DIGITAL COMMUNCICATION

Subject Code : EC0618

Prepared by : Prof. Zalak Patel
Assistant Professor
Electronics & communication
Engineering Department

UNIT-I

BASEBAND MODULATION

- **Pulse Modulation**
- **Line coding**

ANALOG-TO-DIGITAL CONVERSION

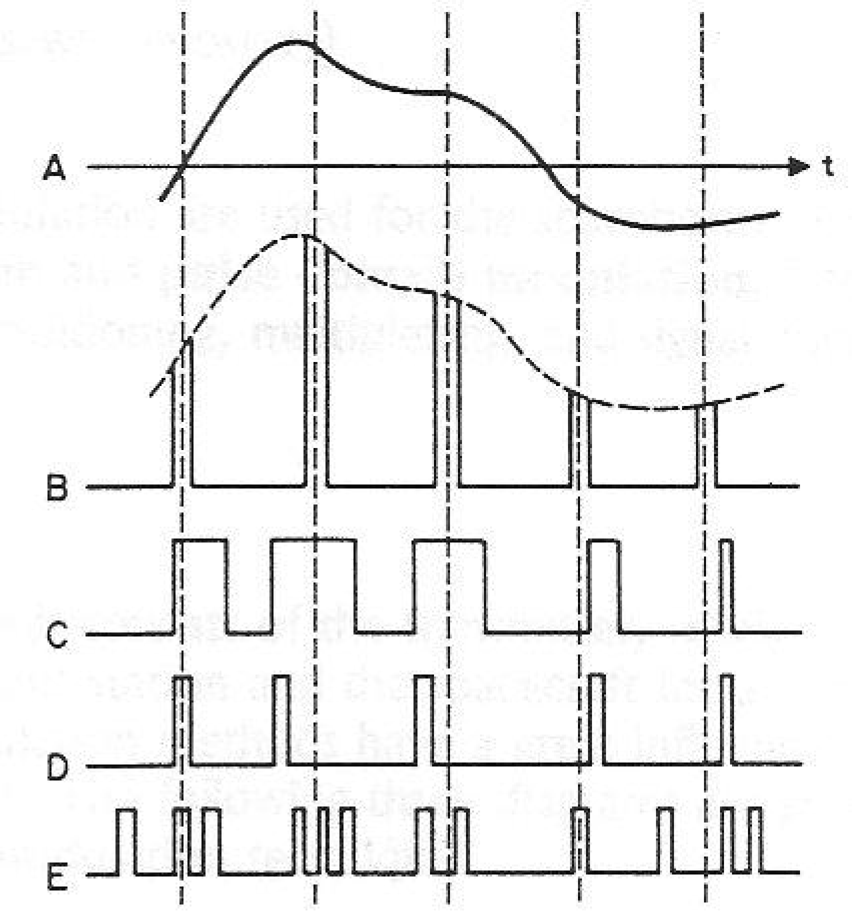
- A digital signal is superior to an analog signal because it is **more robust to noise and can easily be recovered, corrected and amplified**. For this reason, the tendency today is to change an analog signal to digital data.

- To convert analog signal into digital signal pulse modulation is performed.
  1. Pulse Ampltidue Modualtion (PAM)
  2. Pulse Code Modulation (PCM)
  3. Pulse Width Modulation (PWM)
  4. Pulse Position Modulation (PPM)

- **PCM is the most widely used method for conveting an analog signal into a digital signal. (A/D conversion)**

What is Pulse Modulation Technique?

- <u>**The continuous time (analog) signals are sampled & sampled values are used to change the certain parameters of periodic pulse train.**</u>



[Analog signal]

[Pulse Amplitude Modulation]

[Pulse Width Modulation]

[Pulse Position Modulation]

[Pulse Code Modulation]

PAM

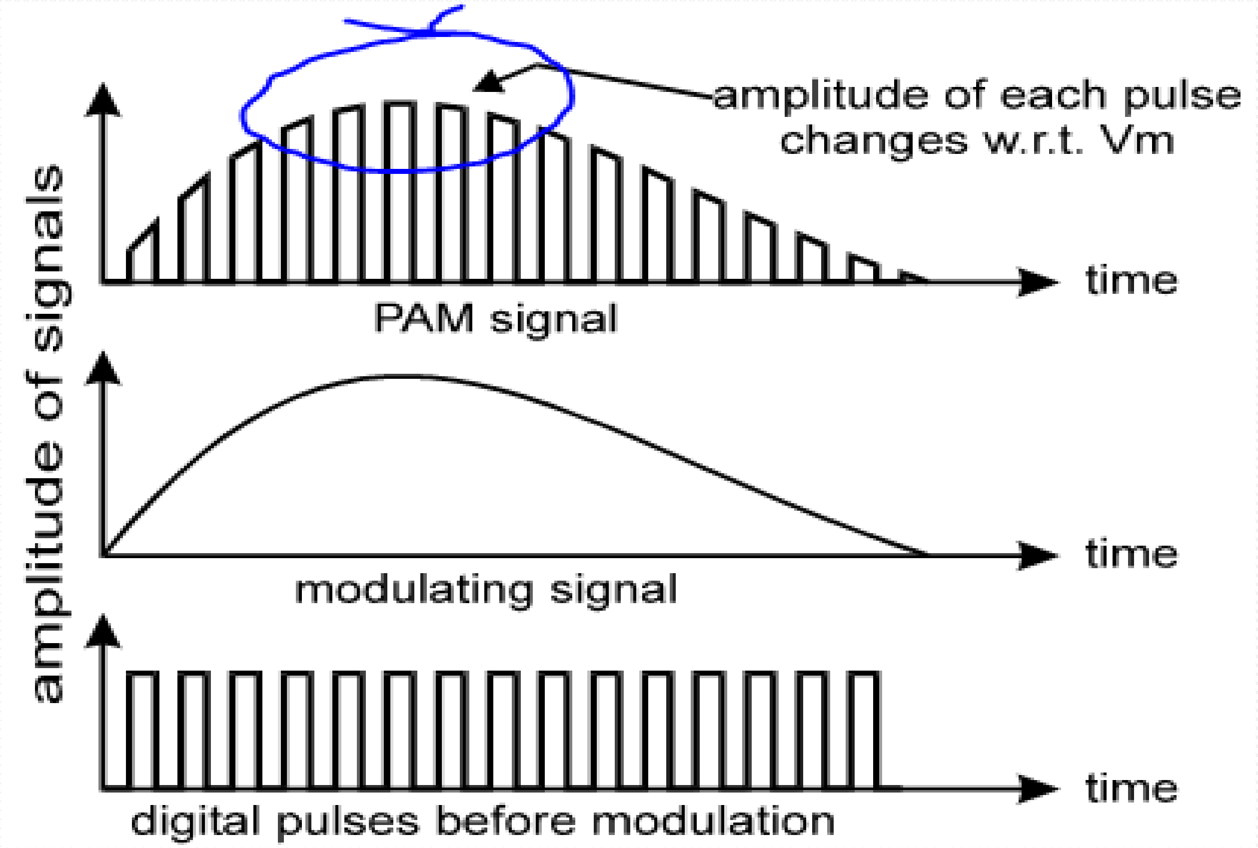
- In pulse-amplitude modulation (PAM) the amplitude of a pulse train (digital carrier) is varied in proportion to the modulating (analog message) signal.

- The pulse train consists of uniform width & uniform amplitude pulses spaced at equal time interval.

- Pulse Amplitude Modulation is achieved by multiplying the carrier (digital pulse train) with the modulating signal.

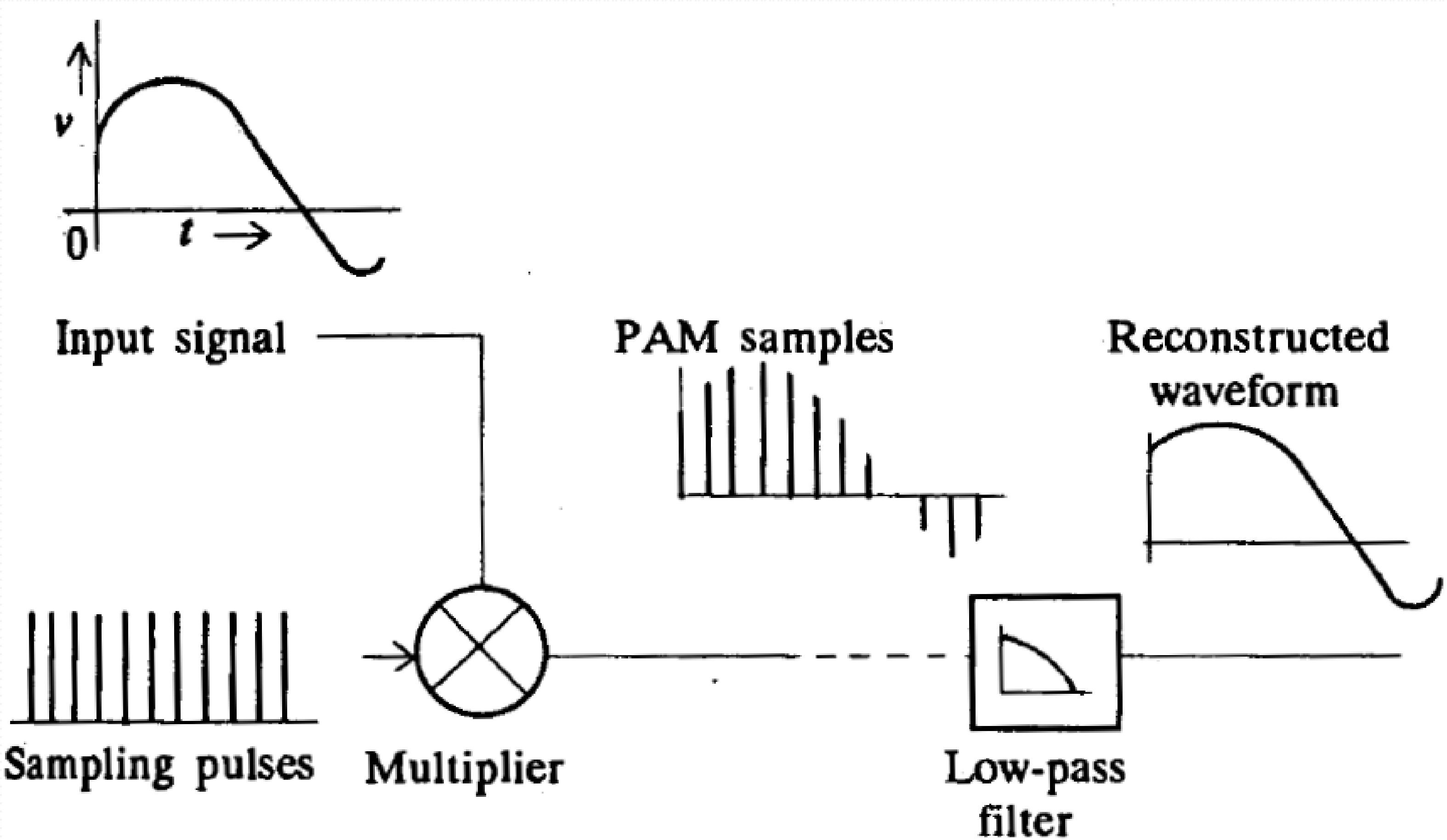
- Periodic time of the pulse train T_s is known as the sampling period. Sampling frequency $f_s = 1/ T_s$

PAM



amplitude of each pulse changes w.r.t. Vm
amplitude of signals
time
PAM signal
time
modulating signal
time
digital pulses before modulation

PAM

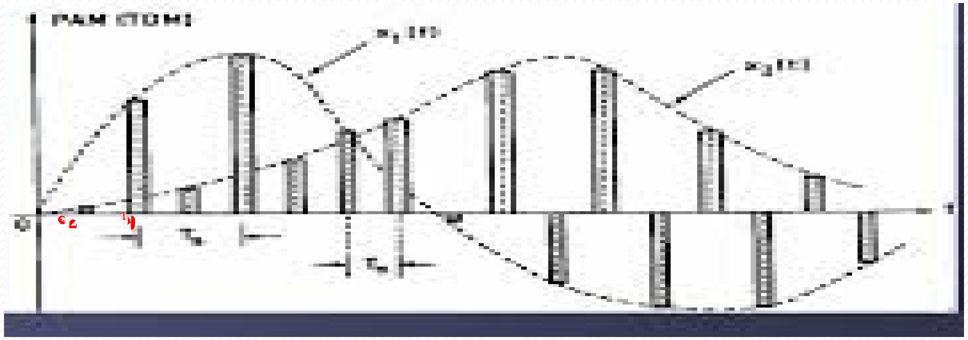


v
0
t
Input signal
PAM samples
Reconstructed waveform
Sampling pulses
Multiplier
Low-pass filter

Advantages of Pulse Modulation

- **It permits the simultaneous transmission of several signals on time-sharing basis. (Time Division Multiplexing).**

- Since pulse modulated signals occupies only part of channel time, several pulse modulated signals can be transmitted on same channel by interweaving.
- **Fig shows multiplexing of two PAM signals on the same channnel by reducing the pulse widths.**

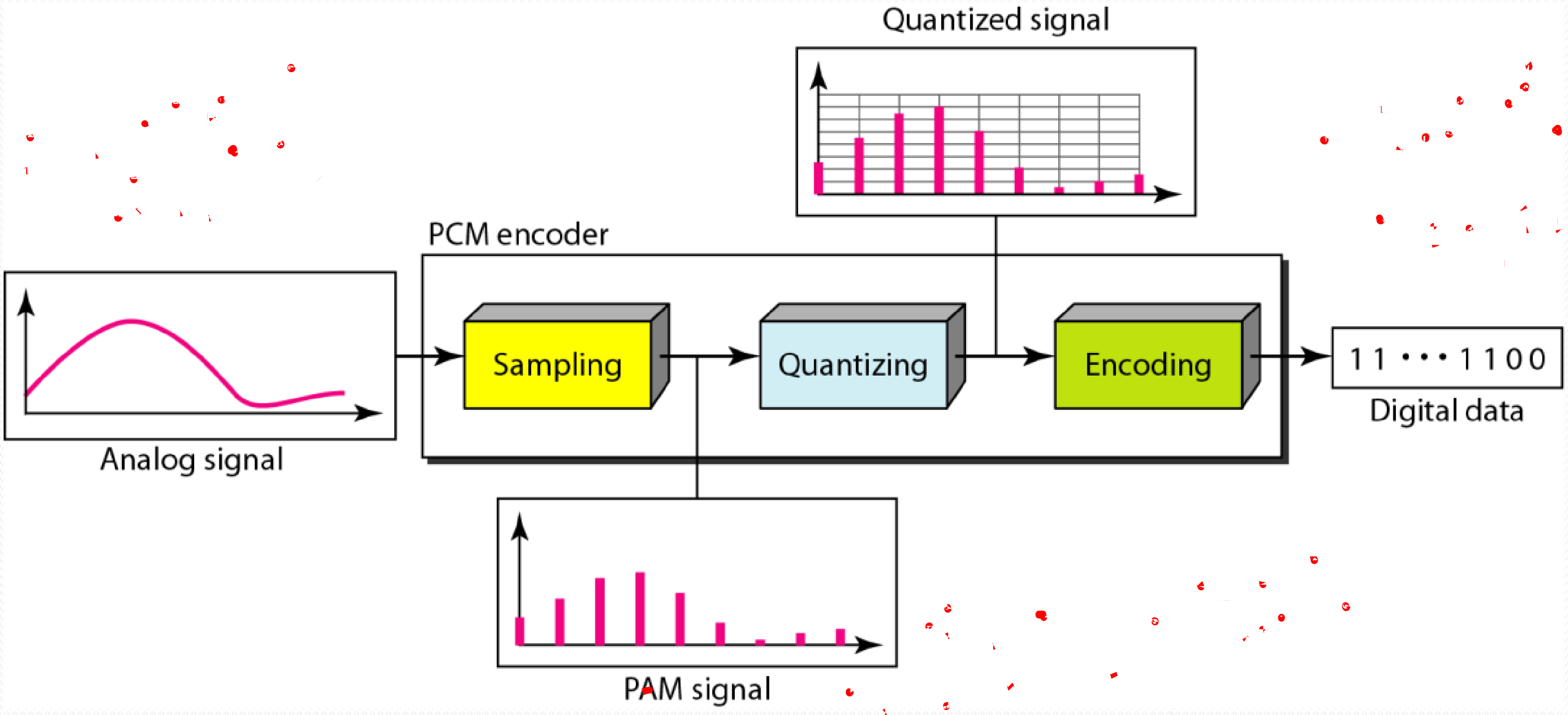


Pulse Code Modulation

- PCM consists of three steps to digitize an analog signal:
  1. **Sampling**
  2. **Quantization**
  3. **Binary encoding**

- Before we sample, we have to filter the signal to limit the maximum frequency of the signal as it affects the sampling rate.

PCM Encoder Block Diagram



PCM- Sampling Process (1)

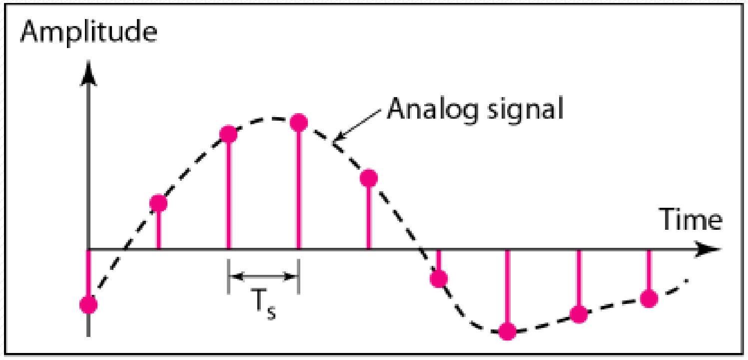
- Analog signal is sampled every T_S secs.
- T_s is referred to as the sampling interval.
- $f_s = 1/T_s$ is called the sampling rate or sampling frequency.

- There are 3 sampling methods:
  - Ideal - an impulse at each sampling instant
  - Natural - a pulse of short width with varying amplitude
  - Flattop - a pulse of short width but with single amplitude value

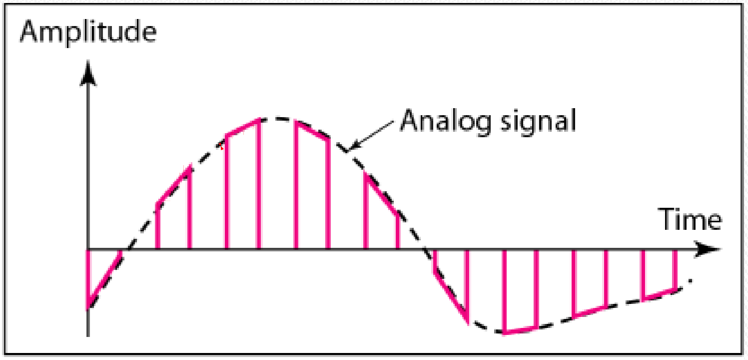
- The process is referred to as pulse amplitude modulation PAM and the outcome is a signal with analog (non integer) values.

PCM-Sampling Methods

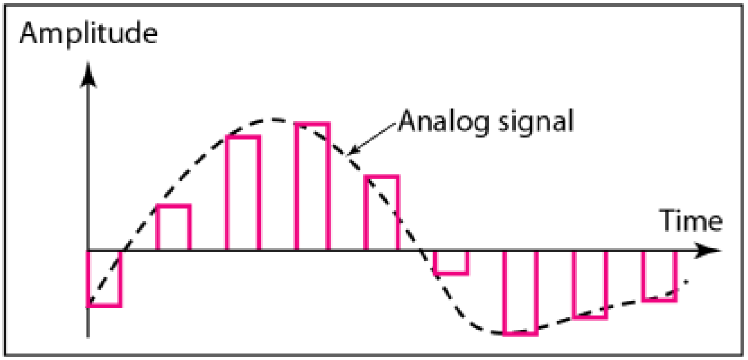
Three different sampling methods for PCM



a. Ideal sampling



b. Natural sampling



c. Flat-top sampling

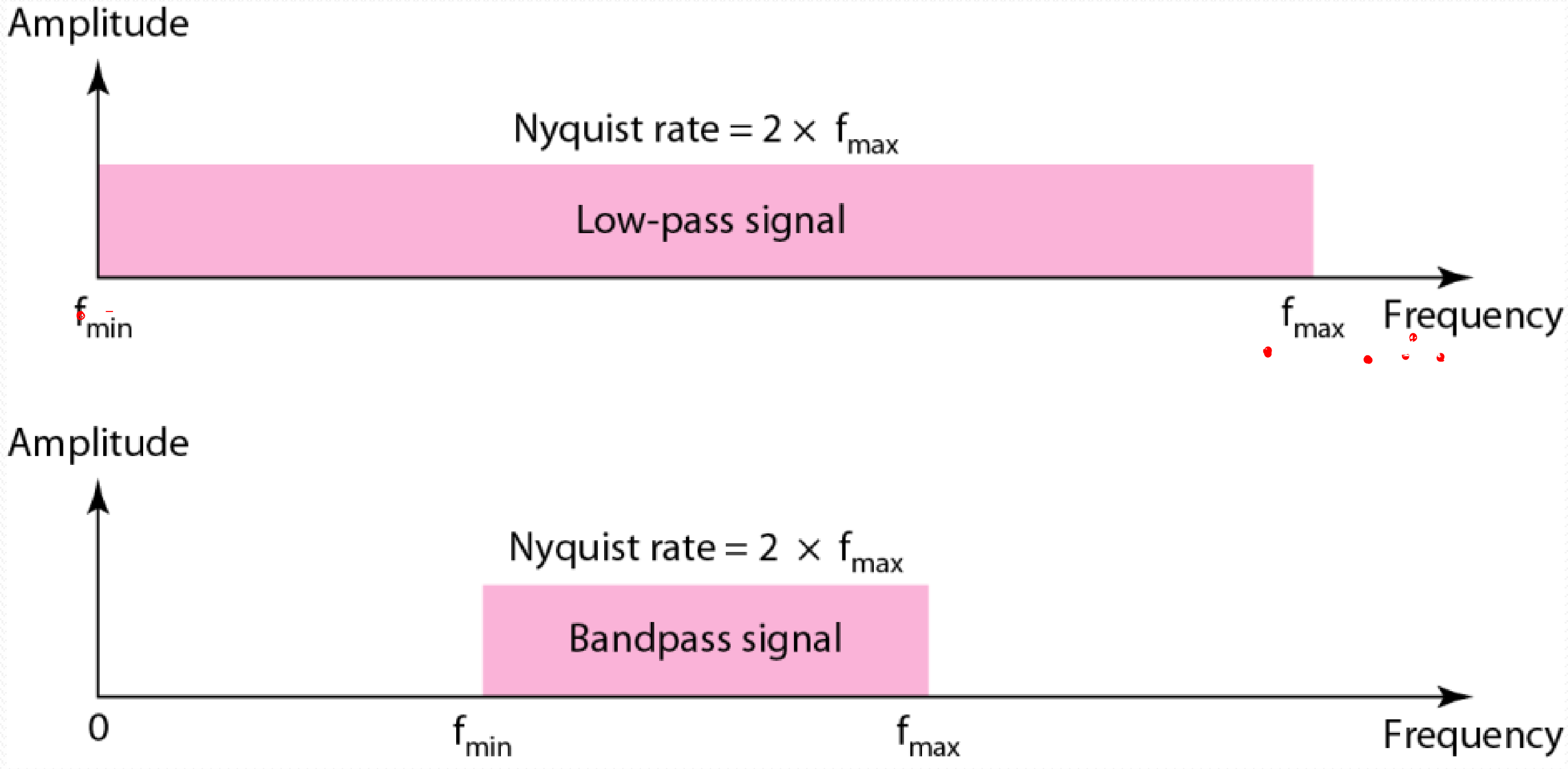
PCM

- **According to the Nyquist theorem, the sampling rate must be at least 2 times the highest frequency contained in the signal.**

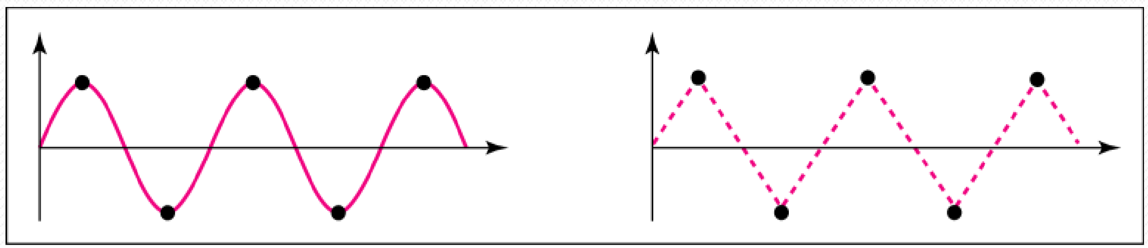
- The sampling rate must be sufficiently large so that analog signal can be reconstructed from the samples with high accuracy.
- The sampling theorem to detemine the proper sampling rate for a given signal, has deep significance in signal processing and communication theory.

PCM

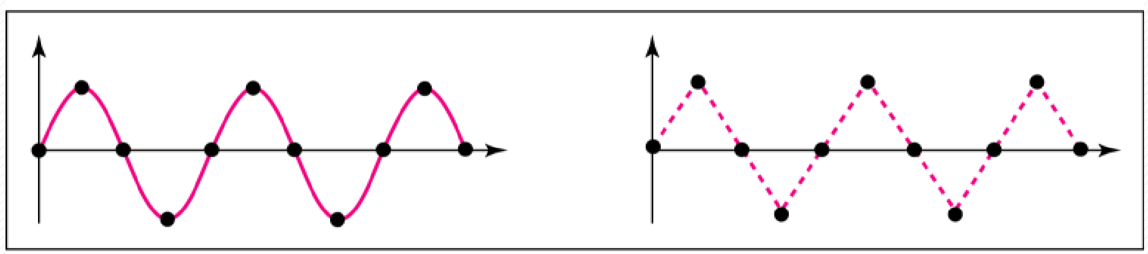
Nyquist sampling rate for low-pass and bandpass signals



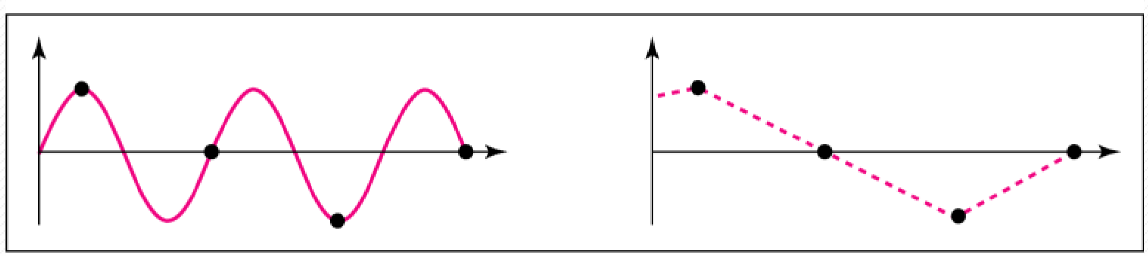
PCM-Effect of Sampling Rate



a. Nyquist rate sampling: $f_s = 2f$



b. Oversampling: $f_s = 4f$



c. Undersampling: $f_s = f$

PCM

Example

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

PCM- Quantization Process (2)

- Sampling results in a series of pulses of varying amplitude values ranging between two limits: a min and a max.

- The amplitude values are infinite between the two limits.

- **Quantization is the process to map the *infinite* amplitude values onto a finite set of known values.**

- This is achieved by dividing the distance between min and max into L zones, each of height Δ.

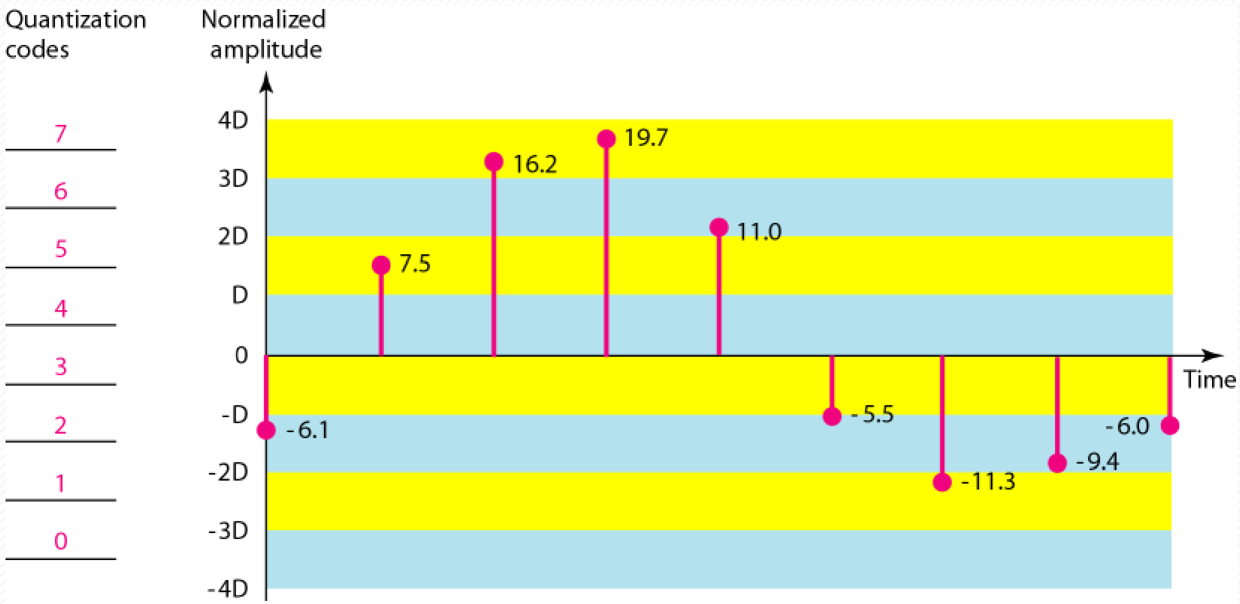
$$\Delta = (max - min)/L$$

PCM

- The midpoint of each zone is assigned a value from 0 to L-1 (resulting in L values)
- **Each sample falling in a zone is then approximated to the value of the midpoint.**

- Assume we have a voltage signal with amplitudes V_{min}=-20V and V_{max}=+20V
- We want to use L=8 quantization levels
- Zone width Δ = (20 - -20)/8 = 5
- The 8 zones are: -20 to -15, -15 to -10, -10 to -5, -5 to 0, 0 to +5, +5 to +10, +10 to +15, +15 to +20
- The midpoints are: -17.5, -12.5, -7.5, -2.5, 2.5, 7.5, 12.5, 17.5

PCM



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

PCM- Encoding (3)

- Each zone is then assigned a binary code.
- The number of bits required to encode the zones, or the number of bits per sample is obtained as follows:

$$n_b = \log_2 L$$

- Given our example, $n_b = 3$
- The 8 zone (or level) codes are therefore: 000, 001, 010, 011, 100, 101, 110, and 111
- Assigning codes to zones:
  - 000 will refer to zone -20 to -15
  - 001 to zone -15 to -10, etc.

Quantization Error

- When a signal is quantized-the coded signal is an approximation of the actual amplitude value.
- The difference between actual and coded value (midpoint) is referred to as the quantization error.
- The more zones, the smaller Δ which results in smaller errors.
- BUT, the more zones the more bits required to encode the samples - higher bit rate.

Find sampling frequency

- **Signal**
- Telephone speech
- FM audio
- CD quality music
- TV signal
- HDTV signal

- **Signal BW**
- 3.1 kHz
- 15 kHz
- 20 kHz
- 5 MHz
- 37 MHz

- **Sampling rate**
- 8 kHz
- 32 kHz
- 44.1 kHz
- 13 MHz
- 74.25 MHz

Bit rate and bandwidth requirements of PCM

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

$$\textbf{Bit rate} = n_b \times f_s$$

$$\textbf{bps} = n_b \times 2 \times B_{analog}$$

- “A noiseless channel having 1 Hz of bandwidth can transmitt 2 independant pieces of information per second errorfree.”

(1 Hz= 2 bps)

- Theoretical minimum bandwidth required to transmit PCM signal is

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

- A digitized signal will always need more bandwidth than the original analog signal. Price we pay for robustness and other features of digital transmission.

Examples of Channels

Channel	Bandwidth	Bit Rates
Telephone voice channel	32 kHz	64 kbps
Copper pair	1 MHz	1-6 Mbps
Coaxial cable	500 MHz (6 MHz channels)	30 Mbps/ channel
5 GHz radio (IEEE 802.11)	300 MHz (11 channels)	54 Mbps / channel
Optical fiber	Many TeraHertz	40 Gbps / wavelength

Problem

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

Solution

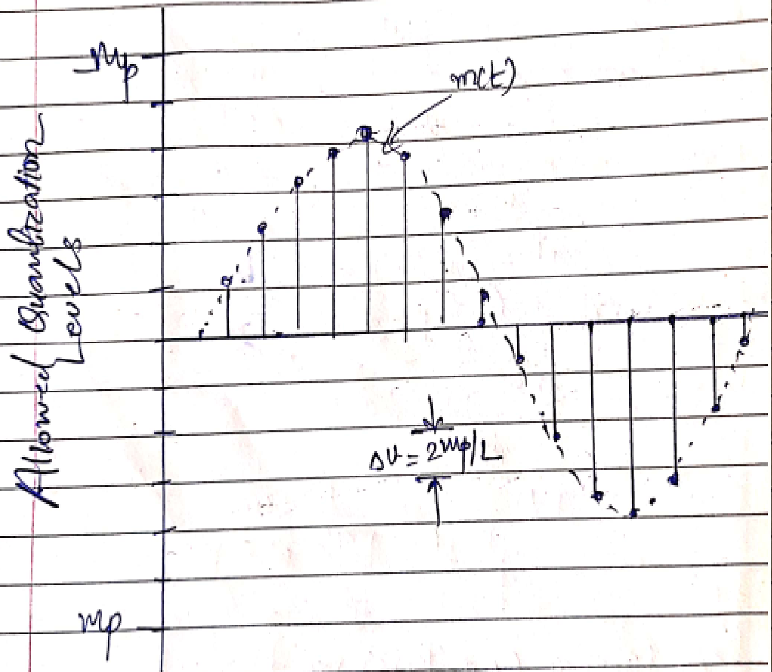
The human voice normally contains frequencies from 0 to 4000 Hz. So the sampling rate and bit rate are calculated as follows:

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

Quantization error

→ The amplitude range $(-m_p, m_p)$ divided into L uniformly spaced interv... each of width

$$\Delta v = \frac{2m_p}{L}$$



→ Since sample value is approx by the midpoint of the subinterval (of height Δv) in which sample falls;

max quantization error is $\pm \Delta v/2$.

∴ quantization error lies in range of $\left(-\frac{\Delta v}{2}, \frac{\Delta v}{2}\right)$

$$\Rightarrow \tilde{q^2} = \frac{1}{\Delta v}\int_{-\Delta v/2}^{\Delta v/2} q^2\, dq$$

$$= \frac{1}{\Delta v}\left[\frac{q^3}{3}\right]_{-\Delta v/2}^{\Delta v/2}$$

$$= \frac{(\Delta v)^2}{12}$$

$$N_q = \frac{(2m_p/L)^2}{12} = \frac{m_p^2}{3L^2}$$

- If pulse detction error is neglegible (considering noise has not been accumulated much in digital signal transmitted over channel-due to the presance of regenrative repeaters)

- The source of error is mainly due to quantization noise. This quantization noise power is directly proportional to the square of step size of quantization zone.

$$N_q = \frac{(\blacktriangle v)^2}{12}$$

- Signal-to-Noise Ratio SNR is an indication of the quality of recieved signal.

As the output signal power is same as message signal power (in absence of pulse detection error)

$$S_0 = \widetilde{m^2(t)}$$

$$N_0 = N_q = \frac{(\Delta v)^2}{12} = \frac{m_p^2}{3L^2} \quad \left[\because \Delta v = \frac{2m_p}{L}\right.$$

$$\therefore \frac{S_0}{N_0} = \frac{\widetilde{m^2(t)}}{(\Delta v)^2/12} = 3L^2 \cdot \frac{\widetilde{m^2(t)}}{m_p^2}$$

Nonuniform Quantization: Principle of Progressive Taxation

- SNR is directly proportional to the signal power in case of unifrom quantization (step size ▲v= constant) where quantization noise power is constant.
- So signals with lower amplitudes will have low SNR and it is statistically found that smaller amplitudes predomaintes in speech and large amplitudes are less frequent.
- we require a constant SNR (same reception quality) for all the values of message signal power.
- this can be done by making samller step size (▲v) for the regions which have smaller amplitudes and so qunatization noise power will be small and signals with small amplitudes can also be recieved with good signal quality.
- however, large amplitudes regions can still be quantized with high step size (▲v) and still can be received with good signal quality.
- So SNR remains almost same over large dynamic range of input signal power.

Nonuniform Quantization

- This approach of equalizing SNR is similar to the use of progressive income tax to equalize the incomes.

- The loud talker and stronger signals are peanlized with higher noise steps (▲v) in order to compensate the soft talkers and weak signals.

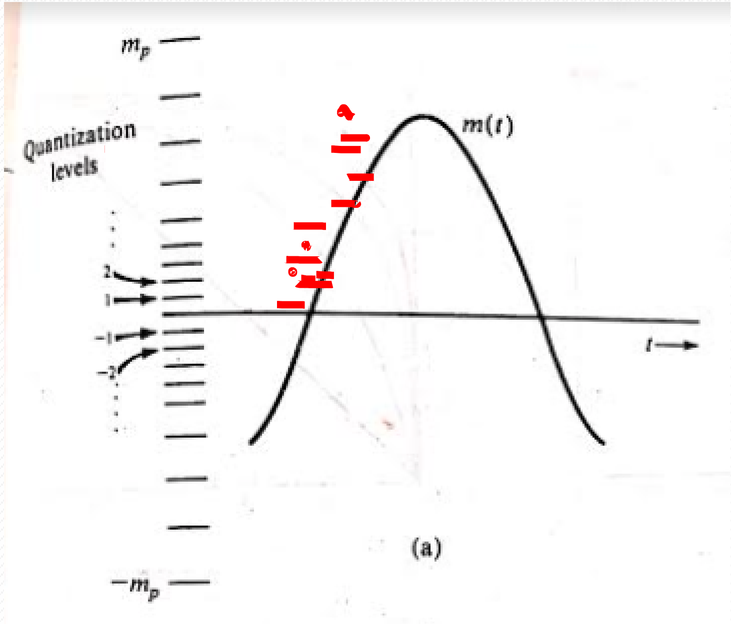


Figure: Non uniform quantization

The small amplitude signals are under represented with uniform quantization.

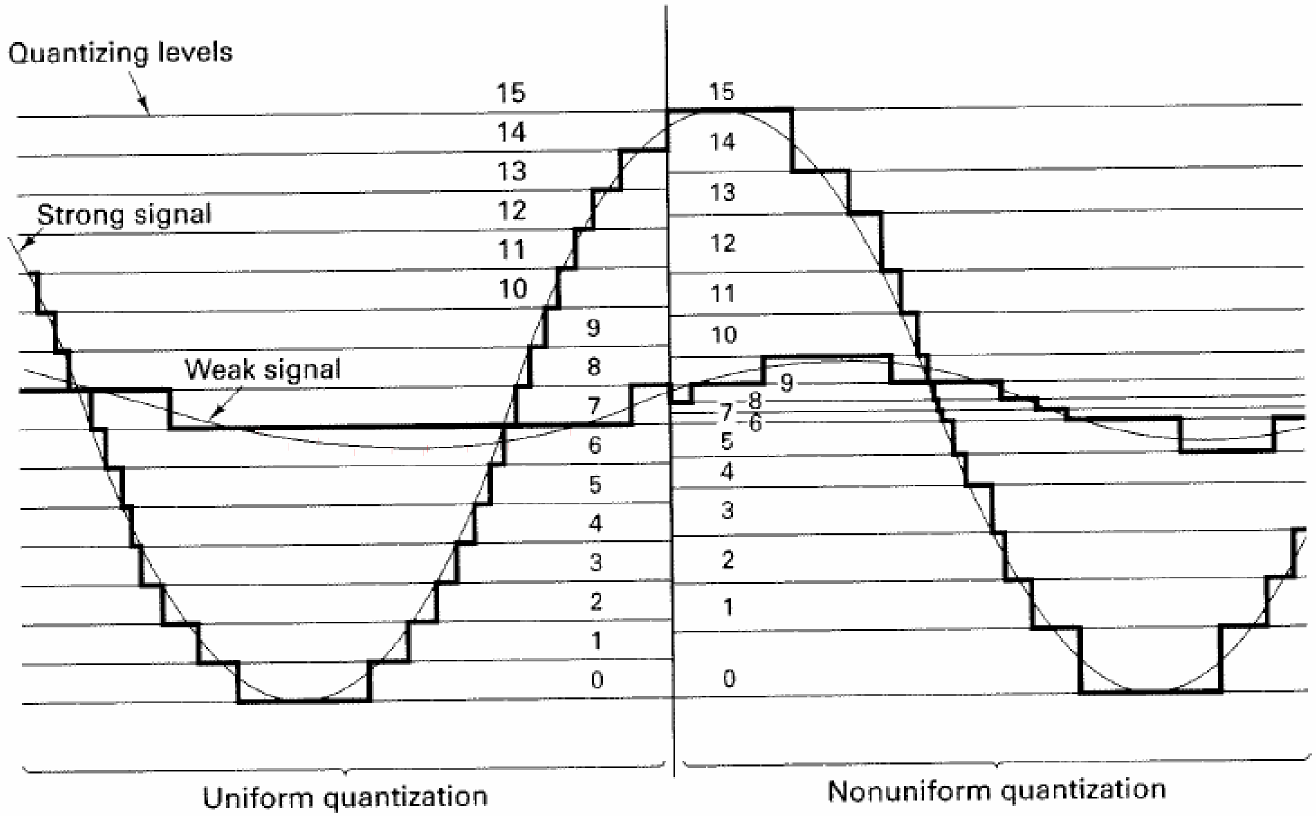
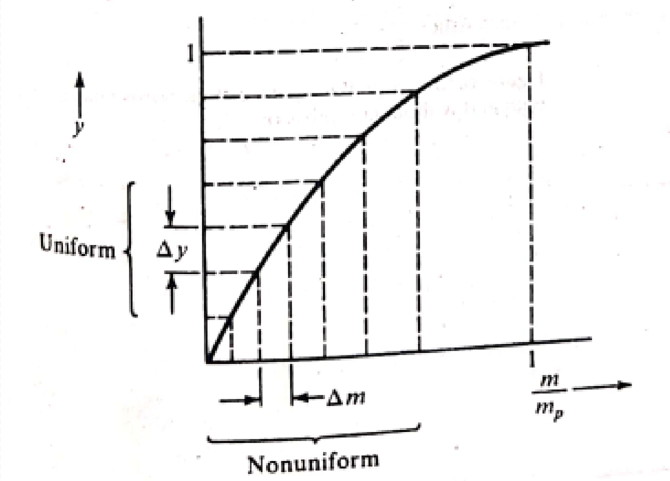


Figure 2.18 Uniform and nonuniform quantization of signals.

The Compander

- To perform nonuniform quantization, it is also possible to first compress the signal smaples and then using a uniform quantization.

- The input-output cahrecteristic of compressor is as shown in below figure. [horizontal axis: normaized input signal m/m_p & vertical axis: output signal y]



- The compressor maps input signal increments ▲m into large increments ▲y for small input signals and vice-versa.

- hence a given interval ▲m contains a larger no of steps (or smaller step size) when m is small.

u law for compression:

- An approximately logarithmic copression charecteristics gives quantization noise power nearly proprtional to signla power and thus making SNR independant of input signal power over a large dynamic range.

- The u law for compression is given by

$$y = \frac{1}{\ln(1+\mu)} \ln\left(1 + \frac{\mu m}{m_p}\right) \qquad 0 \leq \frac{m}{m_p} \leq 1$$

- Where u determines the degree of compression. To obtain nearly constant SNR , u should be grater than 100.

- logarithmic compressor can be realized by a semiconductor diode becuase V-I charecteristic of diode is of the desired form.

- The compressed samples must be restored to their original values at the reciever by using an expander with charecteristics complementary to that of compressor.

- The compressor and expnader together are called the **Compander.**

Encoding---Natural Binary Pulse Code

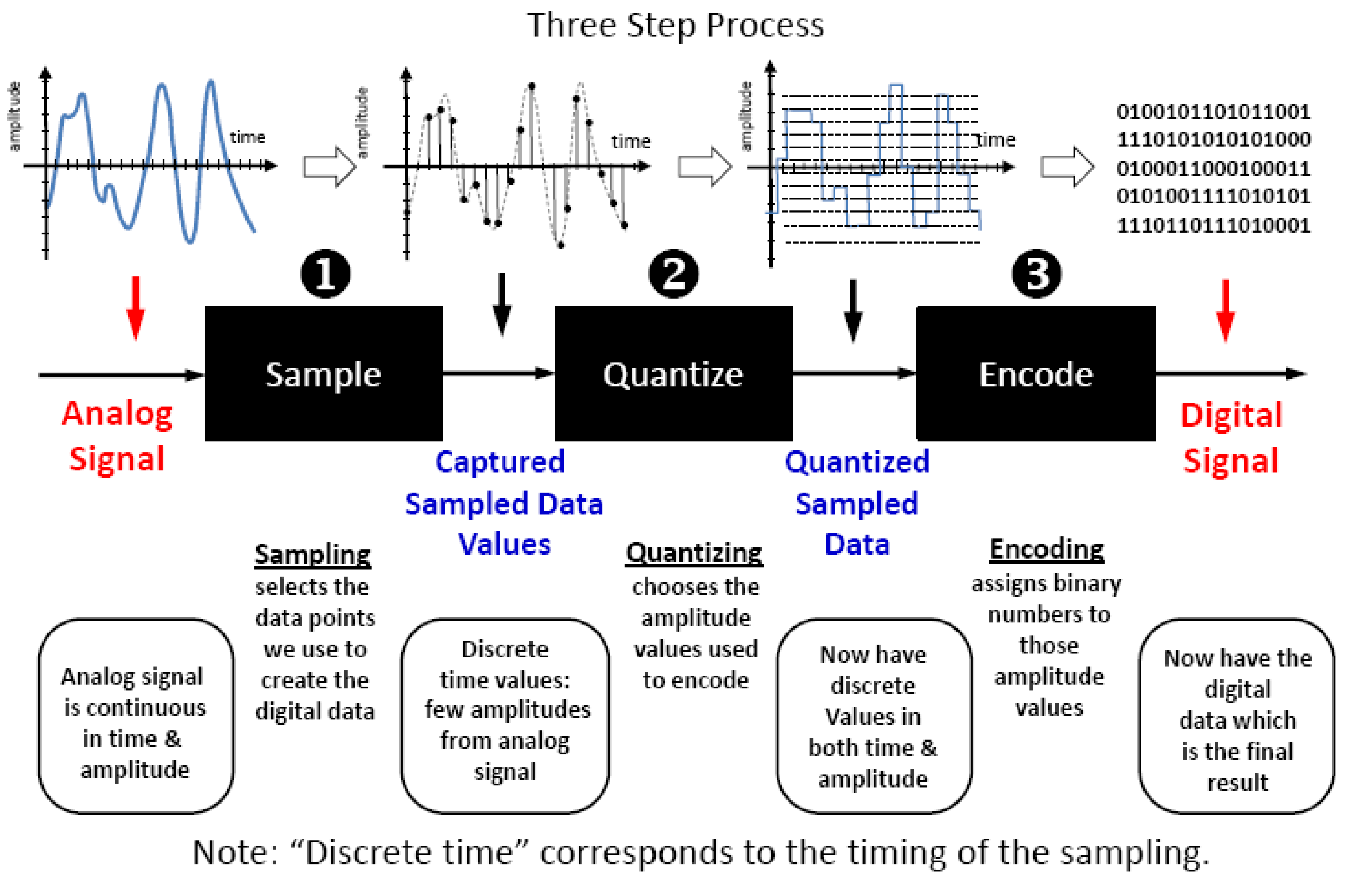
- To communicate sampled values, we send a sequence of bits that represents the quantized value.
- For 16 quantization levels, 4 bits are required.
- PCM can use a binary representation of value.

Digit	Binary equivalent	Pulse code waveform
0	0000	
1	0001	
2	0010	
3	0011	
4	0100	
5	0101	
6	0110	
7	0111	
8	1000	
9	1001	
10	1010	
11	1011	
12	1100	
13	1101	
14	1110	
15	1111	

APPLICATIONS OF PCM

- PCM was originally intended for use in telephone systems.
- But in the 21st century, it is also the standard way for digitalizing analog data such as in digital audio in computers, digital video and CD formats, telemetry, digital telephony and other digital audio applications.
- Also used in space communication .

Analog to Digital Conversion Process (ADC)



ADVANTAGES OF PCM

- **Low Noise Susceptibility:** PCM signals can be transmitted farther than analog signals without signal degradation, information loss, and distortion.
- **Repeatability:** The signal is completely regenerated by each repeater, making it noise-free at the start of each repeated transmission.
- **Encoded Signal:** A PCM signal can be modulated in such a way that only a specific decoder can make sense of the underlying data. This is useful when the data being sent requires a level of security.
- **Storage:** Compact disk is the classic application of PCM.

ADVANTAGES OF PCM

- PWM,PPM and PCM have all the advantages of FM when it comes to noise performance.
- Noise will have no effect at all unless its peaks are so large that they can be mistaken for pulses or so large negatively that they mask pulses.
- For sloping pulses noise superimpose on the pulses' sides and the result may well be a change in width. This will affect PWM and PPM but not PCM because PCM depends only on presence or absence of pulses.

ADVANTAGES OF PCM

- **Low Noise Susceptibility**
- A transmitted pulse that is close enough to the expected value of a binary, can be reliably reproduced into a binary one.
- This low noise susceptibility allows PCM signals to transmit farther than analog signals without signal degradation, information loss, and distortion.

ADVANTAGES OF PCM

- **Repeatability**
- The signal is completely regenerated by each repeater, making it noise-free at the start of each repeated transmission.
- Noise does not accumulate even after many passes through multiple repeaters.

ADVANTAGES OF PCM

- **Storage**
- **Encoded Signal**
  - A PCM signal can be modulated in such a way that only a specific decoder can make sense of the underlying data.
  - This is useful when the data being sent requires a level of security.

Differential PCM

- Voice and video signals represented in PCM exhibit high correlation, which means that PCM signals contain redundant information. The result is an inefficient coding.
- By removing the PCM information redundancy a more efficient coded signal may be obtained. This is done using DPCM.

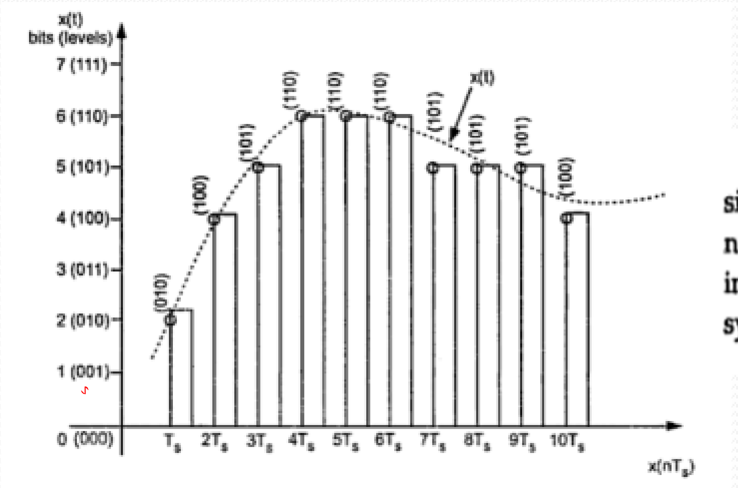


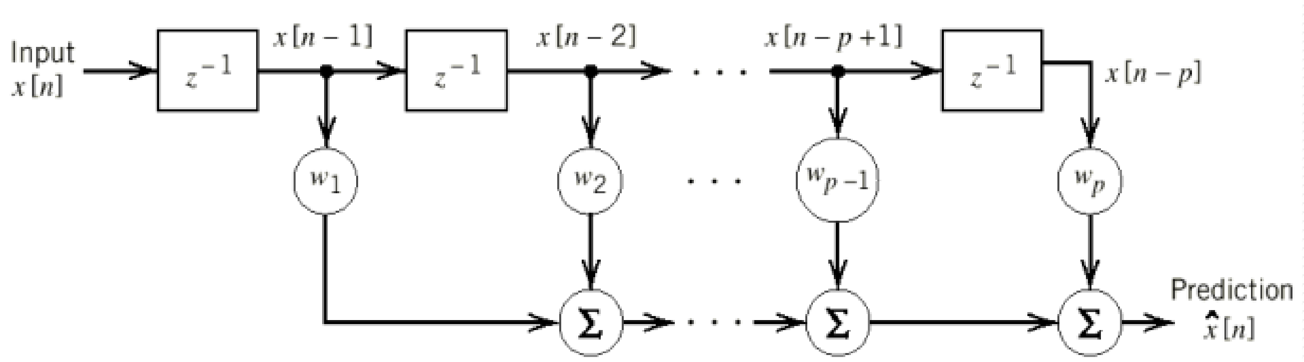
Fig. Redundant Information in PCM

The samples of a signal are highly corrected with each other. This is because any signal does not change fast. That is its value from present sample to next sample does not differ by large amount. The adjacent samples of the signal carry the same information with little difference. When these samples are encoded by standard PCM system, the resulting encoded signal contains redundant information.

Linear Prediction

- Linear prediction is a signal processing function performed by a finite-duration impulse response (FIR) discrete-time filter, as shownin Figure.
- Linear prediction consists of estimating the current sample of a signal from a certain number of previous samples. This is always possible when the signal samples are correlated.
- Linear predictor involves the use of three functional blocks: (1) a set of delay units, (2) a set of multipliers, and (3) a set of adders.
- For a linear predictor, the output is given by
- where p is the prediction order, wk's are the predictor coefficients.
- Since the linear prediction is an estimation, it results into an error called prediction error and given by:

$$e(n) = x(n) - \hat{x}(n)$$



Differential PCM

CASE-A

- Instead of transmitting the sample values, if we transmit the difference between successive samples i.e. instaed of transmitting m(k), if we transmit the difference d(k)= m(k)-m(k-1).
- At reciever, knowing d(k) and previous sample value m (k-1), it is possible to reconstruct m(k).
- Thus peak amplitude m_p of transmitted values is reduced considerably.
- So quantization noise $N_q = \frac{(\Delta v)^2}{12}$ is reduced

as the quantization interval $\Delta v = \frac{2m_p}{L}$ is reduced.

This means for given transmission bandwidth (fixed L and n); we can increase the SNR.

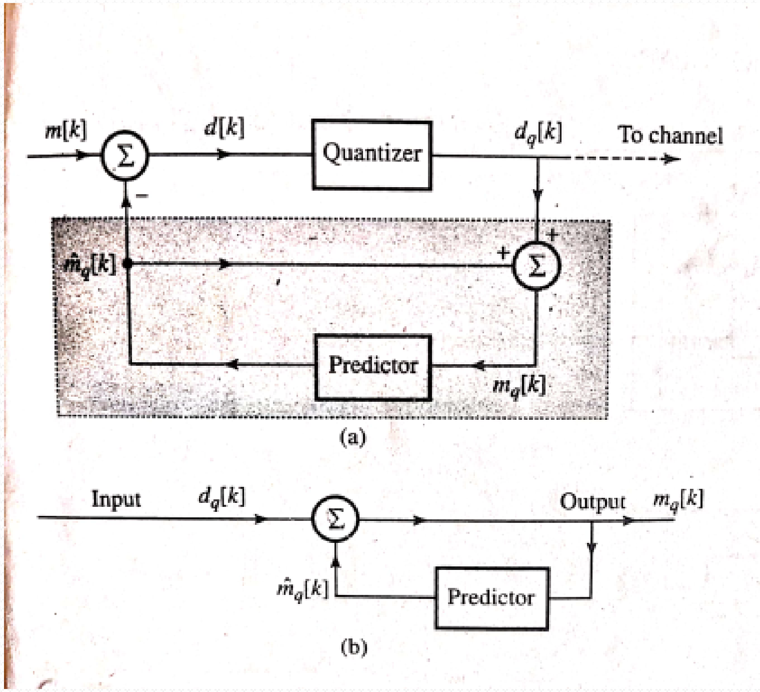
CASE-B

- This scheme is further improved by estimating the value of m(k) from a knowledge of previous sample values.
- If this estimate is $\widehat{m(k)}$ then we transmit the difference (prediction error) d(k)= m(k)-$\widehat{m(k)}$
- At receiver, we determine $\widehat{m(k)}$ from previous sample values and add it to recieved d(k) to generate m(k).
- *If prediction filter are well performing then prediction error d(k) will be even smaller than the difference between successive samples.*
- *So further SNR can be improved.*

Principle of Differential PCM

- In DPCM a linear prediction is performed on samples of a message signal $m(kTs)=m(k)$, then the prediction error (difference) is computed and fed to a quantizer to obtain the quantized value $d_q(k)=d(k)+q(k)$, where $q(k)$ is the quantization error.
- At reciever we can generate $\widehat{m(k)}$ from the past sample values to which recieved $d(k)$ is addded to generate $m(k)$.
- But in actuality, at reciever instaed of past samples $m(k-1)$, $m(k-2)$.... as well as $d(k)$, we have their quantized versions $m_q(k-1)$, $m_q(k-2)$.... So it is possible to determine the $\widehat{m_q(k)}$ instead of $\widehat{m(k)}$. This will increase the error in reconstruction.
- Better strategy is to determine estimate of m (k) instead of estimate of

Block diagram-DPCM Transmitter and Reciever



(a)

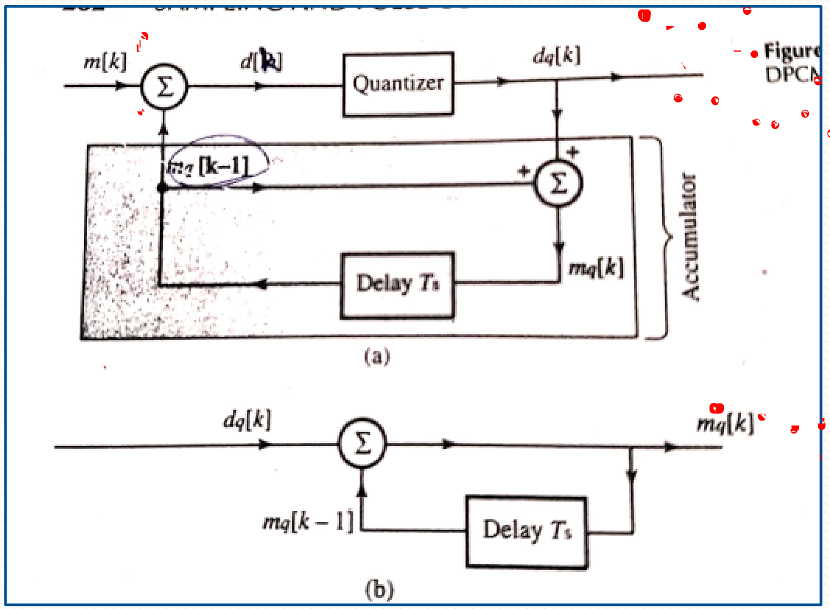
(b)

Delta Modulation (DM)

- In DM, the message signal is over-sampled (typically 4 times the Nyquist rate) to purposely increase correlation between adjacent samples.
- Which results in a small prediction error and that can be encoded using only 1 bit. (L=2)
- So DM is a 1-bit DPCM which is a special case of DPCM that uses only 2 quantization levels.
- The delta modulation uses First order predictor, which is just a time delay of Ts (sampling interval).
- So the prediction error (difference) d(k)=m(k)-$\widehat{mq(k)}$=m(k)-mq(k-1) which is quantized into two levels. So in DM information about the difference between successive samples is transmitted by a 1-bit codeword.
- So if m(k)ım_q(k-1) then positive pulse is generated in d(k) which gives rise to positive step (+▲) in $\widehat{m_q(t)}$. That way the DM attempt to equalize $\widehat{m_q(t)}$ to m(t) in small steps at each sampling instant. And if $m(k) < m_q(k-1)$ *then* negative *pulse is generated in* $d(k)$ *which gives rise to* negative *step* (−▲) *in* $\widehat{m_q(t)}$.
- The pulse train $d_q(k)$ is the delta modulated pulse train. Which can be qunatized into two levles (+ve pulse and -ve pulse) hence bit 1 is used to encode the positive pulse and bit 0 is used to encode the negative pulse.
- So DM provides $\widehat{m_q(t)}$ as a stair case approximation of m(t). Which when passed through a low pass filter , the coarness in stair case is eliminated and we get smoother and better

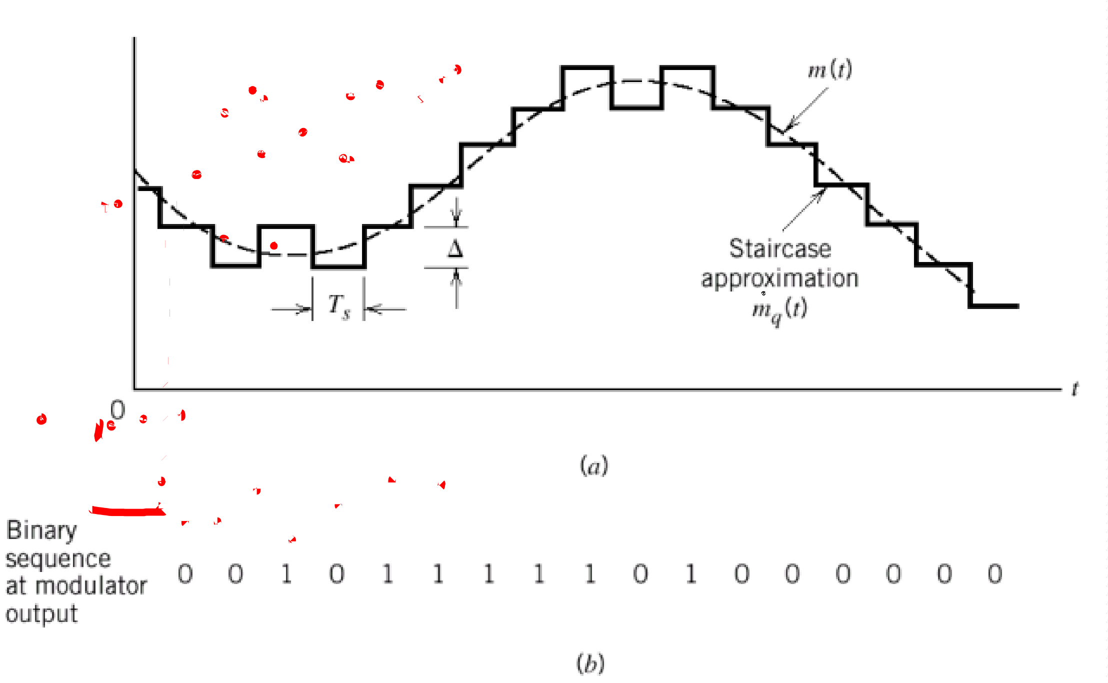
Block Diagram of Delta Modulation (DM)

-Special case of DPCM



- $^{35}_{17}$The transmitter of a DM system is given by a comparator, a one-bit quantizer, an accumulator, and an encoder.
- The receiver of a DM system is given by a decoder, an accumulator, and a low-pass filter.

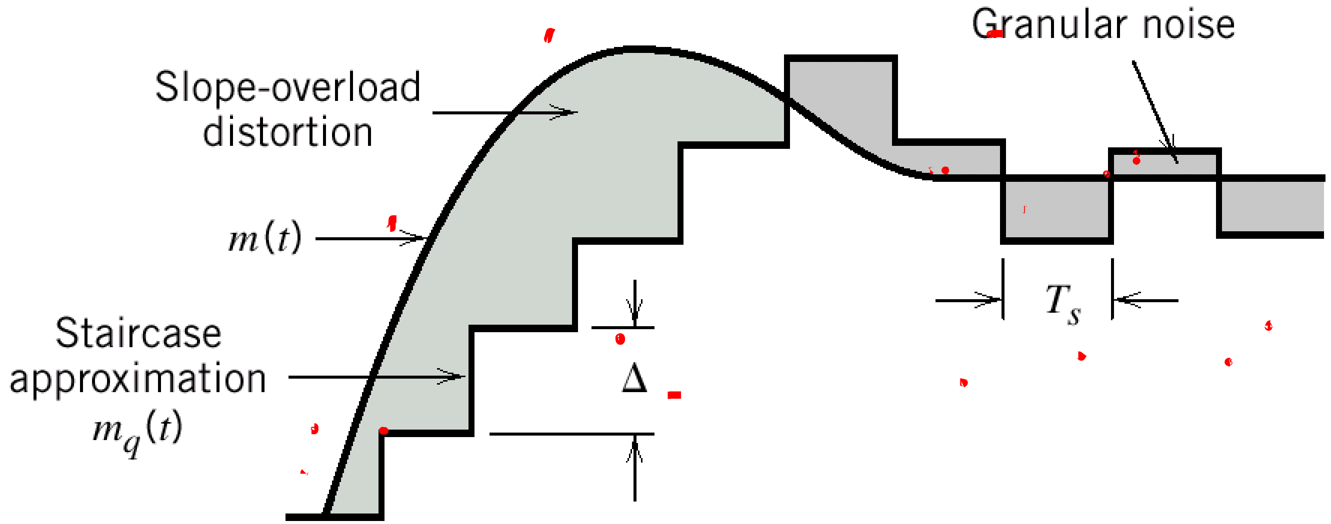
Illustration of Delta Modulation



Drawbacks of DM

- DM is subject to two types of quantization error: **Slope overload distortion and granular noise.**
- Slope overload distortion is due to the fact that the staircase approximation $\widehat{m_q}(t)$can't follow closely the actual curve of the message signal m(t).
- Which is called slope overload and gives rise to slope overload noise.
- In contrast to slope-overload distortion, granular noise occurs when ▲ is too large relative to the local slope characteristics of m(t). (**granular noise is similar to quantization noise in PCM)
- It seems that a large ▲ is needed for rapid variations of m(t) to reduce the slope-overload distortion and a small ▲ is needed for slowly varying m(t) to reduce the granular noise. The optimum can only be a compromise between the two cases.
- To satisfy both cases, an adaptive DM is needed, where the step size can be adjusted in accordance with the input signal m(t).

Illustration of the two different forms of quantization error in delta modulation.



Condition for Slope overload distortion occurrence:

Slope overload distortion will occur if

$$A_m > \frac{\delta}{2\pi f_m T_s}$$

where T_s is the sampling period.

Let the sine wave be represented as,

$$x(t) = A_m \sin(2\pi f_m t)$$

Slope of $x(t)$ will be maximum when derivative of $x(t)$ with respect to 't' will be maximum. The maximum slope of delta modulator is given

$$\text{Max. slope} = \frac{\text{Step size}}{\text{Sampling period}}$$

$$= \frac{\delta}{T_s} \qquad \text{..............(1)}$$

Slope overload distortion will take place if slope of sine wave is greater than slope of delta modulator i.e.

$$\max\left|\frac{d}{dt}x(t)\right| > \frac{\delta}{T_s}$$

$$\max\left|\frac{d}{dt}A_m \sin(2\pi f_m t)\right| > \frac{\delta}{T_s}$$

$$\max|A_m\, 2\pi f_m \cos(2\pi f_m t)| > \frac{\delta}{T_s}$$

$$A_m\, 2\pi f_m > \frac{\delta}{T_s}$$

or

$$\boxed{A_m > \frac{\delta}{2\pi f_m T_s}} \qquad \text{................(2)}$$

Adaptive Delta Modulation

- Adaptive delta modulation (ADM) is a modification of DM, in which the step size is adapted to the slope (variation) of the message signal.
- If successive errors are of opposite polarity, then the delta modulator is operating in the granular mode; in such a case it is advantageous to use reduced step size.
- If successive errors are of the same polarity, then the delta modulator is operating in its slope-overload mode; in this case, the step size should be increased.

- ADM allows much larger dynamic range of message signal amplitudes.

Adaptive delta modulation system: (*a*) Transmitter. (*b*) Receiver.

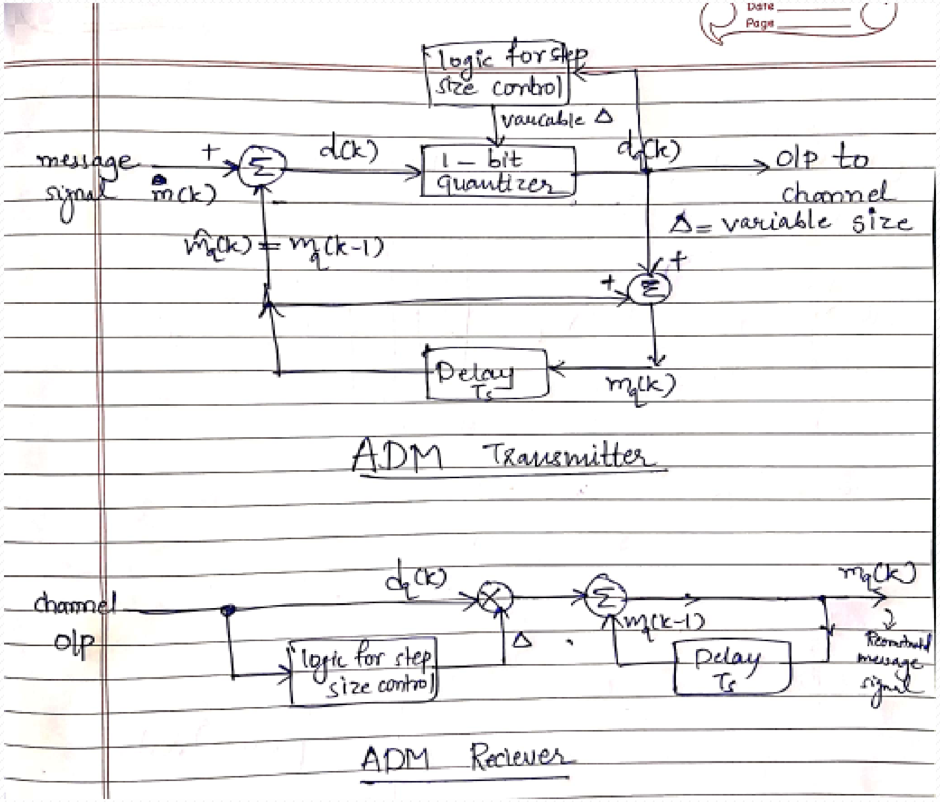


Illustration of Adaptive Delta Modulation

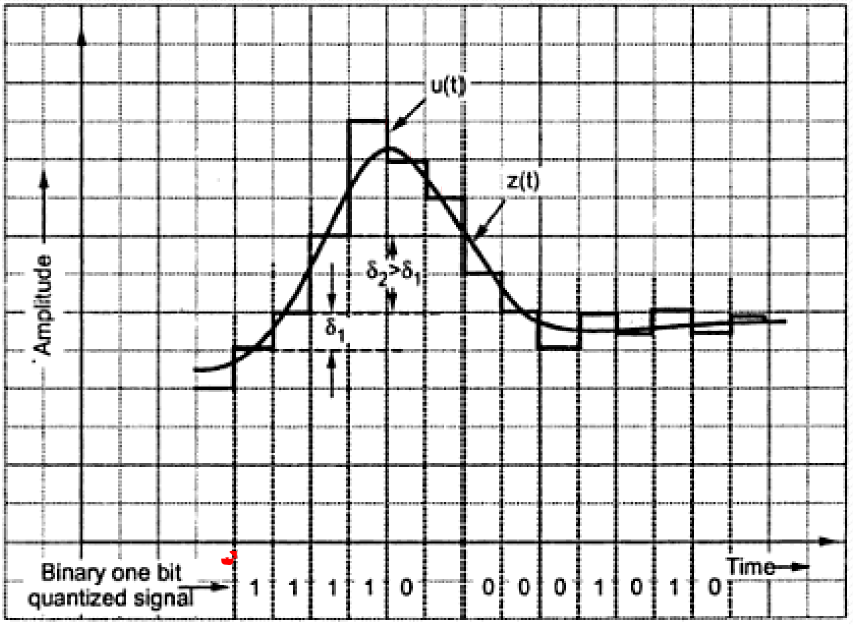


Fig. Waveforms of adaptive delta modulation

Comparison between PCM,DM,ADM and DPCM

Sr. No.	Parameter	PCM	Delta modulation (DM)	Adaptive Delta Modulation (ADM)	Differential Pulse Code Modulation (DPCM)
1	Number of Bits	It can use 4, 8 or 16 bits per sample.	It uses only one bit for one sample.	Only one bit is used to encode one sample.	Bits can be more than one but are less than PCM.
2	Levels, step size	The number of levels depend on number of bits. Level size is fixed.	Step size is fixed and cannot be varied.	According to the signal variation, step size varies (Adapted).	Fixed number of levels are used.
3	Quantization error and Distortion	Quantization error depends on number of levels used.	Slope overload distortion and granular noise is present.	Quantization error is present but other errors are absent.	Slope overload distortion and quantization noise is present.
4	Bandwidth of transmission channel	Highest bandwidth is required since number of bits are high.	Lowest bandwidth is required.	Lowest bandwidth is required.	Bandwidth required is lower than PCM.
5	Feedback.	There is no feedback in transmitter or receiver.	Feedback exists in transmitter.	Feedback exists.	Feedback exists.
6	Complexity of notation.	System is complex.	Simple.	Simple.	Simple.
7	Signal to noise ratio	Good	poor	better than DM	fair
8	Area of applications	Audio and video Telephony	speech and images	speech and images	speech and video

Next part shows the comparison for voice encoding

Sr.No.	Parameter	PCM	DM	ADM	DPCM
9	Sampling rate kHz	8	64-128	48-64	8
10	Bits/sample	7 - 8	1	1	4-6
11	Bit rate	56-64	64-128	46-64 48	32-48

Time Division Multiplexing

- This technique combines time-domain samples from different message signals (sampled at the same rate) and transmits them together across the same channel.

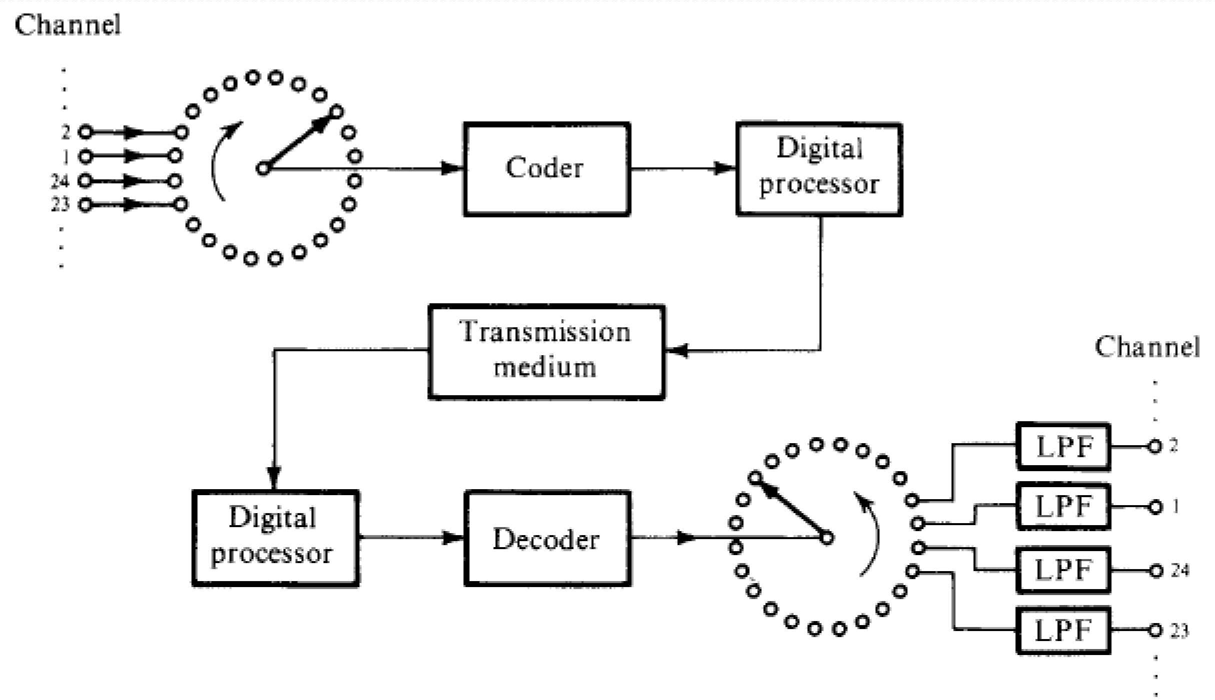
- The multiplexing is performed using a commutator (switch) as shown in Figure . At the receiver a decommutator (switch) is used in synchronism with the commutator to demultiplex the data.

- All 24 channels are sampled in a sequence. Commutators are high speed electronics switching circuits. It is rotating clockwise, takes samples of 1st message, and so on..up to the sample of 24th message till Ts duartion.

- In one rotation lasting for Ts duartion, it takes 24 samples of 24 message signals. The commutator has a speed of rotation set at 8000 Hz (sampling frequency of voice message signal). So it takes 8000 samples of each 24 meassage signals in a period of 1 second.

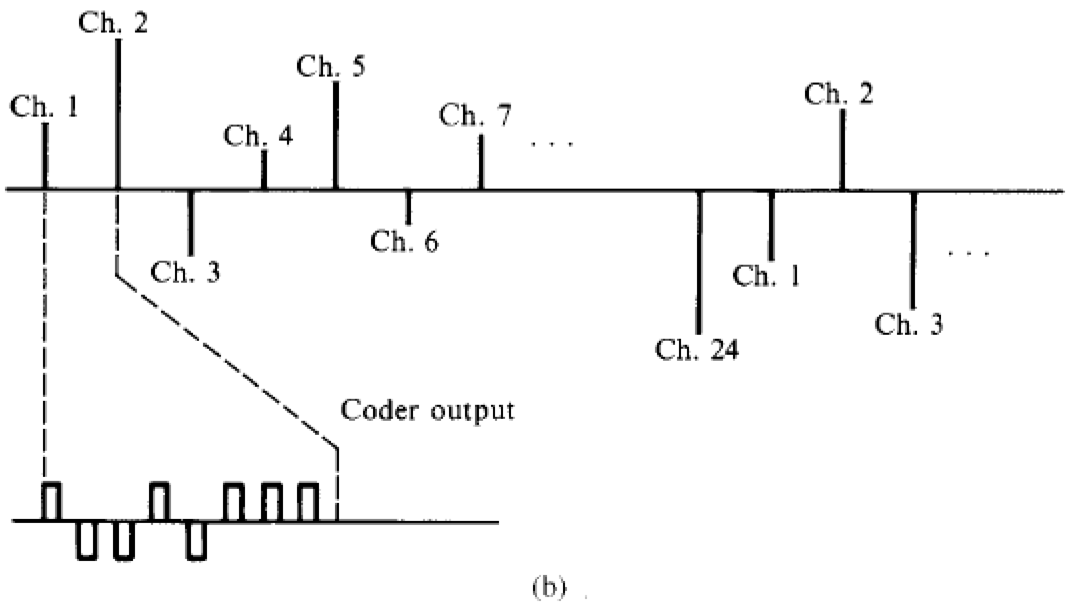
North American PCM Telephony

- Twenty four T1 carriers (64kb/s) i.e. 24 PCM signals are time-division multiplexed to generate one DS1 carrier (1.544 Mb/s)



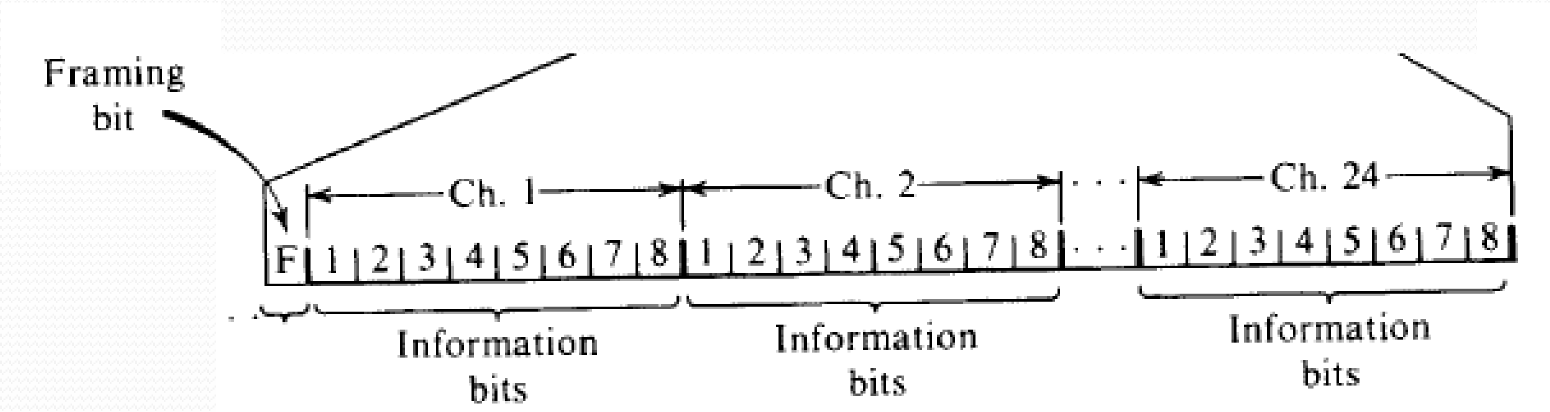
(a)

Each channel has 8 bits – 24 Channels



- Each frame has 24 X 8 = 192 information bits
- Frame time = 1/8000 = 125 μs.

bit rate in T1 system



- There are 8 bits from each channel. There are 24 such channels which are Time division multiplexed. There is one frame bit at the beginning. Thus there are total 24 channels*8 bits/channel + 1frame bit= 193 bits in a frame.

- At rx it is necessary to be sure whree each frame begins in oredr to sepearte the information bits correctly. For this purpose, framing bit is added at the beginning of each frame.

- These frames are transmitted at 8 KHz rate. becuase the successive samples from the same channel occur at 8 KHz.

- Hence bit rate in T1 system = 193 bits/frame* 8000 frames/sec= 1.544 Mbps.

- Using Time division multiplexed PCM signals having overall bit rate of 1.544 Mbps require a channel bandwidth of 32 kHz only. (B=nfs/2)

What is Line Coding?

- **The input to a digital system is in the form of sequence of digits.**

- The input can be from the sources such as data set, computer, digitized voice (PCM), digital TV or Telemetry equipment.

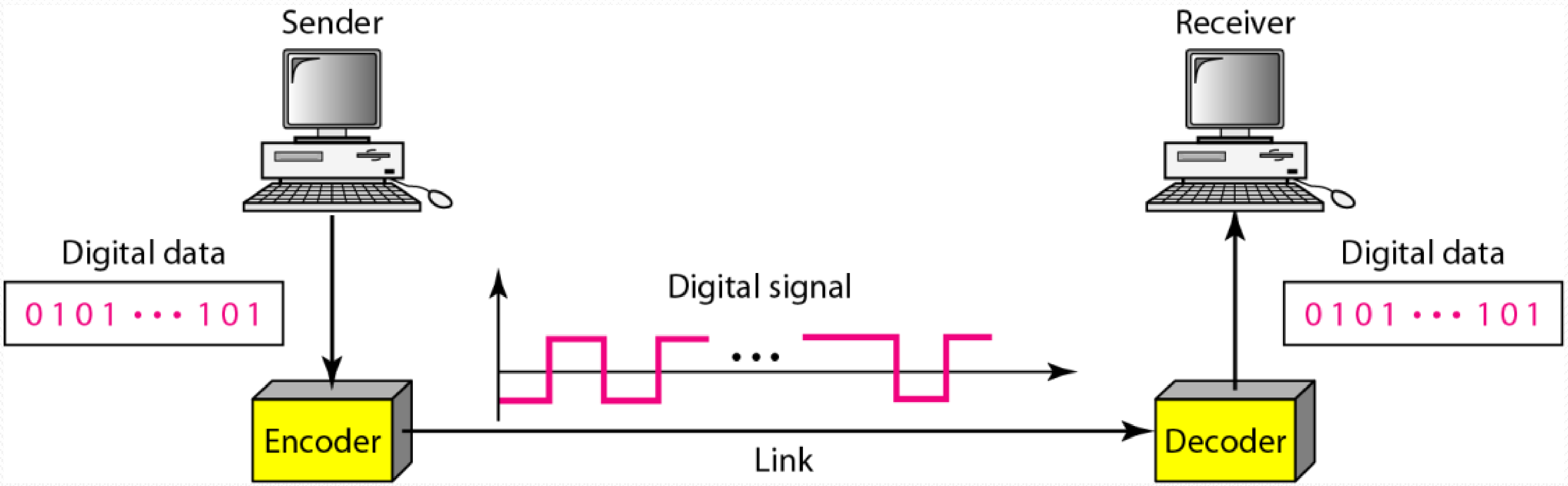
- ***Line coding*** **is the process in which the digital input is coded into electrical pulses or waveforms for the transmission over channel.**

- **Regenerative Repeaters** are used at regular intervals along a digital transmission line to detect the incoming digital signal and to transmit the new clean pulse for the further transmission along the line.

LINE CODING

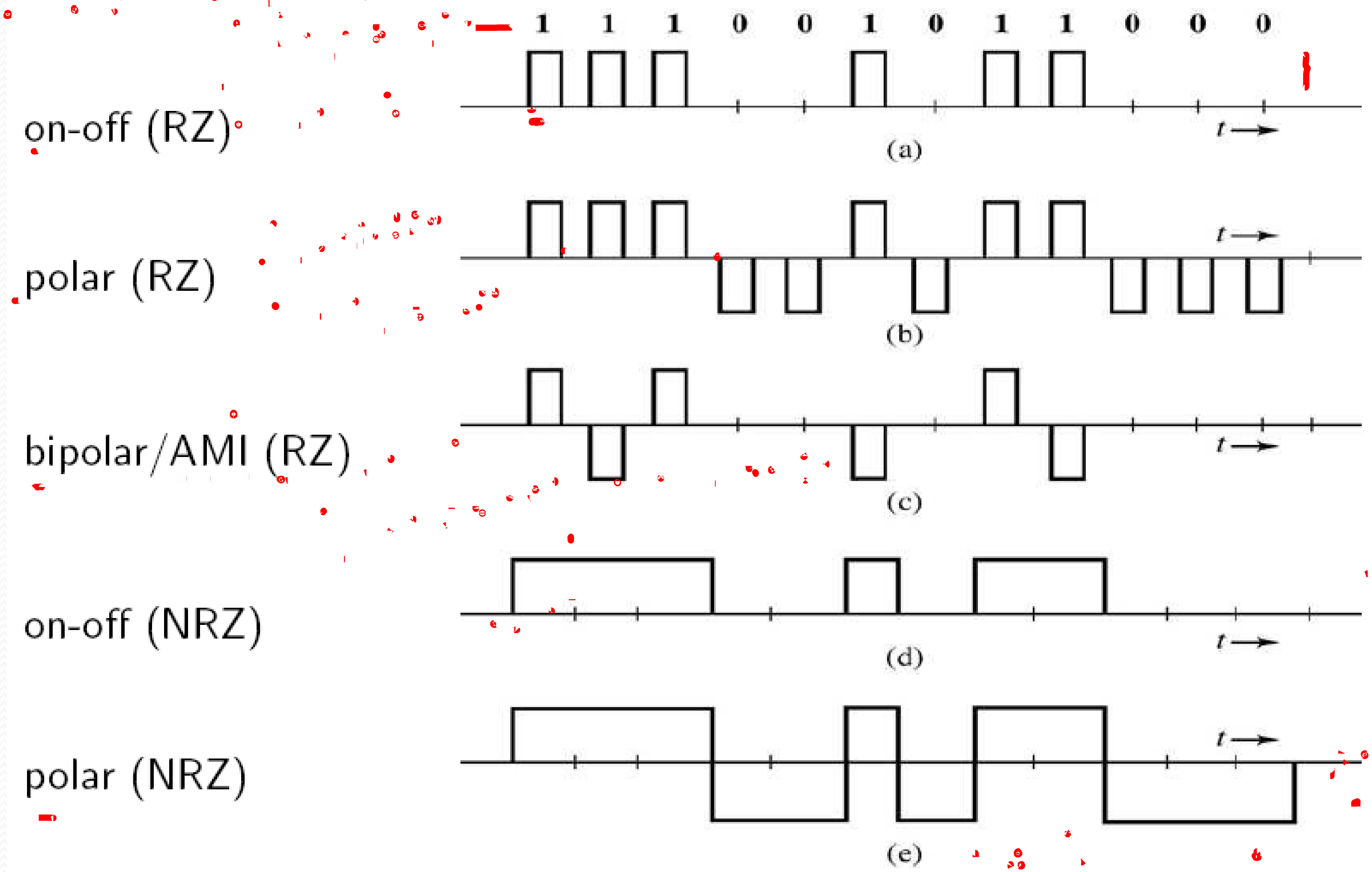
- Converting a string of 1's and 0's (digital data) into a sequence of signals that denote the 1's and 0's.
- For example a high voltage level (+V) could represent a "1" and a low voltage level (0 or -V) could represent a "0"

LINE CODING



Sender
Receiver
Digital data
0101 • • • 101
Digital signal
Encoder
Link
Decoder
Digital data
0101 • • • 101

Line Coding-Examples



RZ = return to zero, NRZ = non return to zero

Line Code Requirements

1. **Ttransmission bandwidth:** as small as possible

2. **Power efficiency:** as small as possible for required data rate and error probability (noise immunity)

   e.g. Ploar scheme requires least power for a given noise immunity

3. **Error detection/correction capability:**

   e.g. AMI where **0** is encoded by no pulse and **1** is encoded by +p(t) or

   - p(t) pulse depending on the previous polarity of the pulse.

4. **Timing information**: clock information must be extracted from the signal

   e.g. if pulses are transmitted at rate Rb pulses per second, the periodic timing information-clock signal at Rb Hz is required at repeater satation to sample the incoming pulses.

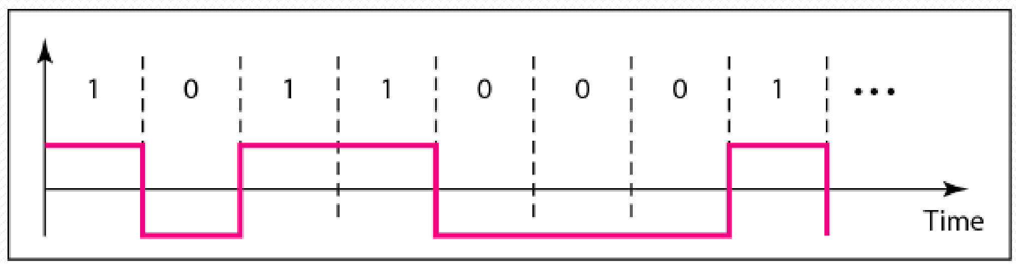
   This can be done by rectification of incoming pulse train which will result in periodic signal of clock frequency Rb hz. The timing information can be extracted by using a resoanant circuit tuned to the clock frequency.

Line Code Requirements

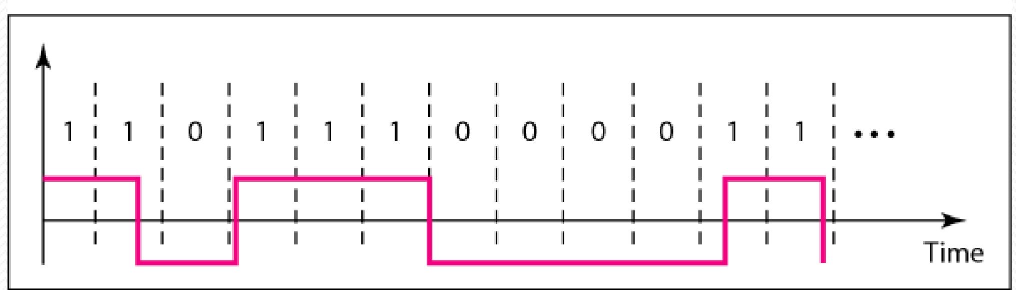
5. **Transparency:** It should transmitt a digital signal correctly regardless of the pattern of 1's and 0's.

   E.g. The long string of 0's in a sequence, will result in no pulses or no signal at the input of resoanant circuit and hence sinusoidal output of resoannt circuit starts decaying which causes error in timing information.

6. **Self synchronization** - The clocks at the sender and the receiver must have the same bit interval. If the receiver clock is faster or slower it will misinterpret the incoming bit stream.



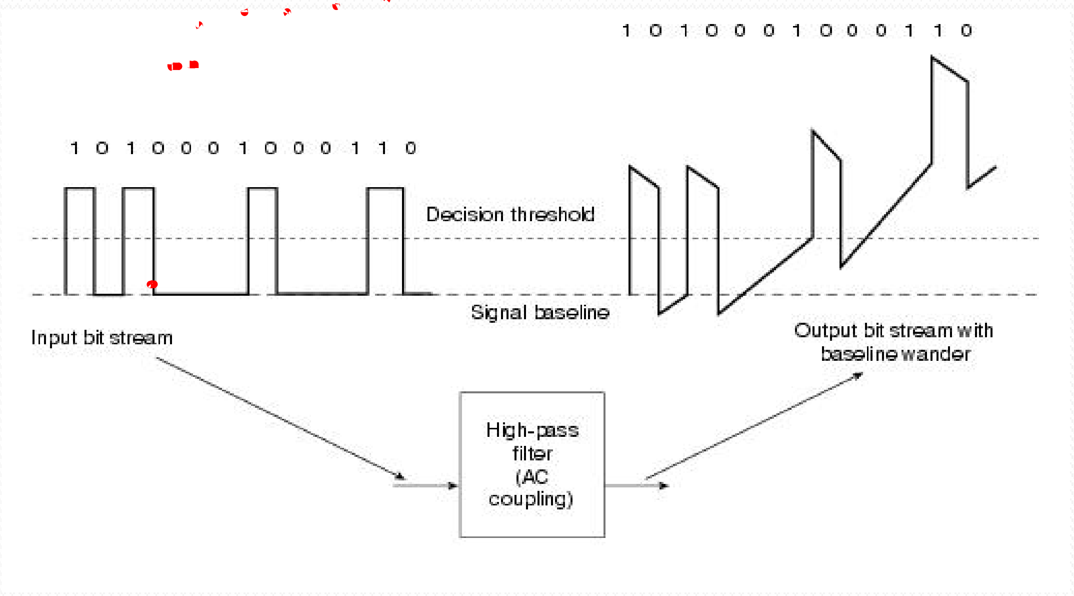
a. Sent



Line Code Requirements

7. **Baseline wandering** - If the incoming signal does not vary over a long period of time, the spectrum creates very low frequencies (DC components) and significant power in low frequency drift the baseline (dc wander) and thus cause errors in detection of incoming data elements.

- A good line encoding scheme will prevent long runs of fixed amplitude.

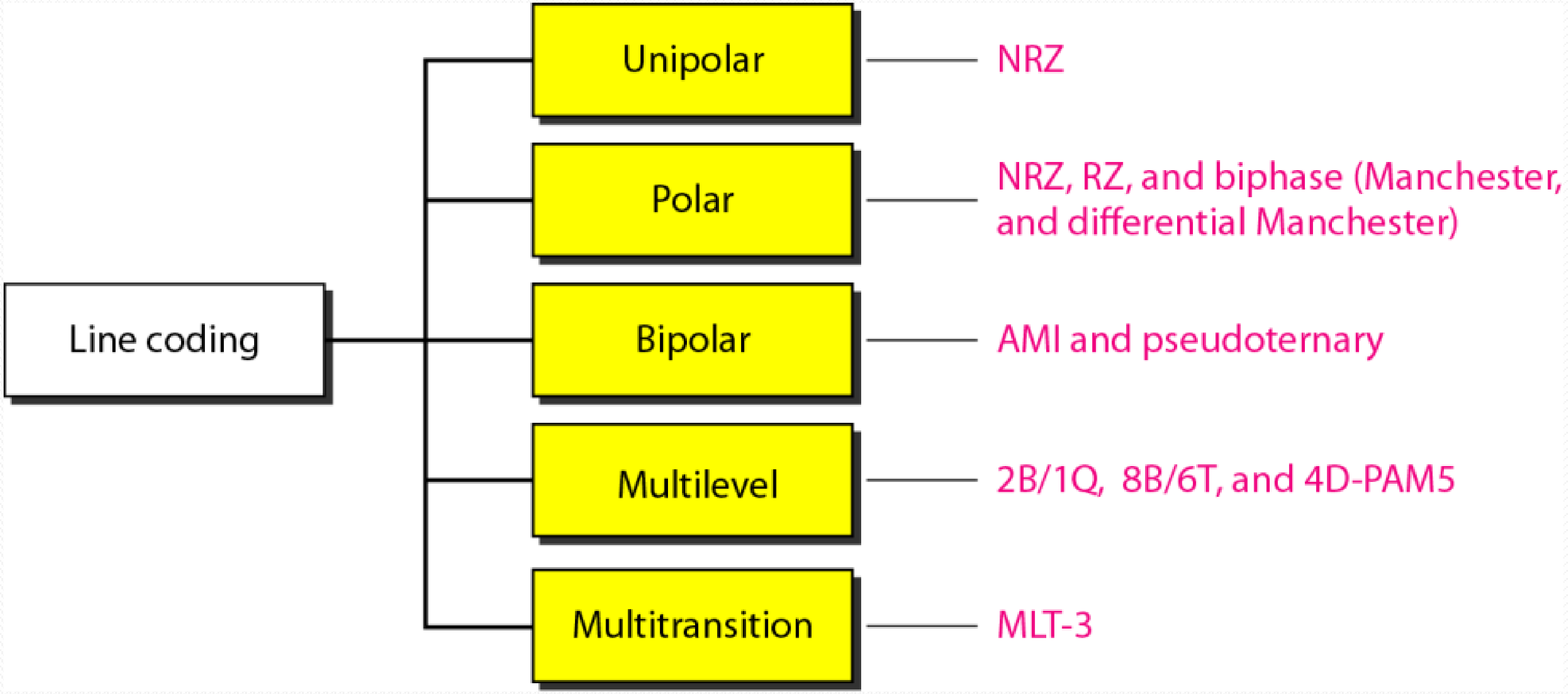


LINE CODING

DC components

- When the voltage level in a digital system is constant for a while, the spectrum creates very low frequencies. These frequencies around zero, called DC components create problems for a system that can not pass low frequencies.
- For example, a telephone line cannot pass frequencies below 200 Hz

LINE CODING

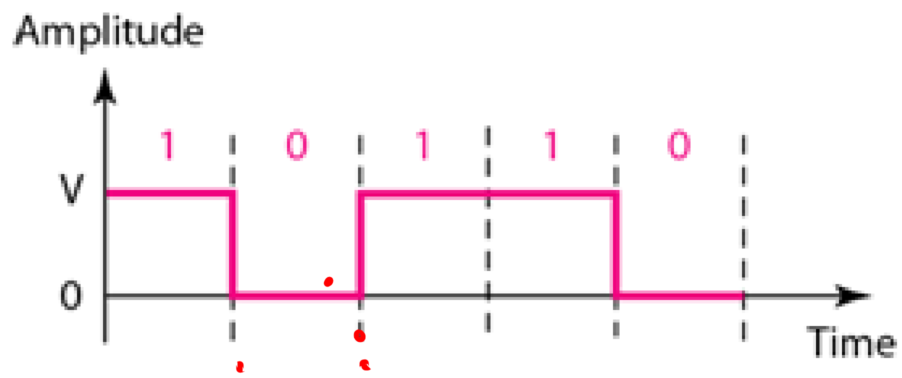


Line coding
Unipolar
Polar
Bipolar
Multilevel
Multitransition
NRZ
NRZ, RZ, and biphase (Manchester, and differential Manchester)
AMI and pseudoternary
2B/1Q, 8B/6T, and 4D-PAM5
MLT-3

Unipolar NRZ

- **All signal levels are on one side of the time axis - either above or below.**

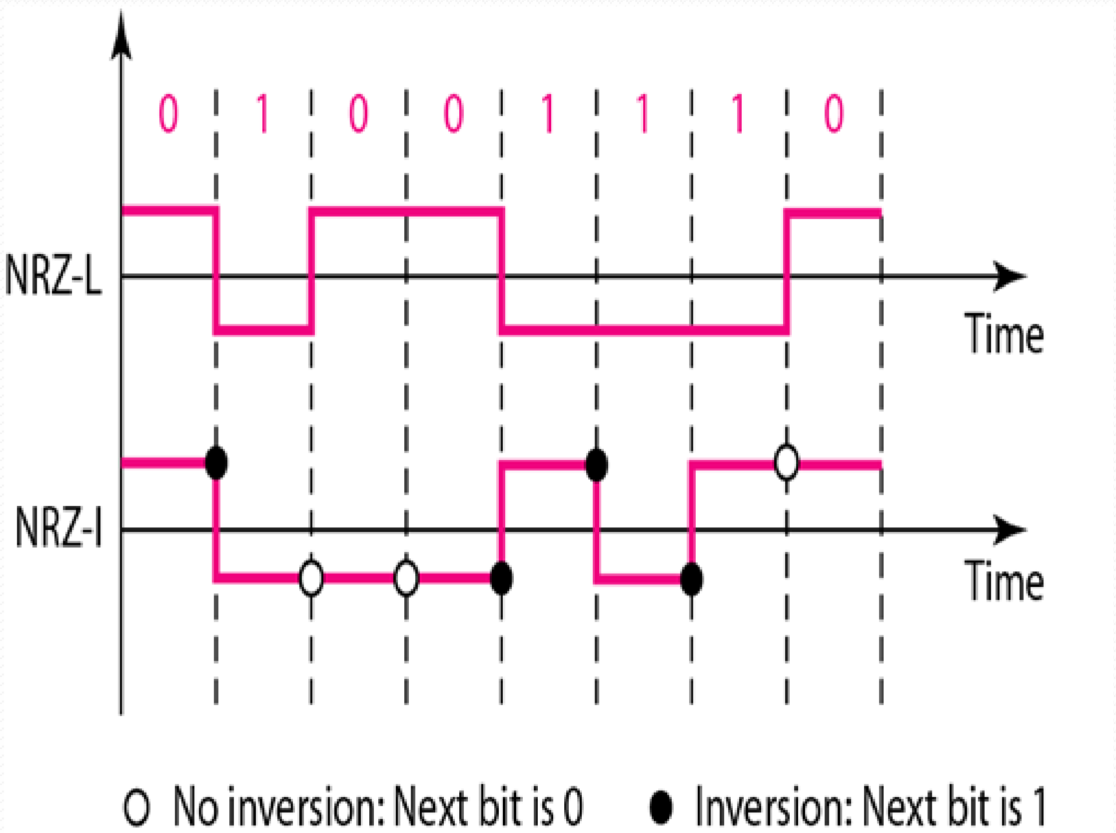
- NRZ - **Non Return to Zero** scheme is an example of this code. The signal level does not return to zero during a symbol transmission.



Polar - NRZ

- **The voltages are on both sides of the time axis.**
- Polar NRZ scheme can be implemented with two voltages. E.g. +V for 1 and -V for 0.
- There are two versions:
  - **NZR - Level (NRZ-L)** - positive voltage for one symbol and negative for the other.
  - **NRZ - Inversion (NRZ-I)** - the change or lack of change in polarity determines the value of a symbol. E.g. a "1" symbol inverts the polarity a "0" does not.

Polar - NRZ


0
1
0
0
1
1
1
0
NRZ-L
Time
NRZ-I
Time
No inversion: Next bit is 0
Inversion: Next bit is 1

Question

1. ***Draw the graph of the NRZ-I scheme using each of the following data stream assuming that the last signal level has been positive.***

   a. 11111111

   b. 00000000

   c.00110011

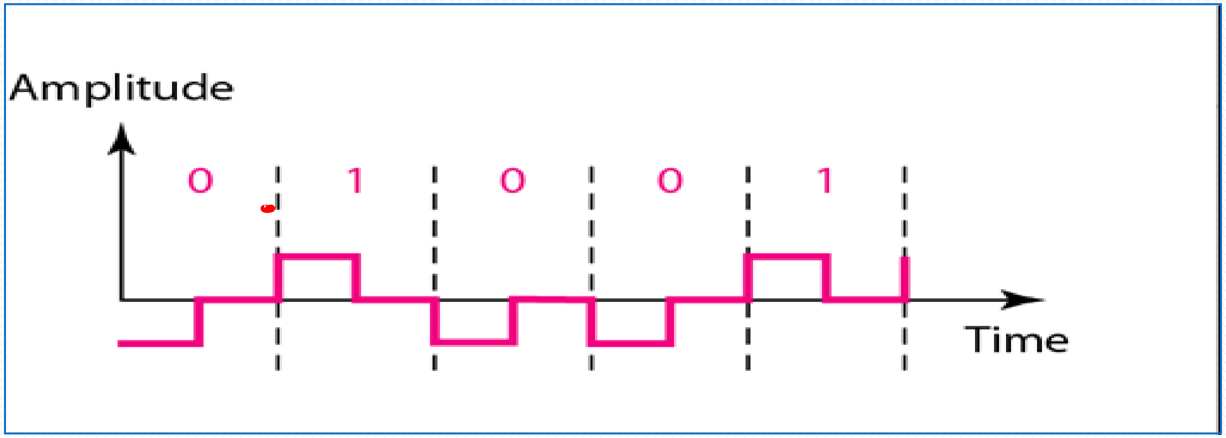
   d. 01010101

Problem with NRZ coding

- The main problem with NRZ encoding occurs when the sender and receiver clocks are not synchronized. The receiver does not know when one bit has ended and the next bit is starting.

Polar - RZ

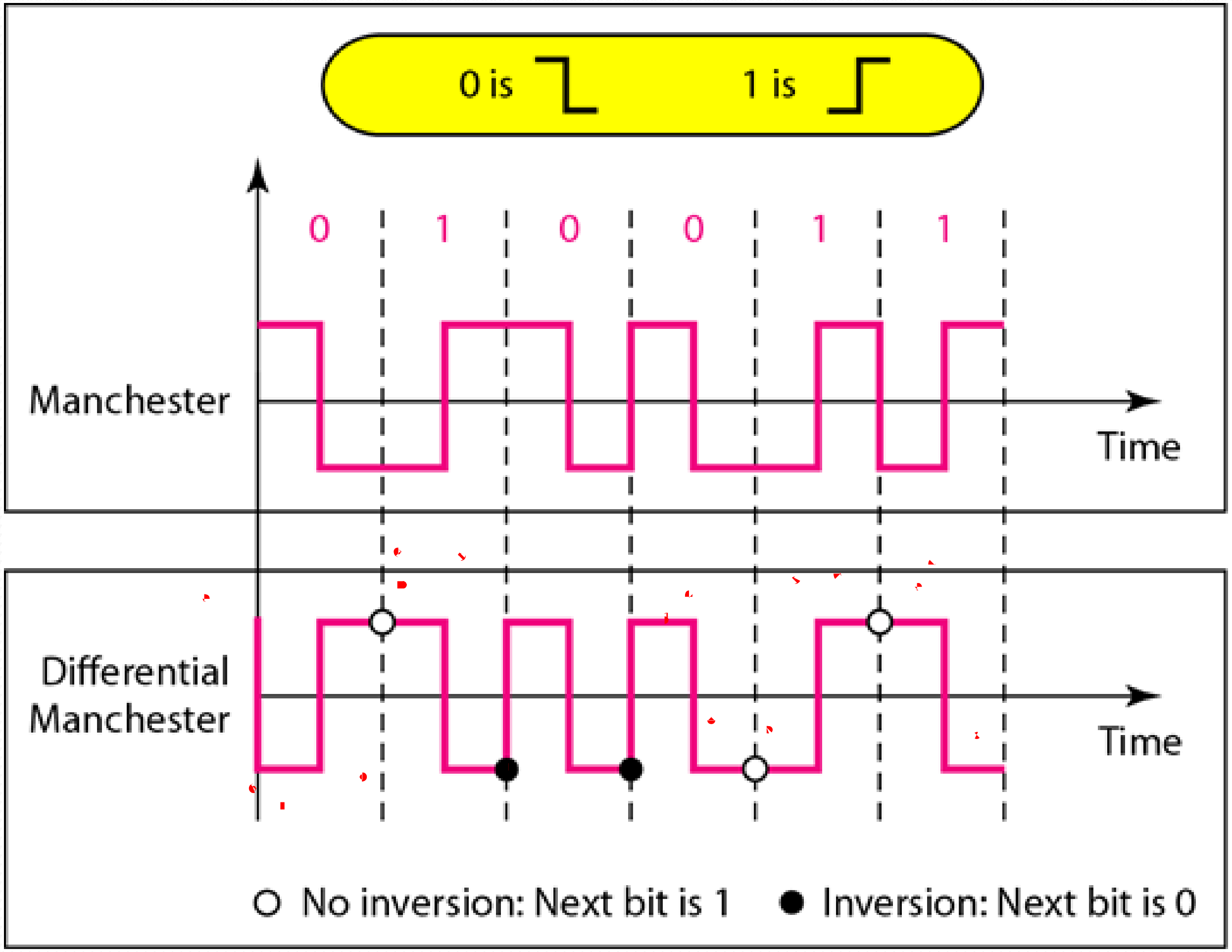
- The Return to Zero (RZ) scheme uses three voltage values. +, 0, -.
- **Each symbol has a transition in the middle. Either from high to zero or from low to zero.**
- **Self synchronization - transition indicates symbol value.**



Polar - Biphase: Manchester and Differential Manchester

- Manchester coding consists of combining the NRZ-L and RZ schemes.
  - Every symbol has a level transition in the middle: from high to low or low to high. Uses only two voltage levels.
- Differential Manchester coding consists of combining the NRZ-I and RZ schemes.
  - Every symbol has a level transition in the middle. But the level at the beginning of the symbol is determined by the symbol value. One symbol causes a level change the other does not.

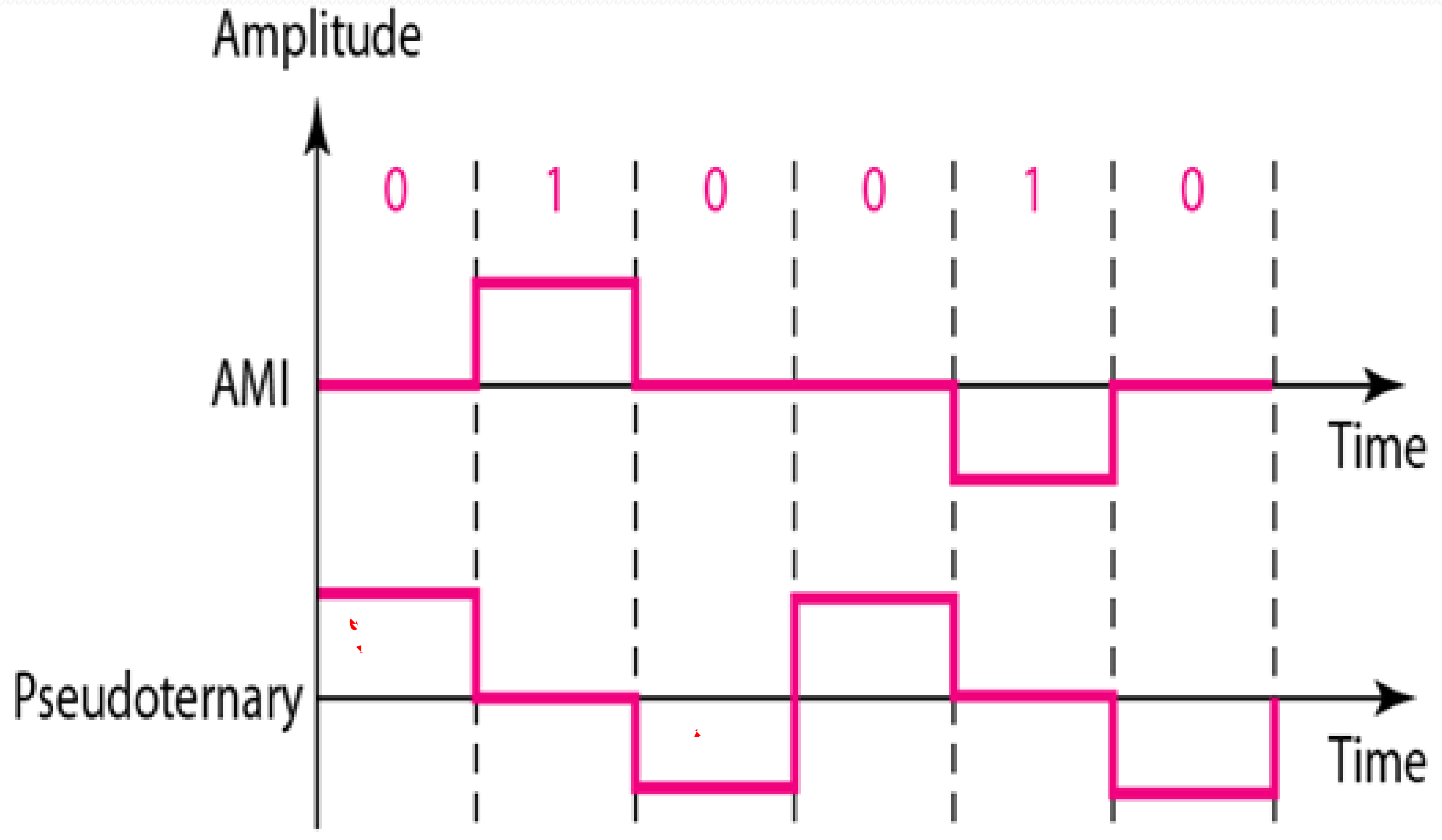
Polar - Biphase: Manchester and Differential Manchester



Bipolar - AMI and Pseudoternary

- Code uses 3 voltage levels: - +, 0, -, to represent the symbols (note not transitions to zero as in RZ).
- Voltage level for one symbol is at "0" and the other alternates between + & -.
- Bipolar Alternate Mark Inversion (AMI) - **the "0" symbol is represented by zero voltage and the "1" symbol alternates between +V and -V.**
- Pseudoternary is the reverse of AMI.

Bipolar - AMI and Pseudoternary



Multilevel Schemes

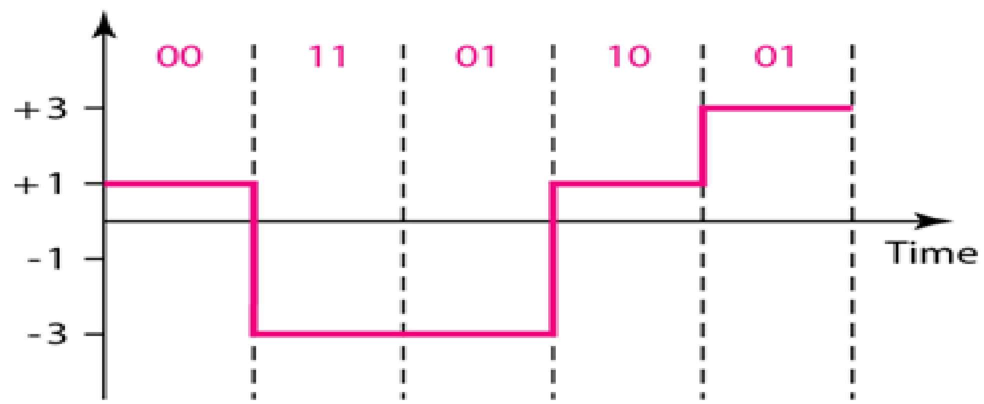
- We use the notation mBnL, where m is the no of data elements in a symbol, B represents binary data, n represents the length of the signal pattern and L is the number of signal levels.

- A letter is often used in place of L : B(binary) for L=2, T ternary) for L=3 and Q(quaternary) for L=4

Multilevel 2B1Q Scheme

	Previous level: positive	Previous level: negative
Next bits	Next level	Next level
00	+1	-1
01	+3	-3
10	-1	+1
11	-3	+3

Transition table



Assuming positive original level

Applications of Line Coding

- **NRZ encoding**: RS232 based protocols
- **Manchester encoding**: Ethernet networks
- **Differential Manchester encoding**: token-ring networks
- **NRZ-Inverted encoding:** Fiber Distributed Data Interface (FDDI)
- **2B1Q scheme**: high-bit-rate digital subscriber line (HDSL)

Scrambling

- **Scrambling** is a technique that does not increase the number of bits and does provide synchronization.

- Problem with technique like Bipolar AMI(Alternate Mark Inversion) is that continuous sequence of zero’s create synchronization problems.

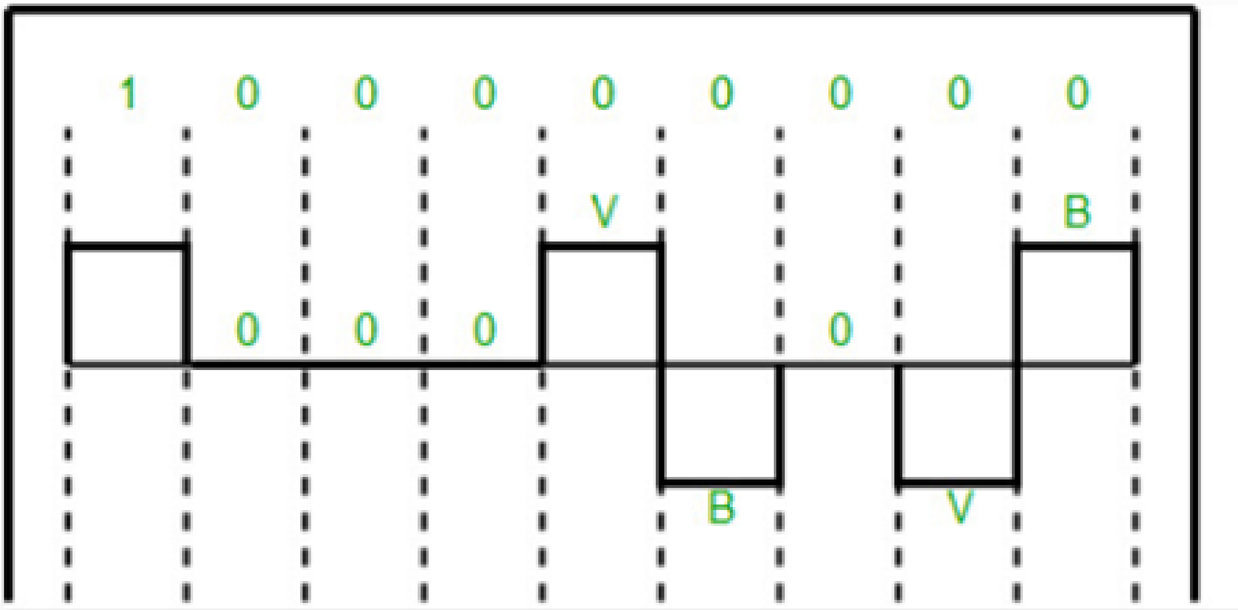
Scrambling techniques

- B8ZS(Bipolar with 8-zero substitution)

  This technique is similar to Bipolar AMI except when eight consecutive zero-level voltages are encountered they are replaced by the sequence,"oooVBoVB".
- V(Violation), is a non-zero voltage which means signal have same polarity as the previous non-zero voltage. Thus it is violation of general AMI technique.
- B(Bipolar), also non-zero voltage level which is in accordance with the AMI rule (i.e., opposite polarity from the previous non-zero voltage).

B8ZS(Bipolar with 8-zero substitution)

Example: Data = 100000000

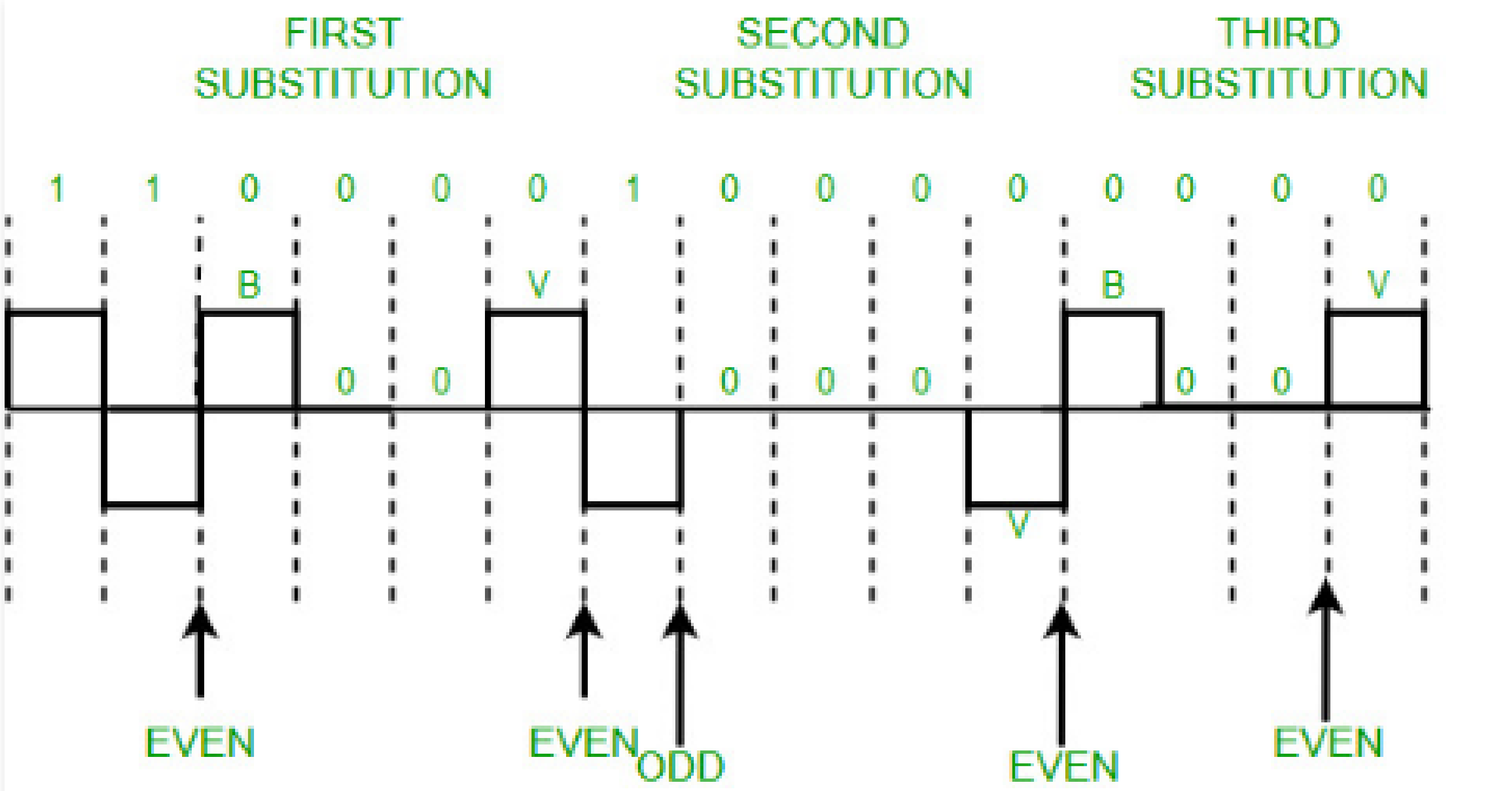


HDB3(High-density bipolar 3-zero)

- In this technique four consecutive zero-level voltages are replaced with a sequence “oooV” or “BooV”.
- Rules for using these sequences:
- If the number of nonzero pulses after the last substitution is odd, the substitution pattern will be “oooV”, this helps maintaining total number of nonzero pulses even.
- If the number of nonzero pulses after the last substitution is even, the substitution pattern will be “BooV”. Hence even number of nonzero pulses is maintained again.

HDB3(High-density bipolar 3-zero)

Example: Data = 110000100000000



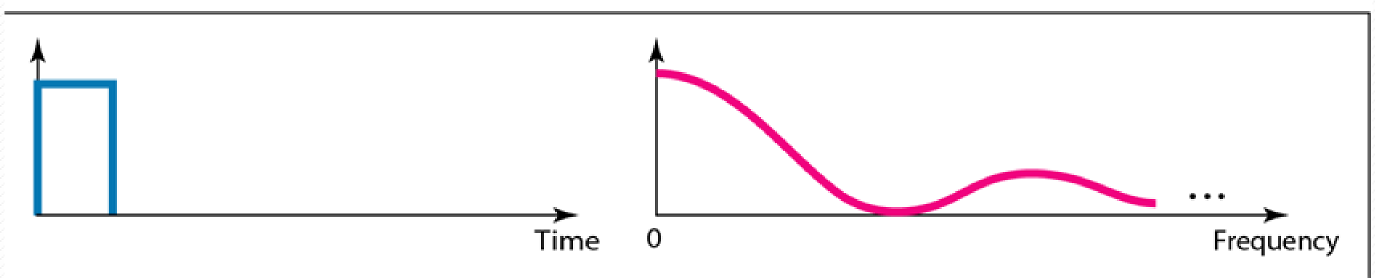
HDB3 scrambling technique:

- **Explanation** – After representing first two 1's of data we encounter four consecutive zeros. Since our last substitutions were two 1's(thus number of non-zero pulses is even).
- So, we substitute four zeros with "BooV".

Digital Data Transmission

Pulse Transmission over Band Limited Channel

- A digital signal is a composite analog signal with an infinite bandwidth.

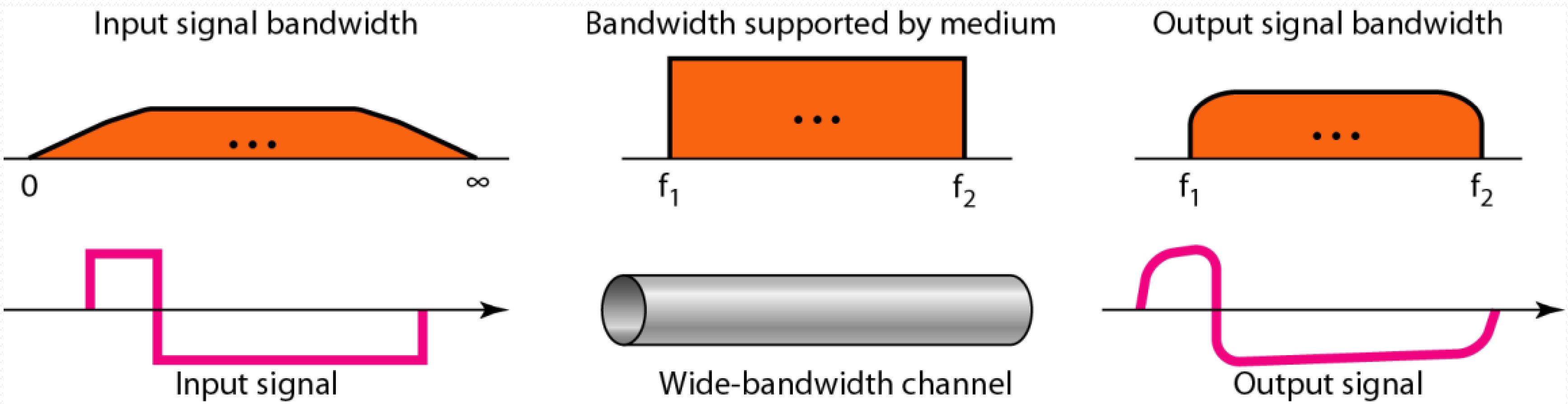


b. Time and frequency domains of nonperiodic digital signal

- A signal **can not be time-limited and band-limited** simultaneously.

Pulse Transmission over Band Limited Channel

- Baseband transmission of a digital signal that preserves the shape of the digital signal is possible only if we have a low-pass channel with an infinite or very wide bandwidth.



Pulse Transmission over Band Limited Channel

- In practice, **communications channels have a limited bandwidth, and hence transmitted pulses tend to deviate from the assumed rectangular shape and be spread during transmission.**

- Spreading of a pulse beyond its interval cause it to interfere with neighboring pulses. This is known as **Inter Symbol Interference (ISI)** which cause error in correct detection of the pulses.

Inter Symbol Interfernce

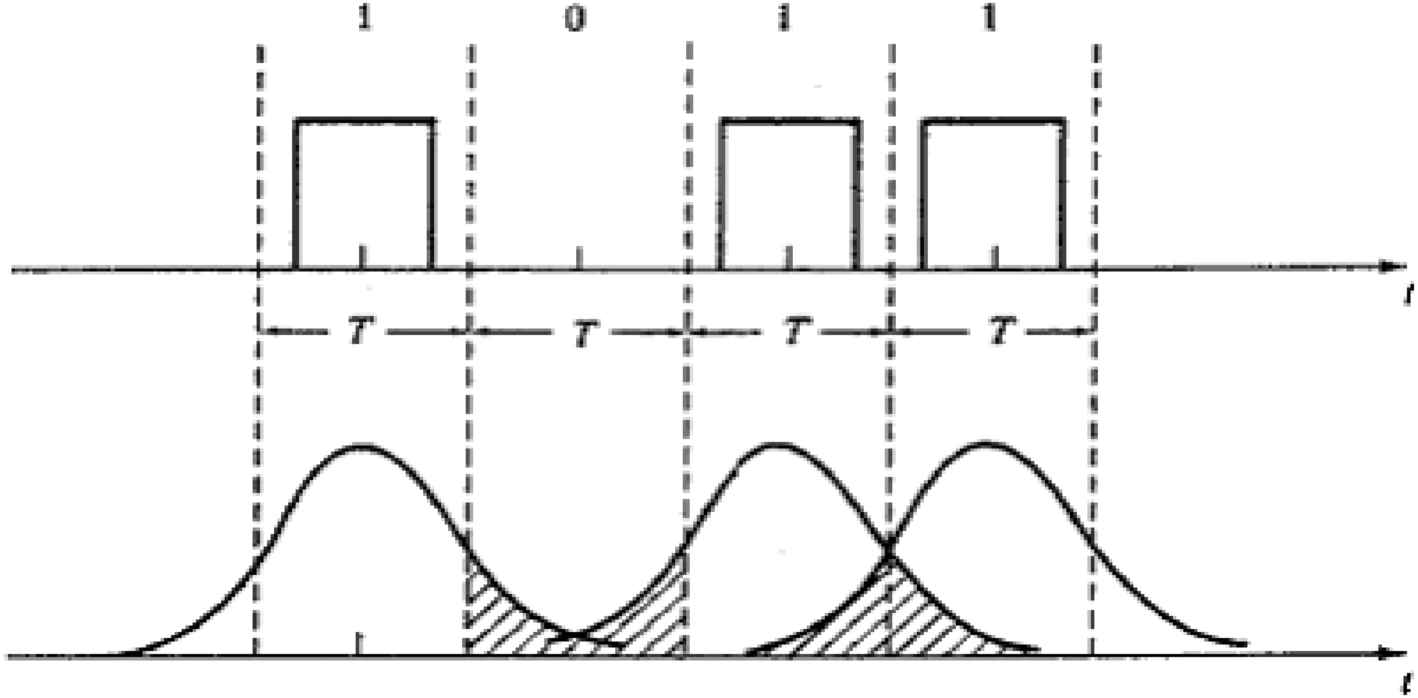
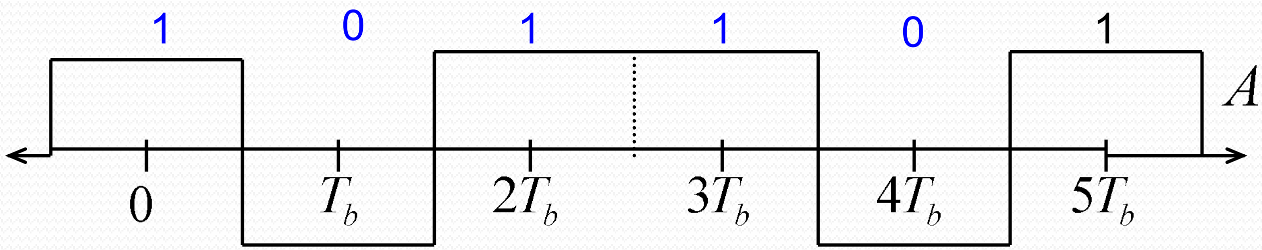


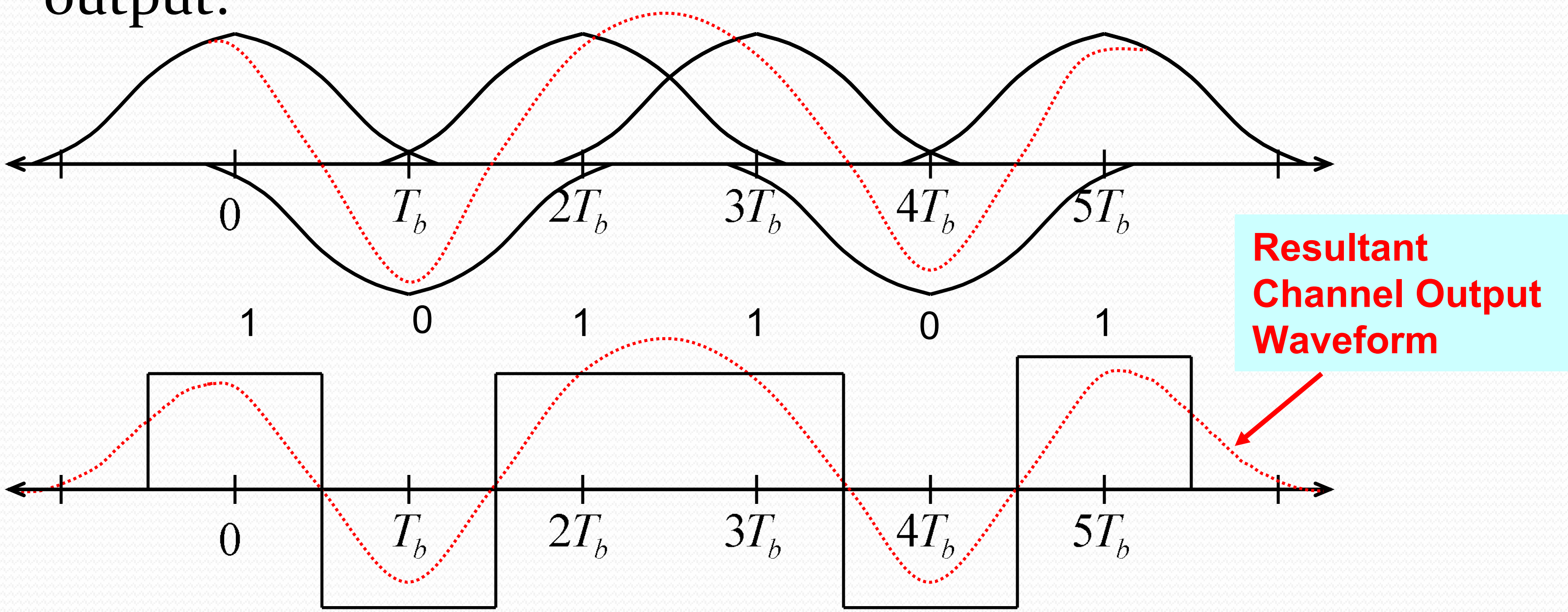
Fig. 5-12 Intersymbol interference in digital transmission

Intersymbol Interference

- For the input data stream:



- The channel output is the superposition of each bit's output:



Pulse Shaping

- Nyquist proposed that a zero–ISI pulse p(t) must satisfy the condition

$$p(t) = \begin{cases} 1, & t = 0 \\ 0, & t = \pm T_b, \pm 2T_b, \ldots \end{cases}$$

sinc pulse

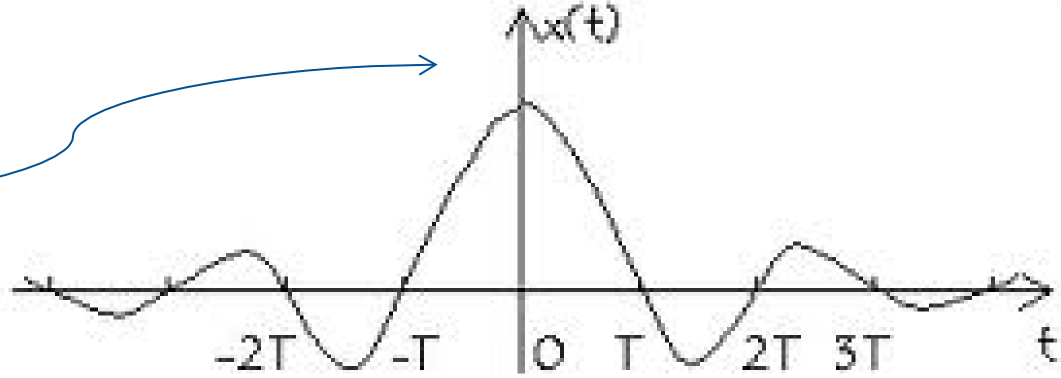
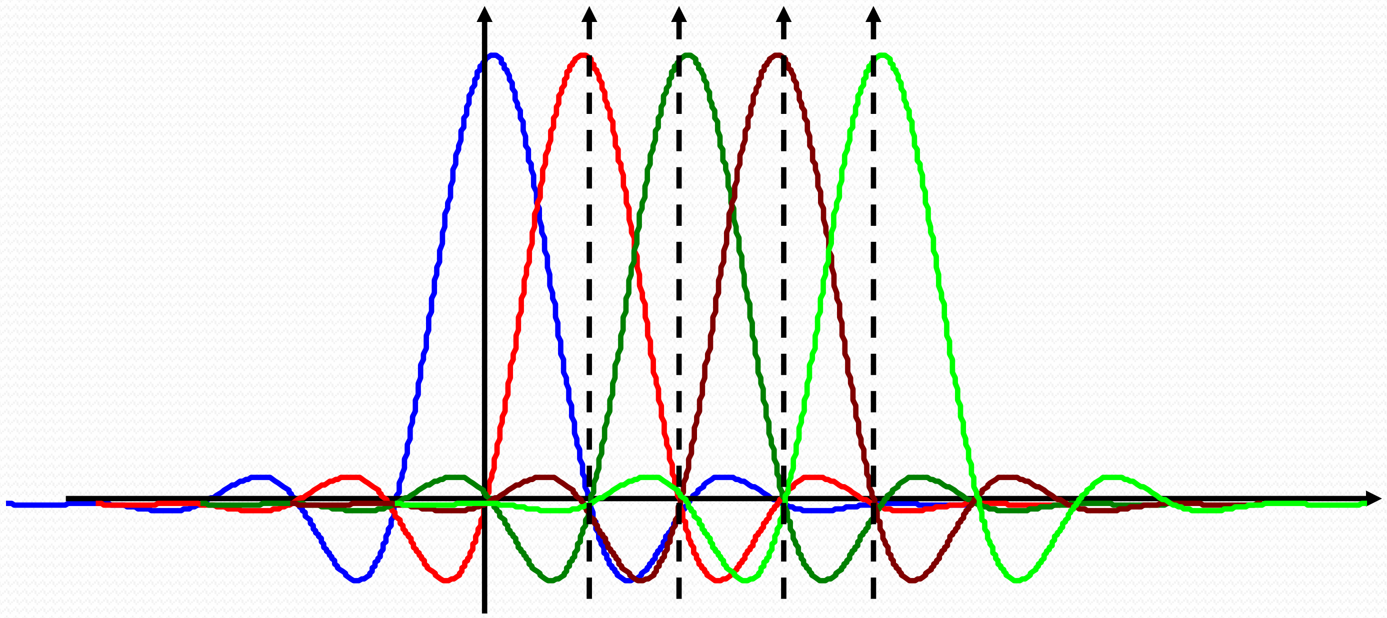


Figure 2.11 – Response (pulse) corresponding to no intersymbol interference

Nyquist's First Method for Zero ISI

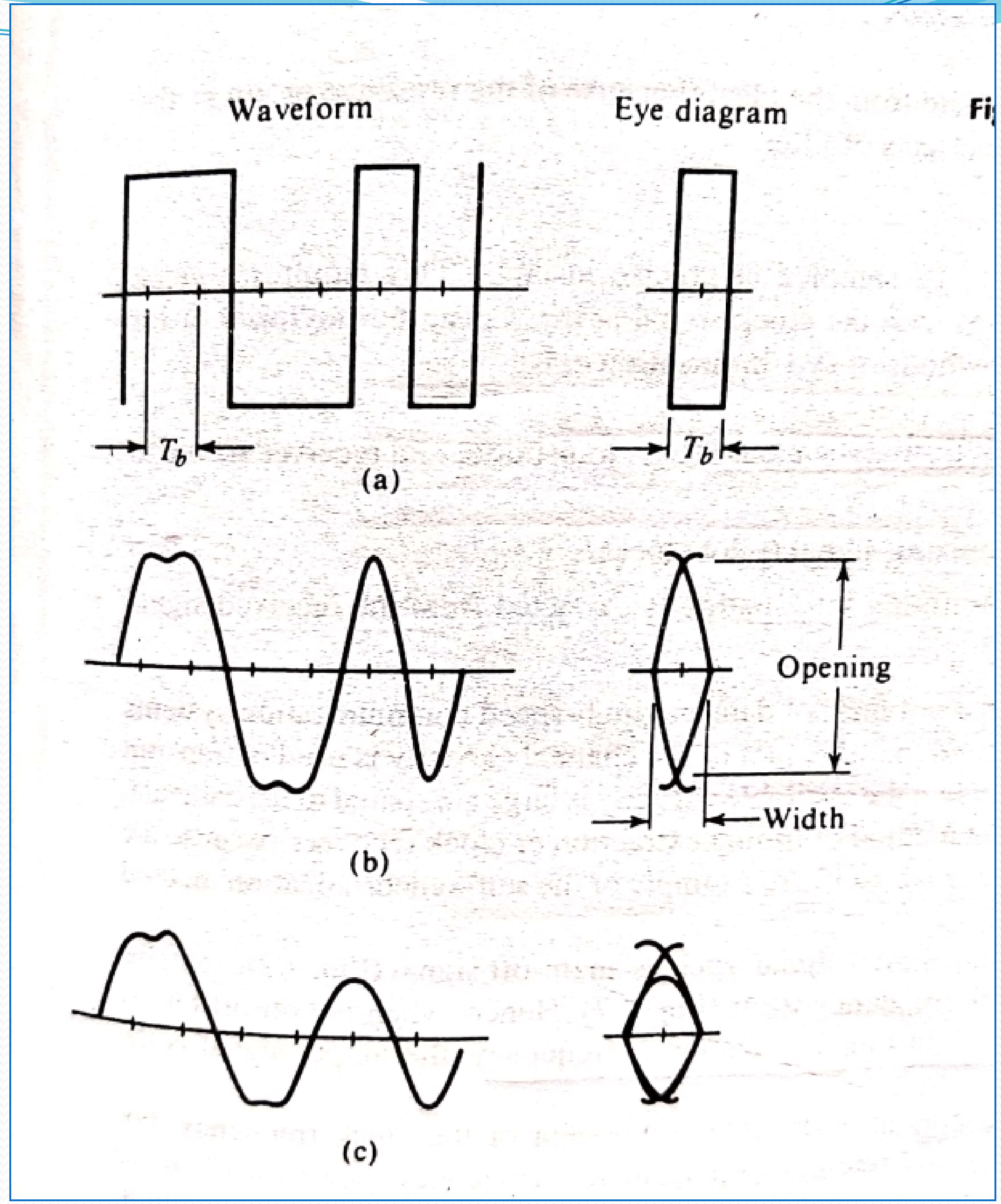


❑ISI occurs but, NO ISI is present at the sampling instants.

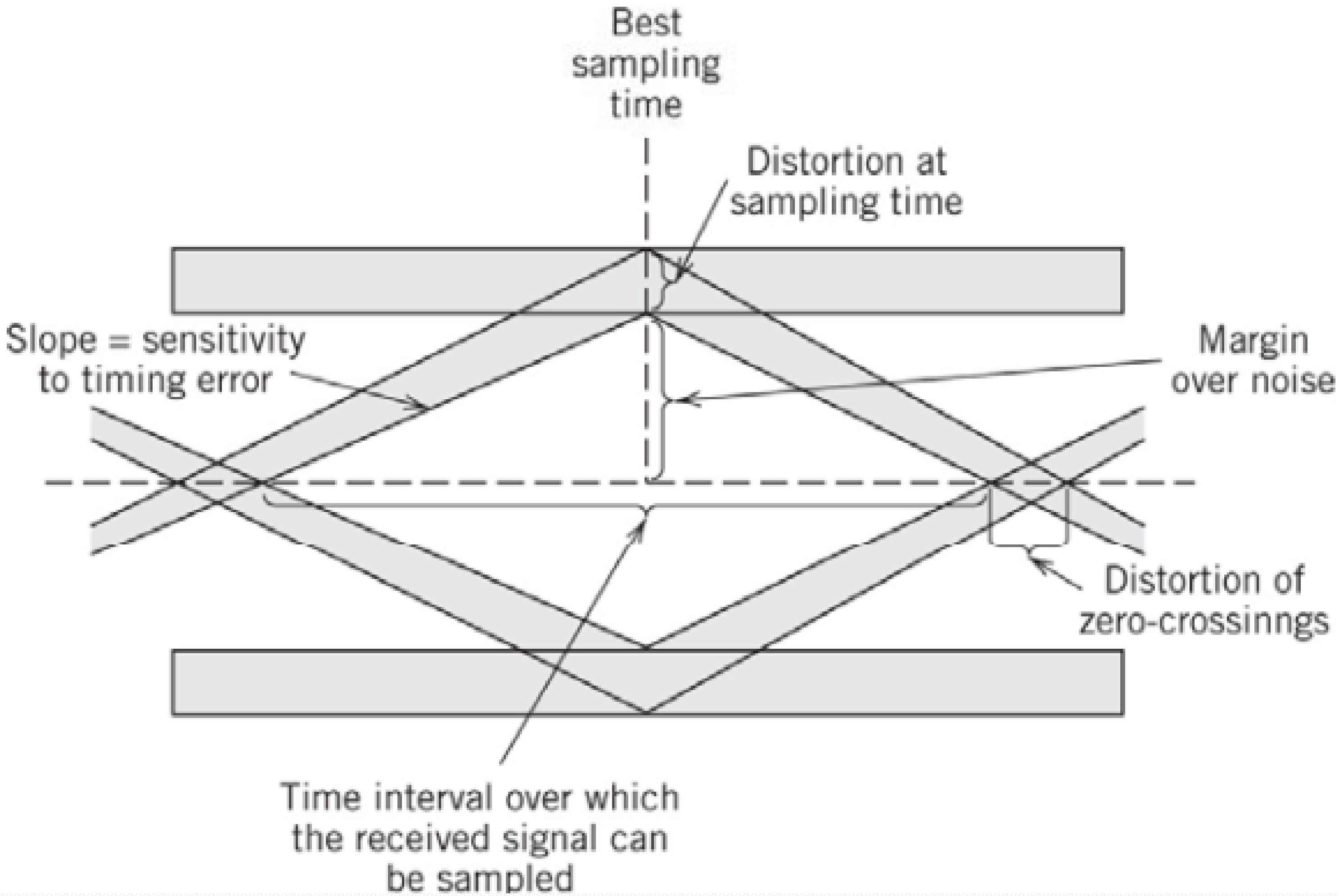
Eye Pattern

- The ISI & other signal degradation can be studied conveniently on oscilloscope through **Eye pattern.**
- A random binary pulse sequence is sent over the channel. The channel output is applied to the input of an oscilloscope.
- **The time base of the scope is kept same as the interval of one pulse (T_b).**
- **The oscilloscope shows the superposition of several traces which is the input signal cut up every Tb and then superimposed.**
- The eye pattern looks like a human eye and hence the name eye diagram.

- Consider the transmission of binary signal by polar rectangular pulses.
- If the channel bandwidth is infinite the eye pattern will be as shown in **fig. a**
- If channel is not distortion less, received pulses will ne rounded and spread out. If ISI is eliminated at the sampling instants, the eye pattern will be as shown in **fig. b.**
- If ISI is not zero at sampling instants, pulses values will deviate from thei full sacle values and which causes a blur and closing eye partially at midpoint**. Fig. c**



EYE PATTERN



Best sampling time
Distortion at sampling time
Slope = sensitivity to timing error
Margin over noise
Distortion of zero-crossinngs
Time interval over which the received signal can be sampled

Eye Pattern

- The **width of the eye opening** defines the time interval over which the received signal can be sampled without error from intersymbol interference. The preferred time for sampling is the instant of time at which the eye is open the widest.
- The sensitivity of the system to timing errors is determined by the **rate of closure of the eye** as the sampling time is varied.
- The **height of the eye opening** defines the noise margin of the system.
- When the effect of ISI is severe, traces from the upper portion of the eye pattern cross traces from the lower portion and the eye is completely closed.
- In this situation, it is impossible to avoid errors due to the presence of intersymbol interference in the system.