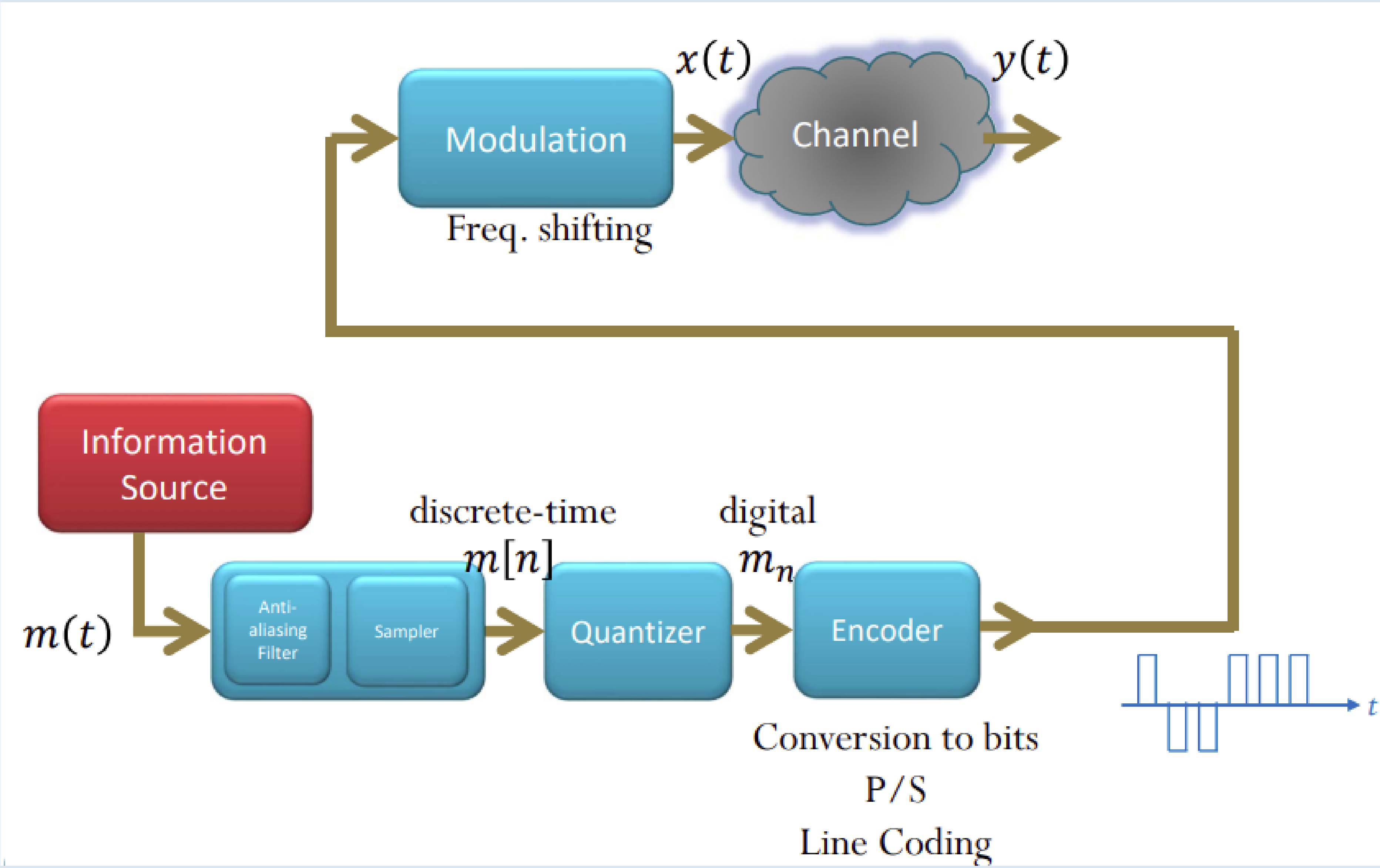
FM has superior noise immunity compared to AM

FM has better noise (or RFI) rejection than AM, as shown in this dramatic **New York publicity demonstration by General Electric in 1940**. The radio contained both AM and FM receivers. With a million-volt arc as a source of interference behind it, the AM receiver produced only a roar of static, while the FM receiver clearly reproduced a music program from Armstrong's experimental FM transmitter W2XMN in New Jersey.

https://en.wikipedia.org/wiki/Frequency_modulation

Note: **RFI** stands for **radio frequency interference.**

https://www.youtube.com/watch?v=cptiIDohFZc



$x(t)$
$y(t)$
Modulation
Channel
Freq. shifting
Information Source
discrete-time
digital
$m[n]$
m_n
$m(t)$
Anti-aliasing Filter
Sampler
Quantizer
Encoder
t
Conversion to bits
P/S
Line Coding

Analog-to-Digital Conversion

- Generally, analog signals are continuous in time and in range (amplitude); that is, they have values at every time instant, and their values can be anything within the range. On the other hand, digital signals exist only at discrete points of time, and their amplitude can take on only finitely (or countably) many values.
- Suppose we want to convey and analog message m(t) from a source to our destination. We now have many options.

(a) Use m(t) to modulate a carrier $A_c Cos(2\pi f_c t)$ using AM,FM,PM techniques studied earlier.

(b) Sample the continuous-time message m(t) to get a discrete-time message m[n]:

m(t)→Sampler→m[n]

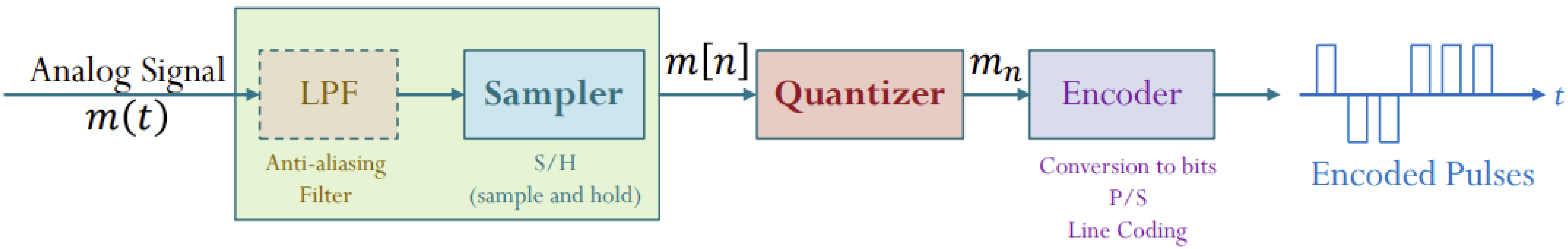
- May LPF(anti-aliasing filter) m(t) before sampling to eliminate aliasing (and reduce out-of-band).
- Need to make sure that sampling rate f_s is enough.

i) Send m[n] using analog pulse modulation techniques (PAM, PWM,PPM)

- Note that m[n] is a sequence of numbers.
- Information is transmitted basically in analog (not digital) form but the transmission takes place at discrete times.

(ii) **In digital pulse modulation**, m[n] is represented by a (discrete) number (symbol) selected from a finite alphabet set.

1) In **Pulse Code Modulation (PCM)**, we further quantize *m[n]* into m_n which has finitely many levels. Then, convert m_n into binary sequence represented by two basic pulses.



2) There are also other forms of "source coding" such as DPCM (Delta Pulse-Code Modulation) and DM (Delta Modulation).

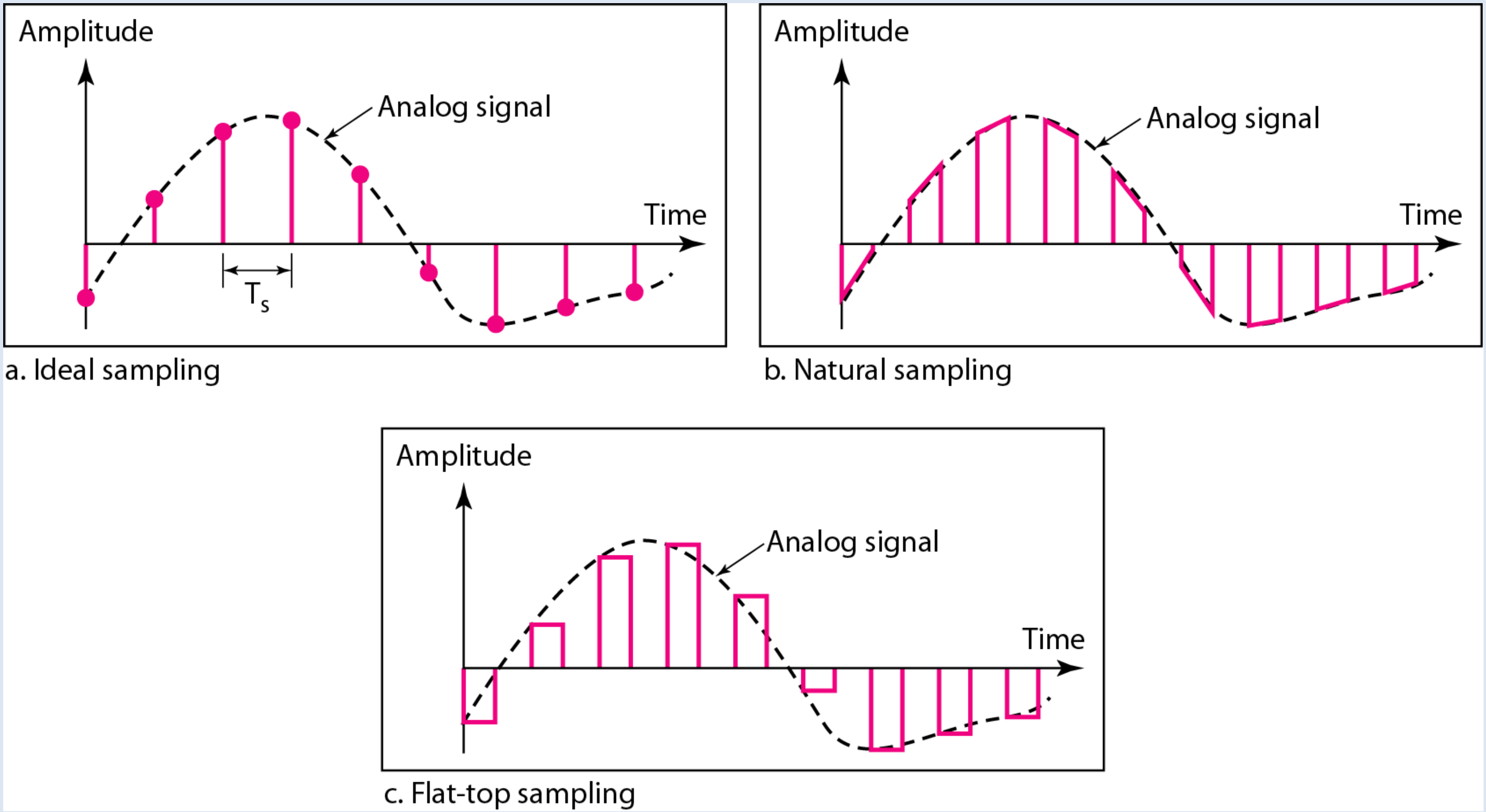
Pulse-Code Modulation (PCM)

- PCM is a discrete-time, discrete-amplitude waveform-coding process, by means of which an analog signal is directly represented by a sequence of coded pulses.
- PCM consists of three steps to digitize an analog signal:
  1. Sampling
  2. Quantization
  3. Binary encoding
- Before we sample, we have to filter the signal to limit the maximum frequency of the signal as it affects the sampling rate.
- Filtering should ensure that we do not distort the signal, ie remove high frequency components that affect the signal shape.

Sampling

- Analog signal is sampled every T_S secs.
- T_s is referred to as the sampling interval.
- $f_s = 1/T_s$ is called the sampling rate or sampling frequency.
- There are 3 sampling methods:
  - Ideal - an impulse at each sampling instant
  - Natural - a pulse of short width with varying amplitude
  - Flattop - sample and hold, like natural but with single amplitude value
- The process is referred to as pulse amplitude modulation PAM and the outcome is a signal with analog (non integer) values

Figure *Three different sampling methods for PCM*



According to the Nyquist theorem, the sampling rate must be at least 2 times the highest frequency contained in the signal.

Uniform Memoryless Quantization

Definition: Through quantization, each sample value m[n] is transformed to (e.g approximated, or "rounded off," to the nearest) quantized value.

Sampling vs. Quantization:

(a) Sampling operates in the time domain. Quantization operates in the amplitude domain.

(b) The sampling process is the link between an analog waveform and its discrete-time representation. The quantization process is the link between an analog waveform and its discrete-amplitude representation.

Suppose the range of the quantizer is $(-m_p, m_p)$.

• Note that, here, m_p is not necessarily the peak value of m(t). The amplitudes beyond $\pm m_p$ will be simply chopped off.

A simple (memoryless) quantizer partitions the range into **L intervals**. Each sample value (m[n]) is approximated by the midpoint (m_n) of the interval in which the sample value falls.

• Each sample is now represented by one of the L numbers.

• Such a signal is known as an L-ary digital signal.

• The length of each interval is denoted by $\Delta = 2m_p / L$.

• Because the quantized levels are uniformly spaced, we say that the quantizer is uniform.

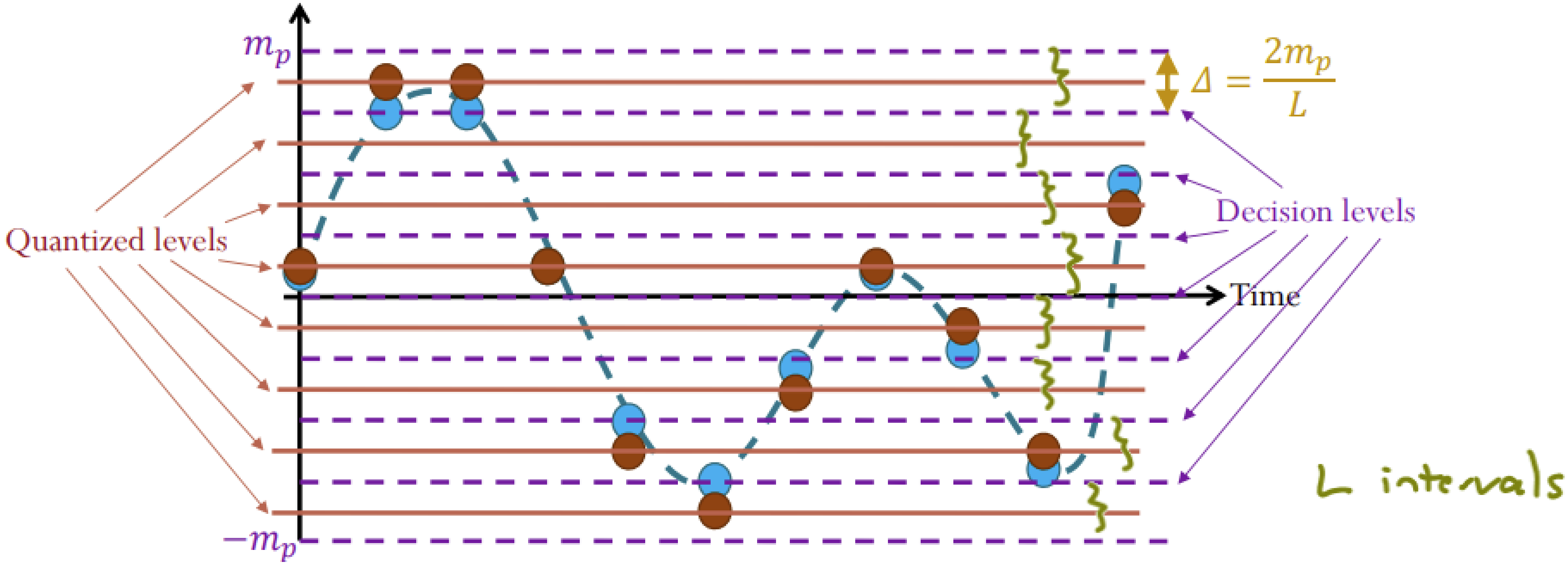


Figure : Quantized levels

In PCM, the quantized samples are coded and transmitted as binary pulses. At the receiver, some pulses may be detected incorrectly. Hence, there are two sources of error in this scheme: (a) quantization error (b) pulse detection error

In almost all practical schemes, the pulse detection error is quite small compared to the quantization error and can be ignored

The quantization error can be reduced as much as desired by increasing the number of quantizing levels, the price of which is paid in an increased required transmission bandwidth

Advantages of PCM

(a) Robustness to channel noise, distortion, and interference

- assuming these corruptions are within limits.
- With analog messages, on the other hand, any distortion or noise, no matter how small, will distort the received signal.

(b) Efficient regeneration of the coded signal along the transmission path by using regenerative **repeaters**.

- For analog communications,
  - A message signal becomes progressively weaker as it travels along the channel, whereas the cumulative channel noise and the signal distortion grow progressively stronger.
  - Ultimately the signal is overwhelmed by noise and distortion.
  - Amplification offers little help because it enhances the signal and the noise by the same proportion.
  - Consequently, the distance over which an analog message can be transmitted is limited by the initial transmission power.

Advantages of PCM

- For digital communications,
  - We can set up repeater stations along the transmission path at distances short enough to be able to detect signal pulses before the noise and distortion have a chance to accumulate sufficiently.
  - At each repeater station the pulses are detected, and new, clean pulses are transmitted to the next repeater station, which, in turn, duplicates the same process.

(c) Digital signals can be **coded** to remove redundancy, protect against channel corruption, and provide privacy.

Application

PCM has emerged as the most favored scheme for the digital transmission of analog information-bearing signals (e.g., voice and video signals) – The method of choice for the construction of public switched telephone networks (PSTNs).

Delta Modulation (DM)

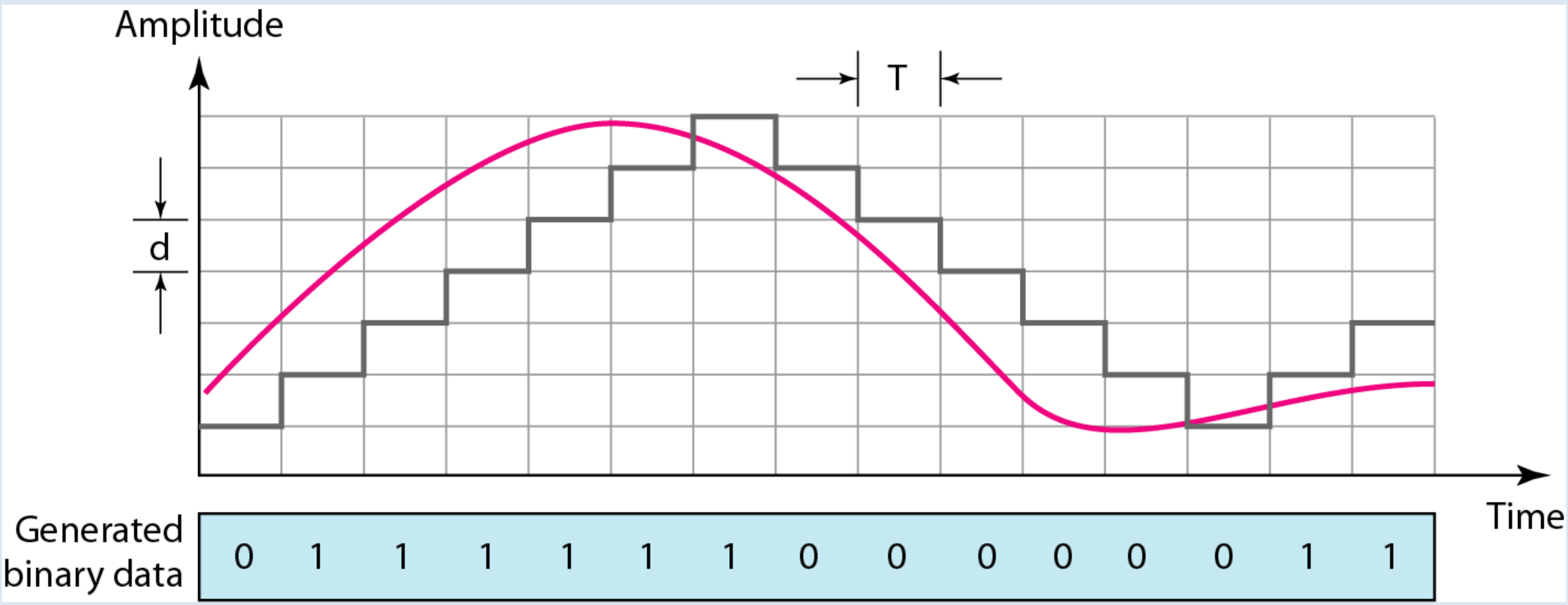
- Delta Modulation is developed to reduce the complexity of PCM.
- In PCM, we transmit all the bits that are used to code a sample hence require large bandwidth, to overcome this problem DM is used.
- Delta modulation transmits only one bit per sample. Here, the present sample value is compared with the previous sample and this result whether the amplitude is increased or decreased is transmitted.
- Input signal x(t) is approximated to step signal $\hat{x}(t)$ by the delta modulator and step sized is kept fixed.
- Difference between the input signal x(t) and staircase approximated signal $\hat{x}(t)$ is confined to two levels, i.e., +δ and −δ

$$x(t) > \hat{x}(t) \rightarrow \text{Increase } \hat{x}(t) \text{ by } \delta$$

$$x(t) < \hat{x}(t) \rightarrow \text{Decrease } \hat{x}(t) \text{ by } \delta$$

- If the δ is positive, the process records a 1; if it is negative, the process records a 0

Figure *The process of delta modulation*



Digital-To-Analog Conversion

- Digital-to-analog conversion is the process of changing one of the characteristics of an analog signal based on the information in digital data.
- Digital data needs to be carried on an analog signal.
- A carrier signal (frequency f_c) performs the function of transporting the digital data in an analog waveform.
- The analog carrier signal is manipulated to uniquely identify the digital data being carried.

Figure *Digital-to-analog conversion*

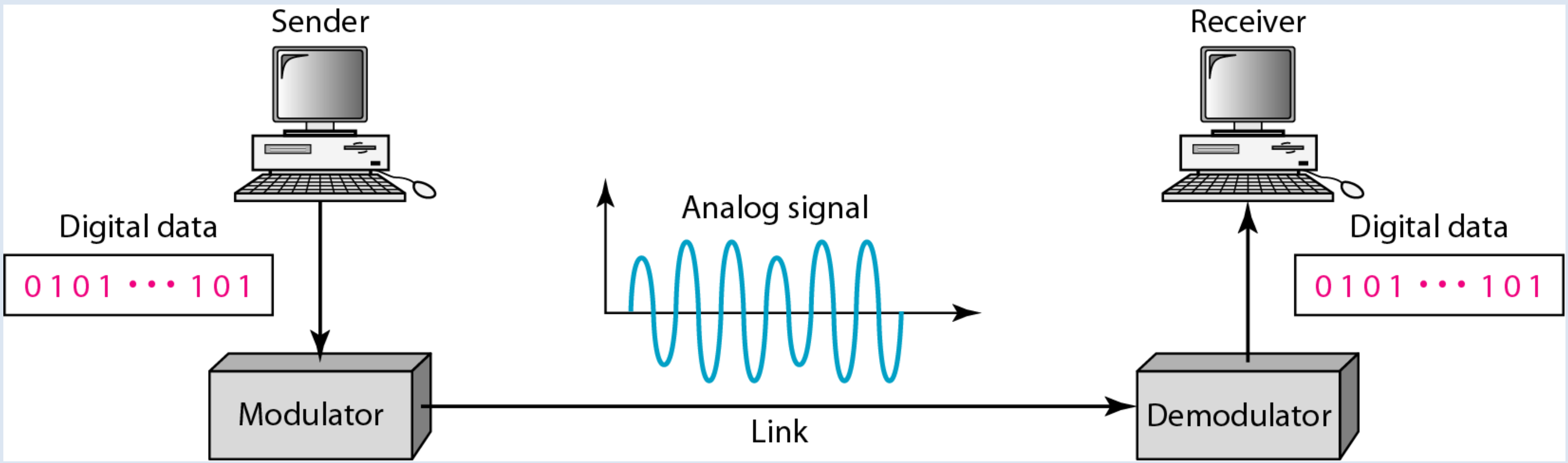
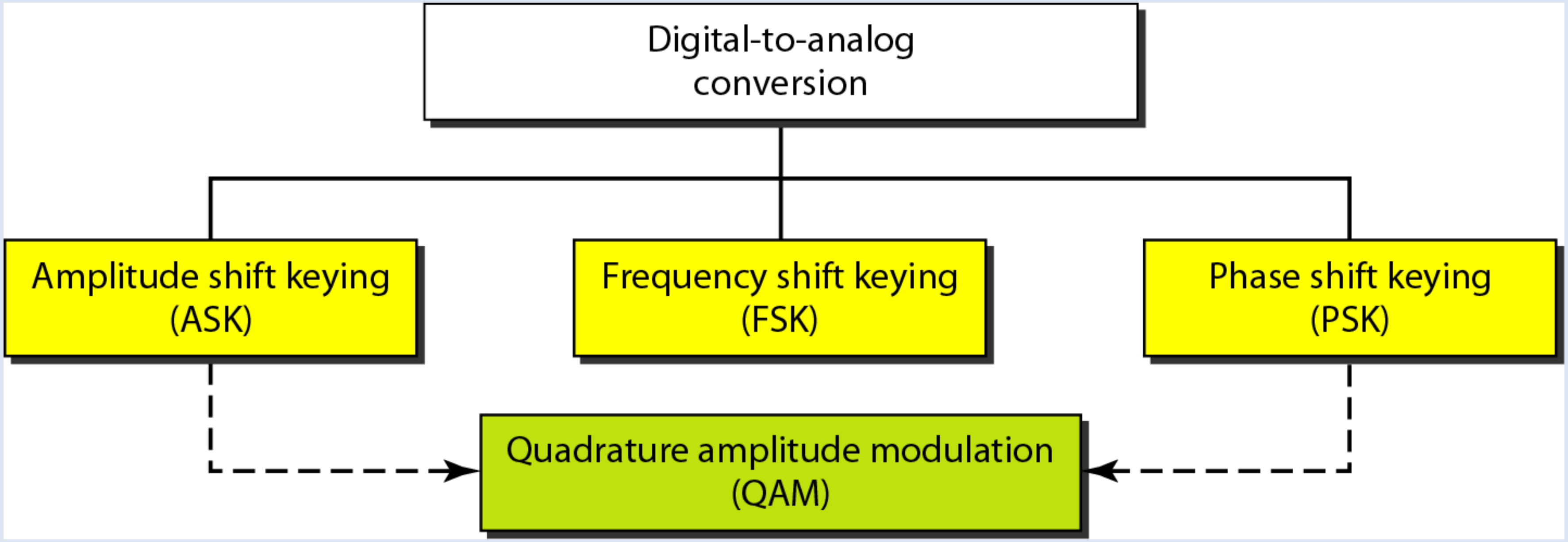


Figure *Types of digital-to-analog conversion*



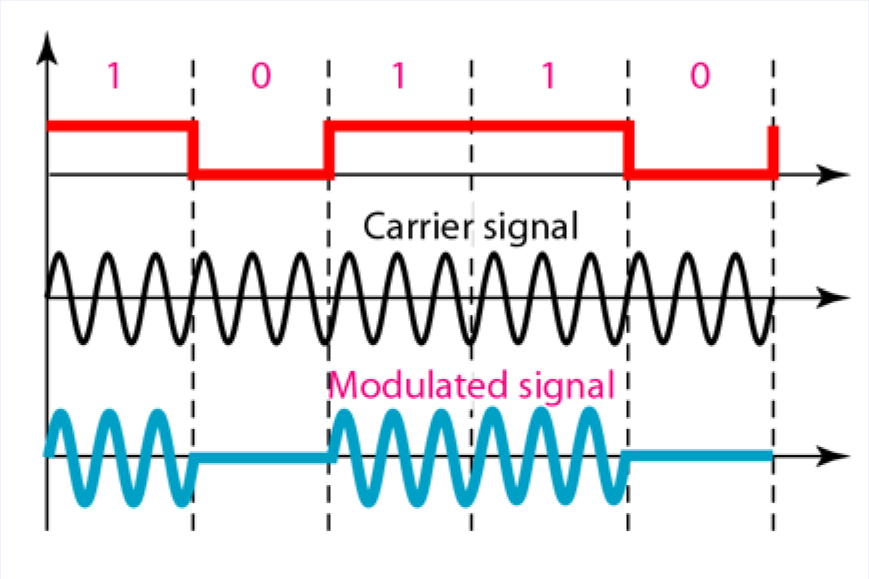
Amplitude Shift Keying (ASK)

- ASK is implemented by changing the amplitude of a carrier signal to reflect amplitude levels in the digital signal.
- Two binary values are represented by two different amplitudes of the carrier frequency. This is referred to as binary amplitude shift keying or on-off keying (OOK).
- The peak amplitude of one signal level is 0; the other is the same as the amplitude of the carrier frequency.
- The modulated signal is

$$s(t) = \begin{cases} A_c Sin(2\pi f_c t) & Symbol \text{ "1"} \\ 0 & Symbol \text{ "0"} \end{cases}$$

- The demodulator for an ASK system needs only to determine the presence or absence of a sinusoid in a given time interval

Amplitude Shift Keying (ASK)



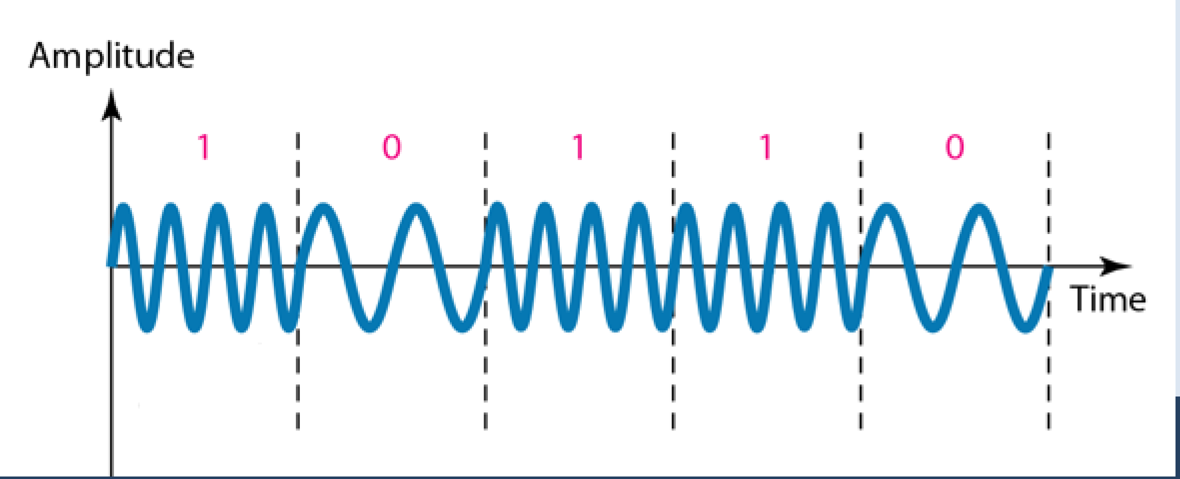
- ASK is susceptible to sudden gain changes and is rather inefficient modulation technique.

Frequency Shift Keying (FSK)

- The frequency of the carrier signal is varied to represent data.
- If the information bit is a 0, the sinusoid has frequency $f_1 = f_c - \epsilon$,low frequency (expand in time) and if it is a 1, the sinusoid has a frequency $f_2 = f_c + \epsilon$, high frequency (compress in time)
- Carrier signal : c(t) = $A_c Cos(2\pi f_c t)$
- The modulated signal :

$$s(t) = \begin{cases} A_c Cos(2\pi f_2 t) & Symbol \text{ "1"} \\ A_c Cos(2\pi f_1 t) & Symbol \text{ "0"} \end{cases}$$

- The demodulator for an FSK system must be able to determine which of two possible frequencies is present at a given time.



Phase Shift Keying (PSK)

- The phase of the carrier is varied to represent two or more different signal elements.
- Both peak amplitude and frequency remain constant as the phase changes.
- Today, PSK is more common than ASK or FSK.

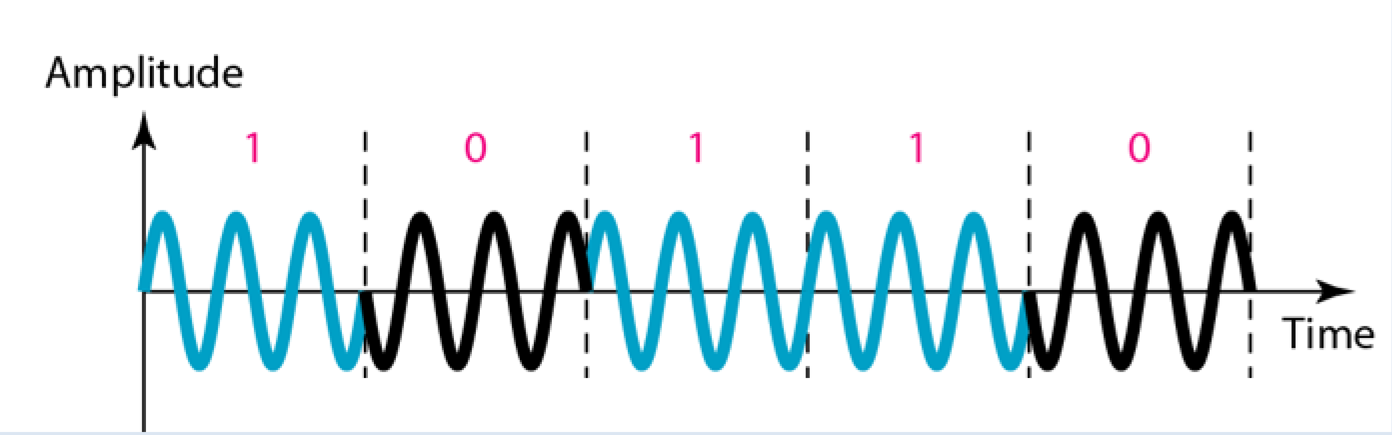
Binary PSK (BPSK)

- Carrier signal : c(t) = $A_c Cos(2\pi f_c t)$
- The modulated signal is

$$s(t) = \begin{cases} A_c Cos(2\pi f_c t) \quad Symbol\ "1" \\ A_c Cos(2\pi f_c t + \pi) = -A_c Cos(2\pi f_c t)\ Symbol\ "0" \end{cases}$$

Binary Phase Shift Keying (BPSK)

- This PSK scheme is equivalent to multiplying the sinusoid signal by +1 when the information is a 1 and by – 1 when the information bit is a 0.
- Thus, the demodulator for a PSK system must be able to determine the phase of the received sinusoid with respect to some reference phase.



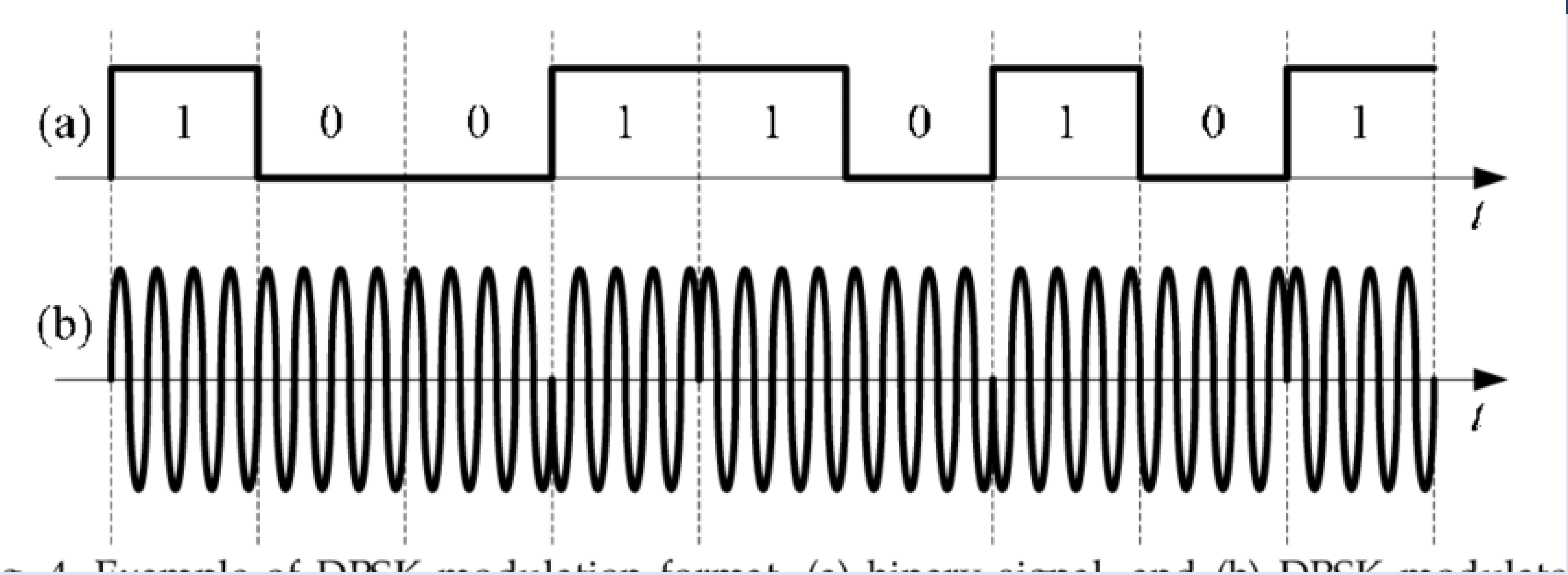
Binary PSK (BPSK)

- Binary PSK is as simple as binary ASK with one big advantage – it is less susceptible to noise. In ASK, the criterion for bit detection is the amplitude of the signal; in PSK, it is the phase. Noise can change the amplitude easier than it can change the phase.
- PSK is superior to FSK because we do not need two carrier signals.
- However, PSK needs more sophisticated hardware to be able to distinguish between phases.
- The bandwidth is the same as that for binary ASK, but less than that for BFSK. No bandwidth is wasted for separating two carrier signals.

Differential Phase Shift Keying DPSK

- Differential PSK is a noncoherent form of phase shift keying which avoids the need for a coherent reference signal at the receiver.
- Noncoherent receivers are easy and cheap to build, and hence widely used in wireless communications.
- In DPSK systems, the input binary sequence is first differentially encoded (NRZ-I) and then modulated using a BPSK modulator.
- While DPSK signaling has the advantage of reduced receiver complexity, its energy efficiency is inferior to that of coherent PSK by about 3 dB.

DPSK



It is seen from the above figure that, if the data bit is Low i.e., 0, then the phase of the signal is not reversed, but continued as it was. If the data is a High i.e., 1, then the phase of the signal is reversed, as with NRZI, invert on 1 a form of differential encoding.

Multilevel Modulation

- Multi-level digital modulation schemes are used to transmit M possible signal during each signal interval.
- These are usually used for bandwidth saving and data rate increasing.
- Ordinary modulation techniques such as ASK, FSK and PSK are 2-level modulation schemes. Only 1-bit can be transmitted in one cycle or symbol.

2 level modulation ----- 2 = 2^1 ----- **1** bit per symbol is transmitted
4 level modulation ----- 4 = 2^2 ----- **2** bits per symbol is transmitted
8 level modulation ----- 8 = 2^3 ----- **3** bits per symbol is transmitted etc.

4- Multilevel Modulation

Consider a system which two successive binary pulses are combined and the resultant set of four binary pairs. 00, 01, 10, 11 are used to trigger a high frequency sine wave of four possible phases one for each of the binary pairs. These bits then are grouped and considered as symbols. These will be four possible signals.

Types of M-ary:

1- M-ary ASK (MASK)

2- M-ary FSK (MFSK)

3- M-ary PSK (MPSK) (Quadrature PSK (QPSK)- Covered in the lecture)

Quadrature PSK (QPSK)

- To increase the bit rate, we can code 2 bits onto one signal element.
- Multilevel Modulation
- Variation of BPSK- it uses two separate BPSK modulations; one is in-phase, the other quadrature (out-of-phase).
- Phase modulation method using four phases.
- QPSK allows the signal to carry twice as much information as ordinary PSK using the same bandwidth.
- QPSK is used for satellite transmission of MPEG2 video, cable modems, video conferencing, cellular phone systems, and other forms of digital communication over an RF carrier.

QPSK

The signal of QPSK can be represented as:

$$s(t) = \sqrt{\frac{2E_s}{T_s}} Cos\left(2\pi f_c t + \frac{\pi(2i-1)}{4}\right), 0 \le t \le T_s, i = 1,2,3,4$$

Assume $\emptyset = \frac{\pi(2i-1)}{4}$

where E_s is the average energy per symbol, f_c is the frequency of carrier wave Furthermore, expanding $s(t)$,

$$s(t) = \sqrt{E_s}\sqrt{\frac{2}{T_s}} Cos(2\pi f_c t + \ \emptyset)$$

$$= \sqrt{E_s}\sqrt{\frac{2}{T_s}} \cos(2\pi f_c t)\ cos\emptyset\ - \sqrt{E_s}\sqrt{\frac{2}{T_s}} \sin(2\pi f_c t)\ sin\emptyset$$

After simplifying, $\varphi_1 = \sqrt{\frac{2}{T_s}} \cos(2\pi f_c t), 0 \le t \le T_s$

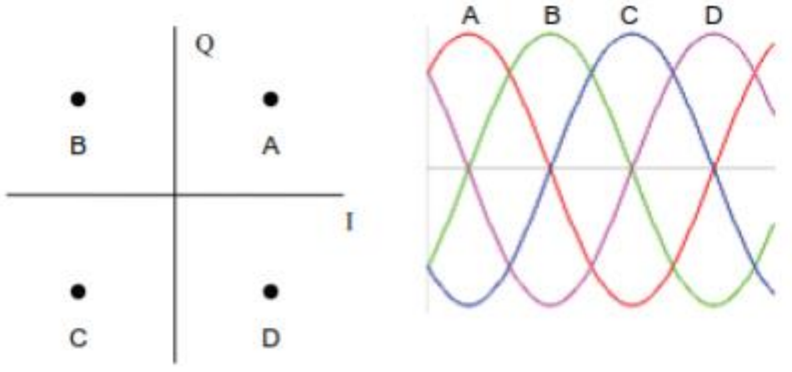
$$\varphi_2 = \sqrt{\frac{2}{T_s}} Sin(2\pi f_c t)\ ,0 \le t \le T_s$$

QPSK

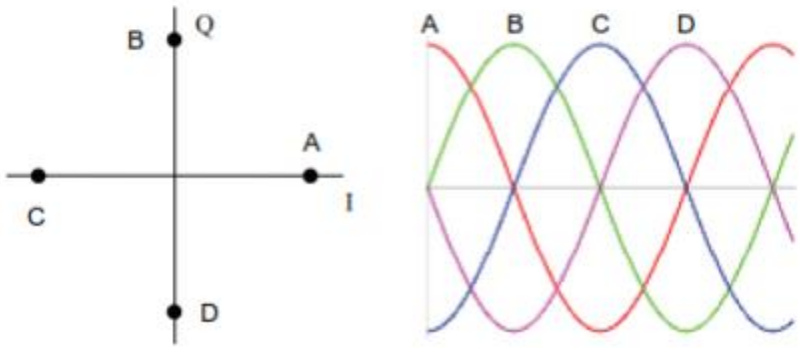
$$s(t) = \sqrt{E_s} cos\emptyset \, \varphi_1(t) - \sqrt{E_s} sin\emptyset \varphi_2(t), 0 \le t \le T_s, i = 1,2,3,4$$

Based on this representation, a QPSK signal can be depicted using a two dimensional constellation

diagram with four points.



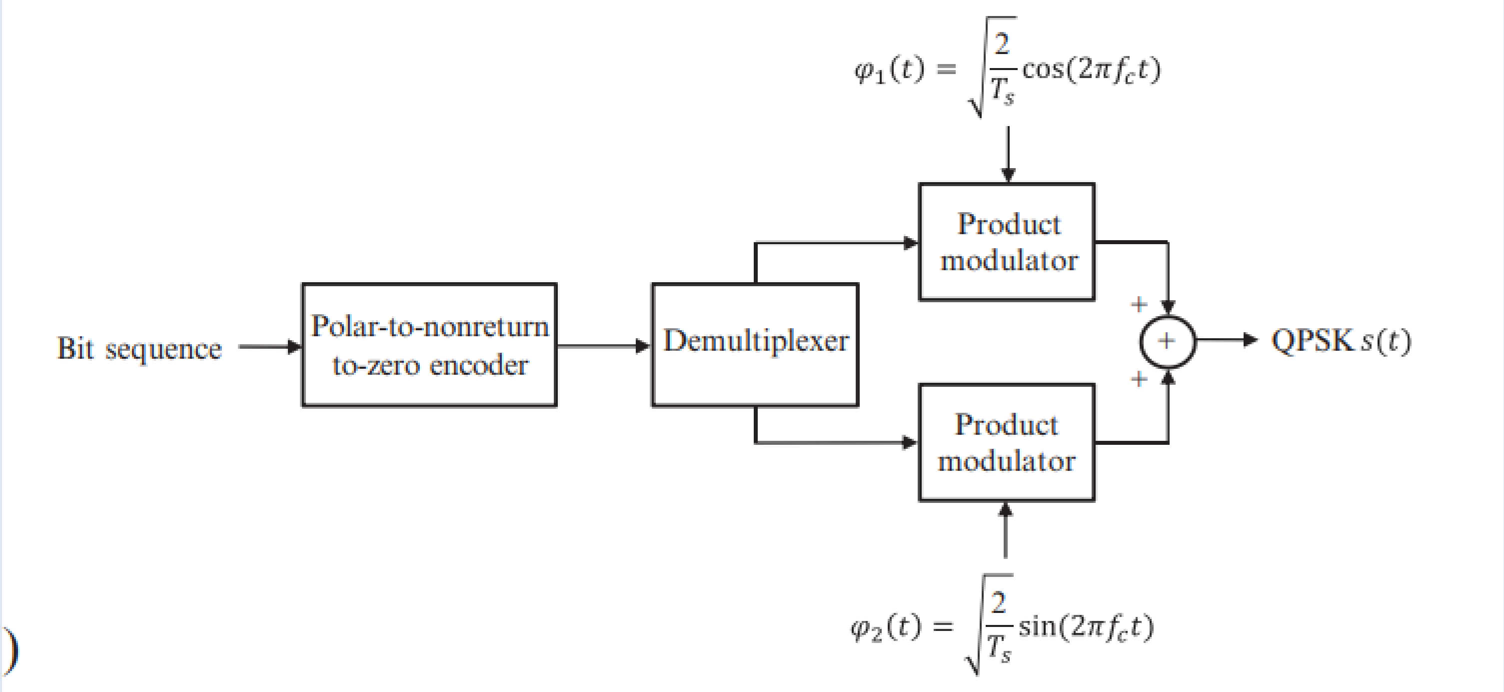
a) QPSK ($\varphi = \pi/4, 3\pi/4, 5\pi/4, 7\pi/4$)



b) QPSK ($\varphi = 0, \pi/2, \pi, 3\pi/2$)

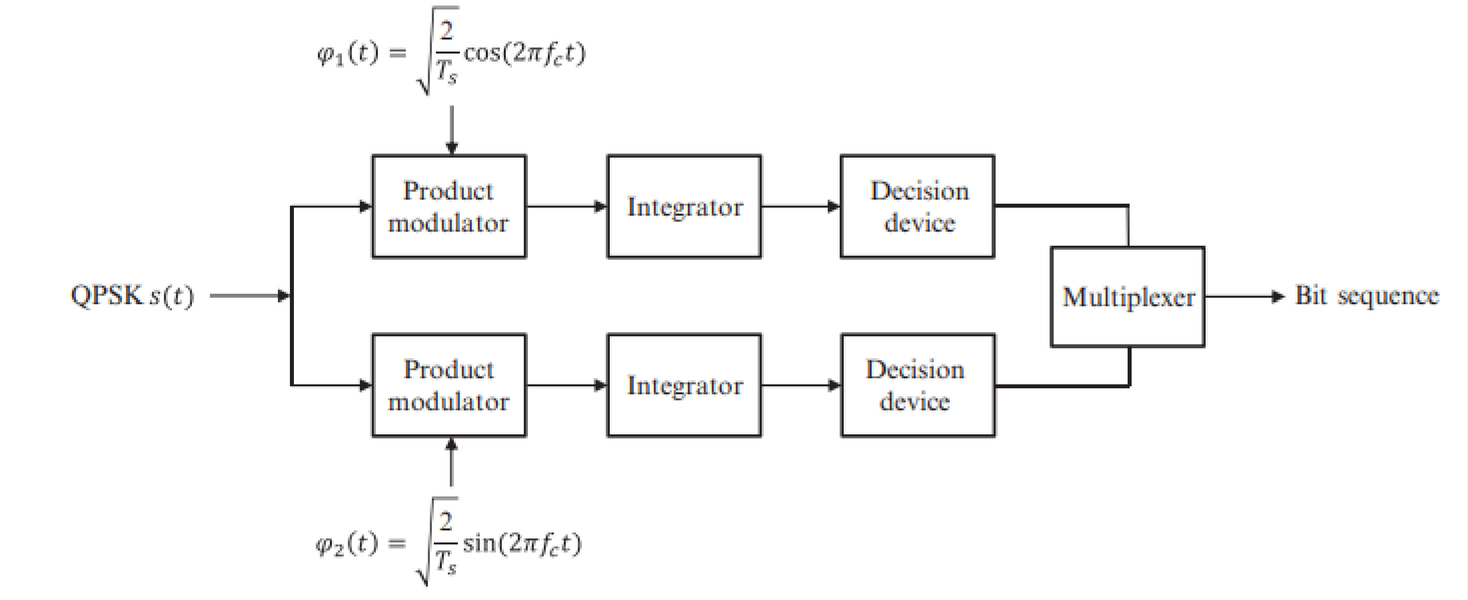
QPSK Transmitter

The bit sequence is first converted into a polar non-return-to-zero NRZ sequence. The bit stream m(t) is then split into two bit streams $m_I(t)$ and $m_Q(t)$ (in-phase and quadrature components).The bit stream $m_I(t)$ is called the “even” stream and $m_Q(t)$ is called the “odd” stream. The two binary sequences are separately modulated by two carriers $\varphi_1(t)$ and $\varphi_2(t)$, which are in quadrature. The two modulated signals, each of which can be considered to be a BPSK signal, are summed to produce a QPSK signal.



QPSK Receiver

The s(t) is split into two parts, and each part is coherently demodulated using the in-phase and quadrature carriers. The outputs of the demodulators are passed through decision circuits which generate the in-phase and quadrature binary streams. The two components are then multiplexed to reproduce the original binary sequence.



Modem and Modem Standards

- Unlike during the last decade of the twentieth century home users today tend to take the functioning of their modem (now DSL or cable modem instead of voice modem for a *land-line* phone) for granted because they are not as involved in configuration issues. The information below was important to CS 240 students in the 1990's and has been left for the perspective it can provide.
- The "Comite Consultatif Internationale de Telegraphie et Telephonie", or CCITT, has established a number standards, including the "V series" for telephone modems. It is a part of the United Nations International Telecommunications Union (ITU). Below we highlight some aspects of the V series standards as applied to modems.
- Recall analog transmission is called *broadband* transmission, particularly when it uses frequency division multiplexing. People are generally able to hear from a frequency of about 20 Hz (recall Hz, or Hertz, refers to cycles per second) to a frequency of about 14,000 Hz. However, a voice grade telephone circuit has a bandwidth from 0 to 4000 Hz (so a nominal bandwidth of 4000 Hz). Moreover, from 0 to 300 Hz at the bottom and 3300 to 4000 Hz at the top are designated as *guard bands*, dropping the bandwidth to just 3000 Hz. These guard bands become important when signals are multiplexed, helping prevent interference. This explains why most speech is adequately transmitted over the telephone, but it cannot convey the high notes of music.

Modem and Modem Standards

- Modems use some combination of *amplitude modulation (AM), frequency modulation (FM)* , and/or *phase modulation (PM)* of a carrier wave to transmit data. The number of times per second that the signal changes is measured as *baud*, so this is a unit of signaling rate. Now one can send bits per signal if there are ^{k} different ways to modulate a signal in a particular modulation scheme, and hence to create a bits per second (bps) that is times the baud. Be fore we chart the V series standard we list two popular modulation schemes.
- *QAM:* Quadrature Amplitude Modulation employs 8 phases and 2 amplitudes to get 16 values, and hence 4 bits per signal.
- *TCM:* Trellis Coded Modulation also uses both PM and AM to generate either 5, 6, 7, or 8 bits per signal. The 6-bit TCM was the most common form when this method was developed.

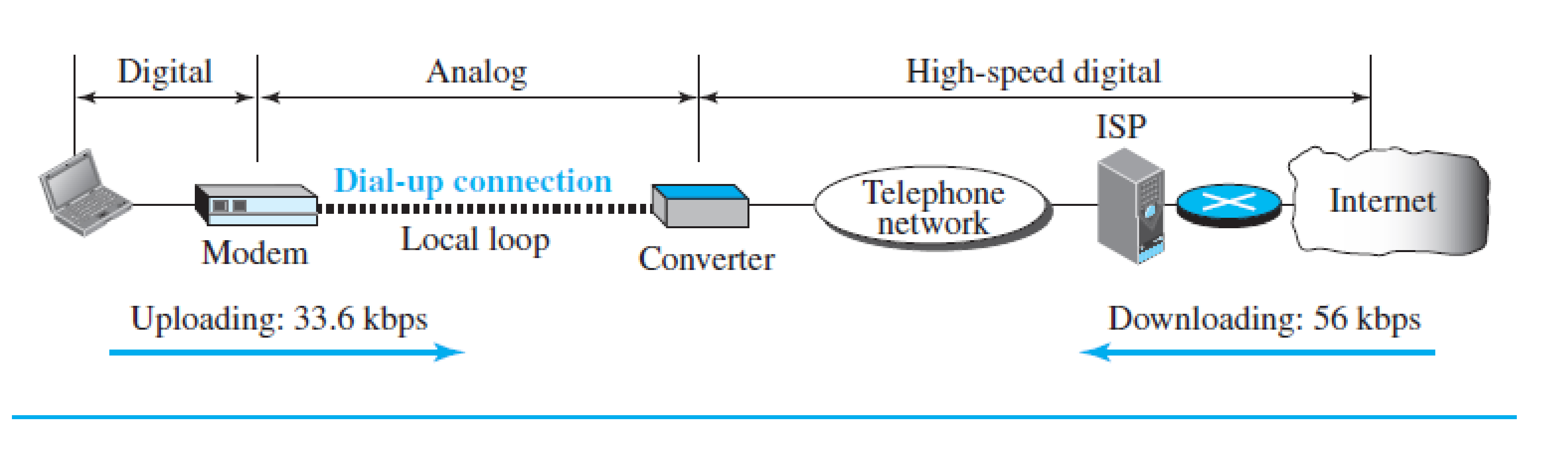
V.22 and V.22bis	These delt with 1200 and 2400 baud full-duplex modems, using frequency modulation to achieve 1 bit per signal (hence 1200 and 2400 bps). These standards choose to stay well below the maximum 3000 baud of a standard voice grade circuit in order to reduce the impact of noise.
V.32	This uses 2400 baud with 4-bit QAM to achieve 9600 bps transmission.
V.32bis	This uses 2400 baud with 6-bit TCM to achieve 14.4 Kbps transmission.
V.34	This is one of the more involved standards in that it allows for the use of 2700, 3000, 3300 or 3600 baud and has over 50 valid baud/modulation schemes. It is thus quite likely that a V.34 modem will be able to negotiate a fairly good rate over even quite noisy wires. The 3600 baud combined with an 8-bit TCM yield 28.8 Kbps.
V.34bis	Here you can get 33.6 Kbps via 4200 baud and 8-bit TCM. This rate may have trouble holding since the 4200 baud could be at the upper limits of the capacity of residential wires (it certainly exceeds the "conventional" maximum 3000 baud).
V.42	This is NOT a standard about baud rates and modulation, but rather deals with error-checking standards.
V.42bis	This is NOT a standard about baud rates and modulation, but rather deals with data compression.
V.90	This is the standard for so called 56Kbps modems that resolved issues arising from the different methods employed on Rockwell and Motorola chipsets (so called X2 and 56Kflex). These are asymmetrical devices, only able to approach their nominal speeds in receive mode, while transmitting at significantly lower rates.

The IEEE 802 committee set up a subcommittee (802.14) to develop cable modem standards, but it disbanded when an industry consortium came out with the DOCSIS open standard. A modern cable modem is an integrated device that actually functions across all seven layers of the OSI reference model, with major emphasis on layers 1 and 2 (and most of the higher-level functions controlled by the cable company, not the client). A cable modem is a network bridge, and hence is subject to the 802.1D standard. Most cable modems are now connected to Hybrid Fiber Cable (HFC) networks owned by the cable provider. Cable networks are full duplex, but have much greater downstream bandwidth than upstream bandwidth. Before the advent of DOCSIS some early systems used POTS (plain old telephone system) for the upstream communication now carried on the cable itself. There aren't the same kind of issues with collisions on these network segments as occur on classic ethernet segments.

modem (= MOdulator-DEModulator) converts digital data to analog form for transmission, then converts it back to digital form when received.

56K Modems

- Traditional modems have a data rate limitation of 33.6 kbps, as determined by the Shannon capacity. However, modern modems with a bit rate of 56,000 bps are available; these are called ***56K modems.***
- These modems may be used only if one party is using digital signaling (such as through an Internet provider). They are asymmetric in that the downloading rate (flow of data from the Internet service provider to the PC) is a maximum of 56 kbps, while the uploading rate (flow of data from the PC to the Internet provider) can be a maximum of 33.6 kbps.
- Do these modems violate the Shannon capacity principle? No, in the downstream direction, the SNR ratio is higher because there is no quantization error.



- In **uploading,** the analog signal must still be sampled at the switching station. In this direction, quantization noise is introduced into the signal, which reduces the SNR ratio and limits the rate to 33.6 kbps.

- However, there is no sampling in the **downloading.** The signal is not affected by quantization noise and not subject to the Shannon capacity limitation. The maximum data rate in the uploading direction is still 33.6 kbps, but the data rate in the downloading direction is now 56 kbps.

- One may wonder how we arrive at the 56-kbps figure. The telephone companies sample 8000 times per second with 8 bits per sample. One of the bits in each sample is used for control purposes, which means each sample is 7 bits. The rate is therefore 8000 · 7, or 56,000 bps or 56 kbps.

Quadrature Amplitude Modulation (QAM)

- Presentation by Arpan Adhikari in Class