Digital transmission is defined as the transmission of digital signals between two or more points in a communication system. The transmitted signal is the form of a digital signal.

The digital signals can be of different types such as binary, octal, hexadecimal etc.

The original information can be analog or digital. If it is analog then it is converted to digital.

The communication medium can be a coaxial cable or optical fiber link.

Examples of digital pulse modulation are PCM, DM and ADM.

digital representation of a signal has following advantages :

Immunity to transmission noise and interference.

Regeneration of the coded signal along the transmission path is possible.

Communication can be kept "private" and "secured" through the use of encryption.

It is possible to use a uniform format for different kinds of baseband signals.

It is possible to store the signal and process it further.

Digital signals are better suited for processing and multiplexing.

Digital transmission systems are more immune to noise.

Measurement and evaluation of digital signals is simpler.

It is possible to evaluate error performance of digital systems.

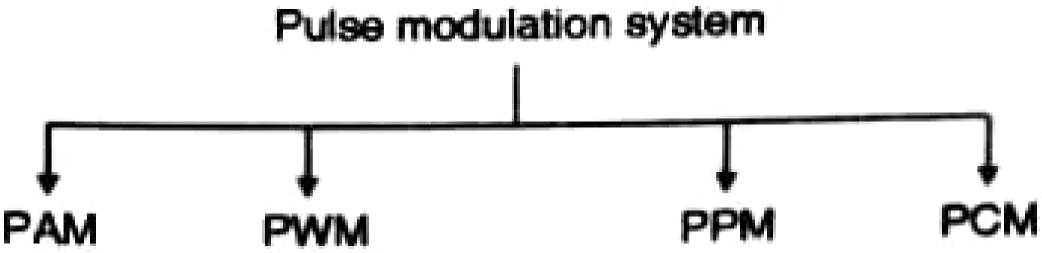
The required bandwidth is increased due to digital technology.

2. System complexity is increased.
3. In order to convert the analog signal to digital prior to transmission and then from digital analog at the receiver, we need to use the additional encoders and decoder circuits.
4. Synchronization is necessary for the digital systems (between transmission and receiver clocks
5. Digital transmission systems are not compatible to the older analog transmission systems.

10.1.3 Pulse Modulation :

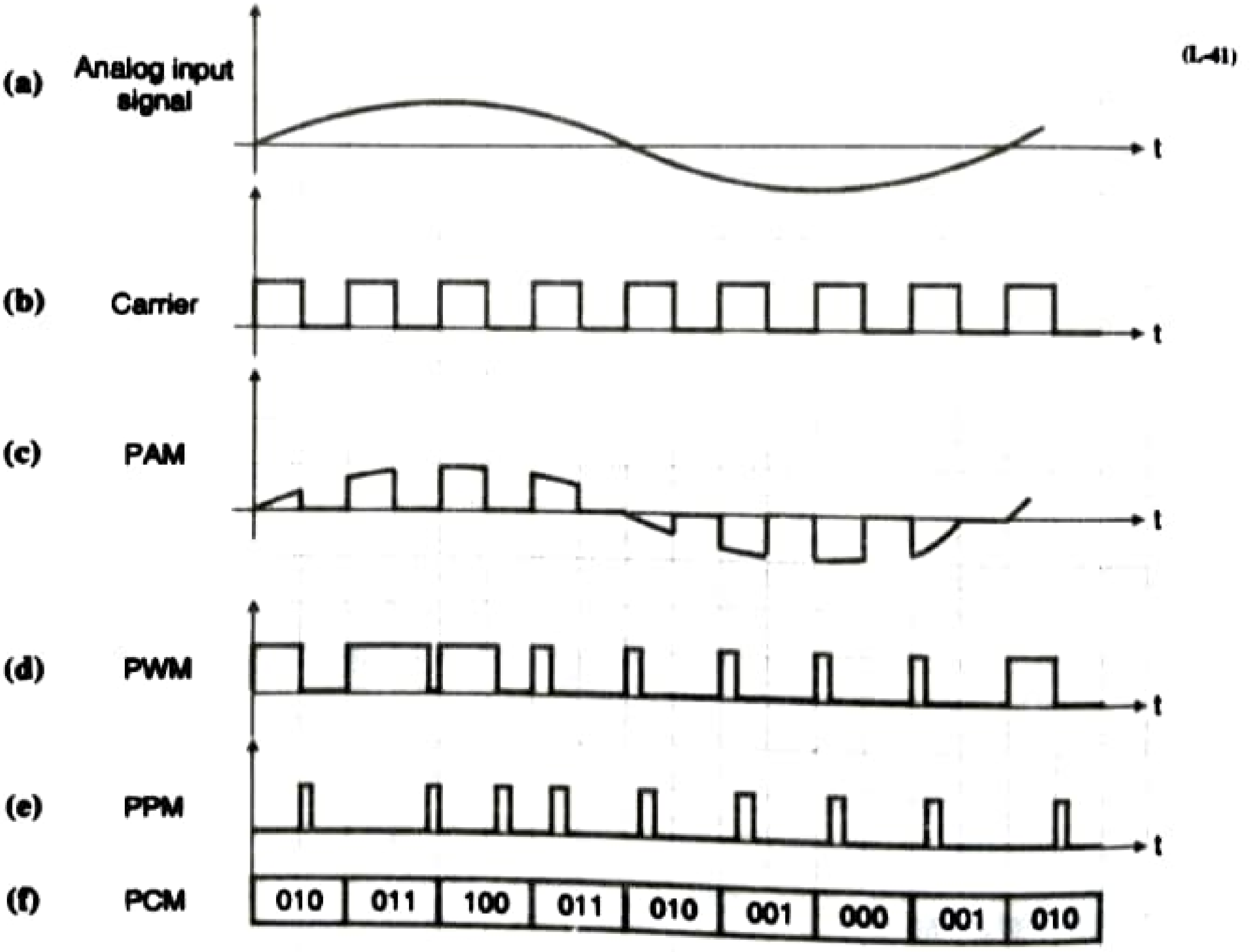
- Pulse modulation is defined as the technique to convert information into pulse form.
- Following are the important types of pulse modulation systems.

(D-1427)



se Code Modulation (PCM) :

The analog message signal is sampled and converted to a fixed length, serial binary number as shown in Fig. 10.1.1(f).



(L-154) **Fig. 10.1.1 : 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.

...at 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 rectangular 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. Codeword are formed by grouping the digital pulses.

Note that for digital pulse communication we do not change amplitude, frequency or phase of the transmitted signal. Thus the transmitted signal in digital pulse communication is a digital signal.

...ctical 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.

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).

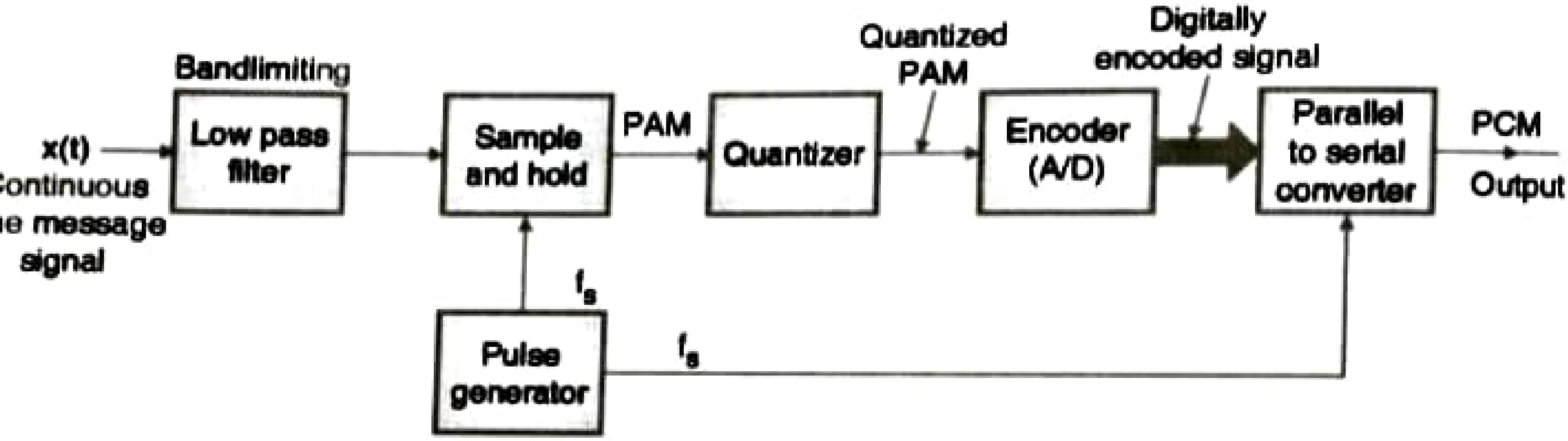
The essential operations in the PCM transmitter are sampling, quantizing and encoding.

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.

Block diagram of the PCM transmitter is as shown in Fig. 10.2.1.



(L-221) **Fig. 10.2.1 : 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.

...forms :

The waveforms at various points in the PCM transmitter are as shown in Fig. 10.2.2.

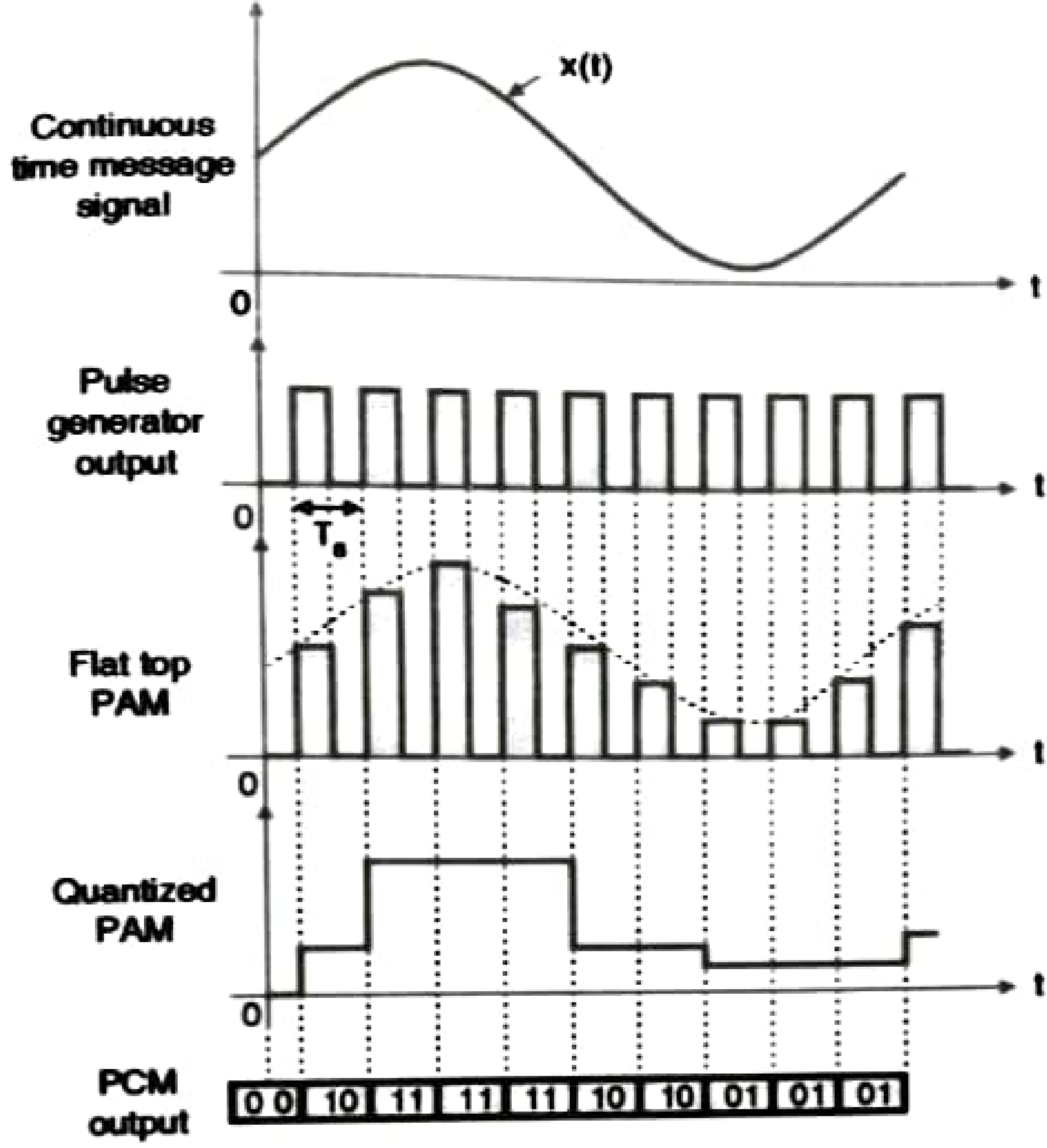
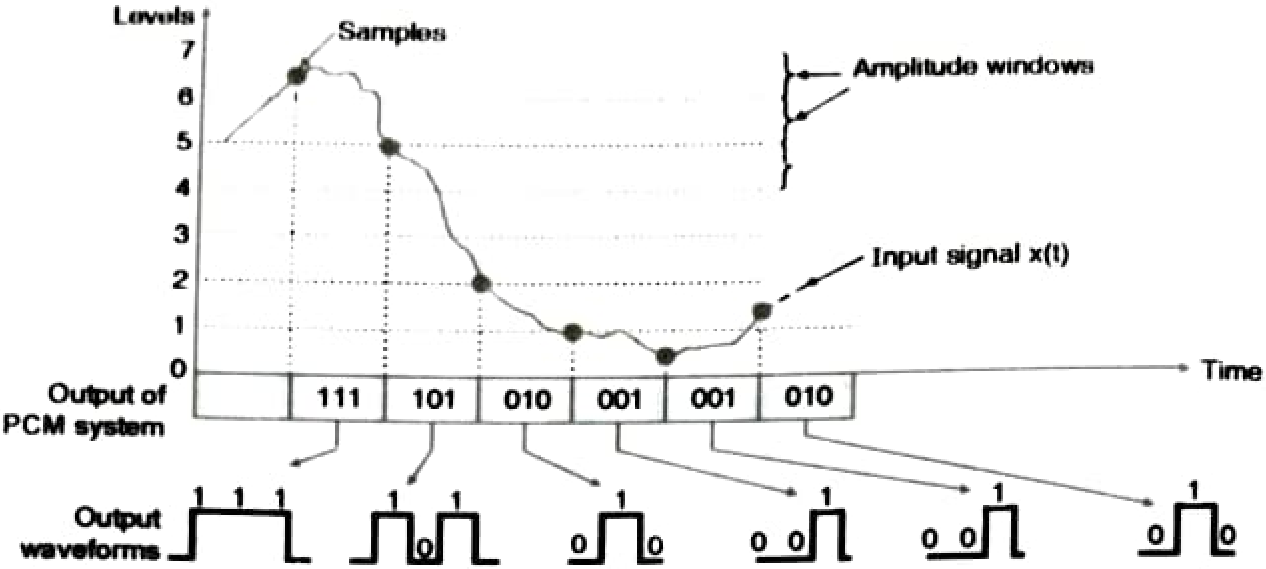


Fig. 10.2.3 shows input to and output of a PCM system. It is important to understand that the output is in the form of binary codes. Each transmitted binary code represents a particular amplitude of the input signal. Hence the "information" is contained in the "code" which is being transmitted.

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 5^{th} - window of amplitude then a digital word 101 is transmitted. If the sample is in the 2^{nd} - window then the transmitted word is 010 and so on.

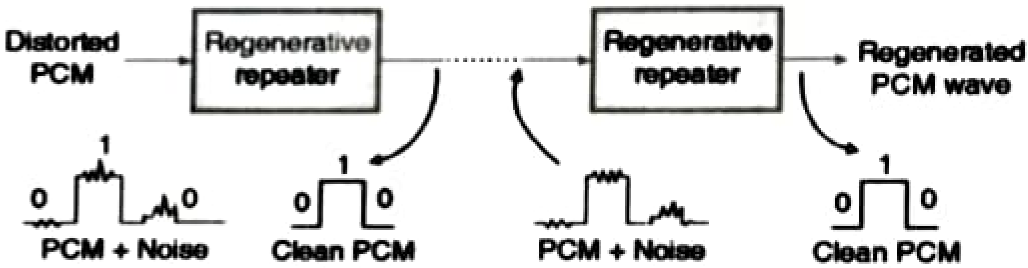
In this example we have converted the amplitudes into 3 bit codes, but in practice the number of bits per word can be as high as 8, 9 or 10.



(L-223) **Fig. 10.2.3 : Input and output waveforms of a PCM system**

10.2.3 PCM Transmission Path and Regenerative Repeaters :

The path between the PCM transmitter and PCM receiver over which the PCM signal travels is called as PCM transmission path and it is as shown in Fig. 10.2.4.



(D-472) **Fig. 10.2.4 : PCM transmission path**

The most important feature of PCM system lies in its ability to control the effects of distortion and noise when the PCM wave travels on the channel.

PCM accomplishes this capacity by means of using a chain of regenerative repeaters as shown in Fig. 10.2.4.

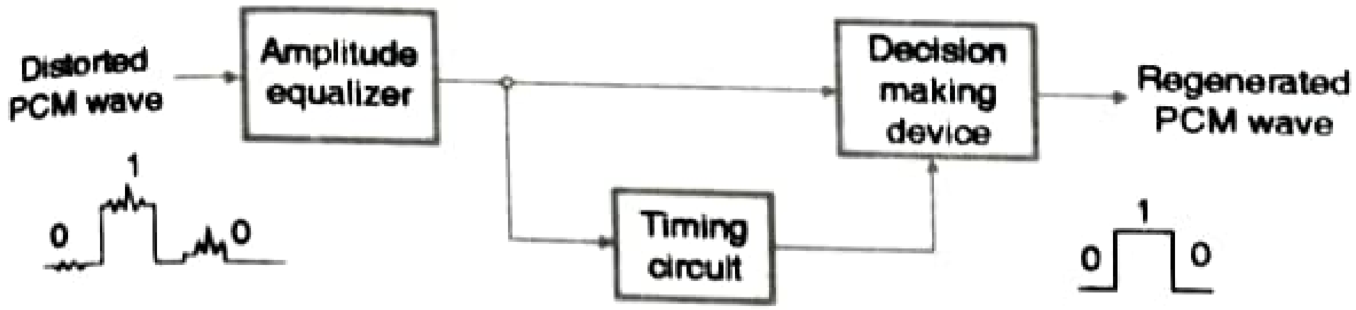
Such repeaters are spaced close enough to each other on the transmission path.

The regenerative repeater performs three basic operations namely equalization, timing and decision making.

- So each repeater actually reproduces the clean noise free PCM signal from the PCM signal distorted by the channel noise. This improves the performance of PCM in presence of noise.

Block diagram of a repeater :

- The block diagram of a regenerative repeater is shown in Fig. 10.2.5.
- The amplitude equalizer shapes the distorted PCM wave so as to compensate for the effects of amplitude and phase distortions.
- The timing circuit produces a periodic pulse train that is derived from the input PCM pulses. This pulse train is then applied to the decision making device.



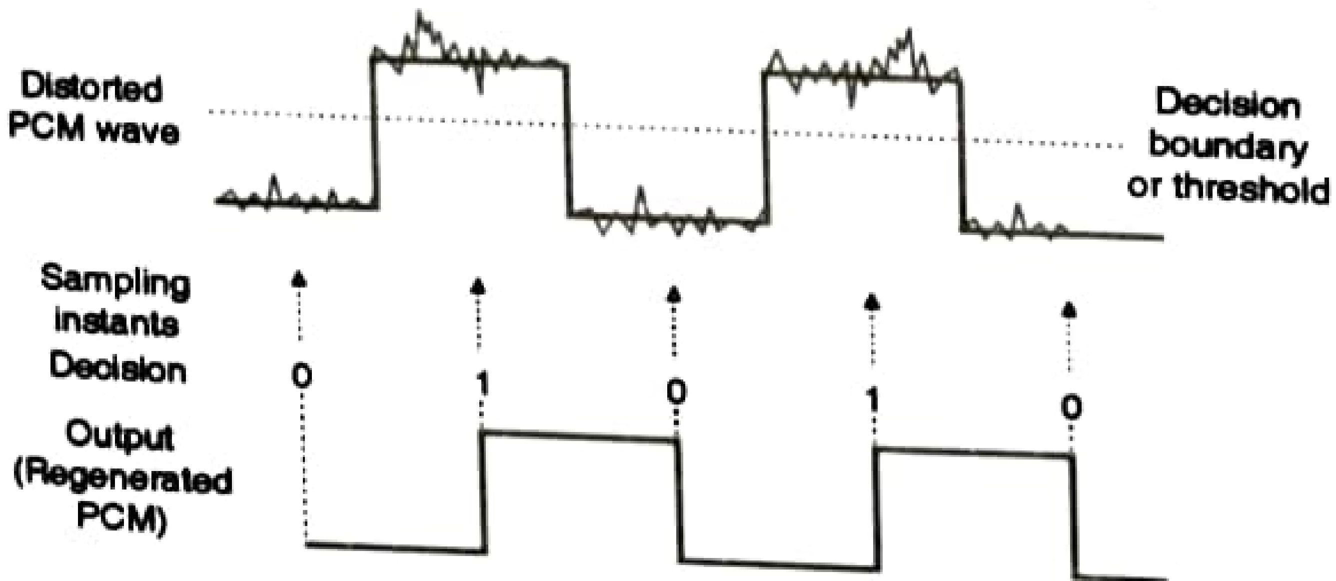
(E-667) **Fig. 10.2.5 : Block diagram of a regenerative repeater**

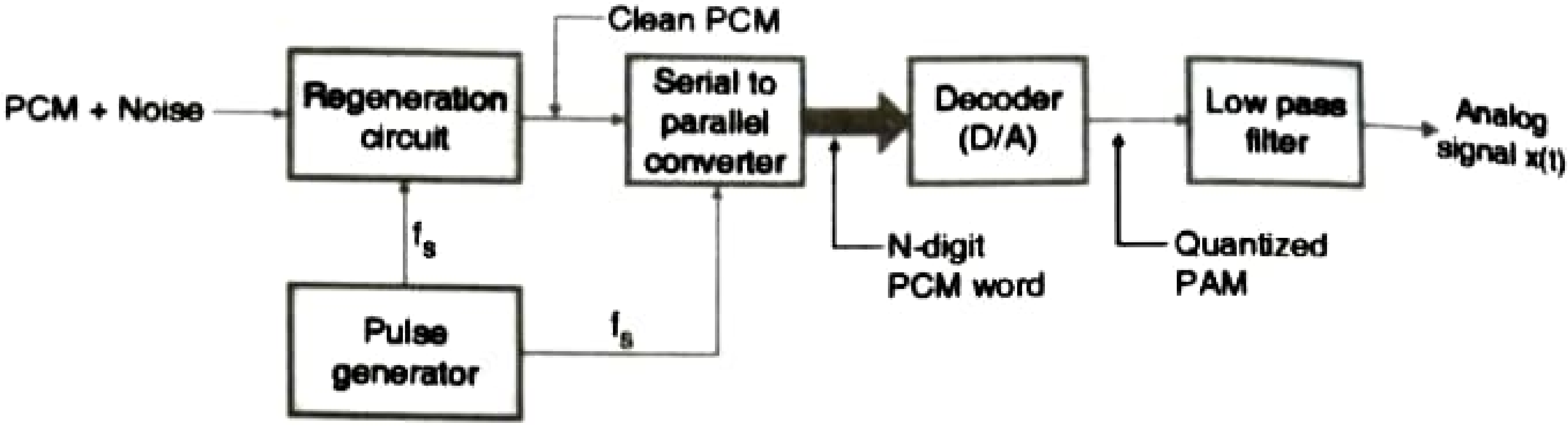
The decision making device uses this pulse train for sampling the equalized PCM pulses. The sampling is carried out at the instants where the signal to noise ratio is maximum.

The decision device makes a decision about whether the equalized PCM wave at its input has a 0 value or 1 value at the instant of sampling.

Such a decision is made by comparing equalized PCM with a reference level called decision threshold as shown in Fig. 10.2.5.

At the output of the decision device we get a clean PCM signal without any trace of noise.





(L-224) **Fig. 10.2.7 : 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.

10.2.5 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. 10.2.8 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,

$$s = \frac{V_H - V_L}{Q} \quad \text{...(10.2.1)}$$

In Fig. 10.2.8, 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. 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 Δ_0, then corresponding to any value of x (t), the quantizer output will be equal to "q_0".

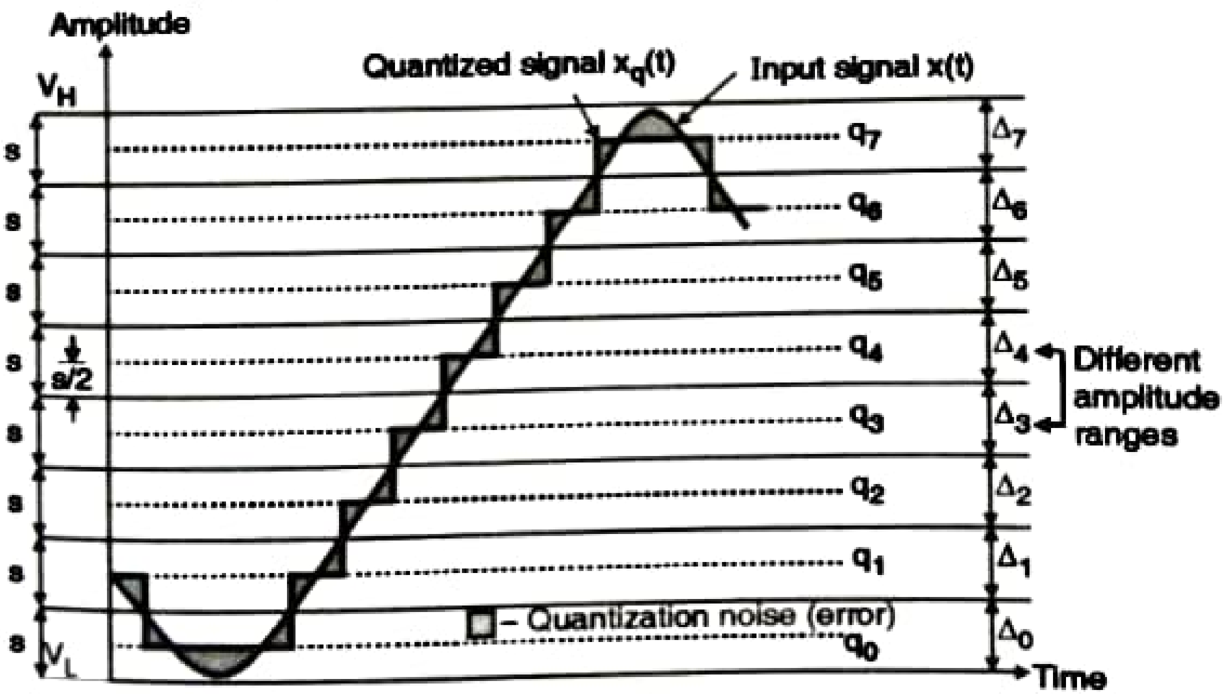
Similarly for all the values of x (t) in the range Δ_1, the quantizer output is constant equal to "q_1".

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.**

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. 10.2.8 : 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.

...antization error or quantization noise ϵ :

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

$$\epsilon = x_q(t) - x(t) \quad \text{...(10.2.2)}$$

The quantization error is shown by shaded portions of the waveform in Fig. 10.2.6.

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,

$$\text{Mean square value of quantization error} = \frac{s^2}{12} \quad \text{...(10.2.3)}$$

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.

$$\therefore \quad Q = 2N \quad \text{...(10.2.4)}$$

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

...al to quantization noise ratio (SNR_q) :

This ratio is the figure of merit for the PCM systems. The signal to quantization noise ratio with a sinusoidal input signal to the PCM system is expressed as,

$$\frac{S_i}{N_q} = [1.8 + 6N]\ \text{dB} \qquad \textbf{For a sinusoidal signal} \quad \text{...(10.2.5)}$$

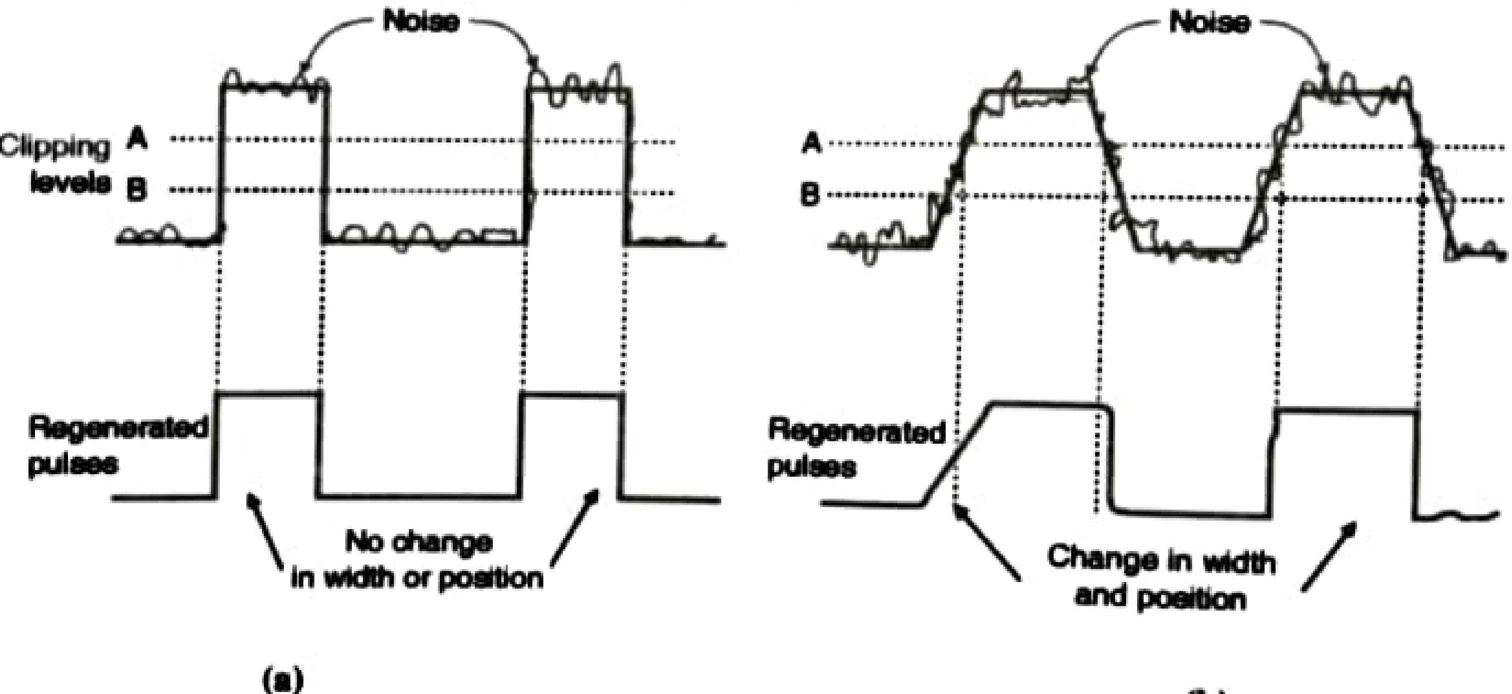
This equation shows that the signal to quantization noise ratio is solely dependent on the number of bits per word i.e. N.

This ratio should be as high as possible, which can be achieved by increasing N. But this increases the bit rate and hence bandwidth of the PCM system.

Therefore the number of bits per word is a compromise between high SNR_q and bandwidth requirements.

10.7 Effect of Noise on the PCM System :

- Look at the two Figs. 10.7.1(a) and 10.7.1(b) which illustrate the effect of noise on the transmitted pulses.
- Consider Fig. 10.7.1(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. 10.7.1(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.



(L-229) Fig. 10.7.1 : Effect of noise on PCM

10.11 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.

10.11.1 Applications of PCM :

Some of the applications of PCM are as follows :

- In digital telephone systems.
- 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.

0.11.2 Advantages of PCM :

Very high noise immunity.

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.

It is possible to store the PCM signal due to its digital nature.

It is possible to use various coding techniques so that only the desired person can decode the received signal. This makes the communication secure.

The increased channel bandwidth requirement for PCM is balanced by the improved SNR.

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.

It is easy to drop or reinsert the message sources in a PCM-TDM system.

0.11.3 Disadvantages of PCM :

The encoding, decoding and quantizing circuitry of PCM is complex.

PCM requires a large bandwidth as compared to the other systems.

0.11.4 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 $(S/N)_D$ 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.

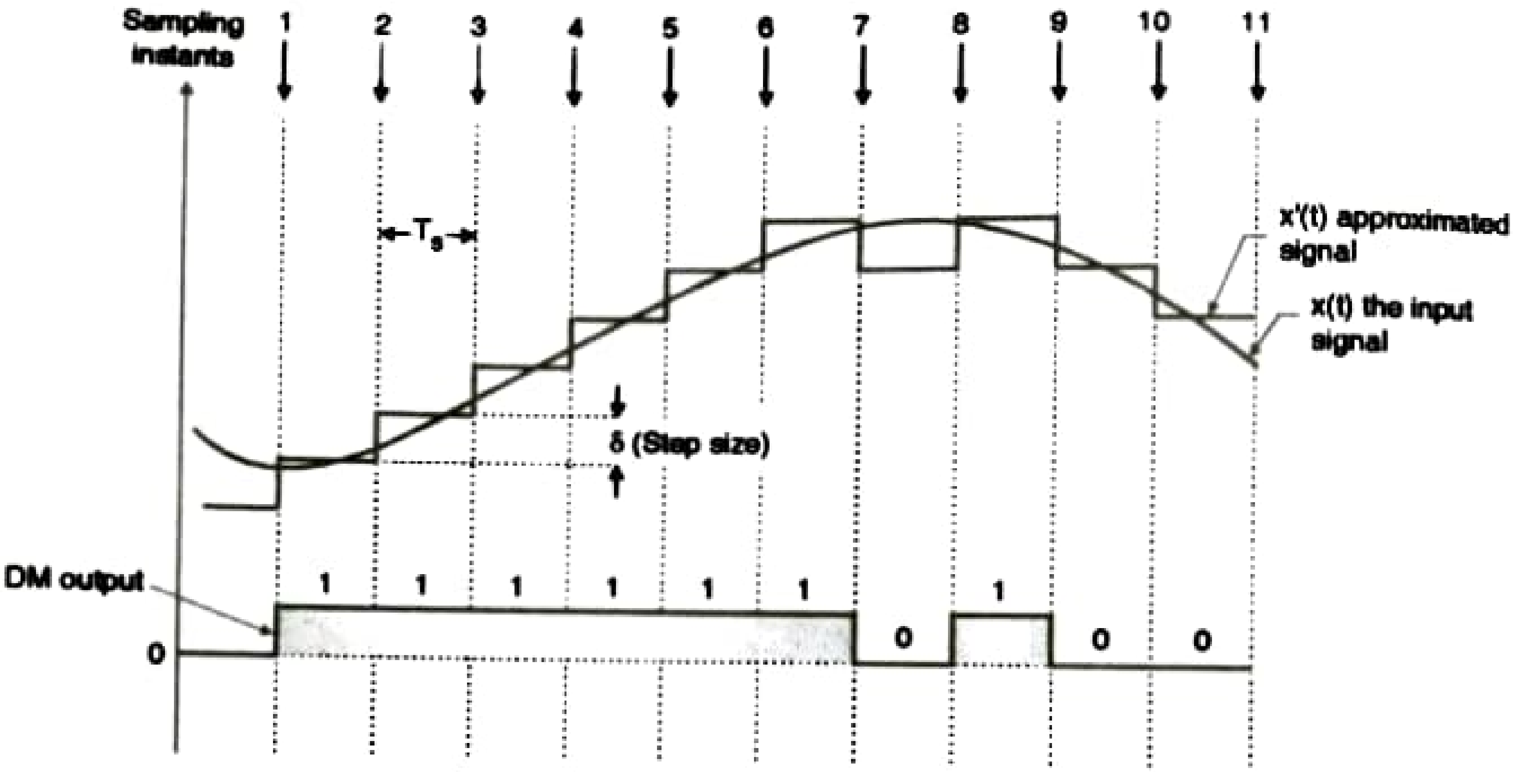
So higher bandwidth requirement and complicated circuitry are the disadvantages of PCM which do not allow its use for the radio, TV broadcasting applications.

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.

rinciple 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.



(L-235) **Fig. 10.13.1 : 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. 10.13.1.

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. 10.13.1.)

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. 10.13.1.)

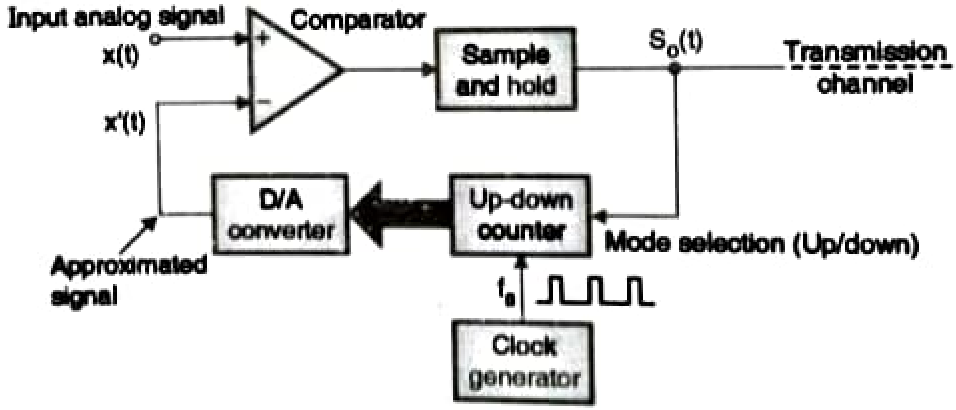
I. output :

As shown in Fig. 10.13.1, 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 appfoximate 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.



(L-236) **Fig. 10.13.4 : D.M. transmitter**

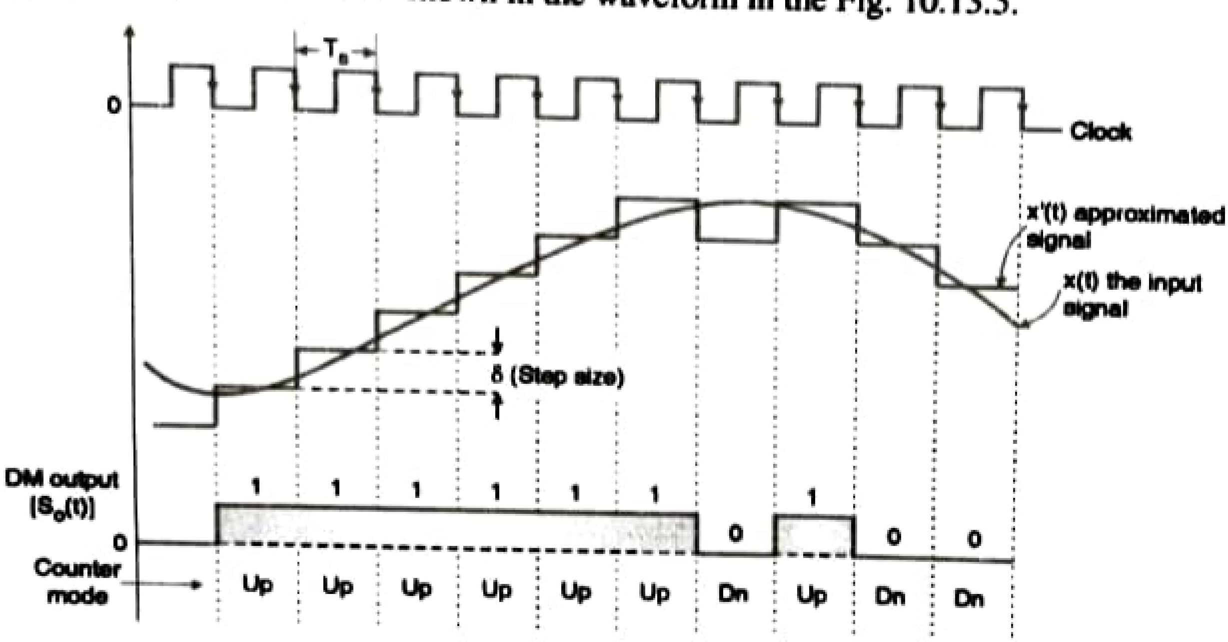
Operation of the circuit is as follows :

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.

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. 10.13.5.



(L-237) **Fig. 10.13.5 : D.M. waveforms**

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.

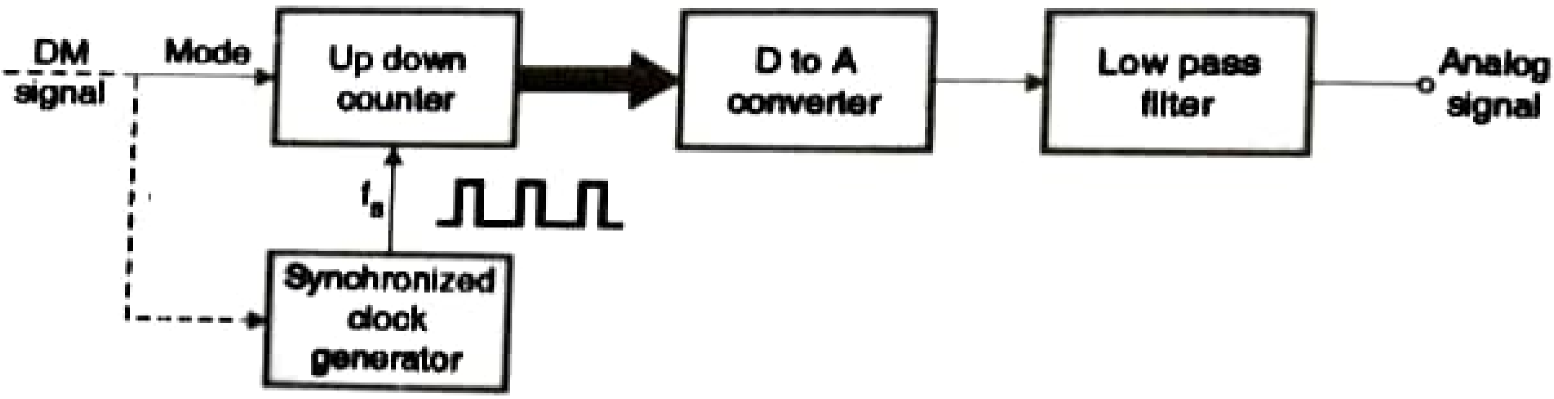
The block schematic of D.M. receiver is shown in Fig. 10.13.6.

The D.M. signal is passed through the accumulator to produce the staircase approximation in a manner similar to that used at the transmitter.

The accumulator output is then applied to a low pass filter to produce the original signal.

10.13.4 D.M. Receiver (Alternate Method) :

- The block diagram of the D.M. receiver is as shown in Fig. 10.13.7.
- 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.



(L-238) **Fig. 10.13.7 : D.M. receiver**

10.13.5 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 60 D.M. is actually a special case of DPCM.

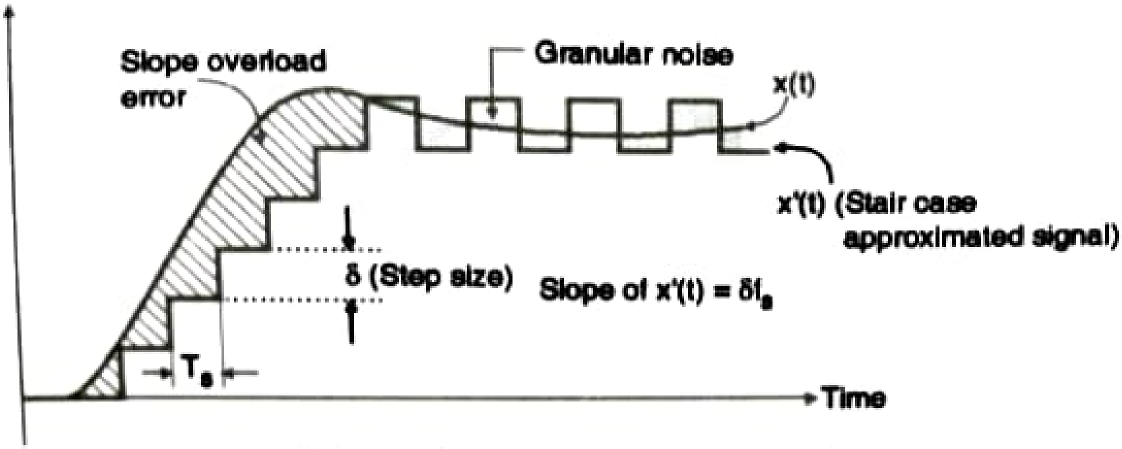
10.13.6 Features of D.M. :

- The output codeword consist of only one bit. Hence no need of framing.
- Simplicity of design for transmitter and receiver.
- Less bit rate and lower bandwidth.

3 Distortions in DM System :

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

1. Slope overload distortion and
2. Granular noise.



(L-239) **Fig. 10.13.8 : Distortions in D.M.**

Slope overload distortion :

- Look at the Fig. 10.13.8 Due to small step size (δ), the slope of the approximated signal $x'(t)$ will be small.

$$\text{The slope of } x'(t) = \frac{\delta}{T_s} = \delta f_s \qquad ...(10.13.4)$$

- If slope of the input 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 the variations in $x(t)$, 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. Thus reducing the slope overload error is not easy.

Granular noise :

- When the input signal $x(t)$ is relatively constant in amplitude, the approximated signal $x'(t)$ will fluctuate above and below $x(t)$ as shown in Fig. 10.13.8. The difference between $x(t)$ and $x'(t)$ is called as granular noise.
- 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).

0.13.9 Advantages of Delta Modulation :

Low signaling rate and low transmission channel bandwidth, because in delta modulation, only one bit is transmitted per sample.

The delta modulator transmitter and receiver are less complicated to implement as compared to PCM.

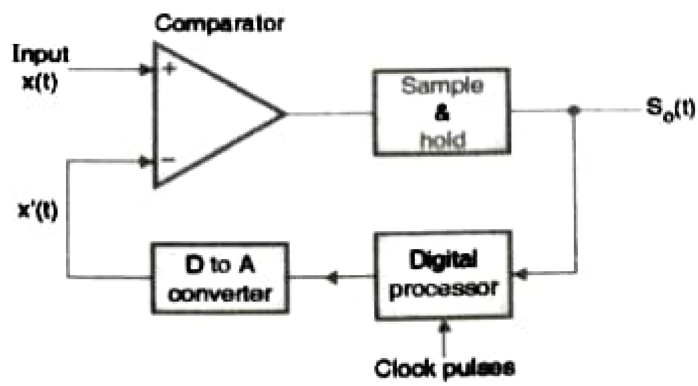
The two distortions discussed earlier i.e. slope overload error and granular noise are present.

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 ptive delta modulation (ADM).

.3 Adaptive Delta Modulation (Alternate Method) :

If you compare this block diagram with that of the linear delta modulator, then you will find that except for the counter being replaced by the digital processor, the remaining blocks are identical. Let us understand the operation of the digital processor. For that carefully see the waveforms of Fig. 10.14.3.



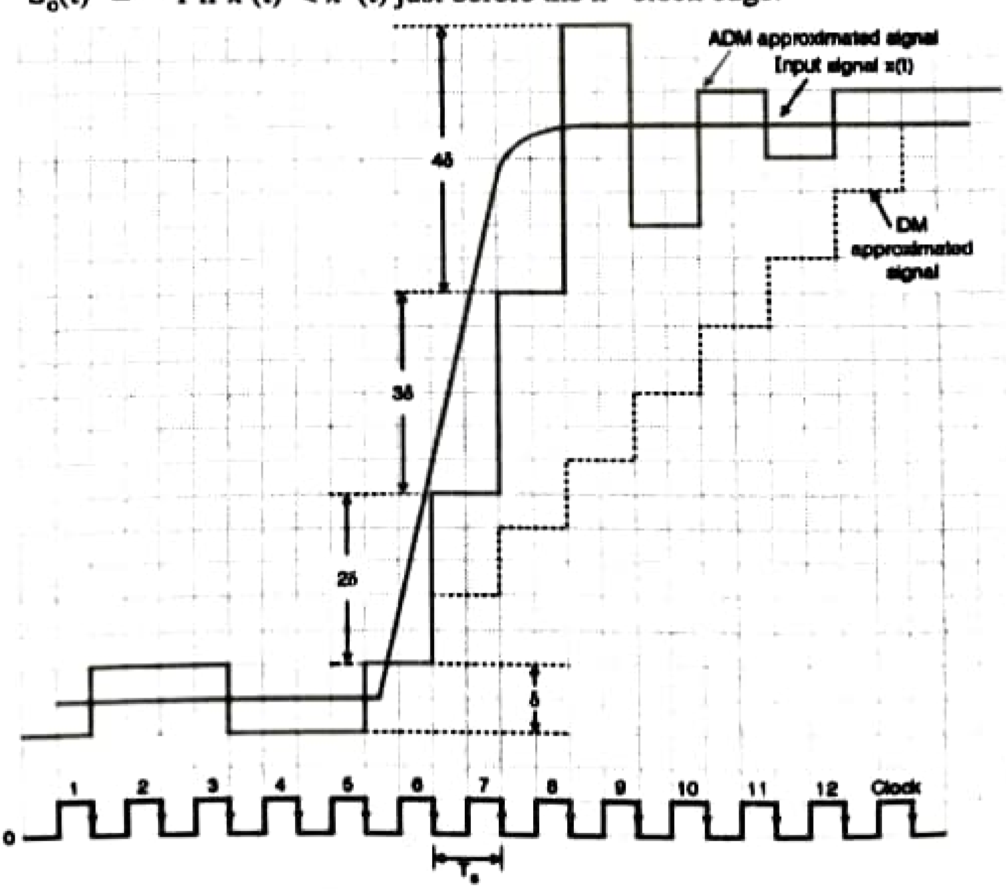
(E-712) **Fig. 10.14.3 : ADM transmitter**

Operation :

- In response to the k^{th} clock pulse trailing edge, the processor generates a step which is equal in magnitude to the step generated in response to the previous i.e. $(k-1)^{th}$ clock edge.
- If the direction of both the steps is same, then the processor will increase the magnitude of the present step by "δ". If the directions are opposite then the processor will decrease the magnitude of the present step by "δ".
- $S_o(t)$ in the Fig. 10.14.4, i.e. the output of the ADM system is given as,

$$S_o(t) = +1 \text{ if } x(t) > x'(t) \text{ just before } k^{th} \text{ clock edge.}$$

and $$S_o(t) = -1 \text{ if } x(t) < x'(t) \text{ just before the } k^{th} \text{ clock edge.}$$



(E-711) **Fig. 10.14.4 : Waveforms of ADM**

- Then the step size at the sampling instant k is given by,

- Then the step size at the sampling instant k is given by,

$$\underset{\substack{\downarrow \\ \text{Step size at } k^{th} \\ \text{clock edge}}}{\delta(k)} = \underset{\substack{\downarrow \\ \text{Step size at} \\ (k-1)^{th} \\ \text{clock edge}}}{[\delta(k-1)]} \; \underset{\substack{\downarrow \\ \text{Output at } k^{th} \\ \text{edge}}}{S_o(k)} + \underset{\substack{\downarrow \\ \text{Basic step} \\ \text{size}}}{\delta} \; \underset{\substack{\downarrow \\ \text{Output at} \\ (k-1)^{th} \\ \text{clock edge}}}{S_O(k-1)} \qquad ...(10.14.8)$$

Let us take an example :

Refer to the waveforms of Fig. 10.14.4. Let us assume k = 6, i.e. consider the 6^{th} clock edge.

$$\therefore \quad k-1 = 5.$$

$$\therefore \delta(k-1) = \delta(5) = \delta$$

$$S_o(k) = S_o(6) = +1$$

$$S_o(k-1) = S_o(5) = +1$$

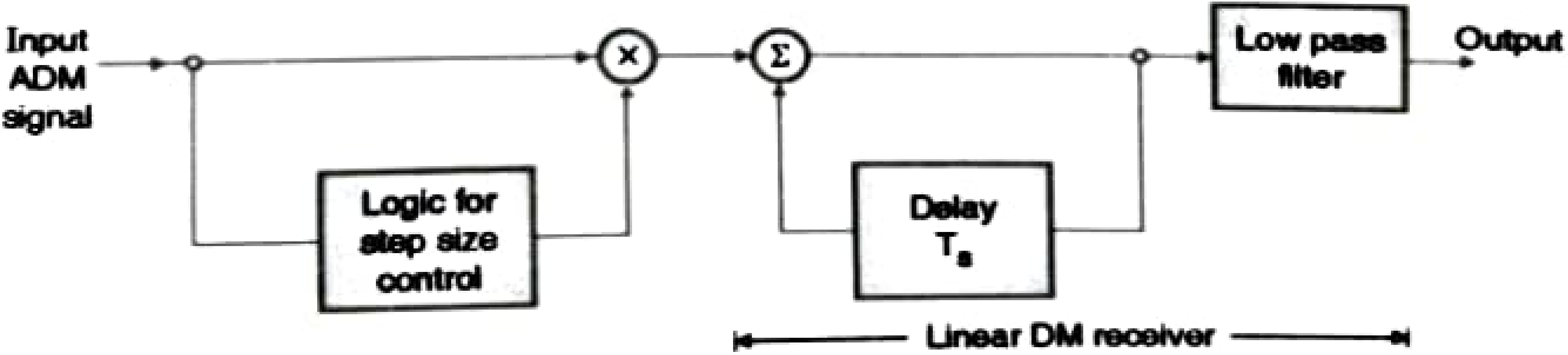
Substitute in Equation (10.14.8) to get,

$$\delta(6) = \delta + \delta = 2\delta \qquad ...(10.14.9)$$

Look at the Fig. 10.14.4, the step size at the 6^{th} clock edge is 2 δ.

As shown in Fig. 10.14.4, due to variable step size, the slope overload error is reduced. But quantization error is increased. Due to the adjustable step size, the slope overload problem is solved. Hence ADM system has a low bit rate than the PCM system. Therefore the BW required is also less than a comparable PCM system.

The block diagram of ADM receiver is shown in Fig. 10.14.5.



(D-497) **Fig. 10.14.5 : ADM receiver**

The ADM signal is first converted into a D.M. signal with the help of the step size control logic and then applied to a D.M. receiver.

At the output of low pass filter we get the original signal back.

The advantages of ADM over DM are as follows :

1. Reduction in slope overload distortion and granular noise.
2. Improvement in signal to noise ratio.
3. Wide dynamic range due to variable step size.
4. Better utilization of bandwidth as compared to delta modulation.

In addition to these advantages, the advantages of delta modulation i.e. low signaling rate and simplicity of implementation are also obtained if we use the adaptive delta modulation.

Sr. No.	Parameter	PCM	DM	ADM	DPCM
1.	Number of bits per sample	N can be 4, 8, 16, 32, 64 etc.	N = 1	N = 1	N is more than 1 but less than that for PCM
2.	Step size	Depends on the number of Q levels.	Step size is fixed	Step size is variable	Step size is fixed
	Distortions / errors	Quantization error	Slope overload and granular noise	Granular noise	Slope overload and granular noise
	Signaling rate and bandwidth	Highest	Low, if the input is slow varying	Lowest	Lower than PCM
	System complexity	Complex	Simple	Simple	Simple
	Feedback from output	No feedback	Feedback is present	Feedback is present	Feedback is present
	Noise immunity	Very good	Very good	Very good	Very good
	Use of repeaters	Possible	Possible	Possible	Possible

4.3 Line Coding :

- The digital data which may be coming from a digital computer or some other source cannot be put directly on the communication channel, because the format of this signal is not suitable for its direct transmission.
- This data is first converted into a suitable format or line code and then transmitted over a communication channel.
- The various formats used are called as line codes.
- The formats or line codes are :
  1. Non-return to zero (NRZ) and return to zero (RZ) unipolar format.
  2. NRZ and RZ polar format.
  3. Non-return to zero bipolar format.
  4. Manchester format.
  5. Polar quaternary NRZ format.

4.4 Definition of Line Coding :

- It is the process of converting binary data, a sequence of bits to a digital signal.
- The data, text, numbers, graphical images, audio and video which are stored in computer memory are in the form of sequences of bits.
- Line coding converts these sequences into digital signals as shown in Fig. 4.4.1.

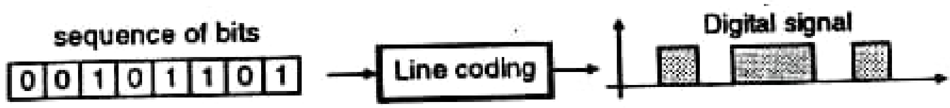


Fig. 4.4.1

4.4.1 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

4.4.2 Signal Level and Data Level :

- A digital signal has a limited number of values. However all of them are not used to represent data. Only some of them only can be used to represent the data.
- The remaining values are used for some other purpose.

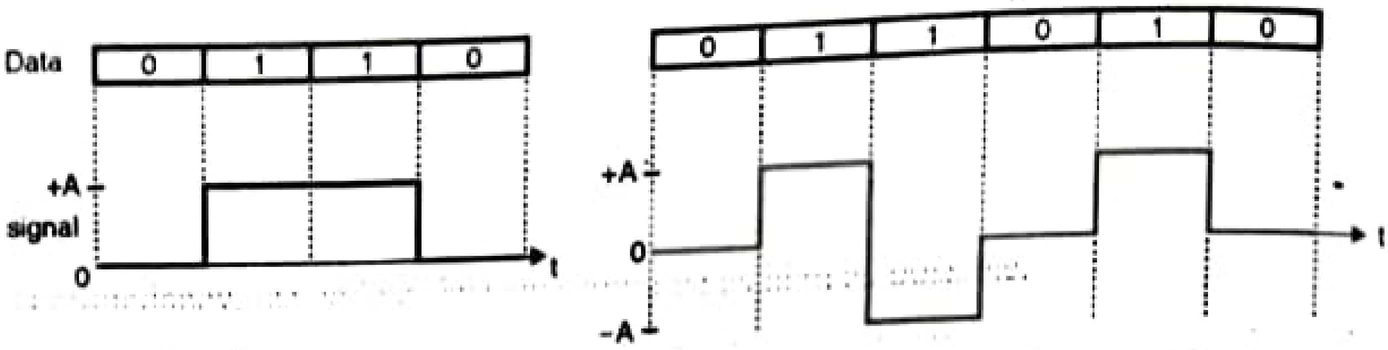
Signal levels :

The number of values allowed in a particular signal is defined as the number of signal levels.

Data levels :

The number of values to represent data is called as the number of data levels.

- Fig. 4.4.2 gives you a clear idea about signal levels and data levels.
- Fig. 4.4.2(a) has two signal levels (0 and A) and two data levels (0 and 1) whereas Fig. 4.4.2(b) has three signal levels and two data levels.



(a) Two signal levels, two data levels .... (b) Two data levels, three signal levels

Fig. 4.4.2

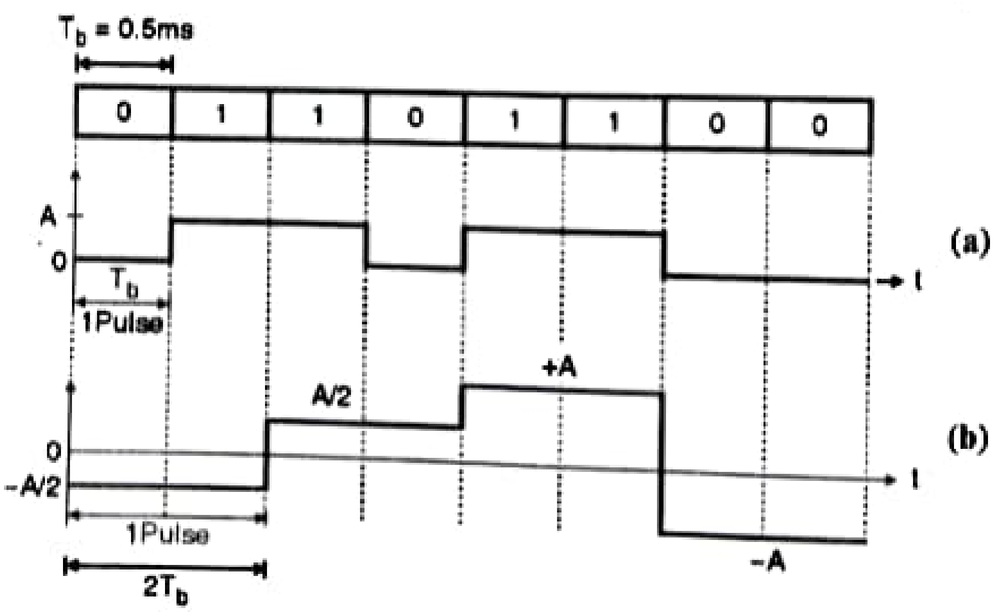
- Fig. 4.4.2(b) has three signal levels (+ A, 0, – A) and two data levels (0 and 1).

4.4.3 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.

$$\text{Bit rate} = \text{Pulse rate} \times \log_2 L \text{ s} \qquad \ldots(4.4.1)$$

Ex. 4.4.1 : For the signals shown in Fig. P. 4.4.1(a) and (b) calculate the bit rate and pulse rate.



Soln. :

- The data has two levels and the bit duration $T_b = 0.5$ ms.
- Refer Fig. P. 4.4.1(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. 4.4.1(b). Now one pulse corresponds to a duration of 2 T_b.

$$\text{So pulse duration} = 2\ T_b = 1 \text{ ms.}$$

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

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

Here L = 4 levels

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

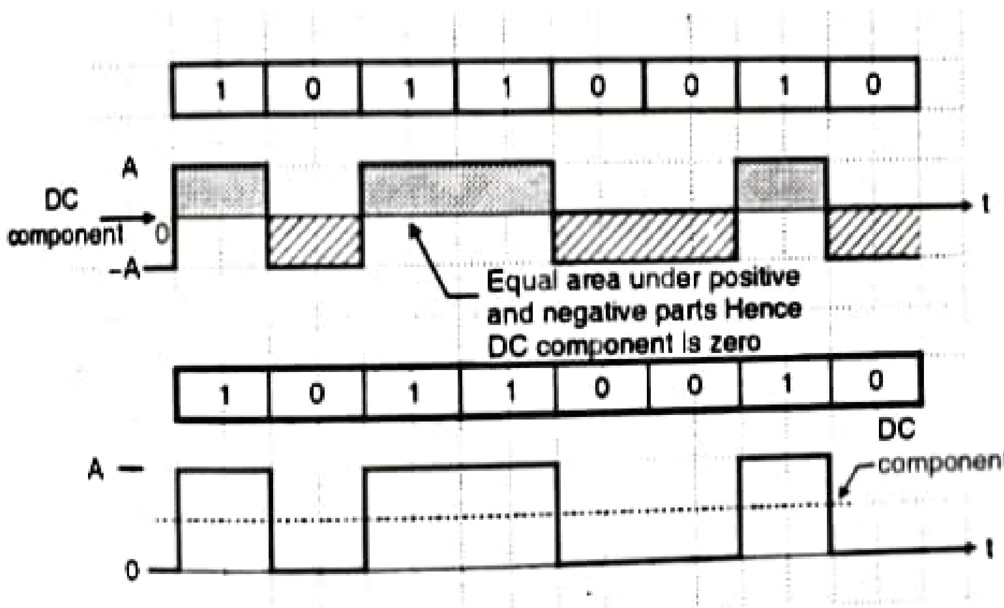
4.4 DC Component :

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. 4.4.3(a)).

But the waveform of Fig. 4.4.3(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.



(a) DC component equal to zero

(b) Signal with a positive dc component

Self Synchronization :

In order to receive the signal correctly, the receiver's bit intervals must correspond exactly to the sender's bit intervals.

The clock frequency of the transmitter and receiver should be the same.

If the clock frequency at the receiver is slower or faster then the bit intervals are not matched and the received signal is different that the transmitted one.

Fig. 4.4.4 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.

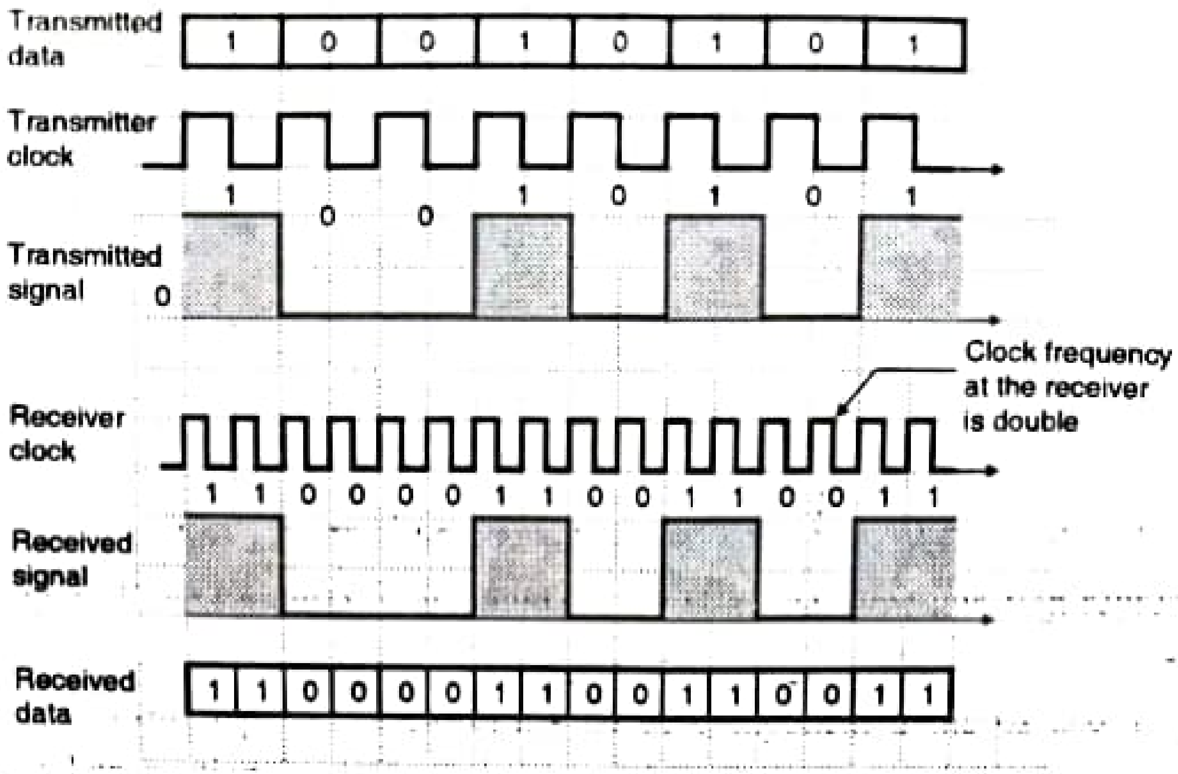


Fig. 4.4.4 : Concept of synchronization

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.

.5 Classification of Line Codes :

Fig. 4.5.1 shows the classification of line codes.

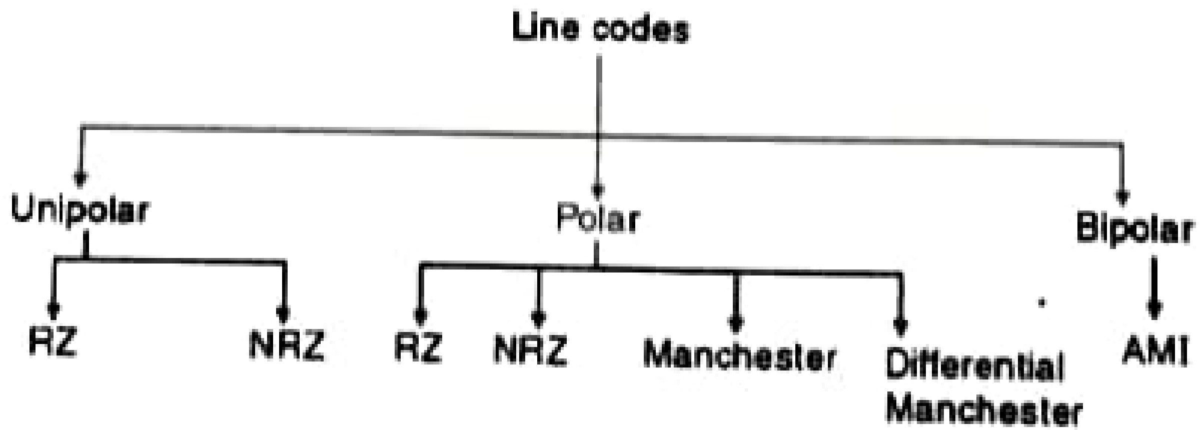


Fig. 4.5.1 : Classification of line codes

The line codes are basically divided into three categories

(1) Unipolar codes (2) Polar codes (3) Bipolar codes

Unipolar codes :

- Unipolar codes use only one voltage level other than zero.
- So the encoded signal will have either + A volts value or 0.
- These codes are very simple and primitive and are almost obsolute now a days.

Polar codes :

- Polar coding using two voltage levels other than zero such as + A / 2 and – A / 2 volts.
- This will bring the dc level for some codes to zero which is a desired characteristics.

Bipolar codes :

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

1 Properties of Line Codes :

owing are some of the important properties of line codes :

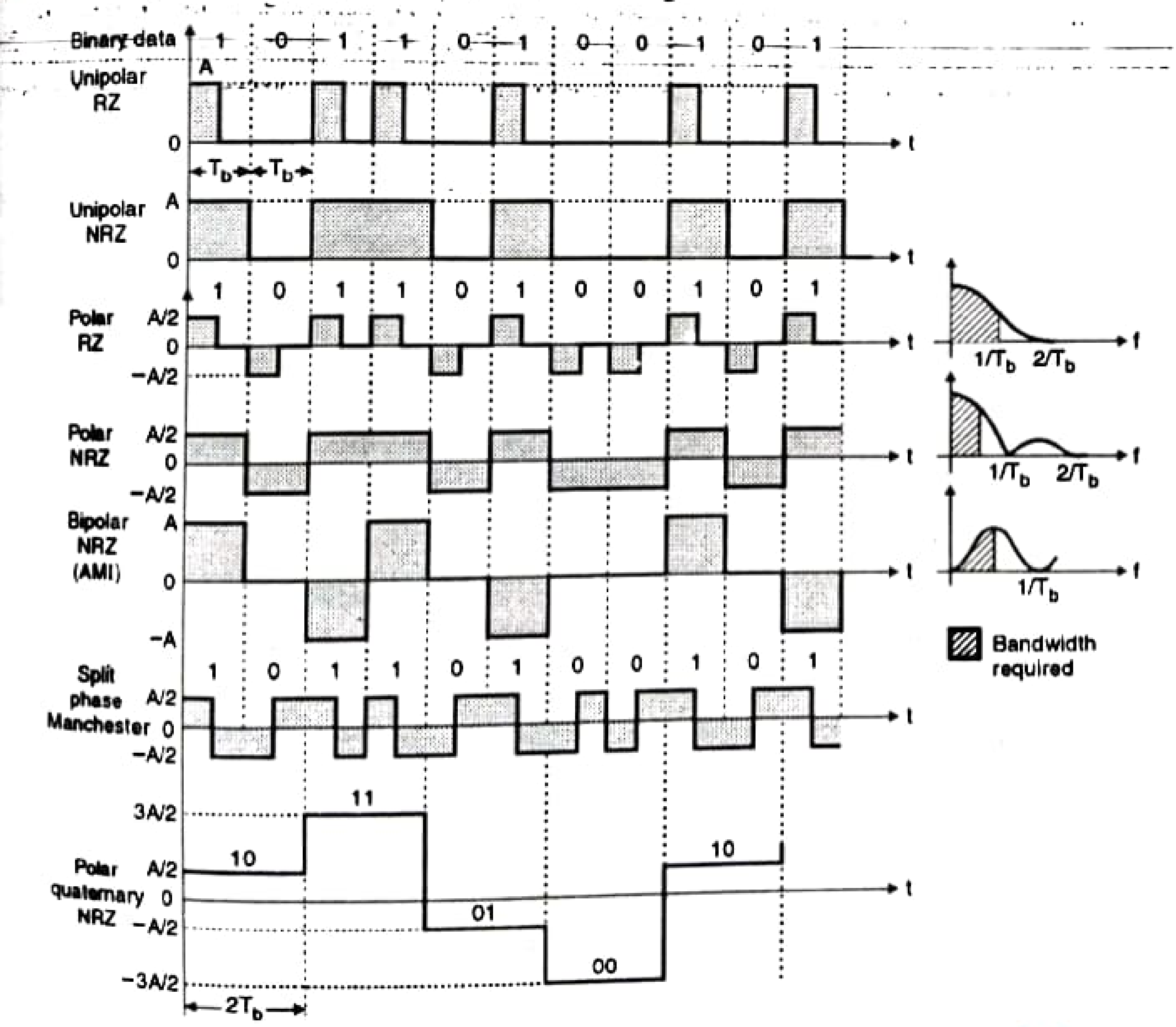
All the cable systems and other communication systems, do not allow transmission of a dc s Therefore the **line signal must have a zero average (dc) value**. NRZ bipolar formats u satisfy this requirement. **For this reason, long strings of element sequences having polarity should not be transmitted.**

As the code adds redundancy, the code efficiency should be as high as possible.

To ensure synchronization at the receiver, the line signal should undergo a **sufficient numb zero crossings that means the transmitted signal should** always **undergo transitions.**

The crosstalk between channels should be minimized. To do so the amount of energy i signal at low frequencies should be small.

Some of the important line codes are as shown in Fig. 4.5.2.



Unipolar Line Codes :

.1 Unipolar RZ Format :

The return to zero (RZ) unipolar format is as shown in Fig. 4.6.1.

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).

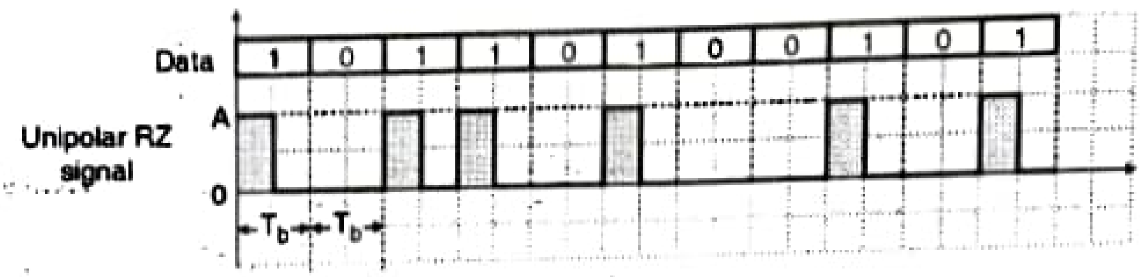


Fig. 4.6.1 : Unipolar RZ format

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

6.2 Unipolar NRZ Format :

A non-return to zero (NRZ) format is as shown in Fig. 4.6.2.

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.

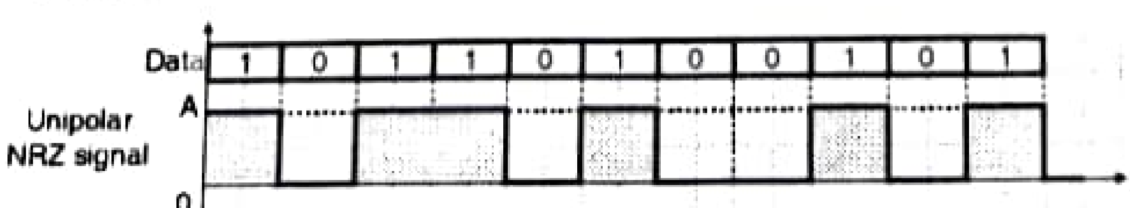


Fig. 4.6.2 : Unipolar NRZ format

.7 Polar Line Codes :

.7.1 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.

The polar RZ format is as shown in Fig. 4.7.1. 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.

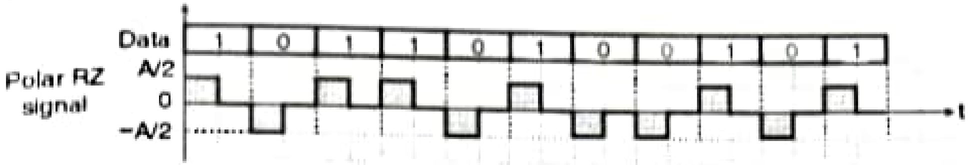


Fig. 4.7.1 : Polar RZ format

2 Polar NRZ Format :

In the polar NRZ format, as shown in Fig. 4.7.2 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.

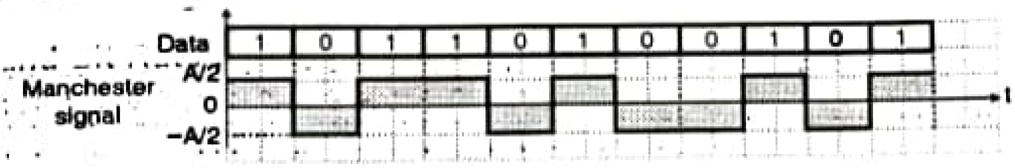


Fig. 4.7.2 : Polar NRZ format

3 Split Phase Manchester Format :

The split phase Manchester format is as shown in Fig. 4.7.3.

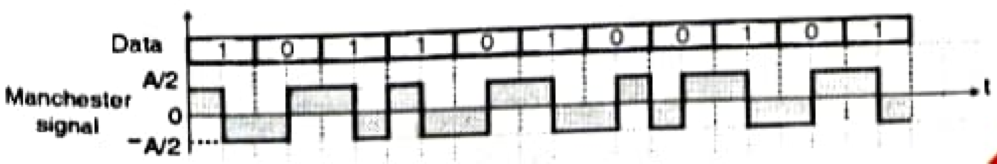
In this format, symbol "1" is represented by transmitting a positive pulse of "+ A/2" amplitude for one half of the symbol duration, followed by a negative pulse of amplitude "– A/2" for remaining half of the symbol duration.

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.



4.7.4 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. 4.7.4.

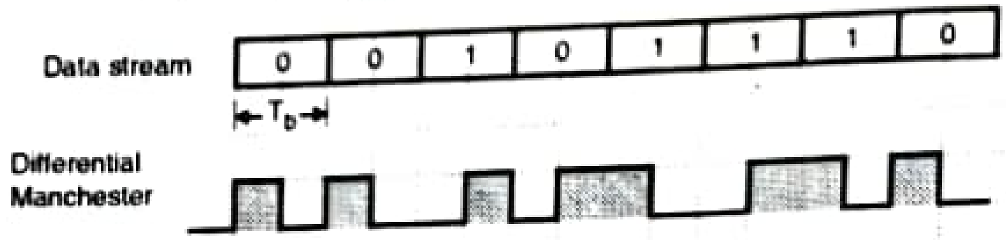


Fig. 4.7.4 : Differential manchester coding

4.8 Bipolar Line Codes :

4.8.1 Bipolar NRZ Format (AMI) :

- The bipolar NRZ format is as shown in Fig. 4.8.1. 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.

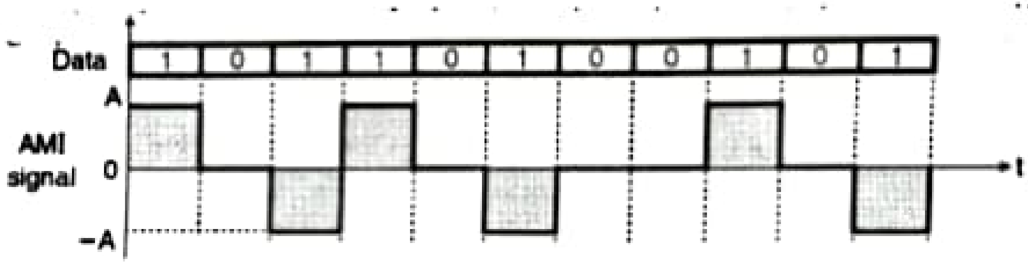


Fig. 4.8.1 : Bipolar NRZ format (AMI)

- 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.

In a communication system when the data is being transmitted in the form of pulses (bits), the output produced at the receiver due to the other bits or symbols interferes with the output produced by the desired bit.

This is called as intersymbol interference (ISI). The intersymbol interference will introduce errors in the detected signal at the receiver.

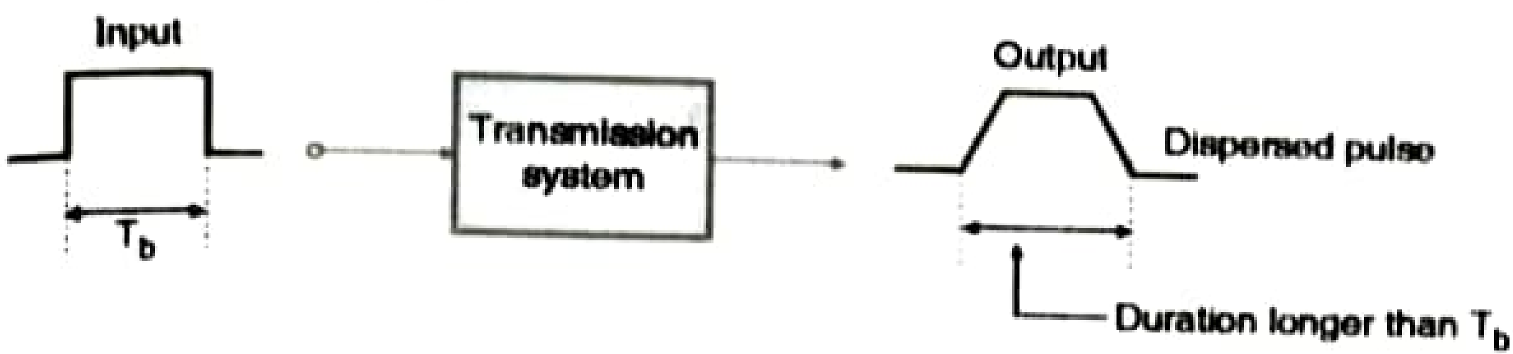
The ISI results because the overall frequency response of the system is never perfect and pulse spreading is bound to take place.

When a short pulse of duration T_b seconds is transmitted through a bandlimited transmission system, then various frequency components present in the input pulse are **differentially attenuated** and more importantly **differentially delayed** by the system.

Due to this the pulse appearing at the output of the system will be "dispersed" over an interval which is longer than "T_b" seconds as shown in Fig. 10.17.1.

Due to this dispersion, the adjacent symbols will interfere with each in time domain other when transmitted over the communication channel. This will result in the intersymbol interference (ISI).

The transmitted pulse of duration T_b seconds and the dispersed pulse of duration more than T_b seconds are shown in Fig. 10.17.1.



(E-298) **Fig. 10.17.1 : Cause of ISI**

ct of ISI :

If the ISI and noise are absent totally then, the transmitted bit can be decoded correctly at the receiver. However errors will be introduced due to presence of ISI at the receiver output.

Due to this the receiver can make an error in deciding whether it has received a logic 1 or a logic 0.

Another effect of ISI is the cross talk which may take place due to overlapping of the adjacent pulses due to spreading.

It is necessary to use the special filters called equalizers in order to reduce ISI and its effect.

ses of ISI :

four important causes for ISI are as follows :

Timing inaccuracies 2. Insufficient bandwidth

Amplitude distortion 4. Phase distortion.

Timing inaccuracies :

The ISI will take place if the transmitter rate of transmission is not same as the ringing frequency of the given channel.

Insufficient bandwidth :

If the transmission rate is less than the channel bandwidth then there is a very small possibility of timing error.

But if the channel bandwidth is reduced, then the possibility of timing error will increase and the possibility of ISI also will increase.

Amplitude distortion :

Generally filters are used in the communication systems in order to bandlimit the signals and reduce the noise.

But the frequency response of the communication channels can not be accurately predicted.

When the frequency characteristics of a communication channel differs from the expected one, the pulse distortion is likely to take place.

The pulse distortion results in reduction of the peaks of the pulses i.e. amplitude distortion. In order to compensate for this, we have to use the amplitude equalization.

Phase distortion :

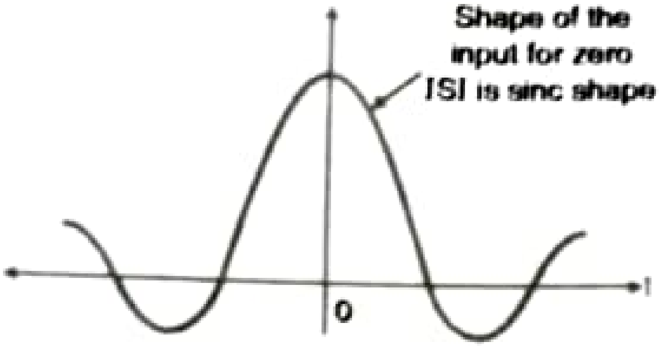
If various frequency components in the input pulse undergo different amounts of time delay while travelling through the channel, then the phase distortion is bound to take place.

This will cause the ISI. Special delay equalizers are required to be used to reduce the phase distortion and the associated ISI.

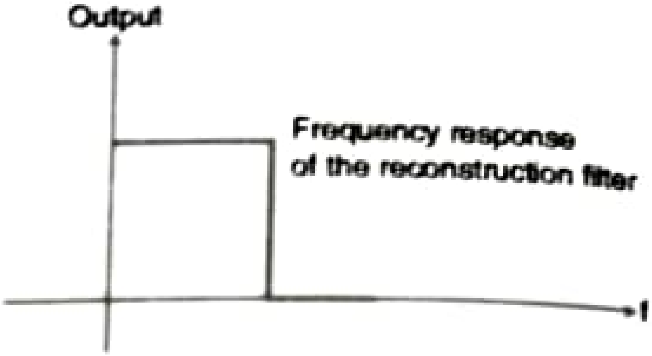
edy to Reduce the ISI :

It has been proved that the function which produces a zero intersymbol interference is a "sinc function". Thus instead of a rectangular pulse if we transmit a sinc pulse then the ISI can be reduced to zero.

Using the since pulse for transmission is known as "Nyquist Pulse Shaping". The sinc pulse transmitted to have a zero ISI is shown in Fig. 10.17.2(a).



(a) Ideal pulse shape for zero ISI

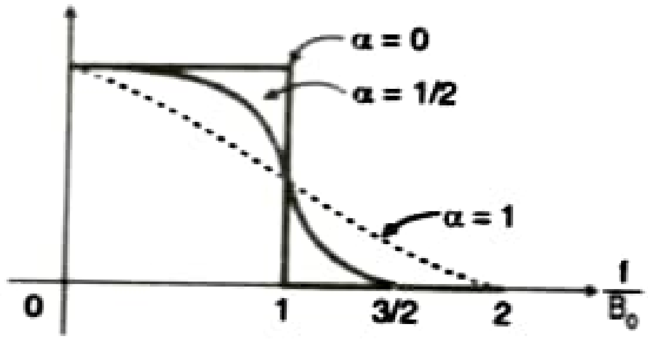


(b) Frequency response of the filter

(E-299) **Fig. 10.17.2**

We know that Fourier transform of a sinc pulse is a rectangular function. Therefore to preserve all the frequency components, the frequency response of the filter must be exactly flat in the pass band and zero in the attenuation band as shown in Fig. 10.17.2(b).

This type of filter is practically not available. Therefore practically the frequency response of the filter is modified as shown in Fig. 10.17.3 with different roll off factors "α" to obtain the practically achievable filter response curves.



(E-300) **Fig. 10.17.3 : Practical filter characteristics**

18 Eye Pattern :

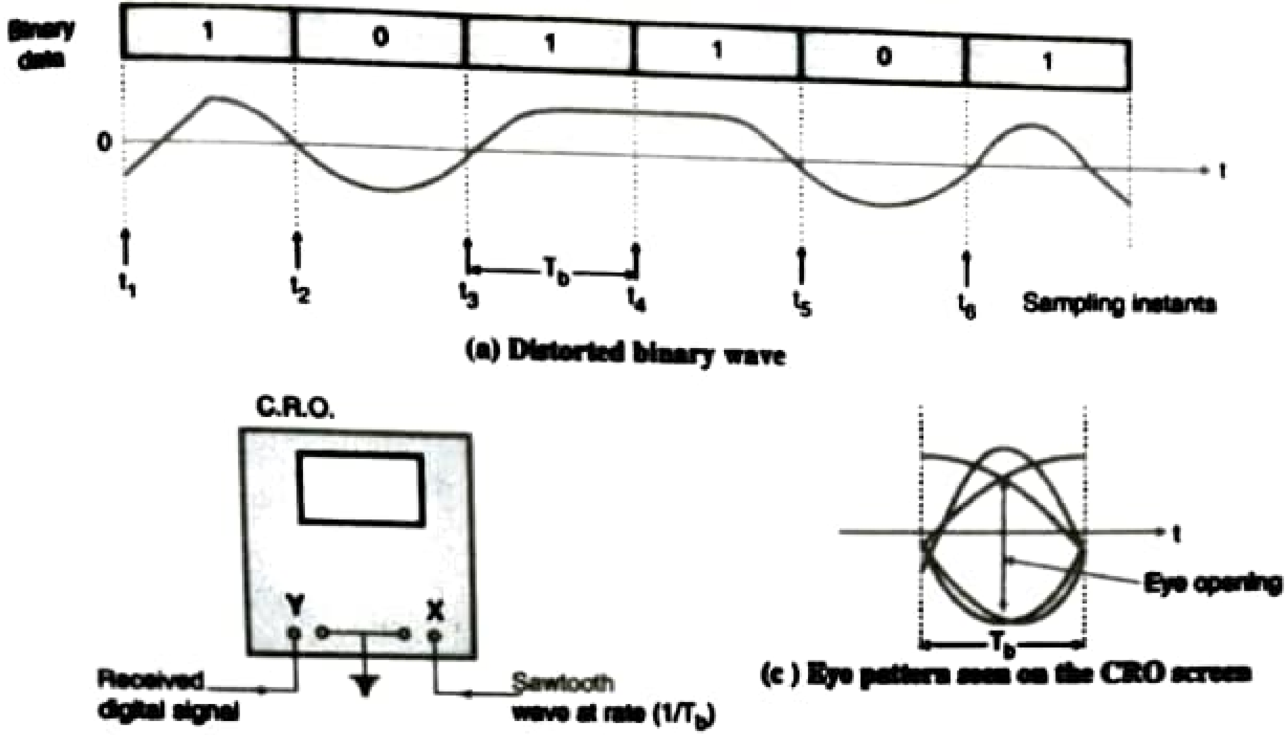
Eye pattern is a pattern displayed on the screen of a cathode ray oscilloscope (C.R.O.). The shape of this pattern is very similar to the shape of human eye. Therefore it is called as **eye pattern**.

Eye pattern is used for studying the intersymbol interference (ISI) and its effects on various communication systems.

The eye pattern is obtained on the C.R.O. by applying the received signal to vertical deflection plates (Y-plates) of the C.R.O. and a sawtooth wave at the transmission symbol rate i.e. $(1 / T_b)$ to the horizontal deflection plates (X-plates) as shown in Fig. 10.18.1(c).

The received digital signal and the corresponding oscilloscope display are as shown in Figs. 10.18.1(a) and (c) respectively.

The resulting oscilloscope display shown in Fig. 10.18.1(c) is called as the "eye pattern". This is due to its resemblance to the human eye.



(a) Distorted binary wave

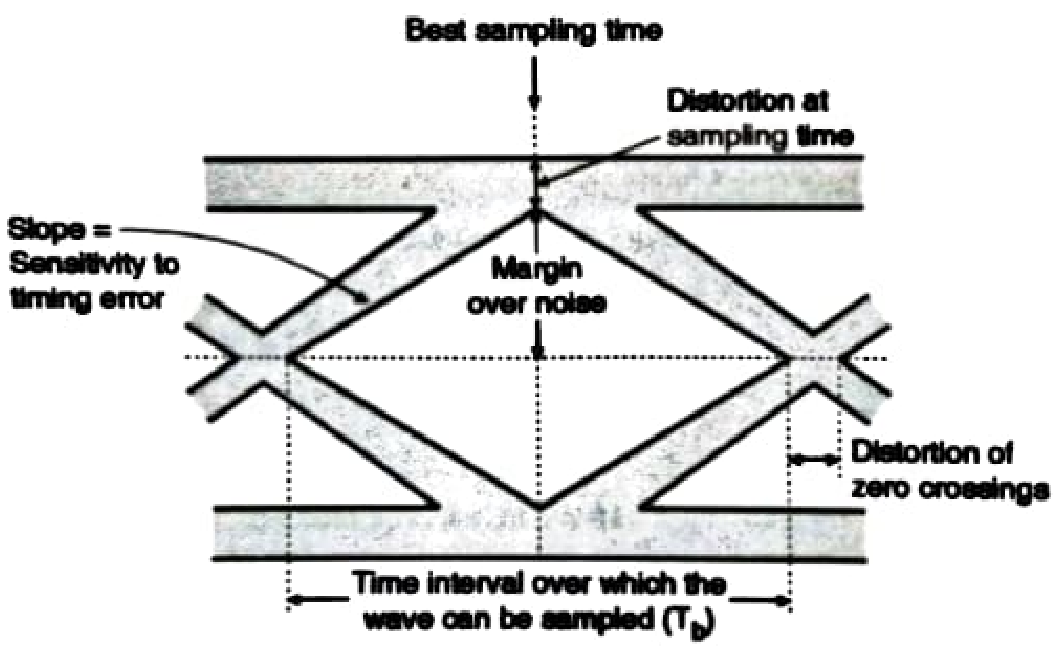
(b) Oscilloscope connections

(c) Eye pattern seen on the CRO screen

(E-353) **Fig. 10.18.1 : Obtaining eye pattern**

The region inside the eye pattern is called as the eye opening.

The eye pattern provides very important information about the performance of the system. The information obtainable is as follows (See Fig. 10.18.2).



(E-354) **Fig. 10.18.2 : Interpretation of eye pattern**

The width of the eye opening defines the time interval over which the received wave can be sampled, without an error due to ISI. The best instant of sampling is when the eye opening is maximum.

The sensitivity of the system to the timing error is determined by observing the rate at which the eye is closing as the sampling rate is varied.

The height of eye opening at a specified sampling time defines the margin over noise.

When the effect of ISI is severe, the eye is completely closed and it is impossible to avoid errors due to the combined effect of ISI and noise in the system.