* Digital Signal :

Digital data take discrete value. eg. data are stored in comp. memory in the form of 0s and 1s. They can be converted to a digital signal or modulated into an analog signal for transmission accross a medium.

Digital Signal can have only a limited no. of defined values. Although each value can be any no, it is often as simple as 1 and 0. Show the signals is by plotting them on a pair of ⊥ axes. ~~the~~
The vertical axis represents the value or strength of a signal. The horizontal axis represents time.

1) Bit interval : It is the time required to send one single bit.

2) Bit Rate :
↳ It refers to the no. of bits intervals in one sec.
↳ units are bps, kbps, Mbps, Gbps.

3) Bit length : It is the distance one bit occupies on the transmission medium.

1 kbps = 1000 bps

1 Mbps = 1000,000 bps

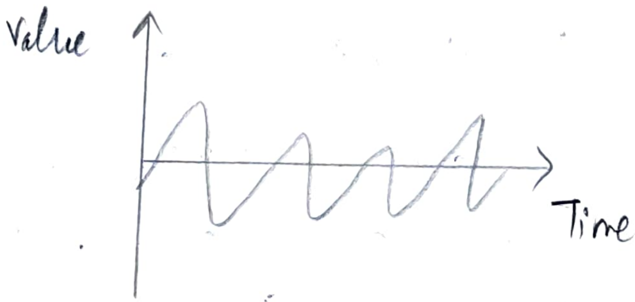
1 Gbps = 1000,000,000 bps

Bit length = Propagation speed × bit duration

* Analog Signals:

Analog data refers to info. i.e. continuous. eg. human voice.

Analog Signal has infinitely many levels of intensity over a period of time. As the wave moves from value A to value B, it passes through & includes an infinite no. of values along its path.

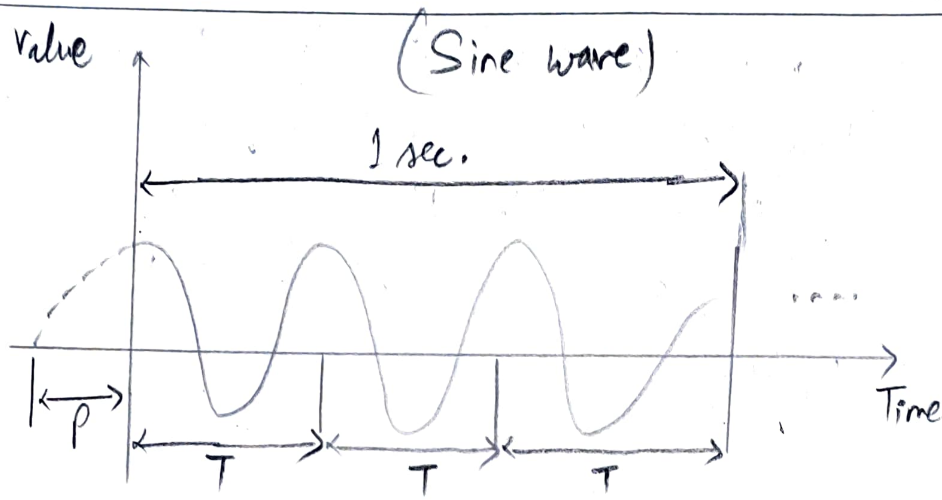


A periodic Analog signal completes a pattern within a measurable time frame, called a period. And repeats that pattern over subsequent identical periods. The completion of one full pattern is called a cycle.

A nonperiodic analog signal changes without exhibiting a pattern or cycle that repeats over time.

~~In~~ simple periodic analog ~~s~~signal, a sine wave cannot be decomposed into simpler signals.

A Composite periodic analog signal is composed of multiple sine waves.



T: Period $T = \frac{1}{3}$ sec.

f: Frequency $f = \frac{1}{T} = 3$ Hz.

P: Phase $P = \left(\frac{1}{4}\right)T = \frac{360'}{4} = 90$ degree.

A Sine wave can be represented by 3 parameters: The peak amplitude, the frequency and the phase.

Peak amplitude of a signal is the absolute value of its highest intensity, proportional to the energy it carries.
Eg. peak amplitude of a electric signal is measured in volts.

Period refers to the amount of time in sec., signal needs to complete 1 cycle.

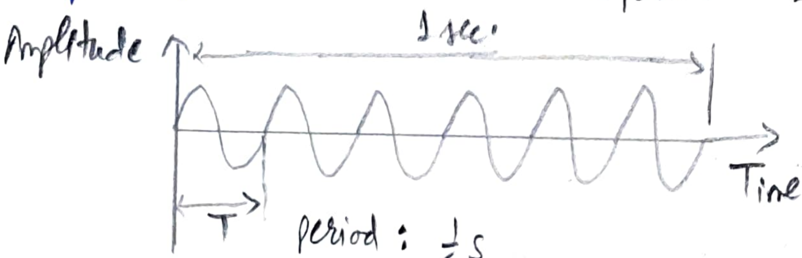
Frequency, measured in Hertz (Hz), refers to the no. of periods in 1 sec.

Period and frequency are inverse of each other ($f = 1/T$).

Phase describes the position of the waveform relative to time 0. Phase is measured in degrees or radians ($360°$ is 2π radian).

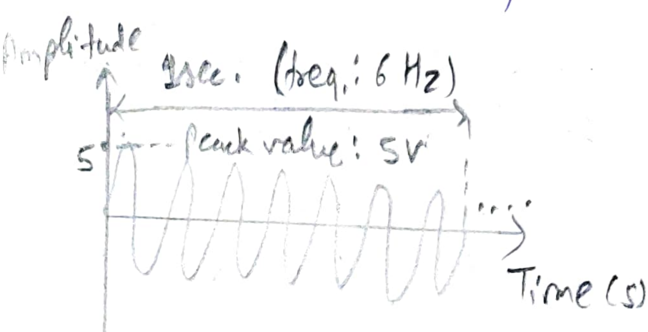
Wavelength means a signal traveling through a transmission medium. Wavelength is a distance. a simple signal can travel in one period. Wavelength binds the period or the frequency of a simple sine wave to the propagation speed of medium.

6 periods in 1 s → frequency is 6 Hz

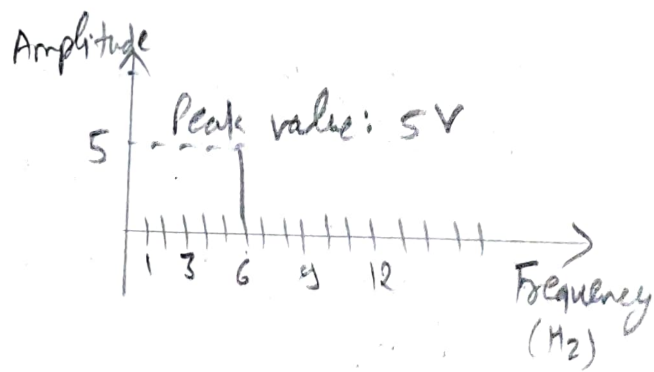


$$\text{Wavelength } (\lambda) = \frac{\text{Propagation speed } (c)}{\text{frequency } (f)}$$

$$\lambda = \frac{c}{f} = c \times T \qquad [\because f = \tfrac{1}{T}]$$

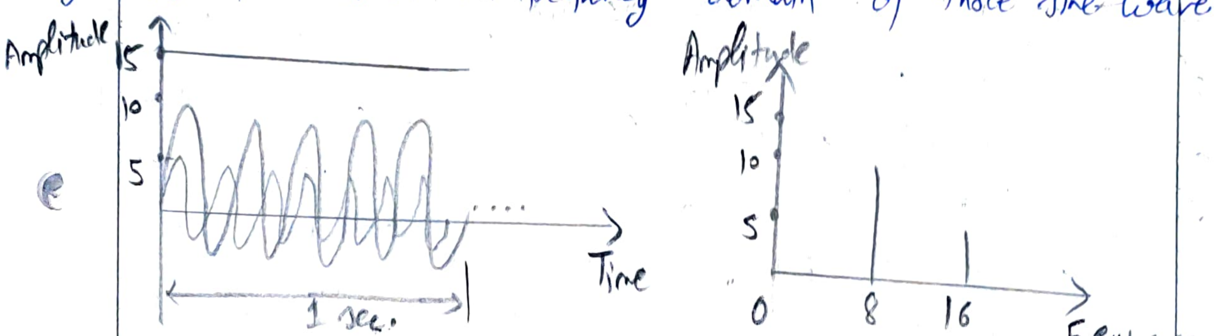


(A Sine wave in the time domain)



(Sine wave in frequency domain)

Eg. The time domain and frequency domain of three sine wave :



(Time-domain) (Frequency-domain)

Composite signal is made up of many simple sine waves with diff. frequencies, amplitudes and phase.

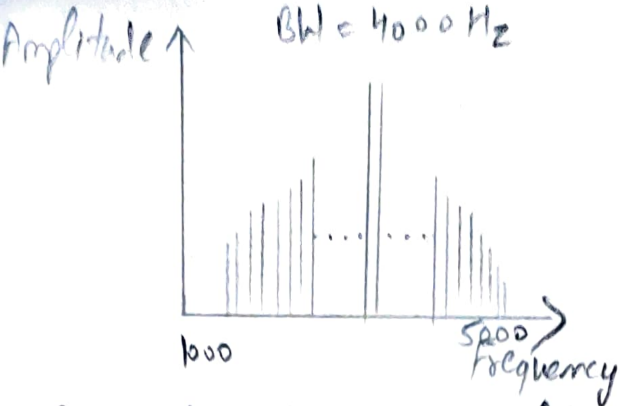
A periodic composite signal can be decomposed into a series of simple sine waves with discrete frequencies that are integral multiple of the fundamental frequency (1f, 2f, 3f and so on).

A Non-periodic composite signal can be decomposed into a combination of an infinite no. of simple waves with continuous frequencies, with real values.

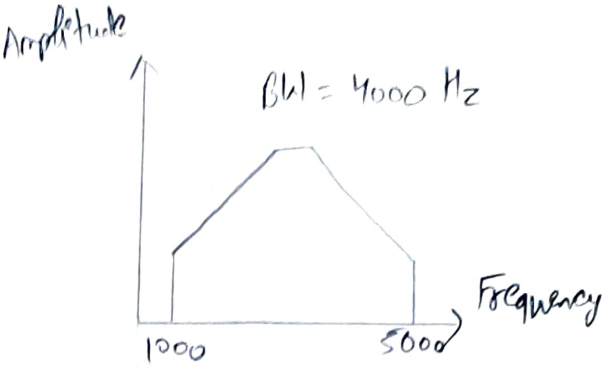
Bandwidth: range of frequency contained in a composite signal. It is a difference b/w two no.

Eg. Composite signal contains ~~BW~~ freq. b/w 1000 and 5000

So BW = 5000 − 1000 = 4000.



[BW of a Periodic Signal]

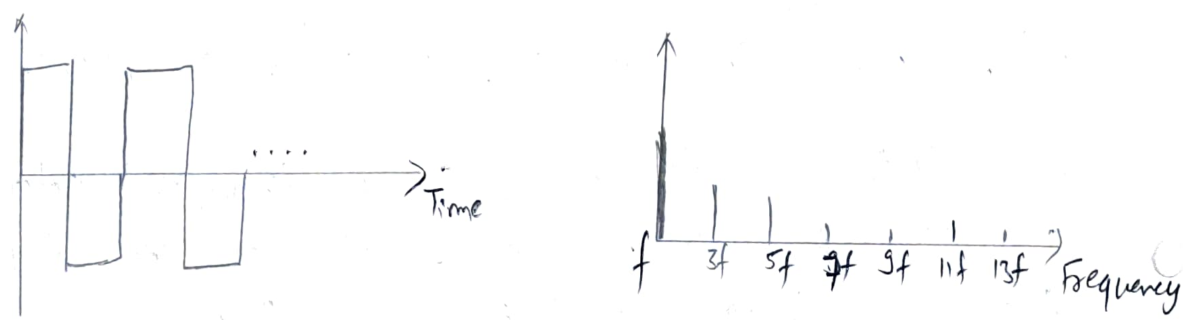


(BW of a Nonperiodic Signal)

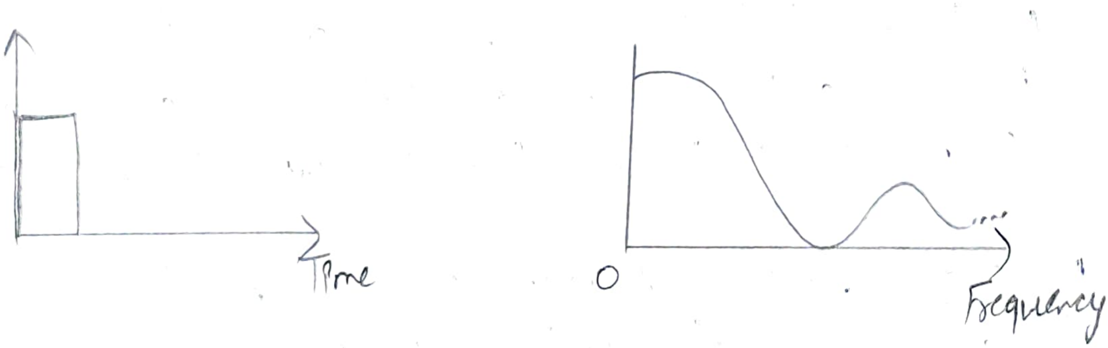
Eg. Assume we need to download text documents at the rate of 100 pages per min. What is the required bit rate of the channel? A page is an avg. of 24 lines with 80 char. in each line. If we assume that one char. requires 8 bits. Calculate Bit rate

$$\text{Bit rate} = 100 \times 24 \times 80 \times 8 = 1536000 \text{ bps}$$

$$= 1.536 \text{ Mbps.}$$



[Time & frequency domains of periodic digital Signal]



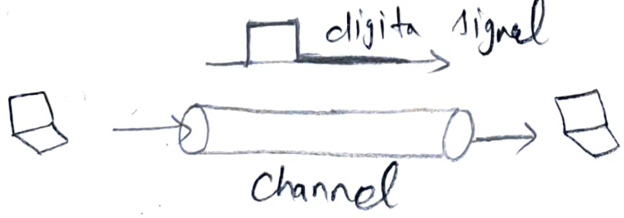
[Time & frequency domains of nonperiodic digital Signal]

Transmission of Digital Signals:

Baseband Transmission means sending a digital signal over a channel without changing the digital signal to an analog signal.

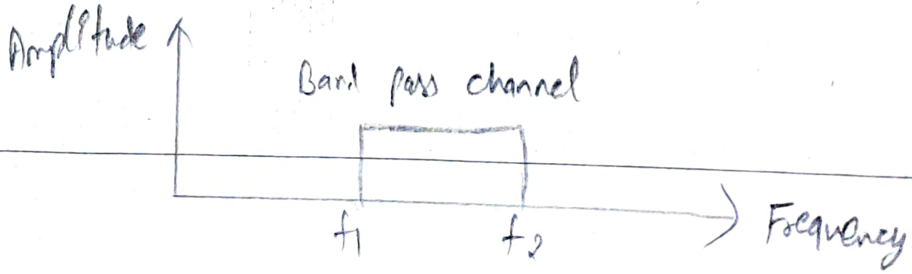
It requires a low pass channel, a channel with a BW that starts from zero.

Eg: Connect several comp. to a medium, but not allow more than two stations to commn at a time.



Broadband Transmission (Modulation): changing the digital signal to an analog signal for transmission.

It requires a band-pass channel, a channel with a BW does not start from zero. This type of channel more available as compared to low pass channel.

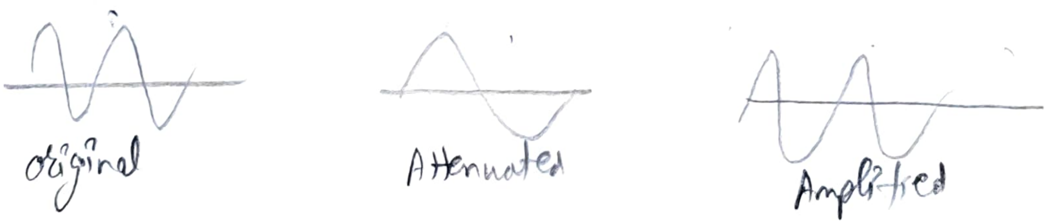


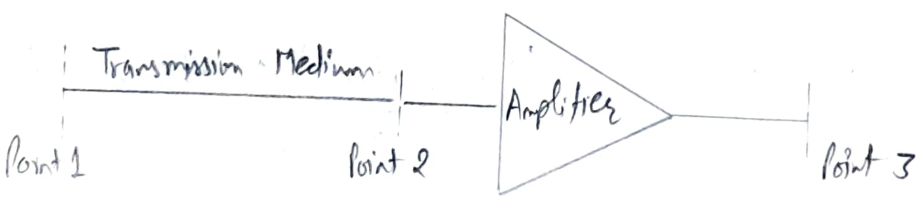
* Transmission Impairment:

Signals travel through transmission media, which are not perfect. The imperfection causes signal impairment. This means signal at the beginning of the medium is not the same as signal at the end of medium. What is sent is not what is received. These causes impairment are attenuation, distortion and noise.

Attenuation:

- → loss of energy.
- → The strength of signal ↓↓ with ↑↑ distance which causes loss of energy in overcoming resistance of medium. This is also known as attenuated signal.
- → Amplifiers are used to amplify the attenuated signal which gives the original signal back and compensate for this loss.





- → It measured in decibels (dB). It measures the relative strength of two signals or one signal at two diff. pts.

$$\boxed{dB = 10 \log_{10} (P_2 / P_1)}$$

variables P_1 & P_2 are Powers of a signal at pts 1 and 2 respectively.

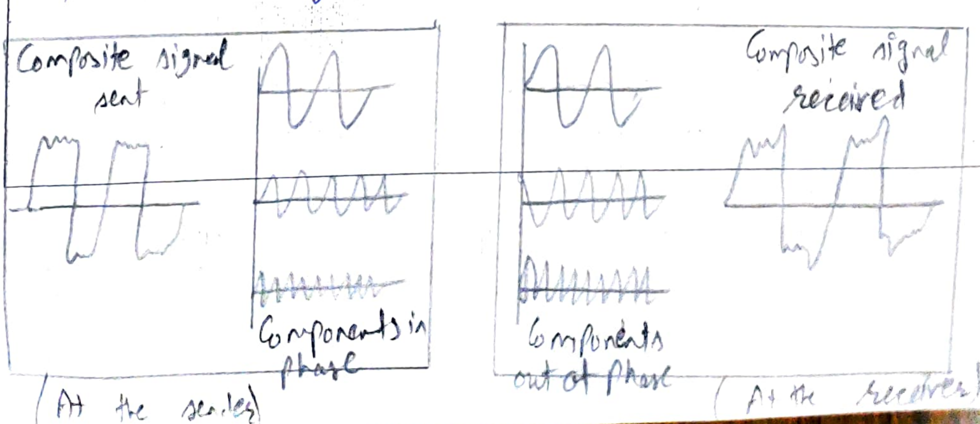
Eg. Suppose a signal travels through a transmission medium and its power is reduced to one half. This means that $P_2 = 0.5\,P_1$. Attenuation is

$$dB = 10 \log_{10} (P_2/P_1) = 10 \log_{10} (0.5P_1/P_1)$$

$$= 10 \log_{10} 0.5 = 10 \times (-0.3) = -3\ dB$$

Distortion:

- Changes in the form or shape of the signal.
- Occurs in composite signal made up of diff. frequencies.
- Each frequency component has its own propagation speed travelling through a medium, thats why it delay in arriving at the final destination.
- Every component arrive at diff. time which leads to distortion. Therefore, they have diff. phases at receiver and from what they had at senders end.



Noise :

→ The random or unwanted signal that mixes up with the original signal is called noise.

→ Noises are induced noise, Crosstalk noise, thermal noise and ~~thermal~~ ~~[illegible]~~ impulse noise, which may corrupt the signal.

→ Induced Noise comes from sources such as motors and appliances. These devices act as sending antenna and transmission medium act as receiving antenna.

→ Thermal Noise is movement of electrons in wire which create an extra signal.

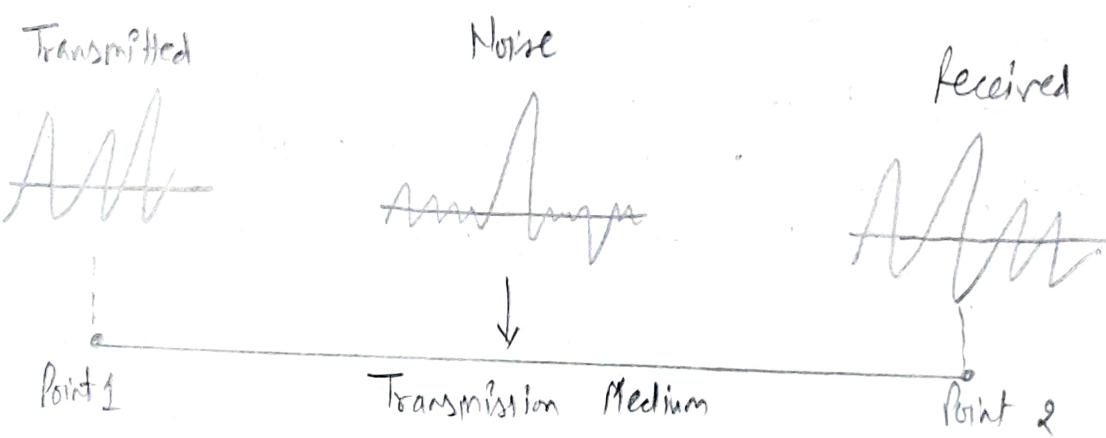
→ Crosstalk Noise is when one wire affects the other wire.

→ Impulse Noise is a signal with high energy that comes from lightning or power lines.

Signal-to-Noise Ratio (SNR) :

$$\boxed{SNR_{dB} = \frac{\text{avg. signal power}}{\text{avg. noise power}}}$$

these may changes with time

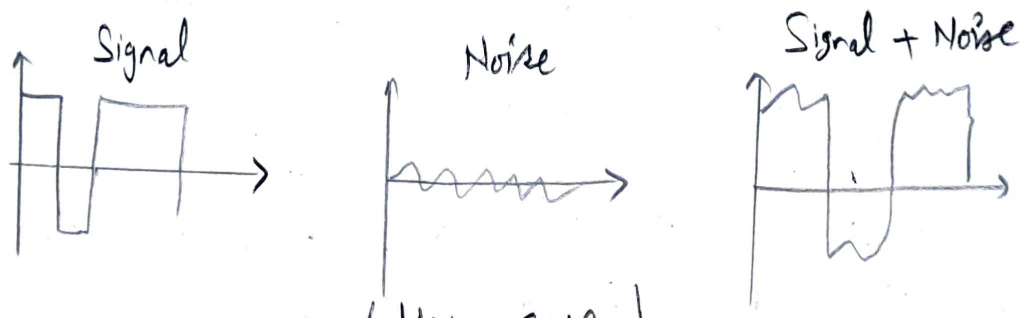


$$\boxed{SNR_{dB} = 10 \log_{10} SNR}$$

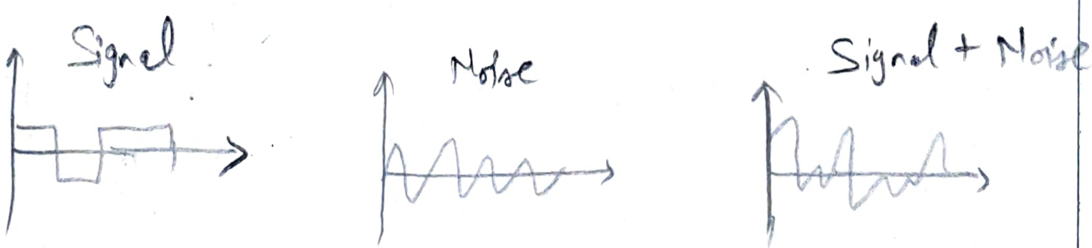
Two Cases of SNR :

High SNR means signal is less corrupted by noise.

Low SNR means signal is more corrupted by noise.



(High SNR)



(Low SNR)

* ~~Data Bit~~ Rate Limits

* **Data Rate Limits:** In data commⁿ is how fast we can send data in bits per sec., over a channel.
Based on 3 factors:
1) Bandwidth available
2) levels of signals
3) quality of channels

Noiseless channel : Nyquist Bit Rate : defines max. Bit rate.

$$\boxed{\text{Bit Rate} = 2 \times B \times \log_2 L}$$

B : bandwidth of channel.
L : no. of signal levels used to represent data
Bit Rate : no. of bits per sec.

eg. We need to send 265 kbps over a noiseless (ideal) channel with a bandwidth of 20 kHz. How many signal levels do we need? By using Nyquist formula:

$$265000 \text{ bps} = 2 \times 20{,}000 \times \log_2 L$$

$$\Rightarrow \log_2 L = \frac{265000}{40000} = 6.625$$

$$\Rightarrow L = 2^{6.625} = 98.7 \text{ levels.}$$

$$\boxed{\begin{array}{c} x = \log_2 n \\ \Downarrow \\ n = 2^x \end{array}}$$

Noisy channel : Shannon Capacity : Channel is always noisy.

$$\boxed{C = B \times \log_2 (1 + SNR)}$$

B: Bandwidth , SNR = Signal-to-Noise Ratio
C: Capacity of channel in bits per sec.

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Eg. Consider an extremely noisy channel in which the value of the signal-to-noise ratio is almost zero. In other words, noise is so strong that signal is faint. The capacity C is:

$$C = B \log_2 (1 + SNR)$$

$$= B \log_2 (1 + 0) \qquad [\because \log_2 1 = 0]$$

$$= B \log_2 1 = B \times 0 = 0$$

This means capacity of channel is zero regardless of the Bandwidth. And data is so corrupted when received.

Eg. Calculate the theoretical highest bit rate of a regular telephone line. A telephone line normally has a BW of 3000 Hz (300 to 3300 Hz) assigned for data comm^n. The SNR = 3162. Calculate capacity:

$$C = 3000 \log_2 (1 + 3162)$$

$$= 34,881 \text{ bps}$$

This means highest bit rate for a telephone line is 34.881 kbps. If we want to send data faster than this, we can either ... the BW of the line or improve the SNR ratio.

Q. have a channel with 1-MHz BW. The SNR for this channel is 63. What are the appropriate bit rate & signal level?

Ans Use Shannon formula to find upper limit

$$C = B \log_2 (1+SNR)$$

$$[1 \text{ MHz} = 10^6 \text{ Hz}]$$

$$= 1 \log_2 (1+63)$$

$$= 1 \log_2 64$$

$$[\log_2 64 = \log_2 2^6 = 6]$$

$$= 1(6)$$

$$= 6 \text{ Mbps}$$

The Shannon formula gives us 6 Mbps, the upper limit. For better performance we choose sth 4 Mbps.. Use Nyquist formula to find no. of signal levels.

$$\text{BitRate} = 2 \times B \times \log_2 L$$

$$\Rightarrow 4 = 2 \times 1 \text{ MHz} \times \log_2 L$$

$$\Rightarrow \log_2 L = 2$$

$$\Rightarrow L = 2^2 = 4$$

&

* Performance Measures:

1) Bandwidth: the term can be used in two diff. contexts with two diff. measuring values: Bandwidth in Hertz and bandwidth in bits per sec.

Bandwidth in Hz is the range of frequencies contained in a composite signal or the range of frequencies a channel can pass.
Eg. BW of a subscriber telephone line is 4 KHz.

Bandwidth in bits per sec. refers a channel, a link or even a n/w can transmit.
Eg. BW of a fast Ethernet n/w is 100 Mbps. It means this n/w can send 100 Mbps.

Relationship : ↑↑ in BW in Hz means ↑↑ in BW in bits per sec. The relationship depends on whether we use baseband transmission or broadband transmission.

Eg. The BW of a subscriber line is 4 KHz for voice or data. The BW of this line for data transmission can be up to 56 kbps. If the telephone company improves the quality of the line and ↑↑ the BW to 8 KHz, we can send 112 kbps.

2) Throughput : It is a measure of how fast we can actually send data through a n/w.
A link may have a BW of B bps, but we can send only T bps ... link with T always less than B.

In other words, BW is a potential measurement of a link; the throughput is an actual measurement of how fast we can send data

Eg: Have a link with a BW of 1 Mbps, but the devices connected to the end of the link may handle only 200 kbps. This means that we cannot send more than 200 kbps through this link.

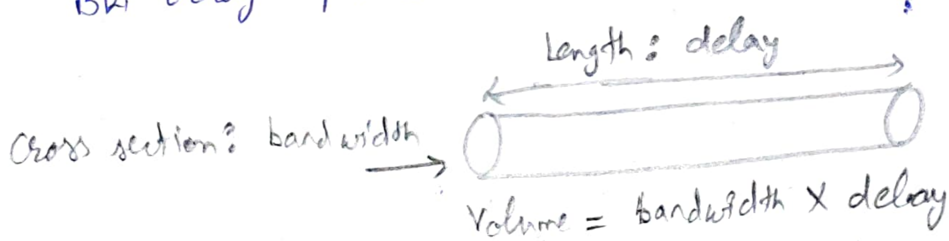
3) Latency: (Delay): How long it takes for an entire msg to completely arrive at the destination from the time the first bit is sent out from the source.

Four types of delay: propagation delay, transmission delay, queuing delay, processing delay.

$$\boxed{\text{Latency} = \text{Propagation delay} + \text{transmission delay} + \text{queuing delay} + \text{processing delay}}$$

NoTe: The BW delay product defines the no. of bits that can fill the link.

Eg. Link b/w two pts as a pipe. The cross section of the pipe represents the BW, and the length of the pipe represents the delay. The volume of the pipe defines the BW delay product.



4) Jitter: Jitter is a prob. if diff. pkt. of data encounter diff. delays and the app. using the data at the receiver site is time-sensitive.
eg. audio & ~~vedi~~ video data

* **Digital Transmission:**

A Comp. n/w is designed to send info from one pt. to another. This info. needs to be converted to either a digital signal or an analog signal for transmission.

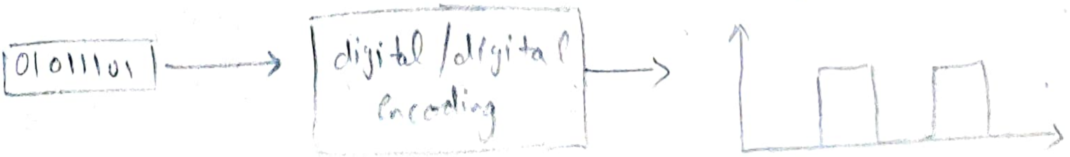
If data is digital, we need to use 'digital-to-digital' conversion technique, in which convert digital data to digital signals.

If data is analog, we use 'analog-to-digital' conversion technique, in which change an analog signal to digital signal.

A) **Digital-to-Digital Conversion:**

data can be either digital or analog. Signals that represent data can be digital or analog.

3 techniques: Line Coding, block Coding and scrambling.



measurement of a link;

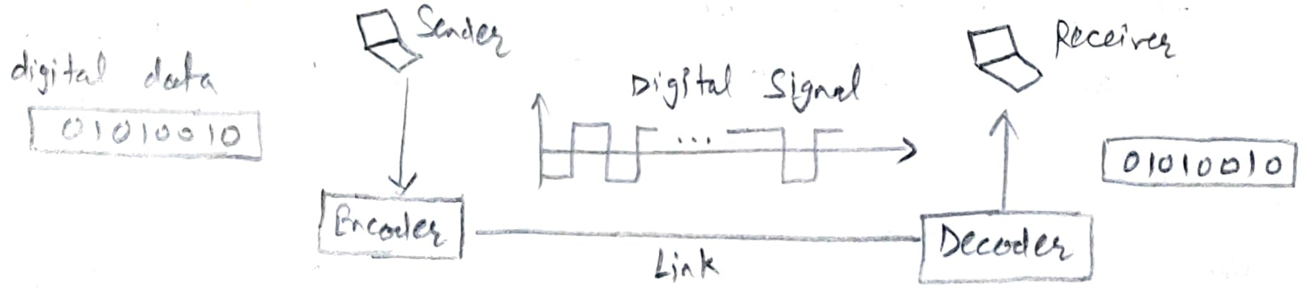
Digital-to-Digital Conversion

- Line Coding
  - Unipolar
  - Polar
    - NRZ
      - NRZ-L
      - NRZ-I
    - RZ
    - Biphase
      - Manchester
      - Differential Manchester
  - Bipolar
    - AMI
    - B8ZS
    - HDB3
- Block Coding
  - 4B/5B Coding
  - 8B/10B Coding
- Scrambling
  - B8ZS Coding
  - HDB3 Coding

1) Line Coding:

It is the process of converting digital data to digital signals. Data in the form of text, numbers, graphical images, audio or video are stored in comp. memory as sequence of bits. Line coding converts a seq. of bits to a digital signal.

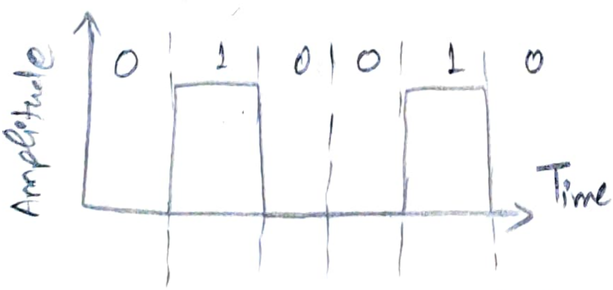
At the sender, digital data are encoded into a digital signal; At the receiver, the digital data are recreated by decoding the digital signal.



1.1) Unipolar:

- Digital transmission system sends the voltage pulses over the medium link such as wire or cable.
- In encoding, one voltage represent 0, and another voltage 1.

- The polarity of each pulse determines whether it is +ve or -ve.
- Unipolar encoding uses only one polarity
- polarity is assigned to the 1 binary state.
- 1 represented as a +ve value and 0 represent as a zero value.
- 1 is considered as a high voltage, and 0 as a zero voltage.
- Unipolar is simpler and inexpensive to implement.



Unipolar has two prob. that make this scheme less desirable:

→ DC Component

→ Synchronization

1.2) Polar:

- two voltage leve: +ve and -ve.
- By using two voltage levels, an avg. voltage level is reduced.

1.2.1) NRZ :

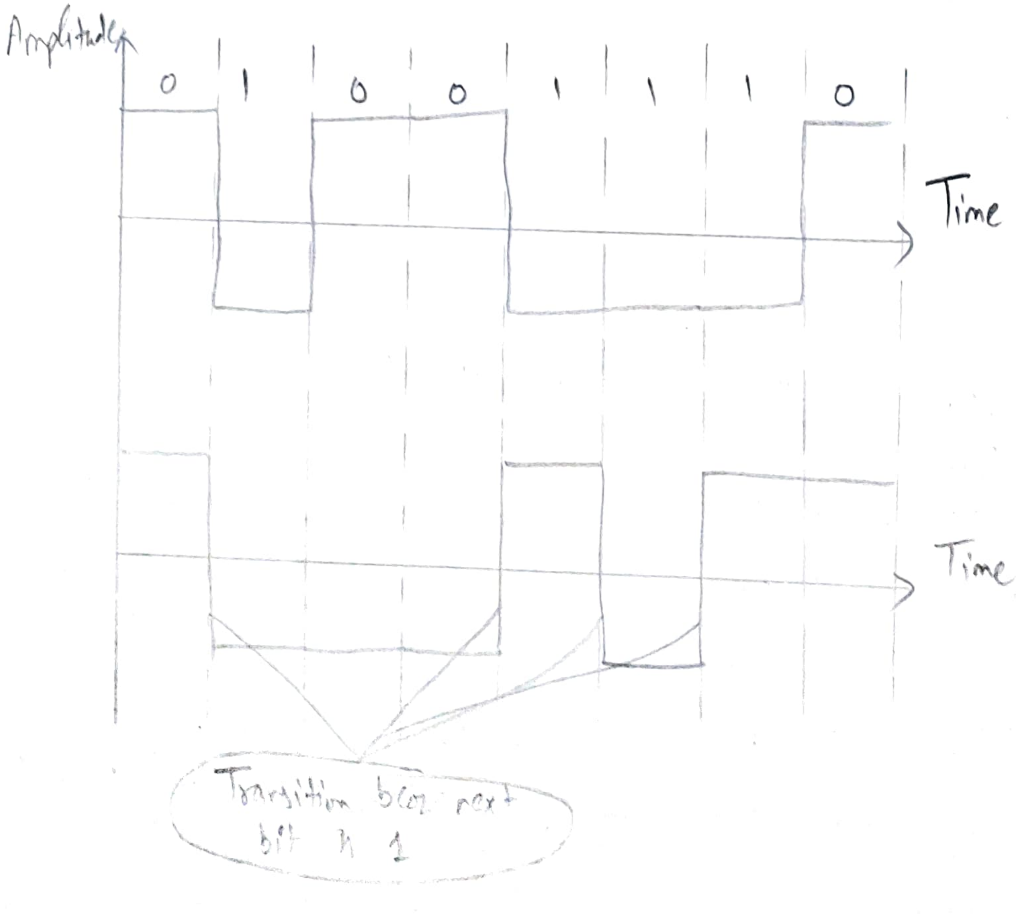
- Stands for Non-Return Zero.
- level of signal can be represented either +ve or -ve.

1.2.1.1) NRZ-L :

- the level of signal depends on the type of bit that it represent.
- If a bit is 0 or 1, then their voltage will be +ve and -ve respectively.
- Therefore, we can say that level of signal is dependent on the status of the bit.

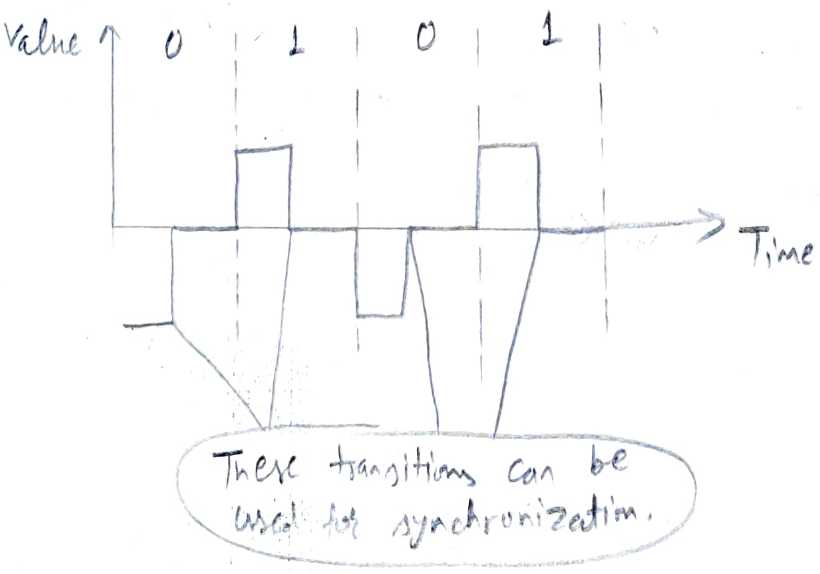
1.2.1.2) NRZ-I : It is an inversion of the voltage level that represent 1 bit.

- A transition occurs b/w +ve and -ve voltage that represent 1 bit.
- 0 bit represent no change or 1 bit represent a change in voltage level.



1.2.2) RZ:

- Return to Zero.
- There must be a signal change for each bit to achieve synchronization. However, to change with every bit, we need to have three values: +ve, -ve and zero.
- Represent +ve by 1, -ve by 0, zero by none.
- halfway through each interval, signal return to zero.
- 1 bit is represented by +ve - to 0 and 0 bit represented by -ve - to - 0.



Disadvantage of RZ:

two signal changes to encode one bit that acquires more BW.

1.2.3) Biphase:

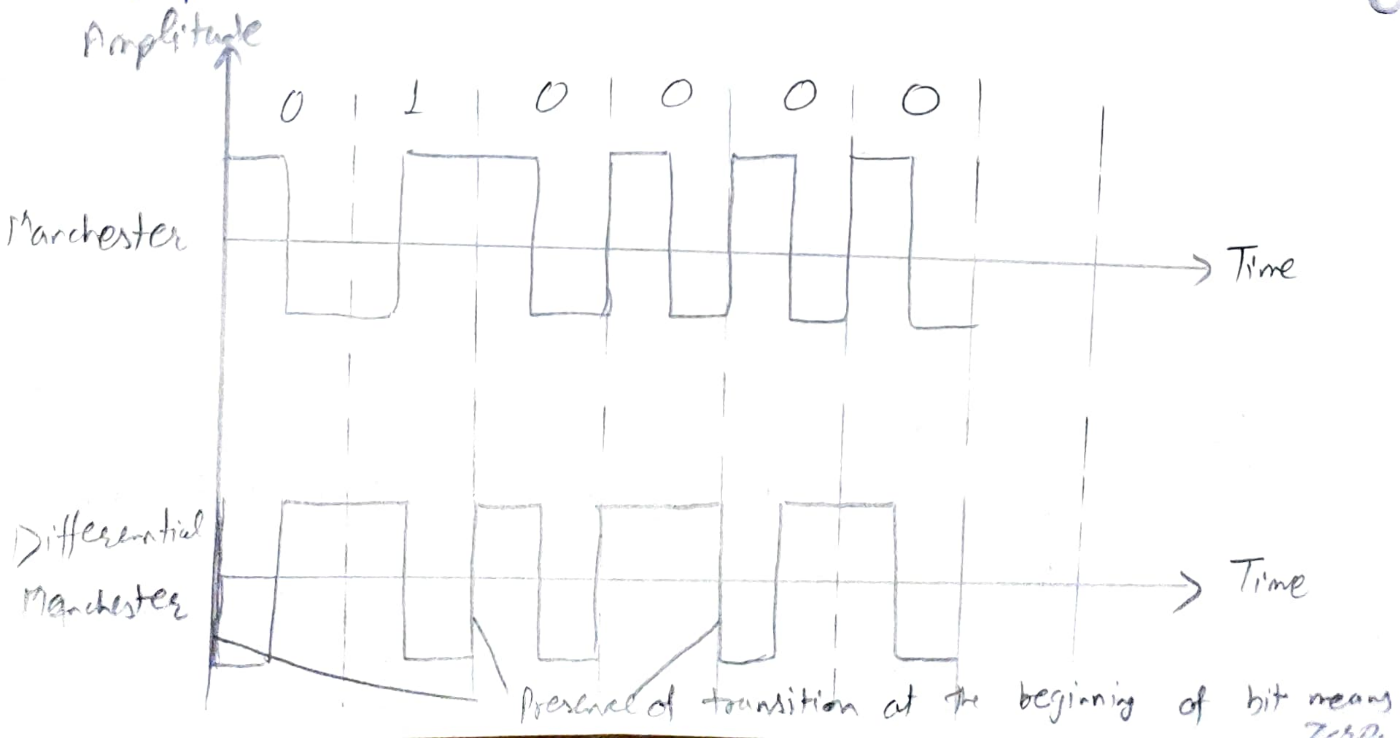
It is an encoding scheme in which signal changes at the middle of the bit interval but does not return to zero.

1.2.3.1) Manchester:

- It changes the signal at the middle of the bit interval but does not return to zero for synchronizatn.
- a -ve - to +ve represent 1 and +ve - to -ve represent 0.
- Manchaster has same level of synchronization as RZ scheme except that It has two levels of Amplitude

1.2.3.2) Differential Manchester:

- It changes the signals at the middle of the bit interval for synchronization, but the presence or absence of the transition at the beginning of the interval determines the bit.
- A transition means binary 0 and no transition means 1.
- two signal changes represent 0 and one signal change represent 1.



1.3) Bipolar:

- represent 3 voltage levels: +ve, -ve and zero.
- zero represent binary 0 and binary 1 represented by alternative +ve & -ve voltages. This alteration can also occur even when the 1 bits are not consecutive.

1.3.1) AMI:

- stands for 'Alternate Mark Inversion' where mark work comes from telegraphy which means 1. So, it can be redefined as alternate 1 inversion.
- 0 bit is represented by zero level and 1 bit is represented by alternative +ve and -ve voltages.

Advantage:

→ seq of 1 bits are synchronized.

Disadvantages:

→ does not ensure the synchronization of a long string of 0's bits.

1.3.2) B8ZS:

- Stands for Bipolar 8-zero substitution.
- This technique is adopted in North America to provide synchronization of a long sequence of 0's bits.

In most of the cases, the functionality of B8ZS is similar to bipolar AMI, but the only difference is, it provides the synchronization when a long sequence of 0s bits occur.

- When eight 0 occurs, then B8ZS implements some changes in 0s string pattern based on the polarity of the previous 1 bit.
- If the polarity of previous 1 bit is +ve, the eight 0s will be encoded as zero, zero, zero, +ve, −ve, zero, −ve, +ve.

polarity of previous bit

+	0	0	0	0	0	0	0	0

will change to

+	0	0	0	+	−	0	−	+

Violation

violation

- If the polarity of previous 1 bit is −ve, then eight 0s will be encoded as zero, zero, zero, −ve, +ve, zero, +ve, −ve.

polarity of previous bit

−	0	0	0	0	0	0	0	0

will change to

−	0	0	0	−	+	0	+	−

violation

violation

1.3.3) HDB3:

- Stands for High-Density Bipolar 3.
- first adopted in Europe and Japan.
- provide the synchronization of a long sequence of 0s bits.
- the pattern of violation is based on the polarity of previous bit.
- When four 0s occur, HDB3 looks at the no. of 1s bits occurred since the last substitution.
- If the no. of 1s bits is odd, then violation is made on fourth consecutive of 0. If the polarity of the previous bit is +ve, then violation is +ve. if the polarity of previous bit is -ve, then violation is -ve.

polarity of previous bit

-	0	0	0	0

↓ will change to

-	0	0	0	-

+	0	0	0	0

↓ will change to

+	0	0	0	+

- If the no. of 1s bits is even, then the violation is made on the place of the first & fourth consecutive 0s. If the polarity of previous bit is +ve, then violation is -ve. And if polarity of previous bit is -ve, the violation is +ve.

... measurement of a link,

Polarity of previous bit

-	0	0	0	0

will change to

-	+	0	0	+

+	0	0	0	0

will change to

+	-	0	0	-

2) Block Coding : Line Coding is always needed block Coding and scrambling may or may not be needed.

→ Block Coding helps in error detection and re-transmission of the signal.

→ It is normally reffered to as mB/nB Coding as it replaces each m-bit data group with n-bit data group (n > m).

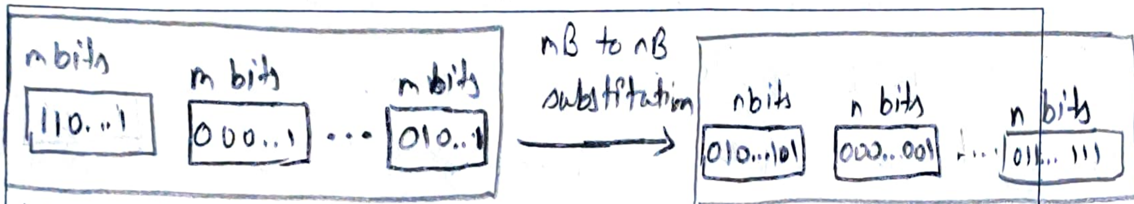
→ Thus, it adds extra bits (redundancy bit) which helps in synchronization at receiver's and sender's end and also providing some kind of error detecting capability.

→ It involves 3 steps: division, substitution & Combination.

→ In the division steps, a sequence of bits is divided into groups of m-bits.

→ In substitution step, we substitute an m-bit group for an n-bit group.

→ Finally, the n-bit groups are combined together to form a stream which has more bits than the original bits.

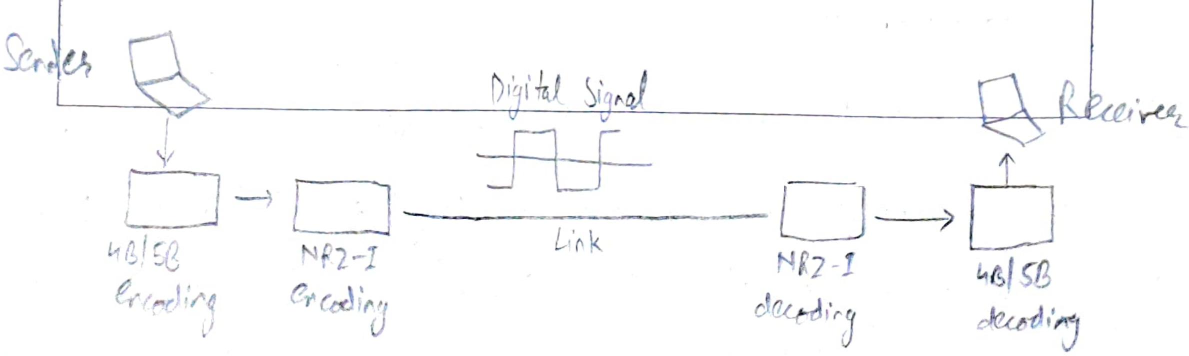


Division of a stream into m-bits group

Combining n-bit groups into a stream.

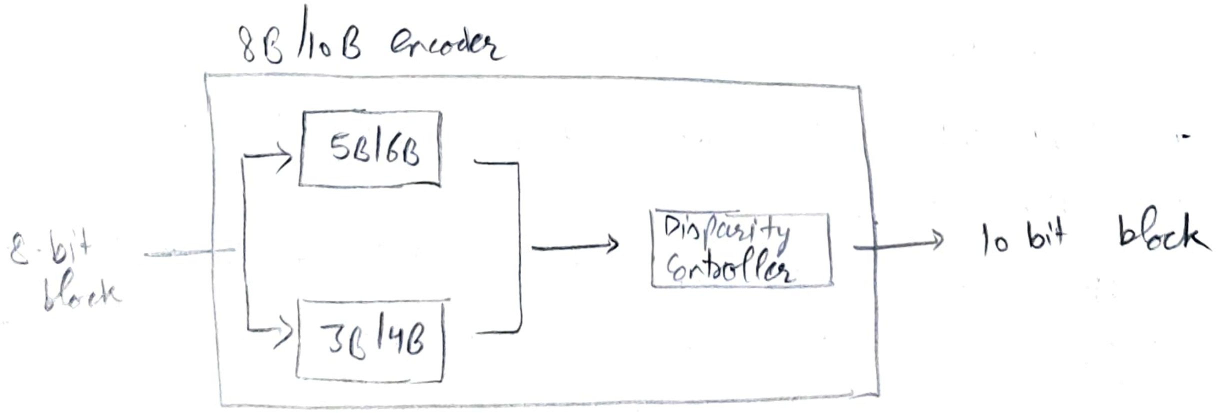
2.1) 4B/5B Coding: (Four Binary/Five Binary)

- This coding scheme is used in combination with NRZ-I.
- It has a synchronization prob. for long sequence of zeros.
- If we substitute the bit stream from 4-bit to 5-bit data group before encoding it with NRZ-I. So that it does not have a long stream of zeros.
- The block code scheme does not have more than three consecutive zeros.
- At the receiver the NRZ-I encoded digital signal is first decoded into a stream of bits and decoded ~~into~~ again to remove the redundancy bits.



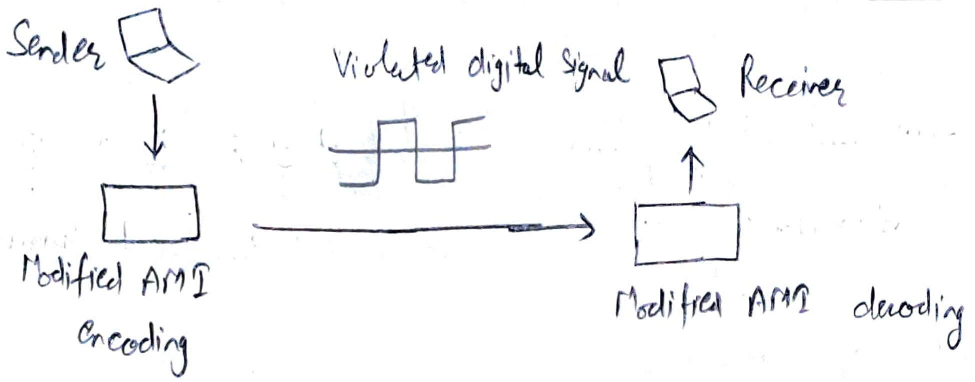
2.2) 8B/10B Coding: (Eight Binary / Ten Binary)

- It is similar to 4B/5B encoding except that a group of 8 bits of data is now substituted by a 10-bit code.
- It provides greater error detection capability than 4B/5B.
- Combinatn of 5B/6B and 3B/4B encoding.
- The most five significant bits of a 10-bit block is fed into the 5B/6B encoder, the least 3 significant bits is fed into a 3B/4B encoder.
- A group of 8 bits can have 2^8 diff combinations while a group of 10 bits can have 2^{10} diff. combination.
- This means that there are $2^{10} - 2^8 = 768$ redundant group that are not used for 8B/10B encoding and can be used for error detection & disparity check.
- Thus, this technique is better than 4B/5B because of better error checking capability and better synchronization.



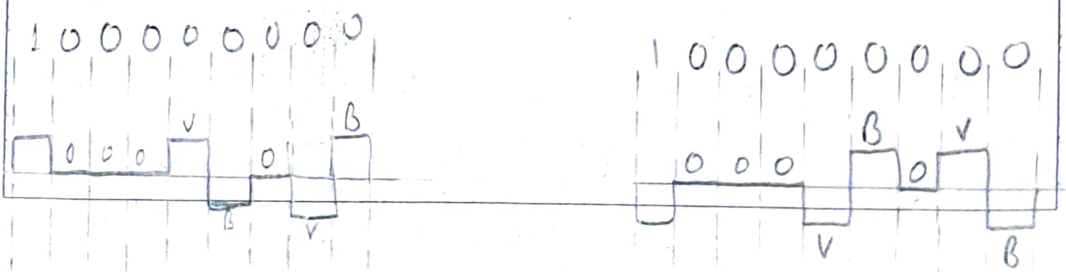
3) Scrambling:

- does not ↑↑ the no. of bits & ~~does~~ provide sync.
- Prob. with technique like Bipolar AMI (Alternate Mark Inversion) is that cont. sequence of zero's create sync. prob. one solⁿ to this is a scrambling.



3.1) B8ZS (Bipolar with 8-zero substitution) :-

- It is similar to Bipolar AMI except when eight consecutive zero-level voltages are encountered they are replaced by the seq. "000VB0VB".
- V (violation) is a non zero voltage which means signal have same polarity as the previous non-zero voltage.
- B (Bipolar), also a non-zero voltage which is in accordance with the AMI rule (i.e. opp. polarity from the previous non-zero voltage)



Previous level is +ve

previous level is -ve

...measurement of a link;

HDB3 Coding: (High density bipolar 3-zero

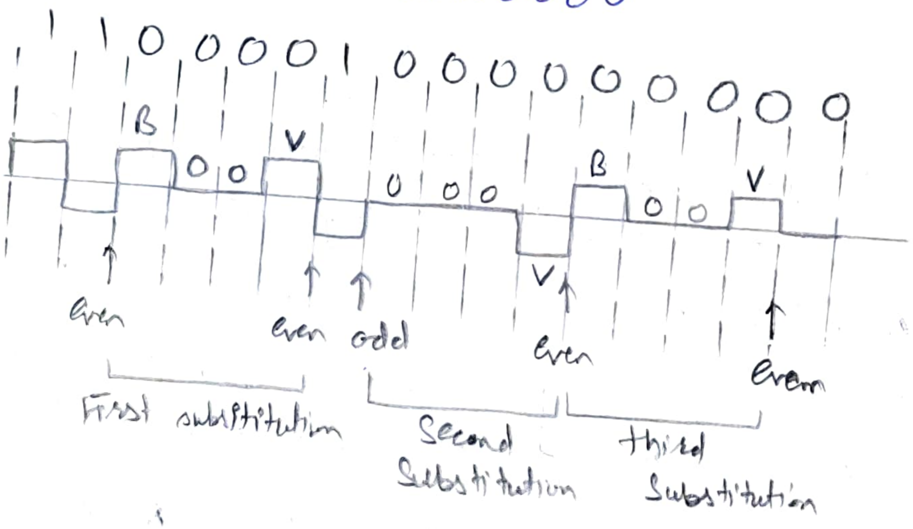
- four consecutive zero level voltages are replaced with a seq. "000V" or "B00V".

Rules for using these seq.:

→ if the no. of non zero pulses after the last substitution is odd, the substitution pattern will be "000V", this helps maintaining total no. of non-zero pulses even.

→ If the no. of non zero pulses after the last substitution is even, the substitution pattern will be "B00V". Hence, even no. of non zero pulses is maintained again.

Eg data: 1100001000000000



B) Analog to Digital Conversion:

- When an analog signal is digitalized, this is called an analog-to-digital conversion.
- Suppose human sends a voice in the form of an analog signal, we need to digitalize the analog signal which is less prone to noise. It requires a reduction in the no. of values in an analog msg. so that they can be represented in the digital stream.

this is a ...

- the info. contained in a continuous wave form is converted in digital pulse.

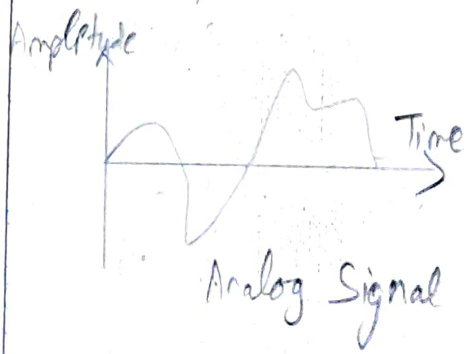
Techiniques:

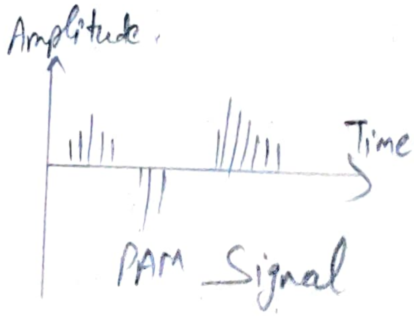
1) PAM:

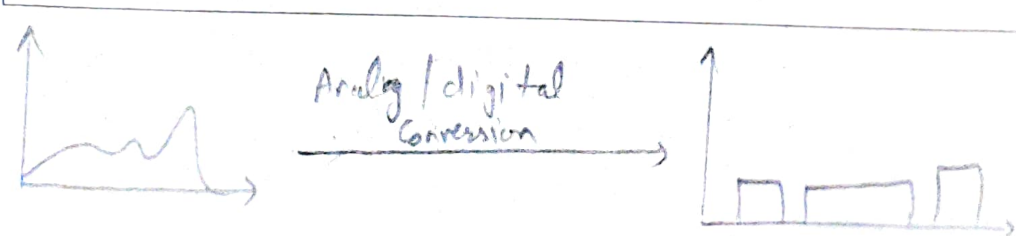
→ Pulse Amplitude Modulation.

→ It takes the analog signal, samples it, and generates a series of digital pulses based on the result of sampling where sampling means measuring the amplitude of a signal at equal intervals.

→ PAM is not useful in data comm^n as it translates the original wave form into pulses, but these pulses are not digital. To make them digital, PAM technique is modified to PCM technique.

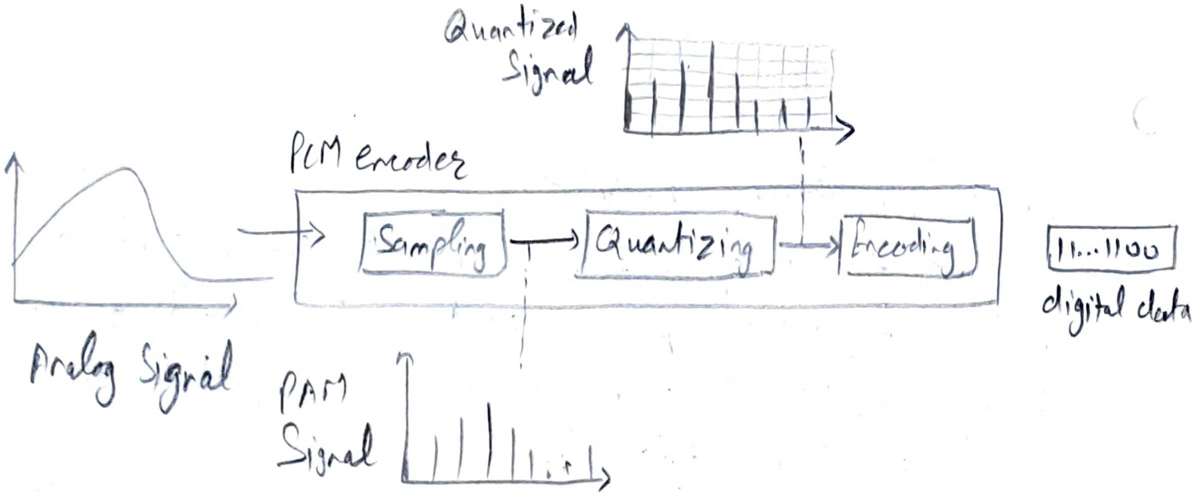






2) PCM:

- Pulse Code Modulation.
- It is used to modify the pulses created by PAM to form a digital signal. To achieve this, PCM quantizes PAM pulses.
- PCM is made of four seperate processes: PAM, quantization, binary encoding and digital-to-digital encoding.
- PCM encoder has three processes:
  - ↳ analog signal is sampled
  - ↳ the sampled signal is quantized
  - ↳ quantized values are encoded as stream of bits

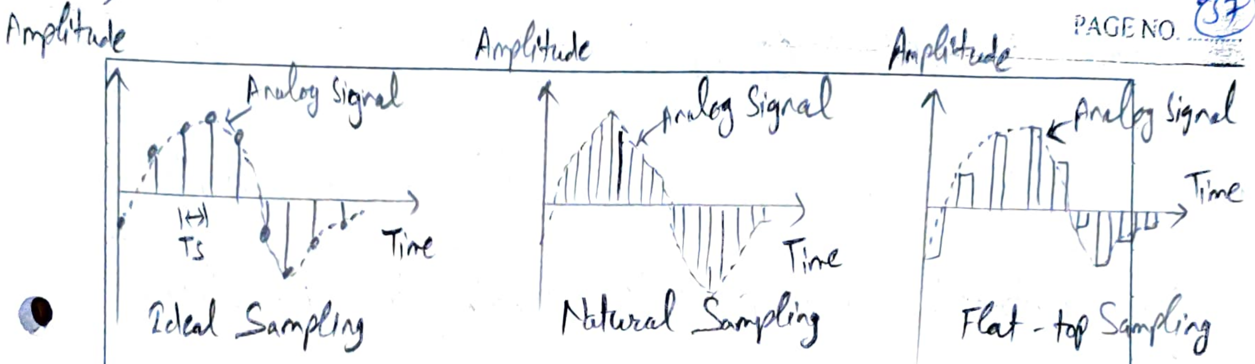


2.1) Sampling: Converting the continuous signal into a discrete signal.

i) Ideal Sampling: also known as Instantaneous sampling pulses from the analog signal are sampled. Cannot be easily impleted.

ii) Natural Sampling: It is a practical method of sampling in which pulse have finite width equal to T. The result is a seq of samples that retain the shape of analog signal.

iii) Flat-top Sampling: easily obtained. top of the samples remains constant by using a circuit.



Nyquise Rate:

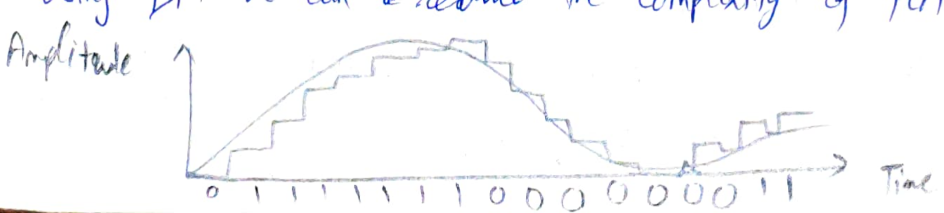
$$\boxed{\text{Nyquist Rate} = 2 \times f_{max}}$$

2.2) Quantization: The result of sampling is a series of pulses with amplitude values b/w the max. and min amplitudes of the signal. The set of amplitudes can be infinite with non ~~te~~ integral values b/w two limits.

2.3) Encoding: The digitization of the analog signal is done by the encoder. After each sample is quantized and the no. of bits per sample is decided, each sample can be changed to an n bit code.

Encoding also minimizes the BW used.

3 Delta Modulation: PCM is a very complex technique. By using DM we can ~~d~~ reduce the complexity of PCM.



✱ Analog Transmission:

→ In digital transmission we need low-pass channel. And in analog transmission we need only band-pass channel.

→ Converting data to bandpass analog signal is called digital-to-analog conversion.

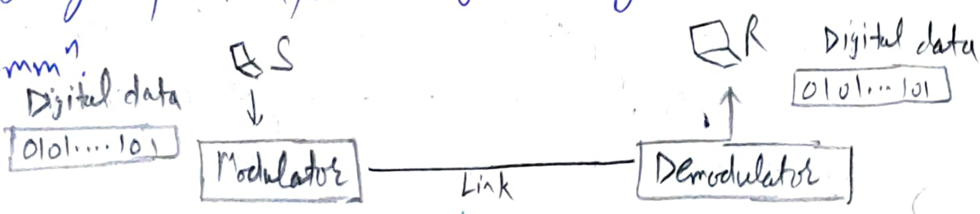
→ Converting a low-pass analog signal to a bandpass analog signal is called analog-to-analog conversion.

1) Digital-to-Analog Conversion / Digital Modulation:

→ This modulation uses discrete signals for modulating a carrier wave.

→ Digital Modulation removes comm^n noise as well as provides enhanced strength for the signal intrusion (break).

→ DM gives more capacity of data, high performance security and accessibility of a faster system by enormous quality comm^n.



Types:

i) ASK (Amplitude Shift keying):

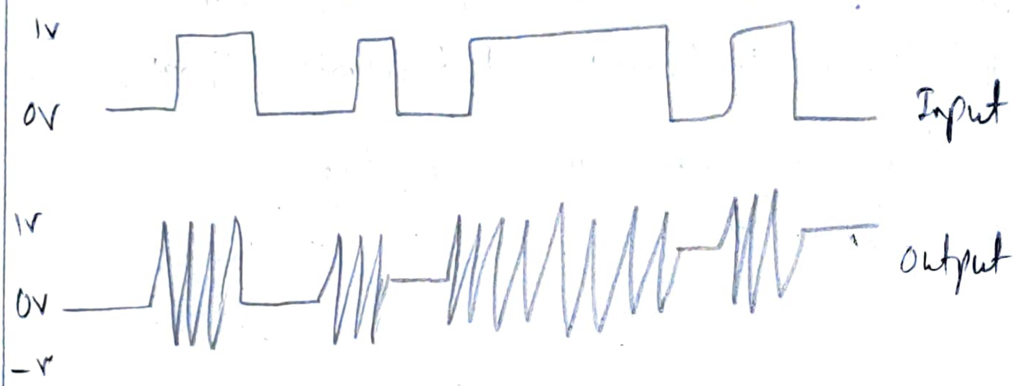
→ carrier signal is analog and data to be modulated is digital.

→ The data is an ON/OFF signal, and the O/P. is also a ON/OFF signal.

→ When data is 1, carrier is present and when data is 0, carrier is not present.

→ Therefore, this modulation scheme is called as OOK or On/Off keying (OOK) or ASK.

→ The app. of ASK includes IR remote controls and fiber optic transmitter & receiver.



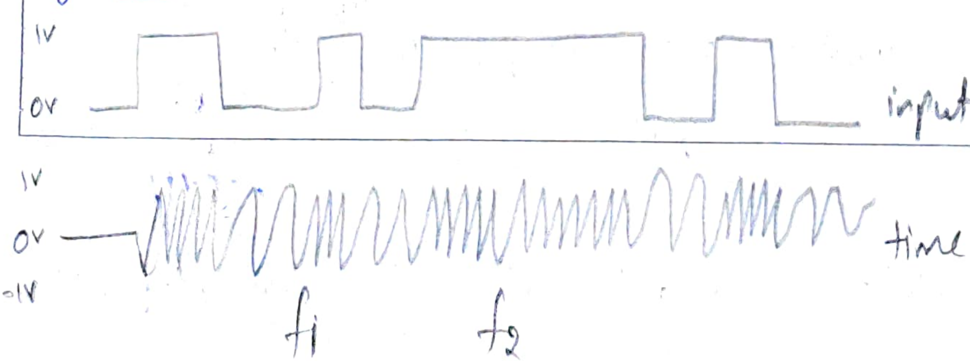
ii) FSK (Frequency Shift keying):

→ When the immediate freq. of the carrier signal is changed then the info. will be transmitted.

→ Carrier signal has two pre-defined freq. namely wc_1, & wc_2.

→ Whenever the data bit is '1' then the carrier signal by wc_1 is transmitted that is $\cos wc_1$.

→ Similarly, when the data bit is '0' then the carrier signal by wc_0 will be transmitted i.e. is $\cos wc_0$.



measurement of a link:

iii) PSK (Phase Shift keying):

→ The instant phase of carrier signal is moved for this modulation.

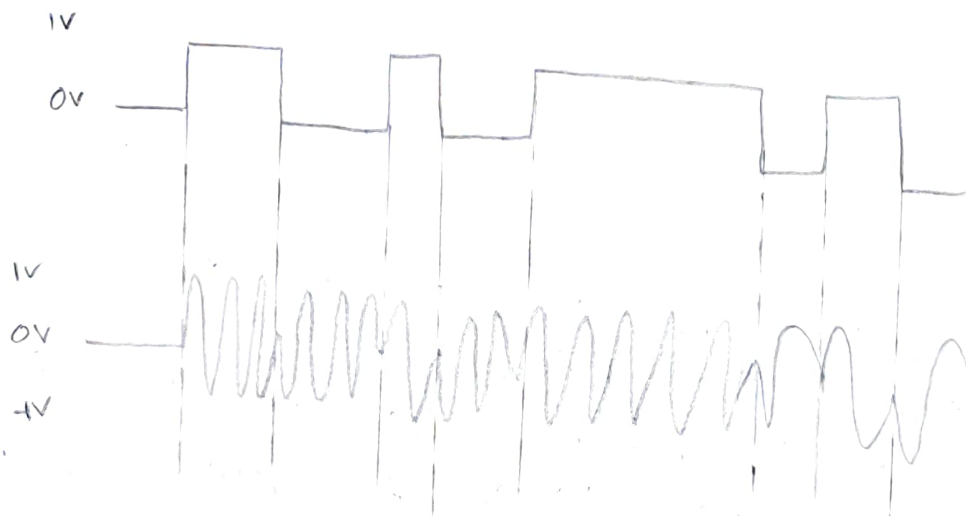
→ If the m(t) baseband signal is =1 then carrier signal within phase will be transmitted.

→ If the baseband signal m(t) = 0 then carrier signal by out of phase is transmitted i.e. $\cos(\omega_c t + \pi)$.

→ If phase shift can be done in four dissimilar quadrants then 2-bit of data will be transmitted at once.

→ This method is an individual case of phase shift keying modulation which is known as Quadrature PSK. (QPSK)

→ app. includes a broadband modem (ADSL), Satellite commⁿ, mobile phones etc.



2) Analog-to-Analog Conversion / Analog Modulation:

→ They modified to represent analog data.

→ AM is required when bandpass is used.

AM

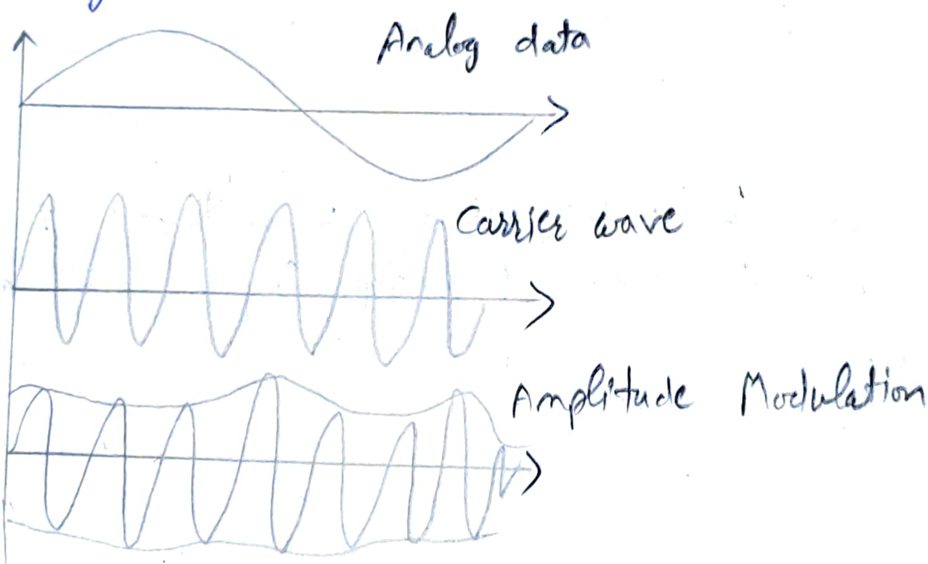
↓ Amplitude Modulatⁿ

↓ Freq. Modulatⁿ

↓ Phase Modulatⁿ

i) Amplitude Modulation:

→ Amplitude of the carrier signal is modified to reflect the analog data.



→ Amplitude modulation is implemented by multiplier.

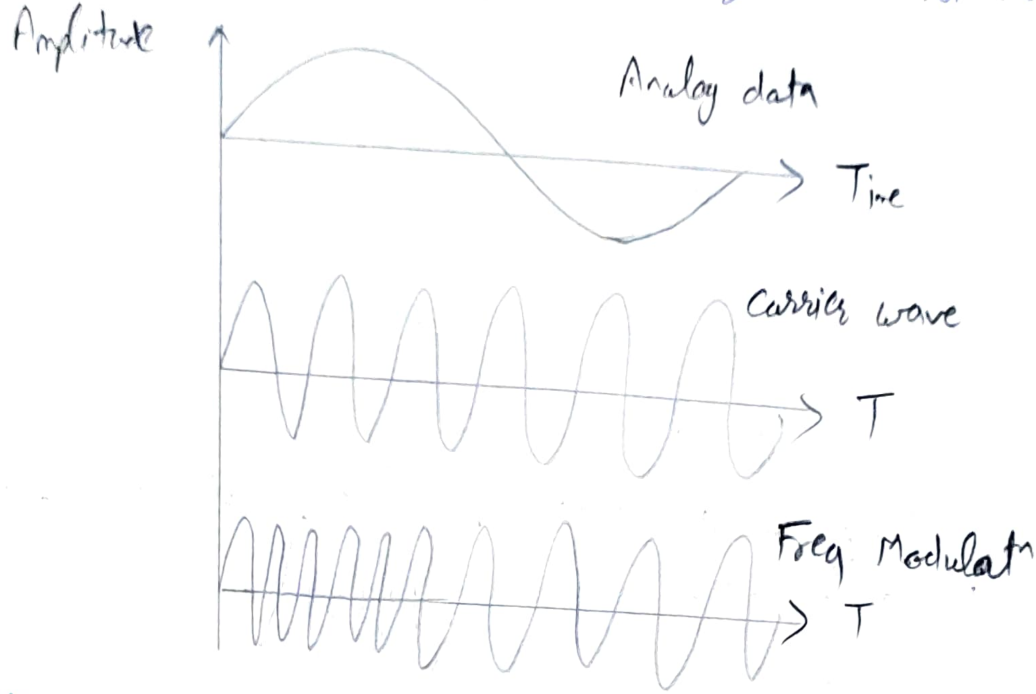
→ The amplitude of modulation signal (analog data) is multiplied by the amplitude of carrier freq., which reflects analog data.

→ The freq. and phase of carrier signal remain unchanged.

ii) Frequency Modulation:

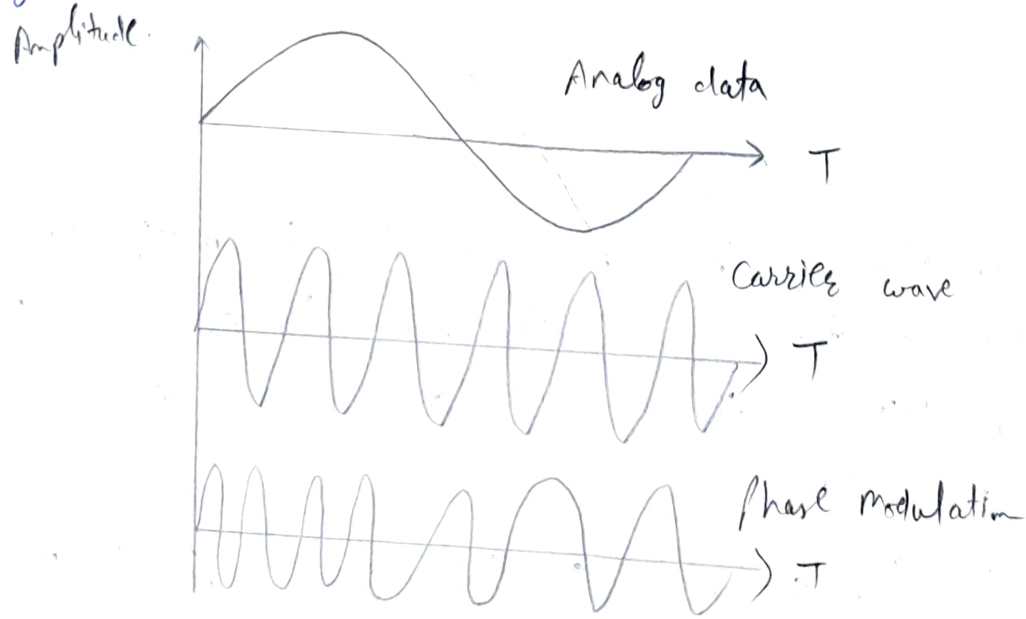
→ The freq. of the carrier signal is modified to reflect the change in the voltage levels of the modulating signal (analog data).

→ The amplitude & phase of carrier signal are not altered.



iii) Phase Modulation:

→ The phase of carrier signal is modulated in order to reflect the change in voltage (amplitude) of analog data signal.



→ Freq. of carrier signal is not ↑↑.

→ Freq. of carrier is signal is changed (made dense & sparse) to reflect voltage change in amplitude of modulating signal.

* Difference b/w Analog & Digital Modulation:

	Analog Modulation	Digital Modulation
1)	AM signal can signify any value in a range.	DM signal can only signify with a set of discrete values.
2)	The i/p must be in the form of analog.	The i/p must be the data in the form of digital.
3)	The value b/w the max & min is considered to be applicable.	Only two binary no. are considered such as 1 and 0.
4)	The AM can generate a signal to carry the frequently changing data.	DM generates a signal whose rate changes at particular time intervals.
5)	not easy to disconnect the signal from noise.	the signal can simply disconnect from noise.