Module-3
" Signal encoding techniques"

Syllabus:
Digital data to digital signals, digital data to analog signals, analog data to digital signals and analog data to analog signals.

3. SIGNAL ENCODING TECHNIQUES

- Both analog and digital information (data) can be encoded as either analog or digital signals.
- The particular encoding that is chosen depends on the transmission media and communications facilities available.
  - ✓ If the transmission media is baseband(use **lowpass channel**) then it is used to transmit digital signals, ,which supports for TDM(time division multiplexing)
  - ✓ If the transmission media is broadband(use **bandpass channel**) then it is used to transmit analog signals, which supports for FDM(frequency division multiplexing)

- **NOTE:** A channel which allows signals of **frequency zero** is ***lowpass channel***, where as zero frequency is not allowed by band pass channel.

- Four possible combinations of conversion from data to signals are:
  1. **Digital data, digital signals:**
     - ✓ The simplest form of digital encoding of digital data is to assign one voltage level to binary one and another to binary zero.
     - ✓ Many encoding schemes are used to improve performance, by altering the spectrum of the signal and providing synchronization capability.
  2. **Analog data, digital signals:**
     - ✓ Analog data, such as voice and video, are often digitized to be able to use digital transmission facilities.
     - ✓ The simplest technique is pulse code modulation (PCM), which involves sampling the analog data periodically and quantizing the samples.
  3. **Digital data, analog signal:**
     - ✓ A modem converts digital data to an analog signal so that it can be transmitted as analog signal.
     - ✓ The basic techniques are amplitude shift keying (ASK), frequency shift keying (FSK), and phase shift keying (PSK).
     - ✓ All involve altering one or more characteristics of a carrier frequency to represent binary data.
  4. **Analog data, analog signals:**
     - ✓ Analog data are modulated by a carrier to produce an analog signal.
     - ✓ The basic techniques are amplitude modulation (AM), frequency modulation(FM), and phase modulation (PM).

➢ **Digital signaling**:

✓ A data source *g*(*t*), which may be either digital or analog, is encoded into a digital signal *x*(*t*).

✓ The actual form of *x*(*t*) depends on the encoding technique and is chosen to optimize use of the transmission medium.

✓ For example, the encoding may be chosen to reduce bandwidth requirement or to minimize errors.

✓ **CODEC** (Encoder & decoder):

- It is a device acts as **encoder at the sender side** used to send analog or digital data *g(t)* in the form of digital signals *x(t)* through the transmission medium and acts as decoder at the receiver side i.e., digital signals are decoded to the data *g(t)* which is in the form sent by the sender

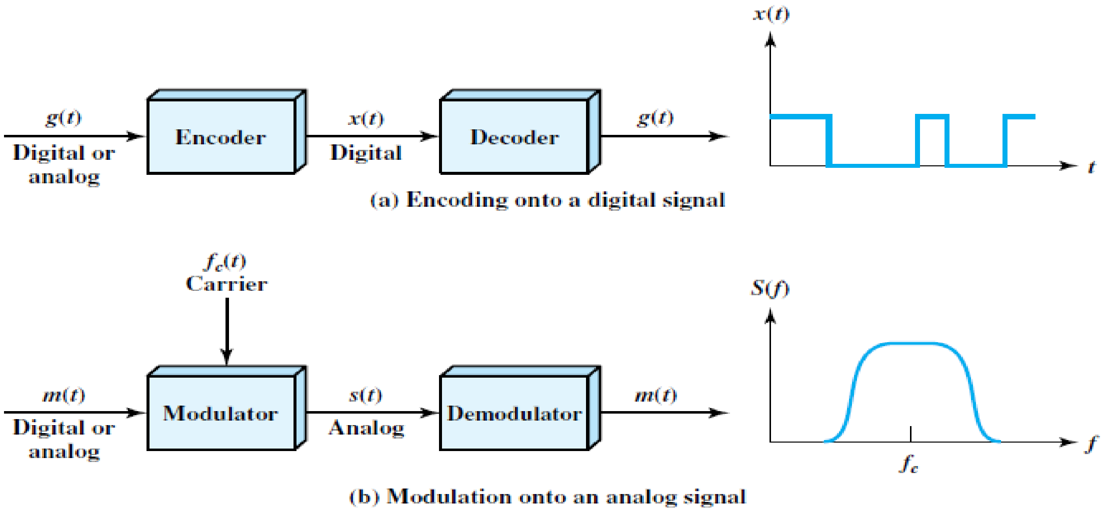


Figure 5.1 Encoding and Modulation Techniques

➢ **Analog signaling:**

✓ **It** is a continuous constant-frequency signal known as the **carrier signal**.

✓ Data may be transmitted using a carrier signal by modulation.

✓ **Modulation:**

- **It** is the process of encoding source data onto a carrier signal with frequency All modulation techniques involve operation on one or more of the three fundamental frequency f_c domain parameters: amplitude, frequency, and phase.
- The input signal *m*(*t*) may be analog or digital and is called the modulating signal.
- The result of modulating the carrier signal is called the modulated signal *s*(*t*).
- As Figure 5.1b indicates, *s*(*t*) is a bandlimited (bandpass) signal i.e., analog signal.

✓ **MODEM**(Modulator & Demodulator):

- It is used to convert the analog or digital data *m(t)* to analog signal *s(t)* at the sender side & transmitted through the transmission medium i.e., modulation takes place & at the receiver side demodulation i.e., analog signal is converted to the data either analog or digital *m(t)* to the form present at the sender side.

3.1 DIGITAL DATA TO DIGITAL SIGNALS

- 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.

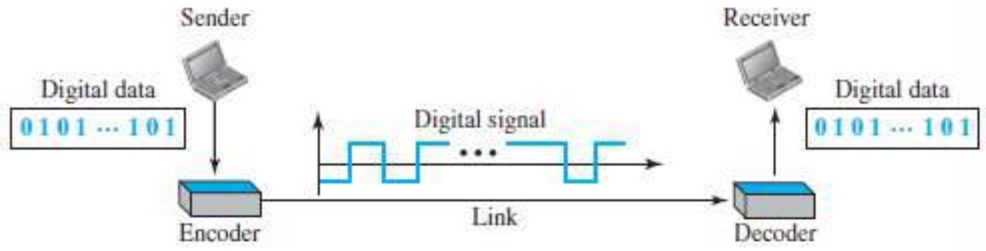


Figure 4.1 *Line coding and decoding*

3.1.1 Data Element Vs Signal element

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.

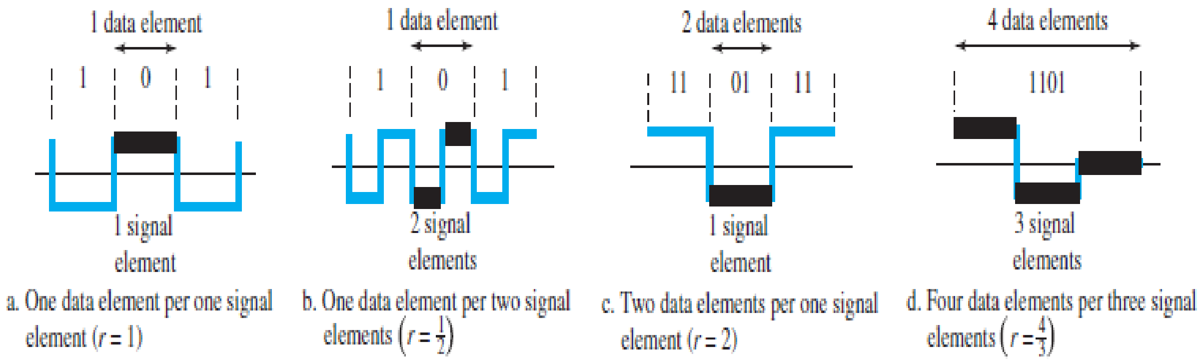


Figure 4.2 *Signal element versus data element*

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

3.1.2 Data Rate versus Signal Rate

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.

NOTE:

Table 5.1 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 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

- **3 factors** that determine how successful the receiver will be in interpreting the incoming signal are: **signal-to-noise ratio(SNR), the data rate, and the bandwidth.**
  - An increase in bandwidth allows an increase in data rate.
  - An increase in data rate increases bit error rate (BER) If SNR is decreasing
  - An increase in SNR decreases bit error rate.

- The encoding scheme is simply the mapping from data bits to signal elements. Before describing these techniques, let us consider the following ways of **evaluating or comparing the various techniques.**

- **Signal spectrum:**

✓ A lack of high-frequency components means that less bandwidth is required for transmission.

✓ In addition, lack of a direct-current (dc) component is also desirable.

- **Clocking:**

✓ There is need to determine the beginning and end of each bit position.

- ✓ To provide some synchronization mechanism that is based on the transmitted signal is achieved with suitable encoding
- **Error detection:**
  - ✓ It is desirable to have a built-in error-detecting capability in the generated code to detect some of or all the errors that occurred during transmission.
  - ✓ Some encoding schemes that we will discuss have this capability to some extent.
- **Signal interference and noise immunity:**
  - ✓ Certain codes exhibit superior performance in the presence of noise. Performance is usually expressed in terms of a BER.
- **Cost and complexity:**
  - ✓ The higher the signaling rate to achieve a given data rate, the greater the cost.
  - ✓ A complex scheme is more costly to implement than a simple one.
  - ✓ For example, a scheme that uses four signal levels is more difficult to interpret than one that uses only two levels.

➢ Line Encoding Mechanisms are broadly divided into 3 categories:

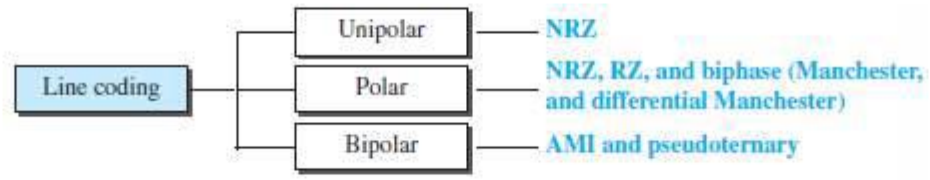
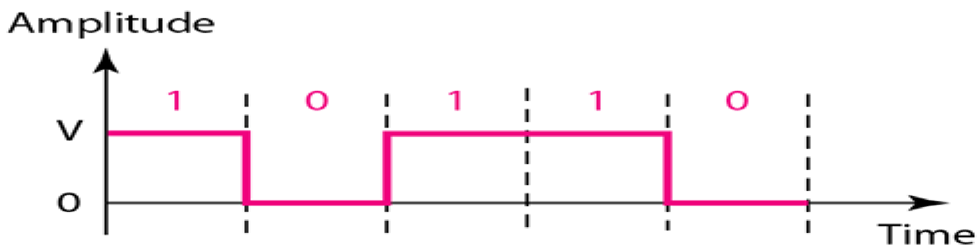


Figure 4.4 *Line coding schemes*

(a) 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.



(b) 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: -Ve for bit 1
  +Ve for bit 0.

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

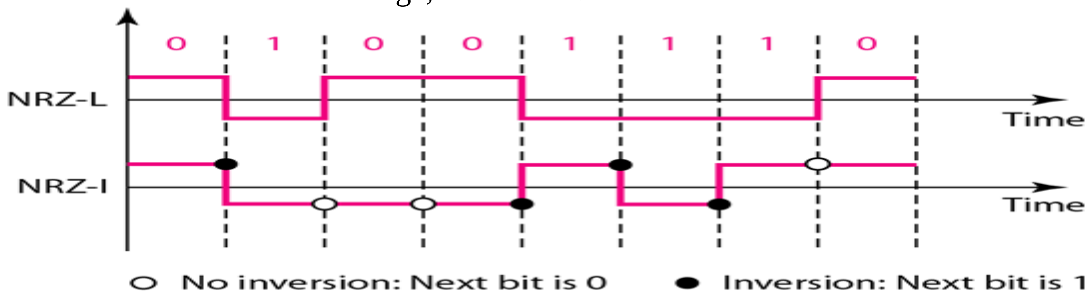
- We use 2 levels of voltage amplitude.
- Two versions of polar NRZ (Figure 4.6):

1. **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.

2. 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.

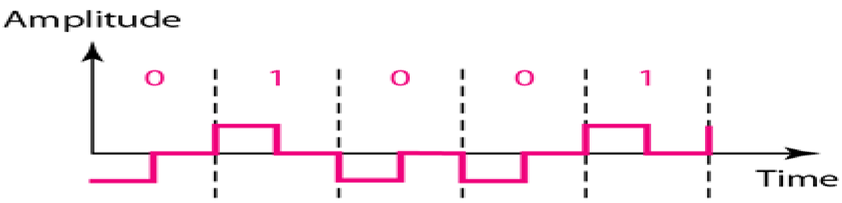


Nonreturn to Zero-Level (NRZ-L)
0 = high level
1 = low level
Nonreturn to Zero Inverted (NRZI)
0 = no transition
1 = transition

- ✓ The main **limitations of NRZ** signals are the presence of a ***dc component and the lack of synchronization capability***.
- ✓ With a long string of 1s or 0s for NRZ-L or a long string of 0s for NRZI, the output is a constant voltage (dc component) over a long period of time. These circumstances, will result in loss of synchronization.
- ✓ Voltage level is constant during the bit interval for NRZ-l or NRZ-I.

(ii) The 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)



(c) Biphase: Manchester & Differential Manchester

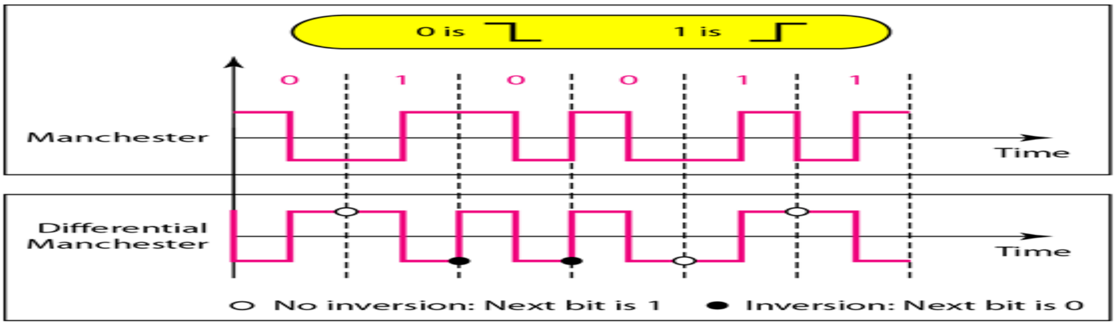
(i) Manchester Encoding

- ✓ 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
  - from high to low (for 0) or
  - 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.



➢ Modulation rate of Manchester and differential Manchester is twice that for NRZ; (as 2 signal elements are required to send 1 data element),this means that the **bandwidth required is correspondingly greater**.

➢ **Advantages**:

• **Synchronization:** Because there is a predictable transition during each bit time, the receiver can synchronize on that transition. For this reason, the biphase codes are known as self-clocking codes.

• **No dc component:** Biphase codes have no dc component, yielding the benefits described earlier.

• **Error detection:** The absence of an expected transition can be used to detect errors. Noise on the line would have to invert both the signal before and after the expected transition to cause an undetected error.

Manchester

0 = transition from high to low in middle of interval

1 = transition from low to high in middle of interval

Differential Manchester

Always a transition in middle of interval

0 = transition at beginning of interval

1 = no transition at beginning of interval

(d) **Bipolar Schemes (or Multilevel Binary)**

✓ This coding scheme uses 3 voltage levels (Figure 4.9):
 i) positive
 ii) negative &
 iii) zero.

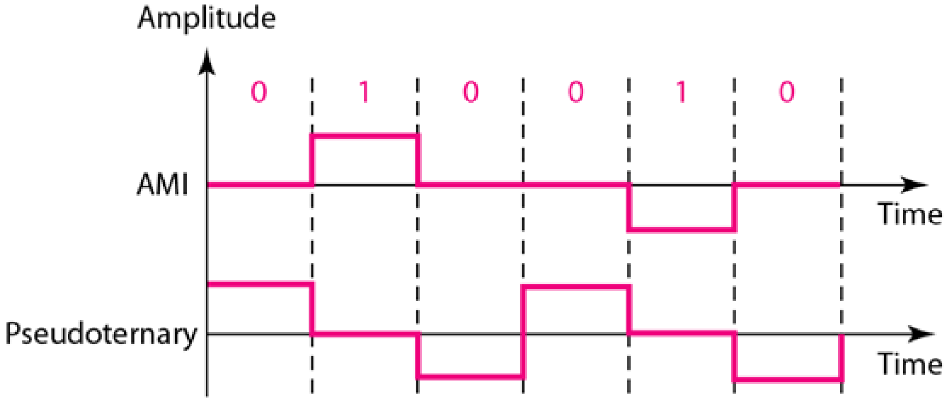
✓ Two variations of bipolar encoding:
 i) AMI (Alternate Mark Inversion)
 ii) Pseudoternary

(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.



✓ The bipolar AMI scheme was developed as an alternative to NRZ. The bipolar AMI scheme has the same signal rate as NRZ, but there is **no DC component**.

✓ For a long sequence of 1s, the voltage level alternates between positive and negative; it is not constant. Therefore, there is no DC component.

✓ For a long sequence of 0s, the voltage remains constant, but its amplitude is zero, which is the same as having no DC component. In other words, a sequence that creates a constant zero voltage does not have a DC component.

✓ AMI is commonly used for long-distance communication, but it has a **synchronization problem when a long sequence of 0s** is present in the data. Scrambling technique can solve this problem.

(e) Scrambling techniques:

✓ The idea behind this approach is simple: Sequences that would result in a constant voltage (here zero) level on the line are replaced by filling sequences that will provide sufficient transitions to maintain synchronization.

✓ The filling sequence must be recognized by the receiver and replaced with the original data sequence.

✓ The filling sequence is the same length as the original sequence, so there is no data rate penalty.

✓ The design goals for this approach can be summarized as follows:

• No dc component
• No long sequences of zero-level line signals
• No reduction in data rate

✓ Two techniques are commonly used in long-distance transmission services;

1. **B8ZS and**
2. **HDB3**

1. Bipolar 8-zeros substitution (B8ZS):

- The coding scheme is based on a bipolar-AMI.
- We have seen that the drawback of the AMI code is that a long string of zeros may result in loss of synchronization.
- To overcome this problem, the encoding is amended with the following rules:

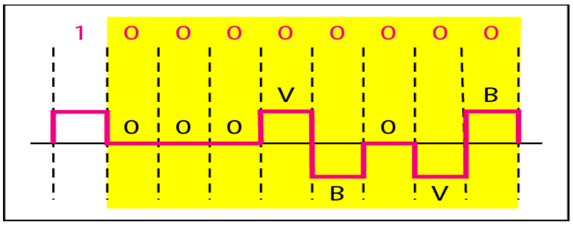
❖ **Case-1: Positive(last known voltage)**

▪ If an octet of all zeros occurs and the last voltage pulse preceding this octet was positive, then the eight zeros of the octet are encoded as **000+ -0- +**

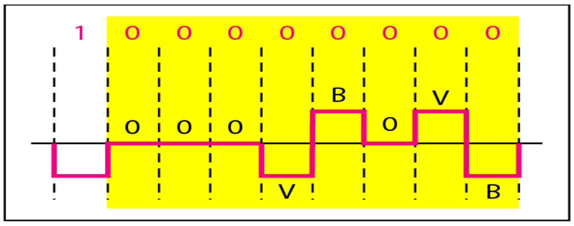
❖ **Case-2: Negative(last known voltage)**

▪ If an octet of all zeros occurs and the last voltage pulse preceding this octet was negative, then the eight zeros of the octet are encoded as **000- +0+ -**.

- This technique forces two code violations (signal patterns not allowed in AMI) of the AMI code, an event unlikely to be caused by noise or other transmission impairment.
- The receiver recognizes the pattern and interprets the octet as consisting of all zeros.
- **Example of B8ZS:**



a. Previous level is positive.



b. Previous level is negative.

2. HDB3:

- A coding scheme that is commonly used in Europe and Japan is known as the **high-density bipolar-3 zeros (HDB3)** code.
- As before, it is based on the use of AMI encoding. In this case, the scheme replaces strings of **four zeros** with sequences containing one or two pulses.
- In each case, the fourth zero is replaced with a code violation. Table 5.4 shows that this condition is tested for by determining

(1) Whether the number of non-zero pulses (1's) since the last violation is even or odd and

(2) The polarity of the last pulse before the occurrence of the four zeros.

Table 5.4 HDB3 Substitution Rules

	Number of Bipolar Pulses (ones) since Last Substitution	
Polarity of Preceding Pulse	**Odd**	**Even**
−	000−	+00+
+	000+	−00−

Or

Case 1: Odd +ve

000+

Case 2: Odd -ve

000-

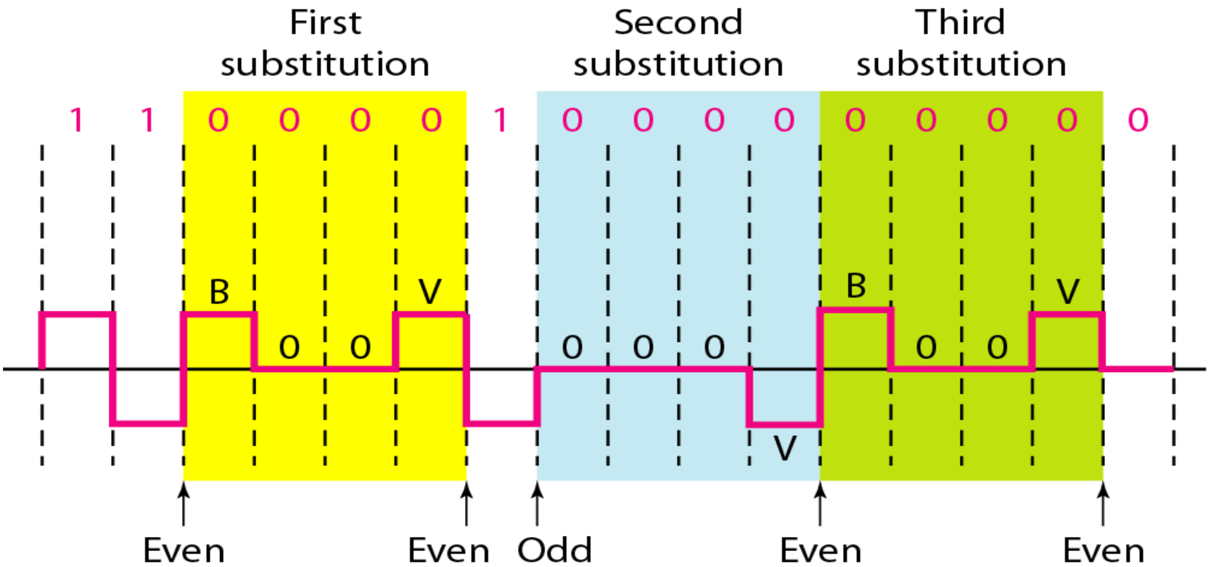
Case 3: Even +ve

-00-

Case 4 : Even -ve

+00+

✓ **Example for HDB3:**



➢ **Modulation Rate**

✓ When signal-encoding techniques are used, a distinction needs to be made between data rate (expressed in bits per second) and modulation rate (expressed in baud).

✓ The data rate, or bit rate, is R The modulation rate is the rate at which signal elements are generated.

$$D = \frac{R}{b} = \frac{R}{\log_2 M}$$

where

D = modulation rate, baud
R = data rate, bps
M = number of different signal elements = 2^b
b = number of bits per signal element

- ✓ Consider, for example, Manchester encoding.
- ✓ The minimum size signal element is a pulse of one-half the duration of a bit interval.
- ✓ For a string of all binary zeroes or all binary ones, a continuous stream of such pulses is generated. Hence the maximum modulation rate for Manchester is 2R.

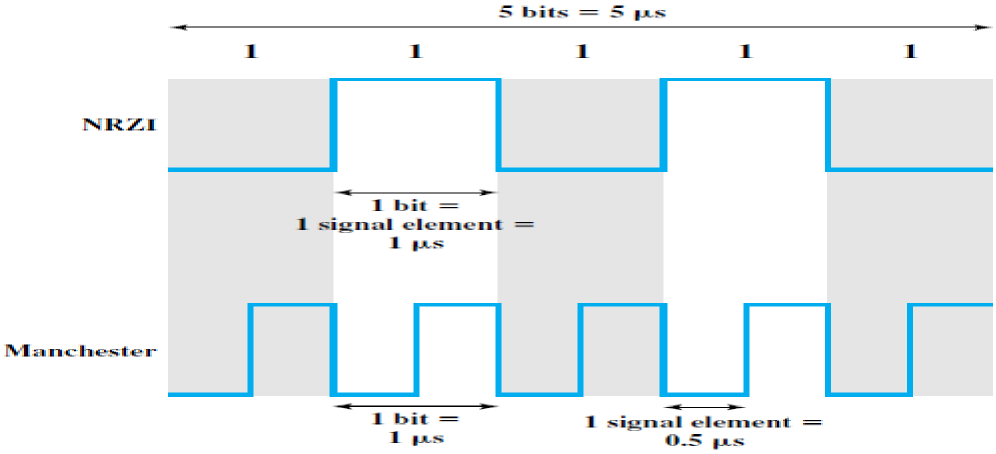


Figure 5.5 A Stream of Binary Ones at 1 Mbps

Nonreturn to Zero-Level (NRZ-L)
0 = high level
1 = low level

Nonreturn to Zero Inverted (NRZI)
0 = no transition at beginning of interval (one bit time)
1 = transition at beginning of interval

Bipolar-AMI
0 = no line signal
1 = positive or negative level, alternating for successive ones

Pseudoternary
0 = positive or negative level, alternating for successive zeros
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 a transition in middle of interval
0 = transition at beginning of interval
1 = no transition at beginning of interval

B8ZS
Same as bipolar AMI, except that any string of eight zeros is replaced by a string with two code violations

HDB3
Same as bipolar AMI, except that any string of four zeros is replaced by a string with one code violation

3.2 DIGITAL DATA TO ANALOG SIGNALS

- Digital-to-analog conversion is the process of changing one of the characteristics of an analog signal based on the information in digital data. i.e., **Modulation involves** operation on one or more of the **three characteristics of a carrier signal:**
  - ✓ **Amplitude,**
  - ✓ **Frequency, and**
  - ✓ **Phase**
- Accordingly, there are three basic encoding or modulation techniques for transforming digital data into analog signals, as illustrated in Figure below:
  - ✓ Amplitude shift keying (ASK),
  - ✓ Frequency shift keying (FSK),
  - ✓ Phase shift keying (PSK)
- **NOTE:** Digital devices are attached to the network via a modem (modulator-demodulator), which converts digital data to analog signals, and vice versa.
- In addition, there is a fourth (and better) mechanism that combines changing both the amplitude and phase, called quadrature amplitude modulation (QAM).

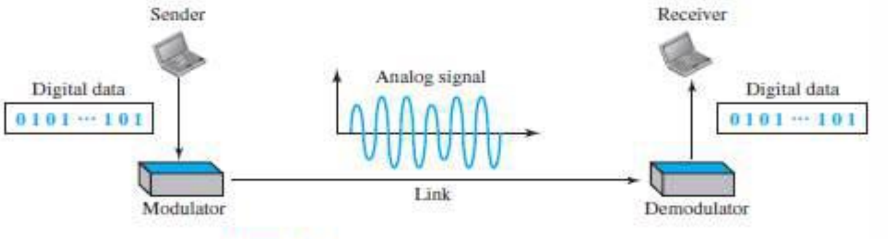


Figure 5.1 *Digital-to-analog conversion*

- Accordingly, there are three basic encoding or modulation techniques for transforming digital data into analog signals, as illustrated in Figure below:
  - ✓ Amplitude shift keying (ASK),
  - ✓ Frequency shift keying (FSK),
  - ✓ Phase shift keying (PSK)
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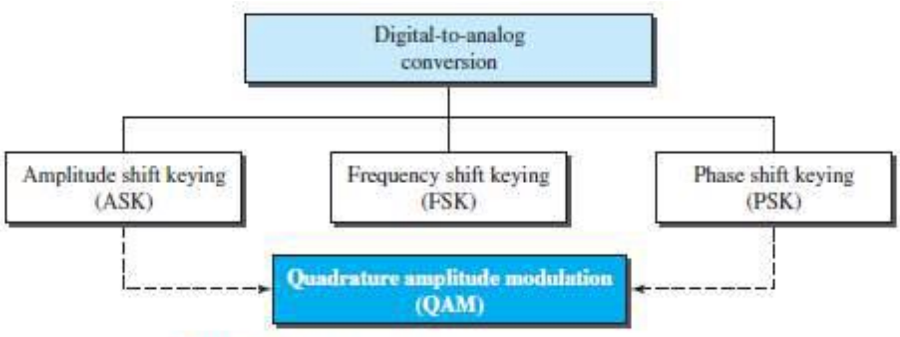


Figure 5.2 *Types of digital-to-analog conversion*

➢ 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

$$r = \log_2 L \text{ or } 2^r = L$$

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.

- ✓ **Example :**

1.

An analog signal carries 4 bits per signal element. If 1000 signal elements are sent per second, find the bit rate.

Solution

In this case, $r = 4$, $S = 1000$, and N is unknown. We can find the value of N from

$$S = N \times (1/r) \quad \text{or} \quad N = S \times r = 1000 \times 4 = 4000 \text{ bps}$$

2.

An analog signal has a bit rate of 8000 bps and a baud rate of 1000 baud. How many data elements are carried by each signal element? How many signal elements do we need?

Solution

In this example, $S = 1000$, $N = 8000$, and r and L are unknown. We first find the value of r and then the value of L.

$$S = N \times 1/r \longrightarrow r = N / S = 8000 / 10{,}000 = 8 \text{ bits/baud}$$

$$r = \log_2 L \longrightarrow L = 2^r = 2^8 = 256$$

3.2.1 Amplitude Shift Keying (ASK)

- ✓ In amplitude shift keying, the amplitude of the carrier signal is varied to create signal elements.
- ✓ Both frequency and phase remain constant while the amplitude changes.
- ✓ In ASK, the two binary values are represented by **two different amplitudes** of the carrier frequency.
- ✓ Commonly, one of the amplitudes is zero; that is, one binary digit is represented by the **presence**, at constant amplitude, of the carrier, the other by the **absence** of the carrier.
- ✓ The resulting transmitted signal for one bit time is

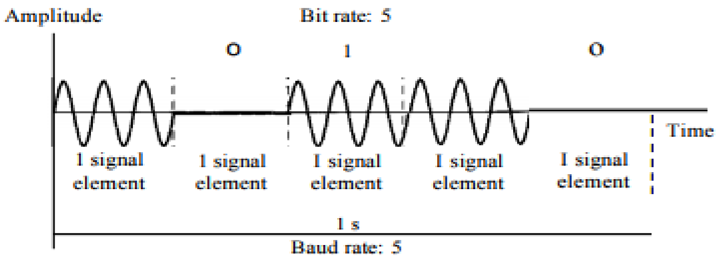
$$\textbf{ASK} \qquad s(t) = \begin{cases} A\cos(2\pi f_c t) & \text{binary 1} \\ 0 & \text{binary 0} \end{cases}$$

where

The carrier signal is **A cos(2πf$_c$t),**
A: Amplitude of carrier signal
f_c :Carrier frequency

(a) Binary ASK (BASK):

- ✓ ASK is normally implemented using only two levels.
- ✓ This is referred to as binary amplitude shift keying or on-off keying (OOK).
- ✓ The amplitude of one signal level is 0; the other is the same as the amplitude of the carrier frequency.



- ✓ **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.

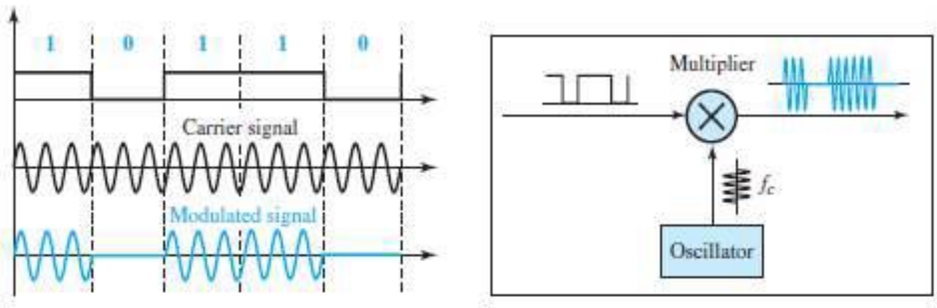


Figure 5.4 *Implementation of binary ASK*

(b) Multilevel ASK

- ✓ The above discussion uses only two amplitude levels.
- ✓ We can have multilevel ASK in which there are more than two levels.
- ✓ We can use 4, 8, 16, or more different amplitudes for the signal and modulate the data using 2, 3, 4, or more bits at a time.
- ✓ In these cases, r = 2, r = 3, r =4, and so on.
- ✓ But ASK is susceptible to sudden gain changes (leads to distortion) and is a rather inefficient modulation technique so generally multi-level ASK is not used.

Bandwidth for ASK

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

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

where d(0<d<1)= this factor depends on modulation and filtering-process.

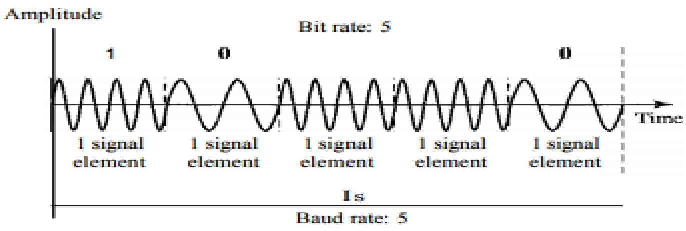
3.2.2. Binary FSK (BFSK)

- ✓ The most common form of FSK is binary FSK (BFSK), in which the two binary values are represented by two different frequencies near the carrier frequency, say two frequencies f_1 and f_2. f_1 if the data element is 0; f_2 if the data element is 1.
- ✓ The resulting transmitted signal for one bit time is-

$$\textbf{BFSK} \qquad s(t) = \begin{cases} A\cos(2\pi f_1 t) & \text{binary 1} \\ A\cos(2\pi f_2 t) & \text{binary 0} \end{cases}$$

Where,
f1 and f2 are typically offset from the carrier frequency
fc by equal but opposite amounts.

- ✓ In the below example f_c is taken as 3Hz, f1=4Hz, f2=2Hz which are offset by 1Hz from carrier frequency.



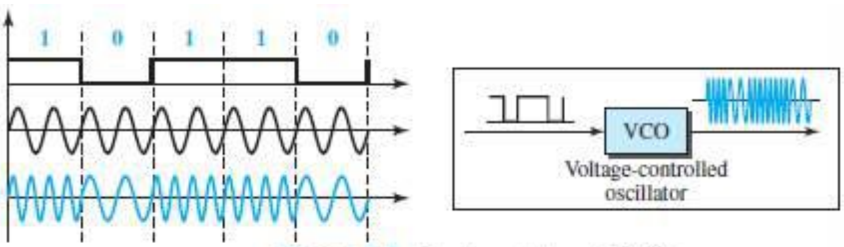


Figure 5.7 *Implementation of BFSK*

✓ **Multilevel FSK**

- Multilevel modulation (MFSK) use more than two frequencies.
- For example, we can use four different frequencies f1, f2, f3 and f4 to send 2 bits at a time.
- To send 3 bits at a time, we can use eight frequencies. And so on.
- In this case each signalling element represents more than one bit. The transmitted MFSK signal for one signal element time can be defined as follows:

$$\textbf{MFSK} \qquad s_i(t) = A\cos 2\pi f_i t, \qquad 1 \le i \le M$$

where

$$f_i = f_c + (2i - 1 - M)f_d$$
$$f_c = \text{the carrier frequency}$$
$$f_d = \text{the difference frequency}$$
$$M = \text{number of different signal elements} = 2^L$$
$$L = \text{number of bits per signal element}$$

✓ **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

$$B = (1 + d) \times \bar{S} + 2\Delta f$$

where 2Δf is the difference between f1 and f2,

EXAMPLE 5.1 With $f_c = 250$ kHz, $f_d = 25$ kHz, and $M = 8$ ($L = 3$ bits), we have the following frequency assignments for each of the eight possible 3-bit data combinations:

$f_1 = 75$ kHz	000	$f_2 = 125$ kHz	001
$f_3 = 175$ kHz	010	$f_4 = 225$ kHz	011
$f_5 = 275$ kHz	100	$f_6 = 325$ kHz	101
$f_7 = 375$ kHz	110	$f_8 = 425$ kHz	111

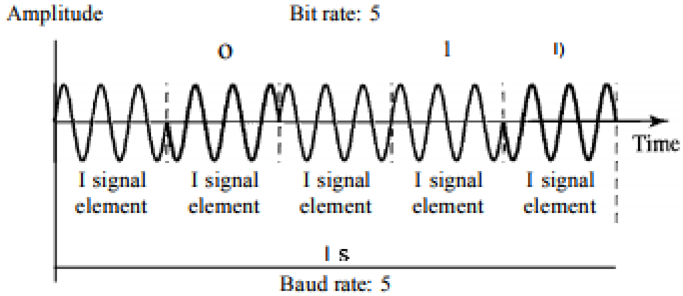
3.2.3. Phase Shift Keying (PSK)

✓ In PSK, the phase of the carrier signal is shifted to represent data.
✓ The simplest scheme uses two phases to represent the two binary digits and is known as binary phase shift keying or Two-Level PSK
✓ The resulting transmitted signal for one bit time is--

$$\textbf{BPSK} \qquad s(t) = \begin{cases} A\cos(2\pi f_c t) \\ A\cos(2\pi f_c t + \pi) \end{cases} = \begin{cases} A\cos(2\pi f_c t) & \text{binary 1} \\ -A\cos(2\pi f_c t) & \text{binary 0} \end{cases}$$

✓ In phase shift keying, 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.

✓ Here we have only two signal elements, one with a phase of 0°, and the other with a phase of 180°.



(a) Binary PSK

- It is as simple as binary ASK with one big advantage-PSK is less susceptible to noise.
- If we have a bit stream, and we define **d (t) as the discrete function** that takes value 1 if the corresponding bit in the bit stream is 1 and the value of d(t) is -1 if the corresponding bit in the bit stream is 0, i.e., d(t)=1 when bit is 1,d(t)=-1 if bit is 0.
- then we can define the transmitted signal as

$$\textbf{BPSK} \qquad s_d(t) = A\, d(t)\cos(2\pi f_c t)$$

- **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°.

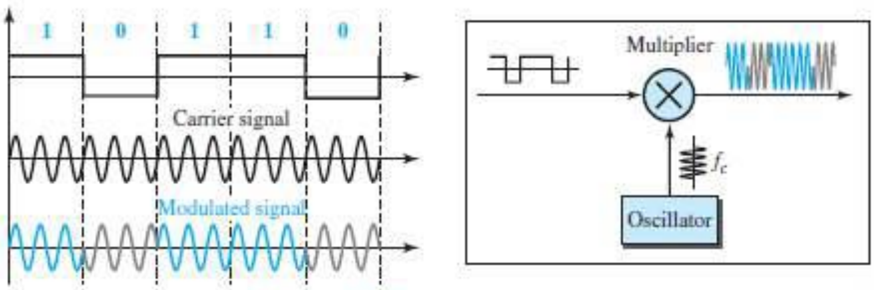


Figure 5.10 *Implementation of BASK*

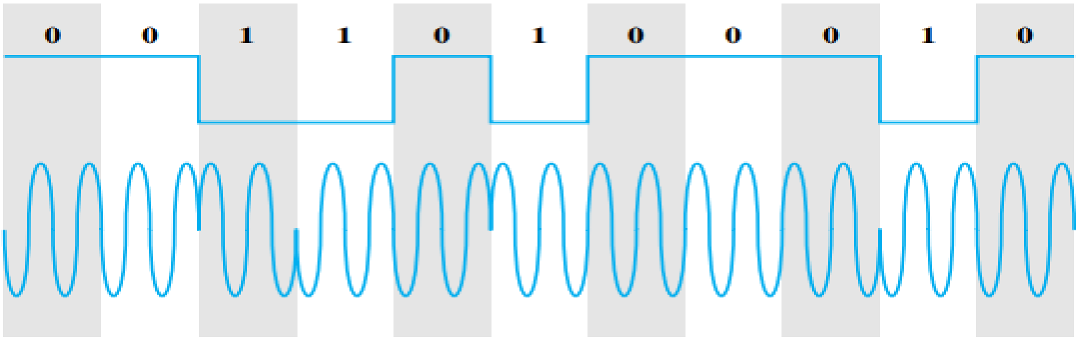
(b)Differential PSK (DPSK):

- An alternative form of two-level PSK is differential PSK (DPSK).
- In this scheme,

Binary 0 is represented by sending a signal of the **same phase** as the previous signal sent.

Binary 1 is represented by sending a signal of **opposite phase** to the preceding one.

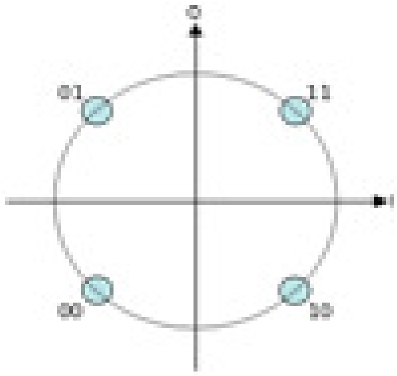
- This term ***differential*** refers to the fact that the phase shift is with reference to the previous bit transmitted rather than to some constant reference signal.



(c) Four-Level PSK (or) QPSK:

- More efficient use of bandwidth can be achieved if each signaling element represents more than one bit.
- For example, instead of a phase shift of 180˚, as allowed in BPSK, a common encoding technique, known as quadrature phase shift keying (**QPSK**), uses phase shifts separated by multiples of **π/2 (90˚)**, where each signal element represents two bits rather than one.

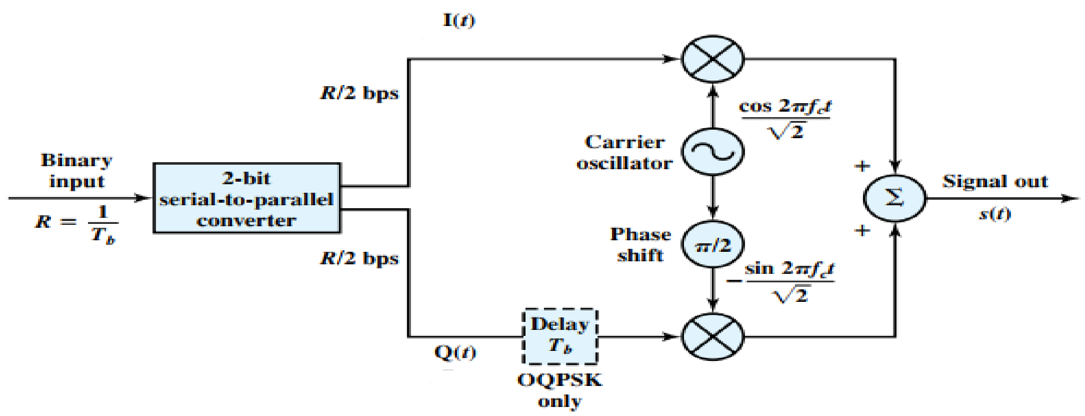
QPSK $$s(t) = \begin{cases} A\cos\left(2\pi f_c t + \frac{\pi}{4}\right) & 11 \\ A\cos\left(2\pi f_c t + \frac{3\pi}{4}\right) & 01 \\ A\cos\left(2\pi f_c t - \frac{3\pi}{4}\right) & 00 \\ A\cos\left(2\pi f_c t - \frac{\pi}{4}\right) & 10 \end{cases}$$



Implementation:

Figure below shows the QPSK modulation scheme in general terms.

- The input is a stream of binary digits with a data rate of $R = 1/Tb$, where Tb is the width of each bit.
- This stream is converted into two separate bit streams of $R/2$ bps each, by taking alternate bits for the two streams.
- The two data streams are referred to as the I (in-phase) and Q (quadrature phase) streams.
- The streams are modulated on a carrier of frequency fc by multiplying the bit stream by the carrier, and the carrier shifted by 90˚.
- The two modulated signals are then added together and transmitted.



- Thus, the combined signals have a symbol rate that is half the input bit rate.
- The use of multiple levels can be extended beyond taking bits two at a time.
- It is possible to transmit bits three at a time using eight different phase angles with phase shift 45degrees.

QPSK $$s(t) = \frac{1}{\sqrt{2}} I(t) \cos 2\pi f_c t - \frac{1}{\sqrt{2}} Q(t) \sin 2\pi f_c t$$

QPSK and OQPSK Modulators

Example: Data = 1 0 1 1 0 0 1 1 0 0 0 1 separated into two parts as

In-Phase part = 1 1 0 1 0 0 --> I(t) = 1 1 -1 1 -1 -1

Quadrature phase= 0 1 0 1 0 1 --> Q(t) = -1 1 -1 1 -1 1)

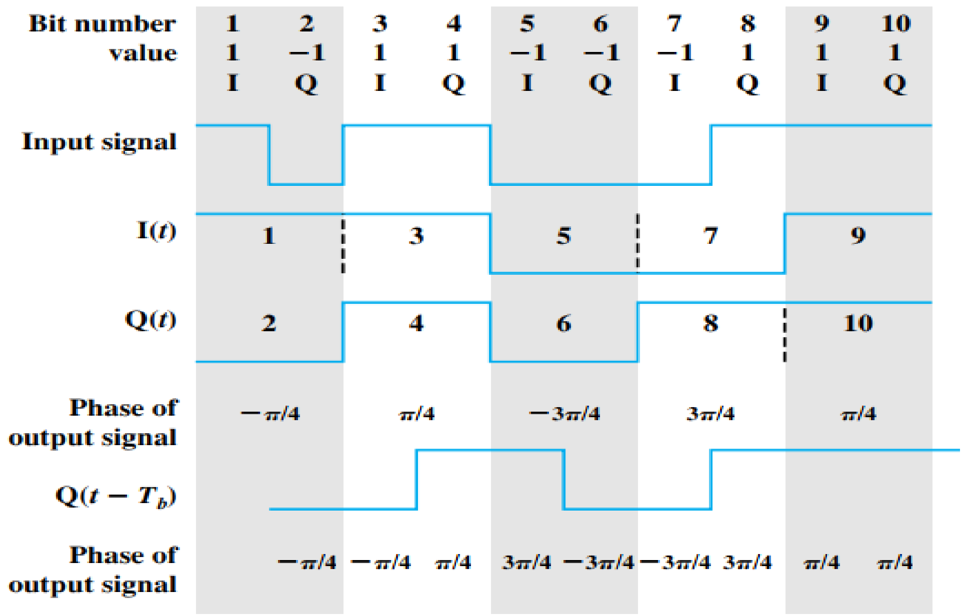


Figure 5.12 Example of QPSK and OQPSK Waveforms

- This figure also shows the variation of QPSK known as offset QPSK (OQPSK).
- For QPSK there are chances to get phase shift of 180 degrees instead of 90 degrees,so OQPSK is introduced.

- The difference is that a delay of one bit time is introduced in the Q stream for OQPSK to maintain phase shift of 90 degrees.

➢ **Performance of Digital to Analog Modulation Schemes:**

✓ The performance of various digital-to-analog modulation schemes, the first parameter of interest is the bandwidth of the modulated signal.

✓ This depends on a variety of factors, including the definition of bandwidth used and the filtering technique used to create the bandpass signal.

✓ The transmission bandwidth $\mathbf{B_T}$ for ASK is of the form

$$\textbf{ASK} \qquad B_T = (1 + r)R$$

where ,
R-> bit rate and
r ->related to the technique by which the signal is filtered to establish a bandwidth for transmission; typically 0<r<1.

✓ With multilevel PSK (MPSK), significant improvements in bandwidth can be achieved.

✓ In general,

$$\textbf{MPSK} \qquad B_T = \left(\frac{1 + r}{L}\right)R = \left(\frac{1 + r}{\log_2 M}\right)R$$

where ,
L is the number of bits encoded per signal element and
M is the number of different signal elements.

✓ For multilevel FSK (MFSK), we have

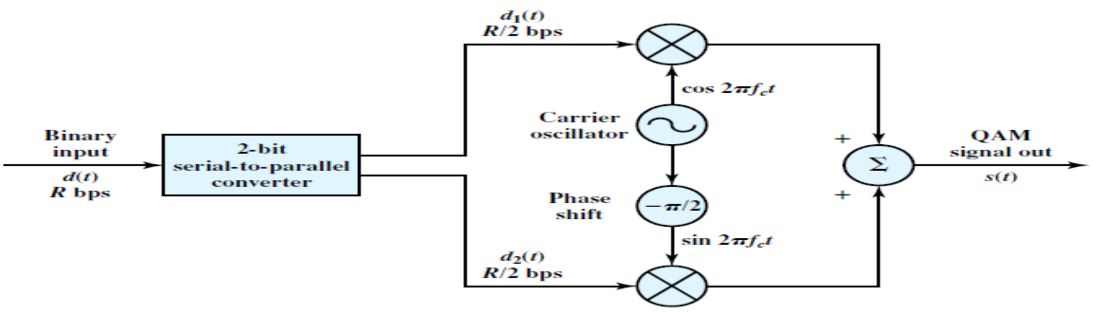
$$\textbf{MFSK} \qquad B_T = \left(\frac{(1 + r)M}{\log_2 M}\right)R$$

3.2.4 Quadrature Amplitude Modulation (QAM)

✓ Quadrature amplitude modulation (QAM) is a popular analog signaling technique that is a combination of ASK and PSK.

✓ QAM can also be considered a logical extension of QPSK and For QAM, each carrier is ASK modulated.

✓ QAM uses two copies of the carrier frequency, one shifted by 90° with respect to the other.



- ✓ Figure above shows the QAM modulation scheme in general terms.
- ✓ The input is a stream of binary digits arriving at a rate of *R* bps.
- ✓ This stream is converted into two separate bit streams of *R*/2 bps each, by taking alternate bits for the two streams.
- ✓ In the diagram, the upper stream is ASK modulated on a carrier of frequency *fc* by multiplying the bit stream by the carrier.
- ✓ Thus, a binary zero is represented by the absence of the carrier wave and a binary one is represented by the presence of the carrier wave at a constant amplitude.
- ✓ This same carrier wave is shifted by 90˚(-ve) and used for ASK modulation of the lower binary stream.
- ✓ The two modulated signals are then added together and transmitted.
- ✓ The transmitted signal can be expressed as follows:

$$\textbf{QAM} \qquad s(t) = d_1(t)\cos 2\pi f_c t + d_2(t)\sin 2\pi f_c t$$

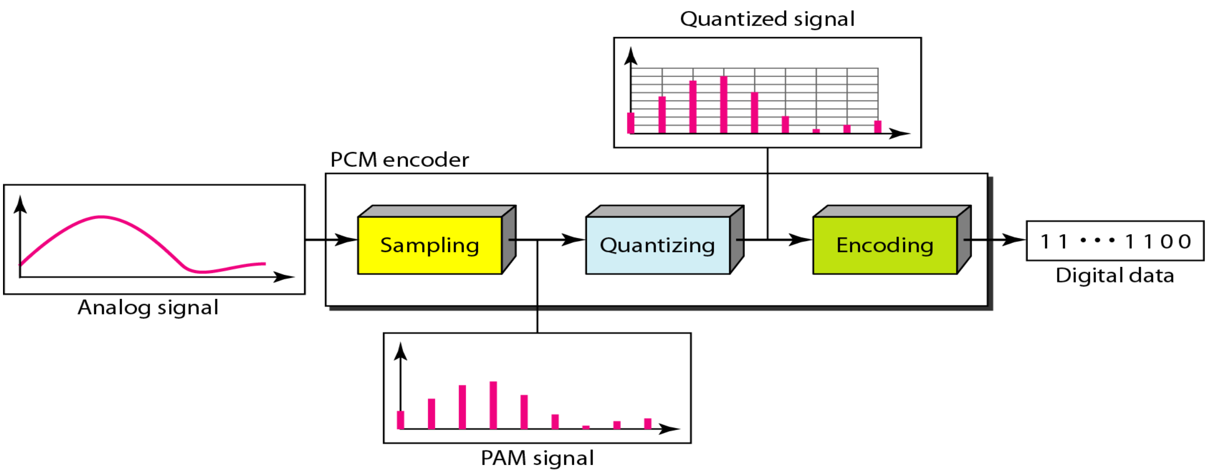
3.3 ANALOG DATA TO DIGITAL SIGNALS

- ➢ A process of converting analog data into digital data is known as digitization.
- ➢ Analog data, such as voice and video, is often digitized to be able to use digital transmission facilities.
- ➢ The device used for converting analog data into digital form for transmission at sender, and subsequently recovering the original analog data from the digital at receiver, is known as a **codec** (coder-decoder).
- ➢ Two principal techniques used in codecs, pulse code modulation and delta modulation.

3.3.1 Pulse Code Modulation (PCM)

- ➢ The most common technique to change an analog signal to digital data (digitization) is called pulse code modulation (PCM).
- ➢ A PCM encoder has three processes, as shown in Figure

  1) Sampling
  2) Quantization &
  3) Encoding.



1. The analog signal is sampled.
2. The sampled signal is quantized.
3. The quantized values are encoded as streams of bits.

(a) Sampling

✓ The first step in PCM is sampling.
✓ The analog signal is sampled every *Ts* s, where *Ts* is the sample interval or period.
✓ The inverse of the sampling interval is called the sampling rate or sampling frequency.

$$f_s = 1/T_s.$$

✓ Three sampling methods (Figure 4.22):

(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).

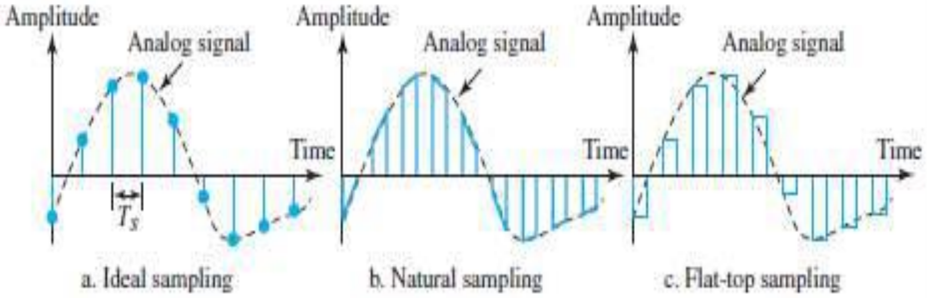
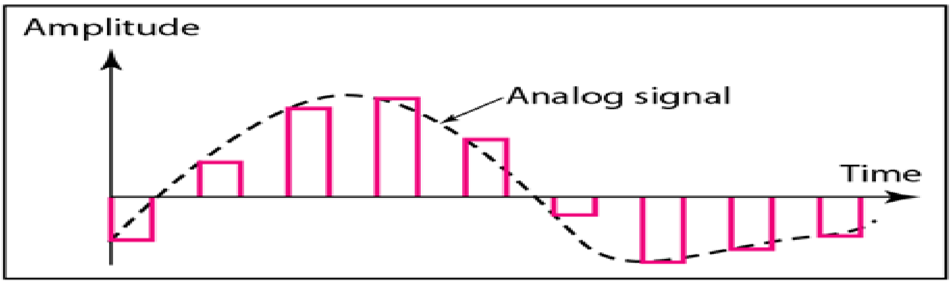


Figure 4.22 *Three different sampling methods for PCM*

✓ The most common sampling method, called sample and hold, create flat-top samples by using a circuit.



Flat-top sampling

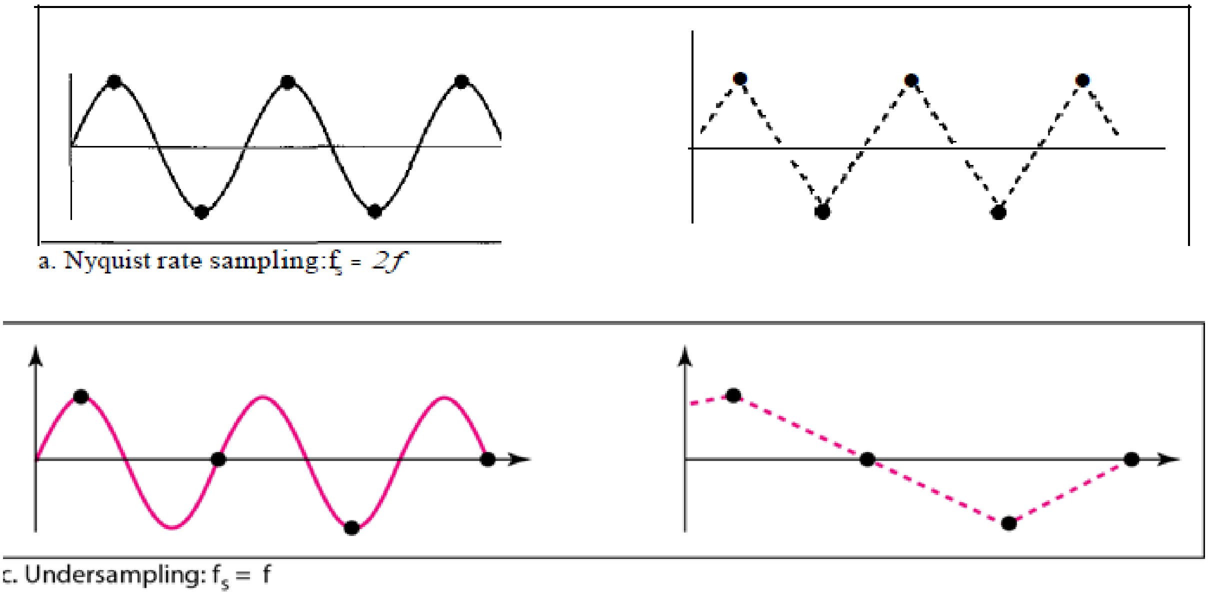
✓ **Sampling Rate:**

- According to the Nyquist theorem, to reproduce the original analog signal, one necessary condition is that the sampling rate be at least twice the highest frequency in the original signal.

✓ ***Example 4.9***

Telephone companies digitize voice by assuming a maximum frequency of 4000 Hz. The sampling rate therefore is 8000 samples per second.

Figure 4.24 ***Recovery of a sampled sine wave for different sampling rates***



✓ ***Example 4.10***

A complex low-pass signal has a bandwidth of 200 kHz. What is the minimum sampling rate for this signal?

Solution

We can sample this signal at 2 times the highest frequency (200 kHz). The sampling rate is therefore 400,000 samples per second.

(b) 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 Vmin & Vmax.

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.

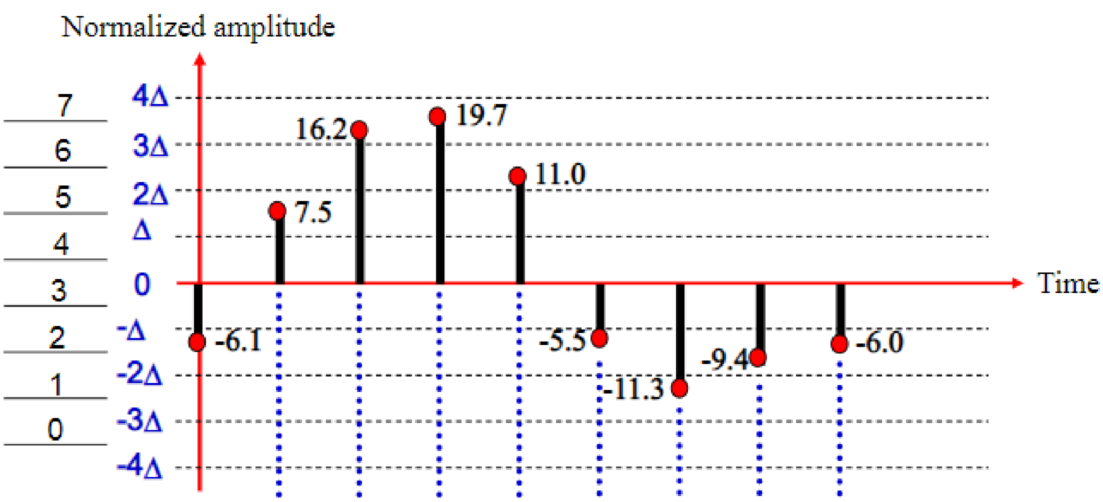
• For example:

Let Vmin=-20 Vmax =+20 V L = 8 Therefore, Δ = [+20-(-20)]/8= 5 In the chart (Figure)

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).

❖ As a simple (above) example, assume that we have a sampled signal and the sample amplitudes are between -20 and +20 V. We decide to have eight levels (L = 8). This means that D=5 V. Figure 4.26 shows this example.
   - ✓ The 8 zones are: -20 to -15, -15 to -10, -10 to -5, -5 to 0, 0 to +5, +5 to +10, +10 to +15, +15 to +20
   - ✓ The midpoints are: -17.5, -12.5, -7.5, -2.5, 2.5, 7.5, 12.5, 17.5
   - ✓ Each sample falling in a zone is then approximated to the value of the midpoint.
   - ✓ Each zone is then assigned a binary code.
   - ✓ The number of bits required to encode the zones, or the number of bits per sample as it is commonly referred to, is obtained as follows:
     $\mathbf{n_b = \log_2 L}$
   - ✓ Given our example, $\mathbf{n_b}$ = 3
   - ✓ The 8 zone (or level) codes are therefore: 000, 001, 010, 011, 100, 101, 110, and 111

Quantization code	2	5	7	7	6	2	1	2	2
Encoded words	010	101	111	111	110	010	001	010	010



NPV	-1.22	1.50	3.24	3.94	2.20	-1.10	-2.26	-1.88	-1.20
NQV	-1.50	1.50	3.50	3.50	2.50	-1.50	-2.50	-1.50	-1.50
NE	-0.38	0	+0.26	-0.44	+0.30	-0.40	-0.24	+0.38	-0.30
QC	2	5	7	7	6	2	1	2	2
EW	010	101	111	111	110	010	001	010	010

Quantization Error:

- ✓ One important issue is the error created in the quantization process. Quantization is an approximation process.
- ✓ The input values to the quantizer are the real values; the output values 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, there is no quantization error; otherwise, there is an error.
- ✓ **NOTE:** It can be proven that the contribution of the quantization error to the SNRdB of the signal depends on the number of quantization levels *L*, or the bits per sample *nb*

$$SNR_{dB} = 6.02nb + 1.76 \quad \mathbf{dB}$$

Uniform versus Nonuniform Quantization:

- ✓ For many applications, the distribution of the instantaneous amplitudes in the analog signal is not uniform.
- ✓ Changes in amplitude often occur more frequently in the lower amplitudes than in the higher ones.
- ✓ The signal is companded i.e,at the sender before conversion it is compressed; it is expanded at the receiver after conversion.
- ✓ Companding means reducing (compressing) the instantaneous voltage amplitude for large values; expanding is the opposite process.

Bit rate and bandwidth requirements of PCM:

- ✓ The bit rate of a PCM signal can be calculated form the number of bits per sample x the sampling rate

Bit rate = n_b x f_s

- ✓ The bandwidth required to transmit this signal depends on the type of line encoding used.

Example 4.14

We want to digitize the human voice. What is the bit rate, assuming 8 bits per sample?

Solution

The human voice normally contains frequencies from 0 to 4000 Hz. So the sampling rate and bit rate are calculated as follows:

$$\text{Sampling rate} = 4000 \times 2 = 8000 \text{ samples/s}$$
$$\text{Bit rate} = 8000 \times 8 = 64{,}000 \text{ bps} = 64 \text{ kbps}$$

4.25

(c) 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

$$n=\log_2 L \quad \text{or} \quad 2^n=L$$

✓ The bit-rate is given by:

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

Example 1:

A complex low-pass signal has a bandwidth of 200 kHz. What is the minimum sampling rate for this signal?

Solution

The bandwidth of a low-pass signal is between 0 and *f*, where *f* is the maximum frequency in the signal. Therefore, we can sample this signal at 2 times the highest frequency (200 kHz). The sampling rate is therefore 400,000 samples per second.

Example 2:

What is the SNR_{dB} in the example of Figure 4.26?

Solution

We can use the formula to find the quantization. We have eight levels and 3 bits per sample, so $SNR_{dB} = 6.02(3) + 1.76 = 19.82$ dB. Increasing the number of levels increases the SNR.

Example 3:

A telephone subscriber line must have an SNR_{dB} above 40. What is the minimum number of bits per sample?

Solution

We can calculate the number of bits as

$$SNR_{dB} = 6.02n_b + 1.76 = 40 \rightarrow n = 6.35$$

Telephone companies usually assign 7 or 8 bits per sample.

Example 4:

We want to digitize the human voice. What is the bit rate, assuming 8 bits per sample?

Solution

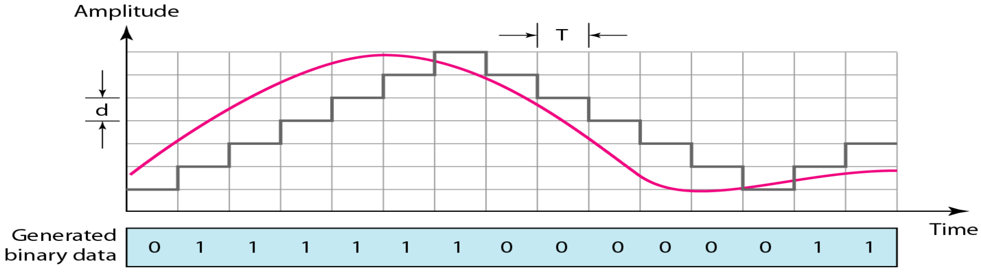
The human voice normally contains frequencies from 0 to 4000 Hz. So the sampling rate and bit rate are calculated as follows:

$$\text{Sampling rate} = 4000 \times 2 = 8000 \text{ samples/s}$$
$$\text{Bit rate} = 8000 \times 8 = 64{,}000 \text{ bps} = 64 \text{ kbps}$$

3.3.2 Delta Modulation:

✓ PCM is a very complex technique. Other techniques have been developed to reduce the complexity of PCM.

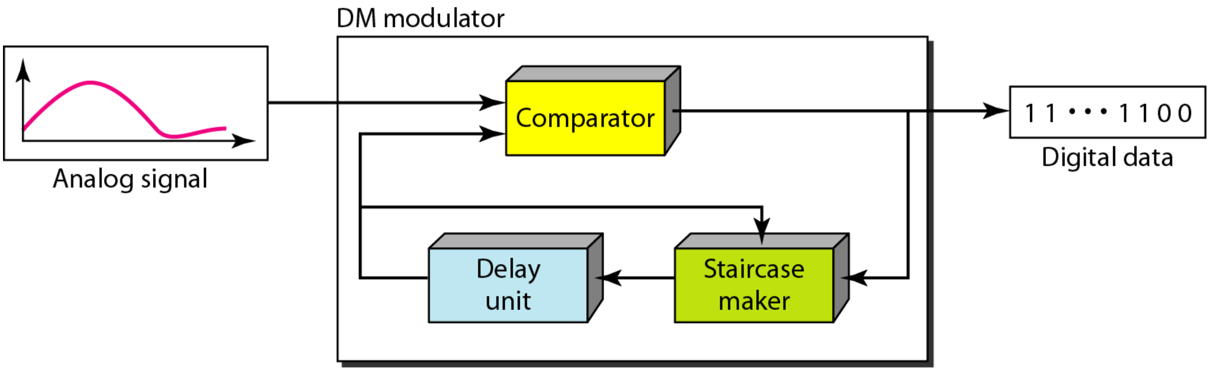
- ✓ The simplest is *delta modulation.*
- ✓ PCM finds the value of the signal amplitude for each sample; DM finds the change from the previous sample.
- ✓ Figure 4.28 shows the process. Note that there are no code words here; bits are sent one after another.



- ✓ This scheme sends only the difference between pulses, if the pulse at time t_{n+1} is higher in amplitude value than the pulse at time t_n, then a single bit, say a "1", is used to indicate the positive value.
- ✓ If the pulse is lower in value, resulting in a negative value, a "0" is used.
- ✓ This scheme works well for small changes in signal values between samples.
- ✓ NOTE: The principal advantage of DM over PCM is the simplicity of its implementation.

(a) Modulator

- ✓ The modulator is used at the sender site to create a stream of bits from an analog signal.
- ✓ The process records the small positive or negative changes, called delta O. If the delta is positive, the process records a 1; if it is negative, the process records a O.
- ✓ However, the process needs a base against which the analog signal is compared.
- ✓ The modulator builds a second signal that resembles a staircase.
- ✓ Finding the change is then reduced to comparing the input signal with the gradually made staircase signal.
- ✓ Figure 4.29 shows a diagram of the process.

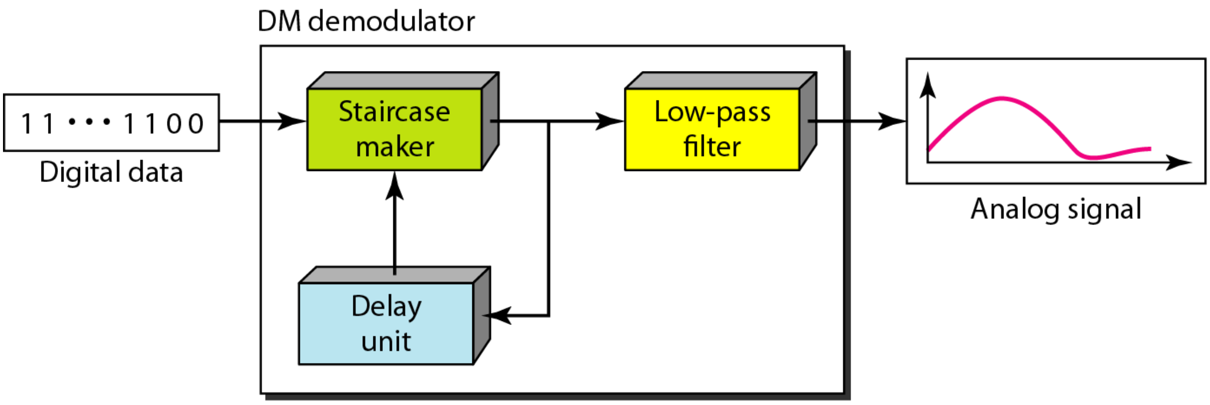


- ✓ The modulator, at each sampling interval, compares the value of the analog signal with the last value of the staircase signal.
- ✓ If the amplitude of the analog signal is larger, the next bit in the digital data is 1; otherwise, it is O.

- ✓ The output of the comparator, however, also makes the staircase itself. If the next bit is I, the staircase maker moves the last point of the staircase signal 0 up; it the next bit is 0, it moves it 0 down.
- ✓ Note that we need a delay unit to hold the staircase function for a period between two comparisons.

(b) Delta demodulation components:

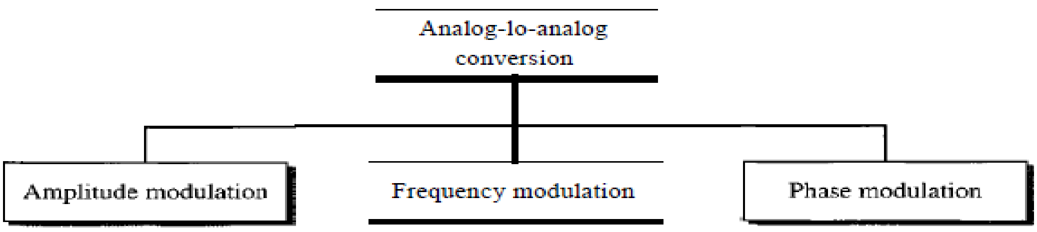
- ✓ The demodulator takes the digital data and, using the staircase maker and the delay unit, creates the analog signal.
- ✓ The created analog signal, however, needs to pass through a low-pass filter for smoothing.



3.4 ANALOG-TO-ANALOG CONVERSION

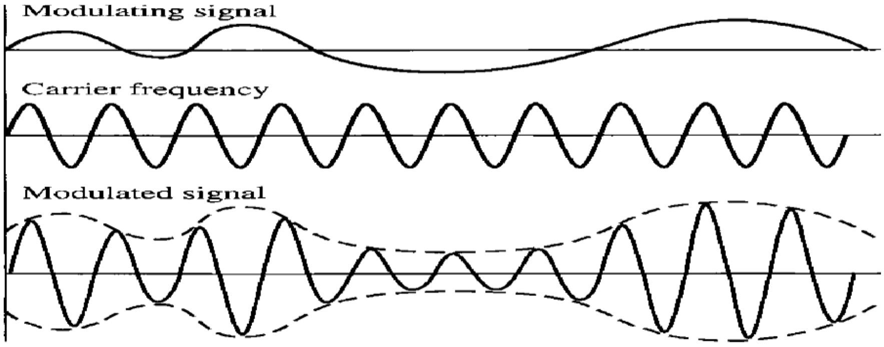
- ➢ Analog-to-analog conversion, or analog modulation, is the representation of analog information by an analog signal.
- ➢ Analog-to-analog conversion can be accomplished in three ways:
  - Amplitude Modulation (AM)
  - Frequency Modulation (FM) and
  - Phase Modulation (PM)

Figure 5.15 *Types ofanalog-to-analog modulation*



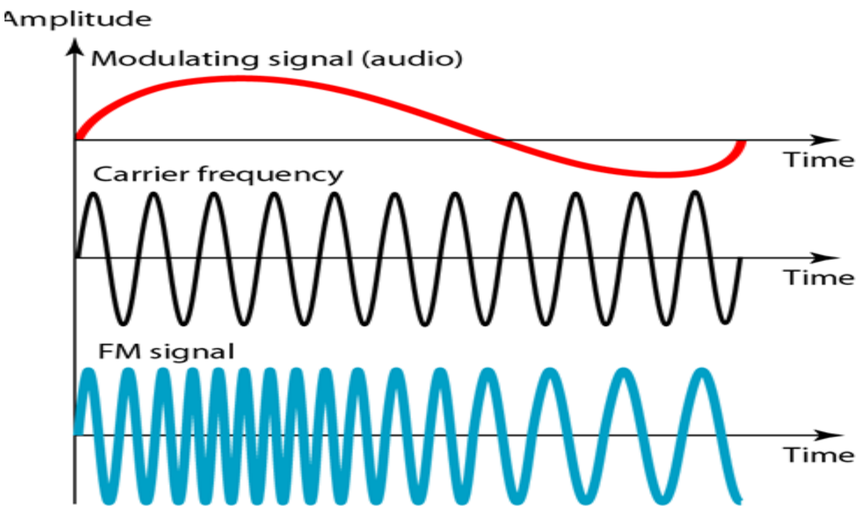
3.4.1 Amplitude Modulation

- In AM transmission, the carrier signal is modulated so that its amplitude varies with the changing amplitudes of the modulating signal (Signal which carries data).
- The bandwidth requirement is low when compared to FM and PM as frequency is not changed in the carrier signal .
- Generally Bandwidth of AM signal is two times the bandwidth of the modulating signal.
- The frequency and phase of the carrier remain the same; only the amplitude changes to follow variations in the information as shown in below fig.



3.4.2 Frequency Modulation

- In FM transmission, the frequency of the carrier signal is modulated to follow the changing voltage level (amplitude) of the modulating signal.
- The peak amplitude and phase of the carrier signal remain constant, but as the amplitude of the information signal changes, the frequency of the carrier changes correspondingly i.e., when amplitude of information signal is high the frequency of the carrier signal is high and vice versa.
- The bandwidth requirement for frequency modulation(FM) is higher than AM and PM.The total bandwidth required for PM can be determined from the bandwidth of modulating signal $B_{PM} = 2(1 + \beta)B$.Where $\beta = 4$ most often.
- Figure shows the relationships of the modulating signal, the carrier signal, and the resultant FM signal.



3.4.3 Phase Modulation

- In PM transmission, the frequency along with phase of the carrier signal is modulated to follow the changing voltage level (amplitude) of the modulating signal.
- The peak amplitude of the carrier signal remain constant, the frequency is high at the starting and ending of the signal and phase of the signal also changes as the amplitude of the information signal changes,the carrier changes correspondingly.
- The bandwidth required is more compared to Amplitude modulation as different frequencies are also used in Phase modulation.
- The total bandwidth required for PM can be determined from the bandwidth of modulating signal $B_{PM} = 2(1 + \beta)B$.Where $\beta = 2$most often

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