CHAPTER 4

Unit II

# Digital and Analog Transmission

**Syllabus :**

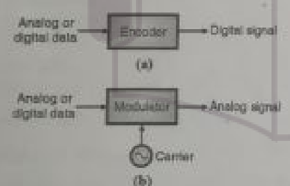
Digital transmission, Digital to digital conversion, Line coding, Line coding schemes, Analog to digital conversion, Pulse Code Modulation (PCM), Transmission modes, Parallel transmission, Serial transmission, Analog transmission, Digital to analog conversion, Aspects of digital to analog conversion, Amplitude shift keying, Frequency shift keying, Phase shift keying, Analog to analog conversion, Amplitude modulation, Frequency modulation, Phase modulation.

## 4.1 Digital Transmission :

- In computer networks, information is sent from one point to the other. We cannot send this information as it is. Instead it has to be converted into either a digital signal or an analog signal, for transmission.
- Use of digital signals for transmission has many advantages. In this chapter various schemes have been discussed for analog and digital transmission.

### 4.1.1 Encoding and Modulation :

- It is possible to encode any type of data into any type of signal as shown in Fig. 4.1.1.




(L-25) **Fig. 4.1.1 : Conversion from analog / digital data to analog / digital signal**

- Fig. 4.1.1(a) illustrates the concept of **digital signalling** in which the input data (analog or digital) is encoded into a digital signal.
- Fig. 4.1.1(b) illustrates the concept of **analog signalling** in which the analog / digital source is used for modulating a continuous time carrier signal to produce an analog signal called modulated signal.

**Encoding Types :**

There are four different possible transformations as follows :

1. Digital data, digital signal.
2. Analog data, digital signal.
3. Digital data, analog signal.
4. Analog data, analog signal.

## 4.2 Digital to Digital Conversion :

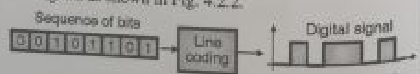
- In this type of encoding, the digital data which is normally binary in nature is converted into a sequence of discrete, discontinuous voltage pulses (digital signal).


(L-254) **Fig. 4.2.1 : Digital to digital conversion**

- The digital data at the input of the encoder may not be suitable for transmission over a longer distance. Hence it is converted into the digital signal which is more suitable for long distance communication.
- The digital signals at the output of the encoder are known as the line codes.

### 4.2.1 Definition of Line Coding :

- The line coding is defined as the process of converting binary data, a sequence of bits to a digital signal.
- The digital data such as text, numbers, graphical images, audio and video are stored in computer memory in the form of sequences of bits.
- Line coding converts these sequences into digital signals as shown in Fig. 4.2.2.




(L-255) **Fig. 4.2.2**

### 4.2.2 Some Important Characteristics of Line Coding :

Some of the important characteristics of line coding are :

1. Signal level and data level



2. Pulse rate and bit rate
3. DC component
4. Self synchronization

### 4.2.3 Signal Level and Data Level :

- A digital such as binary, octal, hex signal has a limited number of values. However all of them are not used to represent the data. Instead some of them only are used for representing the data.
- The remaining values are used for some other purpose.

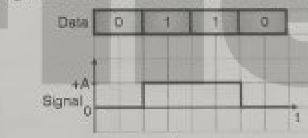
**Signal levels :**

The number of values allowed in a particular signal is defined as the number of signal levels. Refer Fig. 4.2.3 for better understanding.

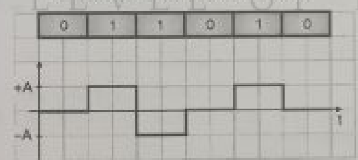
**Data levels :**

The number of values to represent data is called as the number of data levels. The binary data has two values 0 and 1.

- Fig. 4.2.3 gives you a clear idea about signal levels and data levels.
- Fig. 4.2.3 (a) has two signal levels (0 and A) and two data levels (0 and 1) whereas Fig. 4.2.3(b) has three signal levels (0, + A and – A) and two data levels.




**(a) Two signal levels, two data levels**




**(b) Two data levels, three signal levels**

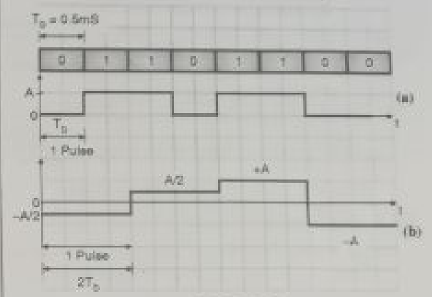
(L-256) **Fig. 4.2.3**

### 4.2.4 Pulse Rate and Bit Rate :

- **Pulse Rate** is defined as the number of pulses per second and a pulse is defined as the minimum amount of time required to transmit a symbol.
- **Bit Rate** is defined as the number of bits per second. It one pulse corresponds to one bit then the pulse rate is equal to the bit rate. But if a pulse carries more than 1 bit then the pulse rate is lower than bit rate.
- The relation between bit rate and pulse rate is as follows :

$$\text{Bit rate} = \text{Pulse rate} \times \log_2 L \qquad \ldots(4.2.1)$$

**Ex. 4.2.1 :** For the signals shown in Fig. P. 4.2.1(a) and (b) calculate the bit rate and pulse rate.




(L-257) **Fig. P. 4.2.1**

**Soln. :**

- The data has two levels and the bit duration $T_b = 0.5$ mS.
- Refer Fig. P. 4.2.1(a) which shows that there are two signal levels (0 or A)

$$\therefore \quad \text{Pulse rate} = \frac{1}{0.5 \times 10^{-3}} = 2000 \text{ pulses / sec.}$$

$$\text{and} \quad \text{Bit rate} = 2000 \times \log_2 2 = 2000 \text{ bps}$$

- Refer Fig. P. 4.2.1(b). Now one pulse corresponds to a duration of 2 $T_b$.

So pulse duration = $2\,T_b = 1$ mS.

$$\therefore \quad \text{Pulse rate} = \frac{1}{1 \times 10^{-3}} = 1000 \text{ pulses / sec.}$$

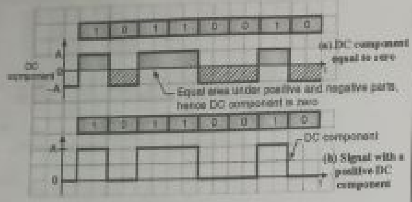
$$\text{And} \quad \text{Bit rate} = \text{Pulse rate} \times \log_2 L$$

$$\text{Here } L = 4 \text{ levels}$$

$$\text{Bit rate} = 1000 \times \log_2 4 = 2000 \text{ bps.}$$

### 4.2.5 DC Component :

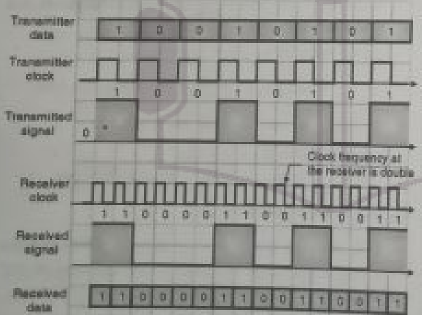
- Over one cycle period of a waveform, if all the positive voltages are cancelled by negative voltages then the DC component of the waveform is zero. (See Fig. 4.2.4(a)).
- But the waveform of Fig. 4.2.4(b) has a positive dc component because the instantaneous voltage can be either zero or positive.
- In line coding, the signal with a non-zero dc component is treated as a distorted one and it can create errors in the received signal.
- The signals with a dc component cannot pass through a transformer. Hence the signals with zero dc component are preferred.




(L-258) Fig. 4.2.4 : Concept of DC component

### 4.2.6 Self Synchronization :

- If the receiver's bit intervals correspond exactly to the sender's bit intervals, only then it is possible to receive (recognize) a signal correctly.
- The clock frequency of the transmitter and receiver should be the same.
- If the clock frequency at the receiver is slower or faster than the bit intervals are not matched and the received signal is different that the transmitted one.
- Fig. 4.2.5 illustrates the effect of change in clock frequency. The receiver clock frequency is twice that of the transmitter frequency. So received data is totally different than the transmitted one.




(L-259) Fig. 4.2.5 : Concept of synchronization

- Such thing would not happen if the receiver clock is **synchronized** with the transmitter clock.
- To achieve this, the transmitted digital signal includes the timing information. This will force the self synchronization.
- For achieving synchronization, the transmitted signal should cross the zero frequently. So if the transmitted signal consists of long trains of 0's or 1's, then the synchronization is affected.

### 4.2.7 Built in Error Detection :

- A line code should have a built in error detect... capability, so that at least some if not all the err... which have been introduced during the transit can b... detected. Some encoding schemes discussed in t... chapter have the error detection capability.

### 4.2.8 Immunity to Noise Interference :

- Noise immunity is another desirable property of a li... code. The noise interference should not be allowed... induce errors in the line codes.

### 4.2.9 Complexity :

- A line code should be as simple as possible. A li... coding scheme which uses four signal levels is mo... complex than that using two signal levels.

### 4.2.10 Bandwidth :

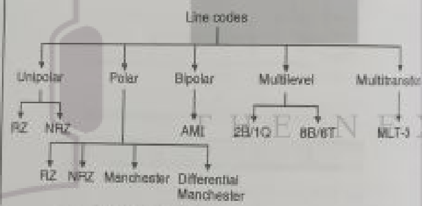
- Most digital signals that we come across have a fini... bandwidth. This is not the absolute bandwidth but it... the effective bandwidth.
- The bandwidth is proportional to the signal rate (bau... rate). The minimum bandwidth is given by,

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

Where C = Case factor, N = Data rate bps...

## 4.3 Classification of Line Codes :

- Fig. 4.3.1 shows the classification of line codes.




(L-260) Fig. 4.3.1 : Classification of line codes

- The line codes are basically divided into thre... categories :
  1. Unipolar codes 2. Polar codes
  3. Bipolar codes

1. **Unipolar codes :**
   - Unipolar codes use only one voltage level othe... than zero.
   - So the encoded signal will have either + A volt... value or 0.
   - These codes are very simple and primitive and ar... not used now a days.
2. **Polar codes :**
   - Polar coding using two voltage levels other tha... zero such as + A/2 and – A/2 volts.
   - This will bring the dc level for some codes to zer... which is a desired characteristics.

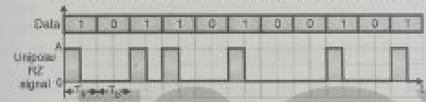


3. **Bipolar codes :**
   - Bipolar coding uses three voltage levels positive, negative and zero which is similar to polar codes.
   - But here the zero level is always used for representing the "0" in the data stream at the input.

## 4.4 Unipolar Line Codes :

### 4.4.1 Unipolar RZ Format :

- The return to zero (RZ) unipolar format is as shown in Fig. 4.4.1.
- In this format each "0" is represented by an off pulse (0) and each "1" by an on pulse with amplitude A and a duration of $T_b/2$ , followed by a return to zero level.
- Therefore this is called as return to zero (RZ) format. As the voltage level is either + A or zero, this is a unipolar format. (Unipolar means only one polarity).

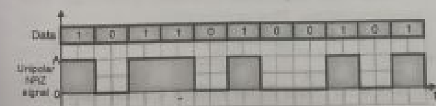



(L-262) Fig. 4.4.1 : Unipolar RZ format

- Due to the unipolar nature, the unipolar RZ format has a nonzero dc value. The dc value does not contain any information.

### 4.4.2 Unipolar NRZ Format :

- A non-return to zero (NRZ) format is as shown in Fig. 4.4.2.
- In this format a logic "1" is represented by a pulse of full bit duration $T_b$ and amplitude + A while a logic "0" is represented by an off pulse or zero amplitude.
- During the on time, the pulse does not return to zero after half bit period. Therefore the name NRZ format.
- As the pulses have either + A or 0 amplitude it is called as a unipolar format.
- Internal computer waveforms are usually of this type. Due to the unipolar nature, the unipolar NRZ format also will have a nonzero average (dc) value which does not carry any information.
- Due to longer pulse duration, the NRZ pulses carry more "energy" than the RZ pulses. But they need synchronization at the receiver as there is no separation between the adjacent pulses.

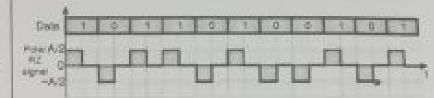



(L-263) Fig. 4.4.2 : Unipolar NRZ format

## 4.5 Polar Line Codes :

### 4.5.1 Polar RZ Format :

- The disadvantage of the two unipolar formats discussed earlier is that they result in a dc component that does not carry any information and wastes power.

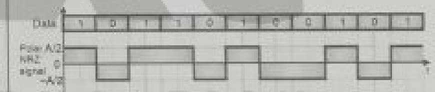



(L-264) Fig. 4.5.1 : Polar RZ format

- The polar RZ format is as shown in Fig. 4.5.1. It shows that opposite polarity pulses of amplitude ± "A/2" are used to represent logic "1" and "0".
- Therefore it is called as a "polar" format. As the pulses return to zero after half the bit duration "$T_b/2$" this format is a RZ format.

### 4.5.2 Polar NRZ Format :

- In the polar NRZ format, as shown in Fig. 4.5.2 a pulse of amplitude "+ A/2" of duration $T_b$ is used to represent a logic "1" and a pulse of amplitude "– A/2" of the same duration represents a logic "0".
- Unlike the unipolar waveform, a polar waveform has no dc component if the 0s and 1s in the input data occur in equal proportion.

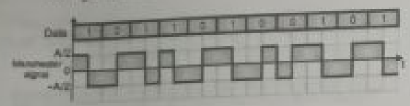



(L-265) Fig. 4.5.2 : Polar NRZ format

### 4.5.3 Split Phase Manchester Format :

- The split phase Manchester format is as shown in Fig. 4.5.3.
- In this format, symbol "1" is represented by transmitting a positive pulse of "+ A/2" amplitude for one half of the symbol duration, followed by a negative pulse of amplitude "– A/2" for remaining half of the symbol duration.
- For symbol "0" these two pulses are transmitted in reverse order.
- This waveform does not have any dc component.
- The Manchester format has a built in synchronization capability as it crosses zero at regular intervals. But this capability is attained at the expense of a bandwidth requirement of twice that of the NRZ unipolar, polar and bipolar formats.

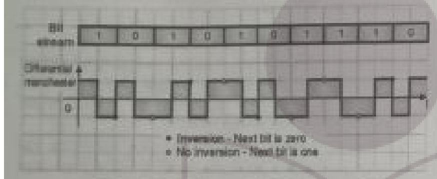
– Local area networks (LAN) such as Ethernet and Cheapernet are increasingly using the Manchester code for signal transmission over the network.




(L-266) Fig. 4.5.3 : Split phase manchester format

### 4.5.4 Differential Manchester Code :

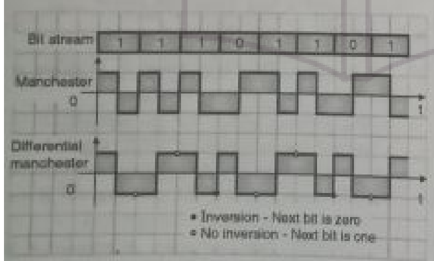
In this code there is always a transition in the middle of a bit interval. The binary zero has an additional transition at the beginning of the bit interval. This is as shown in Fig. 4.5.4 as inversion. There will be no additional transition (inversion) if the next bit is a 1.




(L-286(a)) Fig. 4.5.4 : Differential Manchester Coding

**Ex. 4.5.1 :** Show the Manchester and differential Manchester encoding pattern for the bit stream 11101101.

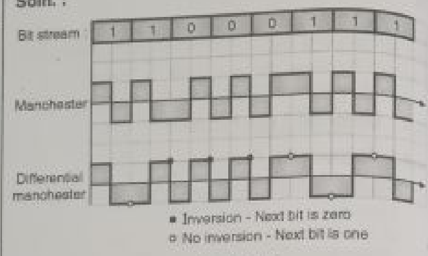
**Soln. :**




(L-287) Fig. P. 4.5.1

**Ex. 4.5.2** Show Manchester and differential Manchester encoding pattern for the bit stream 11000111.

**Soln. :**




(L-288) Fig. P. 4.5.2

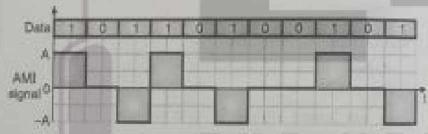
## 4.6 Bipolar Line Codes :

### 4.6.1 Bipolar NRZ Format (AMI) :

– The bipolar NRZ format is as shown in Fig. 4.6.1. Here the successive "1s" are represented by pulses with alternating polarity, and no pulse is transmitted for a logic "0".

– Note that in this representation there are three levels : + A, 0 and – A.

– Therefore this is also known as "pseudoternary or alternative mark inversion (AMI)" format.




(L-268) Fig. 4.6.1 : Bipolar NRZ format (AMI)

– An attractive feature of the bipolar format is the absence of a dc component eventhough the input binary data may contain long strings of "0s" and "1s".

– Moreover the bipolar format eliminates ambiguity that may arise because of polarity inversion during the course of transmissions. (This problem is observed in the switched telephone networks).

– This is the reason why the bipolar NRZ format is used in the PCM-TDM $T_1$ system for digital telephony.

– The absence of dc component allows the use of transformers for coupling.

## 4.7 Some Other Line Codes :

Some other line codes have been created for some special applications. They are :

1. 2 B1Q 2. Polar Quaternary 3. MLT - 3
4. Biphase M 5. Biphase S

### 4.7.1 2 B1Q (Two Binary, One Quaternary) :

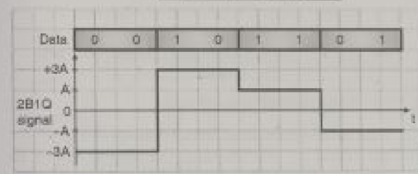
– The 2 B1Q code uses four voltage levels – 3A, – A, + 3A and + A in order to represent the digital input sequence.



– The input data bits are divided into groups of two bits each and each group is represented by one level as shown in Fig. 4.7.1.

**Table 4.7.1**

| Message | Level assigned |
|---|---|
| 0 0 | – 3A Volts |
| 0 1 | – A Volts |
| 1 0 | + 3A Volts |
| 1 1 | + A Volts |

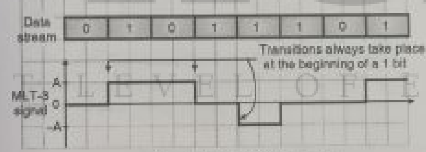



(L-269) Fig. 4.7.1 : 2 B1Q signal

### 4.7.2 MLT-3 (Multi Line Transmission, Three Level) :

– The MLT-3 is similar to NRZ-I but it uses three levels of signals (+ A, 0, – A).

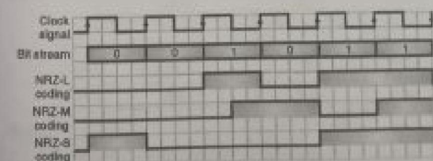
– The signal will change its value from one level to the next at the beginning of a 1 bit and there is no change in the signal value at the beginning of a 0 bit, as shown in Fig. 4.7.2.




(L-271) Fig. 4.7.2 : MLT-3 signal

### 4.7.3 NRZ-L (Non Return to Zero Level) :

– In NRZ-L coding a bit 0 or 1 is represented by a voltage level which remains constant during the bit duration. A binary "1" is represented by a high level and "0" is represented by a low level.




(L-272) Fig. 4.7.3 : NRZ signal coding

### 4.7.4 NRZ-M (Non Return to Zero Mark) :

As shown in Fig. 4.7.3, in NRZ-M the waveform changes its level when the binary digit is "1". The waveform does not change its level when the binary digit is "0".

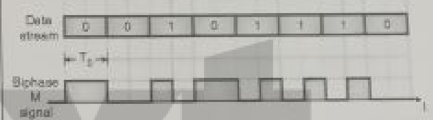
### 4.7.5 NRZ-S (Non Return to Zero Space) :

As shown in Fig. 4.7.3, in NRZ-S the waveform changes its level when the binary digit is "0". The waveform does not change its level when the binary digit is "1".

The other line codes are RZ (return to zero), AMI, manchester etc.

### 4.7.6 Biphase-M Code :

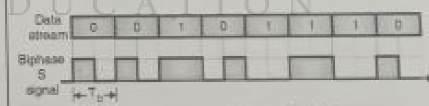
In this code, there is always a transition at the beginning of every bit interval. If the bit is a binary 1 then the coded signal returns to zero in the middle of the bit duration as shown in Fig. 4.7.4.




(L-273) Fig. 4.7.4 : Biphase-M coding

### 4.7.7 Biphase-S Code :

– In this code also there is always a transition at the beginning of every bit interval. If the data bit is a binary 0 then the coded signal returns to zero in the middle of the bit duration as shown in Fig. 4.7.5. No such return transition takes place if the bit is 1.




(L-274) Fig. 4.7.5 : Biphase-S coding

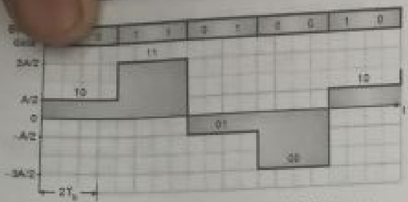
### 4.7.8 Polar Quaternary NRZ Format :

– Fig. 4.7.6 shows quaternary NRZ format derived by grouping the message bits in the blocks of two and using four amplitude levels to represent the four possible combinations 00, 01, 10 and 11.

– To these four combinations, four amplitude levels are assigned, as shown in Table 4.7.2.

**Table 4.7.2**

| Message combination | $x(t) = a_k$ |
|---|---|
| 0 0 | – 3 A/2 |
| 0 1 | – A/2 |
| 1 0 | A/2 |
| 1 1 | 3 A/2 |




(L-275) **Fig. 4.7.6 : Polar quaternary NRZ format**

- In the waveforms of Fig. 4.7.6, the first combination of two bits is "10" hence it is represented by a level "A/2".
- The second combination is "11" hence it is represented by a level of "3A/2". Thus here for a message of two bits only one pulse of duration $D = 2\,T_b$ is transmitted.

$$\therefore\ D = 2\,T_b \qquad ...(4.7.1)$$

The signaling rate : $r = \frac{1}{2\,T_b}$ messages/sec ...(4.7.2)

- If there are "M" levels obtained from the combination of "k" bits (here M = 4 and k = 2) then :

$$M = 2^k \qquad ...(4.7.3)$$

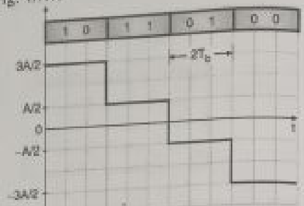
### 4.7.9 Gray Code :

- There is another scheme used for coding the quaternary format. It is called as the gray code.
- The gray coding scheme is illustrated in the following table. The adjacent bits are arranged in such a way that they differ by only one bit.

**Table 4.7.3 : Gray encoding**

| Message combination | x (t) |
|---|---|
| 0 0 | – 3 A/2 |
| 0 1 | – A /2 |
| 1 1 | A/2 |
| 1 0 | 3 A/2 |

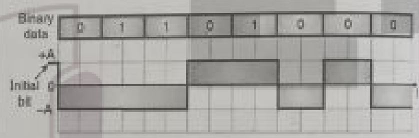
- The polar quaternary format with gray coding is shown in Fig. 4.7.7.




(L-276) **Fig. 4.7.7 : Polar quaternary format with gray coding**

### 4.7.10 Differential Encoding :

- The differential encoding format is shown in Fig. 4.7.8.
- The advantage of differential encoding is that it is immune from the polarity inversion-ambiguity problem.
- Differential encoding starts with an arbitrary initial bit.
- Corresponding to every 0 at the input, the differential encoding format make a transition from + A to – A or – A to + A whereas no transition takes place corresponding to a logic 1 at the input.




(L-277) **Fig. 4.7.8 : Differential encoding format**

- The original binary information can be recovered by sampling the received wave and comparing the polarities of the adjacent samples.

### 4.7.11 Comparison of Line Codes :

| Sr. No. | Parameter | Polar RZ | Polar NRZ | AMI | Manchester | Polar Quaternary NRZ |
|---|---|---|---|---|---|---|
| 1. | Transmission of DC component | Yes | Yes | No | No | Possible |
| 2. | Signaling rate | $1/T_b$ | $1/T_b$ | $1/T_b$ | $1/T_b$ | $1/2T_b$ |
| 3. | Noise immunity | Low | Low | High | High | High |
| 4. | Synchronizing capability | Poor | Poor | Very good | Very good | Poor |
| 5. | Bandwidth required | $1/T_b$ | $1/2T_b$ | $1/2T_b$ | $1/T_b$ | $1/2T_b$ |
| 6. | Crosstalk | High | High | Low | Low | Low |

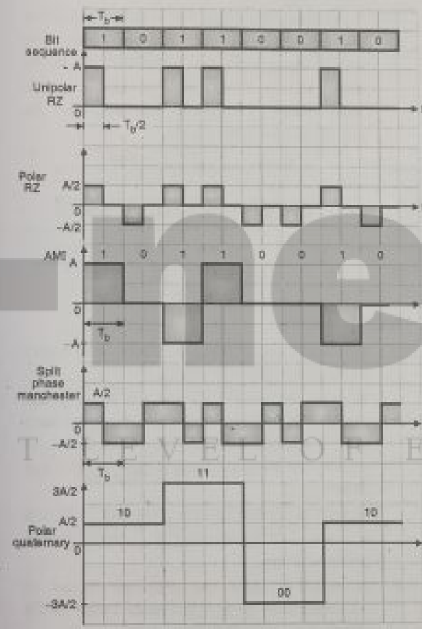


**Ex. 4.7.1 :** Consider that the bit sequence given below is to be transmitted. Bit sequence = 10110010. Draw the resulting waveform if the sequence is transmitted using :

1. Unipolar RZ
2. Polar RZ
3. AMI
4. Split Phase Manchester
5. M-ary where M = 4. (Polar quaternary)

**Soln. :**

The required waveforms are as shown in Fig. P. 4.7.1.



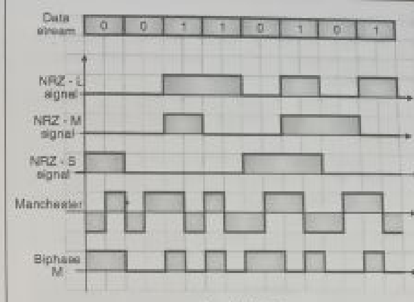

(L-279) **Fig. P. 4.7.1**

**Ex. 4.7.2 :** Sketch the signal waveforms when 0011 0101 is transmitted in the following signal codes :

1. NRZ-L
2. NRZ-M
3. NRZ-S
4. Manchester code
5. Biphase-M.

**Soln. :**

The required waveforms are as shown in Fig. P. 4.7.2.




(L-280) **Fig. P. 4.7.2**

**Ex. 4.7.3 :** How many amplitude levels are there for each of the following methods ?

1. NRZ-L 2. NRZ-S 3. Manchester 4. RZ 5. Differential Manchester 6. NRZ-M.

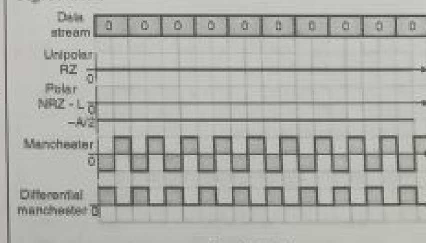
**Soln. :**

| Name of the format | NRZ-L | NRZ-S | Manchester | RZ | NRZ-M |
|---|---|---|---|---|---|
| Number of amplitude levels | 2 | 2 | 2 | 2 | 2 |

**Ex. 4.7.4 :** Assume a data stream is made of ten 0's. Encode this stream using the following schemes. How many changes can you find in each scheme ?

(a) Unipolar RZ (b) Polar NRZ-L (c) RZ (d) Manchester (e) Differential Manchester

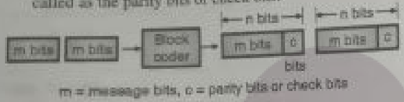
**Soln. :** The required waveforms are as shown in Fig. P. 4.7.4.




(L-281) **Fig. P. 4.7.4**

### 4.7.12 Block Coding :

- In order to ensure better error performance and synchronization, it is necessary to introduce redundancy.
- Block coding is one of the ways to introduce redundancy and enhance the performance of line coding.
- In general block coding is a technique that changes a block of m bits into a block of n bits with n > m. Therefore block coding is referred to as mB/nB coding technique.
- The additional bits added to the original m bits are called as the parity bits or check bits.




(L-282) **Fig. 4.7.9 : Concept of block coding**

- The examples of block codes are 4B/5B encoding in which a 4 bit code is converted into a 5 bit code or an 8B/10B code in which the 8 bit codes are converted into 10 bit codes.
- Fig. 4.7.10 shows the block diagram of a system using 4B/5B encoding.

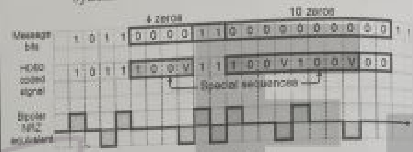

(L-283) **Fig. 4.7.10 : 4B/5B system**

### 4.7.13 Scrambling :

- The Biphase schemes that are suitable for the dedicated links in a LAN prove to be unsuitable for long distance communication as they require large bandwidths.
- The combination of block coding and NRZ line coding also proves to be unsuitable due to the presence of DC component.
- The AMI scheme has smaller BW and no DC component but a long sequence of 0s can create problems for its synchronization.
- For long distance communication, a technique is needed to be found which has small BW, no DC component and no synchronization problems.
- One of the solutions is to use **scrambling.** We can modify AMI for scrambling. Two commonly used scrambling techniques are :
  1. B8ZS
  2. HDB3 encoding

#### 1. High density bipolar (HDB) signalling :

- In case of the bipolar NRZ or AMI signal, the transmitted signal is equal to zero when a binary "0" is to be transmitted. This is true even for the unipolar RZ and unipolar NRZ signals.
- The absence of transmitted signal can cause problems in synchronization at the receiver, if long sequence of binary "0"s are being transmitted.
- This problem can be solved by adding (transmitting) pulses when long strings of 0's exceeding a number n are being transmitted. This type of coding is called as High Density Bipolar coding. It is denoted by HDBN. Here N = 1, 2, 3 ...... The most widely used HDB format is with N = 3 i.e. HDB3.
- In the string of message bits when (N + 1) or more number of zeros occur, they are replaced by special binary sequences of (N + 1) length. As shown in Fig. 4.7.11, these sequences contain some binary 1's which are necessary for synchronization at the receiver.



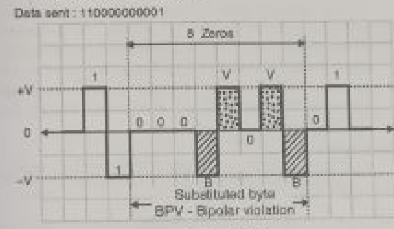

(L-284) **Fig. 4.7.11**

#### 2. B8ZS line code :

- The long form of B8ZS is bipolar with 8-zero substitution.
- We have discussed about the line codes in this chapter. We know that in order to have synchronization between the transmitter and receiver, the line code needs to cross the zero line frequently.
- As per U.S. $T_1$ standard, not more than 15 0's can be sent in succession to ensure proper synchronization.
- In order to solve the problems related to synchronization a new line code called B8ZS (Binary 8-zeros suppression) was developed.
- Whenever eight successive 0's are detected, the implementation of this line code will automatically insert a special 8 bit sequence containing a bipolar violation.
- This can be easily detected and corrected by the CSU/DSU (channel service unit/digital service unit).
- Refer Fig. 4.7.12 to get a clear idea about the B8ZS line code.
- The violations (BPV in Fig. 4.7.12), will distinguish a byte substituted for all 0's from a normal byte which contains 1's.



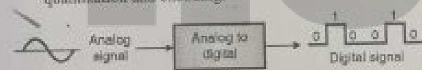
- The B8ZS does not allow more than 8 - consecutive 0's and the bipolar violation pattern uniquely identifies the eight 0's.
- Note that the voltage levels in the "violating byte", has a zero average (dc) value.




(L-285) **Fig. 4.7.12**

## 4.8 Analog to Digital Conversion :

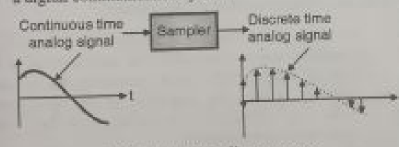
- The process of converting the analog data to digital signal is known as digitisation.
- This process is essential in all the digital communication systems such as pulse code modulation (PCM) or delta modulation (D.M).
- In order to carry out this transformation, one has to follow a sequence of operations such as sampling, quantization and encoding.




(L-218) **Fig. 4.8.1 : Transformation from analog signal to digital signal**

## 4.9 Sampling Process :

- In the pulse modulation and digital modulation systems, the signal to be transmitted must be in the discrete time form.
- If the message signal is coming from a digital source (e.g. a digital computer) then it is in the proper form for a digital communication system.




(L-156) **Fig. 4.9.1 : Sampling process**

- But this is not always the case. The message signal can be analog in nature (e.g. speech or video signal).
- In such a case it has to be first converted into a discrete time signal. We use the "sampling process" to do this.
- Thus using the sampling process we convert a continuous time signal into a discrete time signal.
- For the sampling process to be of practical utility it is necessary to choose the sampling rate properly. The sampling process should satisfy the following requirements :
  1. Sampled signal should represent the original signal faithfully.
  2. We should be able to reconstruct the original signal from its sampled version.
- Fig. 4.9.1 summarizes the sampling process.
- Thus sampling is the process of converting a continuous analog signal to a discrete analog signal and the sampled signal is the discrete time representation of the original analog signal.

### 4.9.1 Sampling Theorem for Low Pass Signals :

- In order to represent the original message signal "faithfully" (without loss of information), it is necessary to take as many samples of the original signal as possible.
- Higher the number of samples, closer is the representation.
- The number of samples depends on the "sampling rate" and the maximum frequency of the signal to be sampled.
- Sampling theorem was introduced to the communication theory in 1949 by Shannon. Therefore this theorem is also called as **"Shannon's sampling theorem".**
- The statement of sampling theorem in time domain, for the bandlimited signals of finite energy is as follows :

**Statement :**

1. If a finite energy signal x (t), contains no frequencies higher than "W" Hz (i.e. it is a band limited signal) then it is completely determined by specifying its values at the instants of time which are spaced (1/2W) seconds apart.
2. If a finite energy signal x (t) contains no frequency components higher than "W" Hz then it may be completely recovered from its samples which are spaced (1/2W) seconds apart.

- Combined statement of sampling theorem : A continuous time signal x (t) can be completely represented in its sampled form and recovered back from the sampled form if the sampling frequency $f_s \geq 2W$ where "W" is the maximum frequency of the continuous time signal x (t).

**Nyquist Rate :**

The minimum sampling rate of 2W samples per second for a signal x(t) having maximum frequency of W Hz is called as Nyquist rate.

$$\therefore \text{Nyquist rate} = f_{s(min)} = 2W \text{ Hz} \quad ...(4.9.1)$$

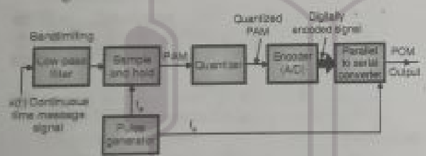
$f_s = 2 f_m$

## 4.10 Pulse Code Modulation (PCM) :

- PCM is a type of pulse modulation like PAM, PWM or PPM but there is an important difference between them. PAM, PWM or PPM are "analog" pulse modulation systems whereas PCM is a "digital" pulse modulation system.
- That means the PCM output is in the coded digital form. It is in the form of digital pulses of constant amplitude, width and position.
- The information is transmitted in the form of "code words". A PCM system consists of a PCM encoder (transmitter) and a PCM decoder (receiver).
- The essential operations in the PCM transmitter are sampling, quantizing and encoding.
- All these operations are usually performed in the same circuit called as analog-to-digital (A to D) converter.
- It should be understood that the PCM is not modulation in the conventional sense.
- Because in modulation, one of the characteristics of the carrier is varied in proportion with the amplitude of the modulating signal. Nothing of that sort happen in PCM.

### 4.10.1 PCM Transmitter (Encoder) :

- Block diagram of the PCM transmitter is as shown in Fig. 4.10.1.




(L-221) Fig. 4.10.1 : PCM transmitter (Encoder)

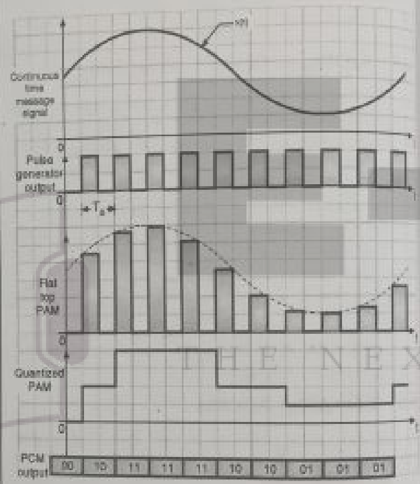
**Operation of PCM transmitter :**

Operation of the PCM transmitter is as follows :

- The analog signal x (t) is passed through a bandlimiting low pass filter, which has a cut-off frequency $f_c$ = W Hz. This will ensure that x (t) will not have any frequency component higher than "W". This will eliminate the possibility of aliasing.
- The band limited analog signal is then applied to a sample and hold circuit where it is sampled at adequately high sampling rate. Output of sample and hold block is a flat topped PAM signal.
- These samples are then subjected to the operation called "Quantization" in the "Quantizer". Quantization process is the process of approximation as will be explained later on. The quantization is used to reduce the effect of noise. The combined effect of sampling and quantization produces the quantized PAM at the quantizer output.
- The quantized PAM pulses are applied to an encoder which is basically an A to D converter. Each quantized level is converted into an N bit digital word by the A to D converter. The value of N can be 8, 16, 32, 64 etc.
- The encoder output is converted into a stream of pulses by the parallel to serial converter block. Thus at the PCM transmitter output we get a train of digital pulses.
- A pulse generator produces a train of rectangular pulses with each pulse of duration "τ" seconds. The frequency of this signal is "$f_s$" Hz. This signal acts as a sampling signal for the sample and hold block. The same signal acts as "clock" signal for the parallel to serial converter. The frequency "$f_s$" is adjusted to satisfy the Nyquist criteria.

**Waveforms :**

The waveforms at various points in the PCM transmitter are as shown in Fig. 4.10.2.



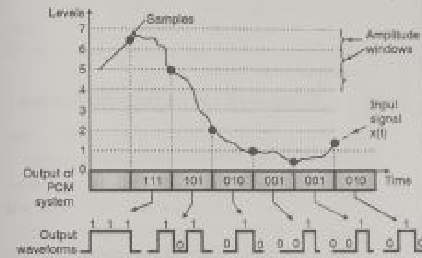

(L-222) Fig. 4.10.2 : Waveforms at different points in PCM transmitter

### 4.10.2 Shape of the PCM Signal :

- Fig. 4.10.3 shows input to and output of a PCM system. It is important to understand that the output is in the form of binary codes. Each transmitted binary code represents a particular amplitude of the input signal. Hence the "information" is contained in the "code" which is being transmitted.
- The range of input signal magnitudes is divided into 8 equal levels. Each level is denoted by a three bit digital word between 000 and 111.
- Input signal x (t) is sampled. If the sample is in the 5th window of amplitude then a digital word 101 is transmitted. If the sample is in the 2nd - window then the transmitted word is 010 and so on.



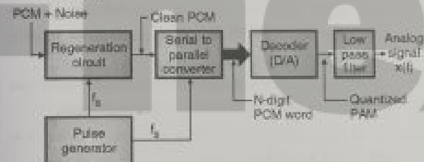
- In this example we have converted the amplitudes into 3 bit codes, but in practice the number of bits per word can be as high as 8, 9 or 10.




(L-223) Fig. 4.10.3 : Input and output waveforms of a PCM system

### 4.10.3 PCM Receiver (Decoder) :

- Fig. 4.10.4 shows the block diagram of a PCM receiver.




(L-224) Fig. 4.10.4 : PCM receiver (Decoder)

**Operation of PCM receiver :**

- A PCM signal contaminated with noise is available at the receiver input.
- The regeneration circuit at the receiver will separate the PCM pulses from noise and will reconstruct the original PCM signal.
- The pulse generator has to operate in synchronization with that at the transmitter. Thus at the regeneration circuit output we get a "clean" PCM signal.
- The reconstruction of PCM signal is possible due to the digital nature of PCM signal. The reconstructed PCM signal is then passed through a serial to parallel converter.
- Output of this block is then applied to a decoder.
- The decoder is a D to A converter which performs exactly the opposite operation of the encoder.
- The decoder output is the sequence of a quantized multilevel pulses. The quantized PAM signal is thus obtained, at the output of the decoder.
- This quantized PAM signal is passed through a low pass filter to recover the analog signal, x (t).
- The low pass filter is called as the reconstruction filter and its cut off frequency is equal to the message bandwidth W.

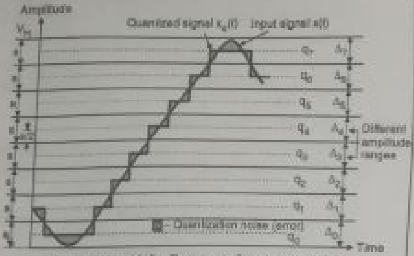
### 4.10.4 Quantization Process :

- Quantization is a process of approximation or rounding off. The sampled signal in PCM transmitted is applied to the quantizer block.
- Quantizer converts the sampled signal into an approximate quantized signal which consists of only a finite number of predecided voltage levels.
- Each sampled value at the input of the quantizer is approximated or rounded off to the nearest standard predecided voltage level.
- These standard levels are known as the "quantization levels". Refer to Fig. 4.10.5 to understand the process of quantization.
- The quantization process takes place as follows :
- The input signal x (t) is assumed to have a peak to peak swing of $V_L$ to $V_H$ volts. This entire voltage range has been divided into "Q" equal intervals each of size "s".
- "s" is called as the step size and its value is given as,

$$s = \frac{V_H - V_L}{Q} \quad \text{...(4.10.1)}$$

In Fig. 4.10.5, the value of Q = 8

- At the center of these ranges, the quantization levels $q_0, q_1, \ldots q_7$ are placed. Thus the number of quantization levels is Q = 8. The quantization levels are also called as decision thresholds.
- $x_q$ (t) represents the quantized version of x (t). We obtain $x_q$ (t) at the output of the quantizer.
- When x (t) is in the range $\Delta_0$, then corresponding to any value of x (t), the quantizer output will be equal to "$q_0$".
- Similarly for all the values of x (t) in the range $\Delta_1$, the quantizer output is constant equal to "$q_1$".
- Thus in each range from $\Delta_0$ to $\Delta_7$, the signal x (t) is rounded off to the nearest quantization level and the quantized signal is produced.
- The quantized signal $x_q$ (t) is thus an approximation of x (t). The difference between them is called as **quantization error or quantization noise.**
- This error should be as small as possible.
- To minimize the quantization error we need to reduce the step size "s" by increasing the number of quantization levels Q.




(L-225) Fig. 4.10.5 : Process of quantization

**Why is quantization required ?**

- If we do not use the quantizer block in the PCM transmitter, then we will have to convert each and every sampled value into a unique digital word.
- This will need a large number of bits per word (N). This will increase the bit rate and hence the bandwidth requirement of the channel.
- To avoid this, if we use a quantizer with only 256 quantization levels then all the sampled values will be finally approximated into only 256 distinct voltage levels.
- So we need only 8 bits per word to represent each quantized sampled value.
- Thus the number of bits per word can be reduced. This will eventually reduce the bit rate and bandwidth requirement.

### 4.10.5 Quantization Error or Quantization Noise $\epsilon$ :

- The difference between the instantaneous values of the quantized signal and input signal is called as quantization error or quantization noise.

$$\epsilon = x_q(t) - x(t) \qquad ...(4.10.2)$$

- The quantization error is shown by shaded portions of the waveform in Fig. 4.10.5.
- The maximum value of quantization error is $\pm s / 2$ where s is step size.
- Therefore to reduce the quantization error we have to reduce the step size by increasing the number of quantization levels i.e. Q.
- The mean square value of the quantization is given by,

$$\text{Mean square value of quantization error} = \frac{s^2}{12} \qquad ...(4.10.3)$$

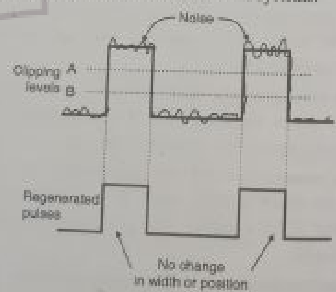
- The relation between the number of quantization levels Q and the number of bits per word (N) in the transmitted signal can be found as follows :
- Because each quantized level is to be converted into a unique N bit digital word, assuming a binary coded output signal.
- The number of quantization levels Q = Number of combinations of bits/word.

$$\therefore \quad Q = 2^N \qquad ...(4.10.4)$$

- Thus if N = 4 i.e. 4 bits per word then the number of quantization levels will be $2^4$ i.e. 16.

### 4.10.6 Effect of Noise on the PCM System :

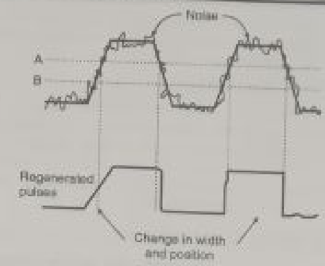
- Look at the two Figs. 4.10.6(a) and 4.10.6(b) which illustrate the effect of noise on the transmitted pulses.
- Consider Fig. 4.10.6(a) first. Due to the noise superimposed on the pulses, only the PAM system will be affected.
- However the PWM, PPM and PCM systems will remain unaffected. The regeneration of the pulses is achieved by using a clipper circuit with reference levels A and B.
- Now consider Fig. 4.10.6(b). Here the sides of the transmitted pulse are not perfectly vertical. In practice the transmitted pulses usually have slightly sloping sides (edges).
- As the noise is superimposed on them, the width and the position of the regenerated pulses is changed.
- Now this is going to distort the information contents in the PWM and PPM signals.
- But PCM is still unaffected as it does not contain any information in the width or the position of the pulses.
- Thus PCM has much better noise immunity as compared to PAM, PWM and PPM systems.




(a)

Fig. 4.10.6 (Contd...)






(b)

(L-229) Fig. 4.10.6 : Effect of noise on PCM

## 4.11 Advantages, Disadvantages, Applications and Modifications in PCM :

- The PCM is considered to be the best modulation scheme to transmit the voice and video signals.
- All the advantages of PCM are due to the fact that it uses coded pulses for the transmission of information.

### 4.11.1 Advantages of PCM :

1. Very high noise immunity (Noise does not affect the information content).
2. Due to digital nature of the signal, repeaters can be placed between the transmitter and the receivers. The repeaters actually regenerate the received PCM signal. This is not possible in analog systems. Repeaters further reduce the effect of noise.
3. It is possible to store the PCM signal due to its digital nature.
4. It is possible to use various coding techniques so that only the desired receiver can decode the received signal. This makes the communication secure.
5. The increased channel bandwidth requirement for PCM is balanced by the improved SNR. This is the due to the fact that PCM obeys an exponential law.
6. There is a **uniform format** used for the transmission of different types of base band signals. Hence it is easy to integrate all these signals together and send them on the common network.
7. It is easy to drop or reinsert the message sources in a PCM-TDM system.

### 4.11.2 Disadvantages of PCM :

1. The encoding, decoding and quantizing circuitry of PCM is complex.
2. PCM requires a large bandwidth as compared to the other systems.

### 4.11.3 Applications of PCM :

Some of the applications of PCM are as follows :

1. In telephony (with the advent of fibre optic cables).
2. In the space communication, space craft transmits signals to earth. Here the transmitted power is very low (10 to 15W) and the distances are huge (a few million km). Still due to the high noise immunity, only PCM systems can be used in such applications.

### 4.11.4 Modifications in PCM :

- Even though PCM is complex, it is possible to implement it using the VLSI technology.
- Due to the improvements in VLSI technology, the use of PCM for digital transmission of analog signals is going to increase.
- But if the simplicity is more important than the performance, then one should use the **Delta Modulation** in place of PCM.
- The requirement of large channel bandwidth for PCM is not a real problem now, due to the availability of wideband communication channels.
- As the problem of limitation on bandwidth has been solved, it has become possible to use the communication satellites and optical fiber communication.
- It is possible to remove the **redundancy** in PCM by using the **data compression techniques.** This will reduce the bit rate of transmitted data without any significant loss of quality in the contents.
- This will increase the complexity of PCM further.

**Why is PCM not used for broadcasting ?**

- In radio broadcasting a relatively large signal to noise ratio (typically of the order of 60 dB) is required.
- To get this level of $(S/N)_o$ the PCM with b > 8 is required, where $b = B_T / W$ i.e. ratio of transmission bandwidth to baseband bandwidth.
- However we can obtain the same performance with an FM system with b = 6 and with much simpler transmitter and receiver circuits.
- So higher bandwidth requirement and complicated circuitry are the disadvantages of PCM which does not make it suitable for the radio, TV broadcasting applications.

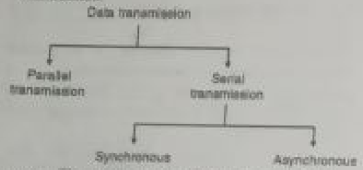
## 4.12 Data Transmission Modes :

- Data transmission means the movement of data which is in the form of bits between two or more digital devices. The data transmission takes place over some physical medium from one computer to the other.
- There are two ways of transmitting the digital data. They are :
  1. Parallel transmission 2. Serial transmission

### 4.12.1 Transmission Mode :

- Various modes of data transmission are shown in Fig. 4.12.1.

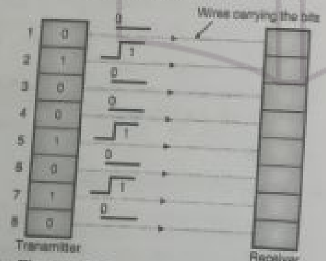
- As seen from Fig. 4.12.1, serial transmission and parallel transmission are the two basic types of transmission. The serial transmission is the most preferred mode of data transmission.
- The serial transmission is further classified into two types namely synchronous and asynchronous transmission.




(G-1039) Fig. 4.12.1 : Modes of data transmission

## 4.13 Parallel Transmission :

- In parallel transmission of data, all the bits of a byte are transmitted simultaneously on separate wires as shown in Fig. 4.13.1.
- This type of transmission requires multiple wires for interconnecting the two devices.
- Parallel transmission is possible practically only if the two devices are close to each other due to the length and the number of wires required.
- For example parallel transmission takes place between a computer and its printer.
- Fig. 4.13.1 shows the parallel transmission of an 8-bit digital data.
- This will require eight wires for connection between a transmitter and a receiver.
- With increase in the number of receivers, the number of wires will increase to an unmanageable number.




(G-1040) Fig. 4.13.1 : Parallel transmission of data

### 4.13.1 Advantages of Parallel Transmission :

The advantage of parallel transmission is that all the data bits will be transmitted simultaneously. Therefore the time required for the transmission of an N-bit word is only one clock cycle.

The serial transmission will require N number of clock cycles for the transmission of same word.

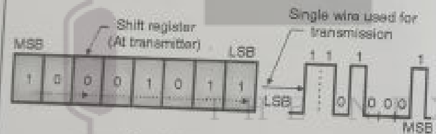
3. Due to this the clock frequency can be kept low without affecting the speed of operation. For serial transmission the clock frequency has to be high.

### 4.13.2 Disadvantages :

To transmit an N-bit word, we need N number of wires. With increase in the number of users, the number of wires increase and it becomes impossible to handle them. The serial transmission uses only one wire, for connecting the transmitter to the receiver. Hence practically the serial transmission is always preferred.

## 4.14 Serial Transmission :

- In serial transmission, the bits of a byte are serially transmitted one by one as shown in Fig. 4.14.1.
- The byte to be transmitted is first stored in a shift register. Then these bits are shifted from MSB to LSB bit by bit in synchronization with the clock. Bits are shifted right [see Fig. 4.14.1] by one position per clock cycle.
- The bit which falls out of the shift register is transmitted. Hence LSB is transmitted first and MSB is the last bit getting transmitted.
- For serial transmission only one wire is needed between the transmitter and the receiver. Hence serial transmission is preferred for long distance data communication. This is the advantage of serial transmission over parallel transmission.



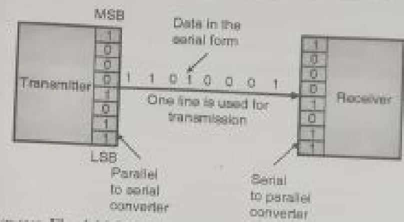

(G-1041) Fig. 4.14.1 : Serial transmission

- The serial transmission has a serious drawback. As only one bit is transmitted per clock cycle, it requires a time corresponding to 8-clock cycles to transmit one byte. (The parallel transmission needs only one clock cycle to transmit a byte). The time can be reduced by increasing the clock frequency.

### 4.14.1 Practical Serial Transmission System :

- Fig. 4.14.2 shows the practical serial transmission system. The transmitter and receiver both are computers.
- Since the communication within a computer is parallel, it is necessary to convert the parallel data into a serial one at the transmitter.
- At the receiver, the serial to parallel conversion is required to be performed as shown in Fig. 4.14.2.






(G-1042) Fig. 4.14.2 : Practical serial transmission system

**Advantages of serial transmission :**

1. Only one wire is required to be used.
2. Reduction in cost due to less number of conductors.

**Disadvantages :**

1. The speed of data transfer is low as only one bit is sent at a time.
2. To increase the speed of data transfer, it is necessary to increase the clock frequency.

**Application :**

It is used for computer to computer communication, specially long distance communication.

### 4.14.2 Comparison of Serial and Parallel Transmission :

| Sr. No. | Parameter | Parallel transmission | Serial transmission |
|---|---|---|---|
| 1. | Number of wires required to transmit N bits. | N wires | 1 wire |
| 2. | Number of bits transmitted simultaneously | N bits | 1 bit |
| 3. | Speed of data transfer | Fast | Slow |
| 4. | Cost | Higher due to more number of conductors | Low, since only one wire is used. |
| 5. | Application | Short distance communication such as computer to printer communication | Long distance computer to computer communication. |

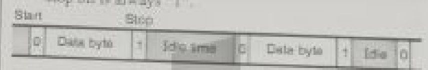
### 4.14.3 Types of Serial Transmission :

There are three types of serial transmission. They are :

1. Synchronous data transmission.
2. Asynchronous data transmission
3. Isosynchronous transmission

## 4.15 Asynchronous Transmission :

- In asynchronous transmission, the transmitter can begin the transmission of data bytes at any instant of time.
- Only one byte is sent at a time. After sending one byte the next byte can be sent after an arbitrary time delay as shown in Fig. 4.15.1.
- The transmitter and receiver can operate at different clock frequencies. There is no synchronization between them on this account.
- As the data transmission can commence at any instant, it becomes difficult for the receiver to understand the instant at which the byte has been transmitted.
- To help the receiver to receive the data bytes "start" and "stop" bits are used alongwith each data byte as shown in Fig. 4.15.1. The start bit is always "0" and stop bit is always "1".




(G-1043) Fig. 4.15.1 : Asynchronous transmission

- The idle time in between the adjacent data bytes is not constant. The idle time is also called as the gaps between the data bytes.
- In the asynchronous transmission the timing of the signal is not important, instead information is received and translated by the patterns which are agreed upon by the sender and receiver.
- As long as these patterns are being followed, the receiver can retrieve the information without any problem.

**Why is it called asynchronous ?**

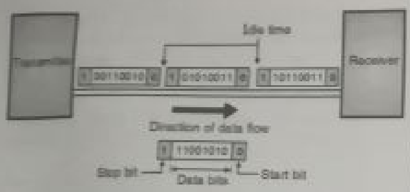
- This mechanism is called as asynchronous because at the byte level the sender and receiver do not have to be synchronized.
- However within each byte, the receiver should still be synchronized with the incoming bit stream.
- This means that some synchronization is required only for the duration of single byte.

**Response to the start and stop bits :**

- When the receiver detects a start bit, it will set a timer and begins counting bits as they come in.
- After "n" bits, the receiver searches for the stop bit.
- As soon as it detects the stop bit, it will wait until it detects the next start bit.
- So the meaning of asynchronous is actually asynchronous at the byte level but the bits are still synchronized. So their durations are same.

### 4.15.1 Block Diagram of Asynchronous Transmission :

- Fig. 4.15.2 shows the block diagram of asynchronous transmission.
- The start bits are 0 and stop bits are 1, as shown in Fig. 4.15.2.




(G-1044) Fig. 4.15.2 : Asynchronous transmission

The use of start and stop bits and the gaps (idle time) between adjacent data units will make the asynchronous transmission slow. This is the major disadvantage of using the start and stop bits.

### 4.15.2 Disadvantages of Asynchronous Transmission :

1. Additional bits called start and stop bits are required to be used.
2. It is difficult to determine the sampling instants hence the timing error can take place.
3. The start/stop bits and idle time makes the asynchronous transmission slow.

### 4.15.3 Advantages of Asynchronous Transmission :

1. Synchronization between the transmitter and receiver is not necessary.
2. It is possible to transmit signals from the sources having different bit rates.
3. The transmission can commence as soon as the data byte to be transmitted becomes available.
4. This mode of transmission is easy to implement.
5. It is a cheap scheme.
6. It is an effective scheme.

### 4.15.4 Application of Asynchronous Transmission :

The connection of a keyboard to a computer is an example of asynchronous transmission.

## 4.16 Synchronous Transmission :

- Synchronous transmission is carried out under the control of a common master clock. Here the bits which are being transmitted are synchronized to a reference clock.
- No start and stop bits are used instead the bytes are transmitted as a block in a continuous stream of bits as shown in Fig. 4.16.1. There is an inter block idle time which is filled with idle characters.
- The receivers operates at exactly the same clock frequency as that of transmitter as both are synchronized with each other.
- This is essential for error free reception of data. Flag is a sequence of fixed number of bits which is prefixed to each block as shown in Fig. 4.16.1. Flag is useful in identifying the beginning of a block.


(G-1045) Fig. 4.16.1 : Synchronous transmission

- In the synchronous transmission the bit stream to be transmitted is combined into longer "frames". A frame would contain more than one bytes.
- There is no gap between the successive frames. The receiver separates the bit stream into bytes for the purpose of decoding.
- Start and stop bits are not used. Instead bits are transmitted serially one after the other.
- The grouping of these bits is responsibility of the receiver.

### 4.16.1 Block Diagram of Synchronous Transmission :

- Fig. 4.16.2 shows the block schematic of synchronous transmission. Note the absence of gaps and start stop bits.


(G-1046) Fig. 4.16.2 : Synchronous transmission

### 4.16.2 Advantages :

1. The speed of transmission is much higher than that of asynchronous transmission. This is due to the absence of gaps between the data units and absence of start stop bits.
2. Start and stop bits are not needed any more.
3. Timing errors are reduced due to synchronization.

### 4.16.3 Disadvantages :

1. The timing is very important. The accuracy of the received data is dependent entirely on the ability of the receiver to count the received bits accurately.
2. The transmitter and receiver have to operate at the same clock frequency. This requires proper synchronization which makes the system complicated.



### 4.16.4 Application of Synchronous Transmission :

The synchronous transmission, due to its high speed is used for the data exchange from one computer to the other.

### 4.16.5 Synchronization :

The byte synchronization is achieved at the data link layer level for the synchronous transmission between computers.

### 4.16.6 Comparison of Synchronous and Asynchronous Transmission :

Table 4.16.1

| Sr. No. | Parameter | Asynchronous transmission | Synchronous transmission |
|---|---|---|---|
| 1. | Synchronization | Not needed | Needed |
| 2. | Start and Stop bits | Used | Not used |
| 3. | Gaps between data blocks | Present | Absent |
| 4. | Speed | Low | High |
| 5. | Application | Communication between a computer and keyboard. | Communication between two computers. |

## 4.17 Isochronous :

- This is the third type of serial transmission. In the real time streaming of audio and video, the time delay introduced during the transmission must remain constant. An uneven time delay would introduce distortion.
- For such applications we cannot use the synchronous serial transmission successfully. The isosynchronous transmission can be used for such applications.

## 4.18 Digital to Analog Conversion :

- In the process of D to A conversion the digital data at the input is converted into an analog signals. These analog signals are transmitted over the transmission medium.
- The most familiar application of D to A conversion is for transmitting digital data through the public telephone network.
- The D to A conversion is done by the modems to convert the digital data from the computers into the analog signals that are sent on the telephone lines for the Internet.


(L-791) Fig. 4.18.1 : Digital data to analog signal

### 4.18.1 Aspects of Digital to Analog Conversion :

- The two most important aspects related to D to A conversion are as stated below :
  1. Data element versus signal element
  2. Data rate versus signal rate.

1. **Data element versus signal element :**
   - We may define data element as the smallest piece of information that can be exchanged and as we know it is a "bit".
   - The signal element is classically defined as the smallest unit of a signal that is constant. This definition is true in the digital context. But the signal will be analog here hence the nature of the signal element is slightly different than that for the digital transmission.
2. **Data rate versus signal rate :**
   - We have defined the data rate (bit rate) and signal rate (baud rate) earlier.
   - The relation between them is as follows :

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

Where N = Data rate (bps)

r = Number of data elements in one signal element

   - The value of "r" in the analog transmission is as follows :

$$r = \log_2 L$$

Where L is the type of signal element not the level.

**Note :**
- Bit rate = Number of bits per second.
- Baud rate = Number of signal elements or symbols per second.
- In analog communication, of digital data the bit rate (bps) is always greater than or equal to the baud rate (bauds).

**Ex. 4.18.1 :** An analog signal carries 4 bits per signal element. If number of signal elements sent per second is 500 calculate the bit rate.

**Soln. :**

**Given :** $r = 4, S = 500, N = ?$

We know that $S = N \times \frac{1}{r}$

$\therefore \quad 500 = N \times \frac{1}{4}$

$\therefore \quad N = 2000 \text{ bps}$ ...Ans.

**Bandwidth :**

The bandwidth requirement for analog transmission of digital data is proportional to the signal rate i.e. the baud rate. But this is not true for FSK system, this has been discussed later.

**Carrier signal :**

- In the D to A conversion, at the sending end a high frequency signal which acts as the base signal for transmission of information is produced. This signal is known as the carrier signal or carrier frequency.
- The input digital signal (which is the information signal) will change one of the characteristics of this carrier such as amplitude, frequency or phase.
- This type of modification or modulation is known as **shift keying**. Depending on which parameter of the carrier is being modified we get Amplitude Shift Keying (ASK), frequency shift keying (FSK) or Phase Shift Keying (PSK).

### 4.18.2 Need of Digital Continuous Wave Modulation :

- PCM converts analog message signal into a digital signal. Now we will learn some techniques which convert the digital message signal into an analog signal and then transmits it.
- Such modulation schemes are called as digital carrier modulation schemes.
- This type of digital to analog conversion is essential when the digital message signal is to be sent over a bandlimited channel such as the telephone line.
- The best application of digital carrier modulation is MODEM.
- The modem will modulate the digital data signal from the DTE (computer) into an analog signal.
- This analog signal is then transmitted on the telephone lines.


(L-61) **Fig. 4.18.2 : Digital carrier modulation**

- The question is why can't we send the digital signal as it is on the telephone lines ? Why should we modulate it ?
- Here is the answer for it. The digital data consists of binary 0s and 1s, therefore the waveform changes its value abruptly from high to low or low to high.
- In order to carry such a signal without any distortion being introduced, the communication medium needs to have a large bandwidth.
- Unfortunately the telephone lines do not have high bandwidth. Therefore we have to convert the digital signal first into an analog signal which needs lower bandwidth by means of the modulation process.

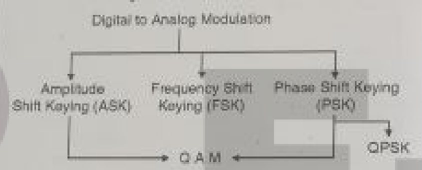
### 4.18.3 Types of Digital Carrier Modulation :

- There are three basic types of modulation techniques for the transmission of digital signals.
- These methods are based on the three characteristics of a sinusoidal signal; amplitude, frequency and phase.

The corresponding modulation methods are then called as :

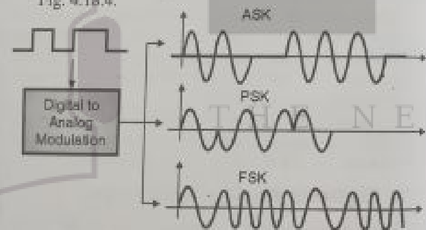
1. Amplitude shift keying (ASK)
2. Frequency shift keying (FSK)
3. Phase shift keying (PSK)
4. Quadrature phase shift keying (QPSK) or 4-psk.
5. Quadrature amplitude modulation (QAM).

- QPSK is a multilevel modulation in which four phase shifts are used for representing four different symbols.
- At high bit rates, a combination of ASK and PSK is employed in order to minimize the errors in the received data.
- This method is known as "Quadrature Amplitude Modulation (QAM)". Let us discuss these methods one by one.
- Fig. 4.18.3 shows the classification of digital to analog modulation systems.




(L-62) **Fig. 4.18.3 : Types of digital to analog modulation**

- Digital to analog modulation is demonstrated in Fig. 4.18.4.




(L-63) **Fig. 4.18.4 : Digital to analog modulation**

### 4.18.4 Advantages and Disadvantages of CW Modulation :

1. The advantage of CW modulation techniques such as ASK, PSK, FSK etc. used for transmission of data is that we can use the telephone lines for transmission of high speed data. Due to the use of CW modulation the BW requirement is reduced.
2. The disadvantage of CW modulation is we need to use a MODEM alongwith every computer. This makes the system costly and complex.

## 4.19 Amplitude Shift Keying (ASK) or Digital Amplitude Modulation :

**Definition :**

ASK is the digital carrier modulation in which the



2π ωct

amplitude of the sinusoidal carrier will take one of the two predetermined values in response to 0 or 1 value of digital input signal.

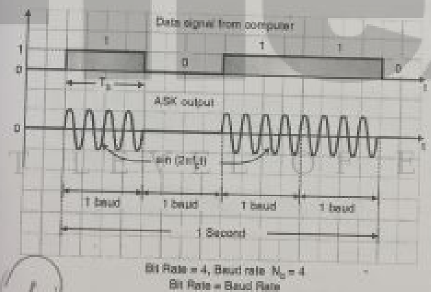
**Generation and waveforms :**

- Amplitude shift keying (ASK) is the simplest type of digital CW modulation. Here the carrier is a sinewave of frequency $f_c$. We can represent the carrier signal mathematically as follows :

$$e_c = \sin(2\pi f_c t) \quad ...(4.19.1)$$

- The digital signal from the computer is a unipolar NRZ signal which acts as the modulating signal. The ASK modulator is nothing but a multiplier followed by a band pass filter as shown in Fig. 4.19.1(a).
- Due to the multiplication, the ASK output will be present only when a binary "1" is to be transmitted.
- The ASK output corresponding to a binary "0" is zero as shown in Fig. 4.19.1(b).


(L-64) **Fig. 4.19.1(a) : ASK generator**




(L-65) **Fig. 4.19.1(b) : ASK waveforms**

- From the waveforms of Fig. 4.19.1(b) we can conclude that the carrier is transmitted when a binary 1 is to be sent and no carrier is transmitted when a binary 0 is to be sent.
- The ASK signal can be mathematically expressed as follows :

$$V_{ASK}(t) = d \sin(2\pi f_c t) \quad ...(4.19.2)$$

where d = Data bit which can take values 1 or 0.

$$\therefore \left.\begin{aligned} V_{ASK}(t) &= \sin(2\pi f_c t) \quad \text{when } d = 1 \\ \text{and} \quad V_{ASK}(t) &= 0 \quad \text{when } d = 0 \end{aligned}\right\} \quad ...(4.19.3)$$

### 4.19.1 Baud Rate ($N_b$) : (symbol)

- For ASK we use 1 bit (0 or 1) to represent one symbol. So the rate of symbol transmission i.e. the baud rate will be same as the bit rate.
- $N_b$ will be same as bit rate R as shown in Fig. 4.19.1(b).

$\therefore$ Baud rate = Bit rate   baud rate

$N_b = f_b$   no of sig state changed

### 4.19.2 Transmission Bandwidth of the ASK Signal :

- The bandwidth of ASK signal is dependent on the bit rate $f_b$. Where bit rate $f_b = 1/T_b$ as shown in Fig. 4.19.1(a). For a bit rate of "$f_b$" bits/sec. the maximum bandwidth required for an ASK signal is

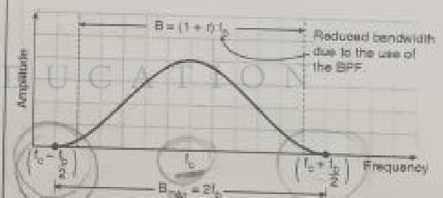
$$BW_{(max)} = f_b \text{ Hz} \quad ...(4.19.4)$$

- The frequency spectrum of an ASK signal is shown in Fig. 4.19.1(c) which shows that the spectrum consists of the carrier frequency $f_c$ with upper and lower sidebands.
- The transmission bandwidth BW of the ASK signal can be restricted by using a filter. The restricted value of bandwidth is given as :

$$BW = (1 + r) f_b \quad ...(4.19.5)$$

where "r" is a factor related to the filter characteristics and its value lies between 0 and 1.

- $f_c$ is the carrier frequency i.e. frequency of the sine wave being transmitted.




(L-66) **Fig. 4.19.1(c) : Frequency spectrum of an ASK signal**

### 4.19.3 Bandwidth of ASK in Terms of Baud Rate :

- For ASK, as shown in Fig. 4.19.1(b), the baud rate = Bit rate.
- The bandwidth of ASK in terms of bit rate is given by,

$$BW = f_c + \left(\frac{f_b}{2}\right) - \left[f_c - \left(\frac{f_b}{2}\right)\right] = f_b$$

Where $f_b = \frac{1}{T_b}$ = Bit rate and $T_b$ = One bit interval.

- Since bit rate and baud rate are equal for ASK, the expression for bandwidth is given by

$$BW = f_c + \frac{N_b}{2} - \left[ f_c - \frac{N_b}{2} \right]$$

Where $N_b$ = Baud rate = $f_b$ i.e. Bit rate

$\therefore$ Bandwidth = $N_b$

– But practically the bandwidth requirement of ASK is given by,

$$BW = (1 + d) \times N_b$$

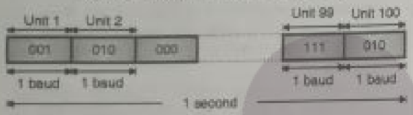
where d is related to modulation process.

**Ex. 4.19.1 :** An analog signal carries 3 bits in each signal unit. If the transmitter send 100 such units per second, calculate the bit rate and baud rate.

**Soln. :**

**Given :** Number of bits per signal unit = 3,

Number of signal units per second = 100




(L-67) Fig. P. 4.19.1

1. Bit rate = Number of bits per second = Number of bits/unit × Number of units / sec.

= 3 × 100 = 300 bits/sec. ...Ans.

2. Baud rate = Number of signal units per second

= 100 bauds/sec. ...Ans.

**Ex. 4.19.2 :** An ASK transmitter transmits 5000 bits per second. Calculate the minimum bandwidth. Assume the transmission to be half duplex.

**Soln. :**

**Given :** Bit rate = 5000 bps, Mode : Half duplex

**To find :** Minimum bandwidth

– The minimum bandwidth of ASK is given by,

$$BW_{(min)} = N_b$$

– For ASK, baud rate $N_b$ is equal to bit rate

$$\therefore BW_{(min)} = 5000 \text{ Hz} \quad \text{...Ans.}$$

**Ex. 4.19.3 :** The bandwidth of an ASK system is 3000 Hz. Calculate the bit rate and baud rate.

**Soln. :**

1. For an ASK system, the baud rate is equal to the bandwidth,

$\therefore$ Baud rate $N_b$ = 3000 bauds per sec. ...Ans.

2. For ASK, bit rate is equal to baud rate.

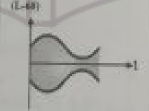
$\therefore$ Bit rate = 3000 bits/sec. ...Ans.

### 4.19.4 Merits and Demerits of ASK :

The advantage of using ASK is its simplicity. It is easy to generate and detect. However its disadvantage is that it is very sensitive to noise, therefore it finds limited application in data transmission. It is used at very low bit rates, upto 100 bits/sec.

### 4.19.5 Comparison of AM and ASK :

Table 4.19.1 : Comparison of AM and ASK

| Sr. No. | Parameter | AM | ASK |
|---|---|---|---|
| 1. | Variable characteristics of the carrier. | Amplitude | Amplitude |
| 2. | Nature of modulating signal. | Modulating signal is analog | Modulating signal is digital |
| 3. | Modulated signal shape. | (L-68)  | (L-68)   |
| 4. | Variation in the carrier amplitude. | Continuous variation in accordance with the amplitude of modulating signal. | Carrier ON or OFF depending on whether a 1 or 0 is to be transmitted. |
| 5. | Number of sidebands produced. | Two | Two |
| 6. | Bandwidth | $2 f_m$ | $(1 - r) f_b$ |
| 7. | Noise immunity | Poor | Poor |
| 8. | Application | Radio broadcasting | Data transmission at low bit rate |
| 9. | Detection method | Envelope | Envelope |



### 4.19.6 Multilevel ASK :

– In the above discussion, there were only two amplitude levels of the input digital data involved. But we can have multilevel ASK in which the digital data can have multiple amplitude levels (e.g. 4, 8, 16 levels etc.). This is possible if we consider a group of consecutive data bits as one symbol.

– 4 levels correspond to 2 bits at a time. Similarly 8 and 16 levels correspond to 3 and 4 bits at a time. Thus multiple bits/symbol would result in multilevel ASK.

– This principle is actually not implemented with pure ASK but it is implemented with QAM.

**Ex. 4.19.4 :** Given bandwidth of 10,000 Hz (100 to 11000 Hz), draw the full duplex ASK diagram of the system. Find the carriers and the bandwidths in each direction. Assume there is no gap between the bands in the two directions.

**Soln. :**

**Given :** System : ASK mode : Full duplex

BW = 10,000 Hz No gap between bands

**Step 1 : Calculate BW in each direction :**

For a full duplex ASK, the bandwidth in each direction is given by,

$$BW = \frac{10,000}{2} = 5000 \text{ Hz.}$$

**Step 2 : Carrier frequencies in each direction :**

– The frequency bands allocated for backward and forward transmission are : 1000 Hz to 6000 Hz and 6000 Hz to 11000 Hz respectively.

– The carrier frequencies are placed at the center of each band.

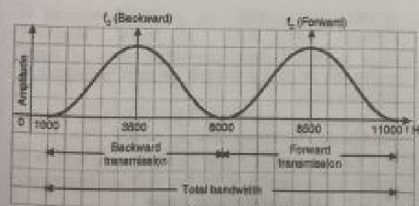
$$\therefore f_c \text{ (backward)} = 1000 + \frac{5000}{2} = 3500 \text{ Hz}$$

and

$$f_c \text{ (forward)} = 6000 + \frac{5000}{2} = 8500 \text{ Hz}$$

**Step 3 : Draw the frequency spectrum :**

– The required frequency spectrum is shown in Fig. P. 4.19.4.




(L-790) Fig. P. 4.19.4 : Frequency spectrum of a duplex ASK system

## 4.20 Frequency Shift Keying (FSK) :

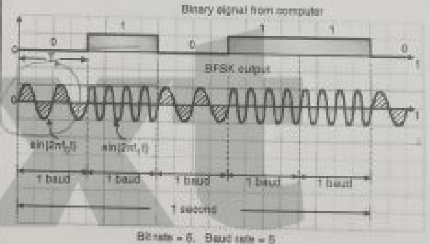
**Definition and waveforms :**

– In "frequency shift keying (FSK)", the frequency of a sinusoidal carrier is shifted between two discrete values, in response to the value (0 or 1) of the digital input signal.

– One of these frequencies ($f_1$) represents a binary "1" and the other value ($f_0$) represents a binary "0".

– The representation of digital data using FSK is as shown in Fig. 4.20.1(b). Note that there is no change in the amplitude of the carrier.


(L-785) Fig. 4.20.1(a) : FSK generation




(L-72) Fig. 4.20.1(b) : Representation of digital signal using FSK

### 4.20.1 FSK Generation :

– Refer to the FSK generator shown in Fig. 4.20.1(a). It is basically a voltage controlled oscillator (VCO) which produce sinewaves at frequencies $f_1$ and $f_0$, respectively.

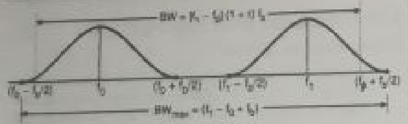
– Corresponding to binary 0 input, the VCO produces a sinewave of frequency $f_0$ whereas corresponding to binary 1 input, the VCO produces a sinewave of frequency $f_1$. ($f_1 > f_0$).

– Thus we obtain the binary FSK (BFSK) signal at the output of VCO corresponding to the input digital data bits.

### 4.20.2 Frequency Spectrum of Binary FSK Signal :

– The FSK signal can be considered to be containing two ASK signals, with the carrier frequencies $f_1$ and $f_0$.

– Therefore the frequency spectrum of the FSK signal is as shown in Fig. 4.20.1(c) which is identical to that of ASK for each of the two frequencies.




(L-793) Fig. 4.20.1(c) : Frequency spectrum of a binary FSK signal

### 4.20.3 Bandwidth of FSK Signal :

- The bandwidth of FSK signal is dependent on the pulse width $T_s$ or bit rate $f_b = 1/T_s$ and the separation between the frequencies $f_0$ and $f_1$, as shown in Fig. 4.20.1(c).
- The maximum bandwidth of FSK system is given by,

$$BW_{max} = \left(f_1 + \frac{f_b}{2}\right) - \left(f_0 - \frac{f_b}{2}\right)$$
$$= (f_1 - f_0 + f_b) \quad ...(4.20.1)$$

- The bandwidth can be restricted by using a bandpass filter after the VCO in the FSK generator. The restricted bandwidth is given as :

$$BW = |f_1 - f_0| + (1 + r)\frac{f_b}{2} \quad ...(4.20.2)$$

Where "r" is the factor related to the filter characteristics and its value lies between 0 and 1.

- The separation between $f_1$ and $f_0$ is kept at least $2 f_b/3$. Substitute this value in Equation (4.20.2) to get

$$BW_{max} = \frac{2}{3} f_b + f_b = \frac{5 f_b}{3} \quad ...(4.20.3)$$

- This shows that FSK requires larger bandwidth than ASK and PSK (to be discussed next).

### 4.20.4 Bandwidth for FSK in terms of Baud Rate :

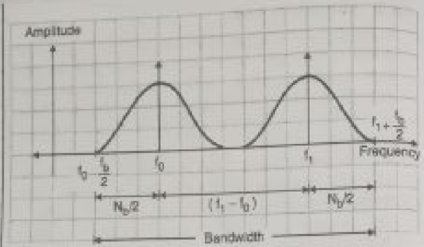
- For FSK also bit rate is equal to baud rate. This is due to the fact that each data bit at the input is treated as a separate symbol.
- We can imagine the FSK spectrum to be a combination of two ASK spectrums centered at frequencies $f_1$ and $f_0$ as shown in Fig. 4.20.1(d).
- From Fig. 4.20.1(d) the expression for bandwidth is given by

$$BW = \frac{N_b}{2} + (f_1 - f_0) + \frac{N_b}{2}$$
$$= (f_1 - f_0) + N_b \quad ...(4.20.4)$$

Where $N_b$ = Baud rate = Bit rate = $f_b$

- Minimum bandwidth will correspond to the situation in which $(f_1 - f_0) = N_b$

$$\therefore BW_{(min)} = N_b + N_b = 2 N_b = 2 f_b \quad ...(4.20.5)$$




(L-74) Fig. 4.20.1(d) : Spectrum of FSK

**Ex. 4.20.1 :** Calculate the bandwidth of an FSK system in which, the transmission takes place at 4000 bits per second rate and the frequency difference between the two carriers is 3000 Hz.

**Soln. :**

**Given :** $(f_1 - f_0)$ = 3000 Hz, Bit rate = 4000 bps.

**To find :** Bandwidth

- Bandwidth = $(f_1 - f_0) + N_b$
- But $N_b$ = baud rate = bit rate = 4000

$\therefore$ BW = 3000 + 4000 = 7000 Hz ...Ans.

**Ex. 4.20.2 :** For a half duplex FSK transmission, the bandwidth of medium is 8000 Hz. If the frequency difference between the two carriers is 4000 Hz calculate the maximum bit rate.

**Soln. :**

**Given :** FSK, half duplex, BW = 8000 Hz, $f_1 - f_0$ = 4000 Hz.

**To find :** Maximum bit rate

1. BW = $(f_1 - f_0) + N_b$

$\therefore$ 8000 = 4000 + $N_b$

$\therefore$ $N_b$ = 4000 bauds/sec.

2. For FSK system baud rate is equal to bit rate.

$\therefore$ Bit rate = 4000 bits per second ...Ans.

### 4.20.5 Multilevel FSK (MFSK) :

- In BFSK (Binary FSK) we use two frequencies to represent two levels of signal amplitude (0 or 1). But if there are multiple levels of signal amplitude, then it is possible to use more frequencies to represent them.
- As discussed in multilevel ASK, even here we use multiple data bits to represent one symbol.
- If 2 bits are used at a time then there will be $2^2 = 4$ levels. So we have to assign four different frequencies $(f_0, f_1, f_2, f_3)$ to represent these levels.
- The bandwidth requirement of MFSK is higher than that of BFSK.



### 4.20.6 Advantages of FSK :

1. FSK is relatively easy to implement.
2. It has better noise immunity than ASK. Therefore the probability of error free reception of data is high.

### 4.20.7 Disadvantages of FSK :

1. The major disadvantage is its high bandwidth requirement as discussed earlier.
2. Therefore FSK is extensively used in low speed modems having bit rates below 1200 bits/sec.
3. The FSK is not preferred for the high speed modems because with increase in speed, the bit rate increases.
4. This increases the channel bandwidth required to transmit the FSK signal.
5. As the telephone lines have a very low bandwidth, it is not possible to satisfy the bandwidth requirement of FSK at higher speed. Therefore FSK is preferred only for the low speed modems.

## 4.21 Phase Shift Keying (PSK) :

**Definition and waveforms :**

- Phase shift keying (PSK) is the most efficient of the three modulation methods.
- Therefore it is used for high bit rates. In PSK, phase of the sinusoidal carrier is changed according to the data bit to be transmitted.
- Fig. 4.21.1(a) shows the simplest form of PSK called Binary PSK (BPSK). The carrier phase is changed between 0° and 180° by the bipolar digital signal. A bipolar NRZ signal is used to represent the digital data from the DTE.
- The BPSK signal can be represented mathematically as :

$V_{BPSK}(t) = \sin(2\pi f_c t)$ when binary "0" is to be represented

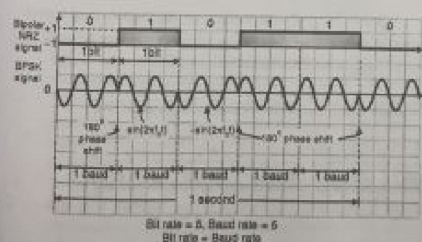
and $V_{BPSK}(t) = -\sin(2\pi f_c t)$

$= \sin(2\pi f_c t + \pi)$ when binary "1" is to be represented.

- Combining the two conditions we can write

$$V_{BPSK}(t) = d \sin(2\pi f_c t) \quad ...(4.21.1)$$

where d = ± 1




(L-80) Fig. 4.21.1(a) : Binary phase shift keying (BPSK)

### 4.21.1 BPSK Generation :

- The BPSK generation takes place as shown in Fig. 4.21.1(b).

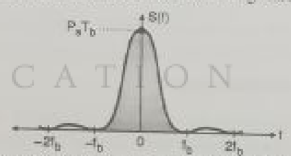

(L-81) Fig. 4.21.1(b) : BPSK generation

- The binary data signal (0s and 1s) is converted into a NRZ bipolar signal by an NRZ encoder, which is then applied to a multiplier (balanced modulator). The other input to the multiplier is the carrier signal $(2\pi f_c t)$.
- The data bits 0s and 1s are converted into a bipolar NRZ signal "d" as shown in the following table.

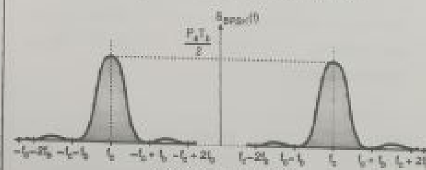
| Digital signal | Bipolar NRZ signal | BPSK output |
|---|---|---|
| Binary 0 | d = 1 | $V_{BPSK}(t) = \sin(2\pi f_c t)$ |
| Binary 1 | d = – 1 | $V_{BPSK}(t) = -\sin(2\pi f_c t)$ |

### 4.21.2 Spectrum of BPSK :

- The spectrum of BPSK is as shown in Fig. 4.21.2.




(a) Power spectral density of the NRZ data b(t)




(b) Spectrum of BPSK

(L-48) Fig. 4.21.2 : Spectrum of BPSK

### 4.21.3 Bandwidth of BPSK ::

- From the frequency spectrum of BPSK signal, shown in Fig. 4.21.2(b), we can come to a conclusion that the bandwidth of a BPSK signal is given by.

$$BW = \text{Highest frequency} - \text{Lowest frequency in main lobe} = (f_c + f_b) - (f_c - f_b)$$

$$\therefore \quad BW = 2 f_b \qquad ...(4.21.2)$$

where $\quad f_b = 1 / T_b$

- Thus the minimum bandwidth of BPSK signal is equal to twice the highest frequency contained in the baseband signal.

**Baud rate :**

In BPSK also each digit (0 or 1) of the input digital data represents a symbol. Hence symbol rate is equal to bit rate.

$$\therefore \quad \text{Baud rate } N_b = \text{Bit rate } f_b$$

$$\therefore \quad BW = 2 N_b$$

### 4.21.4 Advantages of BPSK :

1. BPSK has a bandwidth which is lower than that of a BFSK signal.
2. BPSK has the best performance of all the systems in presence of noise. It gives the minimum possibility of error.
3. BPSK has a very good noise immunity.

### 4.21.5 Disadvantage of BPSK :

The only disadvantage of BPSK is that generation and detection of BPSK is not easy. It is quite complicated.

### 4.21.6 Applications :

- Phase shift keying is the most efficient of the three modulation methods and it is used for high bit rates even higher than 1800 bits/sec.
- Due to low bandwidth requirement the BPSK modems are preferred over the FSK modems, at higher operating speeds.

### 4.21.7 Comparison of Binary Modulation Systems :

| Sr. No. | Parameter | Binary ASK | Binary FSK | Binary PSK |
|---|---|---|---|---|
| 1. | Variable characteristic. | Amplitude | Frequency | Phase |
| 2. | Bandwidth (Hz) | 2R | $\lvert f_1 - f_0 \rvert + (1 + r) R$ | $(1 + r) R$ |
| 3. | Noise immunity. | Low | High | High |
| 4. | Error probability | High | Low | Low |
| 5. | Performance in presence of noise. | Poor | Better than ASK | Better than FSK |
| 6. | Complexity | Simple | Moderately complex | Very complex |
| 7. | Bit rate | Suitable upto 100 bits/sec. | Suitable upto about 1200 bits/sec. | Suitable for high bit rates. |
| 8. | Detection method. | Envelope | Envelope | Coherent |

**Ex. 4.21.1 :** If the data bit sequence consists of the following string of bits, what will be the nature of waveform transmitted by BPSK transmitter ?

The data bit sequence is 1 0 1 1 1 0 1 0.

**Soln. :**

The BPSK signal can also be expressed in terms of cosine wave as :

$$V_{BPSK}(t) = \sqrt{2 P_s}\, b(t) \cos \omega_c t$$

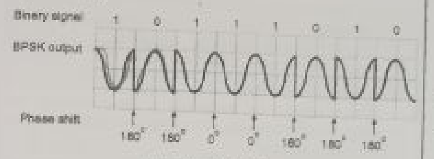
where $b(t) = \pm 1$ depending on the digital input signal. Table P. 4.21.1 lists the values of b (t) and the transmitted signal $V_{BPSK}$ for different bit intervals.

Table P. 4.21.1

| Binary signal | 1 | 0 | 1 | 1 | 1 | 0 | 1 | 0 |
|---|---|---|---|---|---|---|---|---|
| b (t) | +1 | −1 | +1 | +1 | +1 | −1 | +1 | −1 |
| $V_{BPSK}(t)$ | $\cos \omega_c t$ | $-\cos \omega_c t$ | $\cos \omega_c t$ | $-\cos \omega_c t$ | $\cos \omega_c t$ | $-\cos \omega_c t$ | $\cos \omega_c t$ | $-\cos \omega_c t$ |



The transmitted BPSK signal is as shown in Fig. P. 4.21.1.




(L-89) Fig. P. 4.21.1

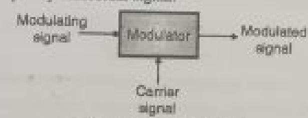
## 4.22 Analog to Analog Conversion :

- In some applications we have to transform analog data such as voice, video etc. into analog signal.
- This process is known as modulation. The analog data at the input is called as modulating signal. It modulates a high frequency sinusoidal signal called carrier to produce another analog signal called modulated signal.


(L-31) Fig. 4.22.1 : Transformation from analog data to analog signals

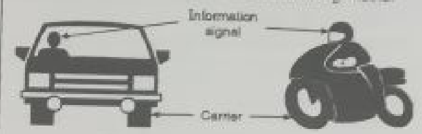
## 4.23 Modulation :

- In the **Modulation** process, two signals are used namely the **modulating signal** and the **carrier**.
- The modulating signal is nothing but the baseband signal or information signal while carrier is a high frequency sinusoidal signal.




(D-8) Fig. 4.23.1 : Modulation

- In the modulation process some parameter of the carrier wave (such as amplitude, frequency or phase) is varied in proportion with the modulating signal.
- The result of this process is called as the modulated signal. This modulated signal is then transmitted by the transmitter over a communication channel or medium.
- The receiver will **"Demodulate"** the received modulated signal and get the original information signal back. Thus demodulation is exactly opposite to modulation.
- In the process of modulation, the carrier wave actually acts as a **carrier** which carries the information signal (modulating signal) from the transmitter to receiver.
- This is similar to a situation in which a person travels in his car or on his bike from one place to the other. The person can be viewed as the modulating signal and the car or bike as the carrier as shown in Fig. 4.23.2.




(D-9) Fig. 4.23.2 : Concept of modulation

### 4.23.1 Need of Modulation :

- A question may be asked as, when the baseband signals can be transmitted directly why to use the modulation ?
- The answer is that the baseband transmission has many limitations which can be overcome using modulation. It is as explained below.
- In the process of modulation, the baseband signal is "translated" i.e. shifted from low frequency side to high frequency side of the frequency spectrum.
- This frequency shift is proportional to the frequency of carrier. The modulation process has the following advantages :

**Advantages of (Reasons for) modulation :**

1. Reduction in the height of antenna
2. Avoids mixing of signals
3. Increases the range of communication
4. Multiplexing becomes possible
5. Improves quality of reception.

**Reduction in height of antenna :**

- For transmission of radio signals, the antenna height must be a multiple of ($\lambda$/4). Here $\lambda$ is the wavelength. $\lambda = c/f$ where c is velocity of light and f is the frequency of the signal to be transmitted.
- The minimum antenna height required to transmit a baseband signal of f = 10 kHz is calculated as follows :

$$\text{Minimum antenna height} = \frac{\lambda}{4} = \frac{c}{4f}$$

$$= \frac{3 \times 10^8}{4 \times 10 \times 10^3}$$

$$= 7500 \text{ meters i.e. } 7.5 \text{ km}$$

The antenna of this height is practically impossible to install.

- Now consider a modulated signal at f = 1 MHz. The minimum antenna height is given by,

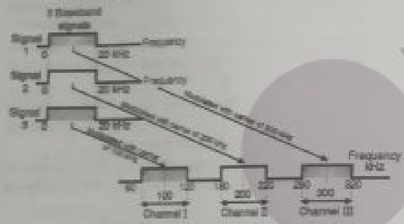
$$\text{Minimum antenna height} = \frac{\lambda}{4} = \frac{c}{4f} = \frac{3 \times 10^8}{4 \times 10 \times 10^6}$$

= 75 meters

- This antenna can be easily installed practically. Thus modulation reduces the height of the antenna.

**Avoids mixing of signals :**

- If the baseband sound signals are transmitted without using the modulation by more than one transmitter, then all the transmitted signal by multiple transmitters will be in the same frequency range i.e. 0 to 20 kHz.
- Therefore the signals from different stations get mixed together and a receiver cannot separate them from each other.
- So if each baseband sound signal is used to modulate a different carrier which corresponds to a different station then they will occupy different slots in the frequency spectrum (different channels).
- This is as shown in Fig. 4.23.3. Thus modulation avoids mixing of signals.




(D-18) Fig. 4.23.3 : Modulation avoids mixing of signals

**Increases the range of communication :**

- The frequency of baseband signals is low, and the low frequency signals can not travel a long distance when they are transmitted. They get attenuated (suppressed) quickly.
- The attenuation reduces with increase in frequency of the transmitted signals, and they travel longer distance.
- The modulation process increases the frequency of the signal. Hence it increases the range of communication.

**Multiplexing becomes possible :**

- Multiplexing is a process in which two or more signals can be transmitted over the same communication channel simultaneously.
- This is possible only with modulation. The multiplexing allows the same channel to be used by many signals.
- So many TV channels can use the same frequency range, without getting mixed with each other. OR different frequency signals can be transmitted at the same time.

**Improves quality of reception** :With frequency modulation (FM), and the digital communication techniques like PCM, the effect of noise is reduced to a great extent. This improves quality of reception.

### 4.23.2 Demodulation or Detection :

- The modulated signals are transmitted by the transmitter via air medium or wire medium. These signals then reach the receivers by travelling over the communication medium.
- At the receiver, the original information signal is separated from the carrier. This process is called as demodulation or detection. Detection is exactly the opposite process of modulation.

**Frequency translation :**

The frequency up conversion at the transmitter and frequency down conversion at the receiver together is called as frequency translation.

### 4.23.3 Types of Analog to Analog Conversion :

The three basic types of analog to analog conversion are shown in Fig. 4.23.4. They are AM, FM and PM.


(L-780) Fig. 4.23.4 : Types of analog to analog conversion

## 4.24 Amplitude Modulation (AM) :

**Definition :**

**Amplitude modulation (AM)** or Amplitude Modulation with Full Carrier (AM-FC) is the process of changing the amplitude of a high frequency sinusoidal carrier signal in proportion with the instantaneous value of modulating signal.

- Fig. 4.24.1 shows the amplitude modulated wave when the modulating signal is a sinusoidal signal.

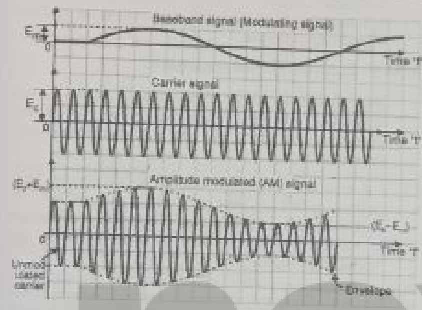
**Observations :**

1. The frequency of the sinusoidal carrier is much higher than that of the modulating signal.
2. In AM the instantaneous amplitude of the sinusoidal high frequency carrier is changed in proportion to the instantaneous amplitude of the modulating signal. This is the principle of AM.
3. The time domain display of AM signal is as shown in Fig. 4.24.1. This AM signal is transmitted by a transmitter. The information in the AM signal is contained in the amplitude variations of the carrier or the envelope shown by dotted lines in Fig. 4.24.1.
4. Note that the frequency and phase of the carrier remain constant.



5. AM is used in the applications such as radio transmission, TV transmission etc.

**Note :** The modulating signal in practice may or may not be purely sinusoidal. Most of the times it will have a complex shape.




(D-12) Fig. 4.24.1 : AM waveform for sinusoidal modulating signal

### 4.24.1 Mathematical Representation of an AM Wave :

**Expression of AM wave :**

- Let the modulating signal be sinusoidal and be represented as,

$$\boxed{e_m = E_m \cos \omega_m t} \quad ...(4.24.1)$$

nsg

where "$e_m$" is the instantaneous amplitude of the modulating signal, $E_m$ is the peak amplitude, $\omega_m = 2\pi f_m$ and $f_m$ = Frequency of the modulating signal.

- Let the carrier signal also be sinusoidal at a much higher frequency than that of the modulating signal. The instantaneous carrier signal $e_c$ is given by,

$$\boxed{e_c = E_c \cos \omega_c t} \quad ...(4.24.2)$$

where $E_c$ = Peak carrier amplitude,

$f_c$ = Carrier frequency and $\omega_c = 2\pi f_c$.

- The AM wave is expressed by the following expression,

$$e_{AM} = A \cos (2\pi f_c t) \quad ...(4.24.3)$$

where A = Envelope of AM wave

- Where A represents the instantaneous value of the envelope. The modulating signal either adds or gets subtracted from the peak carrier amplitude $E_c$ as shown in Fig. 4.24.1. Hence we can represent the instantaneous value of envelope as,

$$A = E_c + e_m = E_c + E_m \cos (2\pi f_m t) \quad ...(4.24.4)$$

- Hence the AM wave is given by,

$$e_{AM} = A \cos (2\pi f_c t)$$

$$= [E_c + E_m \cos (2\pi f_m t)] \cos (2\pi f_c t)$$

$$\therefore \quad e_{AM} = E_c \left[ 1 + \frac{E_m}{E_c} \cos (2\pi f_m t) \right] \cos (2\pi f_c t)$$

- Let $m = E_m / E_c$ be the modulation index.

$$\therefore \quad e_{AM} = E_c [1 + m \cos (2\pi f_m t)] \cos (2\pi f_c t) \quad ...(4.24.5)$$

This expression represents the time domain representation of an AM signal.

**Note :** It is not necessary to always consider the cosine waves to obtain the mathematical expression. We can even use the sine waves to obtain the mathematical expression for AM.

### 4.24.2 Modulation Index or Modulation Factor :

- In AM wave the modulation index (m) is defined as the ratio of amplitudes of the modulating and carrier waves as follows :

$$m = \frac{E_m}{E_c} \quad ...(4.24.6)$$

- When $E_m \le E_c$ the modulation index "m" has values between 0 and 1 and no distortion is introduced in the AM wave. But if $E_m > E_c$ then m is greater than 1. This will distort the shape of AM signal. The distortion is called as "over modulation."
- The modulation index is also called as modulation factor, modulation coefficient or degree of modulation. However if modulation index is expressed as percentage it is called as "percentage modulation."

$$\therefore \quad \% \text{ Modulation} = \frac{E_m}{E_c} \times 100 \quad ...(4.24.7)$$

- Note that "m" is a dimensionless quantity.

### 4.24.3 Frequency Spectrum of the AM Wave (Frequency Domain Description) :

- The frequency spectrum is a graph of amplitude on Y axis versus frequency on X axis. The frequency spectrum of AM wave tells us about which frequency components are present in the AM wave and what are their amplitudes. So consider the equation for AM wave.

$$e_{AM} = (E_c + E_m \cos \omega_m t) \cos \omega_c t$$

$$E_c = \left[ 1 + \frac{E_m}{E_c} \cos \omega_m t \right] \cos \omega_c t$$

- As per the definition of the modulation index, $m = E_m / E_c$.

$$\therefore \quad e_{AM} = E_c (1 + m \cos \omega_m t) \cos \omega_c t \quad ...(4.24.8)$$

Simplifying we get,

$$e_{AM} = E_c \cos \omega_c t + m E_c \cos \omega_m t \cos \omega_c t \quad ...(4.24.9)$$

- For the second term in the above expression use the following standard identity :

$$2 \cos A \cos B = \cos (A + B) + \cos (A - B)$$

– Therefore Equation (4.24.9) gets simplified as follows :

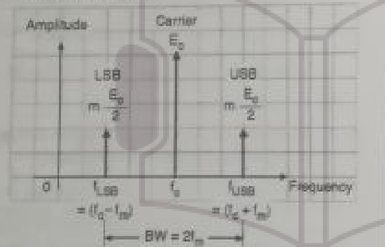
$$e_{AM} = \underbrace{E_c \cos \omega_c t}_{\text{Carrier}} + \underbrace{\frac{m E_c}{2} \cos (\omega_c + \omega_m) t}_{\text{Upper sideband}} + \underbrace{\frac{m E_c}{2} \cos (\omega_c - \omega_m) t}_{\text{Lower sideband}} \quad ...(4.24.10)$$

**Observations :**

The expression for the AM wave shows that it consists of three terms :

1. First term is nothing else but the unmodulated carrier signal.
2. The second term is a sinusoidal signal at frequency $(f_c + f_m)$. This is called as the upper sideband (USB). Its amplitude is $\frac{m E_c}{2}$.
3. The third term represents a sinusoidal signal at frequency $(f_c - f_m)$. It is called as the lower sideband (LSB). Its amplitude is $\frac{m E_c}{2}$.

– Hence the frequency spectrum of an A.M. wave is as shown in Fig. 4.24.2. Note that it is a single sided spectrum i.e. the spectrum plotted for only the positive values of frequency.




(B-37) **Fig. 4.24.2 : Single sided frequency spectrum of AM wave**

## 4.24.4 Concept of Sidebands :

– The AM wave consists of three frequency components (i.e. three sinewaves of different frequencies) namely, the carrier, the lower sideband and upper sideband.

– The lower sideband (LSB) is a sinusoidal component which has a frequency of $(f_c - f_m)$ and an amplitude of $(mE_c / 2)$.

– The upper sideband (USB) is another sinusoidal component which has a frequency of $(f_c + f_m)$ and an amplitude of $(mE_c / 2)$.

– The carrier has a frequency $f_c$ and an amplitude of $E_c$.

**Information content in AM wave :**

– Note that the amplitude of LSB and USB is $(mE_c / 2)$ i.e. it is directly proportional to modulation index m. But the amplitude of carrier $(E_c)$ does not depend on the modulation index.

– This indicates that only the sidebands contain all the information to be conveyed from the transmitter to receiver. Also note that both the sidebands contain identical information.

– The carrier does not contain any information.

– So in order to recover back the transmitted information, it is essential to recover the sidebands at the receiver without any distortion.

## 4.24.5 Bandwidth Requirement :

– The bandwidth of the AM signal is equal to the difference between the highest and the lowest frequency component in the frequency spectrum. Therefore :

$$BW = f_{USB} - f_{LSB} = (f_c + f_m) - (f_c - f_m)$$

$$BW = 2f_m \quad ...(4.24.11)$$

– This shows that the minimum bandwidth requirement of the DSBFC AM system is equal to twice the modulating frequency $(f_m)$.

**Solved Examples :**

**Ex. 4.24.1 :** A modulating signal 10 sin $(2\pi \times 10^3 t)$ is used to modulate a carrier signal 20 sin $(2\pi \times 10^4 t)$. Find the modulation index, percentage modulation, frequencies of the sideband components and their amplitudes. What is the bandwidth of the modulated signal ? Also draw the spectrum of the AM wave.

**Soln. :**

– The modulating signal $e_m$ = 10 sin $(2\pi \times 10^3 t)$. So comparing this with the expression

$e_m = E_m \sin (2 \pi f_m t)$ we get,

$E_m$ = 10 Volts, $f_m = 1 \times 10^3$ Hz = 1 kHz

– The carrier signal $e_c$ = 20 sin $(2\pi \times 10^4 t)$.

Comparing this with the expression $e_c = E_c \sin (2 \pi f_c t)$ we get,

$E_c$ = 20 Volts, $f_c = 1 \times 10^4$ Hz = 10 kHz

**Step 1 : Modulation index and percentage modulation :**

$$m = \frac{E_m}{E_c} = \frac{10}{20} = 0.5 \text{ and } \% \text{ modulation}$$

$$= 0.5 \times 100 = 50\%.$$

**Step 2 : Frequencies of sideband components :**

1. Upper sideband $f_{USB} = f_c + f_m = (10 + 1) = 11$ kHz
2. Lower sideband $f_{LSB} = f_c - f_m = (10 - 1) = 9$ kHz

**Step 3 : Amplitudes of sidebands :**

The amplitudes of upper as well as the lower sideband is given by,

$$\text{Amplitude of each sideband} = \frac{m E_c}{2} = \frac{0.5 \times 20}{2} = 5 \text{ Volts}$$

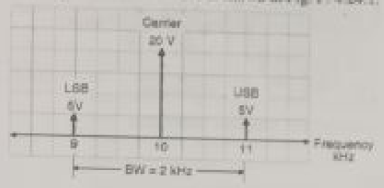


**Step 4 : Bandwidth :**

Bandwidth = $2 f_m = 2 \times 1 = 2$ kHz

**Step 5 : Spectrum :**

The spectrum of AM wave is shown in Fig. P. 4.24.1.




(B-703) **Fig. P. 4.24.1 : Spectrum of the AM wave**

**Ex. 4.24.2 :** In AM, modulating signal frequency is 10 kHz and carrier frequency is 1 MHz. Determine the resultant frequency components.

**Soln. :**

**Given :** $f_m$ = 10 kHz, $f_c$ = 1 MHz

The resultant frequency components will include the carrier, upper sideband and lower sideband.

1. Carrier frequency, $f_c$ = 1 MHz
2. Upper sideband, $(f_c + f_m)$ = 1000 kHz + 10 kHz = 1010 kHz
3. Lower sideband, $(f_c - f_m)$ = 1000 kHz – 10 kHz = 990 kHz

## 4.24.6 Effects of Modulation Index on the A.M. Wave :

Depending on the value of percentage modulation (m) the AM wave can be classified into two categories :

1. Linear modulation
2. Overmodulation

**Linear modulation :**

– If $m \le 1$ or if the percentage modulation is less than 100% then the type of amplitude modulation is linear amplitude modulation.

– The waveforms of AM waves with linear modulation are in Figs. P. 4.24.3(a) and (b) respectively (Refer Ex. 4.24.3)

**Overmodulation :**

– If m > 1 i.e. if the percentage modulation is greater than 100% then the type of amplitude modulation is called as overmodulation.

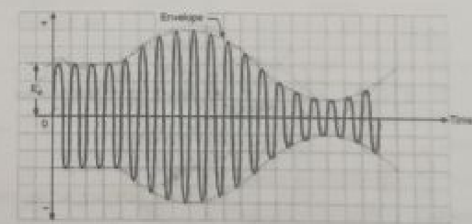
– For m > 1 the envelope can sometimes reverse the phase as shown in Fig. P. 4.24.3(c) in Ex. 4.24.3.

– Overmodulation introduces envelope distortion. Hence it should be avoided.

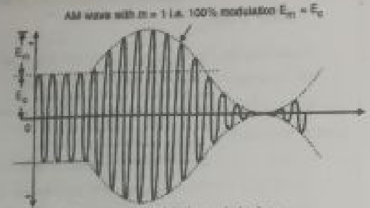
**Ex. 4.24.3 :** Draw the AM waveforms for less than 100 %, with 100 %, more than 100 % and with 0 % percentage modulation. Assume that the modulating signal is a pure sine wave.

**Soln. :**

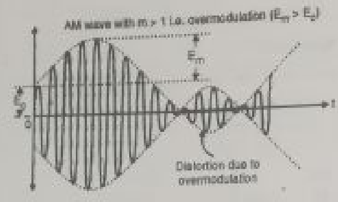
The required waveforms are shown in Figs. P. 4.24.3(a), (b), (c) and (d) respectively.



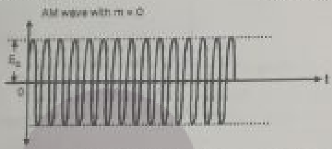

(B-704) **Fig. P. 4.24.3(a) : AM wave for percentage modulation less than 100 %**




(b) AM wave with 100% modulation




(c) AM with over modulation




(B-786) **Fig. P. 4.24.3(d) : AM wave with m = 0**

**Ex. 4.24.4 :** An audio frequency signal $10 \sin 2\pi \times 500\, t$ is used to amplitude modulate a carrier of $50 \sin 2\pi \times 10^5\, t$. Calculate :

1. Modulation index
2. Sideband frequencies
3. Amplitude of each sideband frequencies
4. Bandwidth requirement
5. Total power delivered to a load of 600 Ω.

**Soln. :**

1. The modulating signal $e_m = 10 \sin(2\pi \times 500\, t)$

Comparing it with standard modulating signal given by,

$$e_m = E_m \sin(2\pi f_m t) \text{ we get,}$$
$$E_m = 10\text{ V},\ f_m = 500\text{ Hz}$$

2. The carrier signal $e_c = 50 \sin(2\pi \times 10^5\, t)$

Comparing it with the standard carrier signal given by,

$$e_c = E_c \sin(2\pi f_c t) \text{ we get,}$$
$$E_c = 50\text{ V},\ f_c = 1 \times 10^5\text{ Hz} = 100\text{ kHz}$$

**Step 1 : Modulation index :**

$$m = \frac{E_m}{E_c} = \frac{10}{50} = 0.2$$

**Step 2 : Sideband frequencies :**

1. $f_{USB} = f_c + f_m = 100.5$ kHz
2. $f_{LSB} = f_c - f_m = 99.5$ kHz

**Step 3 : Amplitude of sidebands :**

$$\text{Amplitude of sidebands} = \frac{m E_c}{2} = \frac{0.2 \times 50}{2} = 5\text{ V}$$

**Step 4 : Bandwidth :**

$$BW = 2 f_m = 2 \times 500\text{ Hz} = 1\text{ kHz}$$

**Step 5 : Power delivered to load :**

$$\text{Carrier power, } P_c = \frac{(E_c/\sqrt{2})^2}{R_L} = \frac{E_c^2}{2 R_L}$$
$$= \frac{(50)^2}{2 \times 600} = 2.0833\text{W}$$
$$\text{Total power, } P_t = P_c\left[1 + \frac{m^2}{2}\right]$$
$$= 2.0833\left[1 + \frac{(0.2)^2}{2}\right]$$
$$= 2.125\text{ W}$$

**Ex. 4.24.5 :** A carrier wave of frequency 1 MHz and peak value 10 V is amplitude modulated by a 5 kHz sine wave of amplitude 6 V. Determine the modulation index and draw spectrum.

**Soln. :**

**Given :** $f_c = 1$ MHz, $E_c = 10$ V, $f_m = 5$ kHz, $E_m = 6$ V

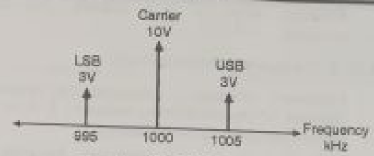
**Step 1 : Modulation index :**

$$m = \frac{E_m}{E_c} = \frac{6}{10} = 0.6 \quad \text{...Ans.}$$

**Step 2 : Spectrum :**

1. $f_{USB} = f_c + f_m = 1000\text{ kHz} + 5\text{ kHz} = 1005\text{ kHz}$
2. $f_{LSB} = f_c - f_m = 1000\text{ kHz} - 5\text{ kHz} = 995\text{ kHz}$
3. Amplitude of each sideband $= \dfrac{m E_c}{2} = \dfrac{0.6 \times 10}{2} = 3$ Volts
4. Spectrum is shown in Fig. P. 4.24.5.






(L-788) **Fig. P. 4.24.5 : Spectrum**

**Ex. 4.24.6 :** For given data, find modulation index, frequencies of the sideband components and their amplitudes and plot the frequency spectrum of Amplitude modulated wave :

Modulating signal $e_m = 5 \cos 2\pi\, 10^3\, t$

Carrier signal $e_c = 10 \cos 2\pi\, 10^4\, t$

**Soln. :**

**Given :** Modulating signal $e_m = 5 \cos 2\pi\, 10^3\, t$

Carrier signal $e_c = 10 \cos 2\pi\, 10^4\, t$

**To find :** 1. Modulation index
2. Sideband frequencies and amplitudes
3. Frequency spectrum

**1. Modulation index (m) :**

From the expressions of $e_m$ and $e_c$ we get,

$$E_m = 5\text{ V},\ f_m = 1000\text{ Hz},\ E_c = 10\text{ V},\ f_c = 10\text{ kHz}.$$
$$\therefore\ m = \frac{E_m}{E_c} = \frac{5}{10} = 0.5 \quad \text{...Ans.}$$

**2. Sideband frequencies :**

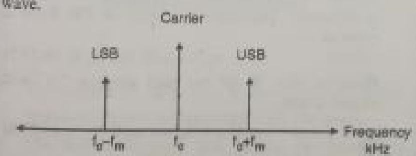
$$f_{USB} = f_c + f_m = 10 + 1 = 11\text{ kHz} \quad \text{...Ans.}$$
$$f_{LSB} = f_c - f_m = 10 - 1 = 9\text{ kHz} \quad \text{...Ans.}$$

**3. Amplitude of sidebands :**

$$\text{Amplitude of both the sidebands} = \frac{m E_c}{2} = \frac{0.5 \times 10}{2}$$
$$= 2.5\text{ V} \quad \text{...Ans.}$$

**4. Frequency spectrum :**

Fig. P. 4.24.6 shows the frequency spectrum of AM wave.




(L-789) **Fig. P. 4.24.6**

### 4.24.7 Generation of AM Wave :

- Fig. 4.24.3 shows the principle of generation of AM waves. The oscillator produces a sinusoidal carrier of desired high frequency.
- This carrier and the modulating signal are applied to a multiplier circuit. At the output of the multiplier we get the A.M. wave as shown. Thus a multiplier is the simplest type of A.M. wave generator.


(L-781) **Fig. 4.24.3 : Generation of AM wave**

## 4.25 Advantages, Disadvantages and Applications of AM :

### 4.25.1 Disadvantages of AM (DSBFC) :

The AM signal is also called as "Double Sideband Full Carrier (DSBFC)" signal. The three main disadvantages of this technique are :

1. Power wastage takes place (Carrier does not contain any information).
2. AM needs larger bandwidth.
3. AM wave gets affected due to noise.

### 4.25.2 Advantages of AM :

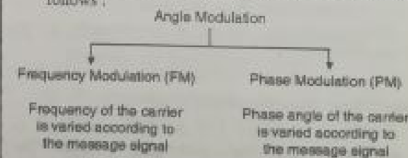
1. AM transmitters are less complex.
2. AM receivers are simple, detection is easy.
3. AM receivers are cost efficient. Hence even a common person can afford to buy it.
4. AM waves can travel a longer distance.
5. Low bandwidth.

### 4.25.3 Applications of AM :

1. Radio broadcasting.
2. Picture transmission in a TV system.

## 4.26 Angle Modulation : Basic Concepts :

- There is another method of modulating a sinusoidal carrier namely the angle modulation. In angle modulation either frequency or phase of the carrier is varied in proportion with the message signal amplitude, but the carrier amplitude remains constant.
- Thus angle modulation systems can be classified as follows :

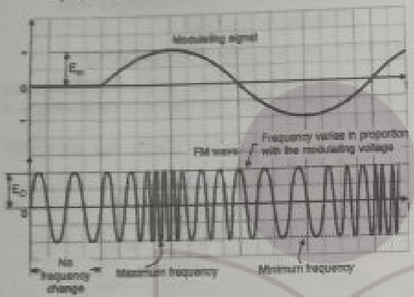



(B-786) **Fig. 4.26.1 : Classification of angle modulation**

## 4.27 Frequency Modulation (FM) :

- In sinusoidal Frequency Modulation (FM), the modulating signal $x(t) = E_m \cos(2\pi f_m t)$ is a pure sinusoidal signal. The carrier signal c (t) is also a sinewave at much higher frequency.

- FM is a system of modulation in which the instantaneous frequency of the carrier is varied in proportion with the amplitude of the modulating signal. The amplitude of the carrier signal remain constant. Thus the information is conveyed via frequency changes.
- FM was first practically tried in 1936 as an alternative to AM. As will be shown later on, FM transmission is more resistant to noise than A.M. The time domain display of FM wave is as shown in the Fig. 4.27.1.




(B-760) **Fig. 4.27.1 : Time domain display of FM wave**

- The amount by which the carrier frequency deviates from its unmodulated value is called as **"deviation"**. The deviation (δ) is made proportional to the instantaneous value of modulating voltage.
- The rate at which these frequency variations or oscillations takes place in the FM wave is equal to the modulating frequency ($f_m$).
- The amplitude of the FM wave always remains constant. This is the biggest advantage of FM.

### 4.27.1 Important Definitions in Frequency Modulation :

- For the F.M. wave the modulating signal x (t) be a sinusoidal signal of amplitude $E_m$ and frequency $f_m$.

$$\therefore \quad x(t) = E_m \cos(2\pi f_m t) \qquad ...(4.27.1)$$

- The unmodulated carrier is represented by the expression,

$$e_c = A \sin(\omega_c t + \phi) \qquad ...(4.27.1(a))$$

**Instantaneous frequency of an FM wave :**

In FM, the frequency f of the FM wave varies in accordance with the modulating voltage. The instantaneous frequency of the FM wave is denoted by $f_i$ (t) and is given by,

$$f_i(t) = f_c[1 + k_f x(t)] = f_c[1 + k_f \cdot E_m \cos(2\pi f_m t)] = f_c + \delta \cos(2\pi f_m t) \qquad ...(4.27.2)$$

Where $\delta = k_f\ E_m\ f_c$ and it is called as **frequency deviation**, where $k_f$ is a constant with units Hz / Volts.

### 4.27.2 Frequency Deviation (δ) :

- **Frequency deviation** δ represents the maximum departure of the instantaneous frequency $f_i$ (t) of the FM wave from the carrier frequency $f_c$.
- Since $\delta = k_f\ E_m$, the **frequency deviation** is proportional to the amplitude of modulating voltage ($E_m$) and it is independent of the modulating frequency $f_m$.

**Maximum and minimum frequency of FM wave :**

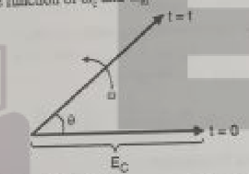
The maximum frequency of FM wave is,

$$f_{max} = f_c + \delta \qquad ...(4.27.3)$$

The minimum frequency of a FM wave is $f_{min} = (f_c - \delta)$.

### 4.27.3 Mathematical Expression for F.M. :

- We know that the FM wave is a sinewave having a constant amplitude and a variable instantaneous frequency. As the instantaneous frequency is changing continuously, the angular velocity "ω" of an FM wave is the function of $\omega_c$ and $\omega_m$.




(D-164) **Fig. 4.27.2 : Frequency modulated vector**

- Therefore the FM wave is represented by,

$$e_{FM} = s(t) = E_c \sin[F(\omega_c, \omega_m)] \qquad ...(4.27.4)$$

$$= E_c \sin\theta(t) \qquad ...(4.27.5)$$

where $\theta(t) = F(\omega_c, \omega_m)$ ...(4.27.6)

- As shown in Fig. 4.27.2, $E_c \sin\theta(t)$ is a rotating vector. If "$E_c$" is rotating at a constant velocity "ω" then we could have written that θ(t) = ωt. But in FM this velocity is not constant. In fact it is changing continuously. The angular velocity of FM wave is given as,

$$\omega = \omega_c[1 + k_f E_m \cos\omega_m t] \qquad ...(4.27.7)$$

- Hence to find "θ (t)" we must integrate "ω" with respect to time.

$$\therefore \theta(t) = \int \omega\, dt = \int \omega_c [1 + k_f E_m \cos\omega_m t]\, dt \qquad ...(4.27.8)$$

$$\therefore \theta(t) = \omega_c \int [1 + k_f E_m \cos\omega_m t]\, dt$$

$$= \omega_c \left[t + \frac{k_f E_m \sin\omega_m t}{\omega_m}\right] = \omega_c t + \frac{k_f E_m \omega_c \sin\omega_m t}{\omega_m}$$

$$\therefore \theta(t) = \omega_c t + \frac{k_f E_m f_c \sin\omega_m t}{f_m} \qquad ...(4.27.9)$$

- As per the definition, $\delta = k_f E_m f_c$



$$\therefore \theta(t) = \omega_c t + \frac{\delta \sin\omega_m t}{f_m} \qquad ...(4.27.10)$$

- Substitute this value of θ (t) in Equation (4.27.5) to get the equation for the FM wave as,

$$e_{FM} = s(t) = E_c \sin\left[\omega_c t + \frac{\delta}{f_m}\sin\omega_m t\right] \qquad ...(4.27.11)$$

- But $\dfrac{\delta}{f_m} = m_f$ i.e. the modulation index of FM wave.

Hence the equation for FM wave is given as,

$$e_{FM} = E_c \sin[\omega_c t + m_f \sin\omega_m t] \qquad ...(4.27.12)$$

- This is the expression for a FM wave, where $m_f$ represents the modulation index.

**Meaning of mathematical representation :**

- The mathematical expression for a FM wave is as follows :

(D-1311)

$$e_{FM} = \underbrace{E_C}_{\text{Peak amplitude of FM wave is constant and equal to the peak amplitude of the carrier}} \underbrace{\sin}_{\text{FM wave is a sine wave}} \underbrace{[\omega_c t + m_f \sin\omega_m t]}_{\text{Frequency of FM wave varies according to the modulating signal}}$$

- The amplitude of FM wave is constant and equal to the amplitude of the carrier i.e. $E_c$.
- FM wave is sinusoidal i.e. it has a shape of sine or cosine wave.
- The frequency of FM wave is not constant. It varies continuously, above and below the carrier frequency $f_c$.

### 4.27.4 Modulation Index of FM :

- The modulation index of an FM wave is defined as :

$$m_f = \frac{\text{Maximum frequency deviation}}{\text{Modulating frequency}} \qquad ...(4.27.13)$$

$$\therefore \quad m_f = \frac{\delta}{f_m} \qquad ...(4.27.14)$$

- The modulation index ($m_f$) is very important in FM because it decides the bandwidth of the FM wave.
- The modulation index also decides the number of sidebands having significant amplitudes.
- In AM the maximum value of the modulation index m is 1. But for FM the modulation index can be greater than 1. The modulation index $m_f$ is measured in radians.

### 4.27.5 Deviation Ratio :

In FM broadcasting the maximum value of deviation is limited to 75 kHz. The maximum modulating frequency is also limited to 15 kHz. The modulation index corresponding to the maximum deviation and maximum modulating frequency is called as the "deviation ratio".

$$\text{Deviation ratio} = \frac{\text{Maximum deviation}}{\text{Maximum modulating frequency}} \qquad ...(4.27.15)$$

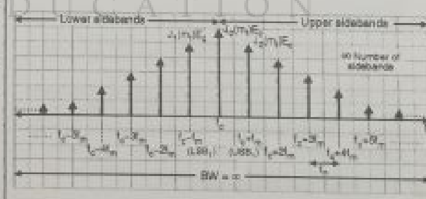
### 4.27.6 Percentage Modulation of FM Wave :

The percent modulation is defined as the ratio of the actual frequency deviation produced by the modulating signal to the maximum allowable frequency deviation.

$$\therefore \quad \%\ \text{Modulation} = \frac{\text{Actual frequency deviation}}{\text{Maximum allowed deviation}} \qquad ...(4.27.16)$$

### 4.27.7 Frequency Spectrum of FM Wave (Frequency Domain Representation) :

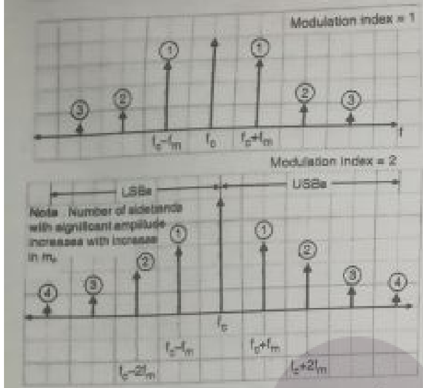
- Frequency domain representation of FM wave is a graph of amplitude plotted on y axis versus the frequency plotted on the x axis.
- To represent the FM wave in the frequency domain, consider the equation of FM wave again.
- The expression for the FM wave is not simple. It is complex since it is sine of sine function. The only way to solve this equation is by using the Bessel functions.




(B-713) **Fig. 4.27.3 : Ideal frequency spectrum of FM wave**

### 4.27.8 Effect of Modulation Index on the Frequency Spectrum of FM :

- As the amplitude of modulating signal varies, the frequency deviation will change. The number of sidebands produced and their amplitudes will change.




(D-167) **Fig. 4.27.4 : Effect of modulation index on the significant number of sidebands**

- Fig. 4.27.4 illustrates the effect of modulation index on the frequency spectrum of FM.
- Higher the value of $m_f$, more will be the number of sidebands having significant amplitudes as shown in Fig. 4.27.4.

### 4.27.9 Bandwidth Requirement of FM :

- Bandwidth of an FM wave is defined as the frequency difference between the highest pair of sidebands.
- Ideally the bandwidth of FM is infinite, because its spectrum consists of infinite number of upper and lower sidebands.
- But practically the bandwidth depends on the number of significant sidebands.
- The number of sidebands having significant amplitudes will increase with increase in the value of modulation index $m_f$. Hence the bandwidth increases with increase in the value of $m_f$.

### 4.27.10 Practical Bandwidth :

- Theoretically the bandwidth of the FM wave is infinite. But practically it is calculated based on how many sidebands have significant amplitude.
- The simplest method to calculate the bandwidth is as follows :

BW = 2 $f_m$ × Number of significant sidebands ...(4.27.17)

- With increase in modulation index, the number of significant sidebands increase. This will increase the bandwidth. The bandwidth of FM is higher than that of AM.

**Carson's Rule :**

- The second method to find the practical bandwidth is a rule of thumb (Carson's rule). It states that the bandwidth of FM wave is equal to twice the sum of the deviation and the highest modulating frequency.

$$BW = 2[\delta + f_{m(max)}] \quad ...(4.27.18)$$

- The Carson's rule gives correct results if the modulation index is greater than 6.

**Ex. 4.27.1 :** Define the term "percent modulation" and determine the percent modulation for an FM wave with a frequency deviation of 10 kHz if the maximum deviation allowed is 25 kHz.

**Soln. :**

The percent modulation is defined as the ratio of the actual frequency deviation produced by the modulating signal to the maximum allowable frequency deviation.

$$\therefore \quad \% \text{ Modulation} = \frac{\text{Actual frequency deviation}}{\text{Maximum allowed deviation}} \quad ...(1)$$

In the example it is given that $\delta = 10$ kHz and $\Delta f_{max} = 25$ kHz

$$\therefore \quad \% \text{ Modulation} = \frac{10 \text{ kHz}}{25 \text{ kHz}} = 40\%$$

**Ex. 4.27.2 :** In an F.M. system, if the maximum value of deviation is 75 kHz and the maximum modulating frequency is 10 kHz calculate the deviation ratio and bandwidth of the system using Carson's rule.

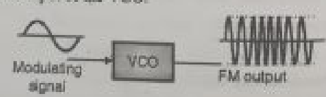
**Soln. :**

**Given :** $\delta_{max} = 75$ kHz , $f_{m(max)} = 10$ kHz

1. Deviation ratio $D = \frac{\delta_{max}}{f_{m(max)}} = \frac{75 \text{ kHz}}{10 \text{ kHz}} = 7.5$ **...Ans.**
2. System bandwidth $B = 2[\delta_{max} + f_{m(max)}] = 2(75 + 10) = 170$ kHz **...Ans.**

### 4.27.11 Generation of FM Wave :

- Fig. 4.27.5 shows the block schematic of a simple FM modulator. It is a simple Voltage Controlled Oscillator (VCO). The output frequency of voltage controlled oscillator is proportional to the control voltage applied to it.
- The modulating signal is applied at its input. This signal acts as the control voltage. The VCO output frequency varies in proportion with the modulating signal instantaneous value. Thus we get an FM wave at the output of the VCO.




(L-782) **Fig. 4.27.5 : Generation of FM**



## 4.28 Advantages, Disadvantages and Applications of FM :

### 4.28.1 Advantages of FM :

1. Improved noise immunity.
2. Low power is required to be transmitted to obtain the same quality of received signal at the receiver.
3. F.M. transmission covers larger area with the same amount of transmitted power.
4. Transmitted power remains constant.
5. All the transmitted power is useful.

### 4.28.2 Disadvantages of FM :

1. Very large bandwidth is required.
2. Since the space wave propagation is used, the radius of transmission is limited by the line of sight.
3. FM transmission and reception equipments are complex.

### 4.28.3 Applications of FM :

Some of the applications of FM are :

1. Radio broadcasting (Vividh Bharti, Radio Mirchi).
2. Sound broadcasting in T.V.
3. Satellite communication.
4. Police wireless.
5. Point to point communication.

**Ex. 4.28.1 :** What is the bandwidth required for FM in which the modulating frequency is 2 kHz and maximum deviation is 10 kHz. Assume highest needed sidebands are 8.

**Soln. :**

$f_m = 2$ kHz, $\delta = 10$ kHz

**Method I :** Bandwidth = 2 $f_m$ × Number of significant sidebands

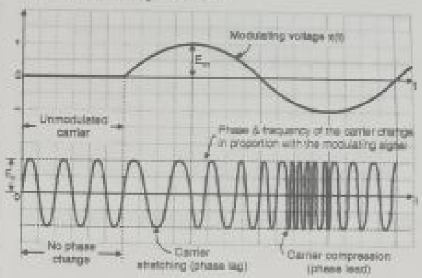
= 2 × 2 kHz × 8 = 32 kHz

**Method II :** Bandwidth = $2[\delta + f_{m(max)}] = 2[10 + 2]$

= 24 kHz

## 4.29 Phase Modulation (PM) :

- Phase modulation is very similar to the frequency modulation. The only difference is that the phase of the carrier is varied instead of varying the frequency. The amplitude of the carrier remains constant.
- As shown in Fig. 4.29.1, as the modulating signal goes positive, the amount of phase lag increases with the amplitude of the modulating signal. The effect of this is that the carrier signal is stretched or its frequency is reduced.
- When the modulating signal goes negative, the phase shift becomes leading. This causes the carrier wave to be effectively compressed. The effect of this is as if the carrier frequency is increased.
- Thus phase modulation is always associated with frequency modulation and vice versa.
- Note that the P.M. wave of Fig. 4.29.1 is the same as the F.M. wave produced by dx(t) / dt i.e. the derivative of x (t) with respect to time.




(L-45) **Fig. 4.29.1 : Time domain display of PM wave**

- So in Fig. 4.29.1 we have plotted the derivative of x (t) which is original x (t) shifted by 90°.
- From the discussion it is clear that the difference between F.M. and P.M. waves can be made only by comparing with the original modulating wave.

### 4.29.1 Mathematical Representation of Phase Modulation (PM) :

- The phase modulation is another type of angle modulation. PM and FM are closely related. It is possible to obtain FM from PM, using the method called "Armstrong method".
- The PM wave is obtained by varying the phase angle $\phi$ of a carrier in proportion with the amplitude of the modulating voltage.

If the carrier voltage is expressed as,

$$e_c = A \sin(\omega_c t + \phi) \quad ...(4.29.1)$$

- Then the PM wave can be expressed as,

$$e_{PM} = A \sin(\omega_c t + \phi_m \sin \omega_m t) \quad ...(4.29.2)$$

- Here $\phi_m$ = Maximum phase change corresponding to the maximum amplitude of the modulating signal. For the sake of uniformity let us modify the Equation (4.29.2) as,

$$e_{PM} = A \sin[\omega_c t + m_p \sin \omega_m t] \quad ...(4.29.3)$$

Where $m_p = \phi_m$ = Modulation index of PM.

- The FM and PM waves look identical when their modulation index are identical. However if we change the modulating frequency $f_m$ then $m_f$ will change but there is no change in the value of $m_p$.

### 4.29.2 Generation of PM :

- Fig. 4.29.2 shows the scheme for the generation of PM wave. The modulating signal x (t) is applied to a differentiator.
- Then the differentiated modulating signal dx (t)/dt is applied to the VCO to produce the PM wave.


(L-798) Fig. 4.29.2 : Generation of PM wave

### 4.29.3 Bandwidth of PM :

- The formula for bandwidth of PM is same as that for FM. But the actual bandwidth of PM is less than that for FM.
- The bandwidth of a PM signal can be calculated from the maximum modulating frequency and the maximum amplitude of the modulating signal.

### 4.29.4 Comparison of FM and PM Systems :

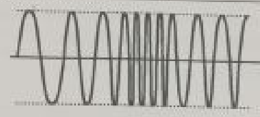
| Sr. No. | FM | PM |
|---|---|---|
| 1. | $s(t) = E_c \sin[\omega_c t + m_f \sin \omega_m t]$ | $s(t) = E_c \sin[\omega_c t + m_p \sin \omega_m t]$ |
| 2. | Frequency deviation is proportional to modulating voltage. | Phase deviation is proportional to the modulating voltage. |
| 3. | Associated with the change in $f_c$, there is some phase change. | Associated with the changes in phase there is some change in $f_c$ |
| 4. | $m_f$ is proportional to the modulating voltage as well as the modulating frequency $f_m$ | $m_p$ is proportional only to the modulating voltage. |
| 5. | It is possible to receive FM on a PM receiver. | It is possible to receive PM on a FM receiver. |
| 6. | Noise immunity is better than AM and PM. | Noise immunity is better than AM but worse than FM. |
| 7. | Amplitude of the FM wave is constant. | Amplitude of the PM wave is constant. |
| 8. | Signal to noise ratio is better than that of PM. | Signal to noise ratio is inferior to that in FM. |
| 9. | FM is widely used. | PM is used in some mobile systems. |
| 10. | In FM the frequency deviation is proportional to the modulating voltage only. | In PM the frequency deviation is proportional to both the modulating voltage and modulating frequency. |

### 4.29.5 Comparison of FM and AM Systems :

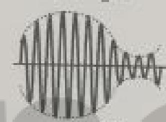
| Sr. No. | FM | AM |
|---|---|---|
| 1. | Amplitude of FM wave is constant. It is independent of the modulation index. | Amplitude of AM wave will change with the modulating voltage. |
| 2. | Hence transmitted power remains constant. It is independent of $m_f$. | Transmitted power is dependent on the modulation index. |
| 3. | All the transmitted power is useful. | Carrier power and one sideband power are useless. |
| 4. | FM receivers are immune to noise. | AM receivers are not immune to noise. |
| 5. | It is possible to decrease noise further by increasing deviation. | This feature is absent in AM. |
| 6. | Bandwidth = 2 [$\delta + f_m$]. The bandwidth depends on modulation index. | BW = $2f_m$. It is not dependent on the modulation index. |
| 7. | BW is large. Hence wide channel is required. | BW is much less than FM. |
| 8. | Space wave is used for propagation. So radius of transmission is limited to line of sight. | Ground wave and sky wave propagation is used. Therefore larger area is covered than FM |
| 9. | Hence it is possible to operate several transmitters on same frequency. | Not possible to operate more channels on the same frequency. |
| 10. | FM transmission and reception equipment are more complex. | AM equipments are less complex. |
| 11. | The number of sidebands having significant amplitudes depends on modulation index $m_f$. | Number of sidebands in AM will be constant and equal to 2. |
| 12. | The information is contained in the frequency variation of the carrier. | The information is contained in the amplitude variation of the carrier. |



| Sr. No. | FM | AM |
|---|---|---|
| 13. | FM wave : Fig. A | AM wave : Fig. B |
| 14. | **Applications** : Radio, TV broadcasting, police wireless, point to point communications. | **Applications** : Radio and TV broadcasting. |



(L-46) Fig. A



(L-46) Fig. B

## Review Questions

Q. 1 What is line coding ? Why line codes are essential ?

Q. 2 What are the disadvantages of RZ codes ?

Q. 3 State the various requirements of a line code.

Q. 4 Why is serial transmission preferred over the parallel transmission ?

Q. 5 Why is the speed of asynchronous transmission is low ?

Q. 6 What is the function of start and stop bits in asynchronous ?

Q. 7 In ______ type transmission, the bits are transmitted simultaneously.

**Ans. : Parallel**

Q. 8 In the asynchronous transmission, the time gap between successive bytes is ______.

**Ans. : Variable**

Q. 9 Write a short note on : Manchester coding.

Q. 10 Explain the term parallel transmission.

Q. 11 State the advantages, disadvantages and applications of parallel transmission.

Q. 12 Explain the serial transmission mode.

Q. 13 Explain the asynchronous transmission.

Q. 14 State advantages, disadvantages and application of asynchronous transmission.

Q. 15 Explain synchronous transmission.

Q. 16 Compare synchronous and asynchronous transmission.

Q. 17 Define pulse modulation. Give the types of pulse modulation.

Q. 18 Define Nyquist rate and Nyquist interval ?

Q. 19 What is quantizing noise ?

Q. 20 State the applications of PCM signals ?

Q. 21 Explain the slope overload distortion. How can it be minimized ?

Q. 22 What is granular noise ?

Q. 23 How is the "information" transmitted in a PCM system ?

Q. 24 What is quantization ?

Q. 25 What is quantization error ? What is its maximum value ?

Q. 26 How to reduce the quantization error ?

Q. 27 State and explain sampling theorem.

Q. 28 Draw and explain the block diagram for generation of PCM signal.

Q. 29 Represent ASK mathematically.

Q. 30 State the bandwidth requirement of ASK system.

Q. 31 What is the maximum B.W. of BPSK system ?

Q. 32 Draw the BPSK signal for the following binary signal. 1 0 1 1 1 0 1 0

Q. 33 Express QPSK mathematically.

Q. 34 How many phases are transmitted in QPSK ?

Q. 35 What are the advantages of QPSK system ?

Q. 36 What is the type of demodulation used for QPSK ?

Q. 37 State the expression for BFSK.

Q. 38 How is a message transmitted in BFSK ?

Q. 39 What is the BW of BFSK ?

Q. 40 What is the BW requirement of QAM ?

Q. 41 Why is QPSK superior to BPSK ?

Q. 42 State merits and demerits of BASK.

Q. 43 What type of receiver is used for the BPSK detection ?

Q. 44 Compare ASK and FSK.

Q. 45 Draw the waveforms for FSK and PSK modulation.

Q. 46 Explain the QPSK modulation scheme with constellation diagram.

Q. 47 What is ASK ? Draw its waveform ?

Q. 48 Draw the block diagram of binary PSK system and explain with signal space diagram.

Q. 49 Write an expression for the BFSK and explain the spectrum of BFSK.

Q. 50 Draw the BFSK waveform to represent the following bit stream.

00101110.

Q. 51 Explain clearly the difference between phase modulation and frequency modulation.

Q. 52 Explain the direct method of FM generation (reactance modulator).

Q. 53 Justify FM is called a constant B.W. system.

Q. 54 Compare and contrast : Frequency modulation and phase modulation.

Q. 55 Define FM and draw the necessary waveforms to explain it.

Q. 56 Derive an equation for FM wave.

Q. 57 Compare AM and FM.

Q. 58 Explain the generation of FM wave.

Q. 59 Write short note on : Frequency spectrum of FM wave.

Q. 60 Compare AM with FM with special reference to power requirements signed to noise ratio and bandwidth required.

Q. 61 Derive the formula for instantaneous value of an FM voltage and define modulation index.

Q. 62 What is angle modulation ?

❑❑❑

CHAPTER 5

Unit II

# Multiplexing

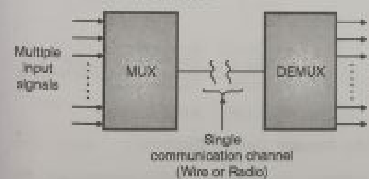
**Syllabus :**
Multiplexing, Frequency division multiplexing, Wavelength division multiplexing, Time division multiplexing.

## 5.1 Introduction to Multiplexing :

- Multiplexing is the process of simultaneously transmitting two or more individual signals over a single communication channel.
- Due to multiplexing it is possible to increase the number of communication channels so that more information can be transmitted.
- The typical applications of multiplexing are in telemetry and telephony or in the satellite communication.

## 5.2 Concept of Multiplexing and Demultiplexing :

- The concept of a simple multiplexer is illustrated in Fig. 5.2.1.
- The multiplexer receives a large number of different input signals.
- Multiplexer has only one output which is connected to the single communication channel.
- The multiplexer combines all input signals into a single composite signal and transmits it over the communication medium.




(L-105) **Fig. 5.2.1 : Concept of multiplexing**

- Sometimes the composite signal is used for modulating a carrier before transmission.
- At the receiving end, of communication link, a demultiplexer is used to separate out the signals into their original form.
- The operation of demultiplexer is exactly opposite to that of a multiplexer. Demultiplexing is the process which is exactly opposite to that of multiplexing.

### 5.2.1 Types of Multiplexing :

- There are three basic types of multiplexing. They are :
  1. Frequency Division Multiplexing (FDM).
  2. Time Division Multiplexing (TDM).
  3. Wavelength Division Multiplexing (WDM).
- The multiplexing techniques can be broadly classified into two categories namely analog and digital.
- Analog multiplexing can be either FDM or WDM and digital multiplexing is TDM.
- Fig. 5.2.2 shows the classification of multiplexing techniques.


(L-106) **Fig. 5.2.2 : Classification of multiplexing techniques**

- Generally the FDM and WDM systems are used to deal with the analog information whereas the TDM systems are used to handle the digital information.
- In FDM many signals are transmitted simultaneously where each signal occupies a different frequency slot within a common bandwidth.
- In TDM the signals are not transmitted at a time, instead they are transmitted in different time slots.

## 5.3 Frequency Division Multiplexing (FDM) :

- The operation of FDM is based on sharing the available bandwidth of a communication channel among the signals to be transmitted.
- That means many signals are transmitted simultaneously with each signal occupying a different frequency slot within the total available bandwidth.
- Each signal to be transmitted modulates a different carrier. The modulation can be AM, SSB, FM or PM.