Module 02

LINE CODING

- Line-coding is the process of converting digital-data to digital-signals (Figure 4.1).
- The data may be in the form of text, numbers, graphical images, audio, or video
- The data are stored in computer memory as sequences of bits (0s or 1s).
- Line-coding converts a sequence of bits to a digital-signal.
- At the sender, digital-data is encoded into a digital-signal.
  At the receiver, digital-signal is decoded into a digital-data.

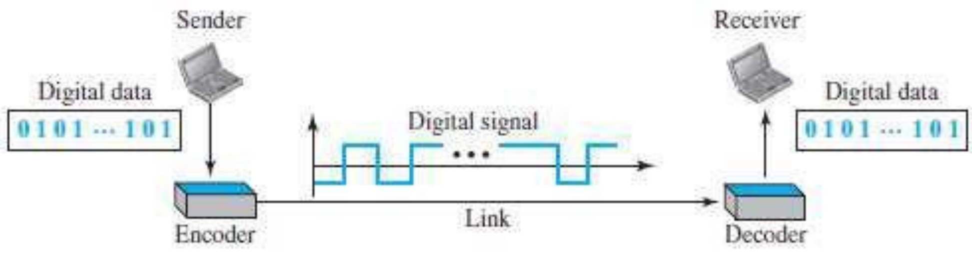


Figure 4.1 *Line coding and decoding*

Characteristics

- Different characteristics of digital signal are
  1) Signal Element Vs Data Element
  2) Data Rate Vs Signal Rate
  3) Bandwidth
  4) Baseline Wandering
  5) DC Components
  6) Built-in Error Detection
  7) Self-synchronization
  8) Immunity to Noise and Interference
  9) Complexity

1) Data Element vs. Signal Element

Data Element	Signal Element
A data-element is the smallest entity that can represent a piece of information (Figure 4.2).	A signal-element is shortest unit (timewise) of a digital-signal.
A data-element is the bit.	A signal-element carries data-elements.
Data-elements are being carried.	Signal-elements are the carriers.

➢ Ratio r is defined as number of data-elements carried by each signal-element.

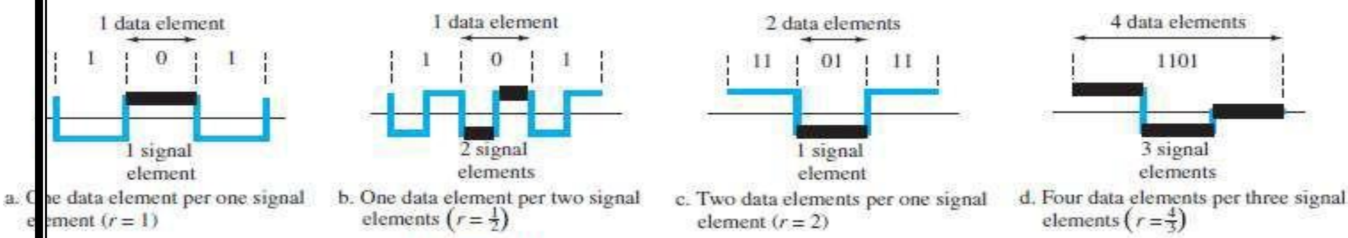


Figure 4.2 *Signal element versus data element*

2) Data Rate vs. Signal Rate

Data Rate	Signal Rate
The data-rate defines the number of data-elements (bits) sent in 1 sec.	The signal-rate is the number of signal-elements sent in 1 sec.
The unit is bits per second (bps).	The unit is the baud.

The data-rate is sometimes called the bit-rate.	The signal-rate is sometimes called the pulse rate, the modulation rate, or the baud rate
Goal in data-communications: increase the data-rate.	Goal in data-communications: decrease the signal-rate.
Increasing the data-rate increases the speed of transmission.	Decreasing the signal-rate decreases the bandwidth requirement.

- The relationship between data-rate and signal-rate is given by

$$S_{ave} = c \times N \times (1/r) \quad \text{baud}$$

where N = data-rate (in bps)
c = case factor, which varies for each case S = number of signal-elements and
r = previously defined factor.

- This relationship depends on
  → value of r.
  → data pattern.

(If we have a data pattern of all 1s or all 0s, the signal-rate may be different from a data pattern of alternating Os and 1s).

Bandwidth

- Digital signal that carries information is non-periodic.
- The bandwidth of a non-periodic signal is continuous with an infinite range.
- However, most digital-signals we encounter in real life have a bandwidth with finite values.
- The effective bandwidth is finite.
- The baud rate, not the bit-rate, determines the required bandwidth for a digital-signal.
- More changes in the signal mean injecting more frequencies into the signal.
  (Frequency means change and change means frequency.)
- Relationship b/w baud rate (signal-rate) and the bandwidth (range of frequencies) is given as

$$B_{min} = c \times N \times (1/r)$$

where N = data-rate (in bps)
c = case factor, which varies for each case r = previously defined factor
B_{min} = minimum bandwidth

Baseline Wandering

- While decoding, the receiver calculates a running-average of the received signal-power. This average is called the baseline
- The incoming signal-power is estimated against this baseline to determine the value of the data-element.
- A long string of 0s or 1s can cause a drift in the baseline (baseline wandering).
  Thus, make it difficult for the receiver to decode correctly.
- A good line-coding scheme needs to prevent baseline wandering.

DC Components

- When the voltage-level in a digital-signal is constant for a while, the spectrum creates very low frequencies.
- These frequencies around zero are called DC (direct-current) components.
- DC components present problems for a system that cannot pass low frequencies.
- For example: Telephone line cannot pass frequencies below 200 Hz.
- For Telephone systems, we need a scheme with no DC component.

*

Built-in Error Detection

- Built-in error-detecting capability has to be provided to detect the errors that occurred during transmission.

Immunity to Noise & Interference

- The code should be immune to noise and other interferences.

Complexity

- ➢ A complex scheme is costlier to implement than a simple one.
- ➢ For ex: A scheme that uses 4 signal-levels is more difficult to interpret than one that uses only 2 levels.

Self-Synchronization

- ➢ To correctly interpret the signals received from the sender, the receiver's bit intervals must correspond exactly to the sender's bit intervals.
- ➢ If the receiver clock is faster or slower, the bit intervals are not matched and the receiver might misinterpret the signals.

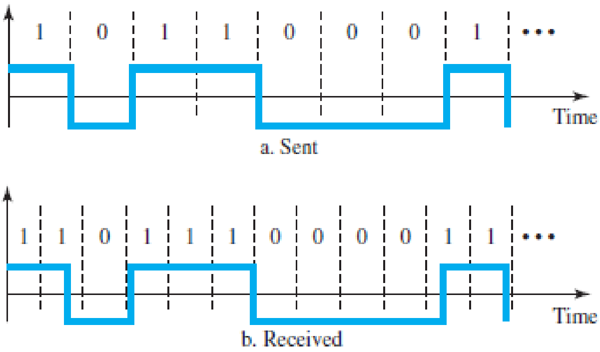


Figure 4.3 *Effect of lack of synchronization*

- ➢ As shown in figure 4.3, we have a situation where the receiver has shorter bit duration.
- ➢ The sender sends 10110001, while the receiver receives 110111000011.
- ➢ A self-synchronizing digital-signal includes timing-information in the data being transmitted.
  - ¤ This can be achieved if there are transitions in the signal that alert the receiver to the beginning, middle, or end of the pulse.
  - ¤ If the receiver's clock is out-of-synchronization, these points can reset the clock.

LINE CODING SCHEMES

- The Line Coding schemes are classified into 3 broad categories (Figure 4.4):

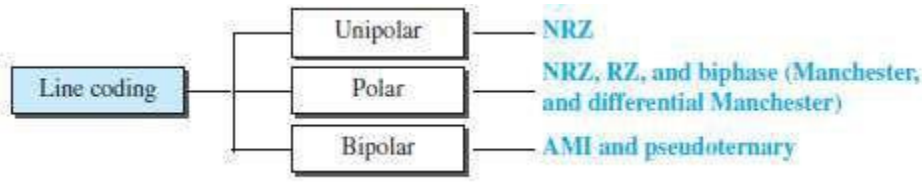


Figure 4.4 *Line coding schemes*

Unipolar Scheme

- All signal levels are either above or below the time axis.

NRZ (Non-Return-to-Zero)

- ➢ The positive voltage defines bit 1 and the zero voltage defines bit 0 (Figure 4.5).
- ➢ It is called NRZ because the signal does not return to 0 at the middle of the bit.

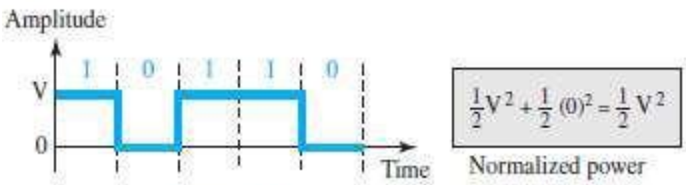


Figure 4.5 *Unipolar NRZ scheme*

- ➢ Disadvantages:
  1) Compared to polar scheme, this scheme is very costly.

2) Also, the normalized power is double that for polar NRZ.
3) Not suitable for transmission over channels with poor performance around zero frequency.
(Normalized power → power needed to send 1 bit per unit line resistance)

Polar Schemes

- The voltages are on the both sides of the time axis.
- Polar NRZ scheme can be implemented with two voltages (V).

For example: -V for bit 1
+V for bit 0.

a) Non-Return-to-Zero (NRZ)

- We use 2 levels of voltage amplitude.
- Two versions of polar NRZ (Figure 4.6):

i) NRZ-L (NRZ-Level)

¤ The level of the voltage determines the value of the bit.
¤ For example: i) Voltage-level for 0 can be positive and
ii) Voltage-level for 1 can be negative.

ii) NRZ-I (NRZ-Invert)

¤ The change or lack of change in the level of the voltage determines the value of the bit.
¤ If there is no change, the bit is 0;
If there is a change, the bit is 1.

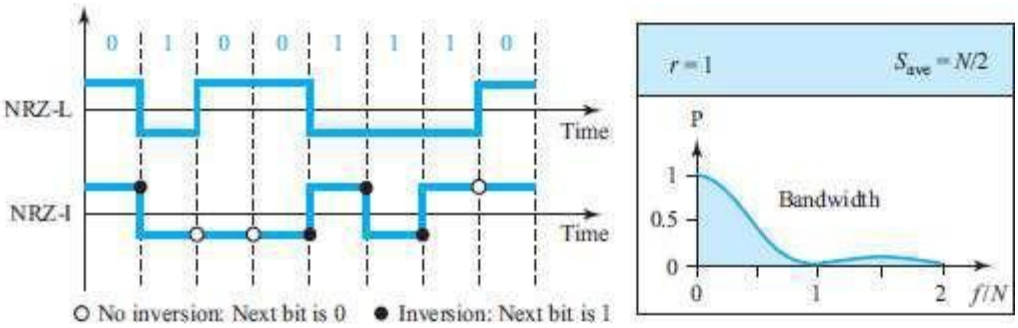


Figure 4.6 *Polar NRZ-L and NRZ-I schemes*

- Disadvantages:

1) **Baseline wandering** is a problem for both variations (NRZ-L NRZ-I).
i) In NRZ-L, if there is a long sequence of 0s or 1s, the average signal-power becomes skewed.
The receiver might have difficulty discerning the bit value.
ii) In NRZ-I, this problem occurs only for a long sequence of 0s.
If we eliminate the long sequence of 0s, we can avoid baseline wandering.
2) The **synchronization problem** also exists in both schemes.
→ A long sequence of 0s can cause a problem in both schemes.
→ A long sequence of 1s can cause a problem in only NRZ-L.
3) In NRZ-L, problem occurs when there is a sudden **change of polarity** in the system.
¤ For example:
In twisted-pair cable, a change in the polarity of the wire results in
→ all 0s interpreted as 1s and
→ all 1s interpreted as 0s.
¤ NRZ-I does not have this problem.
¤ Both schemes have an average signal-rate of N/2 Bd.

NRZ-L and NRZ-I both have a **DC component problem**

b) Return-to-Zero (RZ)

- In NRZ encoding, problem occurs when the sender-clock and receiver-clock are not synchronized.
- Solution: Use return-to-zero (RZ) scheme (Figure 4.7).

➢ RZ scheme uses 3 voltages: positive, negative, and zero.
➢ There is always a transition at the middle of the bit. Either
 i) from high to zero (for 1) or
 ii) from low to zero (for 0)

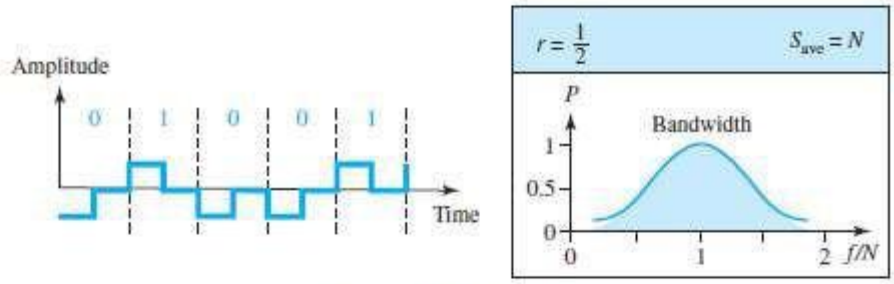


Figure 4.7 Polar RZ scheme

➢ Disadvantages:
 1) RZ encoding requires 2 signal-changes to encode a bit & .'. occupies greater bandwidth.
 2) Complexity: RZ uses 3 levels of voltage, which is more complex to create and detect.
 3) Problem occurs when there is a sudden change of polarity in the system. This result in
 → all 0s interpreted as 1s &
 → all 1s interpreted as 0s.

c) Biphase: Manchester & Differential Manchester

i) Manchester Encoding

➢ This is a combination of NRZ-L & RZ schemes (RZ→transition at the middle of the bit).
➢ There is always a transition at the middle of the bit. Either
 i) from high to low (for 0) or
 ii) from low to high (for 1).
➢ It uses only two voltage levels (Figure 4.8).
➢ The duration of the bit is divided into 2 halves.
➢ The voltage
 → remains at one level during the first half &
 → moves to the other level in the second half.
➢ The transition at the middle of the bit provides synchronization.

ii) Differential Manchester

➢ This is a combination of NRZ-I and RZ schemes.
➢ There is always a transition at the middle of the bit, but the bit-values are determined at the beginning of the bit.
➢ If the next bit is 0, there is a transition. If the next bit is 1, there is none.

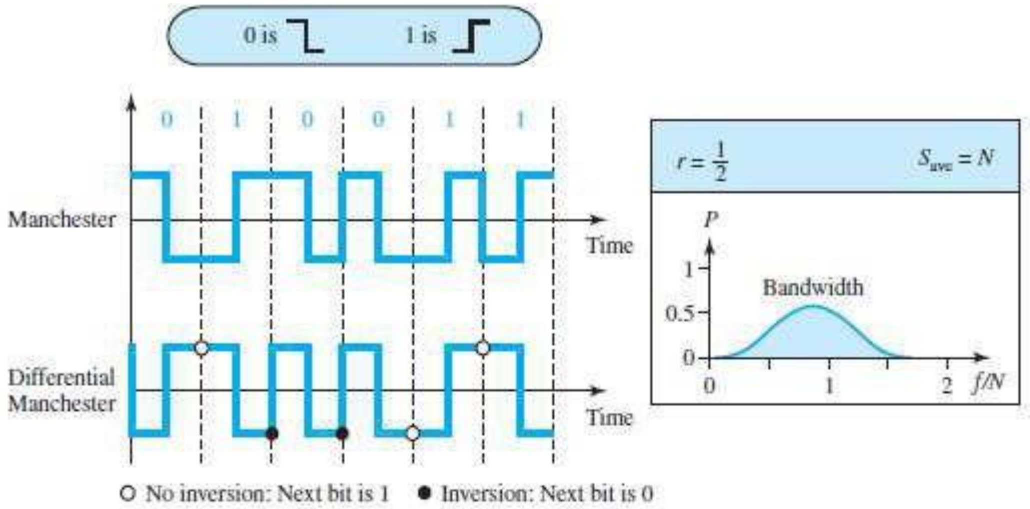


Figure 4.8 *Polar biphase: Manchester and differential Manchester schemes*

- Advantages:
  1) The Manchester scheme overcomes problems associated with NRZ-L.
     Differential Manchester overcomes problems associated with NRZ-I.
  2) There is no baseline wandering.
  3) There is no DC component '.' each bit has a positive & negative voltage contribution.
- Disadvantage:
  1) Signal-rate: Signal-rate for Manchester & diff. Manchester is double that for NRZ.

Bipolar Schemes (or Multilevel Binary)

- This coding scheme uses 3 voltage levels (Figure 4.9):
  - i) positive
  - ii) negative &
  - iii) zero.
- Two variations of bipolar encoding:
  - i) AMI (Alternate Mark Inversion)
  - ii) Pseudoternary

i) AMI

- Binary 0 is represented by a neutral 0 voltage (AMI → Alternate 1 Inversion).
- Binary 1s are represented by alternating positive and negative voltages.

ii) Pseudoternary

- Binary 1 is represented by a neutral 0 voltage.
- Binary 0s are represented by alternating positive and negative voltages.

- Advantages:
  1) The bipolar scheme has the same signal-rate as NRZ.
  2) There is no DC component '.' each bit has a positive & negative voltage contribution.
  3) The concentration of the energy is around frequency N/2.
- Disadvantage:
  1) AMI has a synchronization problem when a long sequence of 0s is present in the data.

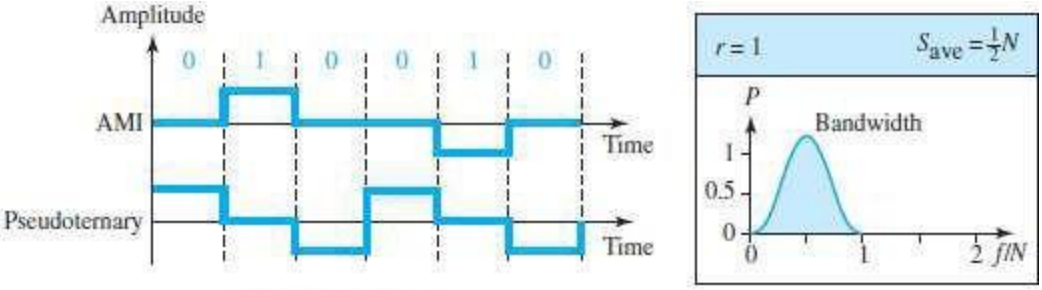


Figure 4.9 *Bipolar schemes: AMI and pseudoternary*

ANALOG TO DIGITAL CONVERSION

- An analog-signal may created by a microphone or camera.
- To change an analog-signal to digital-data, we use PCM (pulse code modulation).
- After the digital-data are created (digitization), then we convert the digital-data to a digital-signal.

PCM

- PCM is a technique used to change an analog signal to digital data (digitization).
- PCM has encoder at the sender and decoder at the receiver.
- The encoder has 3 processes (Figure 4.21):

1) Sampling
2) Quantization &
3) Encoding.

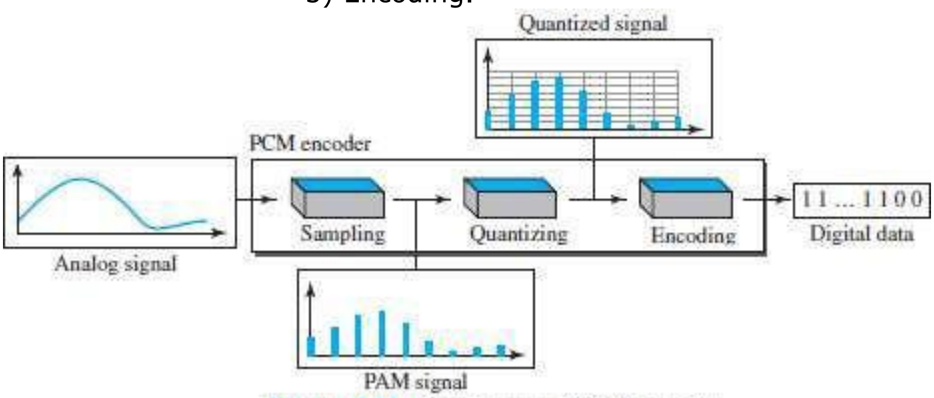


Figure 4.21 *Components of PCM encoder*

Sampling

- We convert the continuous time signal (analog) into the discrete time signal (digital).
- Pulses from the analog-signal are sampled every T_s sec where T_s is the sample-interval or period.
- The inverse of the sampling-interval is called the sampling-frequency (or sampling-rate).
- Sampling-frequency is given by

$$f_s = 1/T_s.$$

Three sampling methods

1) Ideal Sampling
- ➢ This method is difficult to implement.

2) Natural Sampling
- ➢ A high-speed switch is turned ON for only the small period of time when the sampling occurs.
- ➢ The result is a sequence of samples that retains the shape of the analog-signal.

3) Flat Top Sampling
- ➢ The most common sampling method is sample and hold.
- ➢ Sample and hold method creates flat-top samples.
- ➢ This method is sometimes referred to as *PAM* (pulse amplitude modulation).

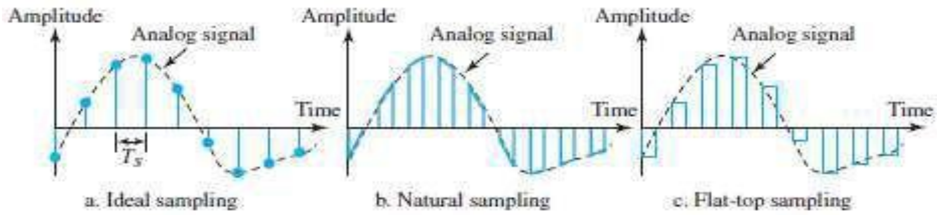


Figure 4.22 *Three different sampling methods for PCM*

Sampling Rate

- According to Nyquist theorem,

  "The sampling-rate must be at least 2 times the highest frequency, not the bandwidth".

  i) If the analog-signal is **low-pass**, the bandwidth and the highest frequency are the same value (Figure 4.23a).

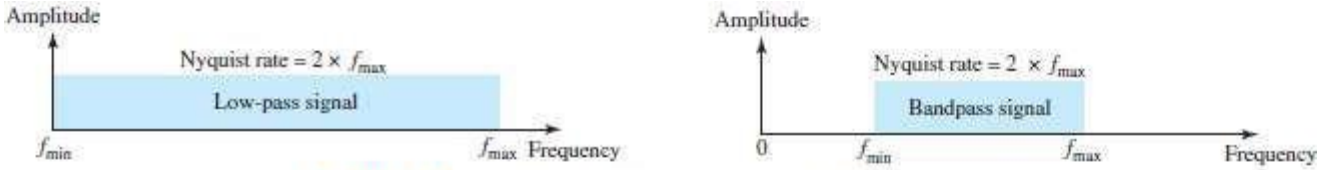


Figure 4.23 *Nyquist sampling rate for low-pass and bandpass signals*

  ii) If the analog-signal is **bandpass**, the bandwidth value is lower than the value of the maximum frequency (Figure 4.23b).

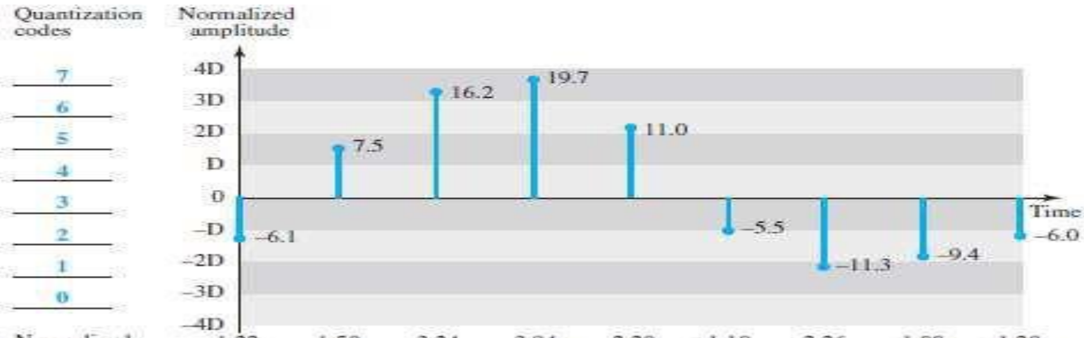
Quantization

- The sampled-signal is quantized.
- Result of sampling is a set of pulses with amplitude-values b/w max & min amplitudes of the signal.
- Four steps in quantization:
  1) We assume that the original analog-signal has amplitudes between V_{min} & V_{max}.
  2) We divide the range into L zones, each of height Δ(delta).

$$\Delta = \frac{V_{max} - V_{min}}{L}$$

where L = number of levels.

  3) We assign quantized values of 0 to (L-1) to the midpoint of each zone.
  4) We approximate the value of the sample amplitude to the quantized values.
- For example: Let V_{min}=-20 V_{max} =+20 V L = 8 Therefore, Δ = [+20-(-20)]/8= 5V
- In the chart (Figure 4.26),
  1) First row is normalized-PAM-value for each sample.
  2) Second row is normalized-quantized-value for each sample.
  3) Third row is normalized error (which is diff. b/w normalized PAM value & quantized values).
  4) Fourth row is quantization code for each sample.
  5) Fifth row is the encoded words (which are the final products of the conversion).



Normalized PAM values	–1.22	1.50	3.24	3.94	2.20	–1.10	–2.26	–1.88	–1.20
Normalized quantized values	–1.50	1.50	3.50	3.50	2.50	–1.50	–2.50	–1.50	–1.50
Normalized error	–0.38	0	+0.26	–0.44	+0.30	–0.40	–0.24	+0.38	–0.30
Quantization code	2	5	7	7	6	2	1	2	2
Encoded words	010	101	111	111	110	010	001	010	010

Figure 4.26 *Quantization and encoding of a sampled signal*

Quantization Level

- Let L = number of levels.
- The choice of L depends on
  → range of the amplitudes of the analog-signal and
  → how accurately we need to recover the signal.
- If the signal has only 2 amplitude values, we need only 2 quantization-levels.
  If the signal (like voice) has many amplitude values, we need more quantization-levels.
- In audio digitizing, L is normally chosen to be 256.
  In video digitizing, L is normally thousands.
- Choosing lower values of L increases the quantization-error.

Quantization Error

- Quantization-error is the difference b/w normalized PAM value & quantized values
- Quantization is an approximation process.
- The input values to the quantizer are the real values.
  The output values from the quantizer are the approximated values.
- The output values are chosen to be the middle value in the zone.
- If the input value is also at the middle of the zone,
  Then, there is no error.
  Otherwise, there is an error.
- In the previous example,
  The normalized amplitude of the third sample is 3.24, but the normalized quantized value is 3.50. This means that there is an error of +0.26.

Uniform vs. Non Uniform Quantization

- Non-uniform quantization can be done by using a process called companding and expanding.
  1) The signal is companded at the sender before conversion.
  2) The signal is expanded at the receiver after conversion.
- Companding means reducing the instantaneous voltage amplitude for large values.
  Expanding means increasing the instantaneous voltage amplitude for small values.
- It has been proved that non-uniform quantization effectively reduces the SNR_{dB} of quantization.

Encoding

- The quantized values are encoded as n-bit code word.
- In the previous example,
  A quantized value 2 is encoded as 010. A quantized value 5 is encoded as 101.
- Relationship between number of quantization-levels (L) & number of bits (n) is given by
  $n=\log_2 L$ or $2^n=L$
- The bit-rate is given by:

$$\text{Bit rate} = \text{sampling rate} \times \text{number of bits per sample} = f_s \times n$$

Original Signal Recovery

- PCM decoder is used for recovery of the original signal.
- Here is how it works (Figure 4.27):
  1) The decoder first uses circuitry to convert the code words into a pulse that holds the amplitude until the next pulse.
  2) Next, the staircase-signal is passed through a low-pass filter to smooth the staircase signal into an analog-signal.
- The filter has the same cut-off frequency as the original signal at the sender.
- If the signal is sampled at the Nyquist sampling-rate, then the original signal will be re-created.
- The maximum and minimum values of the original signal can be achieved by using amplification.

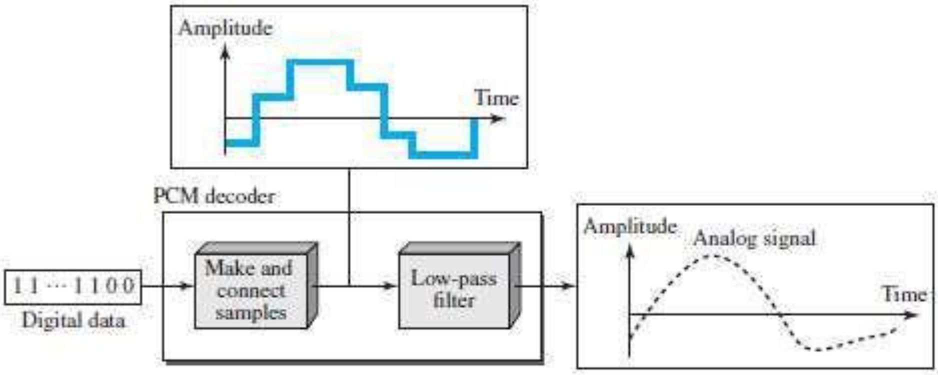


Figure 4.27 *Components of a PCM decoder*

PCM Bandwidth

• The minimum bandwidth of a line-encoded signal is

$$B_{min} = c \times N \times \frac{1}{r}$$

• We substitute the value of N in above formula:

$$B_{min} = c \times N \times \frac{1}{r} = c \times n_b \times f_s \times \frac{1}{r} = c \times n_b \times 2 \times B_{analog} \times \frac{1}{r}$$

• When 1/r = 1 (for a NRZ or bipolar signal) and c = (1/2) (the average situation), the minimum bandwidth is

$$B_{min} = n_b \times B_{analog}$$

• This means the minimum bandwidth of the digital-signal is n_b times greater than the bandwidth of the analog-signal.

Maximum Data Rate of a Channel

• The Nyquist theorem gives the data-rate of a channel as

$$N_{max} = 2 \times B \times \log_2 L$$

• We can deduce above data-rate from the Nyquist sampling theorem by using the following arguments.

3) We assume that the available channel is low-pass with bandwidth B.
4) We assume that the digital-signal we want to send has L levels, where each level is a signal-element. This means r = 1/log$_2$L.
5) We first pass digital-signal through a low-pass filter to cut off the frequencies above B Hz.
6) We treat the resulting signal as an analog-signal and sample it at 2 x B samples per second and quantize it using L levels.
7) The resulting bit-rate is

$$N = f_s \times n_b = 2 \times B \times \log_2 L$$

This is the maximum bandwidth; if the case factor c increases, the data-rate is reduced.

$$N_{max} = 2 \times B \times \log_2 L \quad \text{bps}$$

Minimum Required Bandwidth

• The previous argument can give us the minimum bandwidth if the data-rate and the number of signal-levels are fixed. We can say

$$B_{min} = \frac{N}{(2 \times \log_2) L} \quad \text{Hz}$$

TRANSMISSION MODES

• Two ways of transmitting data over a link (Figure 4.31): 1) Parallel mode & 2) Serial mode.

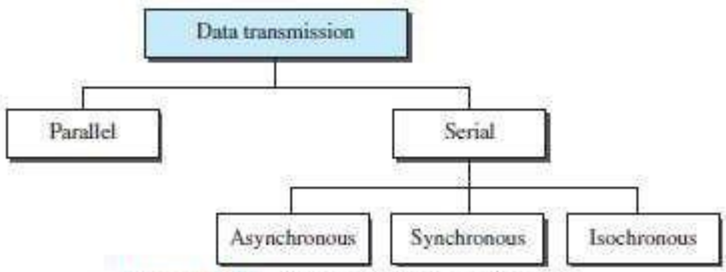


Figure 4.31 *Data transmission and modes*

PARALLEL TRANSMISSION

- Multiple bits are sent with each clock-tick (Figure 4.32).
- „n' bits in a group are sent simultaneously.
- „n' wires are used to send „n' bits at one time.
- Each bit has its own wire.
- Typically, the 8 wires are bundled in a cable with a connector at each end.

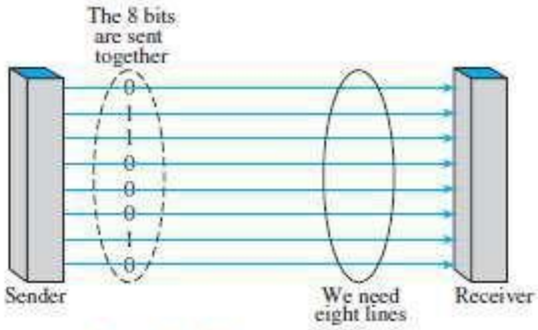


Figure 4.32 *Parallel transmission*

- Advantage:
  1) Speed: Parallel transmission can increase the transfer speed by a factor of n over serial transmission.
- Disadvantage:
  1) Cost: Parallel transmission requires n communication lines just to transmit the data-stream. Because this is expensive, parallel transmission is usually limited to short distances.

SERIAL TRANSMISSION

- One bit is sent with each clock-tick using only a single link (Figure 4.33).

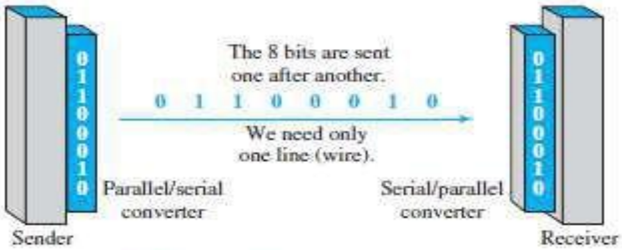


Figure 4.33 *Serial transmission*

- Advantage:
  1) Cost: Serial transmission reduces cost of transmission over parallel by a factor of n.
- Disadvantage:
  1) Since communication within devices is parallel, following 2 converters are required at interface:
     i) Parallel-to-serial converter
     ii) Serial-to-parallel converter
- Three types of serial transmission: asynchronous, synchronous, and isochronous.

Asynchronous Transmission

- Asynchronous transmission is so named because the timing of a signal is not important (Figure 4.34).

- Prior to data transfer, both sender & receiver agree on pattern of information to be exchanged.
- Normally, patterns are based on grouping the bit-stream into bytes.
- The sender transmits each group to the link without regard to a timer.
- As long as those patterns are followed, the receiver can retrieve the info. without regard to a timer.
- There may be a gap between bytes.
- We send
  → 1 start bit (0) at the beginning of each byte
  → 1 stop bit (1) at the end of each byte.
- Start bit alerts the receiver to the arrival of a new group.
  Stop bit lets the receiver know that the byte is finished.
- Here, the term asynchronous means "asynchronous at the byte level".
  However, the bits are still synchronized & bit-durations are the same.

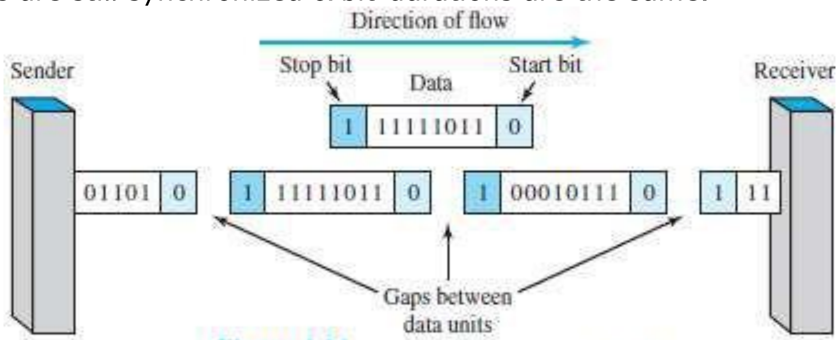


Figure 4.34 *Asynchronous transmission*

- Disadvantage:
  1) Slower than synchronous transmission. (Because of stop bit, start bit and gaps)
- Advantages:
  1) Cheap & effective.
  2) Useful for low-speed communication.

Synchronous Transmission

- We send bits one after another without start or stop bits or gaps (Figure 4.35).
- The receiver is responsible for grouping the bits.
- The bit-stream is combined into longer "frames," which may contain multiple bytes.
- If the sender wants to send data in separate bursts, the gaps b/w bursts must be filled with a special sequence of 0s & 1s (that means idle).

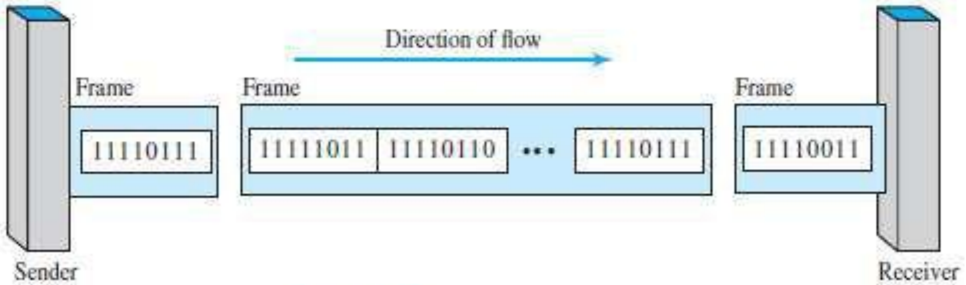


Figure 4.35 *Synchronous transmission*

- Advantages:
  1) Speed: Faster than asynchronous transmission. („" of no stop bit, start bit and gaps).
  2) Useful for high-speed applications such as transmission of data from one computer to another.

Isochronous

- Synchronization between characters is not enough; the entire stream of bits must be synchronized.
- The isochronous transmission guarantees that the data arrive at a fixed rate.
- In real-time audio/video, jitter is not acceptable. Therefore, synchronous transmission fails.
- For example: TV images are broadcast at the rate of 30 images per second. The images must be viewed at the same rate.

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 (Figure 5.1).

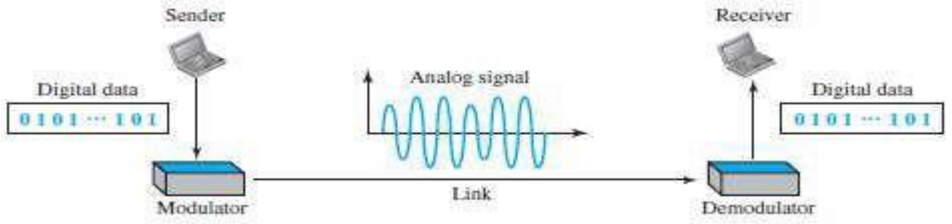


Figure 5.1 *Digital-to-analog conversion*

• A sine wave can be defined by 3 attributes:
1) Amplitude
2) Frequency &
3) Phase.

• When anyone of the 3 attributes of a wave is varied, a different version of the wave will be created.
• So, by changing one attribute of an analog signal, we can use it to represent digital-data.
• Four methods of digital to analog conversion (Figure 5.2):
1) Amplitude shift keying (ASK)
2) Frequency shift keying (FSK)
3) Phase shift keying (PSK)
4) Quadrature amplitude modulation (QAM).

• QAM is a combination of ASK and PSK i.e. QAM combines changing both the amplitude and phase.
QAM is the most efficient of these 4 methods.
QAM is the method commonly used today.

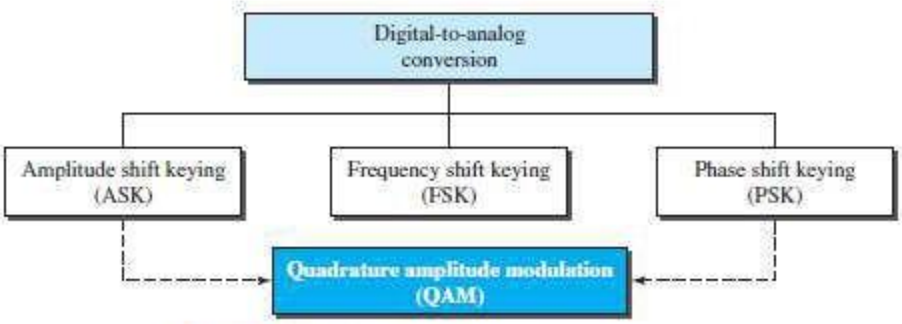


Figure 5.2 *Types of digital-to-analog conversion*

Aspects of Digital-to-Analog Conversion

1) Data Element vs. Signal Element

➢ A data-element is the smallest piece of information to be exchanged i.e. the bit.
➢ A signal-element is the smallest unit of a signal that is transmitted.

2) Data Rate vs. Signal Rate

➢ Data rate (Bit rate) is the number of bits per second.
Signal-rate (Baud rate) is the number of signal elements per second.
➢ The relationship between data-rate(N) and the signal-rate(S) is

$$S = N \times \frac{1}{r} \quad \text{baud}$$

where r = number of data-elements carried in one signal-element.

➢ The value of r is given by r = log2L or 2r=L
where L = type of signal-element (not the level)
(In transportation,
→ a baud is analogous to a vehicle, and

- → a bit is analogous to a passenger.
  We need to maximize the number of people per car to reduce the traffic).

3) Carrier Signal
- The sender produces a high-frequency signal that acts as a base for the information-signal.
- This base-signal is called the carrier-signal (or carrier-frequency).
- The receiver is tuned to the frequency of the carrier-signal that it expects from the sender.
- Then, digital-information changes the carrier-signal by modifying its attributes (amplitude, frequency, or phase). This kind of modification is called modulation (shift keying).

4) Bandwidth
- In both ASK & PSK, the bandwidth required for data transmission is proportional to the signal-rate.
- In FSK, the bandwidth required is the difference between the two carrier-frequencies.

Amplitude Shift Keying (ASK)
- The amplitude of the carrier-signal is varied to represent different signal-elements.
- Both frequency and phase remain constant for all signal-elements.

Binary ASK (BASK)
- BASK is implemented using only 2 levels. (Figure 5.3)
- This is also referred to as OOK (On-Off Keying).

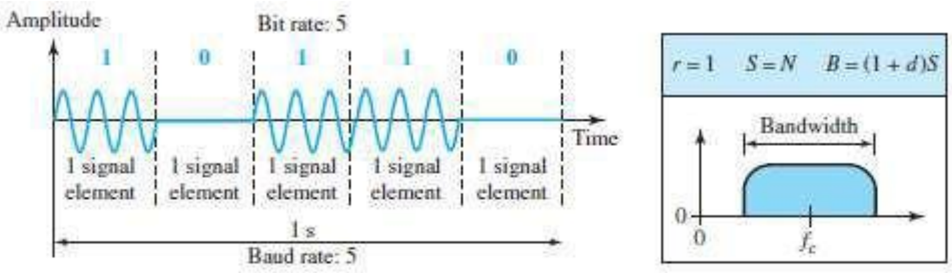


Figure 5.3 *Binary amplitude shift keying*

Implementation of BASK
- Here, line coding method used = unipolar NRZ (Figure 5.4).
- The unipolar NRZ signal is multiplied by the carrier-frequency coming from an oscillator.
  1) When amplitude of the NRZ signal = 0, amplitude of the carrier-signal = 0.
  2) When amplitude of the NRZ signal = 1, the amplitude of the carrier-signal is held.

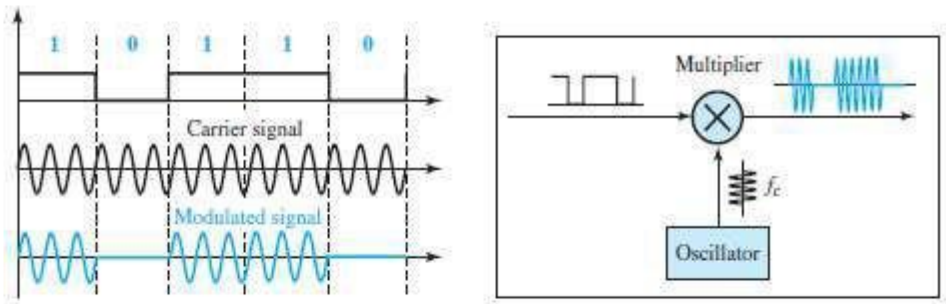


Figure 5.4 *Implementation of binary ASK*

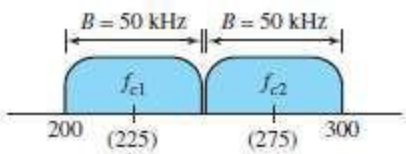


Figure 5.5 *Bandwidth of full-duplex ASK*

Bandwidth for ASK

- Here, the bandwidth (B) is proportional to the signal-rate (S) (Figure 5.5)
- The bandwidth is given by

$$B = (1 + d) \times S$$

Frequency Shift Keying (FSK)

- The frequency of the carrier-signal is varied to represent different signal-elements.
- The frequency of the modulated-signal is constant for the duration of one signal-element, but changes for the next signal-element if the data-element changes.
- Both amplitude and phase remain constant for all signal-elements.

Binary FSK (BFSK)

- This uses 2 carrier-frequencies: f1 and f2. (Figure 5.6)
  1) When data-element = 1, first carrier frequency(f1) is used.
  2) When data-element = 0, second carrier frequency(f2) is used.

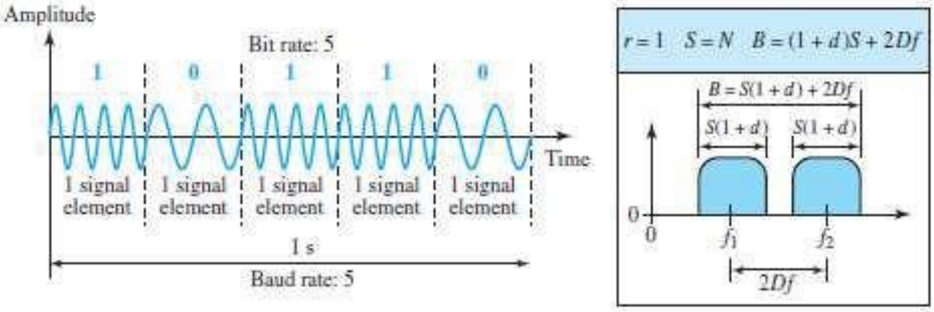


Figure 5.6 *Binary frequency shift keying*

Implementation

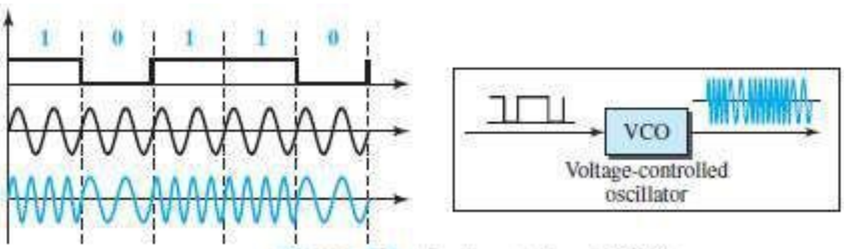


Figure 5.7 *Implementation of BFSK*

Bandwidth for BFSK

- FSK has two ASK signals, each with its own carrier-frequency f1 or f2. (Figure 5.6)
- The bandwidth is given by

$B = (1 + d) \times S + 2\Delta f$ where 2Δf is the difference between f1 and f2,

Phase Shift Keying (PSK)

- The phase of the carrier-signal is varied to represent different signal-elements.
- Both amplitude and frequency remain constant for all signal-elements.

Binary PSK (BPSK)

- We have only two signal-elements:
  1) First signal-element with a phase of 0°.
  2) Second signal-element with a phase of 180° (Figure 5.9).
- ASK vs. PSK
  - In ASK, the criterion for bit detection is the amplitude of the signal.
  - In PSK, the criterion for bit detection is the phase.
- Advantages:
  1) PSK is less susceptible to noise than ASK.

2) PSK is superior to FSK because we do not need 2 carrier-frequencies.

• Disadvantage:

1) PSK is limited by the ability of the equipment to distinguish small differences in phase.

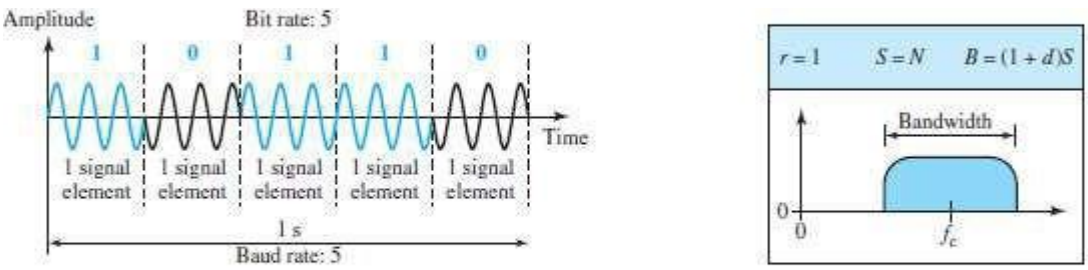


Figure 5.9 *Binary phase shift keying*

Implementation

• The implementation of BPSK is as simple as that for ASK. (Figure 5.10).

• The signal-element with phase 180° can be seen as the complement of the signal-element with phase 0°.

• Here, line coding method used: polar NRZ.

• The polar NRZ signal is multiplied by the carrier-frequency coming from an oscillator.

1) When data-element = 1, the phase starts at 0°.

2) When data-element = 0, the phase starts at 180°.

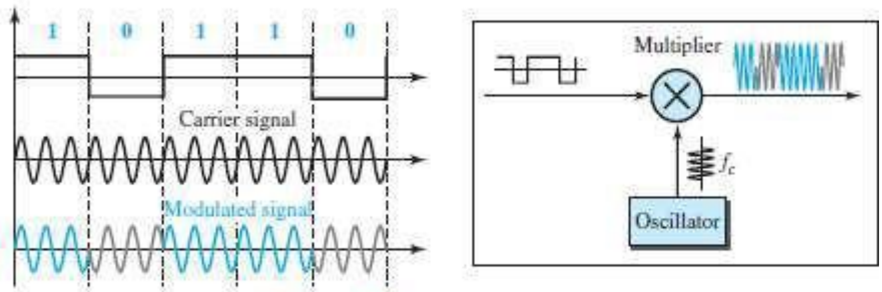


Figure 5.10 *Implementation of BASK*

Bandwidth for BPSK

• The bandwidth is the same as that for BASK, but less than that for BFSK. (Figure 5.9b)

• No bandwidth is wasted for separating 2 carrier-signals.

Quadrature PSK (QPSK)

• The scheme is called QPSK because it uses 2 separate BPSK modulations (Figure 5.11):

1) First modulation is in-phase,

2) Second modulation is quadrature (out-of-phase).

• A serial-to-parallel converter

→ accepts the incoming bits

→ sends first bit to first modulator and

→ sends second bit to second modulator.

• The bit to each BPSK signal has one-half the frequency of the original signal.

• Advantages:

1) Decreases the baud rate.

2) Decreases the required bandwidth.

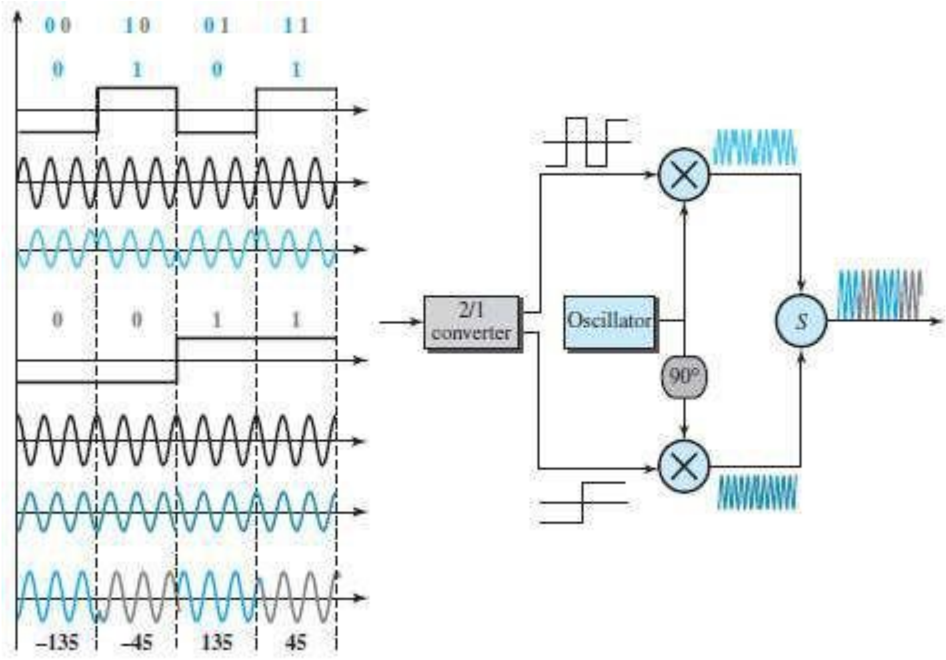


Figure 5.11 *QPSK and its implementation*

• As shown in Figure 5.11, the 2 composite-signals created by each multiplier are 2 sine waves with the same frequency, but different phases.
• When the 2 sine waves are added, the result is another sine wave, with 4 possible phases: 45°, -45°, 135°, and -135°.
• There are 4 kinds of signal-elements in the output signal (L=4), so we can send 2 bits per signal-element (r=2).

Constellation Diagram

• A constellation diagram can be used to define the amplitude and phase of a signal-element.
• This diagram is particularly useful
→ when 2 carriers (one in-phase and one quadrature) are used.
→ when dealing with multilevel ASK, PSK, or QAM.
• In a constellation diagram, a signal-element type is represented as a dot.
• The diagram has 2 axes (Figure 5.12):
3) The horizontal X axis is related to the in-phase carrier.
4) The vertical Y axis is related to the quadrature carrier.
• For each point on the diagram, 4 pieces of information can be deduced.
1) The projection of point on the X axis defines the peak amplitude of the in-phase component.
2) The projection of point on Y axis defines peak amplitude of the quadrature component.
3) The length of the line that connects the point to the origin is the peak amplitude of the signal-element (combination of the X and Y components);
4) The angle the line makes with the X axis is the phase of the signal-element.

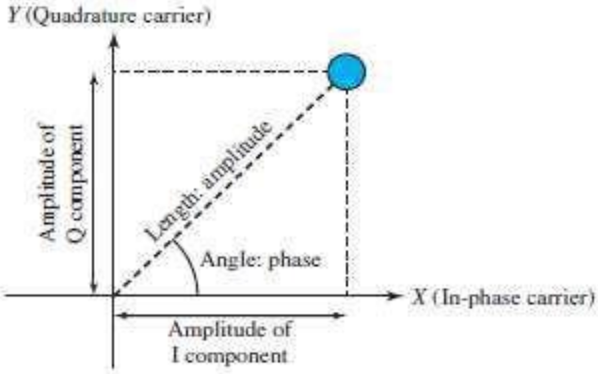


Figure 5.12 *Concept of a constellation diagram*

Quadrature Amplitude Modulation (QAM)

- This is a combination of ASK and PSK.
- Main idea: Using 2 carriers, one in-phase and the other quadrature, with different amplitude levels for each carrier.
- There are many variations of QAM (Figure 5.14).
  - A) Figure 5.14a shows the 4-QAM scheme using a unipolar NRZ signal. This is same as BASK.
  - B) Figure 5.14b shows another QAM using polar NRZ. This is the same as QPSK.
  - C) Figure 5.14c shows another 4-QAM in which we used a signal with 2 positive levels to modulate each of the 2 carriers.
  - D) Figure 5.14d shows a 16-QAM constellation of a signal with 8 levels, 4 positive & 4 negative.

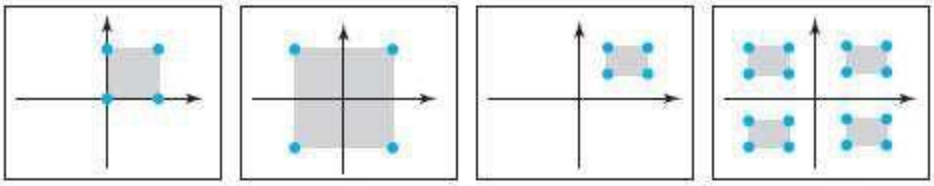


Figure 5.14 *Constellation diagrams for some QAMs*

Bandwidth for QAM

- The bandwidth is same as in ASK and PSK transmission.
- QAM has the same advantages as PSK over ASK.