1.9.2 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.

1.9.3 Types of Serial Transmission :

In data communication, timing control of the reception of bits is important. There are methods of timing control for the reception of bits. They are :

- Synchronous data transmission.
- Asynchronous data transmission

1.10 Asynchronous Transmission :

- In asynchronous transmission, the transmitter commences transmission of data bytes at instant of time.
- Only one byte is sent at a time. After sending one byte the next byte can be sent after an arbit time delay as shown in Fig. 1.10.1.
- The transmitter and receiver operate at different clock frequencies.
- As the data transmission can commence at any instant, it becomes difficult for the receive 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 byte as shown in Fig. 1.10.1. The start bit is always "0" and stop bit is always "1".

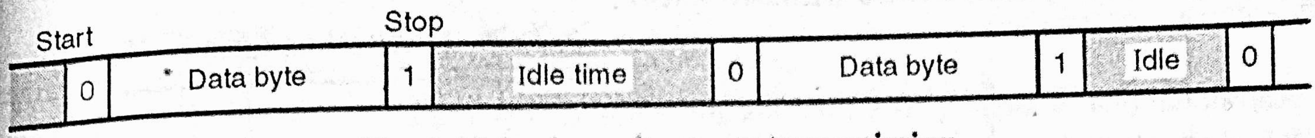


Fig. 1.10.1 : Asynchronous transmission

- The idle time in between two 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 agreed upon patterns.
- 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.

1.10.1 Block Diagram of Asynchronous Transmission :

- Fig. 1.10.2 shows the block diagram of asynchronous transmission.
- The start bits are 0 and stop bits are 1, as shown in Fig. 1.10.2.

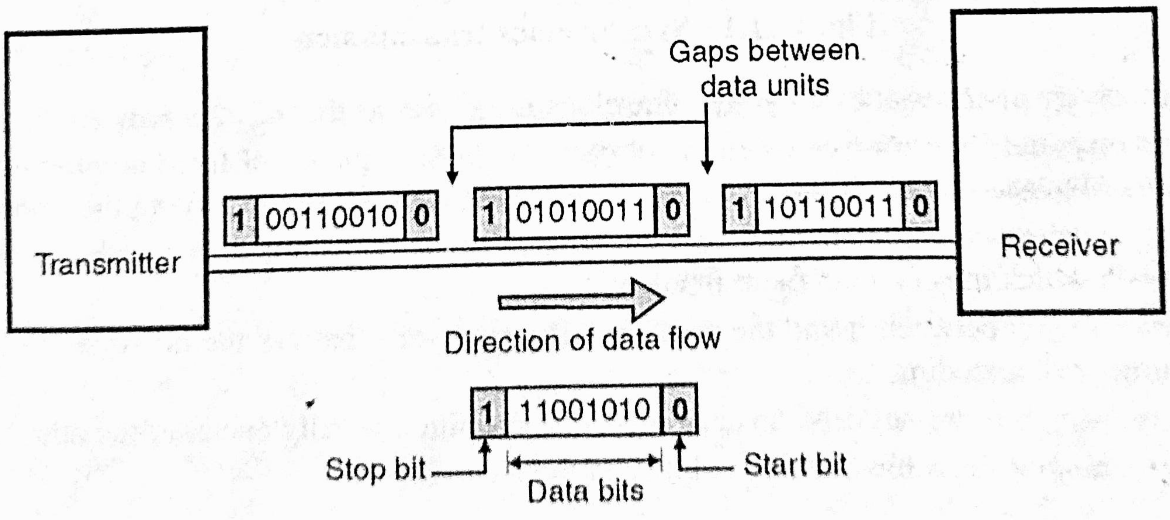


Fig. 1.10.2 : Asynchronous transmission

Disadvantage of using start and stop bits :

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

Disadvantages of asynchronous transmission :

➤➤➤ [Asked in Exam : Oct. 2006 !!!

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.

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.

Application of asynchronous transmission :

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

1.11 Synchronous Transmission :

➤➤➤ [Asked in Exam : Oct. 2004, April 2005, Oct. 2006 !!!]

- 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. 1.11.1. There is an inter block idle time which also is filled with idle characters.

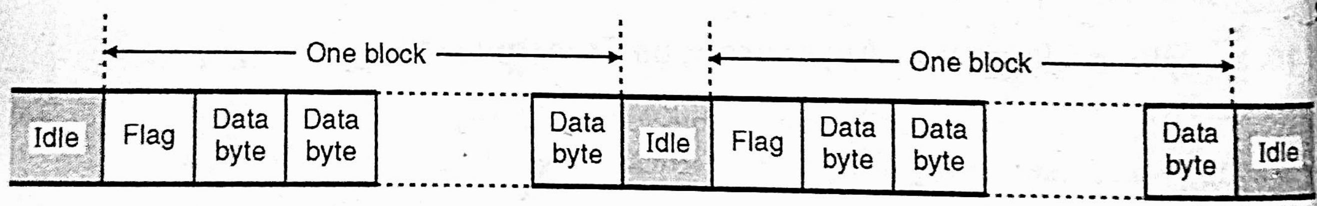


Fig. 1.11.1 : Synchronous transmission

- The receivers operates at exactly the same clock frequency as that of transmitter.
- 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. 1.23.1. Flag is useful in identifying the start of a block.
- In the synchronous transmission the bit stream to be transmitted is combined into longer "frames", which may contain more than one bytes.
- There is no gap between it and the next one. 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.

1.11.1 Block Diagram of Synchronous Transmission :

➤➤➤ [Asked in Exam : Oct. 2006 !!!]

Fig. 1.11.2 shows the block schematic of synchronous transmission. Note the absence of gaps and start stop bits.

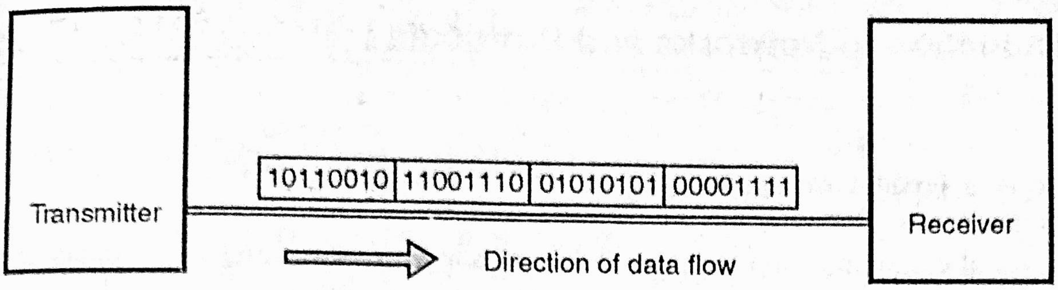


Fig. 1.11.2 : Synchronous transmission

Advantages :

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

Disadvantages :

➤➤➤ [Asked in Exam : April 2006 !!!]

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.

Application of synchronous transmission :

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

Synchronization :

The byte synchronization is accomplished in the data link layer. When synchronous transmission is used between computers.

1.11.2 Comparison of Synchronous and Asynchronous Transmission :

➤➤➤ [Asked in Exam : April 2005, April 2006 !!!]

Table 1.11.1

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.

2.2 Modulation :

➤➤➤ **[Asked in Exam : Oct. 2004 !!!]**

- 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.
- In the modulation process some parameter of the carrier wave (such as amplitude, frequency or phase) is varied in accordance with the modulating signal.

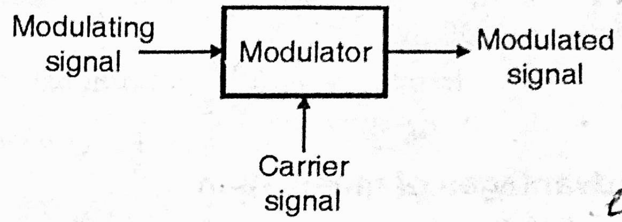


Fig. 2.2.1 : Modulation

- This modulated signal is then transmitted by the transmitter.
- 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. 2.2.2.

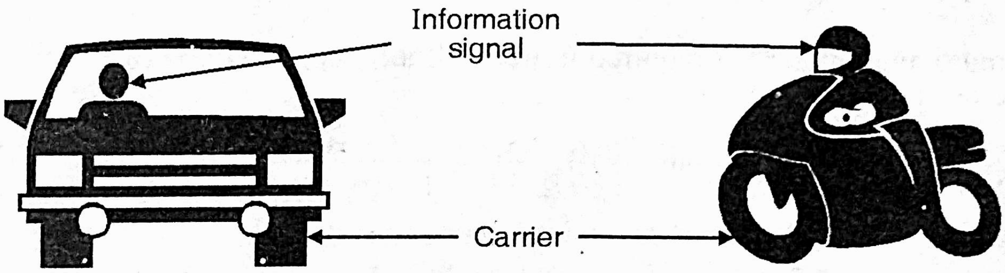


Fig. 2.2.2 : Concept of modulation

2.2.1 Frequency Translation in the Modulation Process :

- The baseband signal or modulating signal is a low frequency signal. For example the audio signal is present in the frequency range from 20 Hz to 20 kHz.
- But due to modulation, the same signal now gets translated to a higher frequency range.
- For example the Vividh Bharati FM pune station is tuned at 101 MHz, or AM Pune station is at 792kHz.

2.2.2 Multiplexing :

- Multiplexing is the process of combining several message signals together and send them over the same communication channel.

2.3 Need of Modulation :

>>> [Asked in Exam : Oct. 2004, Oct. 2006 !!!]

- 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 to high frequency.
- This frequency shift is proportional to the frequency of carrier. The modulation process has the following advantages :

Advantages of modulation :

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

Reduction in height of antenna :

- For the transmission of radio signals, the antenna height must be a multiple of ($\lambda/4$). Here λ 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 :

$$\text{Minimum antenna height} = \frac{\lambda}{4} = \frac{c}{4f} = \frac{3\times10^8}{4\times10\times10^3} = 7500 \text{ meters i.e. } 7.5 \text{ km}$$

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.

$$\text{Minimum antenna height} = \frac{\lambda}{4} = \frac{c}{4f} = \frac{3\times10^8}{4\times10\times10^6} = 75 \text{ meter}$$

- 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 signals will be in the same frequency range i.e. 0 to 20 kHz.
- Therefore all the signals 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 then they will occupy different slots in the frequency domain (different channels).
- This is as shown in Fig. 2.3.1. Thus modulation avoids mixing of signals.

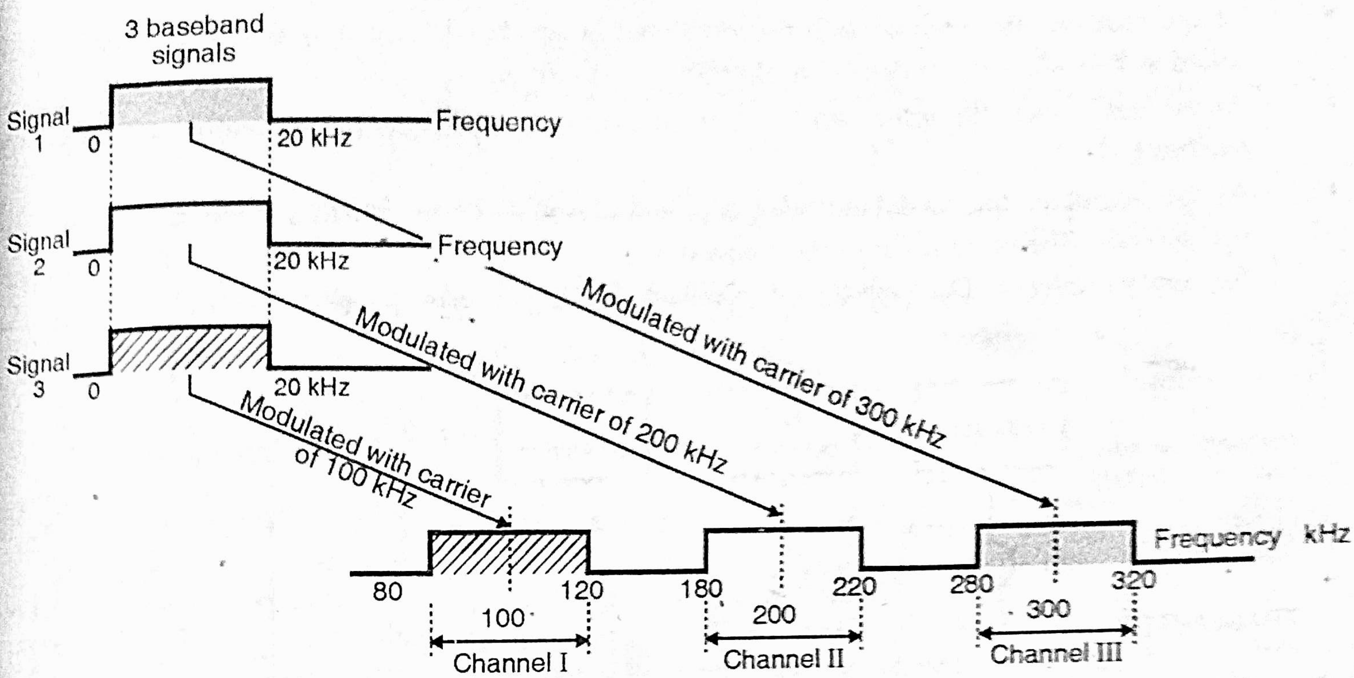


Fig. 2.3.1 : Modulation avoids mixing of signals

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 heavily attenuated (suppressed).
- 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 to be transmitted. Hence it increases the range of communication.
- In addition to the advantages discussed till now, the modulation process has some more advantages.

Multiplexing is 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.

together is called as frequency translation.

2.5 Different Types of Modulation Systems :

Various types of practically used modulation systems are as follows :

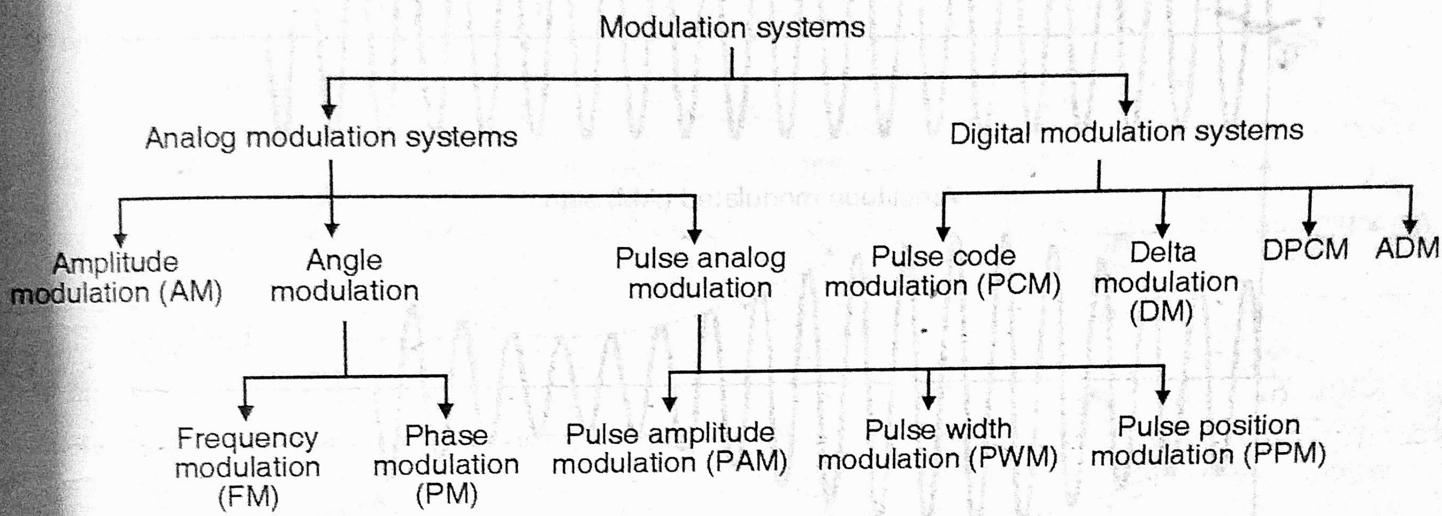
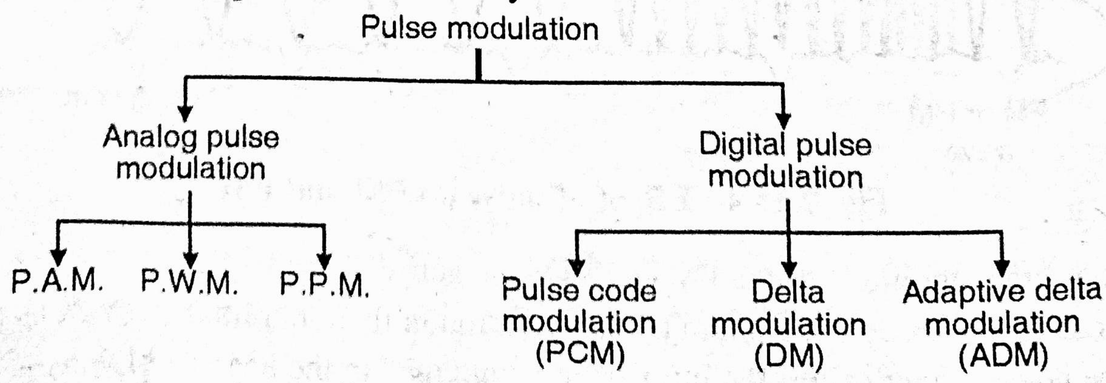


Fig. 2.5.1 : Classification of modulation systems

2.16 Pulse Modulation :

- In pulse modulation, the carrier is in the form of train of periodic rectangular pulses.
- Pulse modulation can be either analog or digital.
- In the analog pulse modulation, the amplitude, width or position of the carrier pulses is changed in accordance with the modulating signal.
- This will result in PAM (pulse amplitude modulation), PWM (pulse width modulation) or PPM (pulse position modulation) respectively.
- PAM, PWM and PPM the examples of analog pulse modulation.
- The pulse modulation can be digital as well. The well known examples of digital pulse modulation are pulse code modulation (PCM), delta modulation (DM), adaptive delta modulation (ADM), etc.
- The classification of the pulse modulation system is as follows :



- In this chapter we are going to discuss only the analog pulse modulation schemes.

1. **Pulse Amplitude Modulation (PAM) :**

The amplitude of a constant width, constant position is varied in proportion with the instantaneous magnitude of the modulating signal as shown in Fig. 2.16.1(c).

2. **Pulse Width Modulation (PWM) :**

- The width of carrier pulses is made to vary in proportion with the instantaneous magnitude of the modulating signal as shown in Fig. 2.16.1(d).
- PWM is also called as pulse duration modulation (PDM) or pulse length modulation (PLM).

3. **Pulse Position Modulation (PPM) :**

- In PPM the amplitude and width of the pulses is kept constant but the position of each pulse is varied in accordance with the amplitudes of the sampled values of the modulating signal. The position of the pulses is changed with respect to the position of reference pulses.
- The PPM pulses can be derived from the PWM pulses as shown in Fig. 2.16.1(e). Note that with increase in the modulating voltage the PPM pulses shift further with respect to reference.

4. **Pulse Code Modulation (PCM) :** ➤➤➤ **[Asked in Exam : Oct. 2004 !!!]**

- The analog message signal is sampled and converted to a fixed length, serial binary number as shown in Fig. 2.16.1(f).

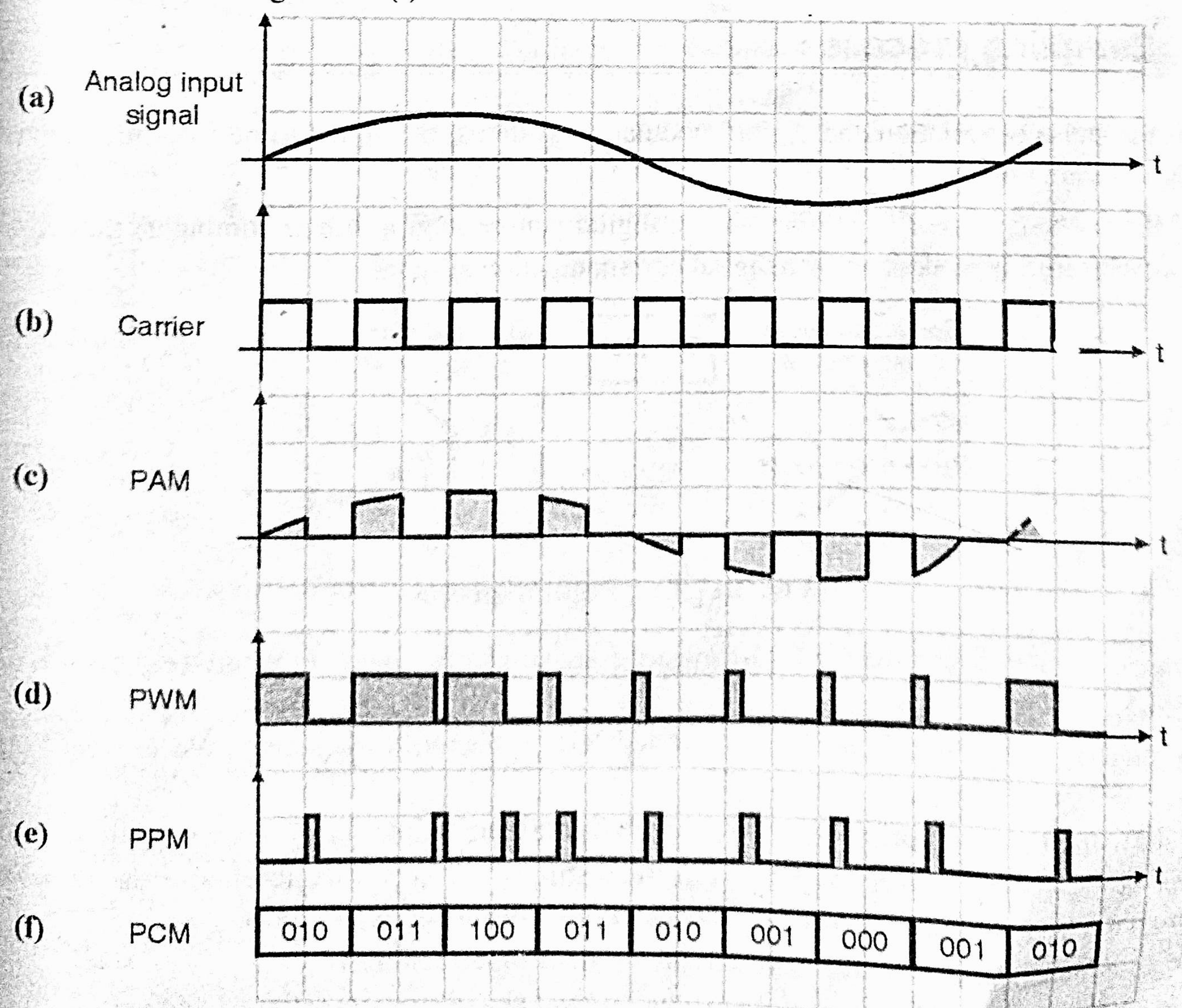


Fig. 2.16.1 : Pulse modulation

- In other words a binary code is transmitted. Hence the name pulse code modulation.
- The PAM, PWM and PPM are called as the analog pulse communication systems whereas PCM, delta modulation (DM) are the examples of digital pulse communication systems.

What is the difference between analog pulse communication and digital pulse communication ?

- For analog as well as digital pulse communication systems, the transmitted signal is a discrete time signal.
- In analog pulse communication, the information is transmitted in the form of change in amplitude, width or position of the carrier pulses. So the transmitted pulsed signal is still an analog signal.
- In digital pulse communication, the information is transmitted in the form of codes. Codewords are formed by grouping the digital pulses.
- Note that we do not change amplitude, frequency or phase of the transmitted signal. Thus the transmitted signal in digital pulse communication is a digital signal.

Practical use :

- PAM does not have a good noise immunity. So its practical use is restricted.
- PWM and PPM are used for some military applications but are not used for commercial communication applications.
- PCM is the most useful method of all.

2.17 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 processing by a digital communication system.

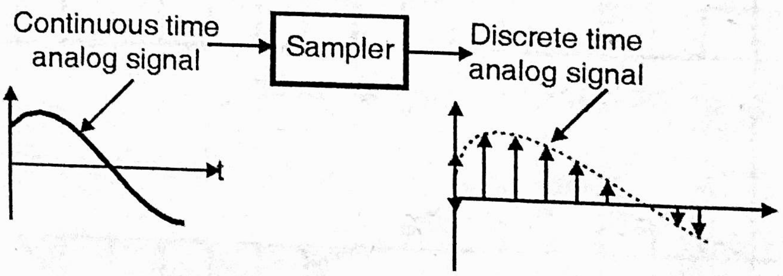


Fig. 2.17.1 : Sampling process

- 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. 2.17.1 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.

2.17.1 Sampling Theorem for Low Pass Signals :

➤➤➤ [Asked in Exam : Oct. 2005, Oct. 2007 !!!]

- 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.

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

2.18 Pulse Amplitude Modulation (PAM) ;

➤➤➤ [Asked in Exam : Oct. 2005, Oct. 2007 !!!]

- In the PAM system, the amplitude of the pulsed carrier is changed in proportion with the instantaneous amplitude of the modulating signal x(t). So the information is contained in the amplitude variation of PAM signal.
- The carrier is in the form of train of narrow pulses as shown in Fig. 2.18.1.
- If you compare the PAM system with the sampling process, you will find that these two processes are identical.
- The PAM signal is then sent by either wire or cable or it is used to modulate a carrier.

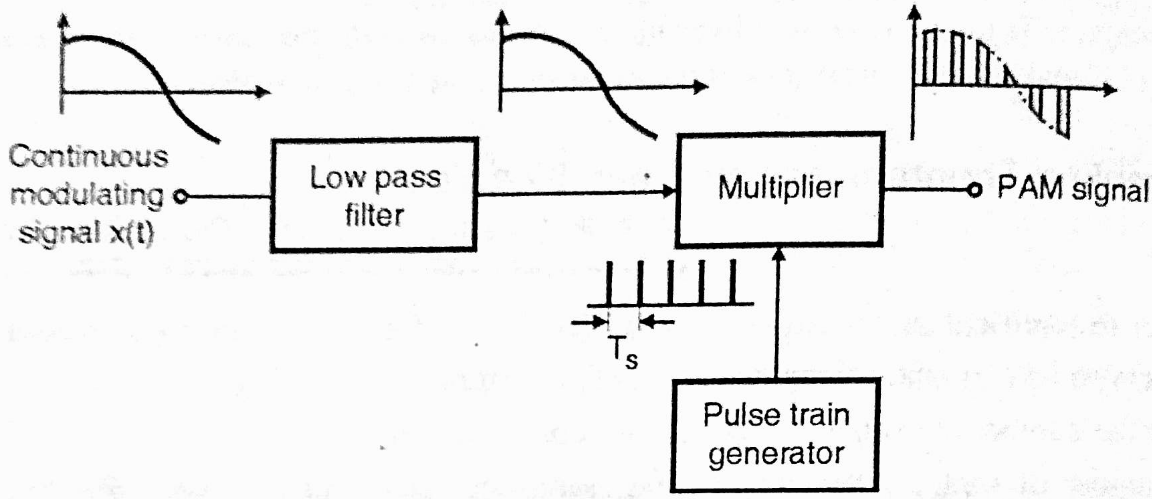


Fig. 2.18.1 : Generation of PAM

Types of PAM :

There are two types of PAM :

1. Natural PAM
2. Flat top PAM

2.19 Pulse Code Modulation (PCM) :

>>> [Asked in Exam : Oct. 2006 ...

- PCM is a type of pulse modulation like PAM, PWM or PPM but there is an important differe... between them. PAM, PWM or PPM are "analog" pulse modulation systems whereas PCM is "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 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.

2.19.1 PCM Transmitter (Encoder) :

>>> [Asked in Exam : April 2006 ...

Block diagram of the PCM transmitter is as shown in Fig. 2.19.1.

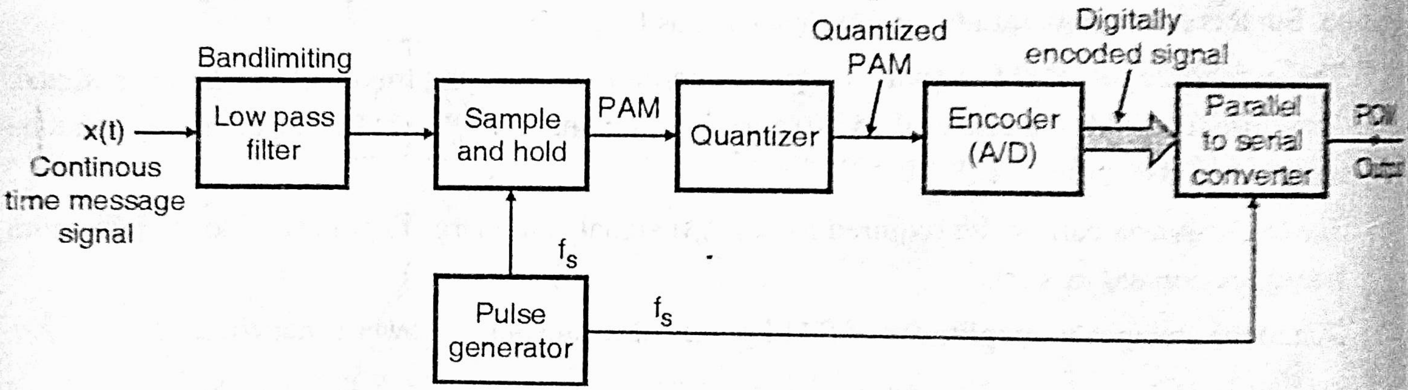


Fig. 2.19.1 : PCM transmitter (Encoder)

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. 2.19.2.

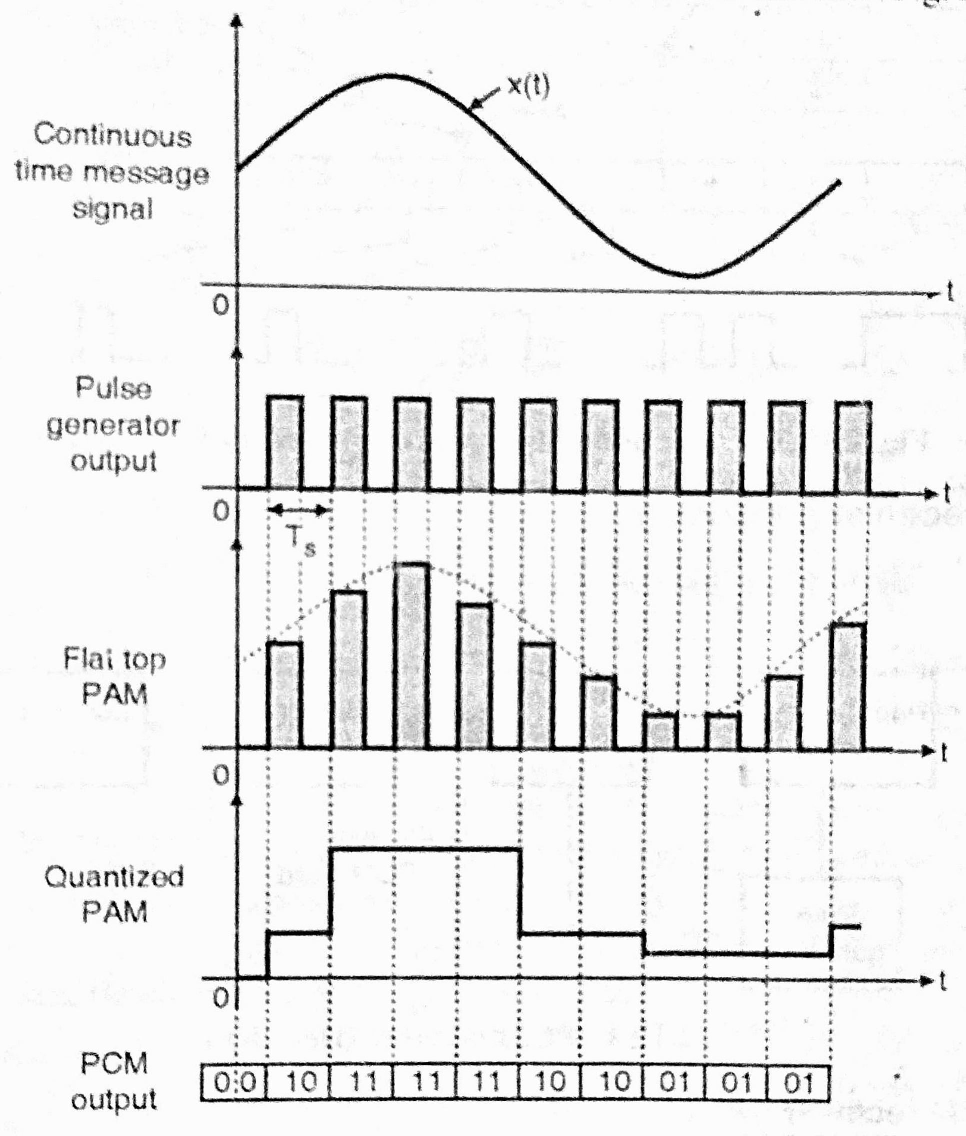


Fig. 2.19.2 : Waveforms at different points in PCM transmitter

Fig. 2.19.3 : Input and output waveforms of a PCM system

2.19.3 PCM Receiver (Decoder) :

Fig. 2.19.4 shows the block diagram of a PCM receiver.

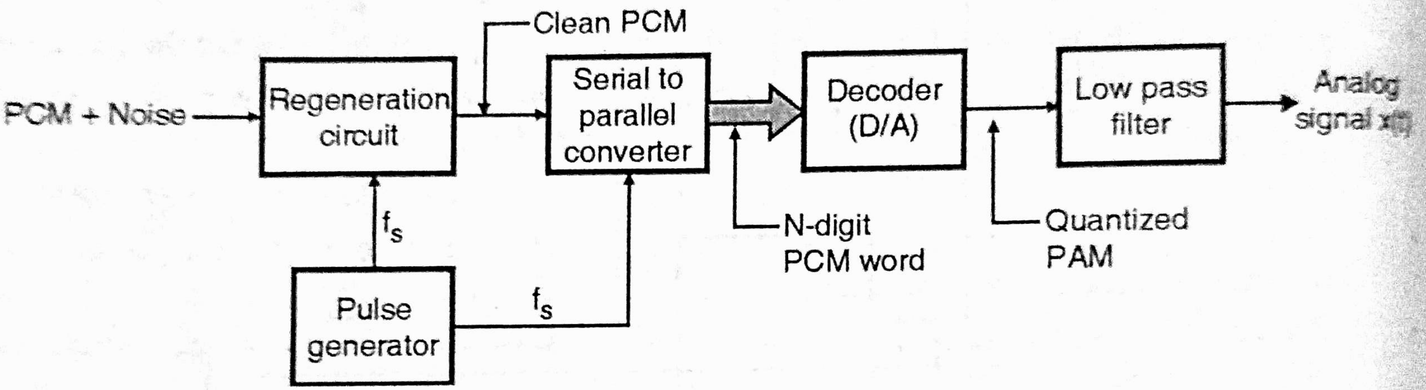


Fig. 2.19.4 : PCM receiver (Decoder)

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 wi reconstruct the original PCM signal.
- The pulse generator has to operate in synchronization with that at the transmitter. Thus at th regeneration circuit output we get a "clean" PCM signal.
- The reconstruction of PCM signal is possible due to the digital nature of PCM signal. Th 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 encode
- The decoder output is the sequence of a quantized multilevel pulses. The quantized PAM sig 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 th message bandwidth W.

2.19.4 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. 2.19.5 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,

$$\boxed{s = \frac{V_H - V_L}{Q}} \quad \text{...(2.19.1)}$$

In Fig. 2.19.5, the value of Q = 8

- At the center of these ranges, the quantization levels q_0, q_1,...q_7 are placed. Thus the number of quantization levels is Q = 8. These 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 Δ_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 Δ_1, the quantizer output is constant equal to "q_1".
- Thus in each range from Δ_0 to Δ_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.**

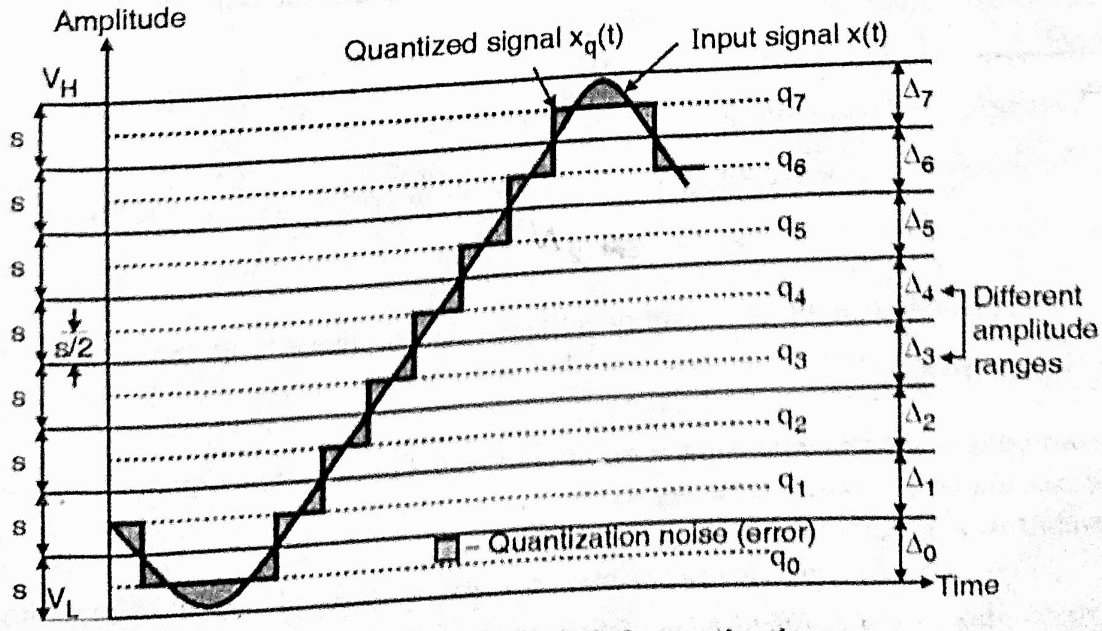


Fig. 2.19.5 : Process of quantization

- 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.

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.

2.19.5 Quantization error or quantization noise $\in$:

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

$$\in \; = \; x_q(t) - x(t) \qquad ...(2.19.2)$$

- The quantization error is shown by shaded portions of the waveform in Fig. 2.19.5.
- 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,

$$\text{Mean square value of quantization error} = \frac{s^2}{12} \qquad ...(2.19.3)$$

The derivation for this expression is given later in this chapter.

- 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.

$$\therefore \quad Q = \cancel{2N}\; 2^N \qquad ...(2.19.4)$$

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

2.20 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.

2.20.1 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.

2.20.2 Advantages of PCM :

1. Very high noise immunity.
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 person 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.

2.20.3 Disadvantages of PCM :

➤➤➤ [Asked in Exam : Oct. 2006 !!!]

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

2.21 Linear Delta Modulation (D.M.) :

➤➤➤ [Asked in Exam : April 2004, April 2006, April 2007, April 2008, Oct. 2008 !!!]

- In PCM system, N number of binary digits are transmitted per quantized sample. Hence the signaling rate and transmission channel bandwidth of the PCM system are very large.
- These disadvantages can be overcome by using the delta modulation.

Principle of operation :

- Delta modulation transmits only one bit per sample instead of N bits transmitted in PCM. This reduces its signaling rate and bandwidth requirement to a great extent.

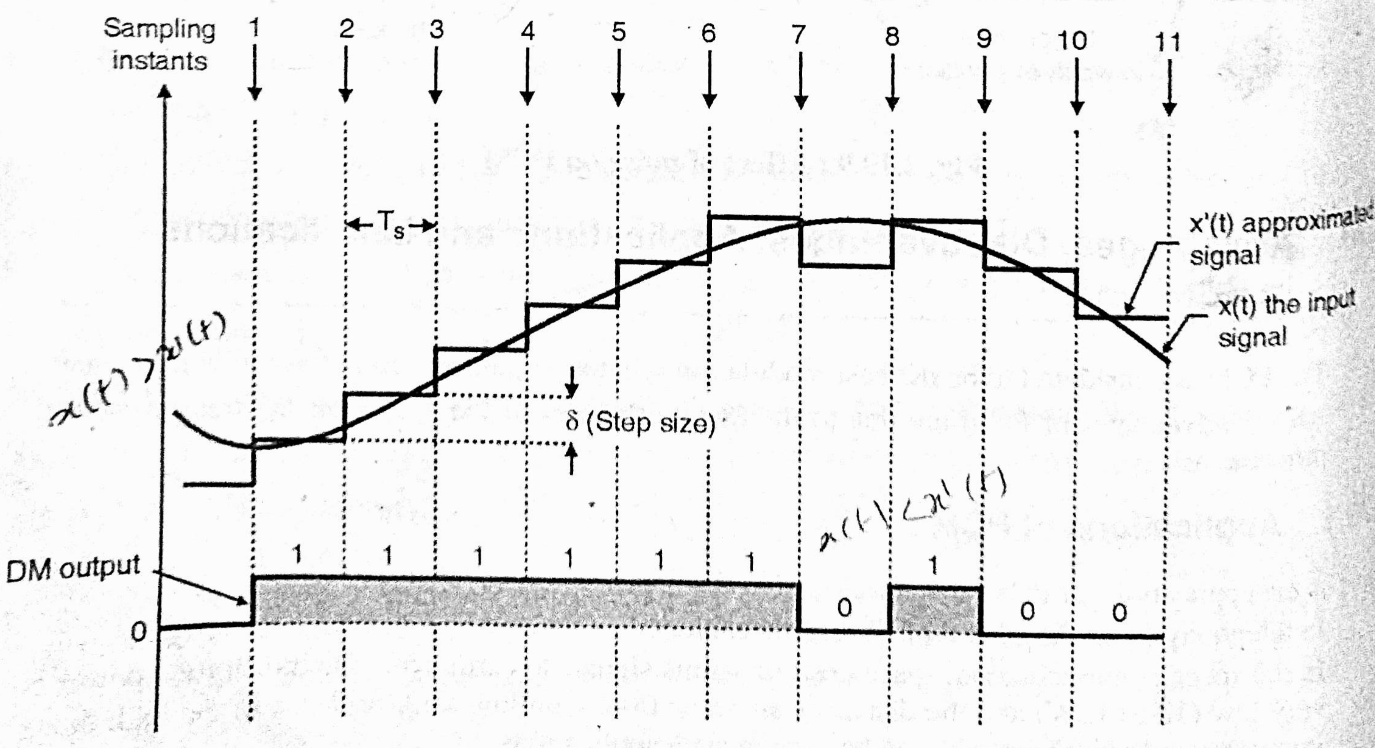


Fig. 2.21.1 : D.M. Waveforms

- In the basic or linear D.M., a staircase approximated version of the sampled input signal is produced as shown in Fig. 2.21.1.
- The original signal and its staircase representation are then compared to produce a difference signal.
- And this difference signal is quantized into only two levels namely $\pm \delta$ corresponding to positive and negative difference respectively.
- That means if the approximated signal $x'(t)$ lies below $x(t)$ at the sampling instant, then the approximated signal is increased by "δ". (See instants 1, 2, 3, 4, 5 and 6 in Fig. 2.21.1.)
- Whereas if $x'(t)$ is greater than $x(t)$ at the sampling instant, then $x'(t)$ is decreased by "δ" (see instants 7, 9 and 10 in Fig. 2.21.1.)

D.M. output :

- As shown in Fig. 2.21.1, the D.M. output is 1 if the staircase signal $x'(t)$ is increased by "δ" i.e. at sampling instants 1, 2, 3, 4, 5 and 6.
- Whereas D.M. output is 0 if $x'(t)$ is decreased by "δ" i.e. at sampling instants 7, 9 and 10.
- In delta modulation, the present sample value $x(t)$ is compared with the approximate value $x'(t)$ and the result of this comparison is transmitted.
- Thus we are sending the information of whether the present sample value is higher than or lower than the approximate value. Note that the actual sampled value is not being transmitted.

2.21.1 Delta Modulator Transmitter :

The block diagram of a delta modulator transmitter is as shown in the Fig. 2.21.2.

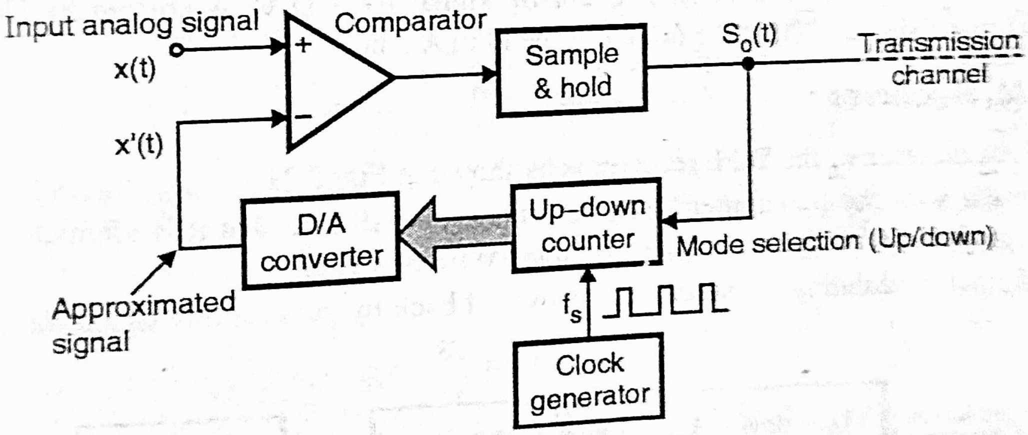


Fig. 2.21.2 : D.M. transmitter

Operation :

The operation of the circuit is as follows :

- $x(t)$ is the analog input signal and $x'(t)$ is the quantized (approximated) version of $x(t)$. Both these signals are applied to a comparator.
- The comparator output goes high if $x(t) > x'(t)$ and it goes low if $x(t) < x'(t)$. Thus the comparator output is either 1 or 0. The sample and hold circuit will hold this level (0 or 1) for the entire clock cycle period.
- The output of the sample and hold circuit is transmitted as the output of the DM system. Thus in DM, the information which is transmitted is only whether $x(t) > x'(t)$ or vice versa. Also note that one bit per clock cycle is being sent. This will reduce the bit rate and hence the BW.

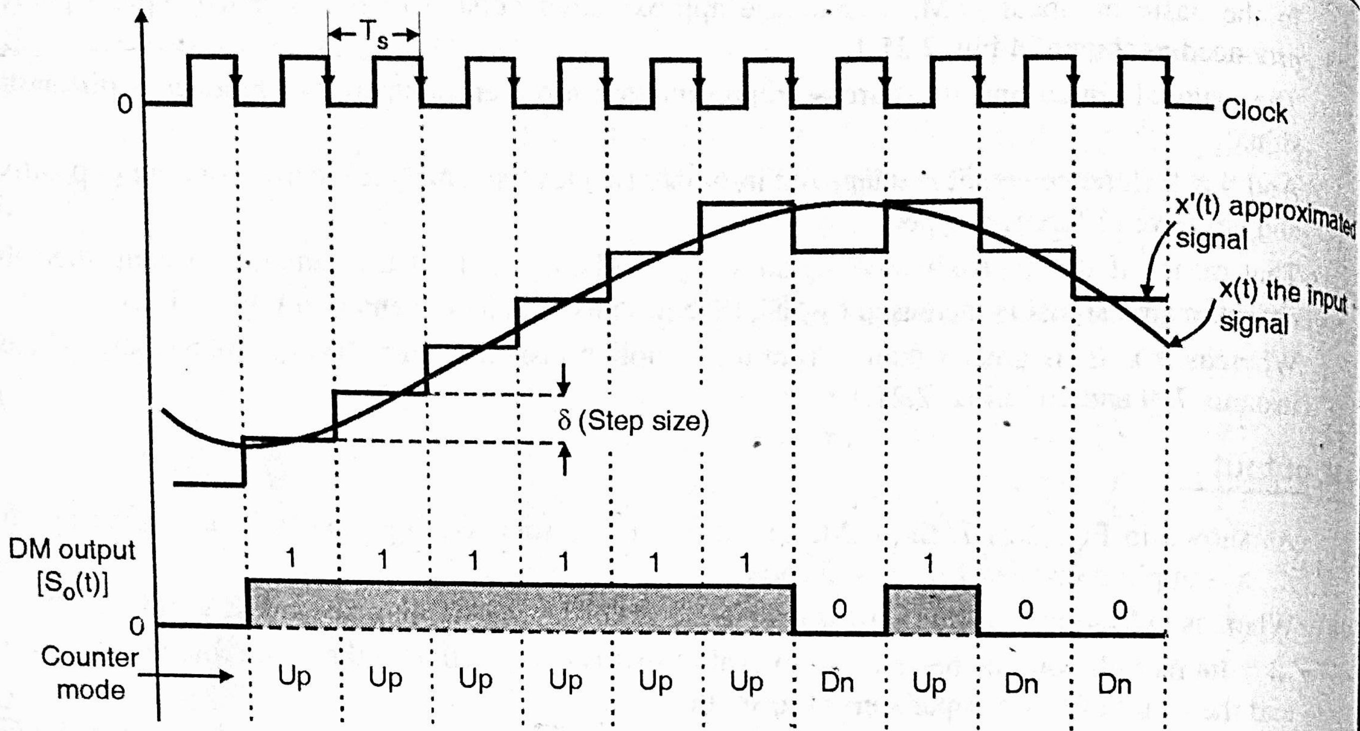


Fig. 2.21.3 : D.M. waveforms

- The transmitted signal is also used to decide the mode of operation of an up/down counter. The counter output increments by 1 if $S_o(t) = 1$ and it decrements by 1 if $S_o(t) = 0$, at the falling edge of each clock pulse. This is as shown in the waveform in the Fig. 2.21.3.
- The counter output is converted into analog signal by a D to A converter. Thus we get the approximated signal x′ (t) at the output of the D to A converter.

2.21.2 D.M. Receiver :

- The block diagram of the D.M. receiver is as shown in Fig. 2.21.4.
- Compare it with the transmitter block diagram, you will find that it is identical to the chain of blocks producing the signal x′ (t) i.e. the approximated signal.
- The original modulating signal can be recovered back by passing this signal through a low pass filter.

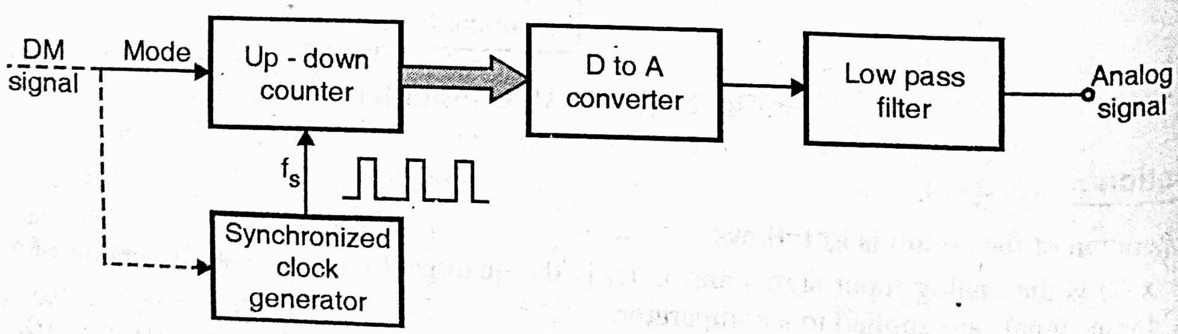


Fig. 2.21.4 : D.M. receiver

Comparison of D.M. and DPCM :

- The comparison of D.M. and DPCM systems revels that except for an output low pass filter, they are identical.
- So D.M. is actually a special case of DPCM.

2.21.3 Features of D.M. :

- A one bit codeword for output. Hence no need of framing.
- Simplicity of design for transmitter and receiver.

2.21.4 Applications of D.M. :

- For some types of digital communications.
- For digital voice storage.

2.21.5 Distortions in the DM System :

The DM system is subjected to two types of quantization error :

1. Slope overload distortion and 2. Granular noise.

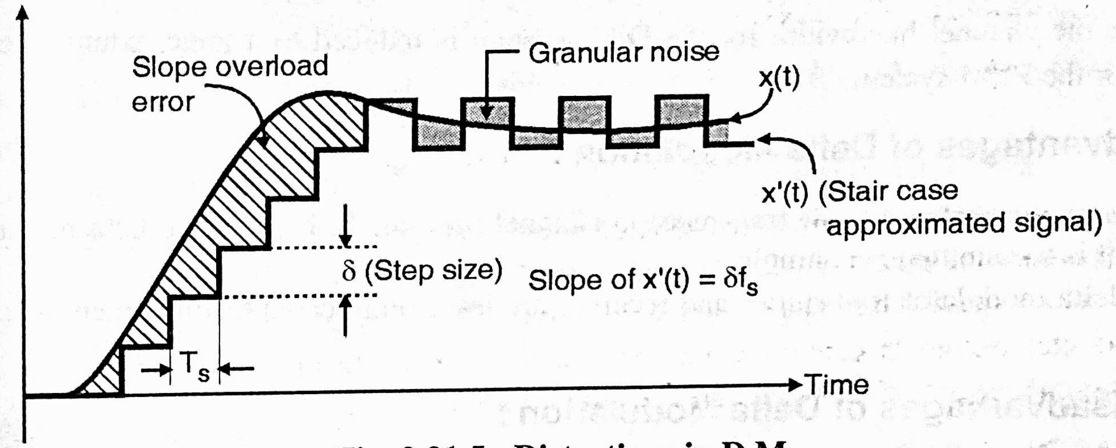


Fig. 2.21.5 : Distortions in D.M.

1. Slope overload distortion :

- Look at the Fig. 2.21.5 Due to small step size (δ), the slope of the approximated signal $x'(t)$ will be small.

$$\text{The slope of } x'(t) = \frac{\delta}{T_s} = \delta f_s \qquad \text{...(2.21.1)}$$

- If slope of the analog signal $x(t)$ is much higher than that of $x'(t)$ over a long duration then $x'(t)$ will not be able to follow $x(t)$, at all.
- The difference between $x(t)$ and $x'(t)$ is called as the slope overload distortion.
- Thus the slope overload error occurs when slope of $x(t)$ is much larger than slope of $x'(t)$.
- The slope overload error can be reduced by increasing slope of the approximated signal $x'(t)$.
- Slope of $x'(t)$ can be increased and hence the slope overload error can be reduced by either increasing the step size "δ" or by increasing the sampling frequency f_s.
- However with increase in δ the granular noise increases and if f_s is increased, signaling rate and bandwidth requirements will go up.

2. Granular noise :

- When the input signal $x(t)$ is relatively constant in amplitude, the approximated signal $x'(t)$ will hunt above and below $x(t)$ as shown in Fig. 2.21.5.
- The granular noise is similar to the quantization noise in the PCM system.

- It increases with increase in the step size δ. To reduce the granular noise, the step size should be as small as possible.
- However this will increase the slope overload distortion.
- In the linear delta modulator the step size δ is not variable. If it is made variable then the slope overload distortion and granular noise both can be controlled.
- A system with a variable step size is known as the adaptive delta modulator (ADM).

2.21.6 D.M. Bit Rate (Signaling Rate) :

- D.M. bit rate (r) = Number of bits transmitted / second

  = Number of samples/sec × Number of bits/sample = $f_s \times 1 = f_s$ ...(2.21.2)
- Thus the D.M. bit rate is (1/N) times the bit rate of a PCM system, where N is the number of bits per transmitted PCM codeword.
- Hence the channel bandwidth for the D.M. system is reduced to a great extent as compared to that for the PCM system.

2.21.7 Advantages of Delta Modulation :

1. Low signaling rate and low transmission channel bandwidth, because in delta modulation, only one bit is transmitted per sample.
2. The delta modulator transmitter and receiver are less complicated to implement as compared to PCM.

2.21.8 Disadvantages of Delta Modulation :

1. The two distortions discussed earlier i.e. slope overload error and granular noise are present.
2. Practically the signaling rate with no slope overload error will be much higher than that of PCM.

The slope overload error can be reduced by using another type of delta modulation, called as adaptive delta modulation (ADM).