MODULE-1

COMMUNICATION :-

Communication is the process of establishing connection or link between two points for information exchange.

(basic Process of exchanging information) ex: Speaking, writing or sending radio signals.

e.g. :- line telephony and line telegraphy, radio telephony and radio telegraphy, point-to-point communication and mobile communi-cation, computer communication etc.

1. Elements of a Communication System -

(i) the generation of a thought pattern or image in the mind of an originator.

(ii) the description of that image, with a certain measure of precision, by a set of oral visual symbols.

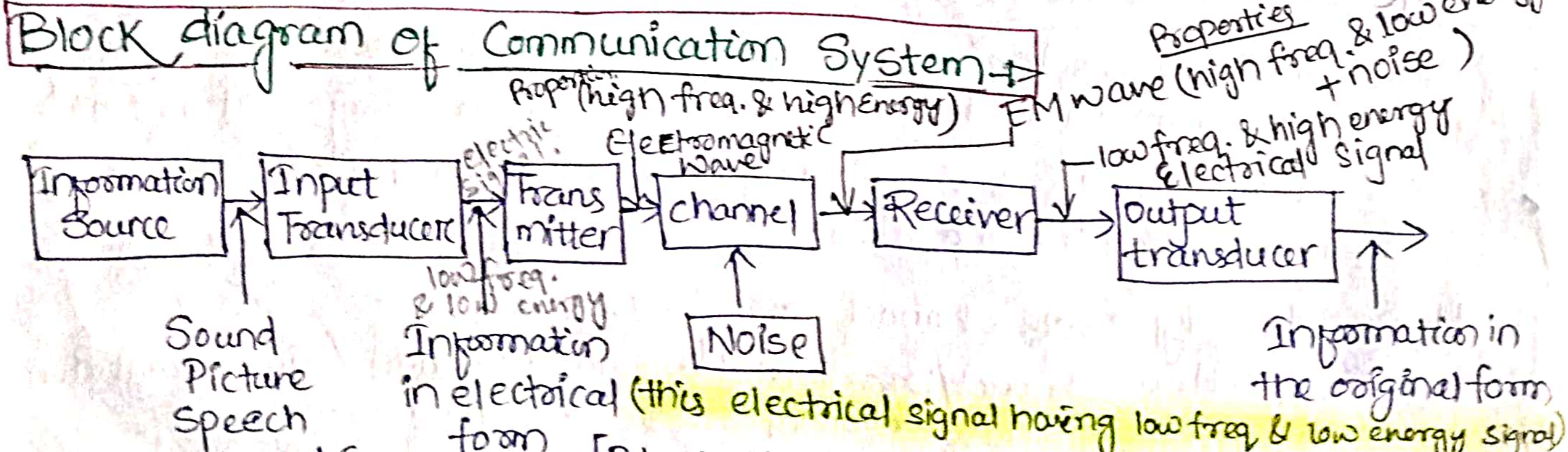
(iii) the encoding of these symbols in a form that is suitable for transmission over a physical medium of interest.

(iv) the transmission of the encoded symbols to the desired destination

(v) the decoding and reproduction of the original symbols.

(vi) the recreation of the original thought pattern or image, with a definable degradation in quality, in the mind of a receipt.

Block diagram of Communication System



[Block diagram of a Communication System]

(These are the essential Components of a Communication System).

1. Information Source -

The function of information Source is to produce required message which has to be transmitted.

(Various messages are in the form of words, groups of words, code, image, Video, Symbols, sound Signal etc.)(out of these messages, only the desired message is selected and Conveyed or Communicated).

2. Input Transducer

A Transducer is a device which converts one form of energy into another form. (means converts non-electrical signal to electrical Signal).
(The message from the information Source may or may not be electrical in nature. In a case when the message produced by the information Source is not electrical in nature, an input transducer is used to Convert it into a time-varying electrical Signal. for example, in case of radio-broadcasting, a microphone Converts the information or message which is in the form of Sound waves into Corresponding electrical Signal).

3. Transmitter

The function of the transmitter is to process the electrical signal from different aspects.

(It is having modulator, amplifier, A/D Conversion, filter, mixer, Antenna)

translate low freq. to high freq. signal

amplify low energy signal to high energy signal

4. The Channel & the Noise

Channel means the medium through which the message travels from the transmitter to the receiver. (We can also Say that the funcn of the channel is to provide a physical connection between the transmitter and the receiver)

→ Here two types of channel are there,

i) wireless channel/broadcast
→ for this wireless channel antenna

ii) wired channel / Point to point channel
→ optical cable
→ copper cable

→ During the process of transmission and reception the signal gets distorted due to noise introduced in the System. (Noise is an unwanted signal which tend to interfere with the required signal).

→ Noise signal is always random in character. It may intefere with signal at any point in a communication System. However, the noise has its greatest effect on the signal in the channel.

5. Receiver –

having demodulator, Amplifier, Antenna

- demodulator ↓ translate high freq. to low freq.
- Amplifier ↓ Low energy to high energy
- Antenna ↓ remove noise

→ The main function of the receiver is to reproduce the message signal in electrical form from the distorted received signal. This reproduction of the original signal is accomplished by a process known as the demodulation or detection.

→ Demodulation is the reverse process of modulation carried out in transmitter.

6. Destination –

It is the final stage which is used to convert an electrical message signal into its original form.

e.g.:- Speaker (Translate Electrical Signal into voice)

(This is the complete process which happens in the communication System).

2. Sources of information :—

→ Some of the important Sources of information in the Communication are 1. Speech 2. Music 3. Pictures 4. Computer data

→ A Source of information is basically a signal which Carries the information.

Signal – A signal is a funcⁿ of one or more independent variables Such as time, space etc. which Contains Some data or information.

→ signal may be a funcⁿ of time, temperature, pressure etc.
Ex: speech, music, picture, video signal etc.

→ signals must be in the form of voltage or current for electronic communication.

Classification of Signals

Signals
- One dimensional Signals → speech, Music, Computer data
- Two dimensional Signals → Pictures
- Three dimensional Signals → Video data
- Four dimensional Signals → Volume data over time.

(Let us discuss these Sources one by one).

1. Speech (Primary method of human Commⁿ)

Speech involves transfer of information from the Speaker to the listener. Such a transfer of information takes place in a language common to both parties. in following three stages:

Source (i) Production (of the voice from the vocal cord)

through which it travels (ii) Propagation (Propagates in air at 300 m/s) meter/second

and (iii) Perception (how to other person interpret the information and response to that) data receives

→ The band of frequencies for the speech communication is 300 Hz to 3100 Hz. This band is utilized for the commercial telephonic communication.

When u want to speak we have the apply some pressure on your vocal track. Due to this our vocal track vibrates and this also fell when u touching your neck. That vocal track is vibrating the pressure that coming out is vibrating the vocal track with producing a sound. the air inside our mouth in a certain rhythm (or) in a certain freq. This sound is coming out from our mouth and travelling in the air. & the sound reaching the listener that through his ear & the mind process & they understand that whatever

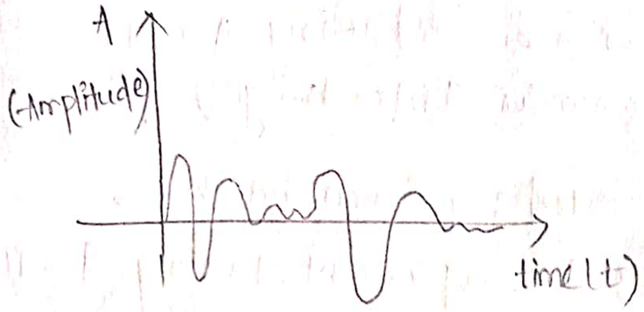
2. Music

(Music is sound signal produced from instruments)

Music signal is originated from the instruments Such as the piano, guitar, violin, flute etc.
tabla

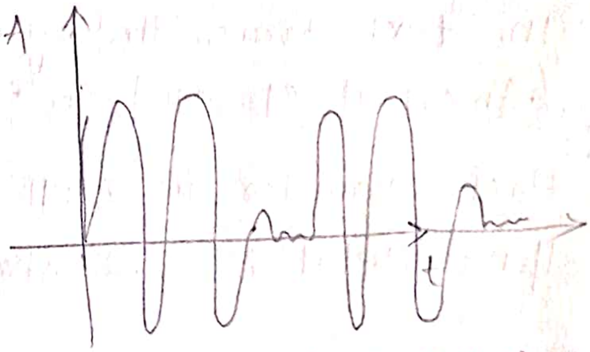
→ Music signal has two possible structures.

(i) Melodic Structure
(ii) Harmonic structure. – same type of Sound is repeated after a certain interval of time.



(i)

which feels good to our ears.



(ii)

→ Melodic structure consists of a ~~Sequence~~ time & sequence of sounds. where as the Harmonic structure consists of a set of simultaneous Sounds.

→ It is Bipolar in nature. (Same as Speech signal).
↳ (+ve & -ve charge carriers)

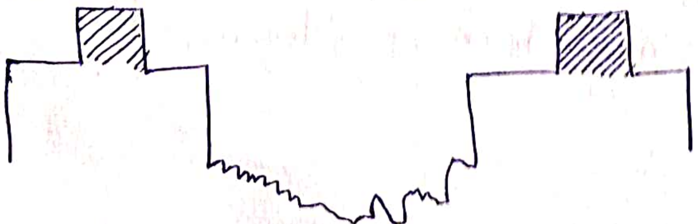
→ Music signal has wider frequency band, that extends upto 15KHZ.

(Diff^n from speech signal due to wider freq. band).

3. Picture –

→ It can be static or dynamic.
(still pictures) video

→ Static picture are the photographs taken by the Camera (Still frames) Whereas dynamic ones are that produced on TV Screens (movable frames or videos).



(a typical video Signal)

→ The bandwidth required for this video Signal is 0 to 5 MHZ.

→ The video signal consists of a luminance signal and chrominance signal.

→ The luminance signal conveys the brightness information while chrominance signal conveys the colour information.

4. Computer data

In computer data, information processed, analysed and stored by a computer.

→ Personal computers are used for electronic mail, exchange of software, and sharing of resources.

→ The text transmitted by a computer is encoded using ASCII (Americal standard code for Information Interchange).

→ Each character in ASCII is represented by seven data bits. Hence, total $2^7 = 128$ characters can be represented using ASCII.

COMMUNICATION CHANNELS : — A channel is the medium, through which the message having some information travel from the Tr. to the Rec.

→ The medium over which the information is passed from the transmitter to the receiver is called as a communication channel.

→ The classification of channels has been shown below,

Communication channels
- Channels based on guided propagation
  - Telephone channels
  - Coaxial Cable
  - optical fiber
- Channels based on Unguided/free propagation
  - ~~Wireless~~ broadcast Channels
  - Satellite channels
  - Mobile radio channels.

[Classification of communication Channels].

Some of the important characteristics of a channel are :-

(i) Power required to achieve the desired S/N ratio.
(ii) Bandwidth of the channel.
(iii) Amplitude & phase response of channel.
(iv) Type of channel (Linear or nonlinear)
(v) Effects of external interference on the channel.

1) Telephone Channels :

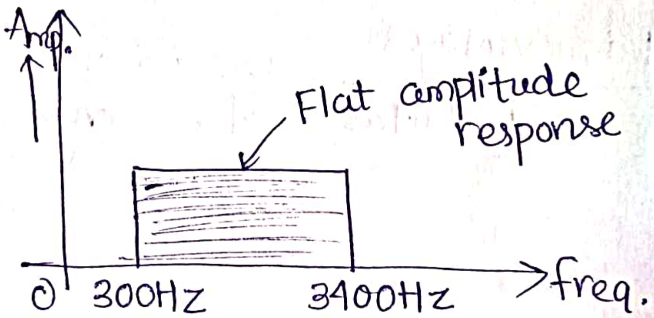
→ It is designed for providing service to voice signals such as telephones.

→ The telephone channels are also used for the World wide Internet Connection.

→ Therefore telephone channel is the best possible option for the data communication over long distances.

Features –

(i) Bandpass characteristics over 300 to 3400 Hz as shown in the below fig.



[∵ Characteristics of telephone Channel]

(ii) High Signal to noise ratio of about 30 dB.

(iii) Approximately linear response.

→ The amplitude response is flat over the entire passband as show in the above fig.

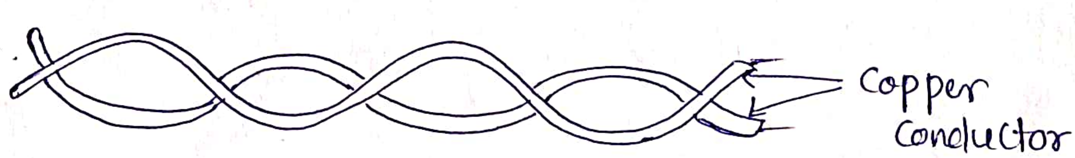


Fig : Twisted pair Cable

→ The telephone channel is built using twisted pairs for signal transmission. A Twisted pair Cable Consist of two solid Copper Conductors ~~each of which is encased in a PVC sheath~~. It can be shielded or unshielded. The unshielded twisted pair Cables are very cheap & easy to install. However, they are badly affected by the noise interfer

→ The characteristics impedance is 90 Ω to 110 Ω.

→ It can allow a signal having 0-200KHz Bandwidth, without any attenuation. (loss of signal)

→ This is also commonly used medium and it is quite cheaper than the co-axial cable.

→ The noise immunity can be improved by using a Shielded twisted pair Cable.

→ The Speed of the twisted Pair cable is 10 mbps to 10gbps
megabyte/sec. gigabyte/sec.

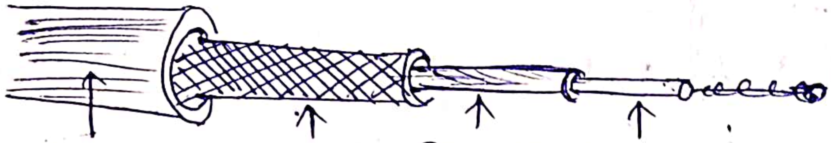
Characteristics

- Most popular
- Used in LAN and Local Telephone Lines
- Can carry voice & data signals
- Copper wires pair are insulated by plastic.
- Wires are twisted together in order to reduce noise.
- ~~It is of~~ In expensive and easy to install & maintain.

Disadvantages

Unsuitable for long distance.

Co-axial Cables –



(construction of a co-axial cable).

→ A co-axial cable consist of an inner conductor and an outer Conductor separated by a dielectric material.

→ The inner conductor is made of a copper wire and outer Conductor is made of copper coated steel.

~~→ Compared to a twisted pair Cable a co-ax~~

→ The speed of the Co-axial cable is 10 mbps to 100mbps.

features –

(i) Two types of cables having 75Ω and 50Ω impedance are available.

(ii) Because of the shield provided, this cable has excellent noise immunity.

(iii) It has a large bandwidth and low losses.

(iv) This cable is Suitable for point to point or point to multipoint applications.

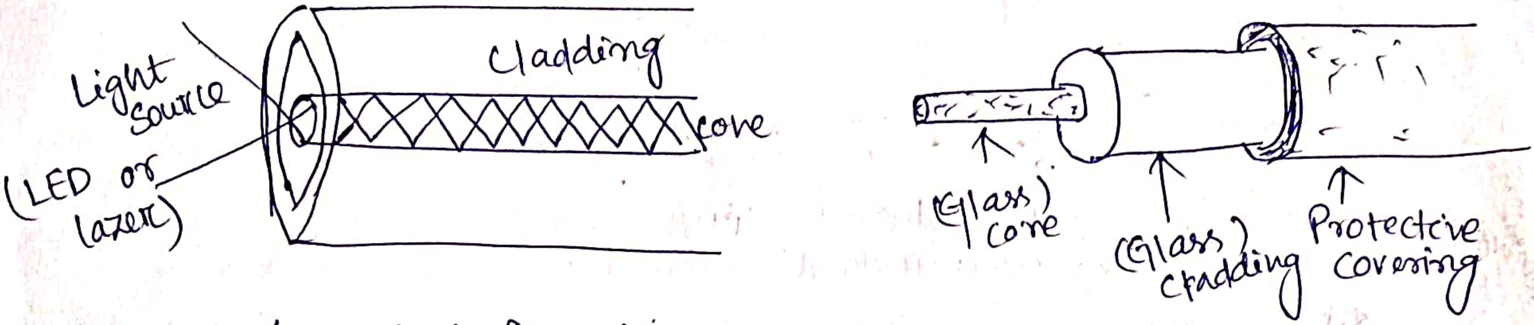
(v) These cables are costlier than twisted pair cables, however, they are cheaper than the optical fiber Cables.

(vi) It is essential to use closely spaced (after every 1 Km) repeaters to achieve the data rates of 8.5 Mb/s to 274 Mb/s.

Disadvantages

→ Expensive than twisted pair Cable.

3) Optical Fiber Cables –



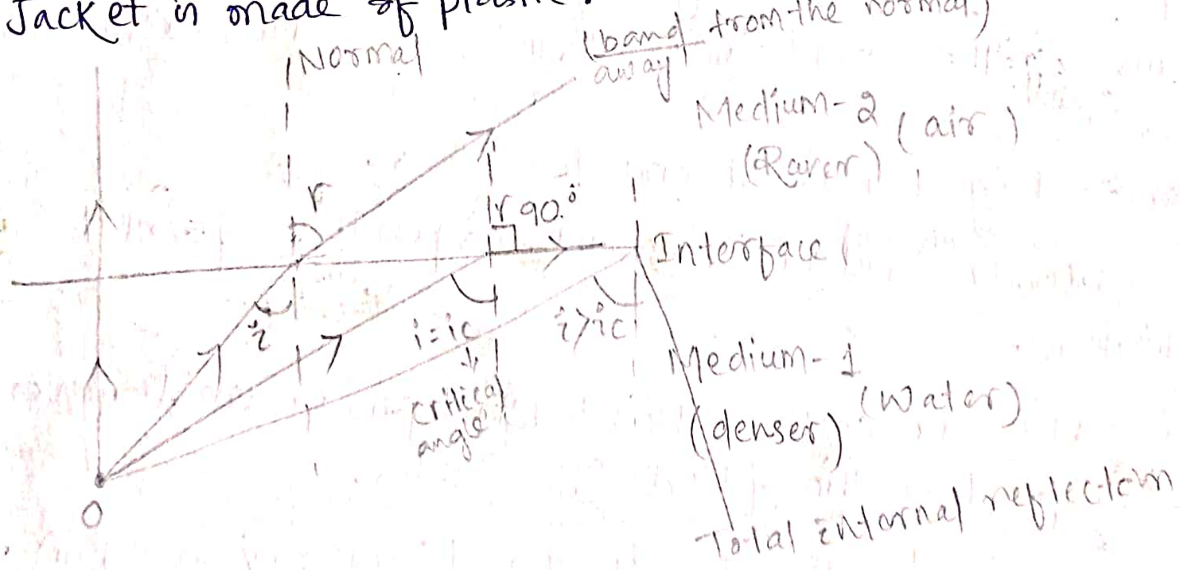
(optical fiber)

→ An optical fiber is a dielectric waveguide that transports signals in the form of light from one place to another place.

→ optical fiber works on the principle of total internal reflection to guide light through the cable.

→ It consists of a central core within which the propagating wave is confined and core is surrounded by a cladding layer, which is itself surrounded by a thin protective Jacket.

→ The core and cladding are both made of pure silica glass and the Jacket is made of plastic.



(When a ray of light strikes the interface at an angle greater than the critical angle, it comes back in the same medium. This phenomenon is called total internal reflection).

Characteristics -

(i) Higher bandwidth therefore can operate at higher data rates.
(ii) the Signal attenuation(loss) is low.
(iii) Noise distortion is reduced.
(iv) Small size and light Weight
(v) Used for point to point communication.

Applications -

(i) The installation cost of optical fibers is higher than that for the co-axial or twisted wire Cables.
(ii) Optical fibers are used in the telephone systems.
(iii) In the local area networks (LANs).

Advantages of Optical Fibers -

(i) Small size and light Weight.
(ii) Easily available.
(iii) Since the transmission takes place in the form of Light rays the signal is not affected due to any electrical or electro-magnetic interference.
(iv) Large bandwidth -(As the light rays have a very high freq. in the GHz range, the bandwidth of the optical fiber extremely large).
~~(v)~~

Disadvantages -

(i) The cost of Optical Fiber Cable is high.
(ii) Joining the optical fibers is a difficult job.

(1) Wireless Broadcast Channels :-

Radio is the first wireless Service to be broadcast.

(1) → These channels are used for the transmission of radio & TV signals.

(3) → The transmitting antenna is mounted on a tower or a hall in order to reach to the receiver.

(2) → Here, the information is transmitted only in one direction and all the users receiving the Same data.

(5) → The receivers uses an antenna to receive the signals.

→ An example of the short-range broadcast radio is Bluetooth.

(4) → The ground wave, sky wave & Space wave are the three types of propagation techniques used for the propagation of EM waves.

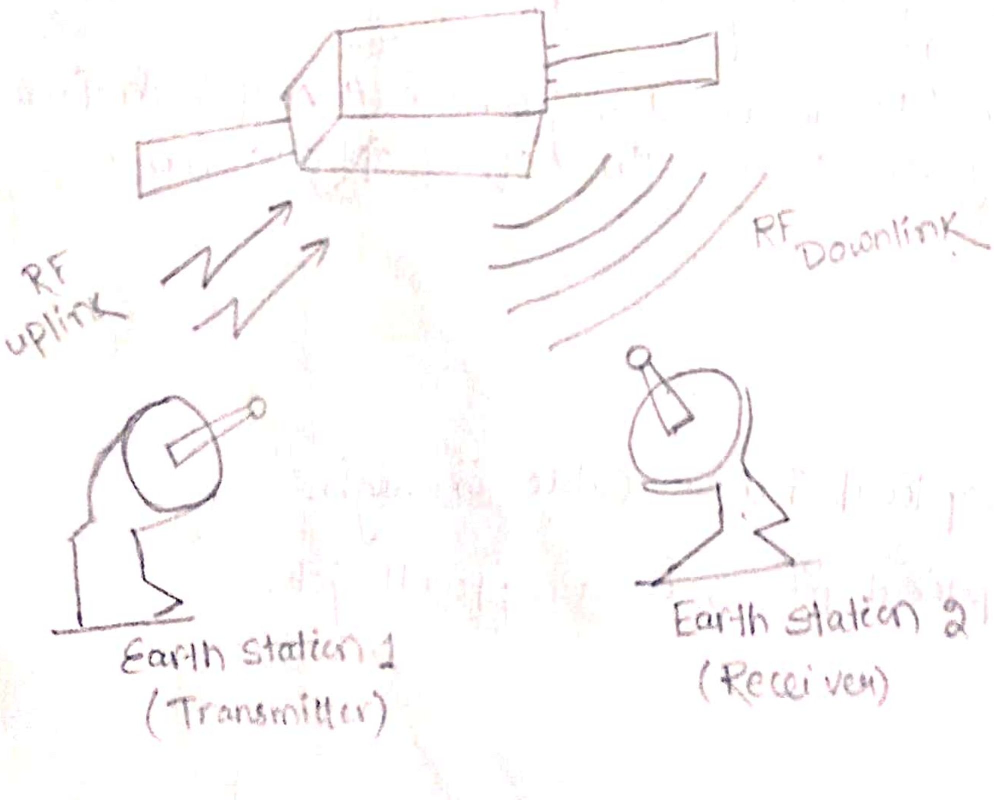
are surface waves which travels along the surface of the earth.

are reflected from the ionosphere.

It travels in a straight line from transmitter to receiver through space.

Disadvantage –
→ It is unidirectional.

(3) Satellite channel –



→ In Satellite based Comm., Comm. takes place between two earth stations through a Satellite.

→ Here, electromagnetic waves are used as carrier signals which carry the information such as voice, audio, video or any other data between ground and space and vice-versa.

→ Satellite microwave systems transmits signals between directional Parabolic antennas. radiates or receives signal in specific dire^n allowing increased performance & reduced interference from unwanted Source

→ They use low gigahertz frequencies and Line of Sight Communication. It can transmit and receive data only where transmit and receive stations are in view of each other without any sort of an obstacle between them. ex: FM radio, microwave and satellite transmission

→ The Satellites act as repeaters with receiving antenna, transponder, ~~with~~ and transmitting antenna.

→ Satellite microwave systems can reach the most remote Places on earth and communicate with mobile devices.

→ This systems works in the following way:

- Signal is Sent through cable media to an antenna which beams the signal to the Satellite.
- The Satellite then transmits the Signal back to another location on earth as Shown in the above fig.

Characteristics

→ It uses frequency range between 11 ~~and~~ to 14GHz.

→ Attenuation depends on frequency, power, antenna size and atmospheric condition.

→ The installation of Satellites is extremely difficult and the alignment of earth station antennas must be perfectly aligned.

→ The cost of building and launching is very high.

→ The Satellites can provide broadcast Services.

→ The message signal transmitted by the earth station to the Satellite is called as an uplink Signal.

→ It is amplified and down converted in frequency by the transponder and then retransmitted back to various earth stations.

→ The signal from Satellite to earth station is called as the downlink signal.

→ The uplink Signal frequency is 6 GHz and downlink Signal frequency is 4GHz.

(2) ## A Mobile Radio channel -

→ In mobile Communication, the sender and the receiver both are allowed to move with respect to eachother.
It means, the users to move from one physical location to another during communication.

→ Here, the Information is transmitted in both the dire^n at the same time.

→ The radio propagation takes place due to scattering of EM waves from the surfaces of the surrounding buildings & diffraction over and around them. Hence, the transmitted energy reaches the receiver via multiple paths. This is called as multipath communication.

→ The signals taking diff^n paths will have to travel diff^n path lengths. So, they have diff^n phase shifts when they reach the receiver.

→ The total signal strength at the receiver is equal to the vector sum of all the signals. Therefore, it keeps changing continuously. Hence, mobile channels are called as the linear time varying channels and it is statistical in nature.

3) Classification of Communication Systems (Line & Wireless or Rad

According to the mode of Propagation, Communication may be divid in ~~the following~~ to two forms.

(i) Line Communication
(ii) Wireless or Radio Communication.

(i) Line Communication –

→ In line communication, the medium of transmission is a pair of conductors called transmission line. This is also called as line channel.

→ This means that in line Communication, the transmitter and the receiver are connected through a wire or line.

Disadvantage –

→ Noise interference on the channel.

→ Power required.

(ii) Wireless or Radio Communication –

→ In wireless or radio communication, a message is transmitted through open space by electromagnetic waves called as radio waves.

→ Radio waves are radiated from the transmitter in open space through a device called Antenna.

→ A receiving antenna intercepts the radio waves at the receiver.

→ All the radio, TV and Satellite broadcasting are wireless or radio communication.

Advantages –

→ cost effectiveness

→ Possible long distance communication

4) Modulation Process, Need of modulation and Classify modulation Process.

- **The Modulation Process –**

→ It is a process of Modification of Carrier signal with respect to modulating (message) signal.

→ Signals Containing information are referred as modulating Signals. This information bearing signal is also called baseband Signal.

→ The Carrier frequency is greater than the modulating frequency. The Signal resulting from the process of modulation is called modulated Signal.

Types of Modulation ⟹

Modulation is basically of two types.

(i) **Continuous Wave Modulation –**

→ When the Carrier Wave is Continuous in nature, the modulation Process is Known as continuous wave (CW) modulation or analog modulation.

Example : Amplitude Modulation and Angle Modulation

→ When the amplitude of the carrier is varied in accordance with the message Signal, it is Known as amplitude modulation.

→ When the angle of the carrier is varied according to the instantaneous value of the modulating signal, it is called angle modulation. Angle modulation may be further Subdivided into Frequency modulation and phase modulation, in which the instantaneous frequency and phase of the Carrier are varied in accordance with the message Signal.

Continuous Wave Modulation
- Amplitude Modulation
- Angle Modulation
  - frequency modulation
  - Phase Modulation.

(ii) <u>Pulse Modulation</u> –

When the carrier wave is a pulse-type waveform, the modulation process is known as pulse modulation.

→ In pulse modulation, the carrier consists of a periodic sequence of rectangular pulses.

→ It can be of an analog or digital type.

→ In analog pulse modulation, the amplitude, duration or position of a pulse is varied in accordance with sample values of the message signal.

→ The analog pulse modulation ~~may~~ are of three types.

(i) pulse-amplitude modulation (PAM)

(ii) Pulse-duration modulation (PDM)

(iii) pulse-position Modulation (PPM).

→ The digital form of pulse modulation is known as pulse-code Modulation.

1.9.2. Need for Modulation or Benefits of Modulation

(Anna, University, Chennai, Semester Exam. 2003-04; 2004-05)

As discussed earlier, the message signal or baseband signal is used to modulate a high frequency carrier signal inside the transmitter. After modulation, the resulting modulated signal is transmitted with the help of an antenna which is connected at the output side of the transmitter. This modulated signal then travels down the channel to reach at the input of the receiver*.

Now, one question can arise why we use modulation in communication system or what will happen if we transmit message signal or audio signal without modulation. The answer is that the modulation serves several purpose in communication system as discussed below:

(i) Practicality of Antenna: We know that in case when free space is used as a transmitting medium (*i.e.* channel), messages are transmitted and received with the help of antennas. For efficient radiation and reception the transmitting and receiving antennas must have lengths comparable to a quarter-wavelength of the frequency used. For example, in AM broadcast systems, the maximum audio frequency transmitted from a radio station is of the order of 5 kHz. If this message audio signal were to be transmitted without modulation, then the height of the antenna required for an effective radiation and reception will be 1/4th of the wavelength given as

$$l = \frac{\lambda}{4} = \frac{c}{4f} = \frac{3 \times 10^8}{4 \times 5 \times 10^3} = 5 \text{ km}.$$

Obviously, it will be totally impracticable to construct and install an antenna of such a height. However, this height of the antenna may be reduced by modulation technique and yet effective radiation and reception is achieved. In modulation process, low frequency or audio signal at radio stations are translated to higher frequency spectrum, *i.e.*, radio frequency range. These higher radio frequencies with the small wavelength act as carrier for the audio frequencies (*i.e.* modulating signal). Thus the height of the antenna required is much reduced and becomes practical.

As an example, if an audio frequency is translated to a radio frequency carrier of frequency 3 MHz, the antenna height required would be

$$l = \frac{\lambda}{4} = \frac{c}{4f} = \frac{3 \times 10^8}{4 \times 3 \times 10^6}$$

$$l = \frac{1}{4} \times 10^2 = 0.25 \times 100 = 25 \text{ metres}$$

This antenna height may be achieved practically.

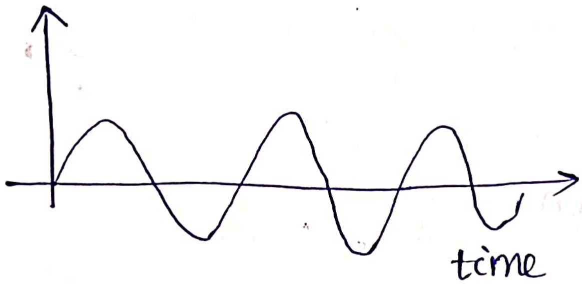
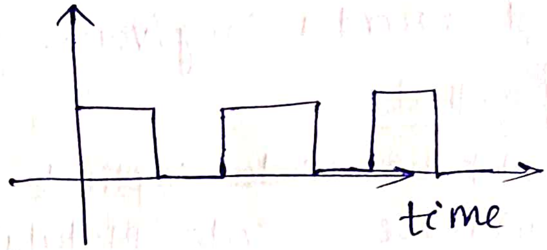
***(ii)* To remove interference:** Another reason for not radiating modulating signal itself is that the frequency range of audio signal is from 20 Hz to 20 kHz. In radio-broadcasting, there are several radio stations. In case, there is no modulation, all these stations transmit audio or sound signals in the range of 20 Hz to 20 kHz. Due to this transmission over same range, the programmes of different stations will get mixed up.

Hence, in order to keep the various signals separate, it is necessary to translate or shift them to different portions of the electromagnetic spectrum. Thus each station is allocated a band of frequency. This also overcomes the drawback of poor radiation efficiency at low frequency.

As an example, in Amplitude Modulation radio-broadcast, the maximum modulating signal frequency permitted is 5 kHz. Amplitude Modulation requires a bandwidth of 10 kHz for each station or channel. Therefore, broadcast channels can be placed adjacent to each other, each channel occupying 10 kHz bandwidth. Hence, different stations may be allotted bandwidths say from 790 to 800 kHz, 800 to 810 kHz and so on. In radio receiver, a tuned circuit at the input selects the desired station and rejects all other stations.

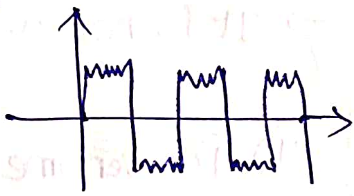
***(iii)* Reduction of noise:** Noise is the major limitation of any communication. Although noise can not be eliminated completely, but with the help of several modulation schemes, the effect of noise can be minimized.

5. Analog and Digital Signals & its Conversion :-

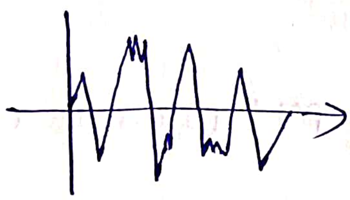
Analog	Digital
	
→ Continous waveform.	→ Discrete data i.e. 0 & 1.
→ Cheap.	→ Expensive
→ Size limitation.	→ No - size limitation
→ Can be distorted with noise.	→ Reliable, privacy more & no disturbance.
→ higher error rate	→ Low error rate.
→ Slow transmission	→ fast transmission.
→ Ex : Human voice in air, analog electronic devices.	→ Ex: Computers, CDs, DVDs and other digital, electronic devices.

★ Analog & digital Signals can be periodic & non-periodic signals.

→ periodic signal ⇒ Continuously repeated patterns



→ Nonperiodic ⇒ has no repetitive patterns.



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→ In Analog signal to digital signal conversion, we representing analog signal by digital signal.

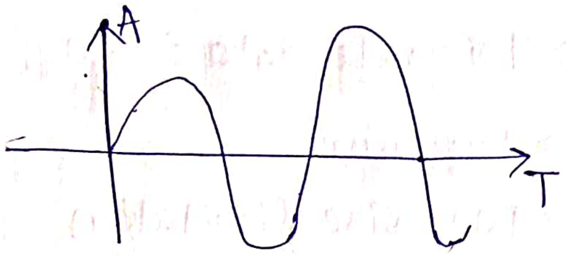
→ In this, we use decoder & encoder.

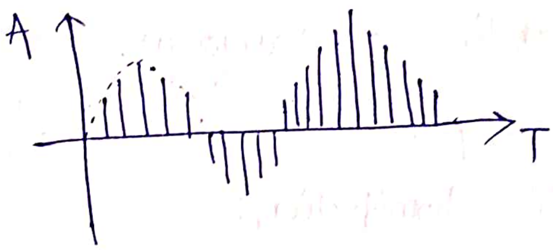
Ex: to record a song/voice.

2 methods

i) Pulse ~~Modal~~ Amplitude Modulation
ii) Pulse Code Modulation.

(i)





(Analog data) to (Digital signal)

→ In PAM, we take analog signal - sample it & generates a series of pulse based on the result of the sampling.

→ Sampling measure the amplitude of a signal to equal enterevals.

→ PAM uses a technique called Sample & hold.

→ It is not used for data communication.

(ii) To overcome disadvantages of PAM, PCM is used.

→ PAM is the 1st step in PCM.

→ PCM modifies the pulse created by PAM to create a completely digital signal.

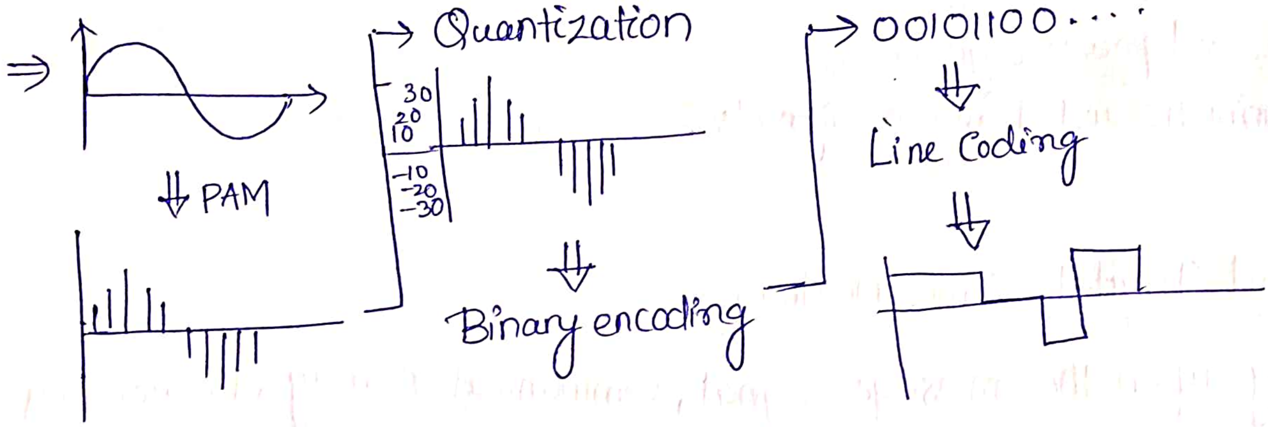
→ There are 3 steps ~~to~~ in PCM.

ⓐ Sampling
ⓑ Quantization
ⓒ line Coding

(a) Sampling – No. of Samples of the signal are taken at regular intervals of a higher frequency of signal.

(b) Quantization – The amplitude is written in binary format.

(c) Line-coding – Conversion to digital signal.



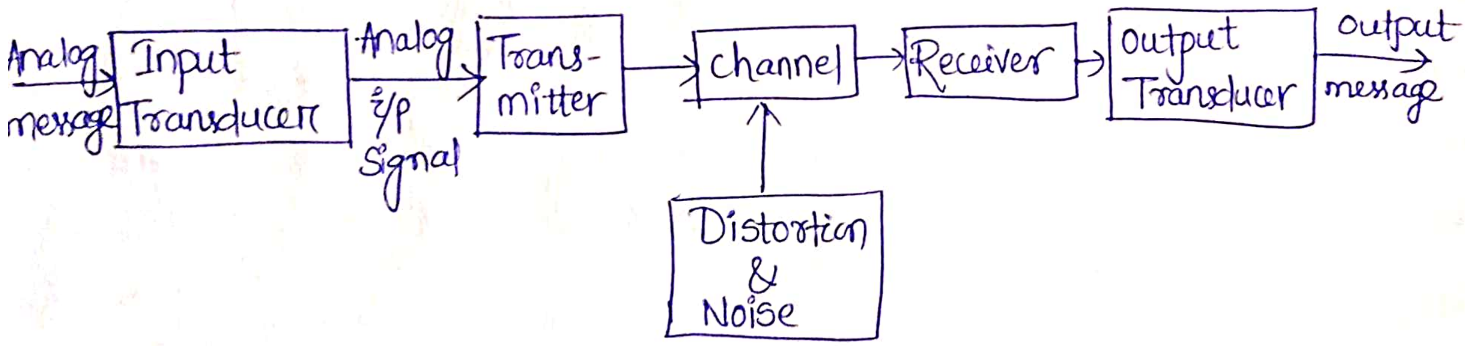
Signals classification :—

1) Continuous time & Discrete time Signals
2) Analog & digital Signals
3) Periodic & Aperiodic Signals
4) Energy and power Signals
5) Deterministic and Random Signals.

Analog and Digital Communication :—

Depending upon the message signal, Communication may classified as
(i) Analog Communication
(ii) Digital Communication.

Analog Communication



[Basic analog Communication System].

→ Analog Communication is that type of Communication in which the message or information signal to be transmitted is analog in nature.

→ This means that in analog communication the modulating Signal (i.e. baseband signal) is an analog signal.

→ This analog message signal may be obtained from Sources Such as Speech, video shooting etc.

→ In Analog communication, the analog message signal modulates Some high Carrier frequency inside the transmitter to produce modulated

Signal.

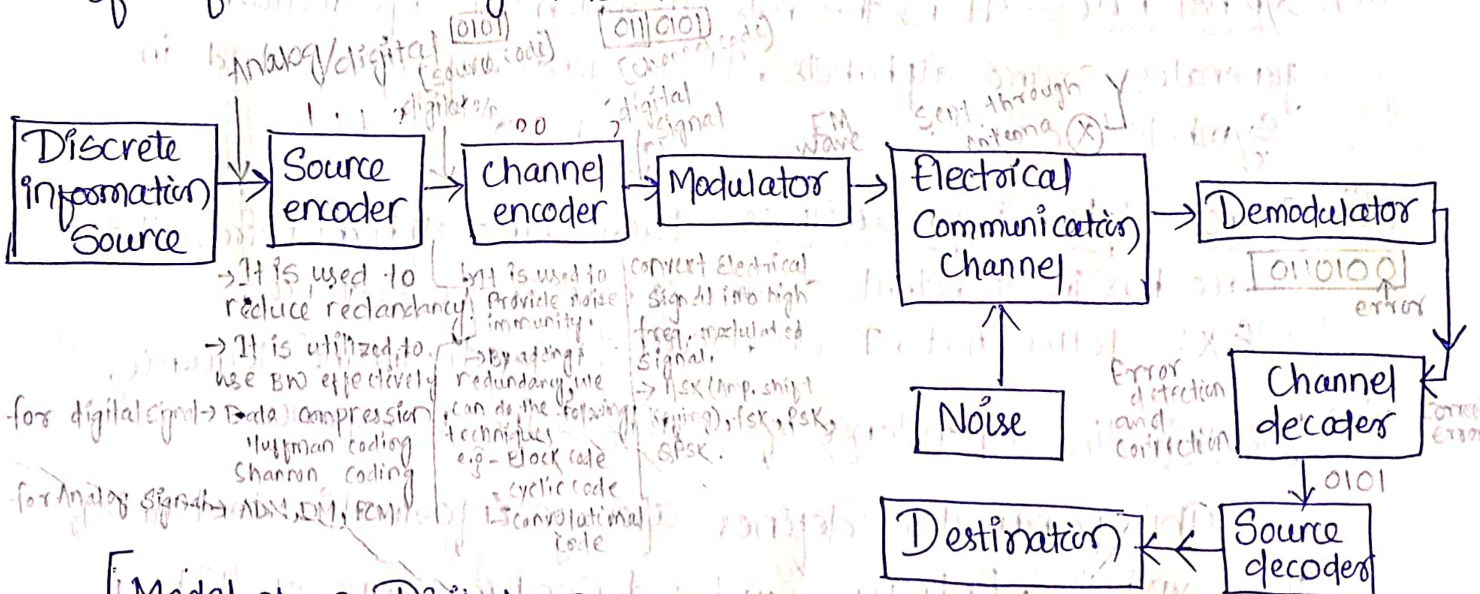
→ This modulated signal is then transmitted with the help of a transmitting antenna to travel through the transmission channel.

→ At the receiver, this modulated signal is received and processed to recover the original message signal.

→ Ex: AM
FM radio transmission
T.V. transmission.

Digital Communication :-

→ In this communication, the message signal to be transmitted is digital in nature. This means that digital communication involves the transmission of information in digital form.



[Model of a Digital Communication System].

Information Source –

It may be classified into two Categories based upon the nature of their output i.e., analog information sources and discrete information Sources.

→ In case of analog communication, the information source is analog. Analog information Sources, Such as m

→ In case of digital Communication, the information Source produces a message signal which is not continuously varying with time.

→ An analog information Source may be transformed into a discrete information Source through the process of Sampling and quantizing.

→ Discrete information Sources are characterized by the following Parameters,

(i) Source alphabet – These are the letters, digits or special characters available from the information Source.

(ii) Symbol rate – It is the rate at which the information source generates Source alphabets. It is generally represented in Symbols/sec unit.

(iii) Source alphabet probabilities – Each Source alphabet from the Source has independent occurance rate in the Sequence.

E.X:– Letters A, E, I etc, Occur frequently in the Sequence.

(iv) Probabilistic dependence of Symbols in a Sequence –

This parameter defines average information content of the Symbols. Entropy may be defined in terms of bits per Symbol.

average information of symbols.

$$\text{Entropy} = \frac{\text{Total Information}}{\text{no. of messages}}$$

→ The Source information rate is the product of Symbol rate and Source entropy i.e.

$$\underset{\text{(Bits/sec)}}{\text{Information rate}} = \underset{\text{(Symbols/sec)}}{\text{Symbol rate}} \times \underset{\text{(Bits/symbol)}}{\text{Source entropy}}$$

→ Thus, the information rate represents minimum average data rate required to transmit information from Source to the destination.

Source Encoder and Decoder –

The Symbols produced by the information source are given to the Source encoder. These Symbols cannot be transmitted directly.

→ They are first converted into digital form by the Source encoder. Each binary '1' and '0' is known as a bit. The group of bits is called a codeword.

→ The Source encoder assigns codewords to the Symbols. for each distinct symbol, there is an unique Codeword.

→ The Codeword can be 4, 8, 16 or 32 bits length. As the number of bits are increased in each codeword, the symbols that may be represented are also increased.

Ex : 8 bits would have $2^8 = 256$ distinct Codewords (means 8-bits may be used to represent 256 symbols)

Similarly, 16 bits would have $2^{16} = 65536$ distinct codewords.

→ Example of Source Encoder are pulse code modulators, delta modulators, vector quantizers etc.

→ Source encoders have some important parameters.

(i) Block Size –

Block Size describes the maximum number of distinct codewords which can be represented by a Source encoder. This depends on the number of bits in the codeword.

Ex:– the block size of 8 bits Source encoder will be 2^8 i.e. 256 Codewords

(ii) Codeword length –

Codeword length is the number of bits used to represent each codeword.

Ex: if 8 bits are assigned to each codeword, then the codeword length will be 8 bits.

(iii) Average data rate –

Average data rate is the output bits per second from the Source encoder. The data rate is generally higher than the Symbol rate.

Ex:- If the length of codeword is 8 bits, then the output data rate from the source encoder would be given as

$$\text{Data rate} = \text{Symbol rate} \times \text{Codeword length}$$

$$= 10 \times 8 = 80 \text{ bits/seconds.}$$

→ The average data rate is higher than the information rate and Symbol rate.

(iv) Efficiency of the Encoder—

The efficiency of the encoder is the ratio of minimum source information rate to the actual output data rate of the Source encoder.

⇒ Decoder is used to perform the reverse operation to that of Source encoder. It converts the binary output of the channel decoder into a Symbol Sequence. Some decoders also use memory to store codewords.

→ The decoders and the encoders can be synchronous or asynchronous.

Channel Encoder and Decoder —

After converting the message or information signal in the form of binary Sequence by the Source encoder, the Signal is transmitted through the channel.

→ The communication channel adds noise and interference to the signal being transmitted. Hence errors are introduced in the binary Sequence received at the receiver end.

→ Therefore, the errors are also introduced in the Symbols generated from these binary codewords. Thus channel coding is done to avoid these types of errors.

→ The channel decoder at the receiver is able to reconstruct error free accurate bit Sequence and reduce the effects of channel noise and distortion.

→ The channel encoder & decoder serve to increase the reliability of a received Signal.

→ However the extra bits which are added by the channel encoders carry no information, rather, they are used by the channel decoder to detect and correct errors if any.

→ The coding and decoding operation at the encoder and decoder needs the memory and processing of binary data.

→ A channel encoder must have the following important parameters.

(i) The coding rate that depends upon the redundant bits added by the channel encoder.

(ii) The coding method used.

(iii) Coding efficiency which is the ratio of data rate at the input to the data rate at the output of the encoder.

(iv) Error control capabilities.

(v) Feasibility of the encoder and decoder.

Digital Modulators and Demodulators –

→ If the modulating signal is digital (i.e. binary codewords), then digita modulation techniques are used.

→ The carrier signal used by digital modulators is always continuous sinusoidal wave of high frequency.

→ If the codeword consists of two bits and they are to be transmitted at a time, then there would be 2^2 i.e., 4 distinct symbols, i.e. codewords. Thus, these codewords will require four distinct waveforms for transmission purpose. Such types of modulators are known as M-ary modulators.

→ Examples of Digital Modulators are –
Amplitude shift Keying (ASK)
Phase Shift Keying (PSK)
frequency shift Keying (FSK)
Differential phase shift Keying (DPSK)
& Minimum shift Keying (MSK).

→ These modulators use a continuous carrier wave, therefore they are also known as digital CW modulators.

→ At the receiver end, the digital demodulator converts the input modulated signal into the sequence of binary bits.

→ A digital modulation method have important Parameters:-

(i) Bandwidth needed to transmit the signal.

(ii) Probability of symbol or bit error.

(iii) Synchronous or asynchronous method of detection.

(iv) Complexity of implementation.

Channel -

→ The connection between transmitter and receiver is established through a communication channel.

→ The communication can take place through wirelines, wireless or fiber optic channels.

→ The other media such as optical disks, magnetic tapes and disks etc. may also be called as a communication channel since they can also carry data through them.

→ However, each and every communication channel has some inherent Problems. These are -

(i) Signal Attenuation -

The signal attenuation in channel occurs due to the internal resistance of the channel and fading of the signal.

(ii) Amplitude and phase distortion -

The transmitted signal is distorted in amplitude and phase due to the non-linear characteristics of the communication channel.

(iii) Additive noise interference -

Additive noise interference is produced due to internal solid state devices and resistors etc., used to implement a Comm. System.

(iv) Multipath distortion -

The multipath distortion occurs mostly in wireless communication channels.

→ The signals coming from different paths tend to interfere with each other.

1.15 ADVANTAGES AND DISADVANTAGES OF DIGITAL COMMUNICATION *(Importan*

In this section, we shall discuss the advantages and disadvantages of digital communicatio briefly.

Advantages:

Following are the advantages of digital communication:

(*i*) The digital communication systems are simpler and cheaper compared to analog communication systems because of the advances made in the IC technologies.

(*ii*) In digital communication, the speech, video and other data may be merged and transmitted over a common channel using multiplexing.

(*iii*) Using data encryption, only permitted receivers may be allowed to detect the transmitted data. This property is of its most importance in military applications.

(*iv*) Since the transmission is digital and the channel encoding is used, therefore the noise does not accumulate from repeater to repeater in long distance communications.

(*v*) Since the transmitted signal is digital in nature, therefore a large amount of noise interference may be tolerated.

(*vi*) Since in digital communication, channel coding is used, therefore the errors may be detected and corrected in the receivers.

(*vii*) Digital communication is adaptive to other advanced branches of data processing such as digital signal processing, image processing and data compression, etc.

Disadvantages

Although digital communication offers, so many advantages as discussed above, it has some drawbacks also. However, the advantages of digital communication outweight disadvantage.

The disadvantages may be listed as under:

(*i*) Due to analog to digital conversion, the data rate becomes high. Therefore more transmission bandwidth is required for digital communication.

(*ii*) Digital communication needs synchronization in case of synchronous modulation.

Performance Comparison of Analog and Digital Modulation

Bandwidth Limitation :-

→ The Frequency range or the band of frequency needed for a Particular given transmission is known as Bandwidth.

→ This band of frequencies required for a particular transmission is also called channel.

→ The information theory states that the greater is the transmission bandwidth of a communication system, the more is the information that can be transmitted.

→ Suppose one is listening to a music in an AM radio. The Complete amount of information available to the human ear is contained in a frequency range upto 15KHz, i.e. musical information extends upto a frequency of 15 KHz. However, in AM radio the maximum modulating frequency is restricted upto 5 KHz and hence the maximum bandwidth of AM transmission is 10 KHz.

Therefore, an AM radio receiver cannot reproduce all the information contained in the music because this will require a bandwidth of 30 KHZ.

→ On the other hand, the bandwidth allocated to a FM transmission is about 200KHz. Thus an FM receiver can easily reproduce the transmitted information without any distortion.

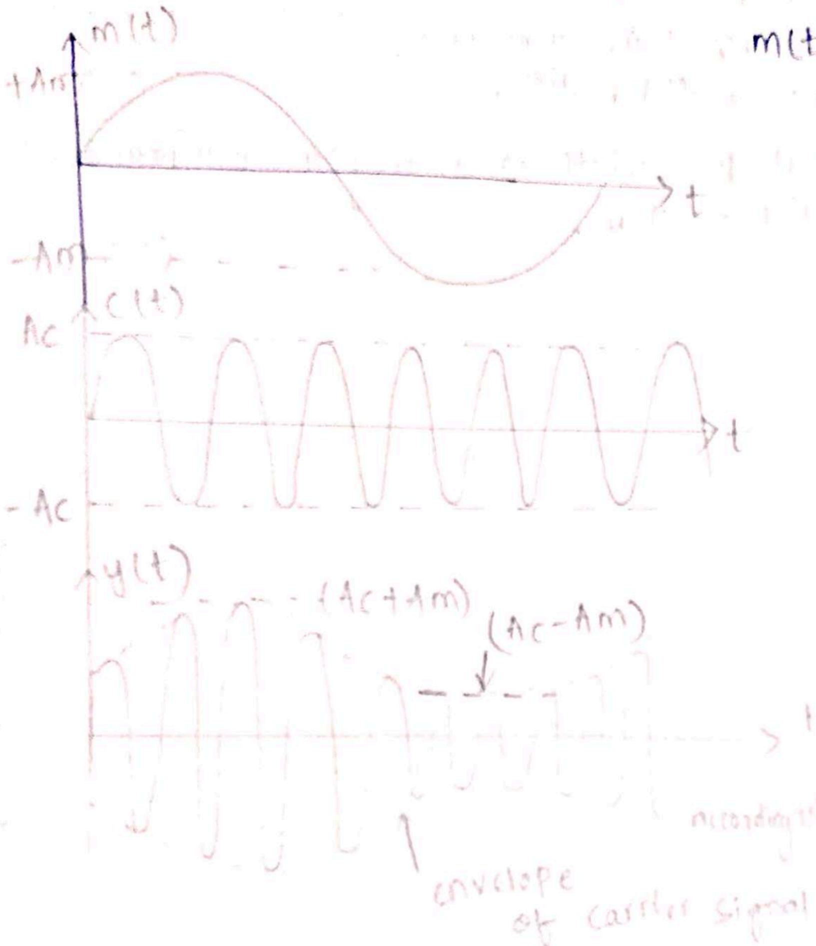
→ This means that a FM System has a better fidelity than an AM System. Also, it may be observed in common life that one prefers to listen to a FM radio than an AM radio.

→ Thus, We can conclude that bandwidth is a major fundamental limitation of a Communication System.

Unit-2 : Amplitude (linear) Modulation System

Amplitude Modulation:-

It is the process in Amplitude of Carrier Signal changes w.r.t message (modulating) signal.



$m(t) \Rightarrow$ modulating Signal (i.e. low freq. signal)
$= A_m \sin \omega_m t$ (If I say freq. is ω_m)

$c(t)$ = Carrier Signal (i.e. high freq. signal)
If its freq is ω_c
$= A_c \sin \omega_c t$

$y(t)$ = Amplitude modulated Signal
Here freq is same as carrier signal and the Amp. of carrier signal change according to modulating signal

→ Here the amplitude of modulated signal is maximum i.e. $A_m + A_c$

→ Amplitude of modulated Signal is minimum i.e. $A_c - A_m$.

→ Envelope of modulated signal change w.r.t to modulating Signal.

→ It is the basic process of amplitude modulation.

Amplitude modulated Signal

$$y(t) = A' \sin \omega_c t$$
$$= (A_c + m(t)) \sin \omega_c t$$
$$= (A_c + A_m \sin \omega_m t) \sin \omega_c t$$

$$= A_c\left(1+\frac{A_m}{A_c}\sin\omega_m t\right)\sin\omega_c t$$

→ Here, $\frac{A_m}{A_c}$ = modulating index = μ

$$\Rightarrow y(t) = A_c(1+\mu\sin\omega_m t)\sin\omega_c t$$

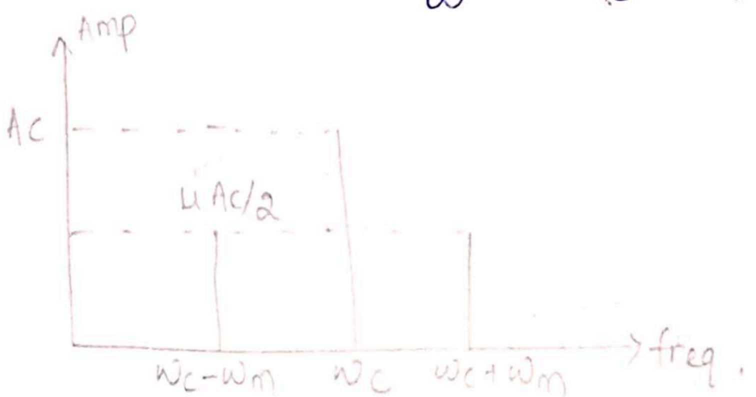
$$= A_c\sin\omega_c t + A_c\mu\,\underbrace{\sin\omega_m t}_{\beta}\,\underbrace{\sin\omega_c t}_{\alpha}$$

$$\boxed{y(t) = A_c\sin\omega_c t + \frac{A_c\mu}{2}\cos(\omega_c-\omega_m)t \mp \frac{A_c\mu}{2}\cos(\omega_c+\omega_m)t}$$

$$\sin\alpha\sin\beta = \frac{\cos(\alpha-\beta)+\cos(\alpha+\beta)}{2} \qquad \text{—(1)}$$

→ This signal is having three freq. Component ω_c, $\omega_c+\omega_m$, $\omega_c-\omega_m$.

→ Side band Amplitude $= \frac{\mu A_c}{2} = \left(\frac{A_m}{A_c}\times\frac{A_c}{2}\right) = \frac{A_m}{2}$



$$\Rightarrow \boxed{BW = 2\omega_m}$$

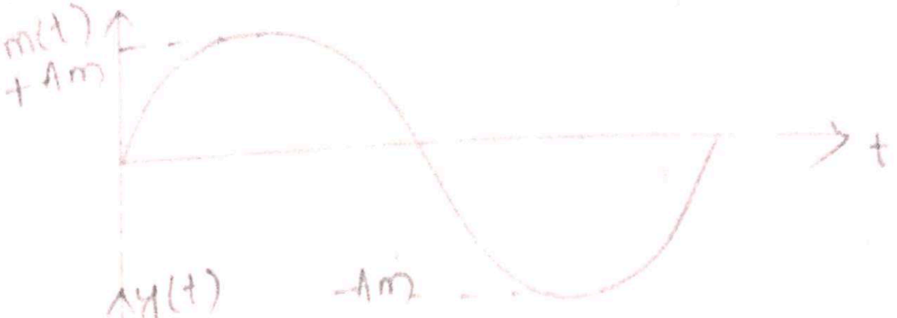
Modulating Index :—

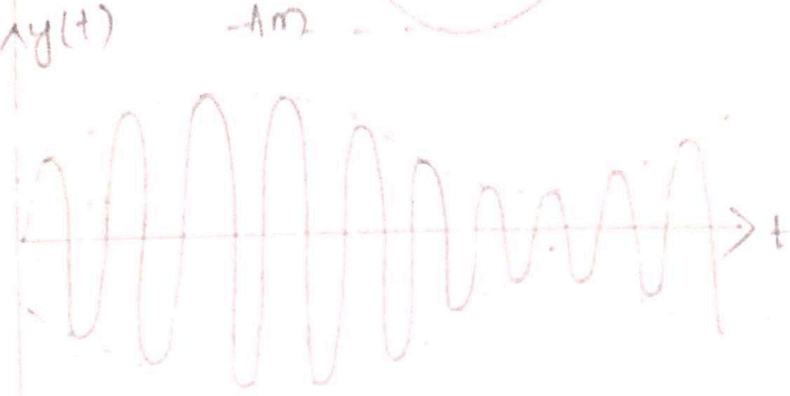
We know the equ^n (1),

$$\boxed{y(t) = A_c\sin\omega_c t + \frac{\mu A_c}{2}\cos(\omega_c-\omega_m)t \mp \frac{\mu A_c}{2}\cos(\omega_c+\omega_m)t}$$

Here, μ is the modulating index.

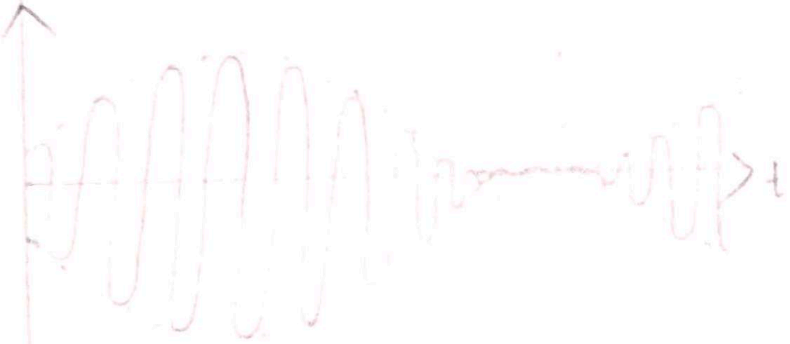
$$\Rightarrow \boxed{\mu = \frac{A_m}{A_c}}$$





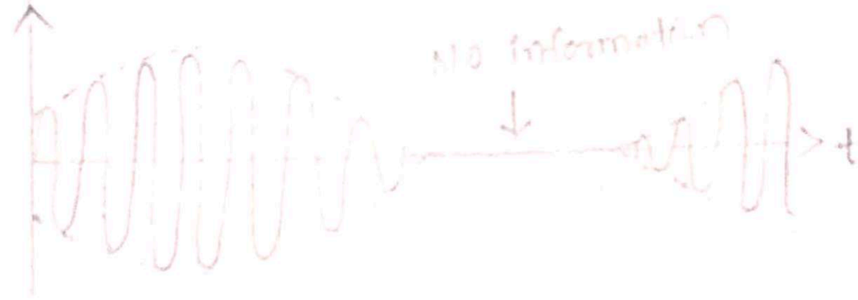
$\boxed{\mu < 1}$

$\boxed{\mu = 1}$

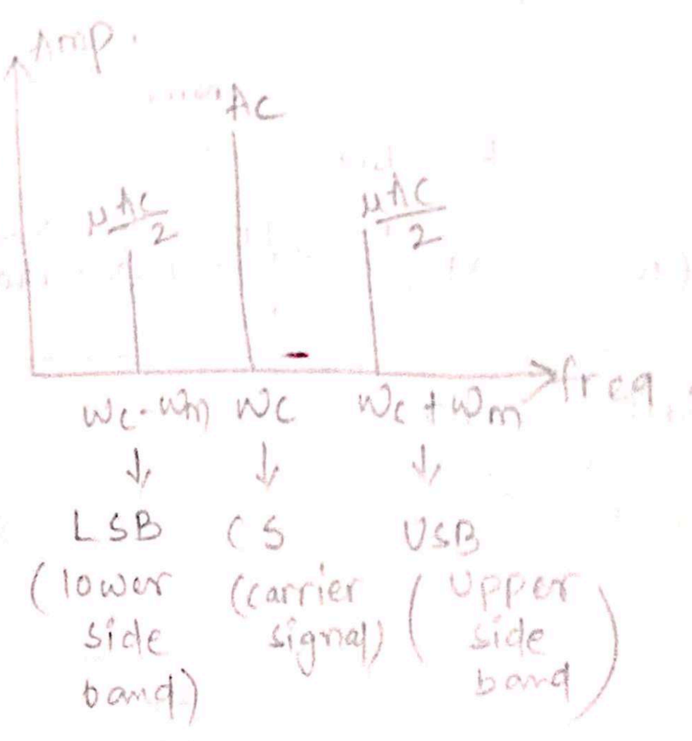


in this modulated signal o/p is 0.

$\boxed{\mu > 1}$



Here distortion is happening. So the signal is getting lost.



Maximum Amplitude, $\boxed{A_{max} = A_c + A_m}$
Minimum " $\boxed{A_{min} = A_c - A_m}$

AM Signal transmitted Power, efficiency & Redundancy :-

We know, the AM signal

$$y(t) = \underbrace{A_c \sin \omega_c t}_{\text{Carrier signal}} + \underbrace{\frac{A_c \mu}{2} \cos(\omega_c + \omega_m)t}_{\text{USB signal}} \mp \underbrace{\frac{A_c \mu}{2} \cos(\omega_c - \omega_m)t}_{\text{LSB signal}}$$

→ Total transmitted Power, $P_t = P_c + P_{USB} + P_{LSB}$

Power of Carrier, $P_c = \frac{A_c^2}{2}$

Power of USB, $P_{USB} = \frac{1}{2}\left(\frac{A_c \mu}{2}\right)^2 = \frac{1}{8} A_c^2 \mu^2$

Power of LSB, $P_{LSB} = \frac{1}{2}\left(\frac{A_c \mu}{2}\right)^2 = \frac{1}{8} A_c^2 \mu^2$

Here, sideband power carrying information but carrier Power not carrying any information.

We also write, $P_t = P_c + P_s$ $[P_s = P_{USB} + P_{LSB}]$

P_c → It is Redundancy → means it doesn't have information.

P_s → It has information.

→ Sideband Power, $P_s = P_{USB} + P_{LSB}$

$$= \frac{1}{4} A_c^2 \mu^2 = \frac{P_c \mu^2}{2}$$

→ Total power, $P_t = P_c + P_s$

$$= \frac{A_c^2}{2} + \frac{1}{4} A_c^2 \mu^2$$

$$= \frac{A_c^2}{2}\left(1 + \frac{\mu^2}{2}\right)$$

$$\boxed{P_t = P_c\left(1 + \frac{\mu^2}{2}\right)}$$ → Transmitted Power.

→ Efficiency is based on information.

$$\boxed{\eta = \frac{P_S}{P_t}}$$ where, P_S = Sideband Power, P_t = Total Power

$$= \frac{\frac{1}{4}A^2\mu^2}{\frac{1}{2}A^2\left(1+\frac{\mu^2}{2}\right)} = \frac{\mu^2}{2+\mu^2}$$

$$\boxed{\eta = \frac{\mu^2}{2+\mu^2}}$$

→ Redundancy, $D = 1 - \eta$

$$= 1 - \frac{\mu^2}{2+\mu^2} = \frac{2+\cancel{\mu^2}-\mu^2}{2+\mu^2} = \frac{2}{2+\mu^2}$$

$\mu = 2$

$$\boxed{D = \frac{2}{2+\mu^2}}$$

Numericals :

→ Efficiency is based on information.

$$\boxed{\eta = \frac{P_s}{P_t}}$$

Where, P_s = Sideband Power

P_t = Total Power

$$= \frac{\cancel{\frac{1}{4}} \cancel{A_c^2} \mu^2}{\cancel{\frac{1}{2}} \cancel{A_c^2} \left(1+\frac{\mu^2}{2}\right)} = \frac{\mu^2}{2+\mu^2}$$

$$\boxed{\eta = \frac{\mu^2}{2+\mu^2}}$$

→ Redundancy, $D = 1 - \eta$

$$= 1 - \frac{\mu^2}{2+\mu^2} = \frac{2+\cancel{\mu^2}-\cancel{\mu^2}}{2+\mu^2} = \frac{2}{2+\mu^2}$$

$$\boxed{D = \frac{2}{2+\mu^2}}$$

Numericals :—

Generation of Amplitude Modulation (AM) :—

AMPLITUDE MODULATOR ⇒

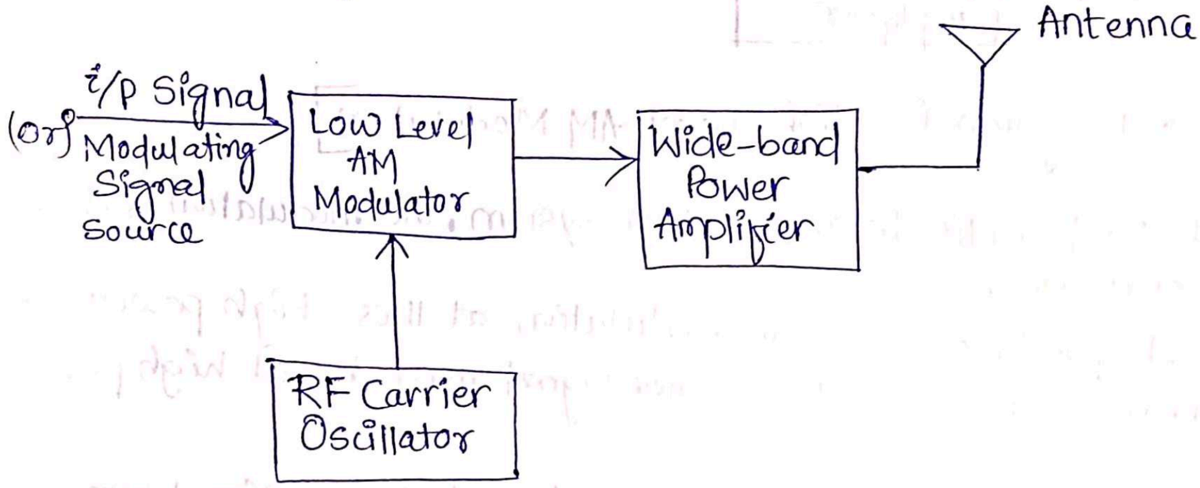
→ The device which is used to generate an Amplitude modulated Wave (AM wave) is known as Amplitude Modulator.

→ The methods of AM Generation may classified into two types.

(i) Low-level AM Modulation

(ii) High-level AM Modulation.

Low-level Amplitude Modulation :—



[Block Diagram for Low Level AM Modulation].

→ In a low-level amplitude modulation system, the modulation is done at low power level.

→ At low power levels, a very small power is associated with the carrier signal and the modulating signal.

→ Because of this, the output power of modulation is low.

→ Therefore, the power amplifiers are required to boost the amplitude-modulated signals upto the desired output level.

→ From the block diagram in the above fig., it is clear that modulation is done at low power level.

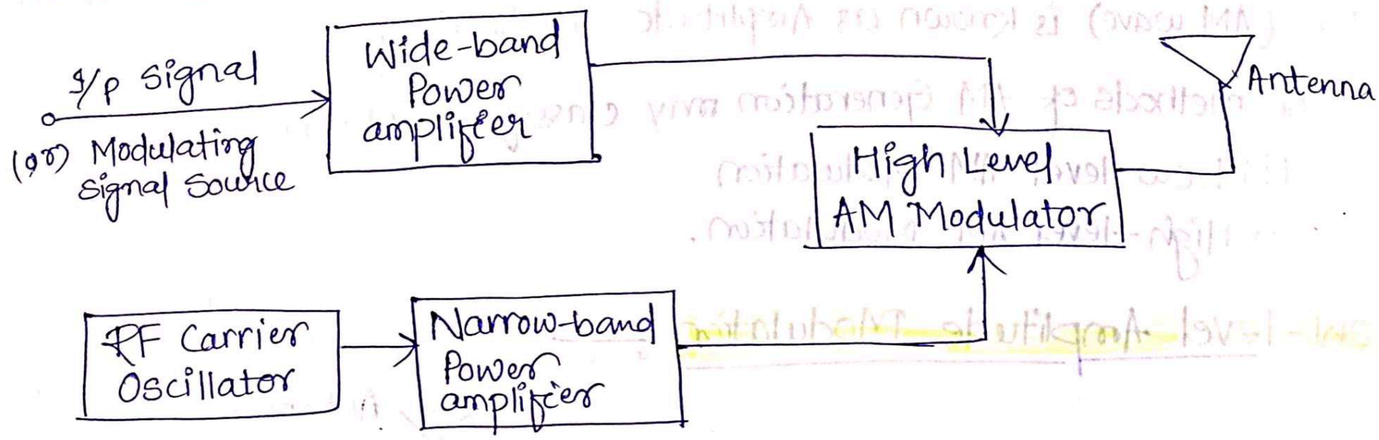
→ After this, the amplitude-modulated signal (i.e. a signal containing a carrier and two sidebands) is applied to a wide-band power amplifier.

→ A wide-band power amplifier is used just to preserve the sidebands of the modulated signal.

→ Amplitude modulated systems, employing modulation at low power levels are also called low-level amplitude modulation transmitters.

→ Example – Square-law diode modulation & Switching modulation.

High Level Amplitude Modulation :—



[Block Diagram for High Level AM Modulation]

→ In a high-level amplitude modulation system, the modulation is done at high power level.

→ Therefore, to produce amplitude-modulation at these high power levels, the baseband signal and the carrier signal must be at high power levels.

→ In block digram, the modulating signal and carrier signal are first power amplified and then applied to AM high-level modulator.

→ For modulating signal, the wide-band power amplifier is required just to preserve all the frequency components present in modulating signal.

→ On the other hand, for carrier signal, the narrow-band power amplifier is required because it is a fixed-frequency signal.

→ Example – Collector modulation method.

Non-linear Resistance or Non-linear Circuits :—

→ We know that the relationship between voltage and current in a linear resistance is expressed as

$$i = bv \quad \text{—— (i)}$$

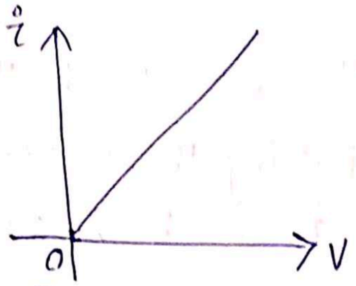
Where,
v = Voltage across the linear resistance
i = Current through linear resistance
& b = any constant of proportionality

→ If equ^n (i) is applied to a resistor, then constant b is clearly its conductance. Also, if equ^n (i) is applied to the linear portion of the transistor characteristic then i is the collector current & v is the voltage applied to the base.

→ Equ^n (i), may be written as

$$i = a + bv \quad \text{—— (ii)}$$

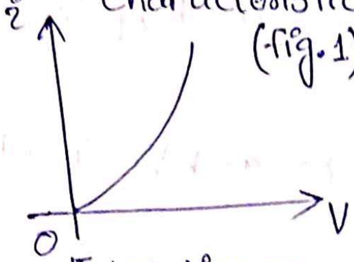
Where a, is the d.c. component of the current.



[Linear resistance characteristics]
(fig. 1)

→ Now, let us consider a non-linear resistance. For a non-linear resistance the current-voltage characteristics will be non-linear as shown in fig. 2. The non-linear relationship between voltage and current may be expressed as,

$$i = a + bv + cv^2 + dv^3 + \dots \quad \text{—— (iii)}$$



[Non-linear resistance characteristics]
(fig. 2)

→ This means that due to non-linearity in the V-i characteristics of a non-linear resistance, the current becomes proportional not only to voltage but also to the square, cube and higher powers of the voltage.

→ For simplicity, neglecting the higher terms in equ^n (iii), we have

$$i = a + bv + cv^2 \quad \text{—— (iv)}$$

→ Devices or circuits having non-linear v-i characteristics can be treated as a non-linear resistance.

→ For example, devices like diodes, transistors, FET etc. exhibit non-linear characteristics and hence work as a non-linear resistance. Equ^n (iv), may be used in relating the output current to the input voltage of a non-linear resistance.

→ We can apply this equ^n to the gate-voltage-drain Current characteristics of a FET, then

$$i = a + bv + cv^2$$

→ If two voltages are applied at gate Simultaneously, then

$$i = a + b(v_1+v_2) + c(v_1+v_2)^2$$

$$\Rightarrow i = a + b(v_1+v_2) + c(v_1^2+v_2^2+2v_1 \cdot v_2) \longrightarrow (V)$$

→ Let us consider that the two input voltages are Sinusoidal having different frequencies, then,

$$v_1 = V_1 \sin \omega t$$

$$\& \quad v_2 = V_2 \sin pt$$

Here, ω and p are the two different frequencies.

→ Putting the values of v_1 & v_2 in equ^n (V), we get

$$i = a + b(V_1 \sin \omega t + V_2 \sin pt) + c(V_1^2 \sin^2 \omega t + V_2^2 \sin^2 pt + 2V_1V_2 \sin \omega t \sin pt)$$

or

$$i = a + bV_1 \sin \omega t + bV_2 \sin pt + cV_1^2 \sin^2 \omega t + cV_2^2 \sin^2 pt + 2cV_1V_2 \sin \omega t \sin pt$$

or

$$i = a + bV_1 \sin \omega t + bV_2 \sin pt + \frac{1}{2}cV_1^2(2\sin^2 \omega t) + \frac{1}{2}cV_2^2(2\sin^2 pt) + cV_1V_2(2\sin \omega t \sin pt).$$

or

$$i = a + bV_1 \sin \omega t + bV_2 \sin pt + \frac{1}{2}cV_1^2(1-\cos 2\omega t) + \frac{1}{2}cV_2^2(1-\cos 2pt) + cV_1V_2[\cos(\omega-p)t - \cos(\omega+p)t]$$

or

$$i = a + bV_1 \sin \omega t + bV_2 \sin pt + \frac{1}{2}cV_1^2 - \frac{1}{2}cV_1^2 \cos 2\omega t + \frac{1}{2}cV_2^2 - \frac{1}{2}cV_2^2 \cos 2pt + cV_1V_2 \cos(\omega-p)t - cV_1V_2 \cos(\omega+p)t$$

or

$$i = \underbrace{\left(a + \frac{1}{2}cV_1^2 + \frac{1}{2}cV_2^2\right)}_{(1)} + \underbrace{bV_1 \sin \omega t}_{(2)} + \underbrace{bV_2 \sin pt}_{(3)} - \underbrace{\left(\frac{1}{2}cV_1^2 \cos 2\omega t + \frac{1}{2}cV_2^2 \cos 2pt\right)}_{(4)} + \underbrace{cV_1V_2 \cos(\omega-p)t}_{(5)} - \underbrace{cV_1V_2 \cos(\omega+p)t}_{(6)} \longrightarrow (VI)$$

→ In equⁿ (VI), if ω is assumed as the Carrier frequency and p is assumed as modulating frequency then we can identify the different terms as follow :

term (1) is the d.c. Component.
term (2) is the carrier Signal.
term (3) is the modulating signal.
term (4) contains harmonics of the carrier signal and the modulating signal.
term (5) is the lower sideband
term (6) is the upper sideband.

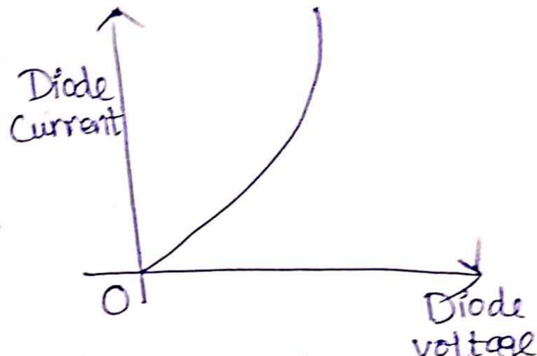
→ Since, the process of amplitude modulation consists of carrier signal, lower sideband and upper sideband; therefore we can conclude from above that when two voltages of different frequencies are passed through a non-linear resistance, the Amplitude Modulation takes place.

Square Law Diode Modulation :–

1. Definition –

Square law diode modulation circuit make use of nonlinear current-voltage characteristics of diode.

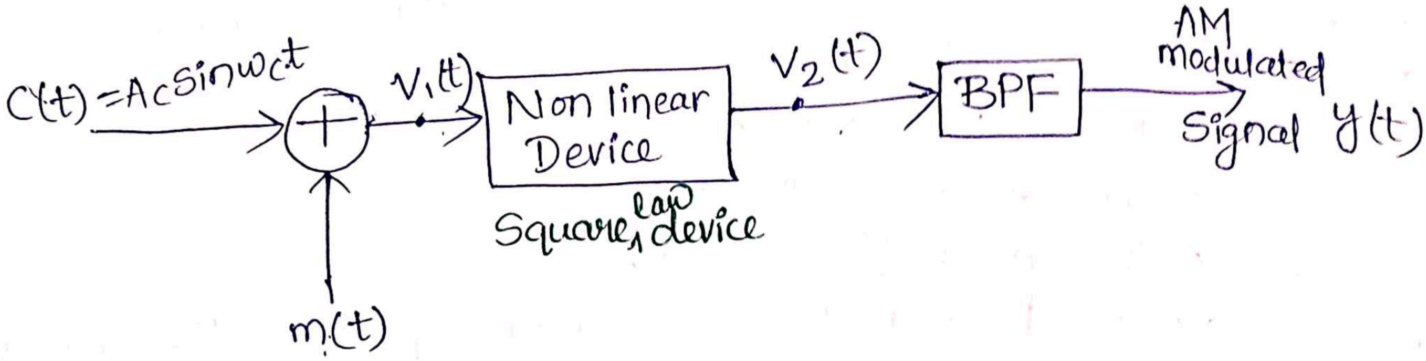
→ This method is suited at low voltage levels because of the fact that current-voltage characteristics of a diode is highly non linear particularly in the low voltage region.



[Non-linear resistance characteristics]

2. Basic Operation –

When two voltages of different frequencies are passed through a non-linear resistance, the amplitude Modulation takes place. So in a practical modulating circuit, the frequencies other than carrier & two sideband frequencies are rejected with the help of a tuned circuit.



→ Square law modulator has three major parts.

1) Adder
2) Non linear device
3) Band Pass filter

→ After Adder, $V_1(t) = C(t) + m(t)$

$$\boxed{V_1(t) = A_c \sin \omega_c t + m(t)}$$

→ After passing through Non-linear device,

$V_2(t) = qV_1(t) + bV_1^2(t)$ where, q & b are Constant.

$$= q(A_c \sin \omega_c t + m(t)) + b(A_c \sin \omega_c t + m(t))^2$$

$$= qA_c \sin \omega_c t + q\cdot m(t) + bA_c^2 \sin^2 \omega_c t + bm^2(t) + 2bA_c(\sin \omega_c t)m(t)$$

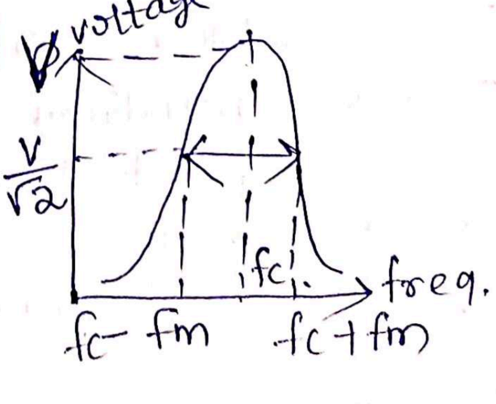
$$= \underbrace{\boxed{qm(t) + bA_c^2 \sin^2 \omega_c t + bm^2(t)}}_{\text{unuseful components}} + \underbrace{\boxed{qA_c \sin \omega_c t + 2bA_c m(t) \sin \omega_c t}}_{\text{useful components}}$$

→ Passed the useful Components in to the BPF

→ After passing through BPF,

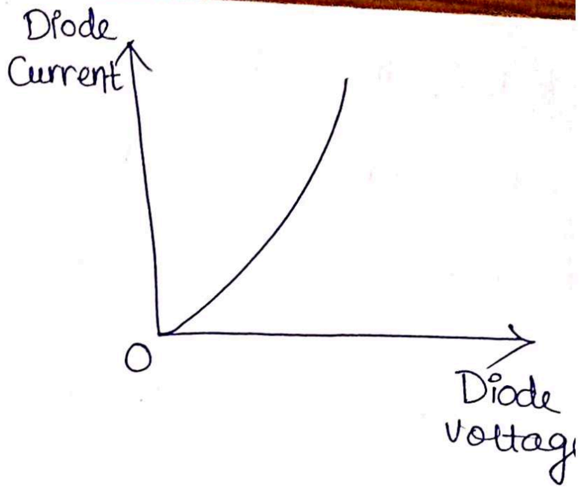
$$y(t) = qA_c \sin \omega_c t + 2bA_c m(t) \sin \omega_c t$$

$$\Rightarrow \boxed{y(t) = qA_c\left(1 + \frac{2b}{q} m(t)\right) \sin \omega_c t}$$

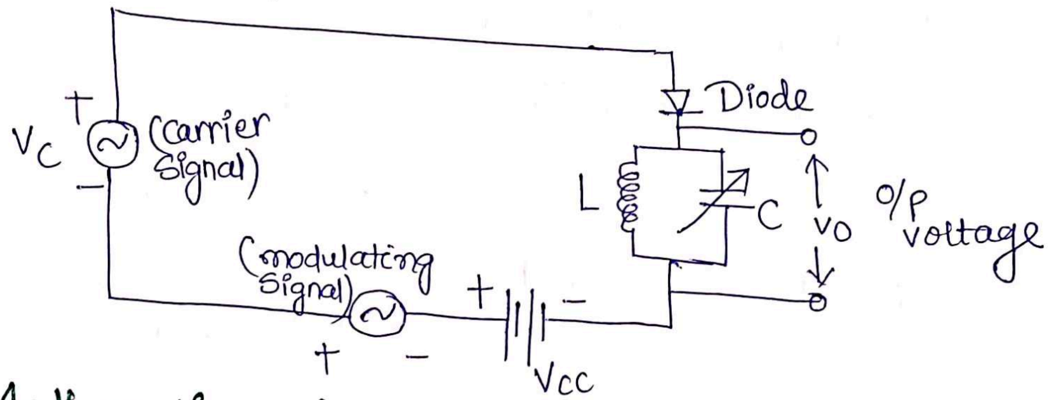


→ In Square law diode modulation, We make use of non-linear $i-v$ characteristic of diode.

Since we know, that $i-v$ characteristic of diode is highly non-linear at low voltages particularly.



Circuit Diagram :—



Mathematical Analysis :—

→ Let us consider that Carrier voltage is expressed as,

$$v_c = V_c \cos \omega_c t \quad \text{———} \quad (i)$$

where, ω_c is the Carrier frequency.

→ Let the modulating voltage be expressed as,

$$v_m = V_m \cos \omega_m t \quad \text{———} \quad (ii)$$

where, ω_m is the modulating frequency.

→ Total AC voltage across diode is

$$v_s = v_c + v_m \quad \text{———} \quad (iii)$$

$$v_s = V_c \cos \omega_c t + V_m \cos \omega_m t \quad \text{———} \quad (iv)$$

→ $v-i$ relationship for non-linear diode is

$$i = a + b v_s + c v_s^2 \quad \text{———} \quad (v)$$

Where, a, b & c are constant.

i = current through the diode

v_s = voltage across the diode

Putting the value of v_s in equ (..)

$$i = a + b(V_c\cos\omega_c t + V_m\cos\omega_m t) + c(V_c\cos\omega_c t + V_m\cos\omega_m t)^2$$

$$\Rightarrow i = a + bV_c\cos\omega_c t + bV_m\cos\omega_m t + c(V_c^2\cos^2\omega_c t + V_m^2\cos^2\omega_m t + 2V_cV_m\cos\omega_c t\cdot\cos\omega_m t)$$

$$\Rightarrow i = a + bV_c\cos\omega_c t + bV_m\cos\omega_m t + cV_c^2\cos^2\omega_c t + cV_m^2\cos^2\omega_m t + 2cV_cV_m\cos\omega_c t\cdot\cos\omega_m t$$

$$\Rightarrow i = a + bV_c\cos\omega_c t + bV_m\cos\omega_m t + \frac{1}{2}cV_c^2(2\cos^2\omega_c t) + \frac{1}{2}cV_m^2(2\cos^2\omega_m t) + cV_cV_m(2\cos\omega_c t\cos\omega_m t)$$

$$\Rightarrow i = a + bV_c\cos\omega_c t + bV_m\cos\omega_m t + \frac{1}{2}cV_c^2(1+\cos 2\omega_c t) + \frac{1}{2}cV_m^2(1+\cos 2\omega_m t) + cV_cV_m[\cos(\omega_c+\omega_m)t + \cos(\omega_c-\omega_m)t]$$

$$\left[\because \cos 2\theta = 2\cos^2\theta - 1 \Rightarrow 2\cos^2\theta = (1+\cos 2\theta) \quad \& \quad 2\cos A\cos B = \cos(A+B) + \cos(A-B)\right]$$

$$\Rightarrow i = a + bV_c\cos\omega_c t + bV_m\cos\omega_m t + \frac{1}{2}cV_c^2 + \frac{1}{2}cV_c^2\cos 2\omega_c t + \frac{1}{2}cV_m^2 + \frac{1}{2}cV_m^2\cos 2\omega_m t + cV_cV_m\cos(\omega_c+\omega_m)t + cV_cV_m\cos(\omega_c-\omega_m)t$$

$$\Rightarrow i = \underbrace{\left(a + \frac{1}{2}cV_c^2 + \frac{1}{2}cV_m^2\right)}_{(1)} + \underbrace{bV_c\cos\omega_c t}_{(2)} + \underbrace{bV_m\cos\omega_m t}_{(3)} + \underbrace{\left(\frac{1}{2}cV_c^2\cos 2\omega_c t + \frac{1}{2}cV_m^2\cos 2\omega_m t\right)}_{(4)} + \underbrace{c\cdot V_cV_m\cos(\omega_c+\omega_m)t}_{(5)} + \underbrace{cV_cV_m\cos(\omega_c-\omega_m)t}_{(6)} \quad \text{—— (VI)}$$

Equ^n (VI), Consists of six terms as follows

term (1) is the d.c. term

term (2) is the Carrier Signal

term (3) is the modulating signal

term (4) Consists of harmonics of carrier and modulating signal.

term(5) represents the upper sideband.
term (6) represents the lower sideband.

→ The load impedance used in the modulator circuit i.e. Tank circuit is tuned to the carrier freq. ω_c & responds to narrowband frequencies centred about ω_c.

So, only frequencies ω_c, $(\omega_c+\omega_m)$ & $(\omega_c-\omega_m)$ are retained & others are rejected.

∴ O/P current would be –

$$i_0 = bV_c\cos\omega_c t + cV_cV_m\cos(\omega_c+\omega_m)t + cV_cV_m\cos(\omega_c-\omega_m)t$$

$$= bV_c\cos\omega_c t + cV_cV_m[\cos(\omega_c+\omega_m)t + \cos(\omega_c-\omega_m)t]$$

$$= bV_c\cos\omega_c t + 2cV_cV_m\cos\omega_c t\cos\omega_m t$$

$$\cancel{= bV_c\cos\omega_c t + 2cV_cV_m}$$

$$= bV_c\left(1+\frac{2cV_m}{b}\cos\omega_m t\right)\cos\omega_c t$$

$$\Rightarrow i_0 = bV_c(1+\mu\cdot\cos\omega_m t)\cos\omega_c t \quad \text{——— (VII)}$$

Where, $\mu = \frac{2cV_m}{b}$ is the modulation index.

Equ^n (VII) is the required expression for AM current.

Demodulation of AM Waves :—

1. Definition –

→ The process of extracting a modulating or baseband signal from the modulated signal is called demodulation or detection.

→ In other words, demodulation or detection is the process by which the message is recovered from the modulated signal at receiver.

→ The devices used for demodulation or detection are called demodulators or detectors.

2. Types —

For amplitude modulation, detectors or demodulators are categorized as :

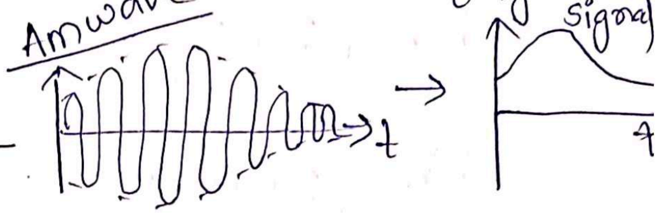
(i) Square-Law detectors
(ii) Envelope detectors
(iii) PLL (Phase Lock Loop)

→ AM signal with large carrier are detected by using the envelope detector.

→ The envelope detector uses the circuit which extracts the envelope of the AM wave.

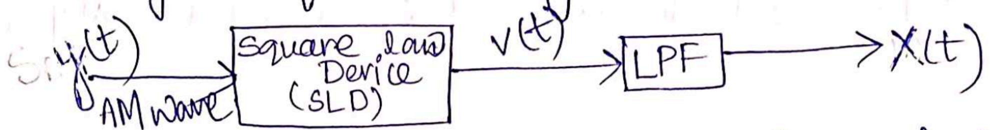
→ The envelope of the AM wave is the baseband or modulating signal. But a low-level amplitude modulated signal can only be detected by using square-law detectors in which a device operating in the non-linear region is used to detect the modulating signal.

Square-Law Detector :—

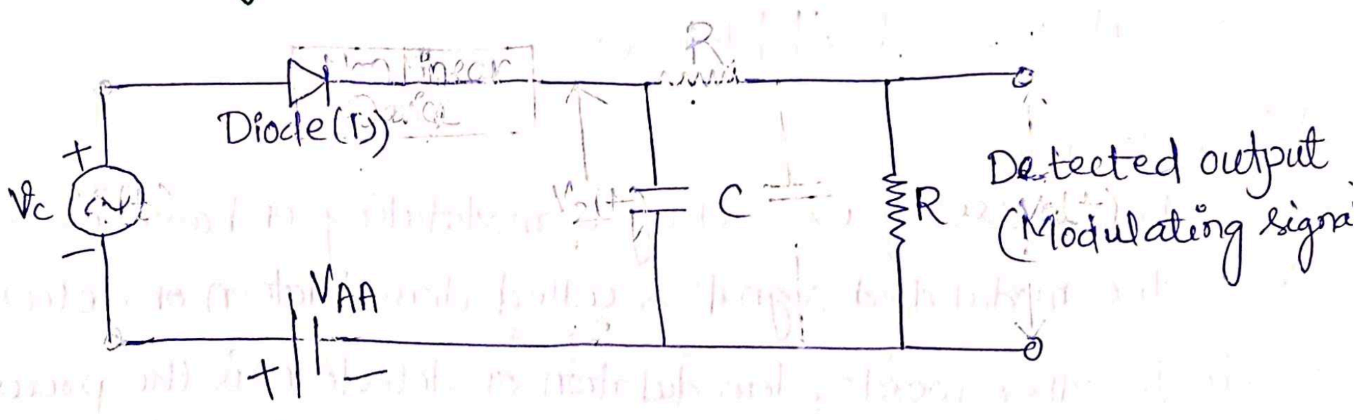


1. Definition —

The square law detector circuit is used for detecting modulated signal of small magnitude (i.e. below 1 volt)



2. Circuit Diagram — [Block diagram of Square law demodulator]



[Basic Circuit of Square law diode detector]

The above figure shows the Circuit of a Square-law detector.

→ It may be observed that the Circuit is very Similar to the Square-law modulator.

→ The only difference lies in the filter Circuit. In a Square law modulator, the filter used is a band-pass filter whereas in a Square law detector, a low-pass filter is used.

⇒ We Know that AM wave is expressed as

$$y(t) = A_c(1+m\,x(t))\sin\omega_c t$$

→ ~~If~~ When we pass the AM Signal through Square law device that is having characteristics.

→ $v(t)$ is o/p of Square law device

$$v(t) = a\,y(t) + b\,y^2(t)$$

$$v(t) = a(A_c(1+m\,x(t))\sin\omega_c t) + b(A_c(1+m\,X(t))\sin\omega_c t)^2$$

$$= aA_c\sin\omega_c t + aA_c m\,x(t)\sin\omega_c t + bA_c^2(1+m^2x^2(t)+2m\,x(t))\sin^2\omega_c t$$

$$= aA_c\sin\omega_c t + aA_c m\,x(t)\sin\omega_c t + bA_c^2\sin^2\omega_c t + bA_c^2m^2x^2(t)\sin^2\omega_c t + \cancel{2}\,mbA_c^2X(t)\left(\frac{1-\cos 2\omega_c t}{\cancel{2}}\right)$$

$$\left[\begin{array}{l} \because \sin^2 x = 1-\cos^2 x \\ \sin^2 x = \dfrac{1-\cos 2x}{2} \end{array}\right]$$

$$= aA_c\sin\omega_c t + aA_c m\,X(t)\sin\omega_c t + bA_c^2\sin^2\omega_c t + bA_c^2m^2x^2(t)\sin^2\omega_c t + \underline{mbA_c^2X(t)} - mbA_c^2x(t)\cos 2\omega_c t$$

(no Carrier only information is there)

→ After LPF,

message Signal o/p will be ⟶ $mbA_c^2x(t)$

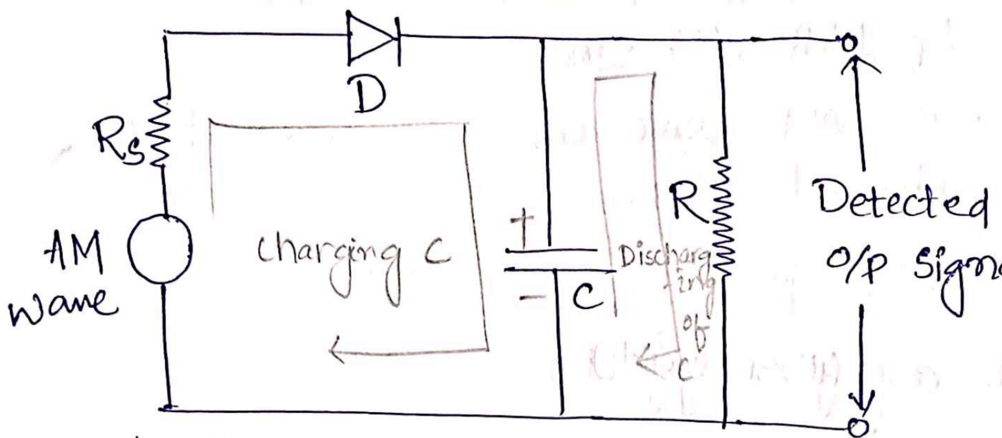
Linear Diode or Envelope Detector :—

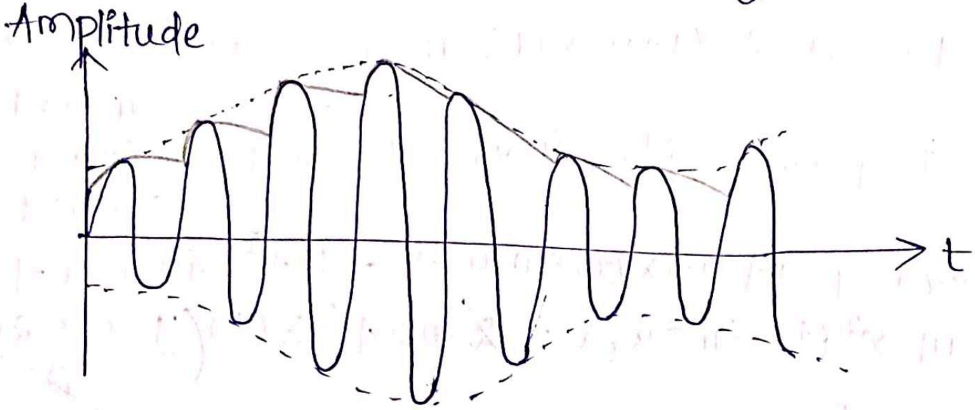
1. Definition :-

An envelope detector is an electronic circuit that takes a high-frequency amplitude modulated signal as input and provides an output, which is the demodulated envelope of the original signal.

→ Envelope detector is most popular in commercial & receiver circuits since it is very simple and is not expensive, also at the same-time, it gives satisfactory performance for the reception of broadcasting programmes.

2. Circuit Diagram :-





3)

→ During charging R_SC, (∵ R_SC = time Constant)

$$\boxed{R_SC << \frac{1}{f_c}}$$

∵ Here charging should be Very fast,
[$\frac{1}{f_c}$ = time period of one cycle for that carrier signal]

→ During Discharging RC, (discharging time period is very large)

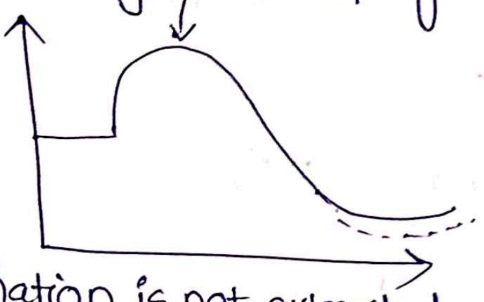
$$\boxed{\frac{1}{f_c} << RC < \frac{1}{f_m}}$$

(∵ Here RC = time Constant)

∴ Here discharging should be Very slow.

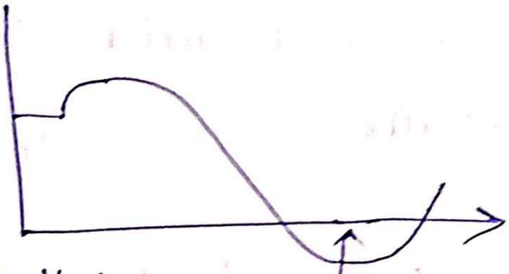
There is having two problems during envelope detection.

① Diagonal clipping



→ Information is not extracted

② Negative Cycle clipping



→ -Ve half cycle can not be detected as the original signal by using envelop detector.

3. Principle & Description :-

→ Envelope Detector having two parts. One is rectifier and another one is low pass filter.

→ Here the Diode is acting as a rectifier and the RC Circuit is acting as a low pass filter.

→ There are 3 components i.e. R_s, C & R, and these 3 components together act as an rectifier Circuit.

→ The Capacitor is gets charged through R_s resistance & Diode and the capacitor is discharged through R & C.

→ So the diode is clip -ve part of the AM signal and the charging operation is happen through R_s resistance and discharging is happen through R-C.

Other Types of AM :—

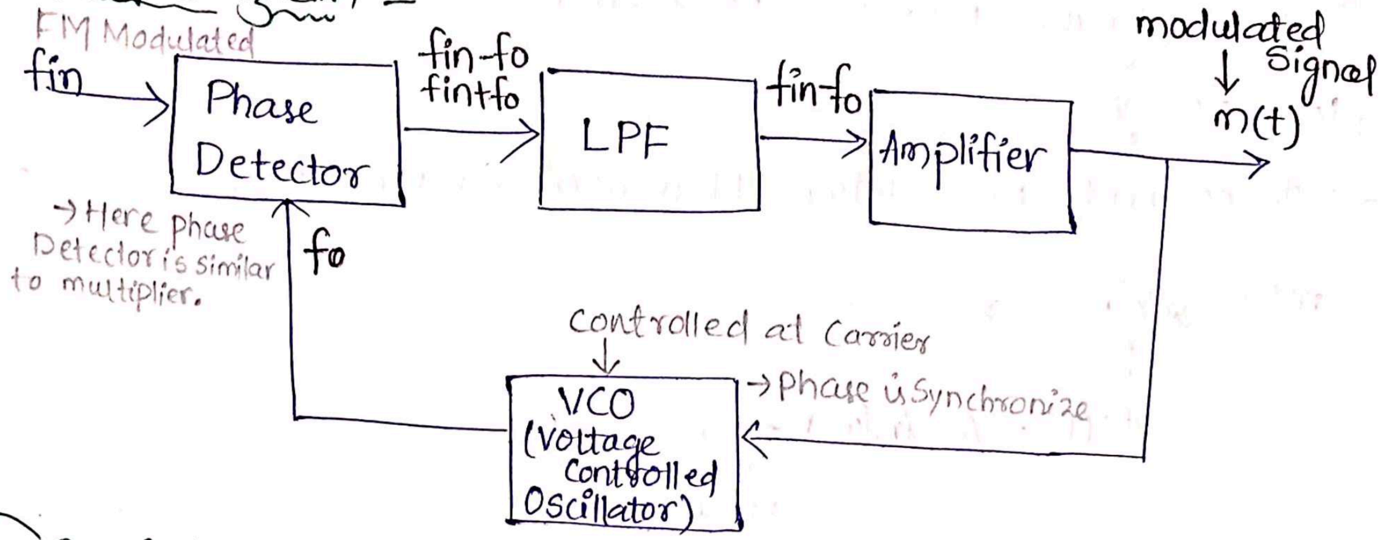
→ The Amplitude Modulation is also called as the Double side-band full carrier system (DSB-FC).

Amplitude Modulation (AM)

- DSB-FC
- DSB-SC (Double Sideband Suppressed Carrier System)
- SSB-SC (Single Sideband Suppressed Carrier System)
- ISB (Independent Sideband System)
- VSB (Vestigial Sideband System).

PLL (Phase Lock Loop): — (In Demodulation)

Block Diagram –



Description –

→ The o/p freq. of VCO is equal to the freq. of unmodulated carrier
→ The phase detector generates voltage proportional to difference between FM Signal & VCO o/p.
→ Then o/p of ~~of~~ phase detector passes through LPF & amplifier
→ Here freq. correction is not required at VCO as it is done at Tx.

Advantages :—

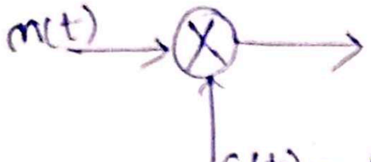
→ No need of tuned Circuit
→ Simple Circuit that Can be Implemented in Integrated Circuit.

Phase Lock Loop:-

→ A PLL is a non-Linear feedback system that tracles the phase of Input Signal & Minimized phase error at local Oscillator.

Why PLL?

→ In coherent demodulation PLL is used for removing phase error.



$$c(t) = A_c \sin(\omega_c t + \underbrace{\phi}_{\text{Phase error}})$$

Block diagram description:-

Phase Detector ⇒

- Compares fin and fo.
- The o/p of the phase detector is proportional to phase difference between fin and fo.
- O/p of phase detector is DC or min. freq. signal. So, it is reflected as the error voltage e.

LPF ⇒

- It removes high freq. noise.
- It produces DC signal.
- We can use active low pass filter or passive low pass filter.

VCO ⇒

- It generates high freq. signal.
- The Instantaneous VCO freq. is controlled by its input voltage.

$$f_o = f + \underbrace{(K)}_{\text{It is based on Vp voltage.}}\omega_m$$

- The freq. of VCO is directly Controlled by DC Vp voltage.

DSB-FC :— (Double Sideband Full Carrier System)

→ In DSB-FC, the carrier signal does not Convey any information. The information is contained in the two sidebands only. But, the sidebands are images of each other and hence both of them contain the same information. Thus, all the information can be conveyed by only one sideband.

→ As we know that the total power transmitted by an AM wave is given by —

$$P_t = P_c + P_{USB} + P_{LSB}$$

$$(\text{or})\quad P_t = P_c + P_c\frac{\mu^2}{4} + P_c\frac{\mu^2}{4}$$

→ Out of the three terms in the above equ^n, the carrier component does not contain any information and one sideband is redundant.

→ Hence, out of the total power $P_t = \left[P_c\left(1+\frac{\mu^2}{2}\right)\right]$ the wasted power is given by;

$$\text{Power Wastage} = P_c + \frac{\mu^2}{2}P_c = P_c\left[1+\frac{\mu^2}{2}\right]$$

Imp. Point :—

This shows that we have to transmit much higher power than what is actually required. Hence, DSB-FC System is a power inefficient System.

Bandwidth Requirement of DSB-FC ⇒

→ The bandwidth (BW) of DSB-FC System is $2f_m$.

→ This is due to the Simultaneous transmission of both the Sidebands out of which only one is Sufficient to convey all the information. Thus, the BW of DSB-FC is double than actually required. Therefore, DSB-FC is a bandwidth inefficient System.

→ Due to these reasons, it was thought that if only one sideband is transmitted by Suppressing the Carrier and another Sideband then a lot of power can be saved and BW also can be reduced.

→ The next step is a System called Double Sideband Suppressed Carrier (DSB-SC) System.

DSB-SC :— (Double Sideband Suppressed Carrier System)

→ It is sometimes called as DSB-AM System.

* **AM Signal** →

$$\Rightarrow y(t) = \underbrace{A_c \cos \omega_c t}_{\text{Carrier}} + \underbrace{\frac{A_c \mu}{2} \cos(\omega_c + \omega_m)t}_{\text{USB}} + \underbrace{\frac{A_c \mu}{2} \cos(\omega_c - \omega_m)t}_{\text{LSB}}$$

$$\Rightarrow y(t) = A_c(1 + \mu x(t)) \cos \omega_c t$$

$$\Rightarrow y(t) = \underbrace{A_c \cos \omega_c t}_{\text{Carrier}} + \underbrace{\mu A_c x(t) \cos \omega_c t}_{\text{Sidebands}}$$

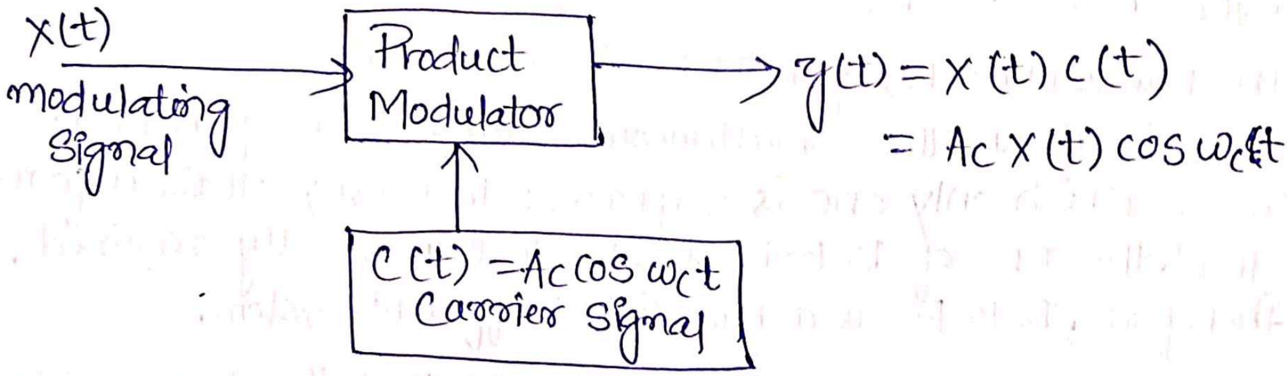
* **In power transmission** ⟹

$$\Rightarrow P_t = P_c + P_S$$
$$= P_c + \frac{\mu^2}{2} P_c$$

* **for $\mu = 1$** ⟹

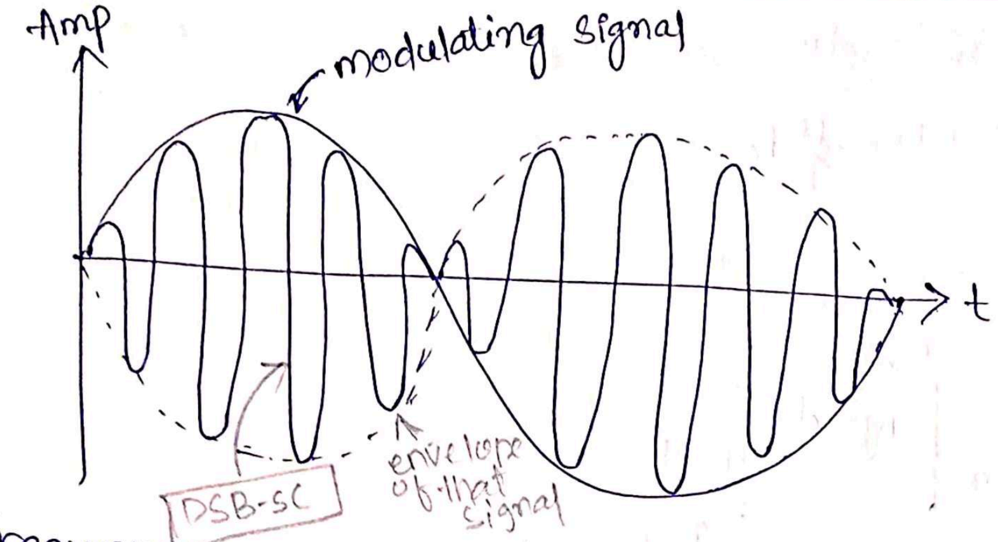
$$\rightarrow P_t = P_c + 0.5 P_c$$

BLOCK DIAGRAM —

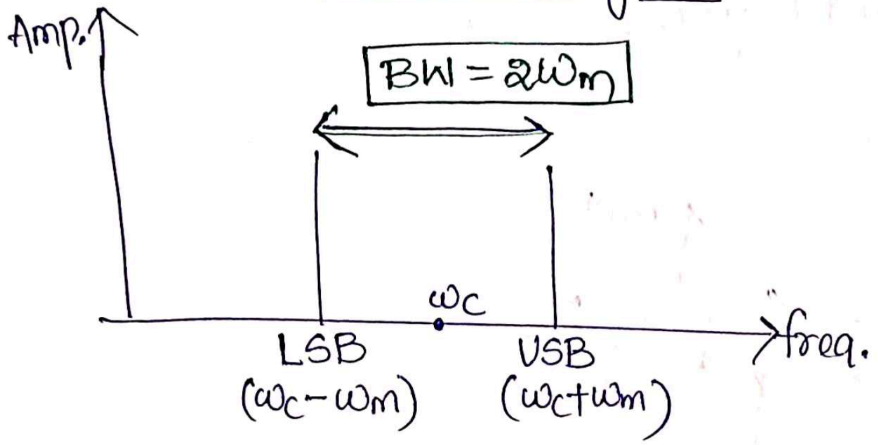


→ In DSB-SC

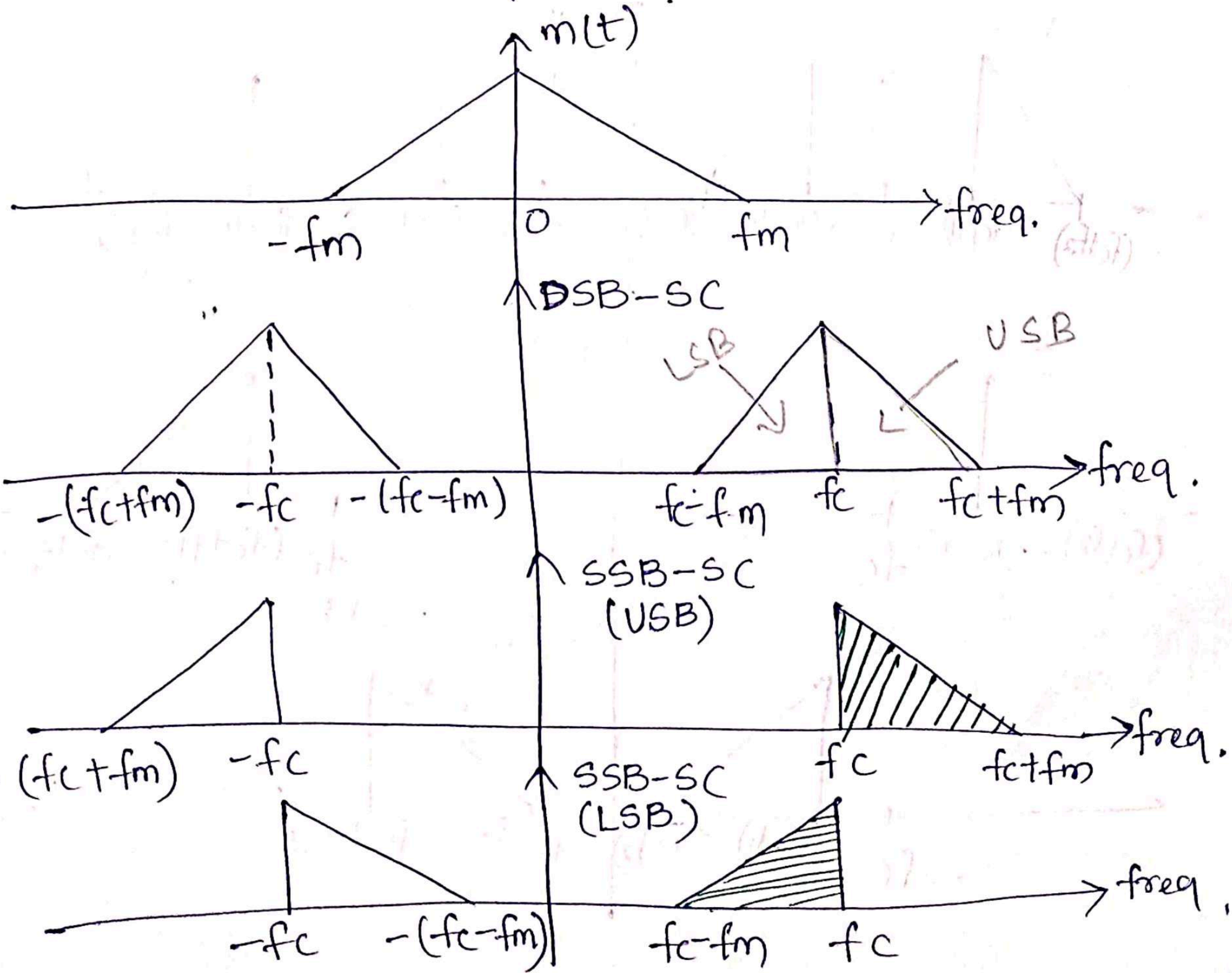
- We don't send carrier signal.
- Only LSB & USB signal is there.
- It has 180° phase reversal at zero crossing of modulating signal.



Frequency response of DSB-SC Signal ⟹



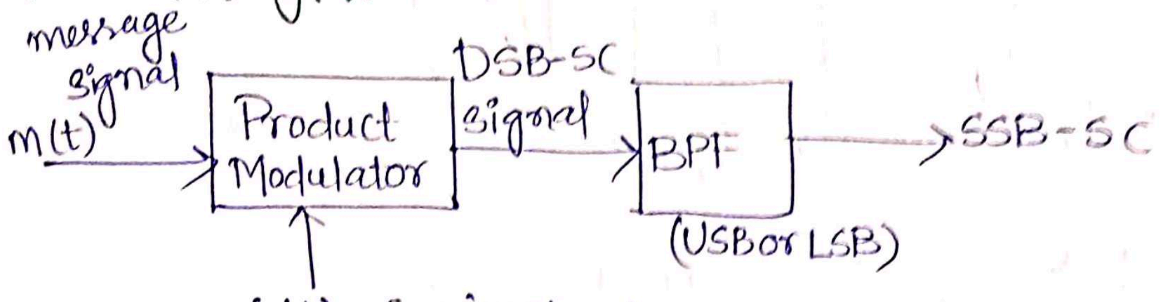
SSB-SC :— (Single sideband Suppressed Carrier System)



→ BW of SSB-SC = f_m

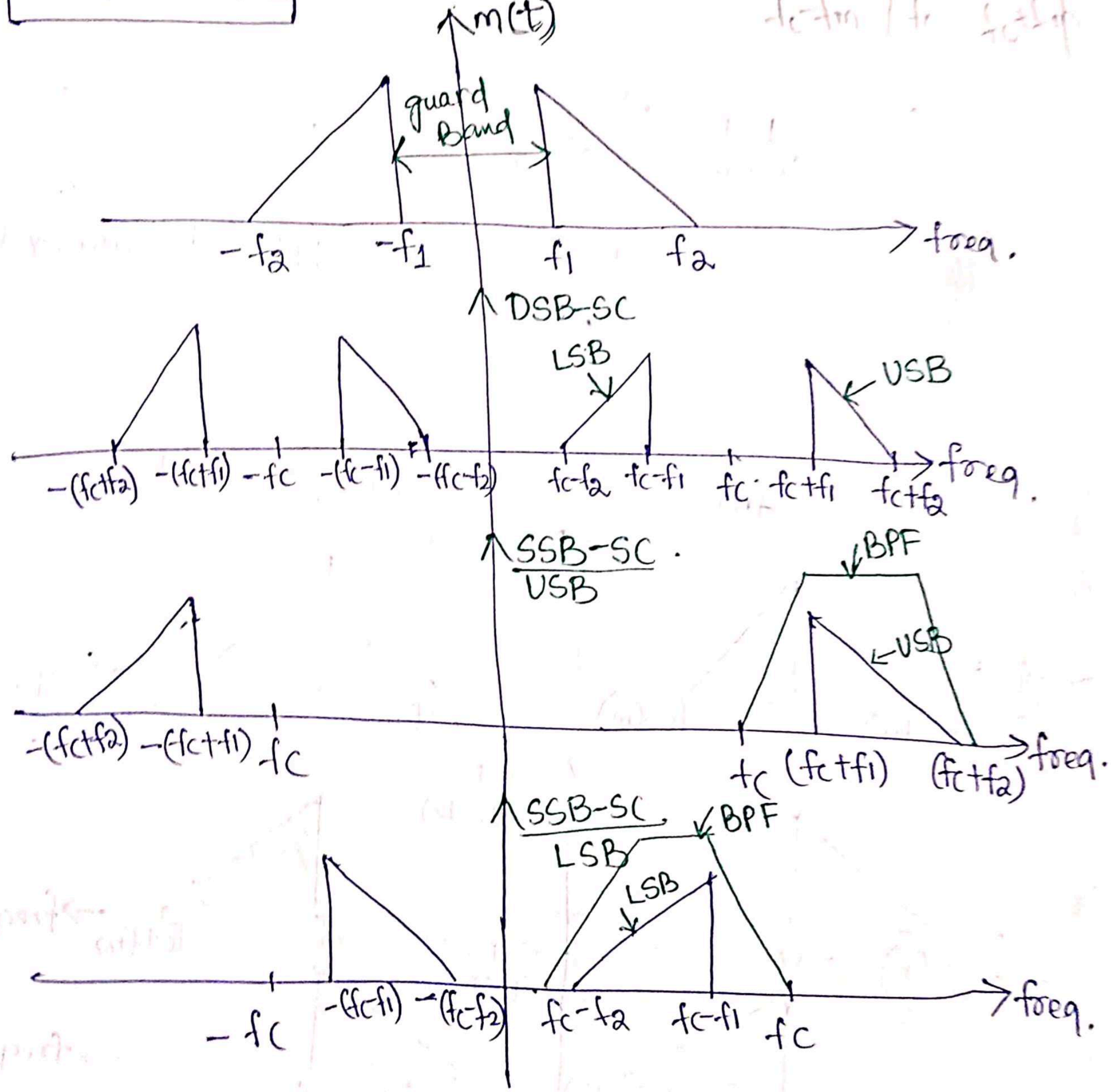
→ Power transmission = $\frac{\mu^2}{4} P_c$

Block Diagram –



Draw Backs –

→ Voice signal
300Hz – 3.4KHz

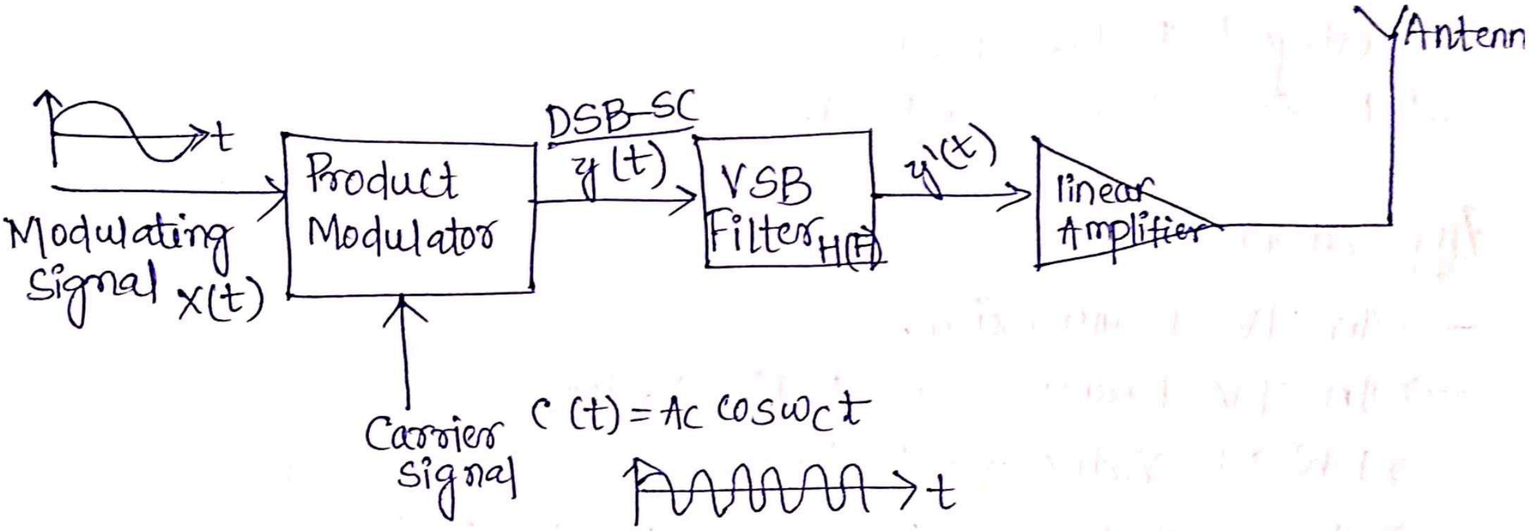


→ Q-factor (1000 to 2000).

VSB :— (Vestigial sideband System)

Generation —

$\boxed{\text{Block diagram}}$ —



→ O/p of Product Modulator

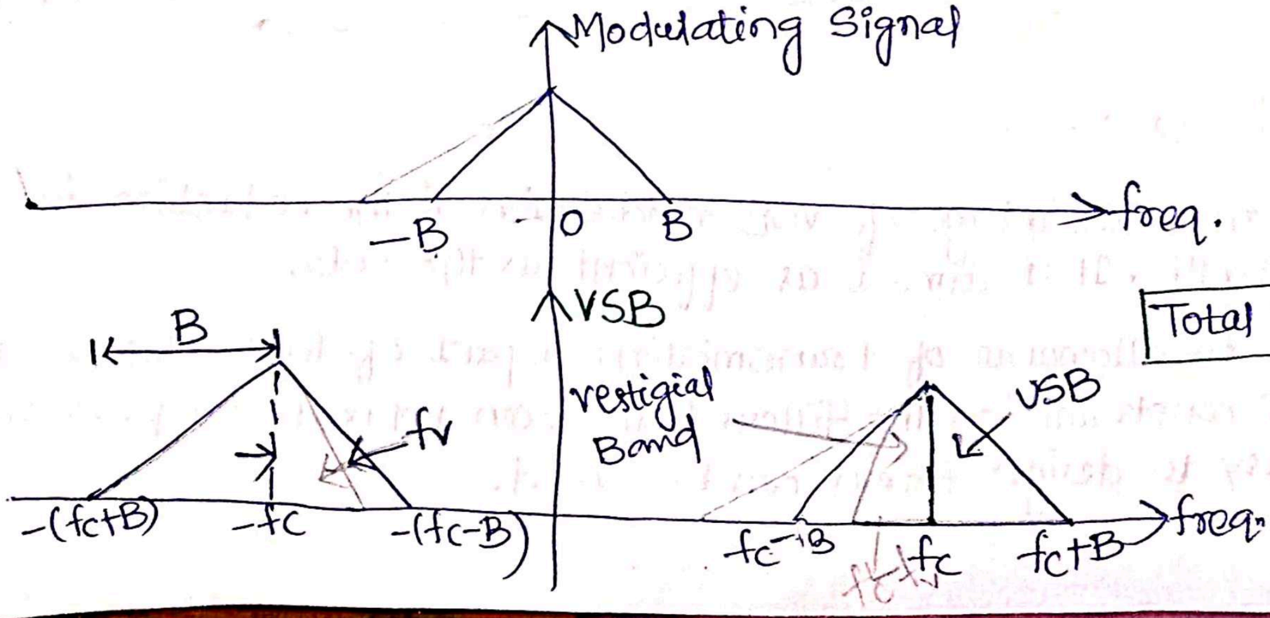
$\Rightarrow y(t) = x(t)\,c(t)$

$\Rightarrow y(F) = \frac{A_c}{2}\left[X(f_c - f_0) + X(f_c + f_0)\right]$

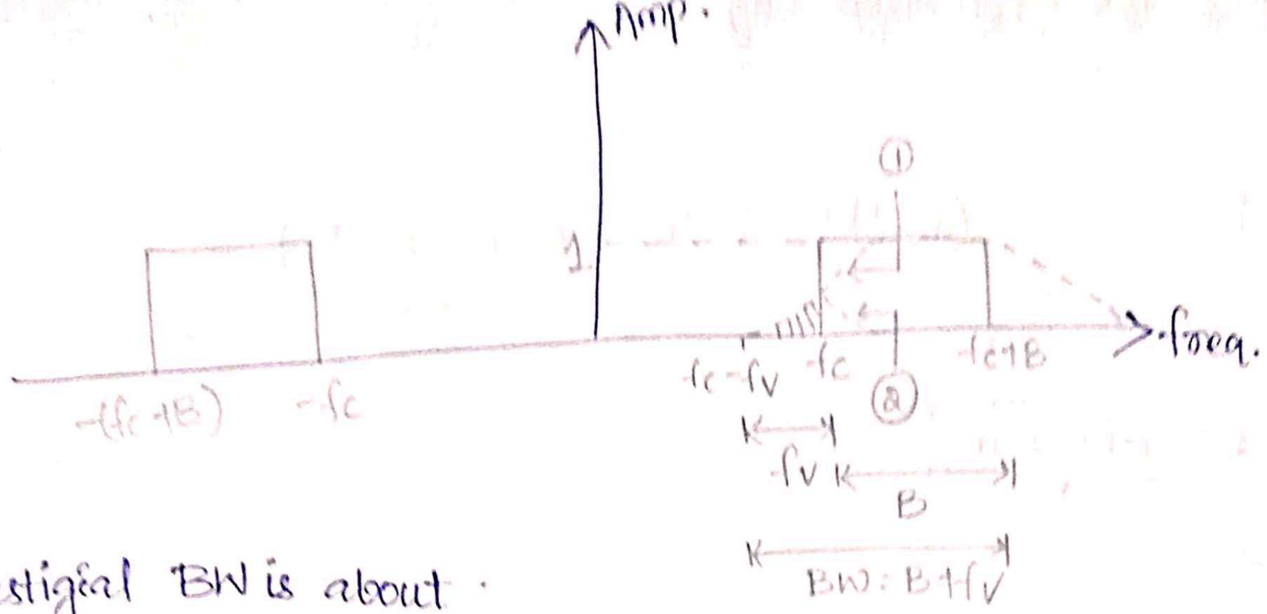
LSB freq. Component | VSB freq. component

→ O/p of VSB Filter

$\Rightarrow y'(F) = y(F)\,H(F) = \frac{A_c}{2}\left[X(f_c - f_0) + X(f_c + f_0)\right] H(F)$



$\boxed{\text{Total BW} = B + f_v}$



→ Vestigial BW is about 25 to 33% of one Side Band.

Application :—

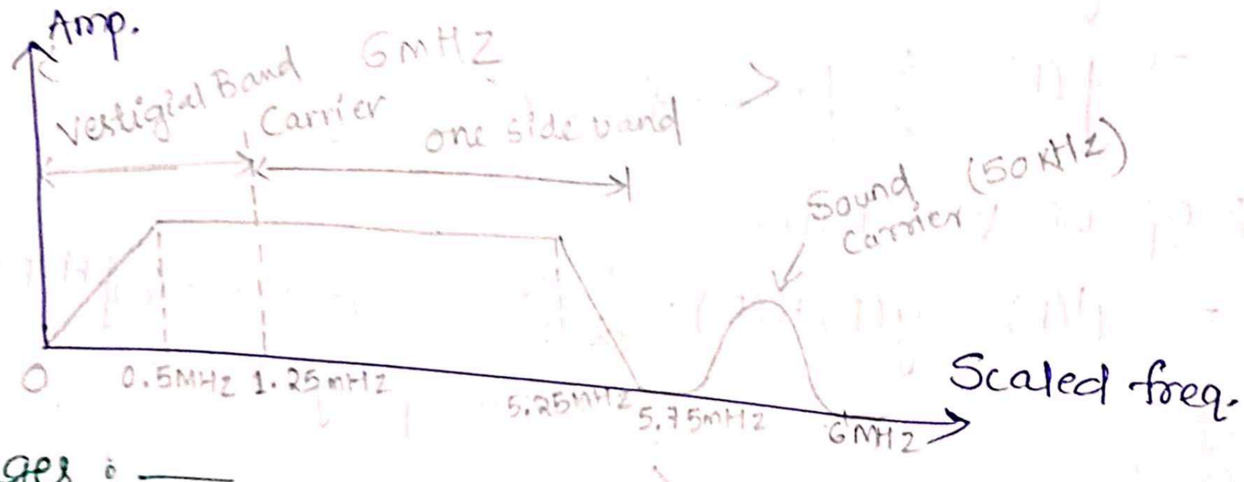
→ In TV transmission.

→ In TV transmission = Audio + Video.

→ BW of Video = 4.2 MHz

→ By DSB-FC = 2(4.2) + guard Band + Audio
= 9 MHz

→ By VSB = 6 MHz



Advantages :—

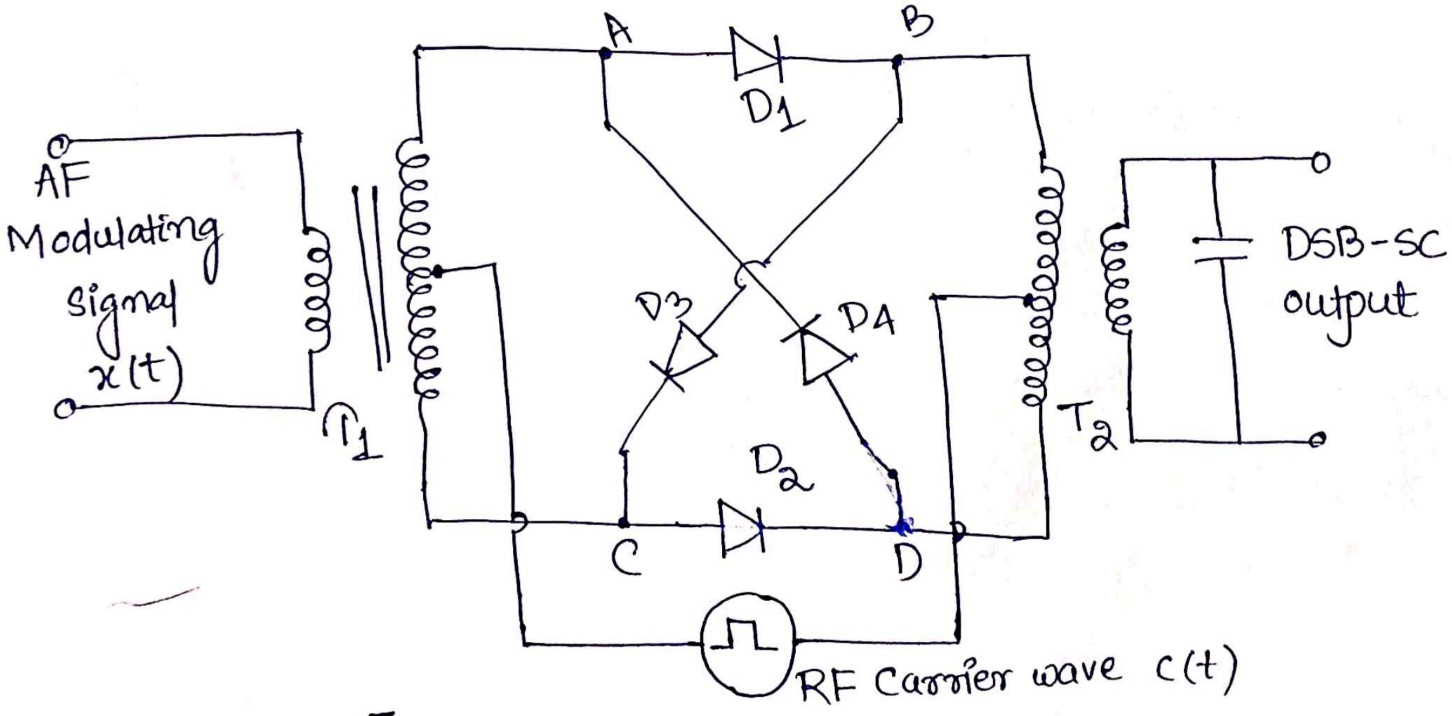
(i) The main advantage of VSB modulation is the reduction in bandwidth. It is almost as efficient as the SSB.

(ii) Due to allowance of transmitting a part of lower sideband, the constraint on the filters have been relaxed. So practically, easy to design filters can be used.

(iii) It possesses good phase characteristics and makes the transmission of low frequency components possible.

Sl. NO.	Parameter of Comparison	DSBFC (Standard AM)	DSBSC	SSB	VSB
(i)	Carrier Suppression	N.A.	fully	fully	N.A.
(ii)	Sideband Suppression	N.A.	N.A.	One S.B. Completely	One S.B. Suppresse, Partially
(iii)	Bandwidth	$2f_m$	$2f_m$	f_m	$f_m < BW < 2f$
(iv)	Transmission efficiency	Minimum	Moderate	Maximum	Moderate
(v)	No of modulating inputs	1	1	1	2
(vi)	Application	Radio broadcasting	Radio broadcasting	Point-to Point mobile Communication	T.V.

Ring Modulator for the DSB-SC Generation :—



[A diode ring Modulator]

→ The above fig. shows the circuit diagram of a diode ring modulator.

→ It consists of four diodes, an audio frequency transformer T_1 and an RF transformer T_2.

→ The carrier signal is assumed to be a square wave with frequency f_c and it is connected between the centre taps of the two transformers.

→ The DSB-SC output is obtained at the secondary of the RF transformer T_2.

→ The above fig. shows the circuit diagram of a diode ring modulator.

→ It consists of four diodes, an audio frequency transformer T_1 and an RF transformer T_2.

→ The carrier signal is assumed to be a square wave with frequency f_c and it is connected between the centre taps of the two transformers.

→ The DSB-SC output is obtained at the secondary of the RF transformer T_2.

Working Operation :—

→ The operation of the ring modulator is explained with the assumptions that the diodes act as perfect switches and that they are switched ON and OFF by the RF carrier signal.

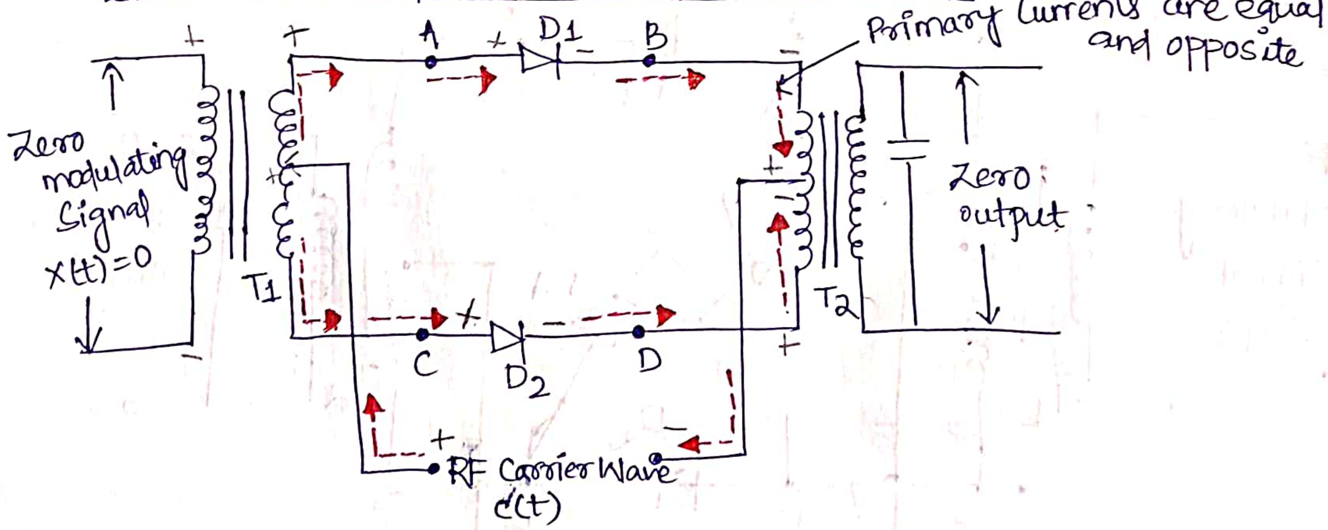
→ This is because the amplitude and frequency of the carrier is higher than that of the modulating signal.

→ The operation can be divided into different modes without the modulating signal and with the modulating signal as follows:

Mode 1 : Carrier Suppression ⇒

To understand how carrier suppression takes place, let us assume that the modulating signal is absent and only the carrier signal is applied. Hence $x(t) = 0$

(i) Operation in the positive half-cycle of carrier —



[Equivalent Circuit of in Mode 1] fig.2(a)

→ As shown in the above fig, the diodes D1 and D2 are forward biased and the diodes D3 and D4 are reverse biased.

→ We can observe that the direction of currents flowing through the Primary windings of output transformer T2 are equal & opposite to each other.

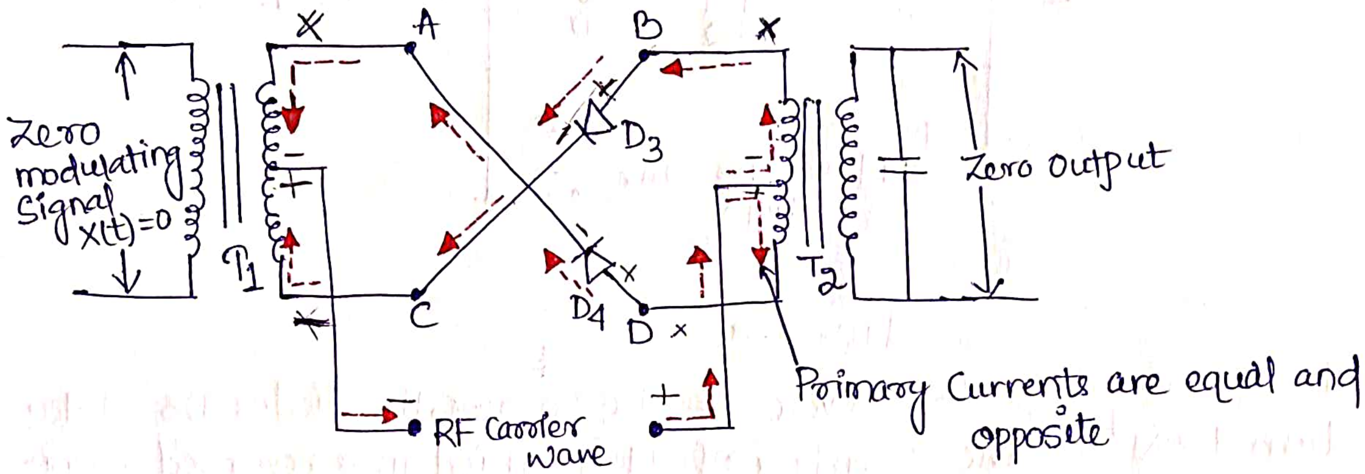
→ Therefore, the magnetic fields produced by these currents are equal & opposite and cancel each other.

→ Hence, the induced voltage in secondary winding is zero. Thus, the Carrier is supported in the positive half-cycle.

(ii) $\boxed{\text{Operation in the Negative half-Cycle of Carrier}}$ –

→ In this mode also let us assume that the modulating Signal is zero.

→ In the negative half-cycle of the Carrier, the diodes D3 and D4 are forward biased and the diodes D1 and D2 are reverse biased.



[Equivalent Circuit of in Mode 1]

fig. 2(b)

→ In fig. 3, the currents flowing in the upper and lower halves of the Primary Winding of T2 are again equal and in opposite directions.

→ This cancels the magnetic fields as explained in mode 1 (i). Thus, the output voltage in this mode also is zero.

→ Thus, the Carrier is Suppressed in the negative half-cycle as well.

→ It is important to note that the perfect cancellation of the Carrier will take place if and only if he characteristics of the diodes are Perfectly matched and the Centre tap is placed exactly at the Centre of the primary transformer T2.

Mode 2: Operation in Presence of Modulating Signal ⇒

In this, we discuss the operation - when RF Carrier and modulating signal both are applied.

(i) $\boxed{\text{Operation in the positive half-Cycle of Modulating Signal}}$ –

As we apply the low frequency modulating Signal through the input audio transformer T1, there are many cycles of the Carrier Signal, in the positive half Cycle of the modulating Signal.

→ In the positive half-cycle of the carrier, D1 and D2 are ON and secondary of T1 is applied as it is across the primary of T2.
→ Hence, during the positive half cycle of carrier, the output of T2 is positive as shown in fig.3 (a).

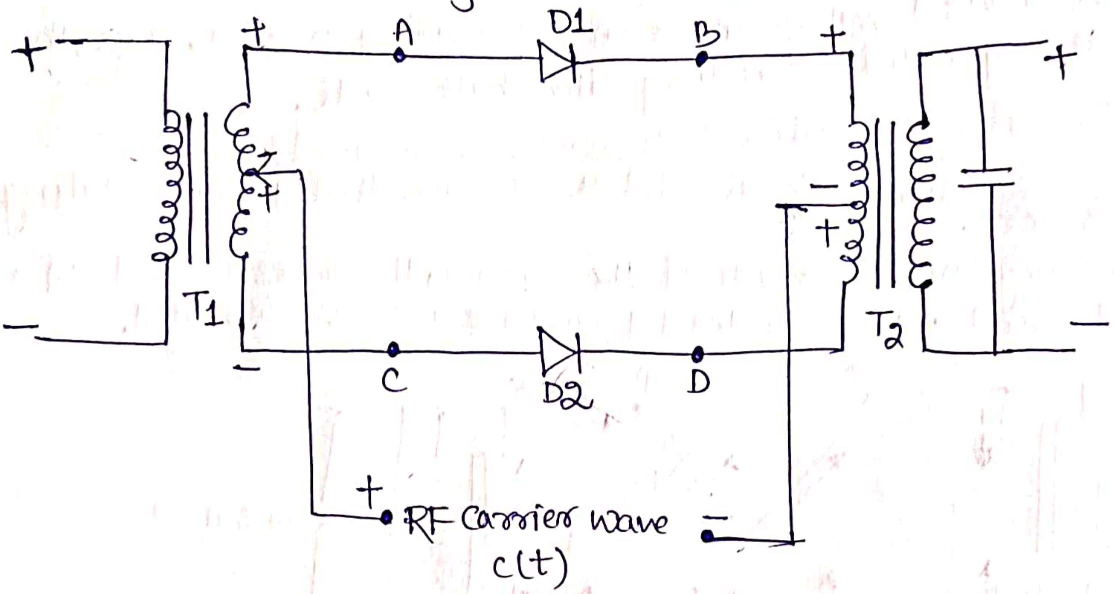


fig. 3 (a)

→ In the negative half-cycle of the carrier, the diodes D3 & D4 are turned ON and the secondary of T1 is applied in a reversed manner across the primary of T2 as shown in the equivalent ckt in fig 3(b).
→ Thus, the primary voltage of T2 is negative and output voltage also becomes negative.

(ii) Operation in the Negative half-cycle of Modulating Signal —

When modulating signal reverses the polarities, the operation of the circuit is same as that in the positive half-cycle discussed in the above.

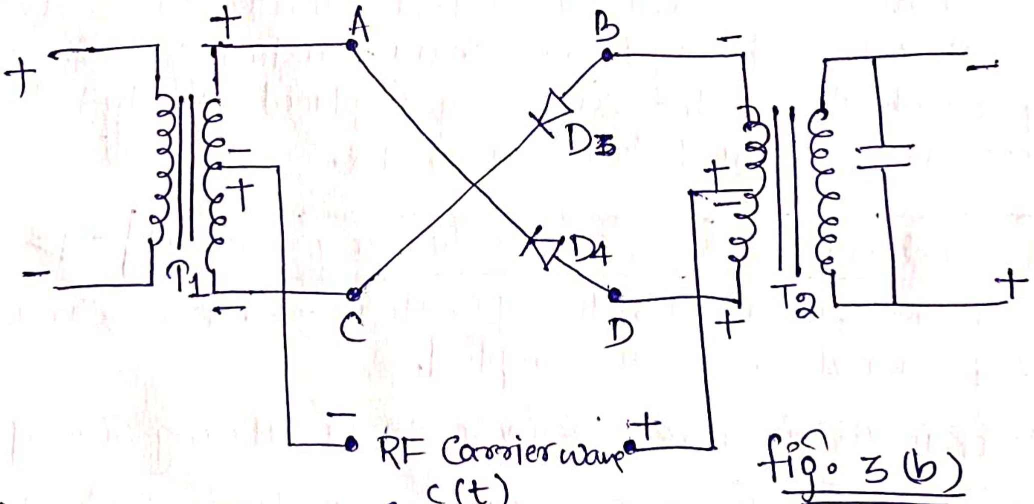


fig. 3 (b)

→ Now, the only difference is that the diode pair D3, D4 will produce a positive output voltage whereas D1, D2 will produce a negative output voltage as shown in the waveforms of fig. 4.

Ch. 3: Frequency Modulation Systems ①

3.1 Frequency modulation: Basic principle.

Both frequency modulation and phase modulation are coming under angle modulation. Frequency and phase angle are related as $\omega = \frac{\theta}{t}$ or $\theta = \omega t$ where θ is the phase angle (in radian), ω is the angular frequency ($= 2\pi f$) in rad/sec.

Frequency modulation is a process in which the frequency (and its rate of change) of the carrier is varied in accordance with the modulating signal.

or in other words: In frequency modulation, the frequency deviation (the amount by which the carrier frequency varies from its unmodulated value) is made proportional to the instantaneous amplitude of the modulating signal.

Let the modulating signal (base band or message) is $e_m(t) = E_m \sin(\omega_m t + \theta_m)$ —①

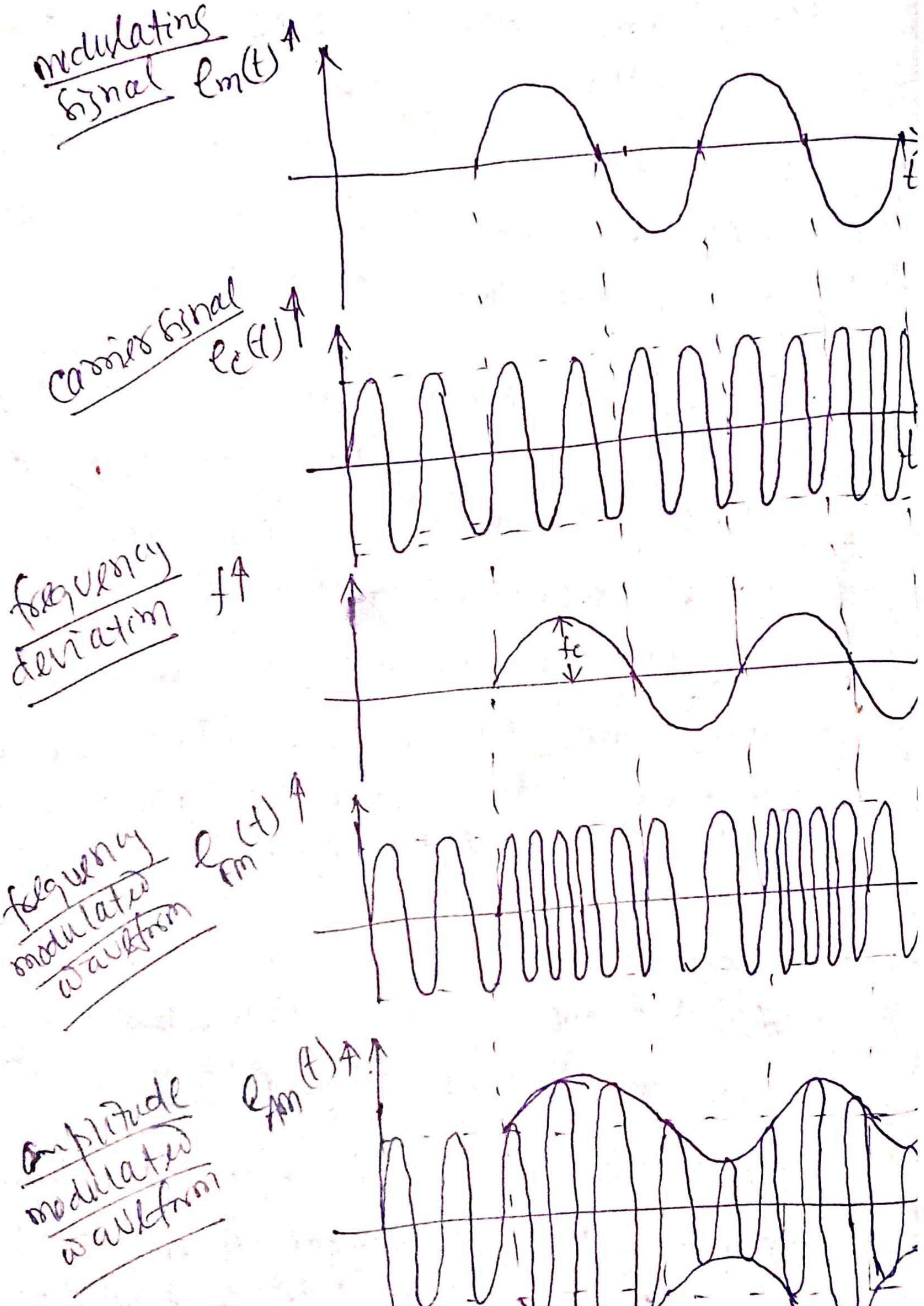
and the carrier signal is $e_c(t) = E_c \sin(\omega_c t + \theta_c)$ —②

where $E_m, E_c \rightarrow$ peak values of $e_m(t)$ and $e_c(t)$ respectively

$\theta_m, \theta_c \rightarrow$ phase angles of modulating signal and carrier signal respectively,

$\omega_m, \omega_c \rightarrow$ angular frequency of modulating and carrier signal respectively.

In frequency modulation, the frequency we q
carrier will vary according to the modulating
In the below figures are shown waveforms
FM and AM signals etc.



(3.2.) Expression for frequency modulated signal, modulation index and side bands.

Let the modulating signal be

$e_m(t) = E_m \sin \omega_m t = E_m \sin 2\pi f_m t$ — (1)

Carrier be $e_c(t) = E_c \sin \omega_c t = E_c \sin 2\pi f_c t$ — (2)

In frequency modulation, the frequency of the modulated signal will be a frequency which is its unmodulated frequency (f_c) plus the frequency deviation (that is proportional to modulating signal $e_m(t)$.

Let the freq. deviation is $\delta f = K_f\, e_m(t)$

or $\delta f = K_f E_m \sin \omega_m t$ — (3)

then frequency of the modulated waveform will be (say) $f = f_c + \delta f = f_c + K_f E_m \sin \omega_m t$ — (4)

where K_f is a proportionality constant (Hz/volt)

maximum deviation will occur when $\sin \omega_m t = \pm 1$. and so $f = f_c \pm K_f E_m$ — (5)

So that maximum deviation is $\delta f = K_f E_m$.

Thus, the FM signal will be given by —

$e_{Fm}(t) = E_c \sin[f(\omega_c, \omega_m)] = E_c \sin \theta$ — (6)

where $\theta = f(\omega_c, \omega_m)$, is a function of both carrier frequency and modulating signal frequency from eqn (4) $f = f_c + K_f E_m \sin\omega_m t$

or $\omega = \omega_c + 2\pi K_f E_m \sin\omega_m t$

as $\omega = \frac{d\theta}{dt}$ or $\theta = \int \omega . dt$

$= \int (\omega_c + 2\pi K_f E_m \sin\omega_m t)\, dt$

$= \omega_c t + \frac{2\pi K_f E_m \cos\omega_m t}{\omega_m}$

[putting $\omega_m = 2\pi f_m$] $= \omega_c t + \frac{K_f E_m \cos\omega_m t}{f_m}$

or $\boxed{\theta = \omega_c t + \frac{\delta f}{f_m} \cos\omega_m t}$

putting θ in (6), $\boxed{e_{fm}(t) = E_c \sin\left(\omega_c t + \frac{\delta f}{f_m} \cos\omega_m t\right)}$

$\frac{\delta f}{f_m} = m_f$ is the modulation index, then

$\boxed{e_{fm}(t) = E_c \sin(\omega_c t + m_f \cos\omega_m t)}$

FM Modulation index (m_f).

modulation index for FM, is defined as the ratio of maximum frequency deviation δf to the modulating signal freq. f_m.

$$\boxed{m_f = \frac{\text{max freq. deviation}}{\text{modulating freq}} = \frac{\delta f}{f_m}}$$

②

Sidebands in FM:

The equation of FM wave from is

$$e_{fm}(t) = E_c \sin(\omega_c t + m_f \cos\omega_m t)$$

which is a sine function of a sinusoidal function, which can be expanded by using Bessel's functions.

$$e_{fm}(t) = E_c \{ J_0(m_f) \sin\omega_c t$$
$$+ J_1(m_f)[\sin(\omega_c+\omega_m)t - \sin(\omega_c-\omega_m)t]$$
$$+ J_2(m_f)[\sin(\omega_c+2\omega_m)t - \sin(\omega_c-2\omega_m)t]$$
$$+ J_3(m_f)[\sin(\omega_c+3\omega_m)t - \sin(\omega_c-3\omega_m)t]$$
$$+ \cdots\cdots \}$$

The above equation shows that the frequency modulated wave form consists of a carrier and an infinite number of ~~set~~ pairs of side bands each preceded by J coefficients. These are called Bessel's functions.

3.3 Frequency spectrum of FM signals.

The analytical eqn. describing FM wave from is given by -

$$e_{fm}(t) = E_c \sin(\omega_c t + m_f \cos\omega_m t).$$

which is a sine function of the cosine function, which can be expanded by

From eqn ④ ,

Bessel's function as —

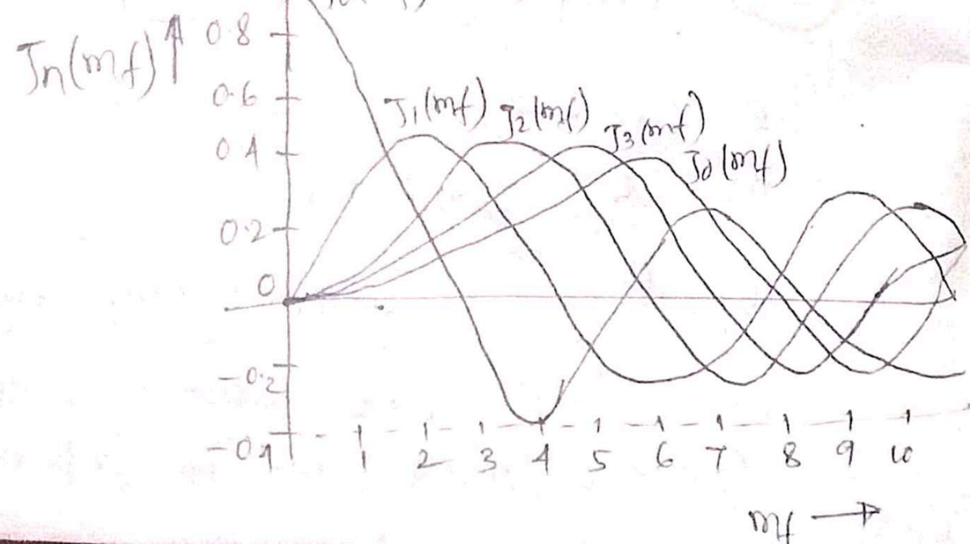
$$e_{FM}(t) = E_c \{ J_0(m_f) \sin w_c t$$
$$+ J_1(m_f) [\sin(w_c + w_m)t - \sin($$
$$+ J_2(m_f) [\sin(w_c + 2w_m)t - \sin(w_c$$
$$+ J_3(m_f) [\sin(w_c + 3w_m)t - \sin(w_c$$
$$+ \cdots + \cdots \}$$

The complex waveform consists of carrier signal and an infinite num of pairs of side bands preceded by J coefficients.

The values of $J_n(m_f)$ can be given

$$J_n(m_f) = \left(\frac{m_f}{2}\right)^n \left[\frac{1}{\lfloor n} - \frac{(m_f/2)^2}{\lfloor 1 \lfloor n+1} + \frac{(m_f/2)}{\lfloor 2 \lfloor} \right.$$
$$\left. - \frac{(m_f/2)^6}{\lfloor 3 \lfloor n+3} + \cdots \right]$$

Graphically Bessel's functions can be given as



Observations:

1. In FM, infinite pairs of sidebands are present along with carrier, separated by $f_m, 2f_m, 3f_m$ --- from carrier.

2. The modulation index ~~β~~, m_f determines the number of sideband components having significant amplitude.

3. Sidebands at equal distances from f_c, have equal amplitude. So sideband distribution is symmetrical about f_c.

(I) 4. In FM, total transmitted power always remains constant, but increasing depth of modulation the required band width is increased.

5. Theoretically B.W. required for FM waveform is infinity. But practically the used BW is extended upto accomodating the significant side bands.

6. In FM, amplitude of the carrier component does not remain constant.

7. It is possible for the carrier component of FM wave to disapear completely. This happens for certain values of modulation index called **eigen values**.

Bandwidth requirement of FM

Bandwidth requirement for FM waves is theoretically infinity. But practically the required bandwidth is upto accomodating the sideband components having significant amplitude. This band extends over a frequency on either side of the carrier, that is approximately equal to the sum of frequency deviation and modulating frequency. Then BW is twice of the above value.

$$\therefore BW_{fm} = 2[m_f + 1] f_m$$

$$= 2\left[\frac{\text{freq. deviation}}{\text{modulating freq}} + 1\right] f_m$$

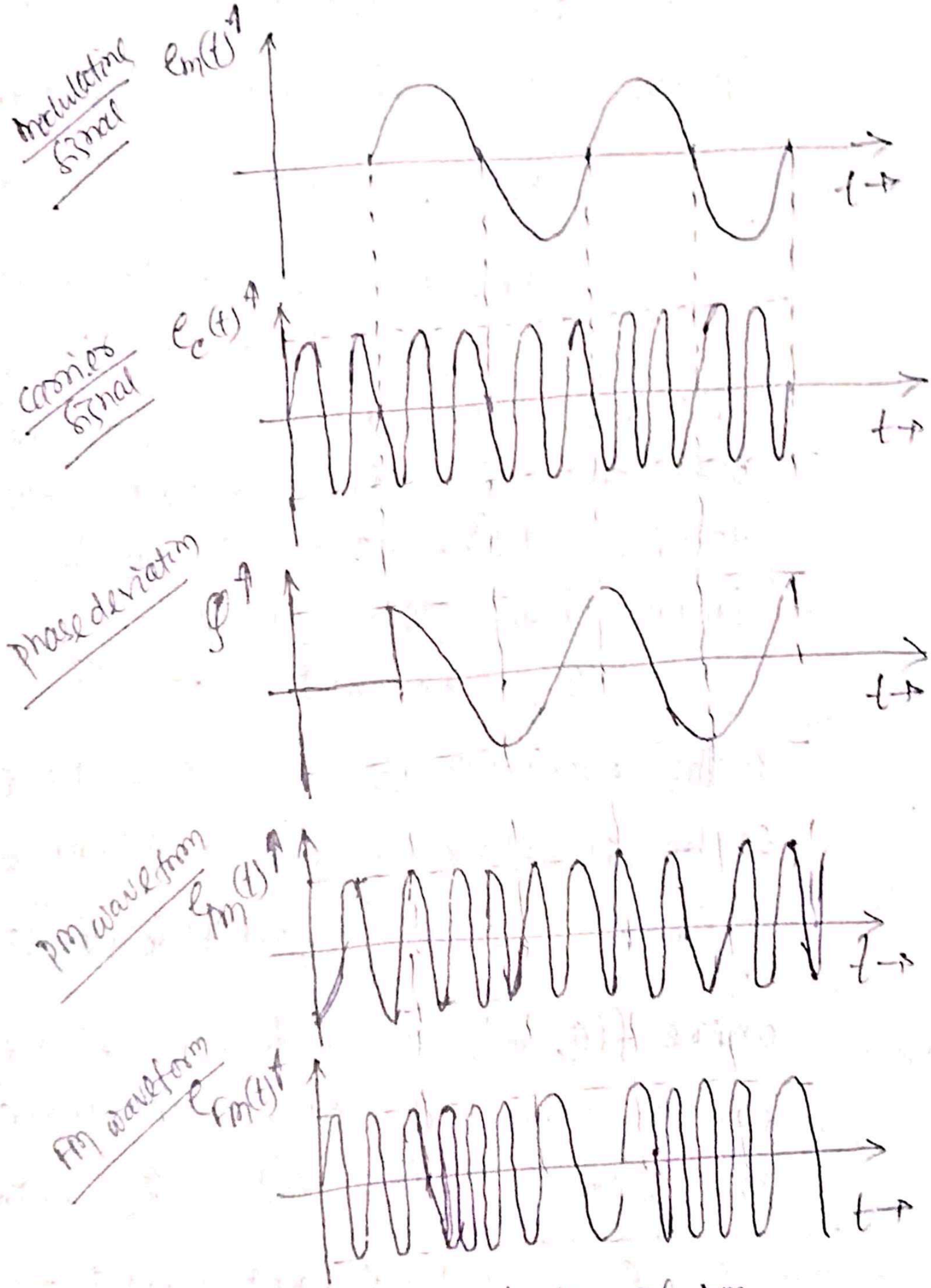
$$= 2\left(\frac{\delta f}{f_m} + 1\right) f_m$$

$$\therefore \boxed{BW_{fm} = 2(\delta f + f_m)}$$

3.4. Phase Modulation:

Phase modulation is a process, in which the phase of the carrier will be varied from its unmodulated value, called the phase deviation, is made proportional to the instantaneous amplitude of the modulating signal.

In the figures are shown the waveforms (3) of phase modulated (PM) waveform and for comparison the frequency modulated (FM) waveform.



Mathematical representation of PM.

Let the modulating signal be –

$$e_m(t) = E_m \sin(\omega_m t + \phi_m) \quad (1)$$

and carrier be $e_c(t) = E_c \cancel{\cos} \sin(\omega_c t + \phi_c)$ (2)

The instantaneous phase φ of the modulated wave will be -

$$\varphi = \varphi_c + K_p E_m \cos \omega_m t \quad \text{--- (3)}$$

where $\varphi_c \rightarrow$ phase of the unmodulated car

$K_p \rightarrow$ proportionality constant (radia

$E_m \cos \omega_m t$ is the phase shifted version of modulating signal.

The maximum deviation will take place when the cosine term will have its maxim value ($= \pm 1$), under this condition, the maximum phase deviation is -

$$\varphi = \varphi_c + K_p E_m \quad \text{--- (4)}$$

so the maximum deviation $\delta_p = K_p \cdot E_m$.

so the phase modulated waveform will be -

$$e_{pm}(t) = E_c \sin[\omega_c t + f(\varphi_c, \varphi_m)] = E_c \sin\theta \quad \text{--- (5)}$$

where $f(\varphi_c, \varphi_m)$ is a function of phases of carrier signal and modulating signal and can be given as $f(\varphi_c, \varphi_m) = \varphi_c + K_p E_m \cos \omega_m t$

$$\text{so, } \theta = \omega_c t + \varphi_c + K_p E_m \cos \omega_m t \quad \text{--- (6)}$$

and the phase modulated wave from, will be -

$$\boxed{e_{pm}(t) = E_c \sin(\omega_c t + \varphi_c + K_p E_m \cos \omega_m t)}$$

modulating index (m_p)

modulating index for PM, is defined as the maximum deviation in phase of the carrier from its unmodulated value.

$$\boxed{m_p = \delta_p = K_p E_m}$$

So, now the phase modulated waveform can be given as — $\boxed{e_{pm}(t) = E_c \sin(\omega_c t + \phi_c + m_p \cos\omega_m t)}$

3.5 Compairision between AM & FM.

① In FM, amplitude of the modulated waveform is constant unlike that in AM. In FM the amplitude of the modulated wave is independent of depth of modulating unlike AM. So the transmitted power in FM is all useful whereas in AM, most part of the transmitted power is in the carrier, which is unused.

② In FM receivers, amplitude limiters can be used to remove amplitude variations caused by noise, which is not possible in AM reception.

③ In FM, increasing frequency deviation decreases the noise. But in AM it is not possible.

④ standard frequency allocations provides

a guard band between ...
so there is less adjacent channel interference than AM.

(5) FM broadcasts involve the upper VHF & UHF range where there is less noise than the MF & HF ranges used in AM.

(6) In FM broadcast, space wave is used for propagation. So several independent transmitters can operate at the same frequency with less interference which is not possible in AM.

(7) In FM, much wider bandwidth is required (about 10 times that in AM).

(8) FM transmitting & receiving equipments are very complex where as in AM, these are simple.

(9) Since reception is limited to line of sight (for space wave propagation), the area of operation is much smaller than AM.

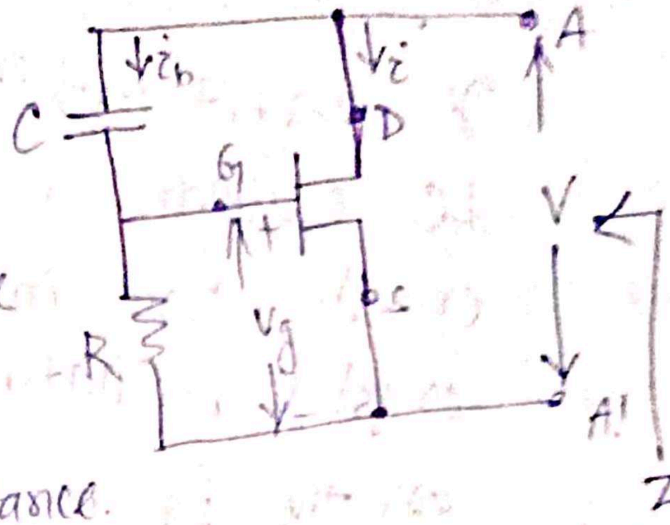
3.6 Methods of FM Generation.

a) Parameter variation Method:
(Direct method)

When either L or C parameters of an oscillator tank circuit is varied, its frequency changes, that is frequency modulation results. If this parameter variation is proportional to a voltage (modulating signal), therefore FM signal will be generated which is due to the variation in reactance of the oscillator tank circuit

Basic Reactance Modulator :

In the fig. is shown an FET reactance modulator. The impedance Z calculated across AA' is a pure reactance. (may be capacitive or inductive by varing the parameter).



Reactance calculation.

To calculate the reatance Z across AA', let a voltage V is applied at AA',

then $v_g = i_b \cdot R = \dfrac{V}{R - jX_c} \cdot R$

The drain current of FET, $i = g_m v_g = \dfrac{g_m \cdot R \cdot V}{R - jX_c}$

$$\therefore Z = \frac{V}{i} = V \Big/ \frac{g_m \cdot R \cdot V}{R - jX_c} = \frac{R - jX_c}{g_m \cdot R}$$

$$= \frac{1}{g_m}\left(1 - \frac{jX_c}{R}\right)$$

if $X_c >> R$, then $Z = -j \dfrac{X_c}{g_m \cdot R}$.

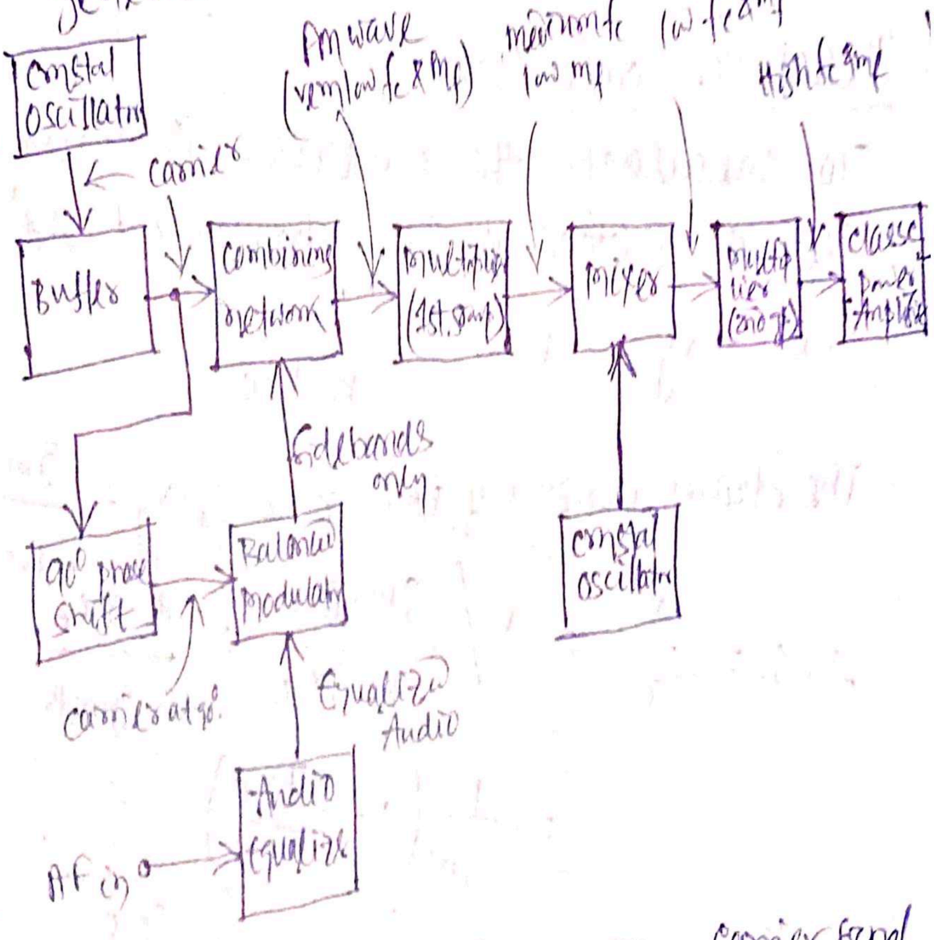
This impedance is a capacitive reactance given by. $X_{eq} = \dfrac{X_c}{g_m \cdot R} = \dfrac{1}{2\pi f C g_m R} = \dfrac{1}{2\pi f C_{eq}}$.

where $C_{eq} = C \cdot g_m \cdot R$.

ii) Indirect Method (Armstrong FM)

In direct method of FM generation, a LC oscillator is used which is not sta[ble] enough. But in indirect method, a crystal oscillator is used.

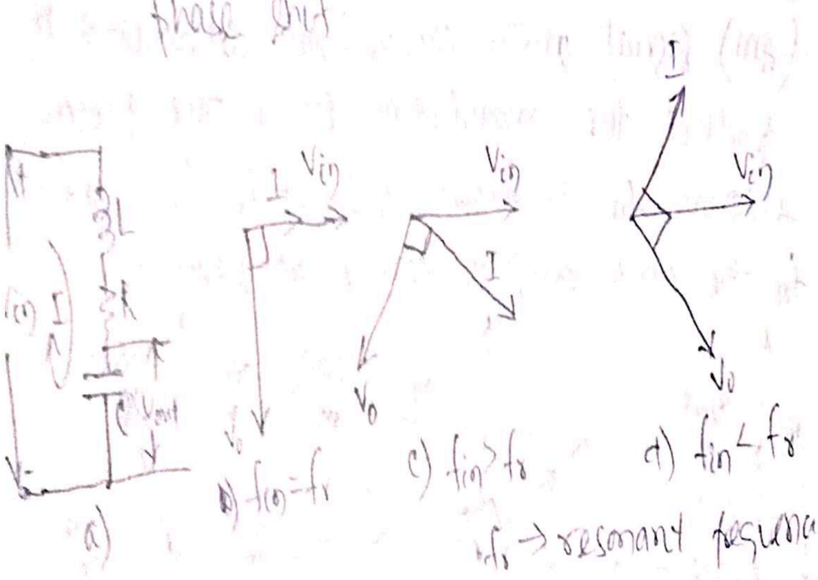
In the fig. is shown the basic block of the Armstrong (indirect) method of F[M] generation.



The crystal oscillator produces the carrier signal which is input to a combining network via a buffer. The other input to the combining network is the output of the balanced modulator. The two inputs of the balanced modulator are the equalized AF signal (modulating signal) and 90° phase shifted carrier. The output of the combining network is the FM signal but with low centre frequency f_c and low modulation index, which is then passed through groups of multipliers to produce wide band FM. Finally the class C power amplifier, amplifies the amplitude of the WBFM signal, transmitted through antenna.

3.7 Foster-seely FM descriminator

Foster seely descriminator is also called a phase shift demodulator. Its basic principle is based upon frequency dependent phase shift



$f_r \rightarrow$ resonant frequency.

Let consider the series RLC circuit, where the output is taken across the capacitor.

At resonant frequency — input voltage V_{in} and

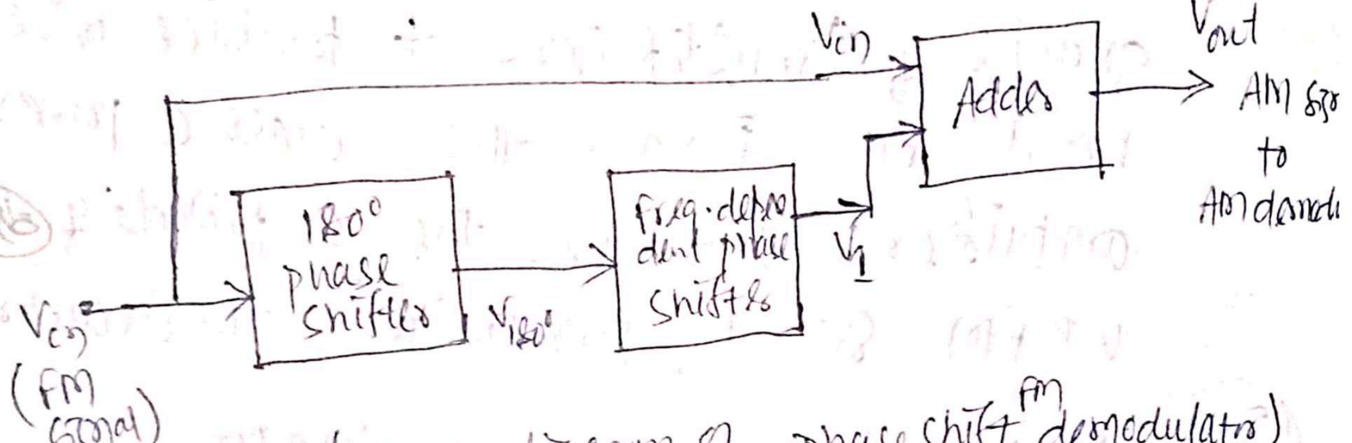
ait I are in phase. fig (b)

At freq. greater than f_r — the current is [illegible]

ctive and I lags V_{in}. — fig (c)

At freq. less than f_r — the circuit is ca

citive and I leads V_{in}. — fig (a).



(Block diagram of phase shift FM demodulator)

The input FM signal is 180° phase shifted and applied to a freq. dependent phase shifter. This output is added with the original FM signal by an adder. The output is amplitude modulated (AM) signal which can be AM demodulated to produce the modulating signal. The phasor diagram for 3 frequencies ($f_{in} = f_r$, $f_{in} > f_r$ & $f_{in} < f_r$) for the above diagram can be as shown —

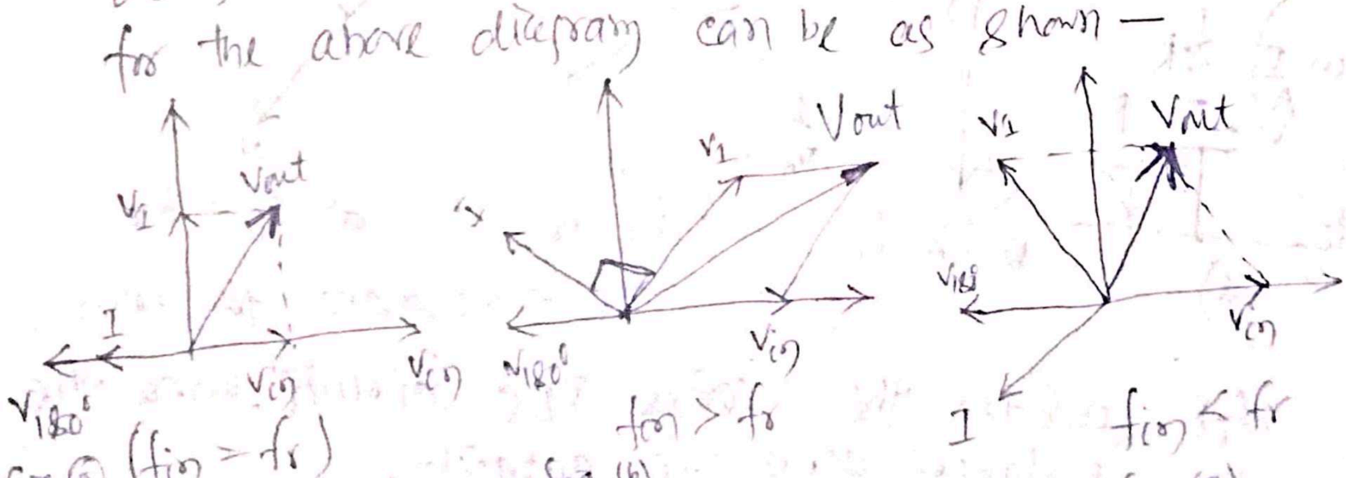
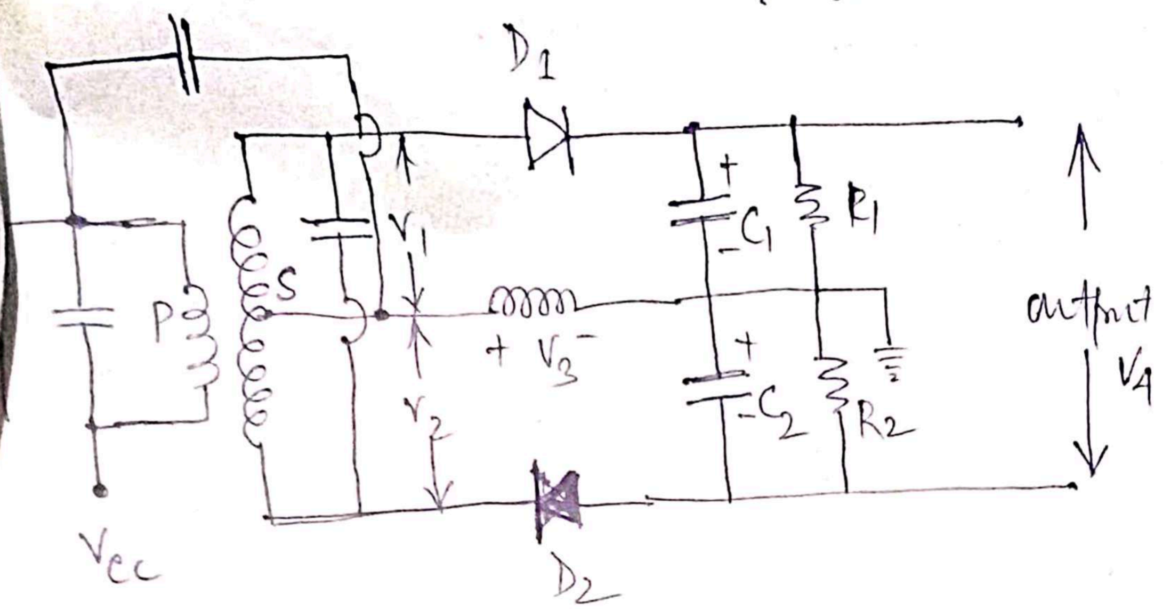


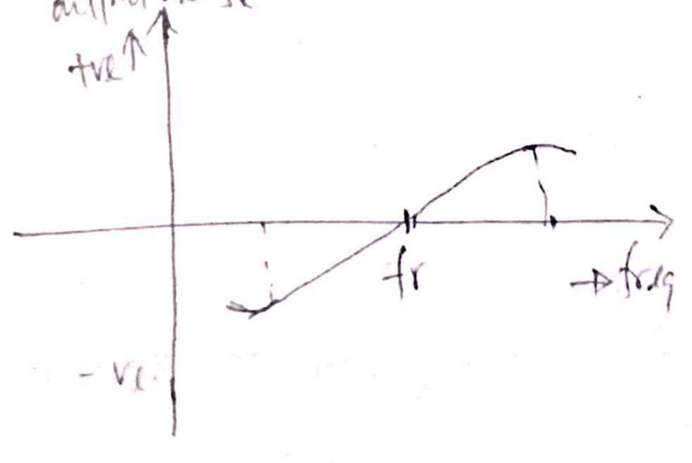
fig (a) ($f_{in} = f_r$) $f_{in} > f_r$ fig (b) $f_{in} < f_r$ fig (c)

Basing upon this principle (frequency dependent phase shift), the foster seely descriminator circuit is as shown — (5)



There are two tuned circuits P & S. The centre of the secondary S is connected to the top of primary P through a coupling capacitor. The RF voltage V_1 and V_3 (V_{1-3}) and V_2 and V_3 (V_{2-3}) are applied to the two diodes D_1 and D_2 and produces the output voltage V_4. V_4 is the arithmetic difference between V_{1-3} and V_{2-3}. $V_4 = [V_{1-3}] - [V_{2-3}]$.

In the fig. is shown the output voltage versus freq. graph of the foster seely descriminator.



— x —

UNIT-4: AM & FM TRANSMITTER & RECEIVER

4.1 Classification of Radio Receivers -

→ Radio Receivers can be classified as

→ ~~Tuned Radio-frequency~~

1) The first Vacuum-tube receivers
2) Regenerative (autodyne) receiver.
3) Superregenerative receiver.
4) TRF (Tuned Radio frequency) receiver.
5) Neutrodyne receiver
6) Reflex receiver.
7) Superheterodyne receiver.

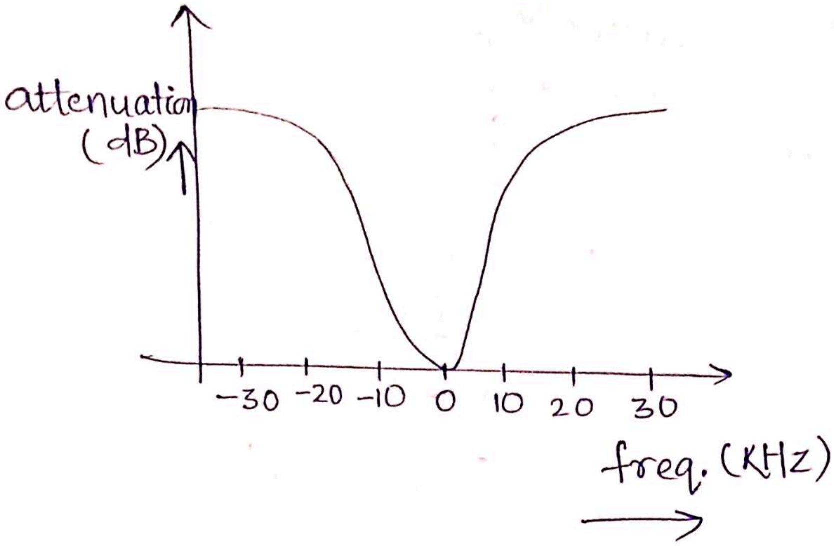
→ Radio receivers perform three basic functions on the signal from the antenna: filtering, amplification & demodulation.

→ During filtering of radio frequencies, the wanted information is allowed to pass while the undesired signal is cancelled out.

4.2 Define the terms Selectivity, Sensitivity, Fidelity & Noise Figure -

Selectivity -

Selectivity of a receiver is its ability to reject unwanted signal. Graphically it is represented as a curve, which shows the attenuation that the receiver offers to signals at frequencies near to the one to which it is tuned.



→ Selectivity is a measure of Quality factor Q of a tuned Circuit i.e. $\boxed{Q = \frac{X_L}{R}}$.

→ It is related to bandwidth as $\boxed{BW = \frac{f_r}{Q}}$, where

f_r → resonant frequency.

→ Narrower is the bandwidth, the better is Selectivity.

Sensitivity –

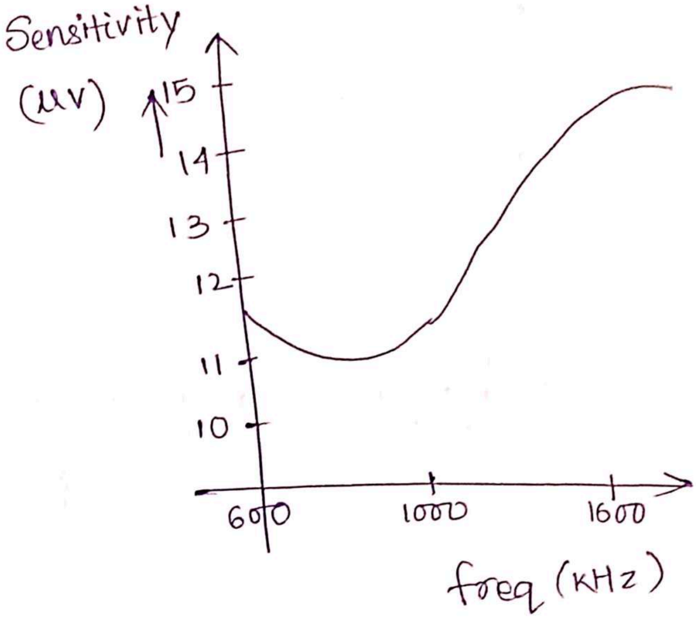
Sensitivity of a receiver is its ability to amplify weak Signals.

→ It is defined interms of voltage that must be applied to the receiver input terminals to give standard output Power.

→ More is the gain of the receiver, smaller will be the value of input voltage to produce desired output Power.

→ Sensitivity, sometimes expressed in micro volt(μV) or in decibels(dBs).

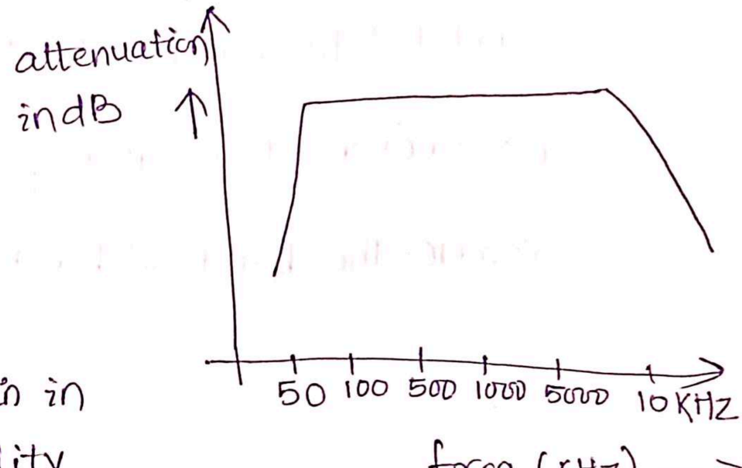
→ In the fig. is shown the Sensitivity graph of a typical receiver. It is seen that Sensitivity varies over the tunning band.



Fidelity –

fidelity is the ability of the receiver to reproduce all the modulating frequencies equally.

→ In the fig. is shown the fidelity curve for a radio receiver. It is seen that the attenuation is flat (Constant) over a band of frequencies.



→ Fidelity is difficult to obtain in AM receiver because good fidelity requires more bandwidth of IF amplifier resulting in poor Selectivity.

Noise Figure –

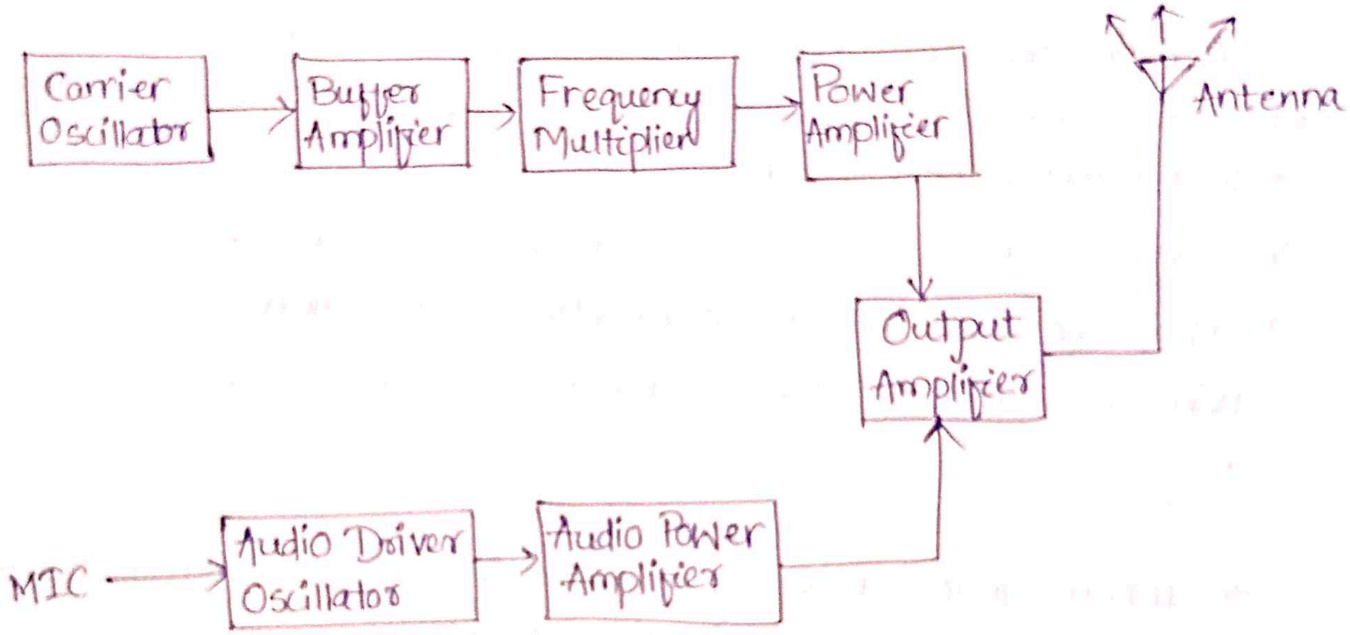
Noise figure or noise factor (F) is defined as the ratio of Signal-to-noise power supplied to the input terminals of a receiver or amplifier to the signal-to-noise power supplied to the output or load resistor.

$$F = \frac{\text{input } S/N}{\text{output } S/N} = \frac{S_i/N_i}{S_o/N_o} = \frac{S_i N_o}{S_o N_i}$$

→ Alternatively, noise figure can be defined as the S/N of an ideal system divided by the S/N at the output of the receiver or amplifier under test, both working at the same temperature, over the same bandwidth and fed from the same source. In addition both must be linear.

→ Noise figure may be expressed as an actual ratio or in decibels (dB).

4.3 AM transmitter - Working Principle With Block Diagram:-



[A.M Transmitter Block Diagram]

→ A.M Transmitter is a high-Power Transmitter means high-Power modulation is done.

→ AM transmitter takes the audio signal as an input and delivers amplitude modulated wave to the antenna as an output to be transmitted.

→ The RF Oscillator generates the Carrier Signal.

→ Both the Modulating and the Carrier Signal is sent to AM Modulator.

→ Power amplifier is used to increase the power levels of AM wave.

4.4: Concept of Frequency Conversion, RF amplifier & IF amplifier, Tuning, S/N ratio :—

Frequency Conversion –

Frequency Conversion converting the Carrier frequency of a received signal from its original value to the intermediate frequency value in a Superheterodyne receiver.

→ It converting data from one frequency to another, including moving from high to low frequencies, low to high frequencies and Converting between different types of panel data.

RF Amplifier –

A radio frequency power amplifier (RF power amplifier) is a type of electronic amplifier that Converts a low-power radio-frequency signal into a higher power signal.

→ Typically, RF Power amplifiers drive the antenna of a transmitter.

→ They are used in a wide variety of applications, including Wireless communication, TV Transmissions, Radar and RF heating.

IF Amplifier –

→ Intermediate-frequency (IF) amplifiers are amplifier stages used to raise signal levels in radio and television receivers, at frequencies intermediate to the higher radio-frequency (RF) signal from the antenna and the lower (baseband) audio or video frequency that the receiver is recovering.

→ IF amplifiers can change the frequency levels in circuits that are too selective, difficult to tune and unstable.

→ They also help by changing the frequency levels in circuit which improve image display and tuning range. They are fixed frequency amplifiers which reject unwanted Signals.

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Tuning – [To adjust (an electronic receiver) to a desired frequency].

→ Tuning means selecting.

→ Tuned amplifiers are the amplifiers that are employed for the Purpose of tuning.

→ Among a set of frequencies available, if there occurs a need to select a particular frequency, while rejecting all other frequencies, Such a process is called selection.

S/N ratio –

S/N ratio is defined as the ratio of Signal Power to the noise Power.

→ It is the measurement used to describe how much desired Sound is present in an audio recording, as opposed to unwanted Sound (noise).

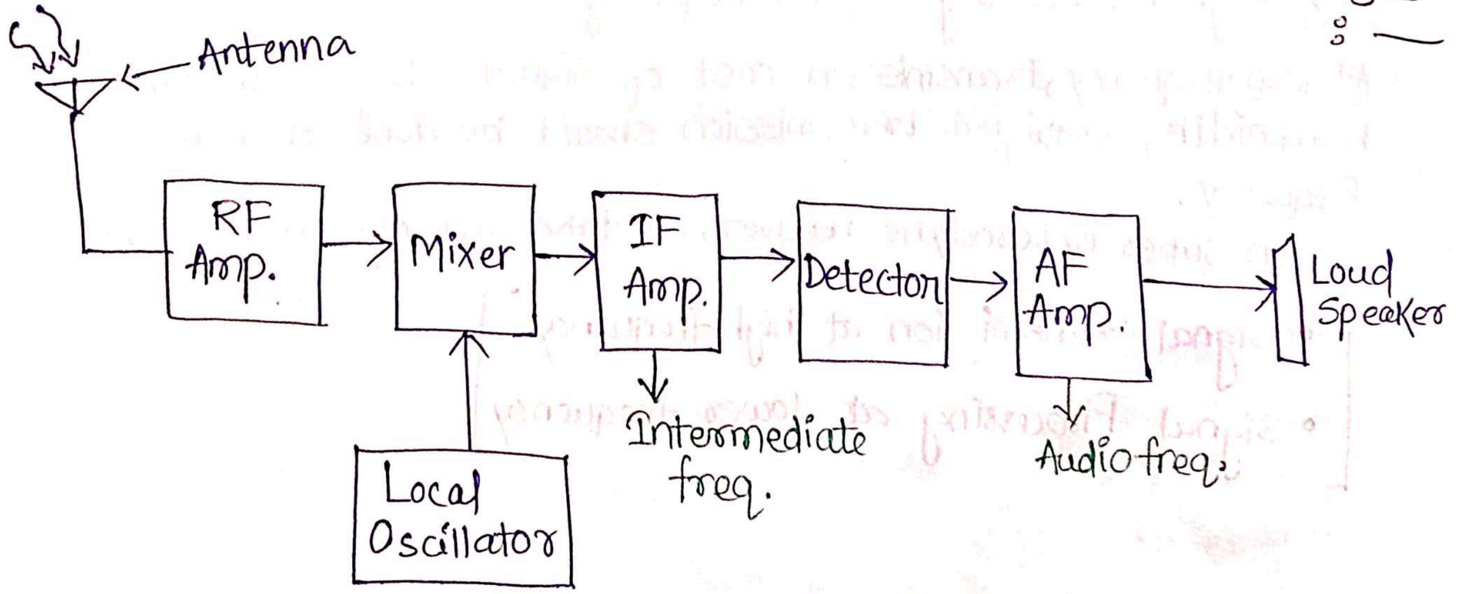
$$\therefore \frac{S}{N} = \frac{P_s}{P_n}$$

where, P_s = Signal Power

P_n = Noise Power at the Same point.

→ The Signal to noise ratio is normally expressed in dB and the typical values of S/N ratio range from about 10dB to 90dB.

4.5 Working of Super heterodyne radio receiver with Block diagram :–



① Antenna → accepting electromagnetic wave
→ Conversion of electromagnetic wave into electrical wave.

② RF Amplifier → Select desired frequency
→ Amplify weak signal

③ Mixer → Tuned frequency into medium frequency
→ Known as intermediate frequency.

Detector → Perform Demodulation
↓
→ detect audio signal

Basic :—

→ At high frequency, processing cost of circuit is high, so we should do signal processing at low frequency.

→ At low frequency, transmission cost of signal is high with low bandwidth, so signal transmission should be done at high frequency.

→ So, in super heterodyne receiver, we take care of above points.

- Signal transmission at high frequency.
- Signal Processing at lower frequency

* Heterodyning :-

It is the process of mixing two Signals having different frequencies in a non-linear manner to produce a signal with new frequency.

→ A special type of receiver is used for this operation :
Super heterodyne receiver (or) Superhet.

→ Higher RF (Radio frequency) to fixed lower Intermediate frequency (IF).

Therefore all the Successding stages have to operate on a fixed frequency.

Antenna -

→ first of all, Antenna is receiving the electromagnetic wave of different frequencies.

→ Then it convert the electromagnetic wave into electrical wave.

RF Amplifier -

→ The working of Radio frequency (RF) Amplifier is to Select the desired frequency and the desired frequency is high freq. signal.

→ It also amplify weak signal. (which is transmitted through a long distance).

Local Oscillator -

→ It produced low level frequency Signal.

Mixer -

→ The Mixer Convert the desired freq. into medium freq. with the help of Local Oscillator.

→ Means the Mixer get the desired freq. as well as the local Oscillator freq, and then they will adding freq.es (or) Substracting freq.es to get the medium frequencies.

The Process is Known as hetrodyne.

→ We can also say that, we converting high freq. Signal into mixed band freq. Signal. The process is Known as hetrodyne and the receiver who received the Signal is Known as Superheterodyne Receiver.

[∵ The Superheterodyne Receiver actually perform the heterodyne Process and the heterodyne Process is nothing but the difference (or) addition between the high freq. and the local Oscillator freq. Signal, So that it is Producing medium freq.]

→ The medium frequency is Known as intermediate frequency.

IF Amplifier –

→ In this, the intermediate frequency Signal will amplify with the help of this IF Amplifier.

→ So that, we can increase the Selectivity, Sensitivity & Fidelity.

Detector –

→ Detector will perform Demodulation.

→ After Demodulation, we can get the audio Signal.

AF Amplifier –

→ The audio Signal which we can get from the detector is low freq. signal. So we the low freq. Signal need to be boost up.

→ In this case, the audio freq. amplifier will help us to amplify the audio Signal.

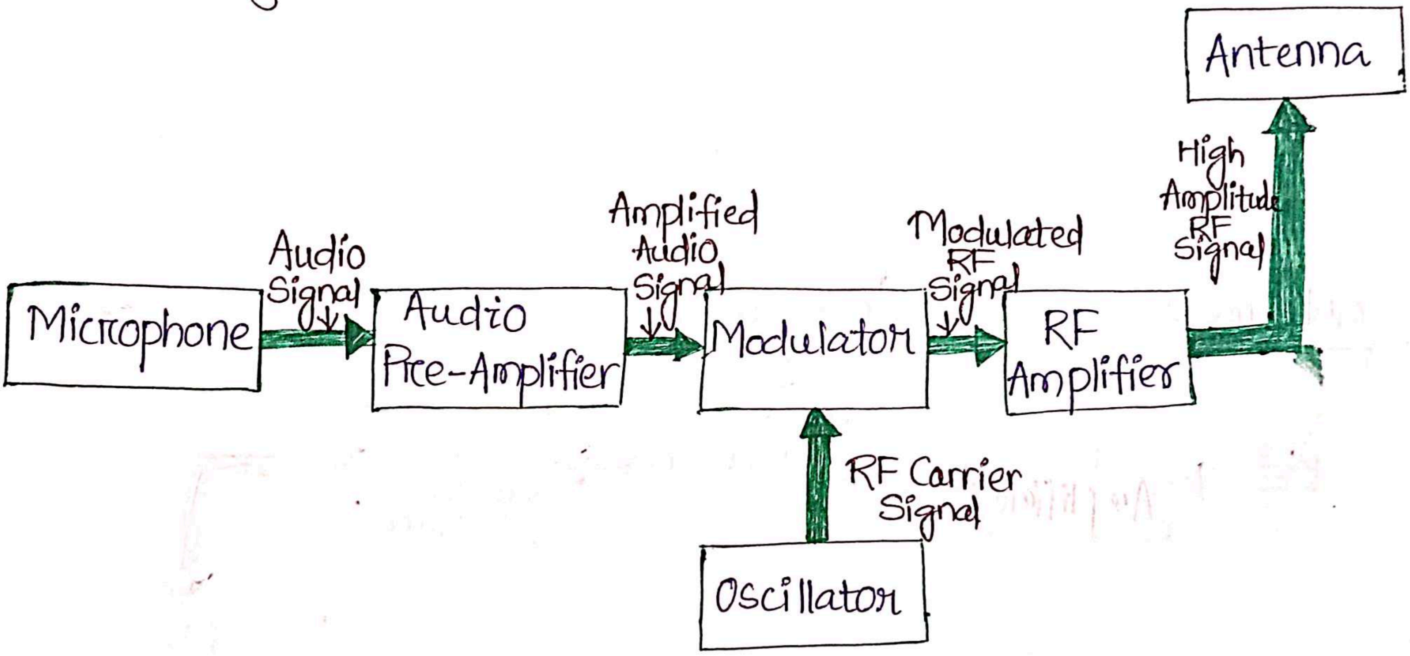
3/01/06 11:07

→ After the amplification, the Signal is transmitted to the Loud Speaker.

Loud Speaker –

→ The electrical wave will Convert into Sound Wave (which is our actual transmitted information).

4.6. Working of FM Transmitter & Receiver with Block Diagram :–



[Block Diagram of FM Transmitter]

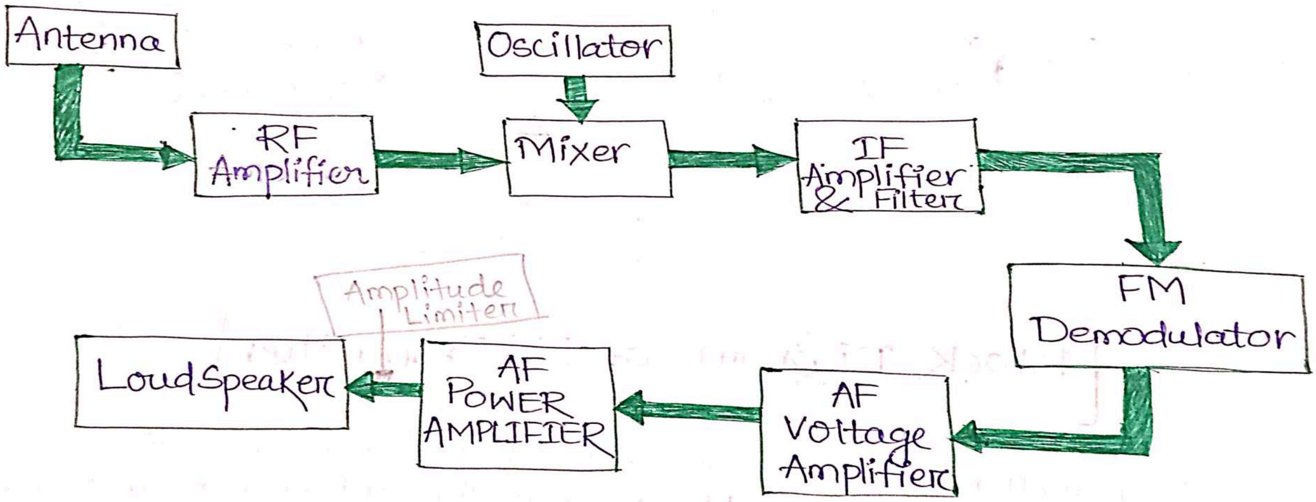
→ FM transmitter is the whole unit, which takes the audio Signal as an input and delivers FM wave to the antenna as an output to be transmitted.

→ The audio Signal from the output of the microphone is sent to the pre-amplifier, which boosts the level of the modulating Signal.

→ This signal is then passed to high pass filter, which acts as a Pre-emphasis network to filter out the noise and improve the signal to noise ratio.

→ This signal is further passed to the FM modulator Circuit. The Oscillator circuit generates a high frequency carrier, which is Sent to the modulator along with the modulating signal.

→ Several stages of frequency multiplier are used to increase the operating frequency. Even the power of the signal is not enough to transmit. Hence, a RF power amplifier is used to increase the power of the modulated Signal. This FM modulated output is finally reaching the antenna to be transmitted.



[Block Diagram of FM Receiver]

→ The two blocks Amplitude limiter and De-emphasis network are included before and after FM demodulator. The operation of the remaining blocks is the same as that of AM receiver.

→ We know that in FM modulation, the amplitude of FM wave remains constant. However, if some noise is added with FM wave in the channel, due to that the amplitude of FM wave may vary. Thus, with the help of amplitude limiter we can maintain the amplitude of FM wave as constant by removing the unwanted Peaks of raise Signal.

NOTE :—

→ In FM transmitter, we have seen the pre-emphasis network (High Pass filter), which is present before FM modulator. This is used to improve the SNR of high frequency audio Signal. The reverse process of pre-emphasis is known as de-emphasis. Thus, in this FM receiver, the de-emphasis network (Low pass filter) is included after FM demodulator. This Signal is passed to the audio amplifier to increase the power level. Finally, we get the original Sound Signal from the loudspeaker.

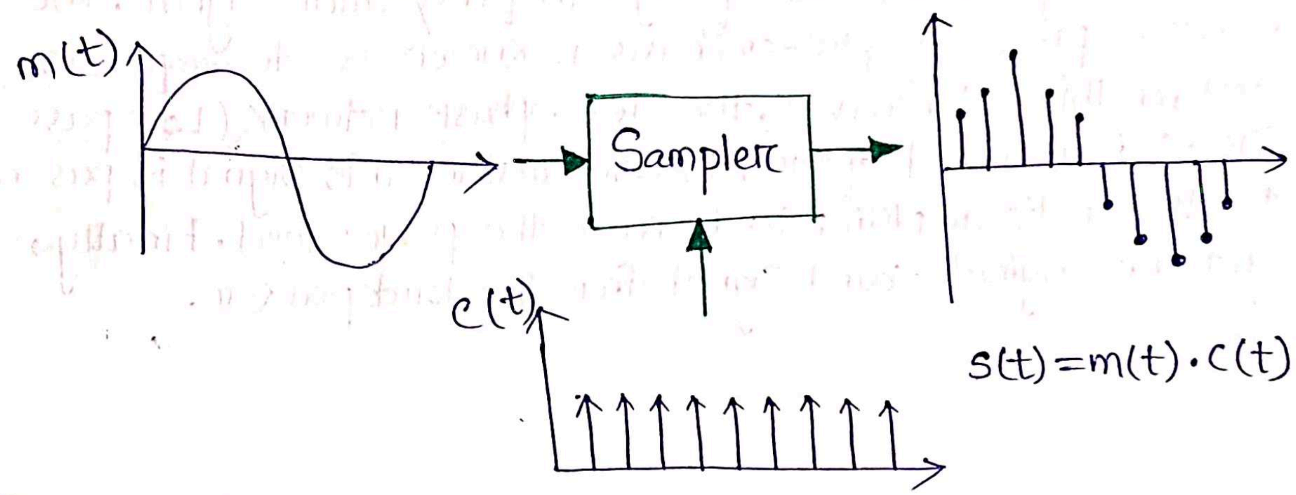
UNIT-5: ANALOG TO DIGITAL CONVERSION & PULSE MODULATION SYSTEM

5.1 Concept of Sampling Theorem, Nyquist rate & Aliasing

Sampling Theorem :- Sampling Process -

→ It is a process to convert continous time signal into discrete signal.

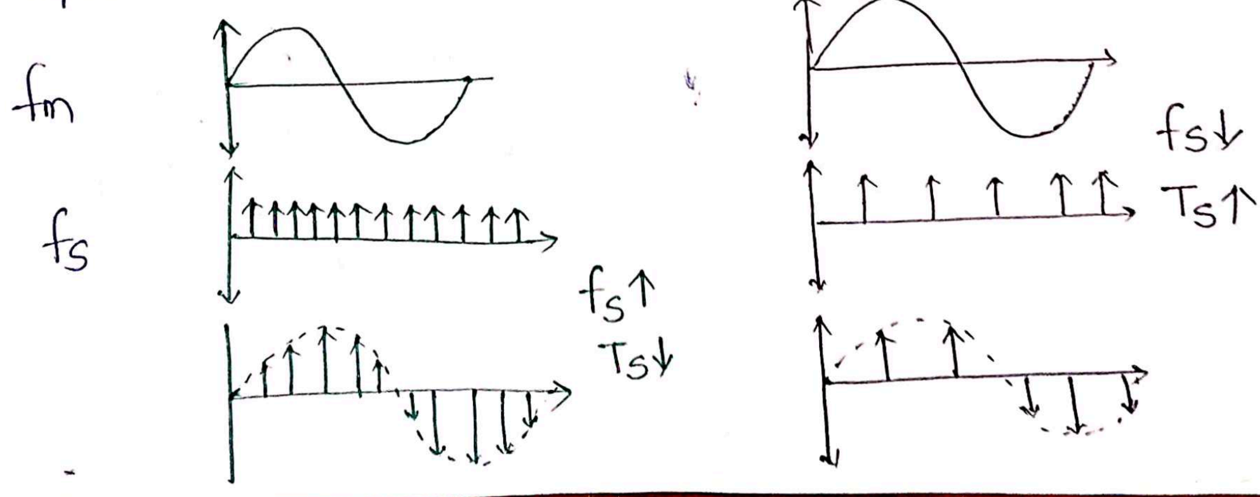
- Sufficient no. of samples must be taken, so that the original signal is reconstructed properly.
- No. of samples to be taken depends on maximum signal freq. present in the signal.



$s(t) = m(t) \cdot c(t)$

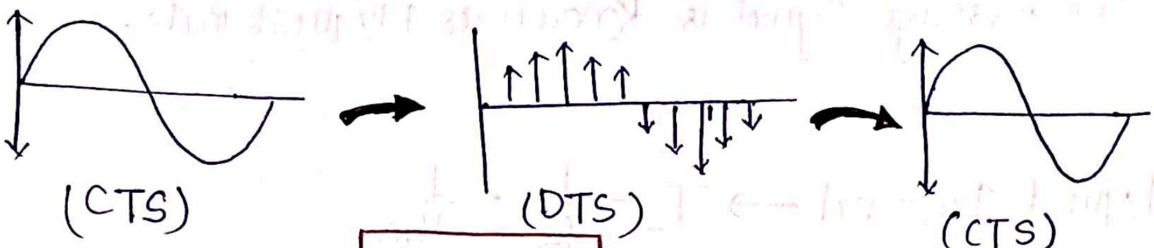
Sampling Process should satisfy :

1. Sampled signal should represent the original signal faithfully.
2. We should be able to reconstruct the original signal from its sampled version.



Sampling Theorem -

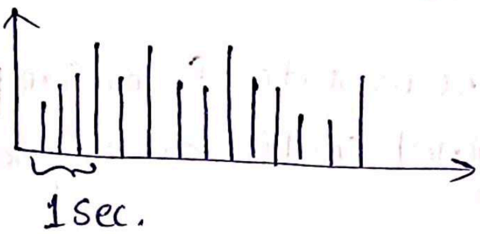
A Continuous time Signal can be completely represented in its Sampled form and recovered back from the Sampled form if Sampling frequency fs is greater than or equal to twice of maximum frequency of Continuous time Signal.



$$\boxed{f_s \geq 2f_m}$$

Sampling Rate (fs) -

The number of Samples per second or per unit time.

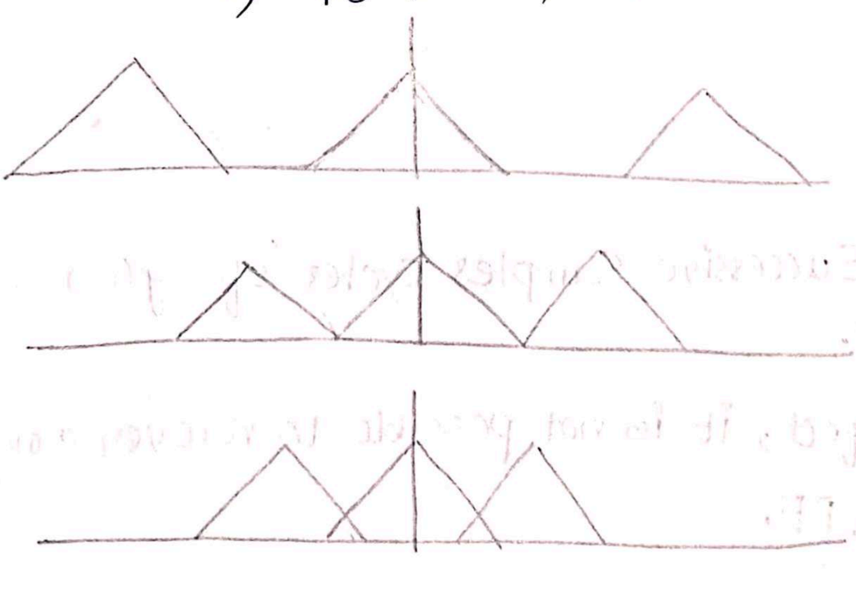


$\Rightarrow$ $\boxed{f_s \geq 2f_m}$ $\rightarrow$ If we written f in w form, then

$$\omega_s \geq 2\omega_m$$

$$2\pi f_s \geq 2 \cdot 2\pi f_m$$

$$\Rightarrow f_s \geq 2f_m$$



$\omega_s > 2\omega_m$ ✓

$\omega_s = 2\omega_m$ ✓

$\omega_s < 2\omega_m$ ✗

Nyquist Rate :-

$$W_S = 2W_M$$

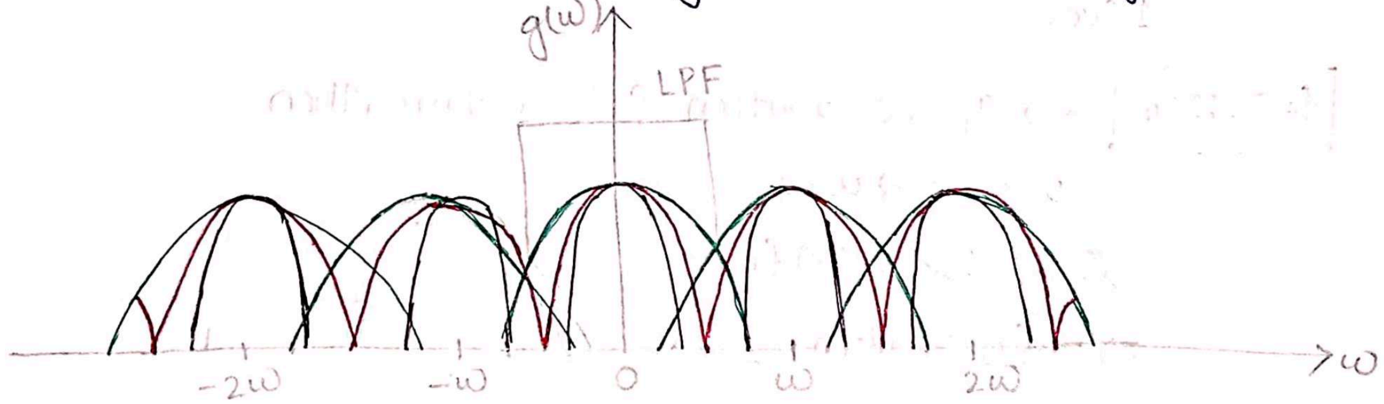
→ The Minimum sampling rate $\omega_s = 2\omega_m$ or $f_s = 2f_m$ required to recover the message signal is known as Nyquist rate.

Nyquist Interval → $T_S = \frac{1}{f_S} = \frac{1}{2f_m}$

Aliasing :-

→ Aliasing is an effect of the Sampling that causes different Signals to become indistinguishable.

→ Due to aliasing, the signal reconstructed from samples may become different than the original continuous Signal.



→ If $f_s < 2f_m$, then Successive Samples cycles of $g(\omega)$ will overlap each other.

→ Due to Aliasing effect, it is not possible to recover original Signal $x(t)$ by LPF.

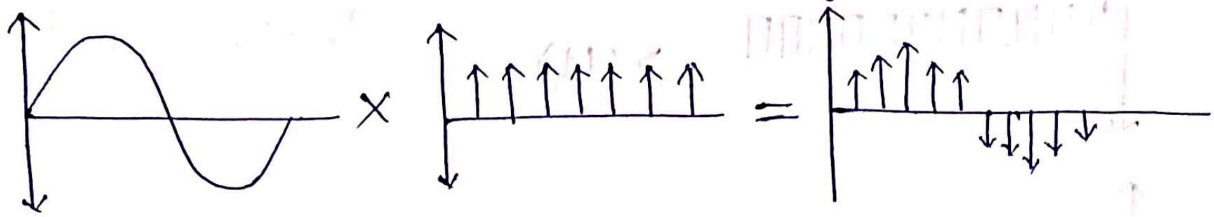
→ Hence, due to overlap at one region to other region, Signal x(t) is distorted.

→ So before we go for Sampling, we pass original Signal through LPF. This is reffered as pre-alias filter, other name is band limit filter.

5.2 Sampling Techniques

(a) Instantaneous Sampling —

Uses unit impulse train as Sampling function.



$$x(t) \cdot \delta(t) = s(t)$$

→ Sampled signal is obtained by multiplication $x(t)$ and $\delta(t)$.

→ Duration of each impulse is extremely short.

→ Sampling rate is infinite.

→ Bandwidth is large.

Disadvantages :

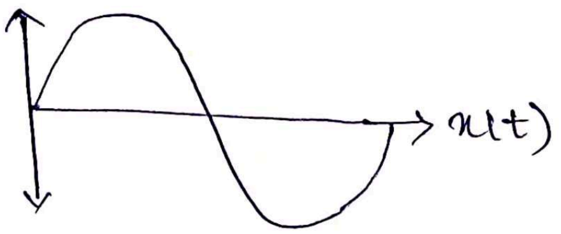
→ Due to very narrow Samples, transmitted power is very Small Signal to noise ratio is low. Thus Sampled pulses may get lost in the background noise.

→ No ideal filters available to Cut off Signals.

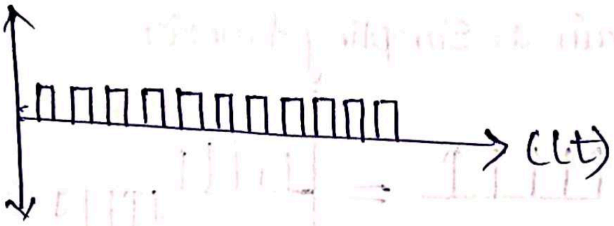
(b) Natural Sampling –

→ Sampling pulses have duration 'τ' and separated by the sampling time 'T_s'.

→ Sampled signal is obtained by multiplication of $x(t)$ and $c(t)$.



Baseband Signal



Sampling Signal

$s(t) = x(t) \cdot c(t)$ Naturally Sampled signal

→ Pulses do not have flat tops but their tops follow the waveform of $x(t)$.

Merits

→ Generation of natural sampling is easy.

→ Practical low pass filter can be used.

Demerits

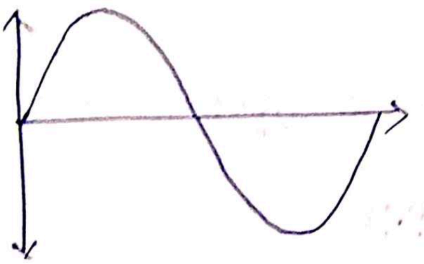
→ Amplitude of sampled pulses is varying.

→ for large value of 'τ', crosstalk is introduced.

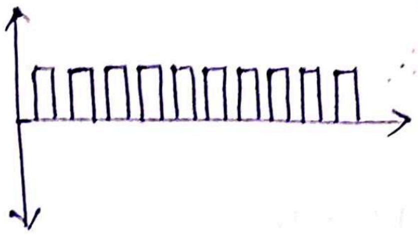
(c) Flat top Sampling —

→ Amplitudes of Sampled pulses are constant.

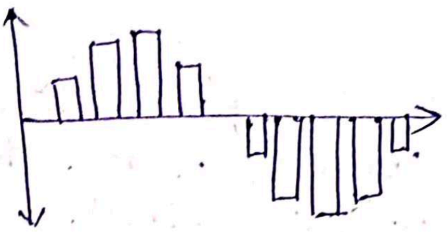
→ Uses Sample and hold Circuit.



$x(t)$ Baseband Signal



$c(t)$ Sampling Signal



$s(t) = x(t).c(t)$ flat top Sampling

Merits

→ Better Signal to noise ratio.

→ Generation of flat top Sampling is easy.

→ Practical filter can be used.

5.3 Analog Pulse Modulation

In Analog Pulse Modulation, the amplitude, duration or Position of a pulse is varied in accordance with Sample values of the message Signal.

→ The Analog Pulse Modulation are of three types.

(i) Pulse - amplitude Modulation (PAM)
(ii) Pulse - width Modulation (PWM)
1 (iii) Pulse - Position Modulation (PPM)

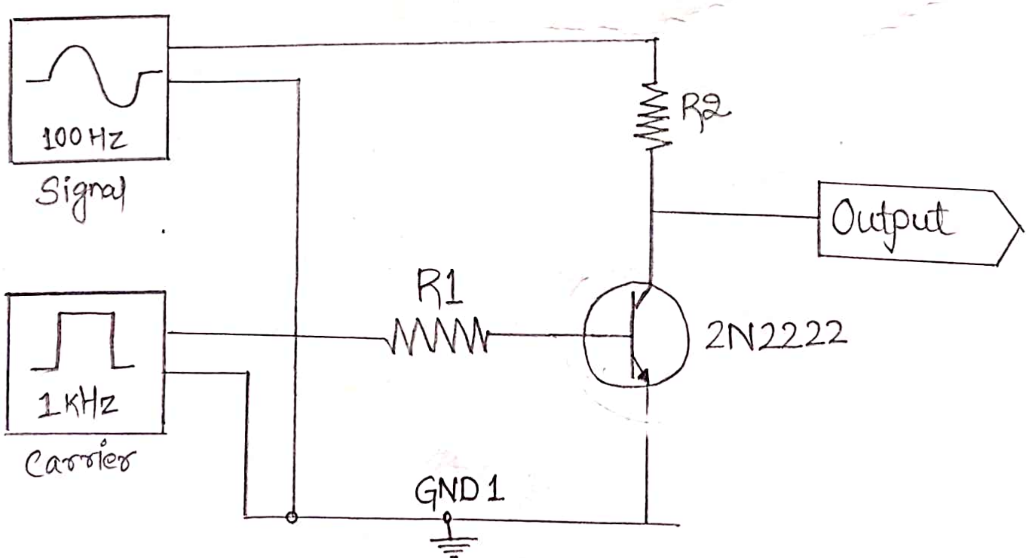
PAM (Pulse- amplitude Modulation) :

→ Pulse Amplitude Modulation is a Pulse Analog modulation Scheme in which the amplitude of a train of Carrier pulses are varied according to the amplitude Variations of message signal.

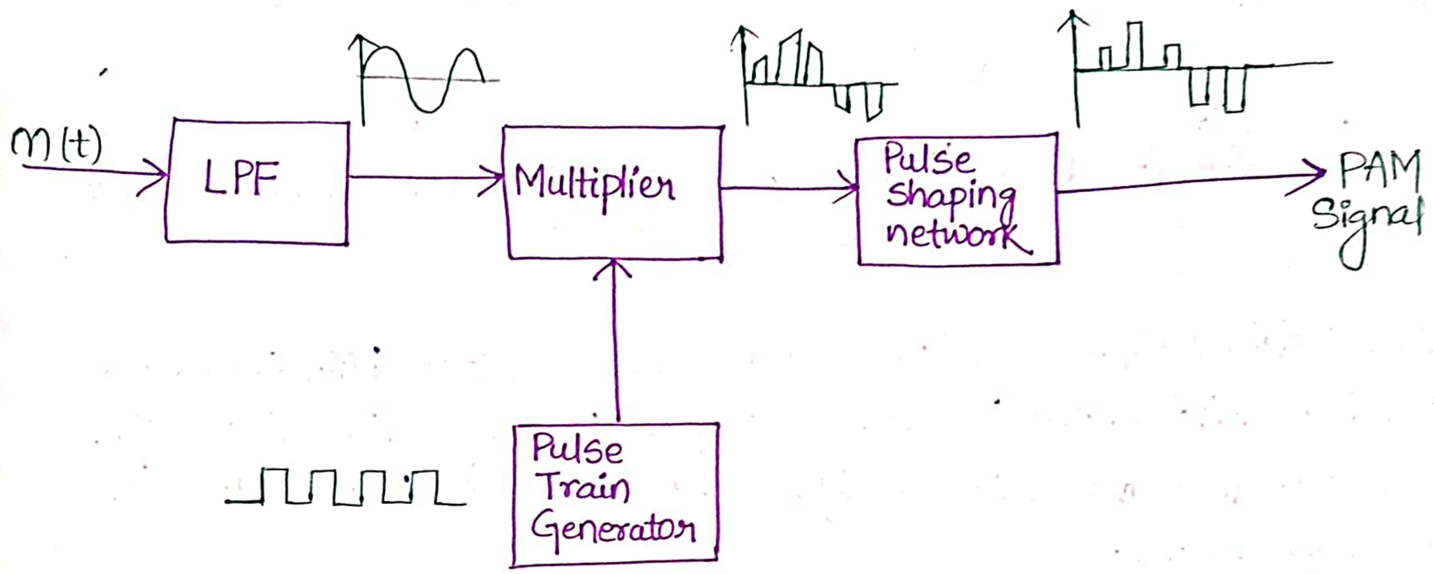
→ In pulse analog modulation pulse by pulse transmission will occured. Each of the pulse to be transmitted Corresponds to baseband Signal and can be directly transmitted through baseband channel only.

Generation

PAM Signal Can be generated either using natural Sampling or using flat-top Sampling as Shown in fig.



[Circuit diagram of PAM]



[Block diagram of PAM Generator]

→ The above fig. shows the block diagram of PAM Generator.

→ It Consists of LPF, a multiplier and a Pulse Generator.

→ Initially the modulating Signal m(t) is passed through LPF to bandlimit the message Signal.

→ The bandlimiting is necessary to avoid aliasing effect in the Sampling Process.

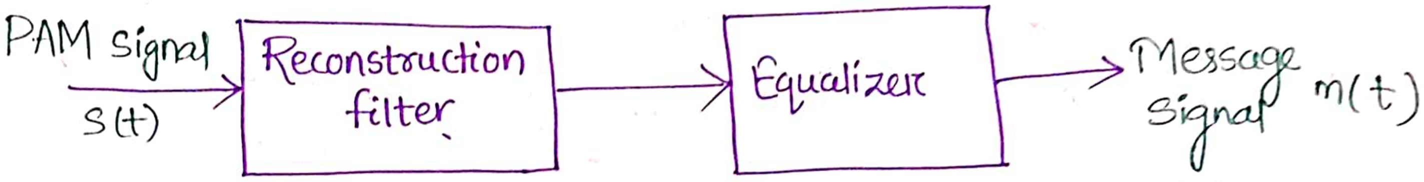
→ Pulse shaping network does the shaping work to give flat tops.

→ If the generated pulses are narrow, PAM signals require little Power for transmission and are Suitable for Time Division Multiplexing (TDM).

→ Flat topped Pulses are easily regenerated by Repeaters and Can be used for transmission over long distances.

→ One disadvantage of PAM Signals is these are affected by noise as much as analog Signals.

Reconstruction of Message Signal (Detection)



[System for Recovering message Signal m(t) from PAM signal s(t)]

→ The Original message signal Can be detected from PAM Signal by Passing PAM signal through a lowpass reconstruction filter with Cut-off frequency Slightly higher than the maximum frequency in message Signal.

→ The equalizer Compensate the aperture effect and attenuation Provided by Low Pass Reconstruction filter.

Pulse Width Modulation (PWM) :-

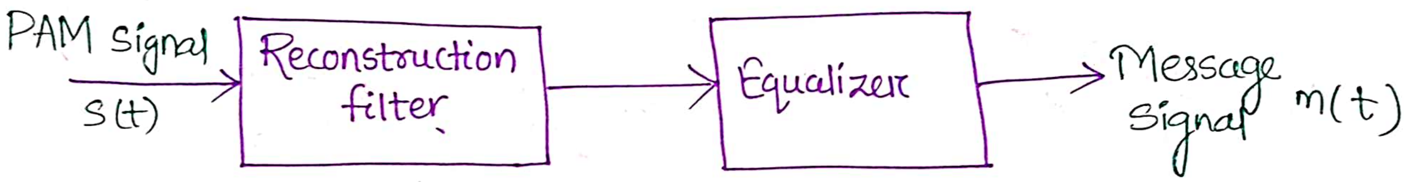
In PWM, the Width of the pulsed Carrier Wave is Varied as per the instantaneous value of the Message Signal (modulating Signal).

→ PWM is less affected by noise due to its Constant amplitude. It is also easier to remove noise from the signal as Compared to PAM.

→ In PWM, the instantaneous Power of the transmitter Varies due to Variations in the Widths of the pulses.

→ One disadvantage of PAM signals is these are affected by noise as much as analog signals.

Reconstruction of Message Signal (Detection)

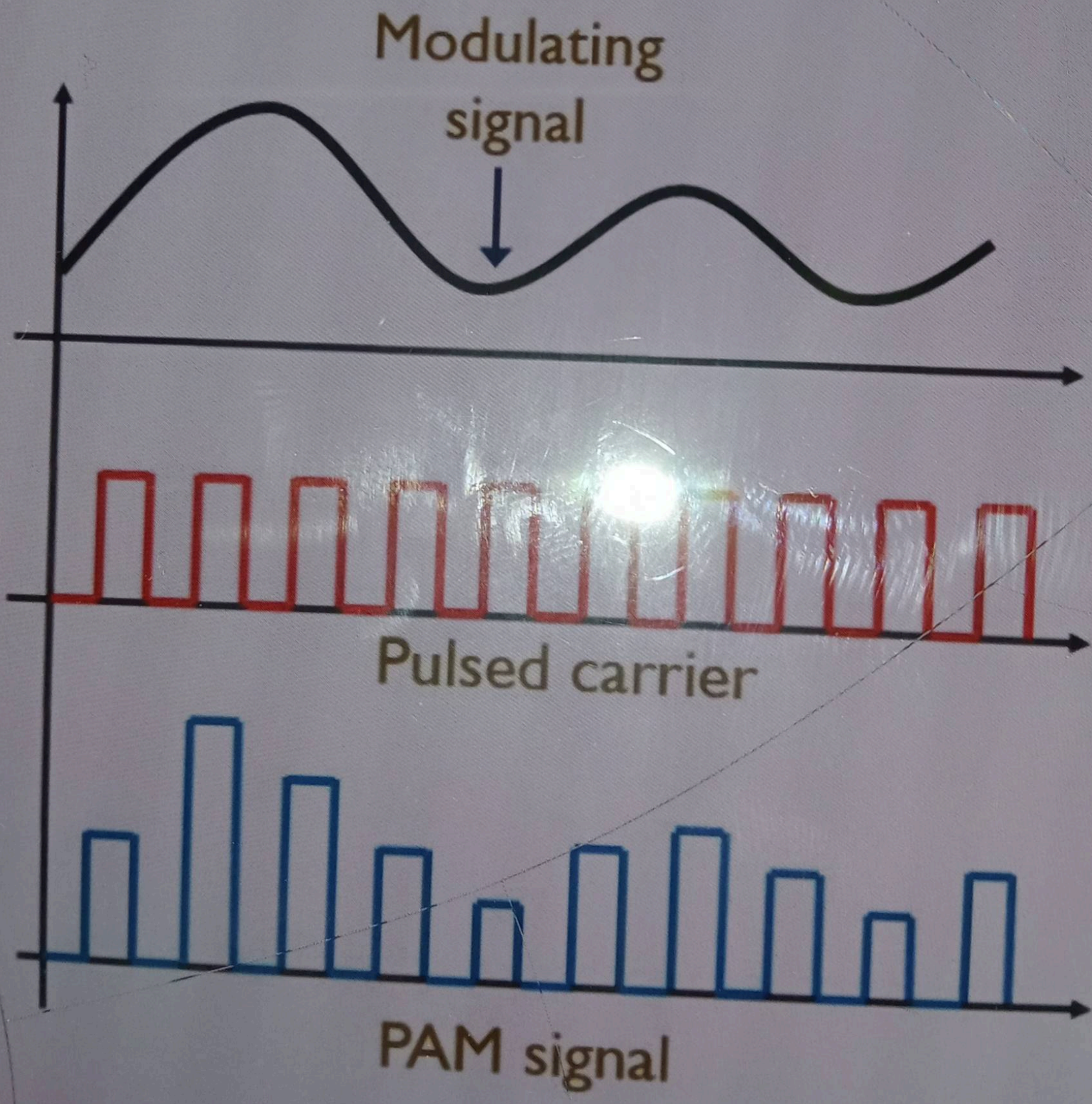


[System for Recovering message signal m(t) from PAM signal s(t)]

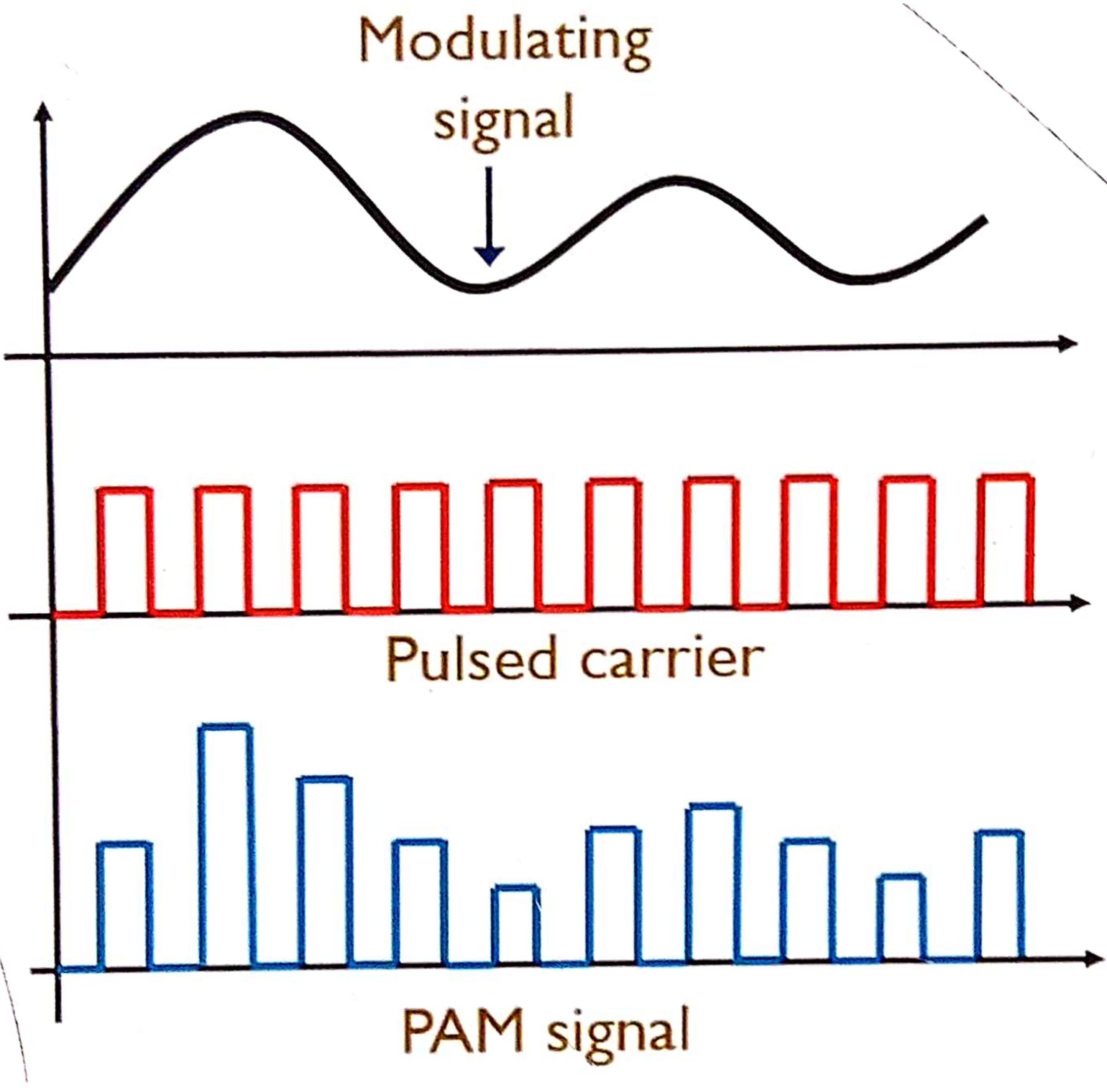
→ The original message signal can be detected from PAM signal by passing PAM signal through a lowpass reconstruction filter with cut-off frequency slightly higher than the maximum frequency in message signal.

→ The equalizer compensate the aperture effect and attenuation provided by Low Pass Reconstruction filter.

PAM stands for pulse amplitude modulation. It is a modulation technique in which the amplitude of the pulsed carrier signal is changed according to the amplitude of the message signal. The figure given below represents a PAM signal:



As we can see in the figure shown

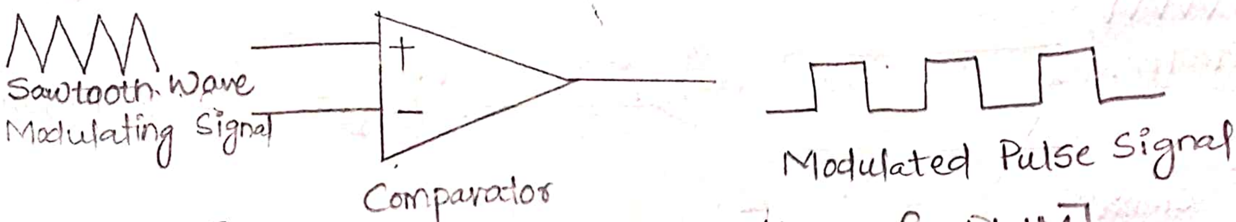

Modulating
signal
Pulsed carrier
PAM signal

Pulse Width Modulation (PWM) :-

In PWM, the width of the pulsed carrier wave is varied as per the instantaneous value of the message signal (modulating signal).

→ PWM is less affected by noise due to its constant amplitude. It is also easier to remove noise from the signal as compared to PAM.

→ In PWM, the instantaneous power of the transmitter varies due to variations in the widths of the pulses.



[CKt diagram of generation of PWM]

→ A pulse width Modulating Signal is generated using a Comparator. The modulating signal forms one part of the input to the Comparator While the non-sinusoidal wave or Sawtooth wave forms the other part of the input. The comparator compares two signals and generates a PWM signal as its output waveform.

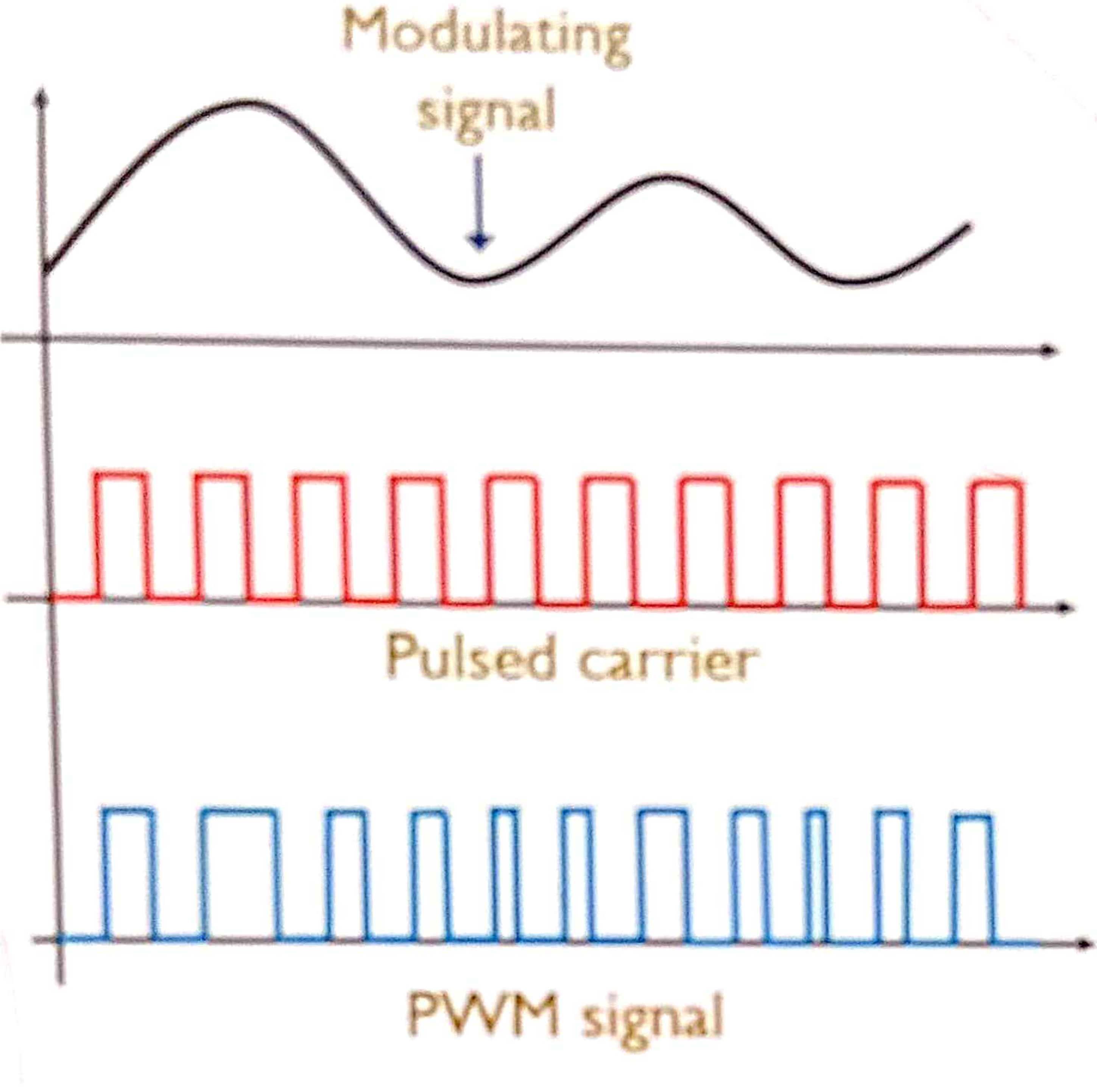
→ If the Sawtooth Signal is more than the modulating signal, then the output signal is in a "High" state. The value of the magnitude determines the comparator output which defines the width of the Pulse generated at the output.

→ The PWM is very Similar to frequency Modulation (FM).

∴ In FM freq. of the Carrier Varies as per the modulating Signal.

$$\boxed{f = 1/T}$$

Similarly, in PWM the P Duration (Time Period/Time Duration) Varies.

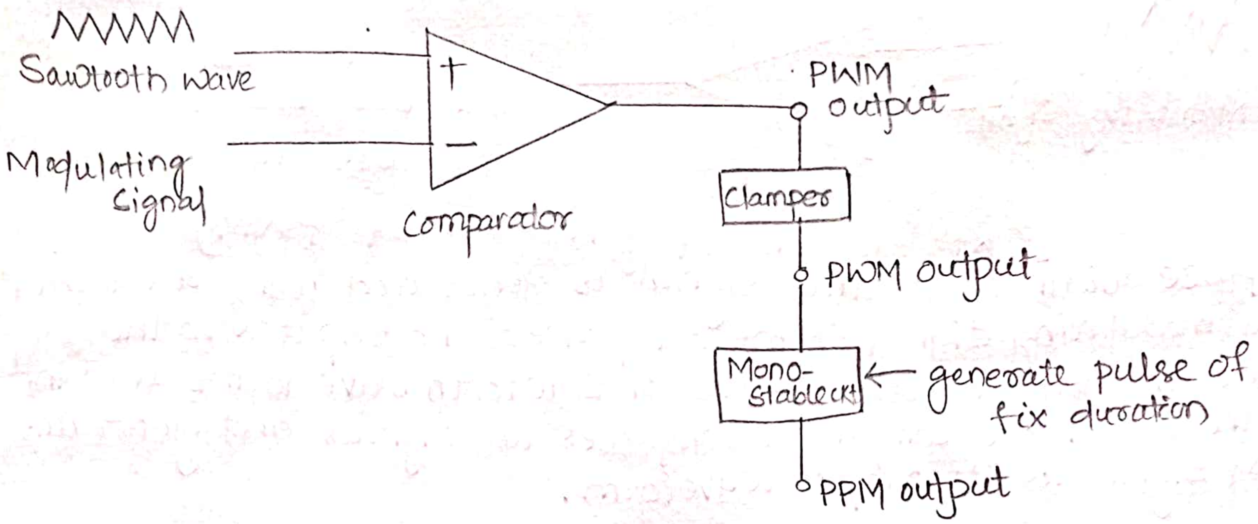

Modulating
signal
Pulsed carrier
PWM signal

Pulse Position Modulation (PPM) :—

Pulse Position Modulation (PPM) is a modulation technique where the position of the pulsed Carrier is varied according to the modulating Signal (message Signal).

→ Because of Constant amplitude in PPM, Noise interference is very less & also noise can easily be Separated from the Signal.

→ As in case of PPM, the amp. & width of each pulse is Same. Therefore the transmission power for each pulse is Same.



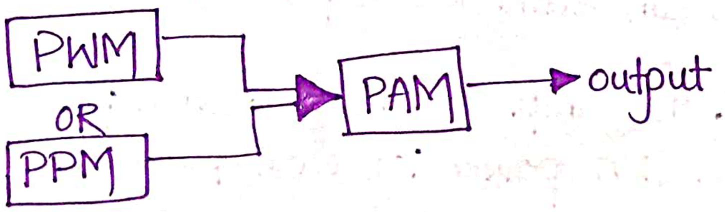
[CKt diagram of generation of PPM]

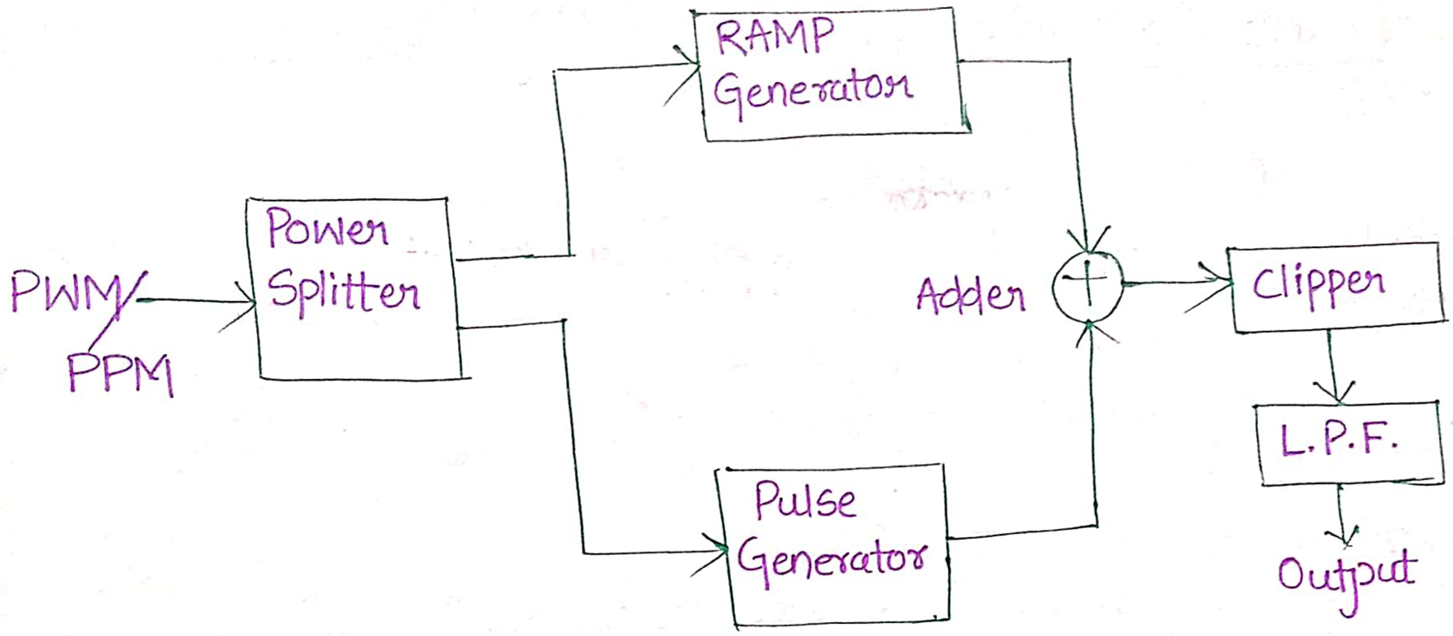
Synchronization b/w transmitter & Receiver —
In PPM, Synchronization b/w transmitter & receiver is required.

Similarity with Phase Modulation (PM) —
Change in Phase – Shifting of the Signal Position on the time axis.
Like PM, in PPM too, the Position of the pulse is varied on the time axis according to the modulating Signal.

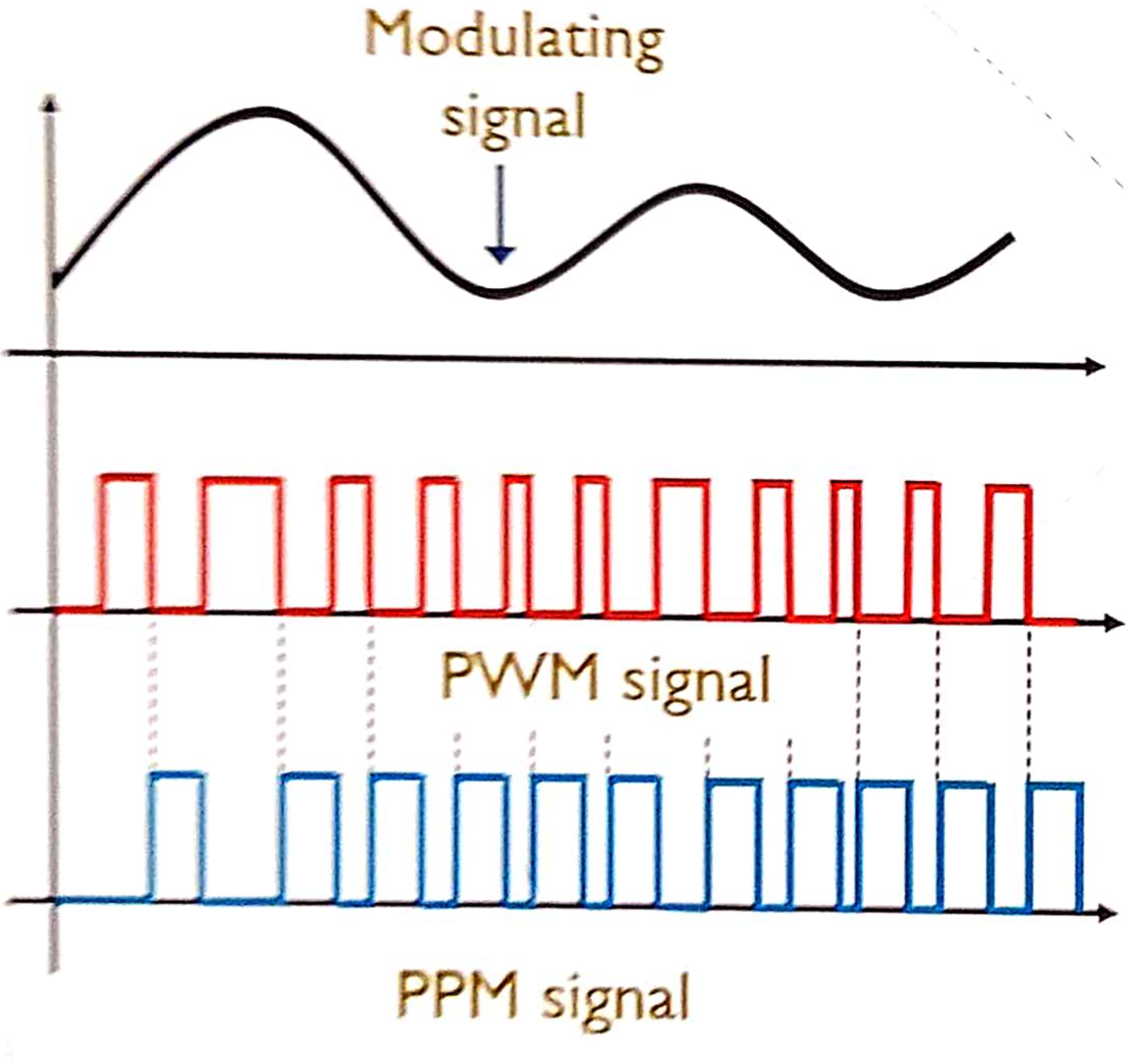
PWM & PPM Demodulation :—

To demodulate (detect) a PWM or PPM Signal, it is first required to be converted into a PAM Signal, then on passing this PAM signal through the Low Pass Filter (LPF), we can get the message Signal in the o/p.

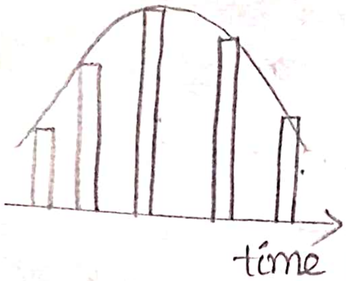
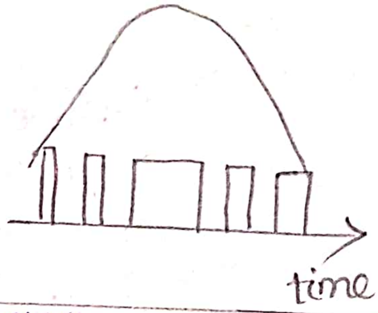




[PWM/PPM Demodulation Block Diagram]


Modulating
signal
PWM signal
PPM signal

Performance Comparison of Pulse Analog Modulation :-

Performance Parameter	PAM	PWM	PPM
Waveform			
Working Principle	Amplitude of pulse Proportional to amp. of modulating signal.	Width of pulse is Proportional to amplitude of modulating signal.	Relative Position of Pulse is proportional to amp. of modulating signal.
Bandwidth	BW is depending on width of pulse.	BW depends on rise time of pulse.	BW depends on rise time of pulse.
Transmitted Power	Varies w.r.t time	Varies w.r.t time	Constant
Noise interference	Max	Min	Less
	It is Similar to AM in Analog Modulation.	It is similar to FM in Analog Modulation.	It is Similar to PM in Analog Modulation.

13.6. Drawbacks of Pulse-Amplitude Modulated (PAM) Signal

Following are the drawbacks of a PAM signal :

(i) The bandwidth required for the transmission of a PAM signal is very large in comparison to the maximum frequency present in the modulating signal.

(ii) Since the amplitude of the PAM pulses varies in accordance with the modulating signal therefore the interference of noise is maximum in a PAM signal. This noise cannot be removed easily.

(iii) Since the amplitude of the PAM signal varies, therefore, this also varies the peak power required by the transmitter with modulating signal.

14.5. Advantages of PWM

(*i*) Unlike, PAM, noise is less, since in PWM, amplitude is held constant.

(*ii*) Signal and noise separation is very easy, as shown in figure 9.32 (*b*).

(*iii*) PWM communication does not require synchronization between transmitter and receiver

.14.6. Disadvantages of PWM

(i) In PWM, pulses are varying in width and therefore their power contents are variable. This requires that the transmitter must be able to handle the power contents of the pulse having maximum pulse width.

(ii) Large bandwidth is required for the PWM communication as compared to PAM.

14.10. Advantages of PPM

(*i*) Like PWM, in PPM, amplitude is held constant thus less noise interference.

(*ii*) Like PPM, signal and noise separation is very easy.

(*iii*) Because of constant pulse widths and amplitudes, transmission power for each pulse is same.

.14.11. Disadvantages of PPM

(*i*) Synchronization between transmitter and receiver is required.

(*ii*) Large bandwidth is required as compared to PAM.

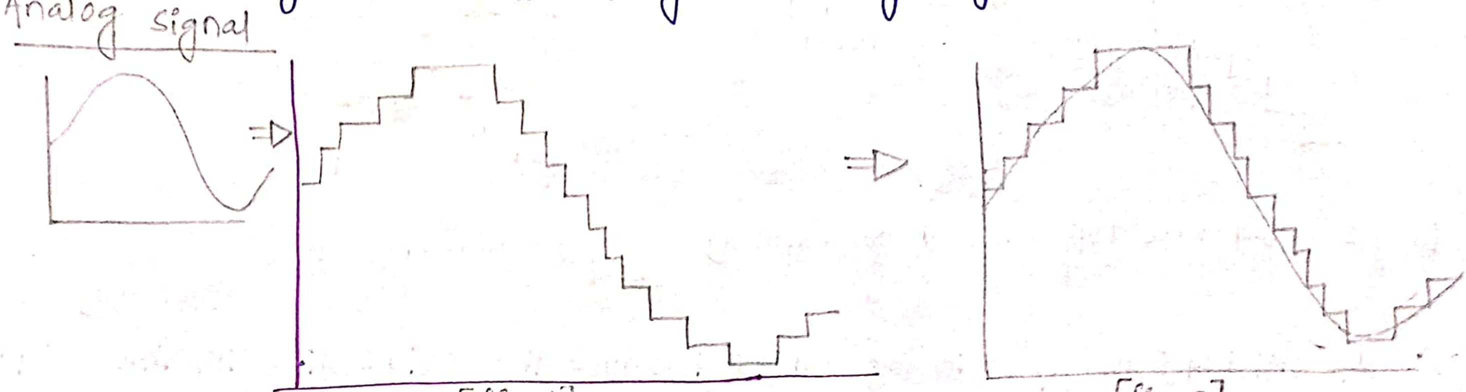
5.4 Concept of Quantization of Signal & Quantization error :-

Quantization is representing the Sampled values of the amplitude by a finite set of levels, which means converting a Continuous-amplitude sample into a discrete-time signal.

→ The discrete amplitudes of the quantized output are called as representation levels or reconstruction levels.

→ The Spacing between the two adjacent representation levels is called a quantum or step-size.

→ The following figure1 shows the resultant quantized signal which is the digital form for the given analog signal.



[fig 1] [fig 2]

→ This is also called as Stair-case waveform, in accordance with its Shape.

→ The above fig 2 shows how an analog signal gets quantized. The plain line represents analog Signal while the Square represents the quantized signal.

→ Both Sampling and quantization result in the loss of information. The quality of a Quantizer output depends upon the number of quantization levels used.

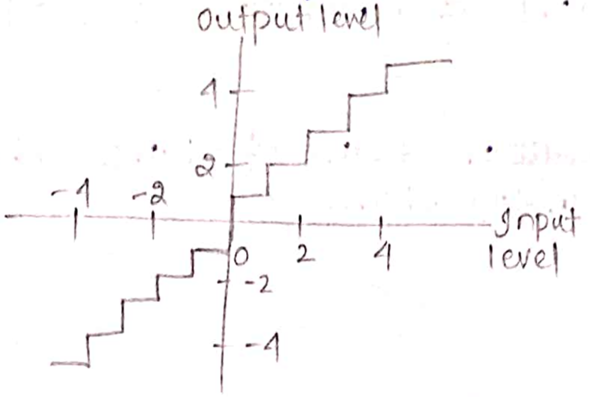
Types of Quantization ⇒

There are two types of Quantization – Uniform Quantization & Non-uniform Quantization

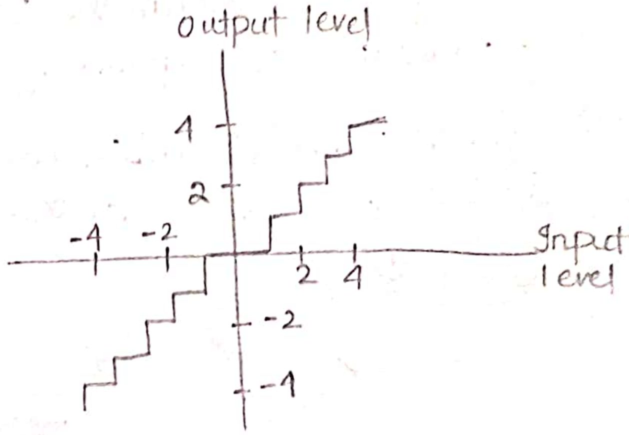
→ The type of quantization in which the quantization levels are uniformly spaced is termed as a Uniform Quantization.

→ The type of quantization in which the quantization levels are unequal and mostly the relation between them is logarithmic, is termed as a Non-uniform Quantization.

→ There are two types of uniform quantization. They are Mid-Rise type and Mid-Tread type. The following figures represent the two types of uniform quantization.



[Fig 1 : Mid-Rise type Uniform Quantization]



[fig 2 : Mid-Tread type Uniform Quantization]

→ The Mid-Rise type is so called because the origin lies in the middle of a raising part of the stair-case like graph. The quantization levels in this type are even in number.

→ The Mid-tread type is so called because the origin lies in the middle of a tread of the staircase like graph. The quantization levels in this type are odd in number.

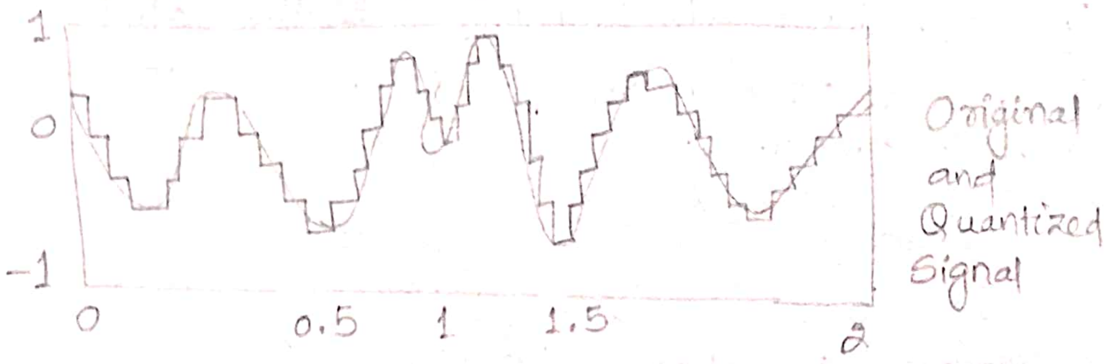
→ Both the mid-rise and mid-tread type of uniform quantizers are symmetric about the origin.

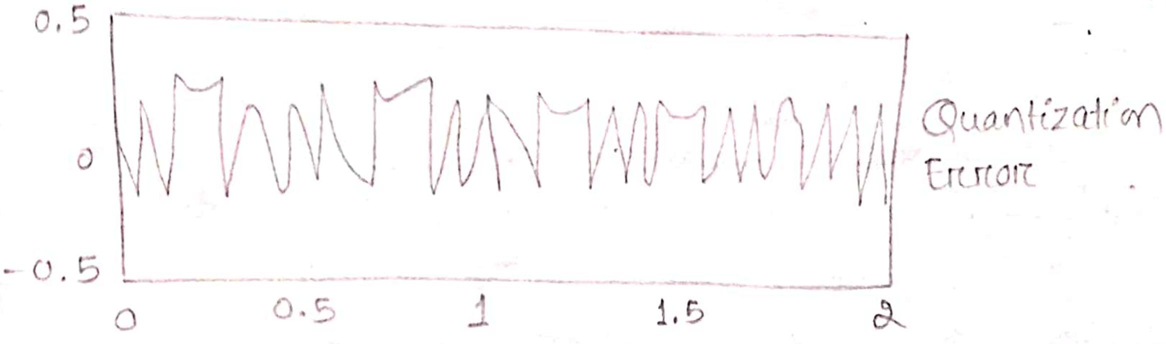
Quantization Error ⟹

The difference between an input value and its quantized value is called a Quantization Error.

→ A Quantizer is a logarithmic function that performs Quantization rounding of the value. An analog to digital converter (ADC) works as a quantizer.

→ The following figure illustrates an example for a quantization error, indicating the difference between the original signal and the quantized signal.





5.5 Generation & Demodulation of PCM System with BLOCK diagram & its applications :—

→ A technique by which analog signal gets converted into digital form in order to have signal transmission through a digital network is known as pulse code Modulation.

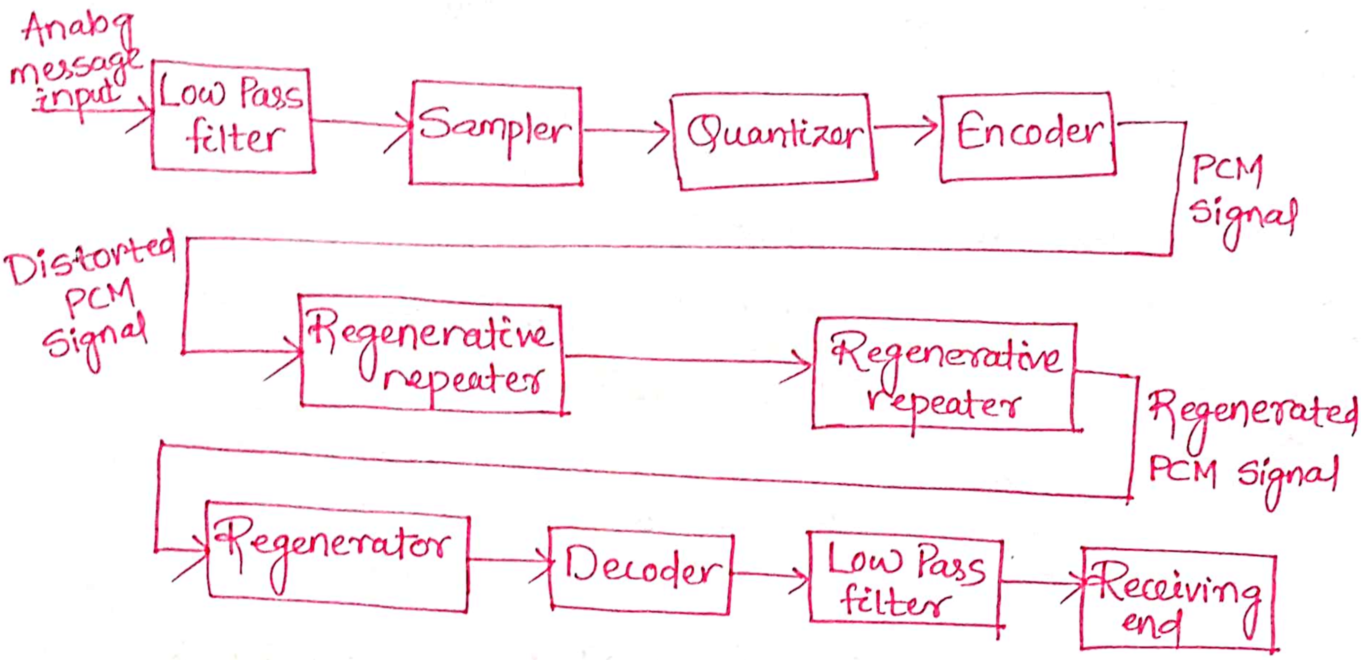
→ PCM systems are basically signal coders also known as waveform coders. PCM allows the representation of the continuous time message signal as a sequence of binary coded pulses. The binary form permits only 2 probable states i.e., 0 and 1.

→ The major steps involved in PCM is sampling, quantizing and encoding.

→ In PCM, the analog message signal is first sampled, and then the amplitude of the sample is approximated to the nearest set of quantization level. This allows the representation of time and amplitude in a discrete manner. Thereby, generating a discrete signal.

→ This discrete signal is then converted into its binary form for the transmission of the signal.

→ In PCM technique the signal gets transmitted in the coded format and must be decoded at the receiver in order to have the original message signal.

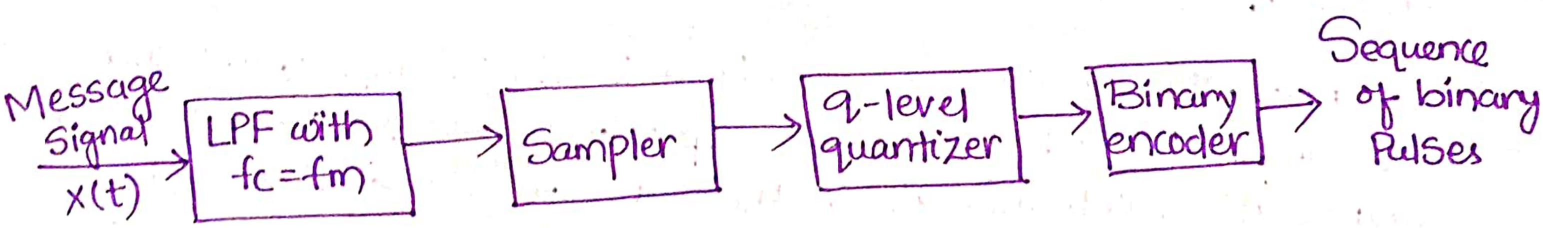


[Block diagram of PCM System]

→ It is basically composed of a transmitter, a transmission path and a receiver. The transmitter performs the sampling, quantizing & encoding of the signal. The transmission path includes regenerative receivers that recover the signal from the undesired noise effects.

→ Lastly, the receiver section that performs decoding of the coded signal after regeneration of the signal at the receiver.

☆ PCM Transmitter ⇒



[PCM transmitter Section]

• LPF : Here, the message signal which is in the continuous time form, is allowed to pass through a low pass filter (LPF). This LPF whose cutoff frequency is f_m elimina high frequency components of the Signal and passes only the frequency Components that lie below fm.

- **Sampler** : The output of the LPF is then fed to a Sampler where the analog input signal is Sampled at regular intervals. The Sampling of the signal is done at the rate of fs. This sampling frequency is so selected that it must follow the Sampling theorem that is expressed as:

$$\boxed{f_s \geq 2f_m}$$

The output of the Sampler is a signal that is discrete time continuous amplitude signal denoted as nTs which is nothing but a PAM signal.

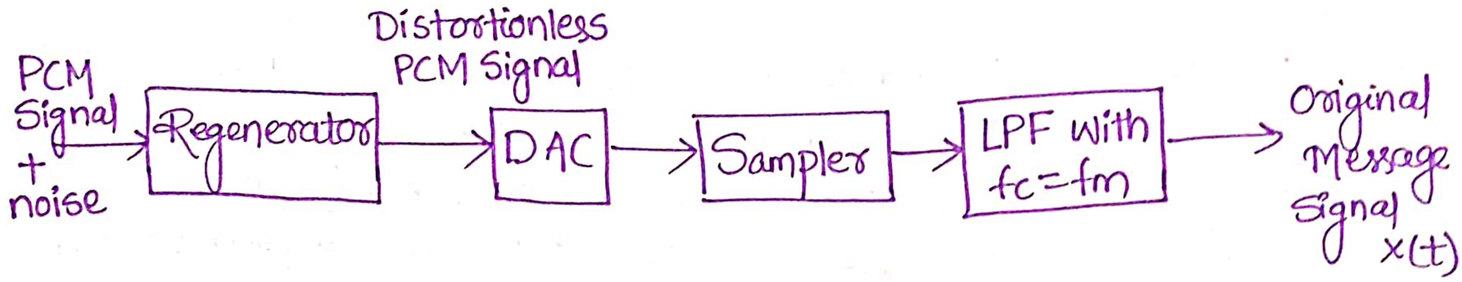
- **Quantizer** : A quantizer is a unit that rounds off each Sample to the nearest discrete level. The Sampler provides a Continuous range Signal and hence still an analog one. The quantizer performs the approximation of each Sample thus assigning it a particular discrete level.

→ As it basically rounds off the value to a Certain level this shows Some variation by the actual amount. Thus we can say, quantizing a Signal introduces some distortion or noise into it. This is known as quantization error.

- **Encoder**: An encoder performs the conversion of the quantized signal into binary codes. This unit generates a digitally encoded signal which is a Sequence of binary pulses that acts as the modulated output.

As it is a binary encoder thus generates a binary coded Sequence. That is transmitted through the transmission path.

☆ PCM Receiver ⟹



[PCM receiver Section]

- **Regenerator** : A regenerative repeater is placed at the receiving end also so as to have an exact PCM transmitted signal. Here, also the regenerator works in a similar manner as that when employed in the transmission path. It eliminates the channel induced noise and reshapes the pulse.

- **DAC and Sampler** : Digital to analog converter performs the conversion of digital signal again into its analog form by making use of the Sampler. As the actual message signal was analog thus at the receiver end there is a necessity to again convert it into its original form.

- **LPF** : The Sampler generates analog signal but that is not the original message signal. Thus, the output of the Sampler is fed to the LPF having cutoff frequency fm. This is sometimes termed as the reconstruction filter that produces the original message signal.

The process done at the transmitter is reversed at the receiver in order to generate the Original analog message signal.

☆ Transmission Bandwidth ⇒

The transmission bandwidth of a PCM System is associated with a number of bits per Sample.

→ If the number of bits per sample increases, the bandwidth also increases.

→ In order to have a good approximation, a large number of levels must be used but that will lead to a larger bandwidth requirement.

→ Let us consider each quantizer level is represented by 'n' binary digits. Then the levels represented by n binary digits is given as,

$$\boxed{q = 2^n}$$

∴ q is the digital level of the quantizer.

→ Every Sample is changed into n bits, thus, a number of bit per Sample is 'n'.

→ As we Know, the number of samples per second is fs. Hence the number of bits per second which is also termed as Signalling rate is given as,

$$\boxed{r = nfs}$$

→ As transmission bandwidth is half the Signalling rate, hence

$$\boxed{BW \geq \frac{1}{2} r}$$

→ Therefore,

$$\boxed{BW \geq \frac{1}{2} nfs} \qquad (\because r = nfs)$$

∴ But we Know, $fs \geq 2fm$

Thus the bandwidth of the PCM System is given as,

$$\boxed{BW \geq nfm}$$

Advantages of PCM :—

1. Immune to channel induced noise and distortion.
2. Repeaters can be employed along the transmitting channel.
3. Encoders allow Secured data transmission.
4. It ensures uniform transmission quality.

Disadvantages of PCM :—

1. Pulse code Modulation increases the transmission bandwidth.
2. A PCM System is Somewhat more Complex than another System.

Applications of PCM :—

(i) With the advent of fibre optic Cables, PCM is used in telephony.

(ii) In Space Communication, space craft transmits signals to earth. Here, the transmitted Power is quite Small (i.e., 10 or 15 W) & the distances are very large (i.e., a few million Km). However, due to the high noise immunity, only PCM systems can be used in Such applications.

5.6. Companding in PCM & Vocoder :—

→ Companding is Non-uniform Quantization.

→ It is required to be implemented to improve SNR of weak Signal.

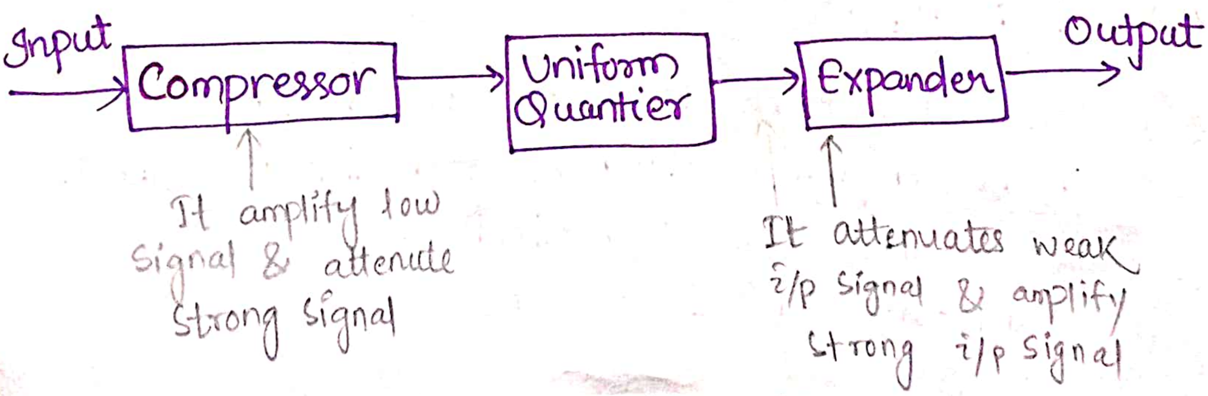
→ Quantization noise is given by,

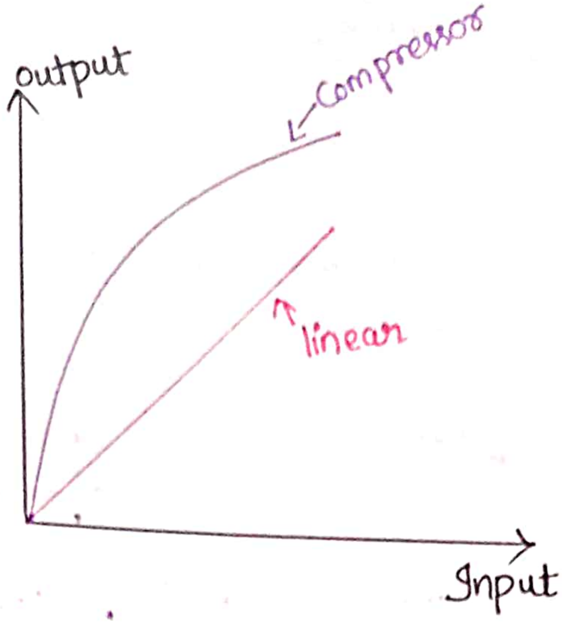
$$\boxed{N_q = \frac{\Delta^2}{12}}$$

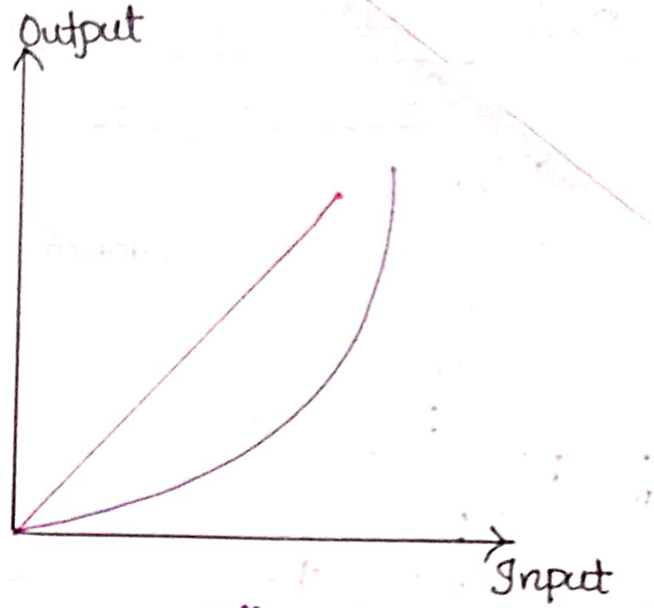
→ For weak Signal noise is Constant.

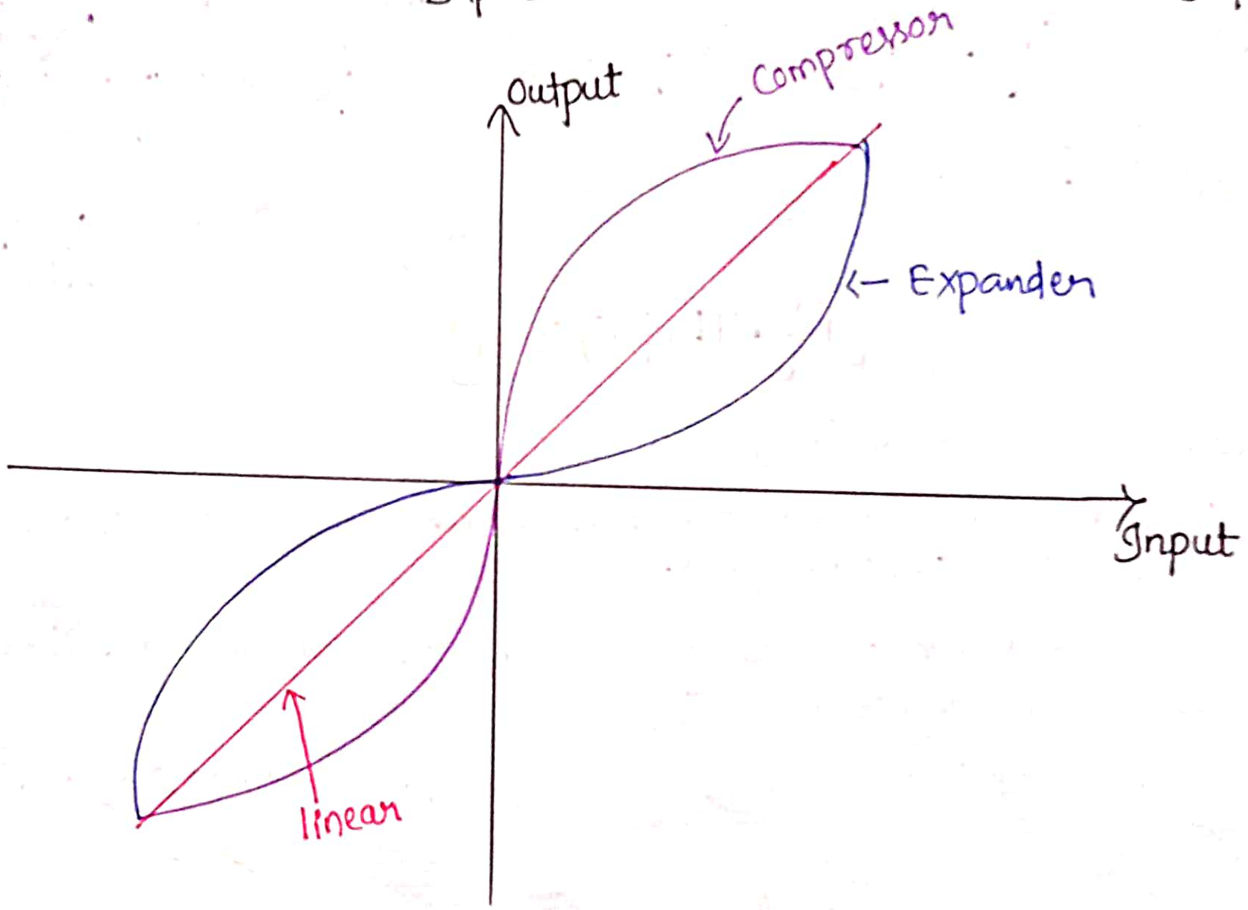
→ Companding is defined from two words.

1. Compression
2. Expansion









<u>Vocoder</u> :— (a Contraction of voice & encoder)

A vocoder is a category of speech coding that analyzes & synthesizes the human voice signal for audio data compression, multiplexing, voice encryption or voice transformation.

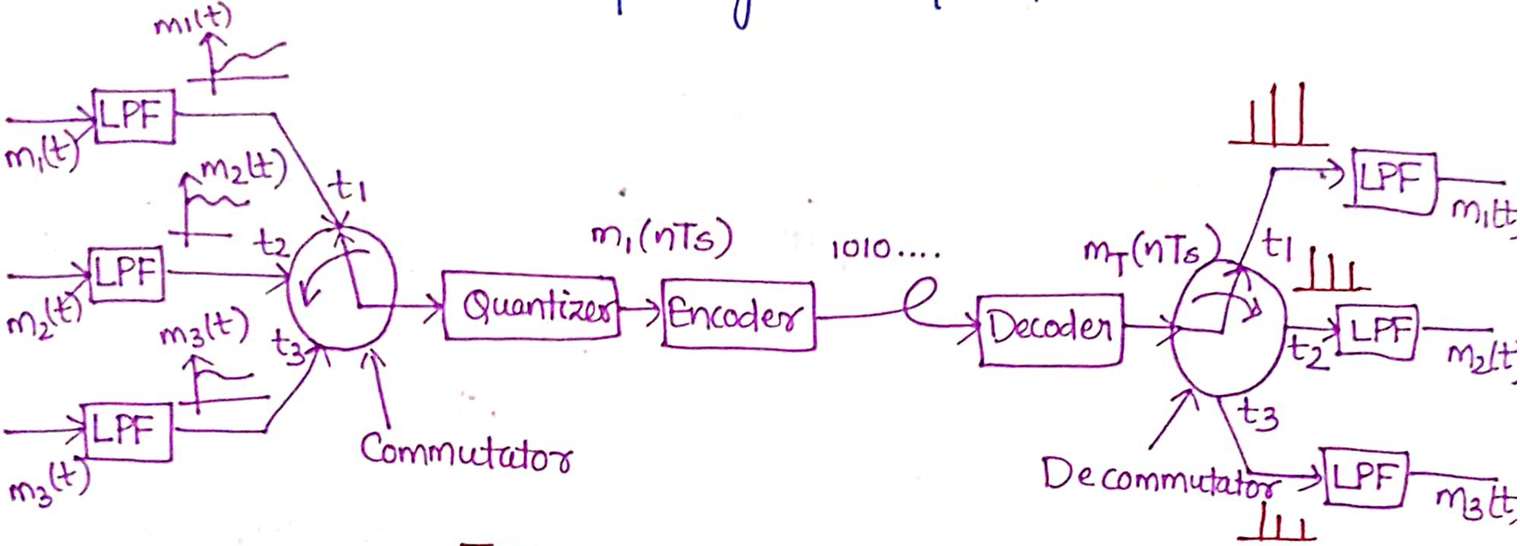
→ By encrypting the control signals, voice transmission can be secured against interception.

→ It is used to reduce the bandwidth of voice information, allowing it to be transferred across further distances.

5.7. Time Division Multiplexing & explain the Operation with Circuit diagram :

→ TDM is used for multiplexing of Digital Signals.

→ When a large number of Signals have to be transmitted over a common channel multiplexing is required.



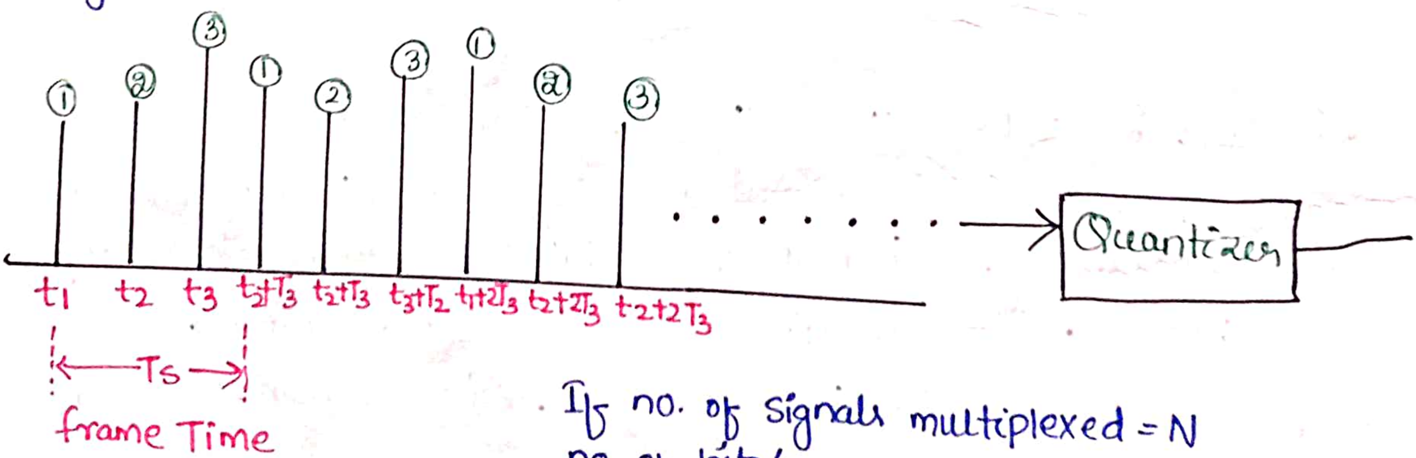
[Fig. TDM System]

→ LPF works as anti-aliasing filter.

→ Commutator is a rotating Switch which rotates in anti-clockwise direction with uniform Speed.

→ Time taken by Commutator to make one complete rotation is Called as frame time(Ts).

→ Commutator is used to Sample multiple number of Continuous Signals.



If no. of signals multiplexed = N

no. of bits/Sample = n

frame Time $T_s = N n T_b$

$T_b = \frac{T_s}{Nn}$ [Where T_b : bit duration]

∴ Bit Rate, $R_b = \frac{1}{T_b} \Rightarrow R_b = \frac{Nn}{T_s}$

$\Rightarrow \boxed{R_b = Nnf_s}$ bits/sec.

If $N = 1$, $R_b = nf_s$ (PCM Case)

→ Decommutator is also a rotating Switch which rotates in clockwise direction with uniform Speed as Commutator.

→ The output of decoder will be quantized Signal, from this quantized signal decommutator extract Samples Corresponding to each of the message Signal and produce at the input of respective LPF.

→ In the Reconstructed Signal, finite amount of Quantization error will be permanantly retained.

→ for proper operation of TDM Speed synchronization to be maintai-ed between commutator & decommutator.

→ To ensure speed Synchronization a no. of additional bits are transmitted at the end of each frame.

Now the frame Time becomes $T_s = (Nn+a)T_b$

$R_b = (Nn+a) f_s$ bits/sec.

10.27 DELTA MODULATION

(Calicut University, Kerala, Sem. Exam., 2006-07)

(i) Reason to use Delta Modulation

We have observed in PCM that it transmits all the bits which are used to code a sample. Hence, signaling rate and transmission channel bandwidth are quite large in PCM. To overcome this problem, Delta Modulation is used.

(ii) Working Principle

Delta modulation transmits only one bit per sample. Here, the present sample value is compared with the previous sample value and this result whether the amplitude is increased or decreased is transmitted. Input signal $x(t)$ is approximated to step signal by the delta modulator. This step size is kept fixed. The difference between the input signal $x(t)$ and staircase approximated signal is confined to two levels, *i.e.*, $+\Delta$ and $-\Delta$. Now, if the difference is positive, then approximated signal is increased by one step, *i.e.*, 'Δ'. If the difference is negative, then approximated signl is reduced by 'Δ'.

When the step is reduced, '0' is transmitted and if the step is increased, '1' is transmitted. Hence, for each sample, only one binary bit is transmitted. Figure 10.23 shows the analog signal $x(t)$ and its staircase approximated signal by the delta modulator.

(iii) Mathematical Expressions

Thus, The principle of delta modulation can be explained with the help of few equations as under:

The error between the sampled value of $x(t)$ and last approximated sample is given as,

$$e(nT_s) = x(nT_s) - \hat{x}(nT_s) \qquad \text{...(10.45)}$$

where
$e(nT_s)$ = error at present sample
$x(nT_s)$ = sampled signal of $x(t)$
$\hat{x}(nT_s)$ = last sample approximation of the staircase waveform.

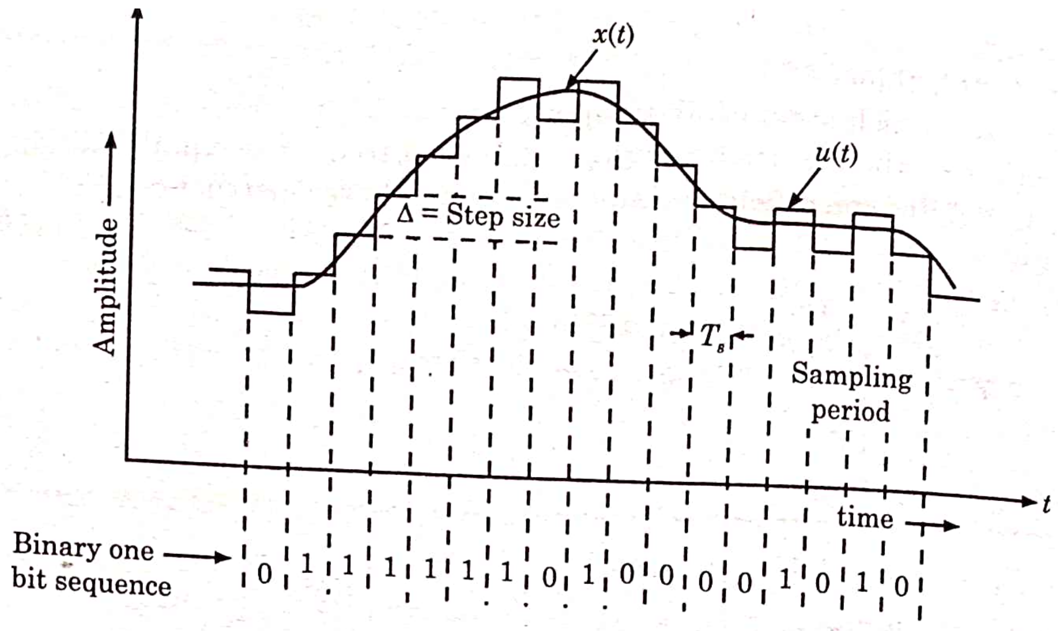


Fig. 10.23 *Delta modulation waveform.*

If we assume $u(nT_s)$ as the present sample approximation of staircase output,

then, $u[(n-1)T_s] = \hat{x}(nT_s)$ ...(10.46)

= last sample approximation of staircase waveform

Let us define a quantity $b(nT_s)$ in such a way that,

$$b(nT_s) = \Delta \; sgn \; [e(nT_s)] \quad \text{...(10.47)}$$

This means that depending on the sign of error $e(nT_s)$, the sign of step size Δ is decided. In other words, we can write

$$b(nT_s) = \begin{cases} +\Delta & \text{if } x(nT_s) \geq \hat{x}(nT_s) \\ -\Delta & \text{if } x(nT_s) < \hat{x}(nT_s) \end{cases} \quad \text{...(10.48)}$$

DO YOU KNOW?

Delta modulation transmits only one bit per sample, indicating whether the signal level is increacing or decreasing, but it needs a higher sampling rate than PCM for equivalent results.

Also, If $b(nT_s) = +\Delta$ then a binary '1' is transmitted

and if $b(nT_s) = -\Delta$ then a binary '0' is transmitted.

Here, T_s = Sampling interval.

(*iv*) Transmitter Part

Figure 10.24 (*a*) shows the transmitter (*i.e.*, generation of Delta Modulated signal).

The summer in the accumulator adds quantizer output ($\pm \Delta$) with the previous sample approximation. This gives present sample approximation. *i.e.*,

$$u(nT_s) = u(nT_s - T_s) + [\pm \Delta]$$

or

$$u(nT_s) = u[(n-1)\,T_s] + b(nT_s) \quad \text{...(10.49)}$$

The previous sample approximation $u[(n-1)\,T_s]$ is restored by delaying one sample period T_s. The sampled input signal $x(nT_s)$ and staircase approximated signal $\hat{x}(nT_s)$ are subtracted to get error signal $e(nT_s)$.

Thus, depending on the sign of $e(nT_s)$, one bit quantizer generates an output of $+\Delta$ or $-\Delta$. If the step size is $+\Delta$, then binary '1' is transmitted and if it is $-\Delta$, then binary '0' is transmitted.

(*v*) Receiver Part

At the receiver end, shown in figure 10.24(*b*), the accumulator and low-pass filter (LPF) are used. The accumulator generates the staircase approximated signal output and is delayed by one sampling period T_s. It is then added to the input signal. If input is binary '1' then it adds $+\Delta$ step to the previous output (which is delayed). If input is binary '0' then one step 'Δ' is subtracted from the delayed signal. Also, the low-pass filter has the cutoff frequency equal to highest frequency in $x(t)$. This low-pass filter smoothens the staircase signal to reconstruct orignal message signal $x(t)$.

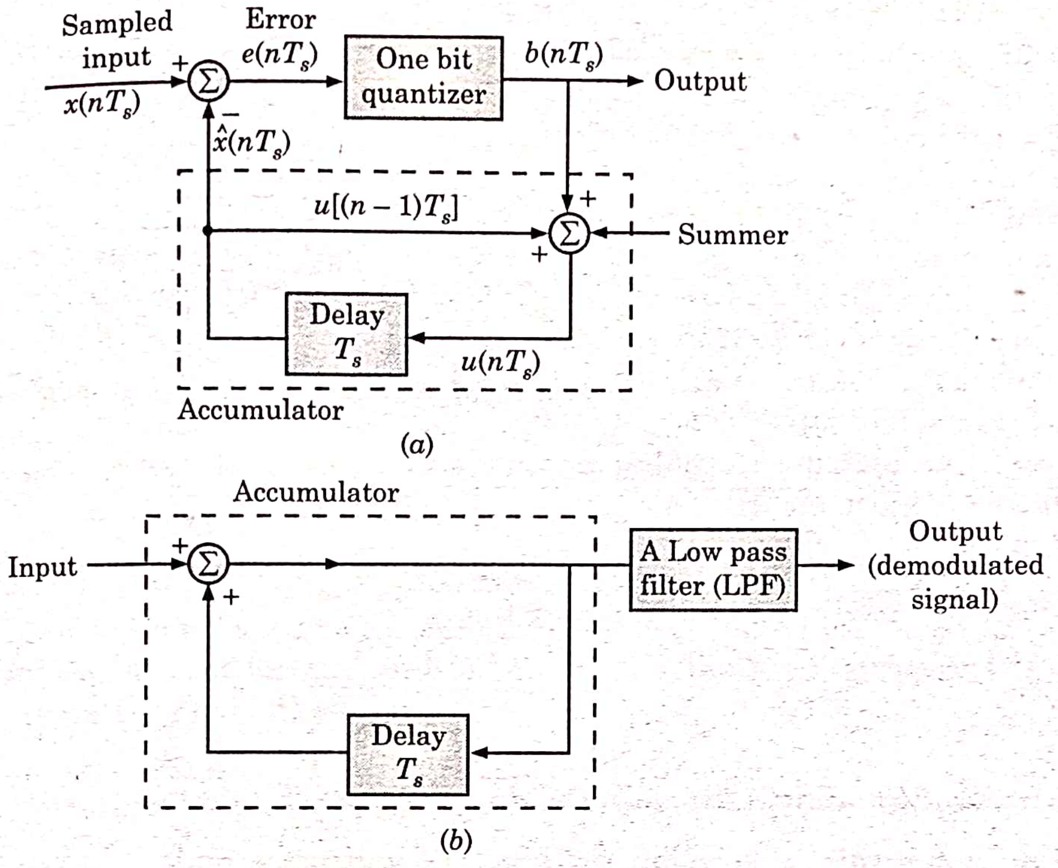


Fig. 10.24 (a) A Delta modulation transmitter (b) A Delta modulation receiver

10.27.1. Advantages of Delta Modulation : Salient Features of Delta Modulation

The delta modulation has certain advantages over PCM as under:

(*i*) Since, the delta modulation transmits only one bit for one sample, therefore the signaling rate and transmission channel bandwidth is quite small for delta modulation compared to PCM.

(*ii*) The transmitter and receiver implementation is very much simple for delta modulation. There is no analog to digital converter required in delta modulation.

10.27.2. Drawbacks of Delta Modulation *(Very Important)*

The delta modulation has two major drawbacks as under:

(*i*) Slope overload distortion,

(*ii*) Granular or idle noise

Now, let us discuss these two drawbacks in detail.

(*i*) Slope Overload Distortion

This distortion arises because of large dynamic range of the input signal.

As can be observed from figure 10.25, the rate of rise of input signal $x(t)$ is so high that the staircase signal cannot approximate it, the step size 'Δ' becomes too small for staircase signal

$u(t)$ to follow the step segment of $x(t)$. Hence, there is a large error between the staircase approximated signal and the original input signal $x(t)$. This error or noise is known as **slope overload distortion.** To reduce this error, the step size must be increased when slope of signal $x(t)$ is high. Since the step size of delta modulator remains fixed, its maximum or minimum slopes occur along straight lines. Therefore, this modulator is also known as **Linear Delta Modulator (LDM).**

(ii) Granular or Idle Noise

Granular or Idle noise occurs when the step size is too large compared to small variations in the input signal. This means that for very small variations in the input signal, the staircase signal is changed by large amount (Δ) because of large step size. Figure 10.25 shows that when the input signal is almost flat, the staircase signal $u(t)$ keeps on oscillating by $\pm\Delta$ around the signal. The error between the input and approximated signal is called **granular noise.** The solution to this problem is to make step size small.

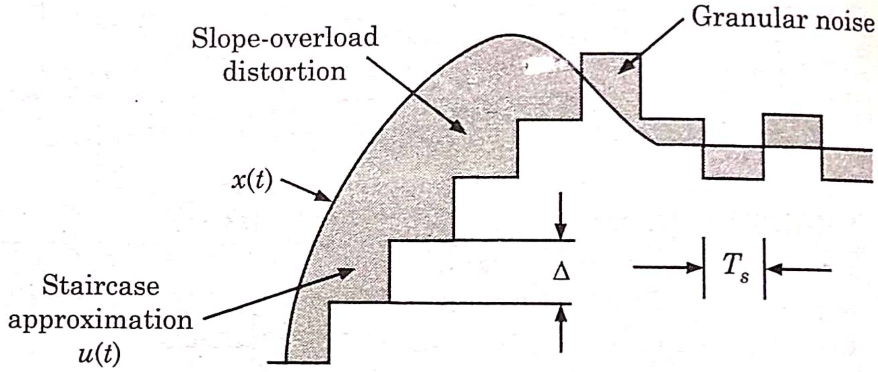


Fig. 10.25 *Quantization errors in delta modulation.*

10.31 DIFFERENTIAL PULSE CODE MODULATION (DPCM)

(GTU, Gujrat, Sem. Exam., 2005-2006)

(i) Reason to use DPCM

It may be observed that the samples of a signal are highly correlated with each other. This is due to the fact that any signal does not change fast. This means that 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 a little difference. When these samples are encoded by a standard PCM system, the resulting encoded signal contains some redundant information. Figure 10.32 illustrates this **redundant information.**

(ii) Redundant Information in PCM

Figure 10.32 shows a continuous time signal $x(t)$ by dotted line. This signal is sampled by flat top sampling at intervals T_s, $2T_s$, $3T_s$ ... nT_s. The sampling frequency is selected to be higher than nyquist rate. The samples are encoded by using 3 bit (7 levels) PCM. The sample is quantized to the nearest digital level as shown by small circles in the figure 10.32. The encoded binary value of each sample is written on the top of the samples. We can observe from figure 10.32 that the samples taken at $4T_s$, $5T_s$ and $6T_s$ are encoded to same value of (110). This information can be carried only by one sample. But three samples are carrying the same information means that it is redundant. Consider another example of samples taken at $9T_s$ and 10 T_s. The difference between these samples only due to last bit and first two bits are redundant, since they do not change.

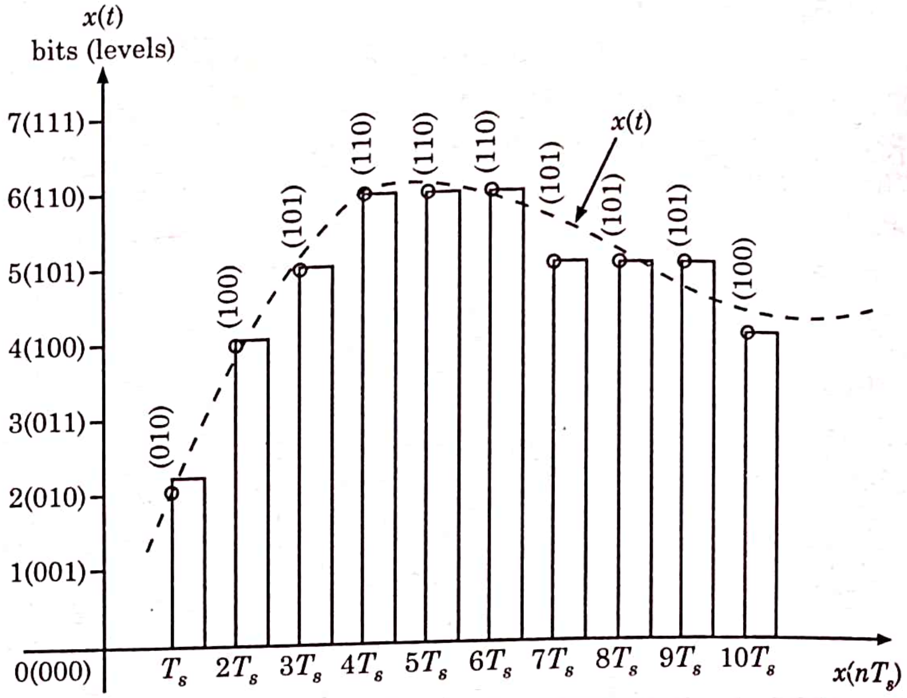


Fig. 10.32 *Illustration of redundant information in PCM.*

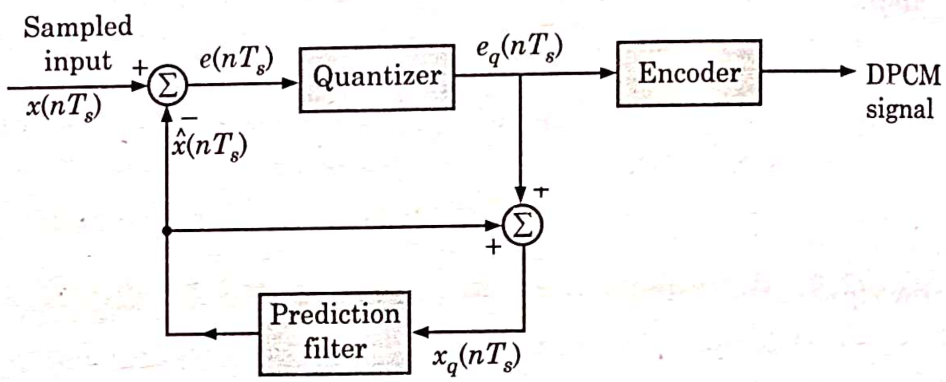


Fig. 10.33 *A Differential pulse code modulation transmitter.*

If this redundancy is reduced, then overall bit rate will decrease and number of bits required to transmit one sample will also be reduced. This type of digital pulse modulation scheme is known as **Differential Pulse Code Modulation (DPCM).**

(*iii*) Working Principle

In fact the differential pulse code modulation works on the principle of prediction. The value of the present sample is predicted from the past samples. The prediction may not be exact but it is very close to the actual sample vlaue. Figure 10.33 shows the transmitter of Differential Pulse Code Modulation (DPCM) system. The sampled signal is denoted by $x(nT_s)$ and the predicted signal is denoted by $\hat{x}(nT_s)$. The comparator finds out the difference between the actual sample value $x(nT_s)$ and predicated sample value $\hat{x}(nT_s)$. This is known as Prediction error and it is denoted by $e(nT_s)$. It can be defined as,

$$e(nT_s) = x(nT_s) - \hat{x}(nT_s) \qquad ...(10.50)$$

Thus, error is the difference between unquantized input sample $x(nT_s)$ and prediction of it $\hat{x}(nT_s)$. The predicted value is produced by using a prediction filter. The quantizer output signal gap $e_q(nT_s)$ and previous prediction is added and given as input to the prediction filter. This signal is called $x_q(nT_s)$. This makes the prediction more and more close to the actual sampled signal. We can observe that the quantized error signal $e_q(nT_s)$ is very small and can be encoded by using small number of bits. Thus number of bits per sample are reduced in DPCM.

The quantizer output can be written as,

$$e_q(nT_s) = e(nT_s) + q(nT_s) \qquad ...(10.51)$$

Here, $q(nT_s)$ is the quantization error. As shown in figure (10.33), the prediction filter input $x_q(nT_s)$ is obtained by sum $\hat{x}(nT_s)$ and quantizer output *i.e.*,

$$x_q(nT_s) = \hat{x}(nT_s) + e_q(nT_s) \qquad ...(10.52)$$

Substituting the value of $e_q(nT_s)$ from equation (10.51) in the above equation, we get,

$$x_q(nT_s) = \hat{x}(nT_s) + e(nT_s) + q(nT_s) \qquad ...(10.53)$$

Equation (10.50) is written as,

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

$$\therefore \quad e(nT_s) + \hat{x}(nT_s) = x(nT_s) \qquad ...(10.54)$$

Therefore, the value of $e(nT_s) + \hat{x}(nT_s)$ from above equation into equation (10.53), we get,

$$x_q(nT_s) = x(nT_s) + q(nT_s) \qquad ...(10.55)$$

Important Point: *Hence, the quantized version of the signal $x_q(nT_s)$ is the sum of original sample value and quantization error $q(nT_s)$. The quantization error can be positive or negative. Thus equation (10.55) does not depend on the prediction filter characteristics.*

(*iv*) Reception of DPCM Signal : Reconstruction of DPCM Signal

Figure 10.34 shows the block diagram of DPCM receiver.

The decoder first reconstructs the quantized error signal from incoming binary signal. The prediction filter output and quantized error sigals are summed up to give the quantized version of the orginal signal. Thus the signal at the receiver differs from actual signal by quantization error $q(nT_s)$, which is introduced permanently in the reconstructed signal.

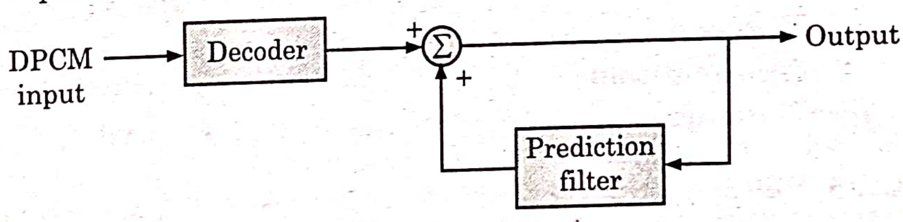


Fig. 10.34 *DPCM receiver*

Table 10.1 **Comparison between PCM, Delta Modulation, Adaptive Delta Modulation and Differential Pulse Code Modulation**

S. No.	*Parameter of comparison*	*Pulse Code Modulation (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 and step size	The number of levels depend on number of bits. Level size is kept fixed.	Step size is kept fixed and cannot be varied.	According to the signal variation, step size varies (*i.e.* Adapted).	Here, 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 are present.	Quantization noise is present but other errors are absent.	Slope overload distor-tion and quantization noise is present.
4.	Transmission bandwidth	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.	Here, Feedback exists.
6.	Complexity of implementation	System complex.	Simple.	Simple.	Simple

Unit : 6 - DIGITAL MODULATION TECHNIQUES

11.2 MULTIPLEXING*

Multiplexing may be defined as a technique which allows many users to share a common communication channel simultaneously. There are two major types of multiplexing techniques. They are as under:

(*i*) Frequency division multiplexing (FDM),

(*ii*) Time division multiplexing (TDM).

11.2.1. Frequency Division Multiplexing (FDM)

(*i*) Definition

This technique permits a fixed frequency band to every user in the complete channel bandwidth. Such frequency slot is allotted continuously to that user. As an example consider that the channel bandwidth is 1 MHz. Let there be ten users, each requiring upto 100 kHz bandwidth. Then the complete channel bandwidth of 1 MHz can be divided into ten frequency bands, *i.e.* each of 100 kHz and every user can be allotted one independent frequency band. This technique is known as **Frequency Division Multiplexing** (FDM).

(ii) Main Application Area

It is mainly used for modulated signal. This is due to the fact that a modulated signal can be placed in any frequency band by just changing the carrier frequency. However, at the receiver, these frequency multiplexed signals can be separated by the use of tuned circuits (*i.e.*, bandpass filters) of their respective frequency band. And for every band, there are independent tuned circuits and demodulators.

> **DO YOU KNOW?**
>
> The number of simultaneous conversation's that can be transmitted using FDM depends on the total bandwidth available which in turn, varies with the medium.

11.2.2. Time Division Multiplexing (TDM)

(i) Definition

As discussed earlier, in PAM, PPM and PDM, the pulse is present for a short duration and for most of the time between the two pulses, no signal is present. This free space between the pulses can be occupied by pulses from other channels. This is known as Time Division Multiplexing (TDM). Thus, time division multiplexing (TDM) makes maximum utilization of the transmission channel.

(ii) Comparision with FMM

Hence, we can say that in FDM, all the signals are transmitted simultaneously over the same communication medium, and the signals occupy frequency slots. However, in TDM, the signals to be multiplexed are transmitted sequentially one after the other. Each signal occupies a short time slot as shown in figure 11.1. Thus, the signals are isolated from each other in the time domain, but all of them occupy the same slot in the frequency spectrum. Therefore, in TDM, the complete bandwidth of the communication channel is available to each signal being transmitted.

(iii) Conceptual Diagram

Figure 11.1 explains the concept of TDM.

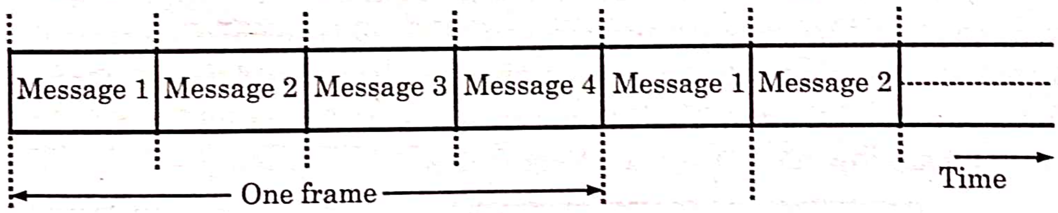


Fig. 11.1 *Illustration of TDM concept*

(iv) Concep of Frame in TDM

At this stage, it may be noted that in context of TDM, we define one important term *i.e.*, frame. One frame corresponds to the time period required to transmit all the signals once on the transmission channel. This has been shown in figure 11.1. Here, we have total four message signals to be transmitted. Hence, one frame will correspond to the time period required to transmit all the four signals once on the channel. The TDM system can be used to multiplex analog or digital signals, however it is more suitable for the digital signal multiplexing.

14.2 DIGITAL MODULATION FORMATS

When we have to transmit a digital signal over a long distance, we need continuous-wave (CW) modulation. For this purpose, the transmission medium can be in form of radio, cable or other type of channel. Also, a carrier signal having some frequency f_c is used for modulation. Then the modulating digital signal modulates some parameter like frequency, phase or amplitude of the carrier. Due to this process, there is some deviation in carrier frequency f_c. This deviation is known as the bandwidth of the channel. This means that the channel has to transmit some range or band of frequencies. Such type of transmission is known as bandpass transmission and the communication channel is known as bandpass channel.

Here, the word bandpass is used since the range of frequencies does not start from zero Hz to f_m Hz. In fact, the range of frequencies from zero Hz to f_m Hz is known as **low-pass signal** and such channel is known as **low-pass channel.**

Now, when it is required to transmit digital signals on a bandpass channel, the amplitude, frequency or phase of the sinusoidal carrier is varied in accordance with the incoming digital data. Since the digital data is in discrete steps, the modulation of the bandpass sinusoidal carrier is also done in discrete steps. Due to this reason, this type of modulation (*i.e.*, Digital modulation) is also known as switching or signaling. Now, if an amplitude of the carrier is switched depending on the input digital signal, then it is called Amplitude shift keying (ASK).

This process is quite similar to analog amplitude modulation. If the frequency of the sinusoidal carrier is switched depending upon the input digital signal, then it is known as the frequency shift keying (FSK). This is very much similar to the analog frequency modulation. If the phase of the carrier is switched depending upon the input digital signal, then it is called phase shift keying (PSK). This is similar to phase modulation. Since the phase and frequency modulation has constant amplitude envelope, therefore FSK and PSK also has a constant amplitude envelope. Because of constant amplitude of FSK and PSK, the effect of non-linearities, noise interference is minimum on signal detection. However, these effects are more pronounced on ASK. Therefore, FSK and PSK are preferred over ASK.

Figure 14.1 shows the waveforms for amplitude-shift keying, phase-shift keying and frequency shift keying. In these waveforms, a single feature of the carrier (*i.e.*, amplitude, phase or frequency) undergoes modulation.

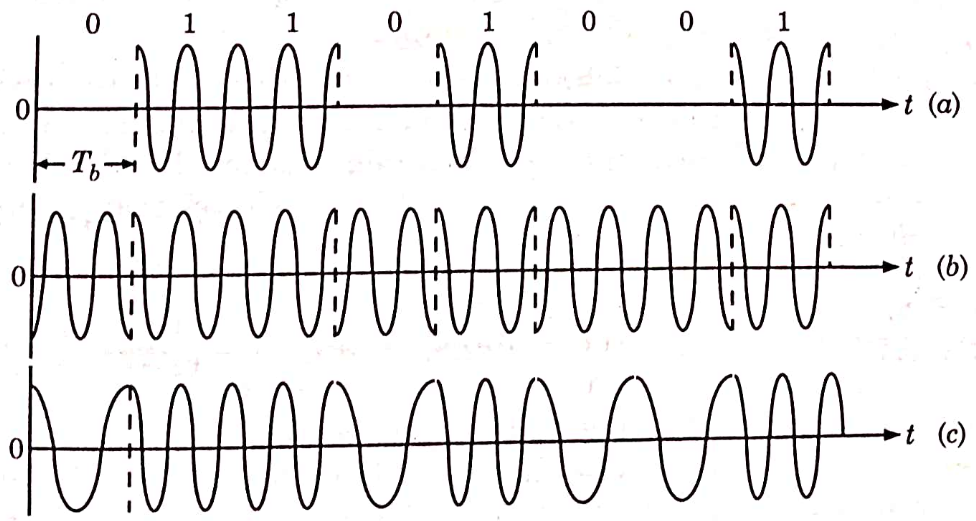


Fig. 14.1 *The three basic forms of signaling binary information, (a) Amplitude-shift keying, (b) Phase-shift keying, (c) Frequency shift keying with continuous phase*

In digital modulations, instead of transmitting one bit at a time, we transmit two or more bits simultaneously. This is known as M-ary transmission. This type of transmission results in reduced channel bandwidth. However, sometimes, we use two quadrature carriers for modulation. This process is known as **Quadrature modulation.**

Thus, we see that there are a number of modulation schemes available to the designer of a digital communication system required for data transmission over a bandpass channel.

Every scheme offers system trade-offs of its own. However, the final choice made by the designer is determined by the way in which the available primary communication resources such as transmitted power and channel bandwidth are best exploited. In particular, the choice is made in favour of a scheme which possesses as many of the following design characteristics as possible:

(*i*) Maximum data rate,

(*ii*) Minimum probability of symbol error,

(*iii*) Minimum transmitted power,

(*iv*) Maximum channel bandwidth,

(*v*) Maximum resistance to interfering signals,

(*vi*) Minimum circuit complexity.

14.3 TYPES OF DIGITAL MODULATION TECHNIQUES

(GGSIPU Delhi, Sem. Examination, 2003-2004)

Basically, digital modulation techniques may be classified into coherent or non-coherent techniques, depending on whether the receiver is equipped with a phase-recovery circuit or not. The phase-recovery circuit ensures that the oscillator supplying the locally generated carrier wave receiver is synchronized* to the oscillator supplying the carrier wave used to originally modulate the incoming data stream in the transmitter.

(*i*) Coherent Digital Modulation Techniques

Coherent digital modulation techniques are those techniques which employ coherent detection. In coherent detection, the local carrier generated at the receiver is phase locked with the carrier at the transmitter. Thus, the detection is done by correlating received noisy signal and locally generated carrier. The coherent detection is a synchronous detection.

DO YOU KNOW?

Digital transmission uses frequency, phase, and amplitude variations, just as does analog transmission.

(*ii*) Non-coherent Digital Modulation Techniques

Non-coherent digital modulation techniques are those techniques in which the detection process does not need receiver carrier to be phase locked with transmitter carrier. The advantage of such type of system is that the system becomes simple. But the drawback of such a system is that the error probability increases. In fact, the different digital modulation techniques are used for various specific application areas.

14.5 COHERENT BINARY AMPLITUDE SHIFT KEYING OR ON-OFF KEYING

(i) Definition

Amplitude shift keying (ASK) or ON-OFF keying (OOK) is the simplest digital modulation technique. In this method, there is only one unit energy carrier and it is switched on or off depending upon the input binary sequence.

Expression and Waveforms

The ASK waveform may be represented as,

$$s(t) = \sqrt{2P_s}\cos(2\pi f_c t) \quad \text{(To transmit '1')} \qquad ...(14.1)$$

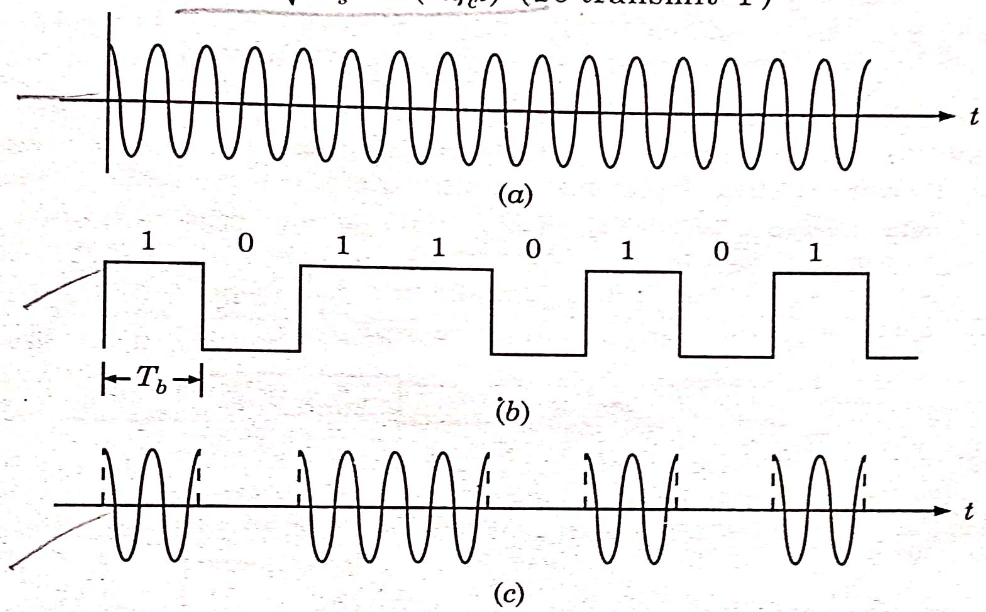


Fig. 14.2 *Amplitude-shift keying waveforms, (a) Unmodulated carrier, (b) NRZ Unipolar bit sequence, (c) ASK waveform.*

To transmit symbol '0', the signal $s(t) = 0$ *i.e.*, no signal is transmitted. Signal $s(t)$ contains some complete cycles of carrier frequency 'f_c'.

Hence, the ASK waveform looks like an ON-OFF of the signal. Therefore, it is also known as the **ON-OFF keying** (OOK). Figure 14.2 shows the ASK waveform.

14.5.1. Signal Space Diagram of ASK

The ASK waveform of equation (14.1) for symbol '1' can be represented as,

$$s(t) = \sqrt{P_s T_b}\cdot\sqrt{2/T_b}\cos(2\pi f_c t) = \sqrt{P_s T_b}\,\phi_1(t) \qquad ...(14.2)$$

This means that there is only one carrier function $\phi_1(t)$. The signal space diagram will have two points on $\phi_1(t)$. One will be at zero and other will be at $\sqrt{P_s T_b}$. Figure 14.3 shows this aspect.

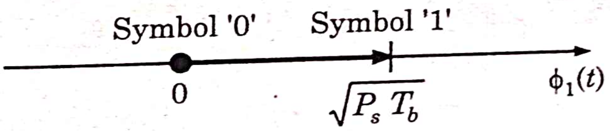


Fig. 14.3 *Signal space diagram of ASK.*

Thus, the distance between the two signal points is,

$$d = \sqrt{P_s T_b} = \sqrt{E_b} \qquad ...(14.3)$$

14.5.2. Generation of ASK Signal

(i) Description and Working Operation

ASK signal may be generated by simply applying the incoming binary data (represented in unipolar form) and the sinusoidal carrier to the two inputs of a product modulator (*i.e.*, balanced modulator). The resulting output will be the ASK waveform. This is shown in figure 14.4. Modulation causes a shift of the baseband signal spectrum.

(ii) Power Spectral Density (psd)

The ASK signal, which is basically the product of the binary sequence and the carrier signal, has a power spectral density (PSD) same as that of the baseband on-off signal but shifted in the frequency domain by $\pm f_c$. This is shown in figure 14.5. It may be noted that two impulses occur at $\pm f_c$.

(iii) Bandwidth of BASK

The spectrum of the ASK signal shows that it has an infinite bandwidth. However for practical purpose, the bandwidth is often defined as the bandwidth of an ideal bandpass filter centered at f_c whose output contains about 95% of the total average power content of the ASK signal. It may be proved that according to this criterion the bandwidth of the ASK signal is approximately $3/T_b$ Hz. The bandwidth of the ASK signal can however, be reduced by using smoothed versions of the pulse waveform instead of rectangular pulse waveforms.

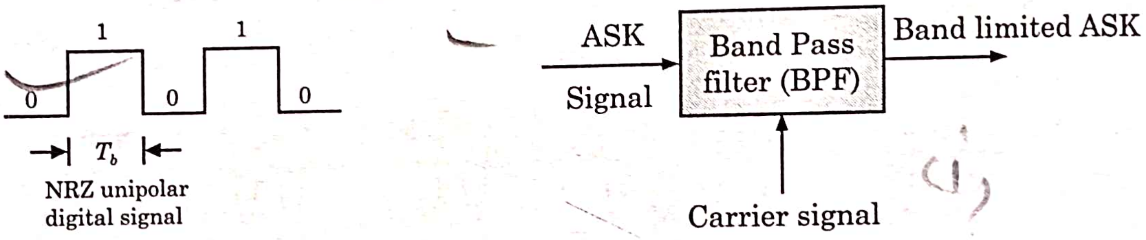


Fig. 14.4 *Generation of binary ASK waveform*

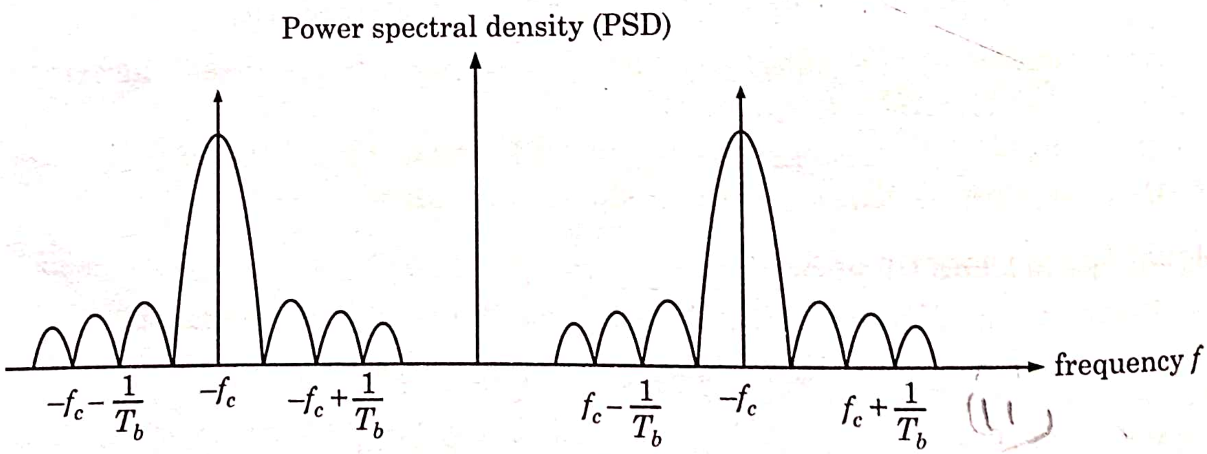


Fig. 14.5 *Power spectral density of ASK signal.*

14.5.3. BASK Reception : Coherent Detection or Demodulation of Binary ASK Signal

(i) Working Operation

The demodulation of binary ASK waveform can be achieved with the help of *coherent detector* as shown in figure 14.6. It consists of a product modulator which is followed by an integrator and a decision-making device. The incoming ASK signal is applied to one input of the product modulator. The other input of the product modulator is supplied with a sinusoidal carrier which is generated with the help of a local oscillator. The output of the product modulator goes to input of the integrator. The integrator operates on the output of the multiplier for successive bit intervals and essentially performs a low-pass filtering action. The output of the integrator goes to the input of a decision-making device.*

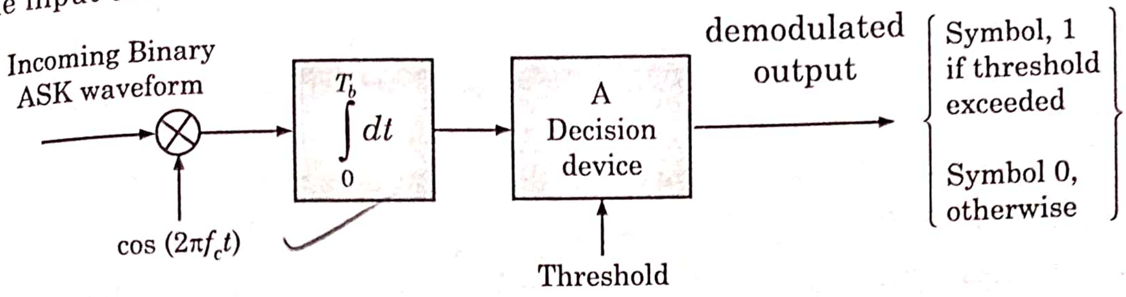


Fig. 14.6 *Coherent detection of binary ASK signals.*

Now, the decision-making device compares the output of the integrator with a preset threshold. It makes a decision in favour of symbol 1 when the threshold is exceeded and in favour of symbol 0 otherwise. The *coherent detection* makes the use of linear operation. In this method we have assumed that the local carrier is in perfect synchronisation with the carriers used in the transmitter. This means that the frequency and phase of the locally generated carrier is same as those of the carriers used in the transmitter.

DO YOU KNOW?

Straight forward amplitude-shift keying (ASK) is rare in digital communication unless we count Morse code, but quadrature AM (QAM) is very common.

(ii) Synchronization Requirement

The following two forms of synchronisation are required for the operation of coherent (or synchronous detector):

(i) *Phase synchronisation which ensures that carrier wave generated locally in the receiver is locked in phase with respect to one that is employed in the transmitter.*

(ii) *Timing synchronisation which enable proper timing of the decision making operation in the receiver with respect to switching instants (switching between 1 and 0) in the original binary data.*

14.5.4. Salient Feature of BASK

The advantage of using BASK is its simplicity. It is easy to generate and detect.

14.5.5. Drawback

But the drawback of BASK is that it is very sensitive to noise, therefore, it finds limited application in data transmission. It is used at very low bit rates, upto 100 bits per sec.

14.5.6. Bit Error Rate (BER) or Probability of Error

As a matter of fact, bit error rate (BER) or probability of error is a very important parameter. This parameter is used to judge the performance of a digital communication system. It is represented by P_e. P_e must be as small as possible.

14.7 COHERENT BINARY FREQUENCY SHIFT KEYING (BFSK)

(JNTU, Hyderabad, Sem. Exam; 2005-06)

In binary frequency shift keying (BFSK), the frequency of a sinusoidal carrier is shifted according to the binary symbol. In other words, the frequency of a sinusoidal carrier is shifted between two discrete values. However, the phase of the carrier is unaffected. This means that we have two different frequency signals according to binary symbols. Let there be a frequency shift by Ω. Then we can write following equations.

If $\quad b(t) = \text{'1'}, \quad$ then $s_H(t) = \sqrt{2P_s}\cos(2\pi f_c + \Omega)t$ ...(14.25)

If $\quad b(t) = \text{'0'}, \quad$ then $s_L(t) = \sqrt{2P_s}\cos(2\pi f_c - \Omega)t$ ...(14.26)

Hence, there is increase or decrease in frequency by Ω. Let us use the following conversion table to combine above two FSK equations:

Table 14.2. Conversion table for BPSK representation

b(t) Input	*d(t)*	$P_H(t)$	$P_L(t)$
1	+ 1V	+ 1V	0V
0	– 1V	0V	+ 1V

The equations (14.25) and (14.26) combinely may be written as

$$s(t) = \sqrt{2P_s}\cos\left[(2\pi f_c + d(t)\Omega)t\right] \quad ...(14.27)$$

Hence, if symbol '1' is to be transmitted, the carrier frequency will be $f_c + \left(\frac{\Omega}{2\pi}\right)$ and is represented by f_H. If symbol '0' is to be transmitted, then the carrier frequency will be $f_c - \left(\frac{\Omega}{2\pi}\right)$ and is represented by f_L.

Therefore, we have

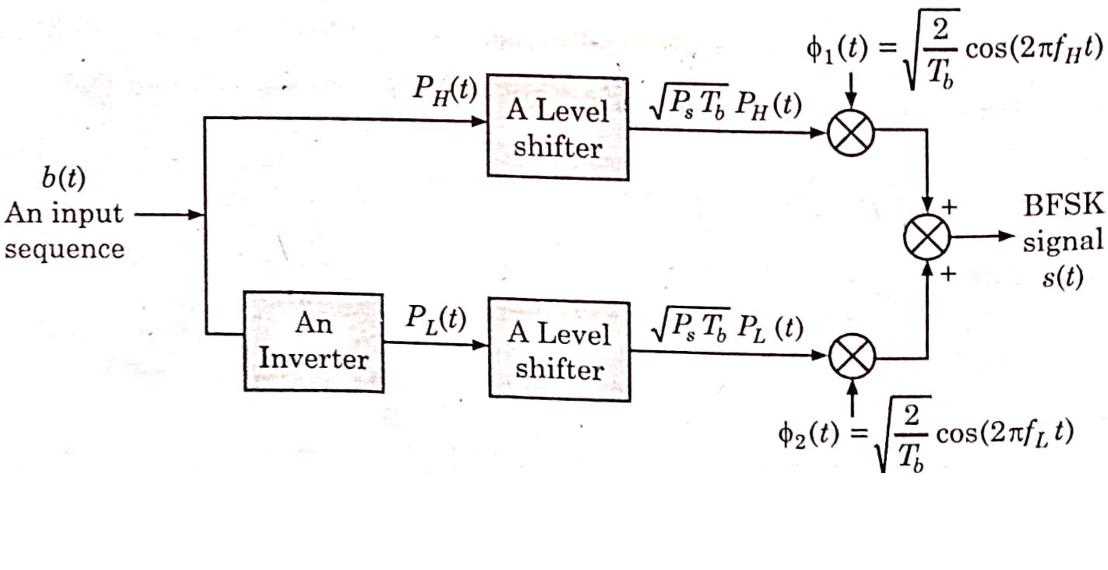
Thus, $\quad f_H = f_c + \frac{\Omega}{2\pi} \quad$ for symbol '1' ...(14.28)

$\quad f_L = f_c - \frac{\Omega}{2\pi} \quad$ for symbol '0' ...(14.29)

14.7.1. Generation of BFSK

(MDU, Rohtak, Sem. Exam; 2005-06) (10 marks)

It may be observed from Table 14.1 that $P_H(t)$ is same as $b(t)$ and also $P_L(t)$ is inverted version of $b(t)$. The block diagram for BFSK generation is shown in figure 14.14.



We know that input sequence $b(t)$ is same as $P_H(t)$. An inverter is added after $b(t)$ to get $P_L(t)$. The level shifter $P_H(t)$ and $P_L(t)$ are unipolar signals. The level shifter converts the '+1' level to $\sqrt{P_sT_b}$. Zero level is unaffected. Thus, the output of the level shifters will be either $\sqrt{P_sT_b}$ (if '+1') or zero (if input is zero). In other words, when a binary '0' is to be transmitted, $P_L(t) = 1$ and $P_H(t) = 0$, and for a binary '1' to be transmitted, $P_H(t) = 1$ and $P_L(t) = 0$. Hence, the ransmitted signal will have a frequency of either f_H or f_L. Further, there are product modulators after level shifter. The two carrier signals $\phi_1(t)$ and $\phi_2(t)$ are used. $\phi_1(t)$ and ϕ_2 (t) are orthogonal o each other. In one bit period of input signal (*i.e.*, T_b), ϕ_1 (t) or $\phi_2(t)$ have integral number of ycles.

Thus, the modulated signal is having continuous phase. Figure 14.15 shows such type of FSK signal. The adder then adds the two signals.

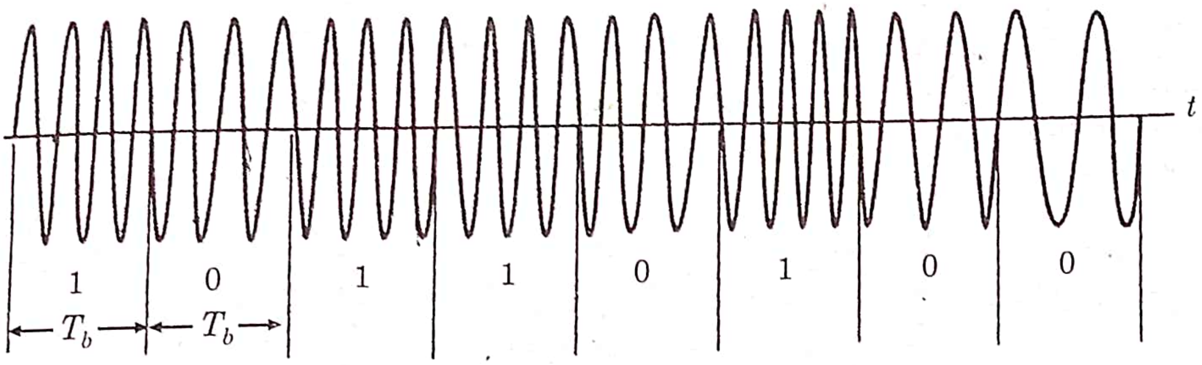


Fig. 14.15 *The BFSK signal.*

14.7.3. Bandwidth of BFSK Signal

From figure 14.16, it is obvious that the width of one lobe is $2f_b$. The two main lobes due to f_H and f_L are placed such that the total width due to both main lobes is $4\ f_b$.

Therefore, we have

Bandwidth of BFSK = $2f_b + 2f_b$

or

$$BW = 4f_b \quad ...(14.37)$$

Now, if we compare this bandwidth with that of BPSK, we note that,

$$BW\ (BFSK) = 2 \times BW(BPSK) \quad ...(14.38)$$

14.7.4. BFSK Receiver: Coherent Detection of BFSK*

Figure 14.17 shows the block diagram of a scheme for demodulation of BFSK wave using coherent detection technique. The detector consists of two correlators that are individually tuned to two different carrier frequencies to represent symbols '1' and '0'. A correlator consists of a multiplier followed by an integrator. Then, the received binary FSK signal is applied to the multipliers of both the correlators. To the other input of the multipliers, carriers with frequency f_{c1} and f_{c2} are applied as shown in figure 14.17. The multiplied output of each multiplier is subsequently passed through integrators generating output l_1 and l_2 in the two paths. The

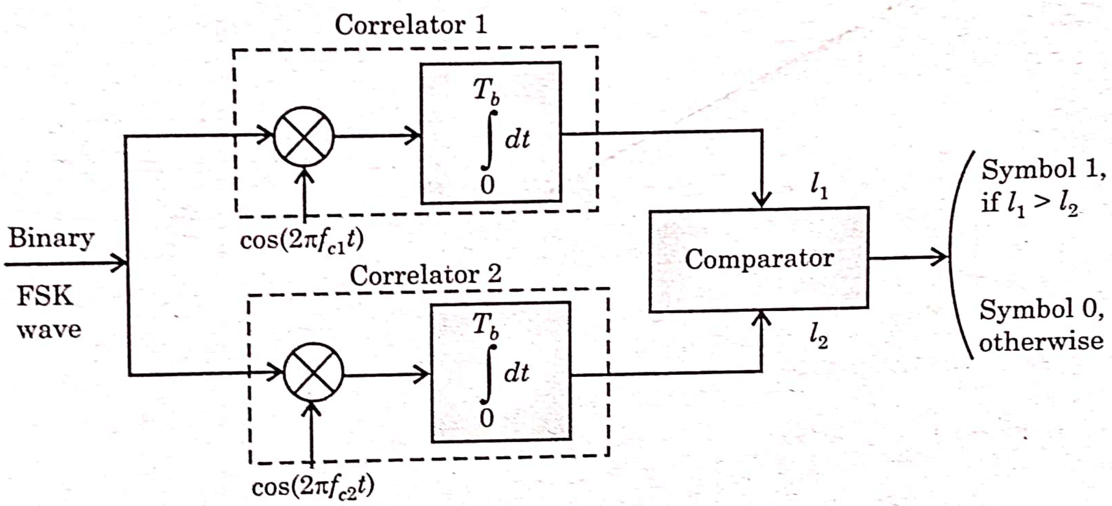


Fig. 14.17 *Block diagram of BFSK receiver (detection of BFSK).*

output of the two integrators are then fed to the decision making device. The decision making device is essentially a comparator which compares the output l_1 (in the upper path) and output l_2 (in the lower path). If the output l_1 produced in the upper path (associated with frequency f_{c1}) is greater than the output l_2 produced in the lower path (associated with frequency f_{c2}), the detector makes a decision in favour of symbol 1. If the output l_1 is less than l_2, then the decision making device decides in favour of symbol 0 (say). This type of digital communication receivers are also called *correlation receivers*. As discussed earlier, the detector based upon coherent detection requires phase and timing synchronisation.

14.6 BINARY PHASE SHIFT KEYING (BPSK)*

(PTU, Jalamdbar, Sem. Exam., 2006-07)

(i) Definition

Binary phase shift keying (BPSK) is the most efficient of the three digital modulation, *i.e.*, ASK, FSK and PSK. Hence, binary phase shift keying (BPSK) is used for high bit rates. In BPSK, phase of the sinusoidal carrier is changed according to the data bit to be transmitted. Also, a bipolar NRZ signal is used to represent the digital data coming from the digital source.

(ii) Expression for BPSK

In a binary phase shift keying (BPSK), the binary symbols '1' and '0' modulate the phase of the carrier. Let us assume that the carrier is given as,

$$s(t) = A\cos(2\pi f_c t) \quad \text{...(14.4)}$$

Here 'A' represents peak value of sinusoidal carrier. For the standard 1 Ω load resistor, the power dissipated would be,

$$P = \frac{1}{2}A^2$$

or

$$A = \sqrt{2P} \quad \text{...(14.5)}$$

Now, when the symbol is changed, then the phase of the carrier will also be changed by an amount of 180 degrees (*i.e.*, π radians). Let us consider, for example,

For symbol '1', we have

$$s_1(t) = \sqrt{2P}\cos(2\pi f_c t) \quad \text{...(14.6)}$$

If next symbol is '0', then we have

For symbol '0', we have

$$s_2(t) = \sqrt{2P}\cos(2\pi f_c t + \pi) \quad \text{...(14.7)}$$

Now, because $\cos(\theta + \pi) = -\cos\theta$, therefore, the last equation can be written as,

$$s_2(t) = -\sqrt{2P}\cos(2\pi f_c t) \quad \text{...(14.8)}$$

With the above equation, we can define BPSK signal combinely as,

$$s(t) = b(t)\sqrt{2P}\cos(2\pi f_c t) \quad \text{...(14.9)}$$

where $b(t) = +1$ when binary '1' is to be transmitted.

-1 when binary '0' is to be transmitted

(iii) Binary Sequence and its Equivalent Signal $b(t)$

Figure 14.7 illustrates binary sequence and its equivalent signal $b(t)$.

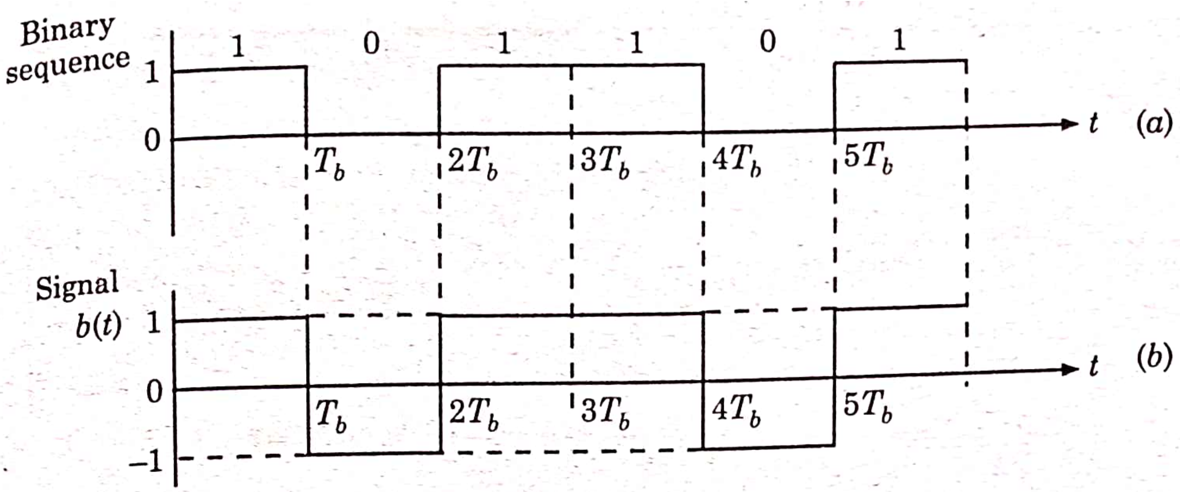


Fig. 14.7 *(a) Binary sequence, (b) The corresponding bipolar signal b(t) .*

14.6.1. Generation of BPSK Signal *(Important)*

BPSK signal may be generated by applying carrier signal to a balanced modulator. The binary data signal (0s and 1s) is converted into a NRZ bipolar signal by an NRZ encoder. Here, the bipolar signal $b(t)$ is applied as a modulating signal to the balanced modulator.

Figure 14.8 shows the block diagram of a BPSK signal generator.

A NRZ level encoder converts the binary data sequence into bipolar NRZ signal.*

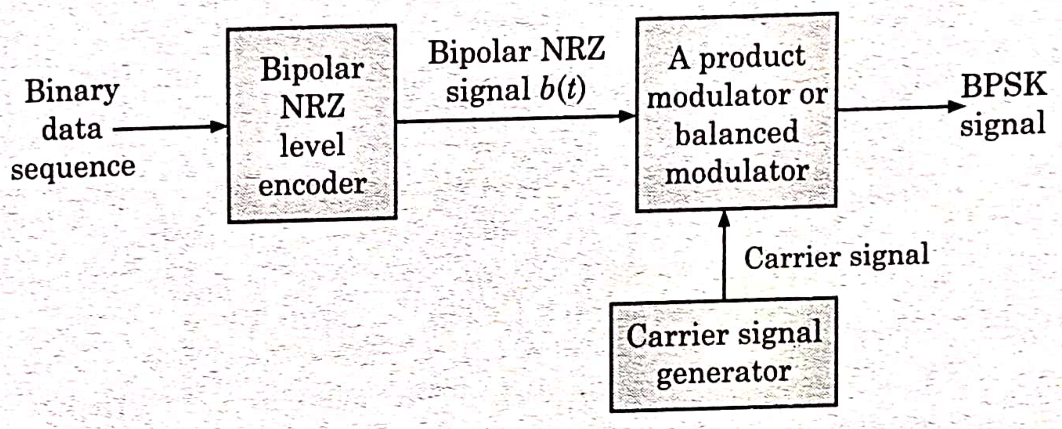


Fig. 14.8 *Generation of BPSK.*

Table 14.1. shows input digital and corresponding bipolar NRZ signal.

S.No.	*Input digital signal*	*Bipolar NRZ signal b(t)*	*BPSK outpt signal*
1.	Binary 0	$b(t) = -1$	$-\sqrt{2P}\cos\omega_c t$
2.	Bianry 1	$b(t) = +1$	$+\sqrt{2P}\cos\omega_c t$

In above table,

(i) $P = \dfrac{E_b}{T_b}$, where, E_b is the signal energy and T_b is the bit duration

(ii) Also, $\omega_c = 2\pi f_c$

14.6.2. Reception of BPSK Signal : Coherent Detection

Figure 14.9 shows the block diagram of the scheme to recover baseband signal from BPSK signal. The transmitted BPSK signal is given as

$$s(t) = b(t)\sqrt{2P}\cos(2\pi f_c t)$$

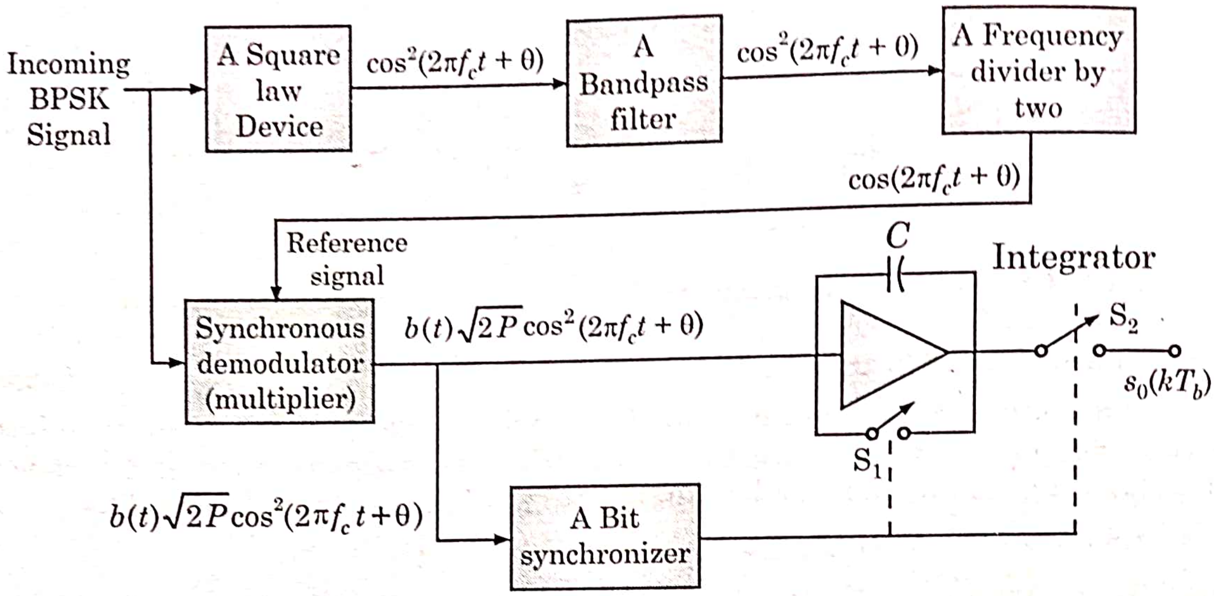


Fig. 14.9 *Reception of baseband signal in BPSK signal.*

This signal undergoes the phase change depending upon the time delay from transmitter end to receiver end. This phase change is, usually, a fixed phase shift in the transmitted signal.

Let us consider that this phase shift is θ. Because of this, the signal at the input of the receiver can be written as

$$s(t) = b(t)\sqrt{2P}\cos(2\pi f_c t + \theta) \quad \text{...(14.10)}$$

Now, from this received signal, a carrier is separated because this is coherent detection. As shown in the figure 14.9, the received signal is allowed to pass through a square law device. At the output of the square law device, we get a signal which is given as

$$\cos^2(2\pi f_c t + \theta)$$

Here, it may be noted that we have neglected the amplitude, since we are only interested in the carrier of the signal.

Again, we know that

$$\cos^2\theta = \frac{1+\cos 2\theta}{2}$$

Therefore, we have

$$\cos^2(2\pi f_c t + \theta) = \frac{1+\cos 2(2\pi f_c t+\theta)}{2} = \frac{1}{2} + \frac{1}{2}\cos 2(2\pi f_c t+\theta)$$

Here, $\frac{1}{2}$ represents a DC level. This signal is then allowed to pass through a bandpass filter (BPF) whose passband is centred around $2f_c$. Bandpass filter removes the DC level of $\frac{1}{2}$ and at the output, we obtain,

$$\cos 2(2\pi f_c t + \theta)$$

This signal is having frequency equal to 2 f_c. Hence, it is passed through a frequency divider by two. Thus, at the output of frequency divider, we get a carrier signal whose frequency is f_c *i.e.*, cos $(2\pi f_c t + \theta)$.

The synchronous (*i.e.*, coherent) demodulator multiplies the input signal and the recovered carrier. Hence, at the output of multiplier, we get

$$b(t)\sqrt{2P}\cos(2\pi f_c t + \theta) \times \cos(2\pi f_c t + \theta) = b(t)\sqrt{2P}\cos^2(2\pi f_c t + \theta)$$

$$= b(t)\sqrt{2P} \times \frac{1}{2}[1 + \cos 2(2\pi f_c t + \theta)]$$

$$= b(t)\sqrt{\frac{P}{2}}[1 + \cos 2(2\pi f_c t + \theta)] \quad ...(14.11)$$

This signal is then applied to the bit synchronizer and integrator. The integrator integrates the signal over one bit period. The bit synchronized takes care of starting and ending times of a bit. At the end of bit duration T_b, the bit synchronizer closes switch S_2 temporarily. This connects the output of an integrator to the decision device. In fact, it is equivalent to sampling the output of integrator. The synchronizer then opens switch S_2 and switch S_1 is closed temporarily. This resets the integrator voltage to zero. The integrator then integrates next bit. Let us assume that one bit period 'T_b' contains integral number of cycles of the carrier. This means that the phase change occurs in the carrier only at zero crossing. This has been shown in figure 14.10. This BPSK waveform has full cycles of sinusoidal carrier.

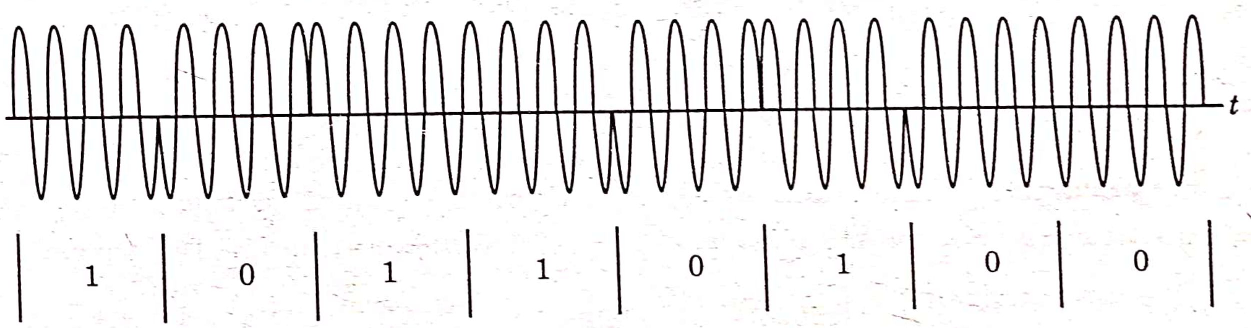


Fig. 14.10 *The BPSK waveform.*

Table 14.3.

S.N.	*Parameter of comparison*	*Binary ASK*	*Binary FSK*	*Binary PSK*
1.	Variable characteristic	Amplitude	Frequency	Phase
2.	Bandwidth (Hz) (spectral efficiency)	$2 f_b$	$4 f_b$	$2 f_b$
3.	Noise immunity	low	high	high
4.	Probability of error	high	low	low
5.	Performance in presence of noise	poor	Better than ASK	Best of three
6.	System complexity	Simple	Moderately complex	Very complex
7.	Bit rate or data rate	Suitable upto 100 bits/sec.	Suitable upto about 1200 bits/sec.	Suitable for high bit rates
8.	Demodulation method	Envelope detection	Envelope detection	Coherent detection

14.12 QUADRATURE PHASE SHIFT KEYING (QPSK)

(Anna University, Chennai, Sem. Exam., 2005-06)

As a matter of fact, in communication systems, we have two main resources. These are the transmission power and the channel bandwidth. The channel bandwidth depends upon the bit rate or signalling rate f_b. In digital bandpass transmission, we use a carrier for transmission. This carrier is transmitted over a channel. If two or more bits are combined in some symbols, then the signalling rate will be reduced. Thus, the frequency of the carrier needed is also reduced. This reduces the transmission channel bandwidth. Hence, because of grouping of bits in symbols, the transmission channel bandwidth can be reduced. In quadrature phase shift keying (QPSK), two successive bits in the data sequence are grouped together. This reduces the bits rate or signalling rate (*i.e.*, f_b) and thus reduces the bandwidth of the channel.*

In case of BPSK, we know that when symbol changes the level, the phase of the carrier is changed by 180°. Because, there were only two symbols in BPSK, the phase shift occurs in two levels only. However, in QPSK, two successive bits are combined. Infact, this combination of two bits forms four distinct symbols. When the symbol is changed to next symbol, then the phase of the carrier is changed by 45° ($\pi/4$ radians). Table 14.7 shows these symbols and their phase shifts.

Table 14.7. Symbol and corresponding phase shifts in QPSK

S. No.	Input successive bits		Symbol	Phase shift in carrier
1	1(1 V)	0 (–1 V)	S_1	$\pi/4$
2	0(–1 V)	0(–1 V)	S_2	$3\pi/4$
3	0(–1 V)	1(1 V)	S_3	$5\pi/4$
4	1(1 V)	1(1 V)	S_4	$7\pi/4$

Hence as shown in Table 14.7, there are four symbols and the phase is shifted by $\pi/4$ for each symbol.

14.13 GENERATION OF QPSK

Figure 14.24 shows the block diagram of QPSK transmitter. Here, the input binary sequence is first converted to a bipolar NRZ type of signal. This signal is denoted by $b(t)$. It represents binary '1' by + 1 V and binary '0' by – 1 V. This signal has been shown in figure 14.25(*a*). The demultiplexer divides $b(t)$ into two separate bit streams of the odd numbered and even numbered bits. Here, $b_e(t)$ represents even numbered sequence and $b_0(t)$ represents odd numbered sequence. The symbol duration of both of these odd and even numbered sequences is $2T_b$. Hence, each symbol consists of two bits. Figure 14.25(*b*) and (*c*) illustrate the waveform of $b_e(t)$ and $b_0(t)$.

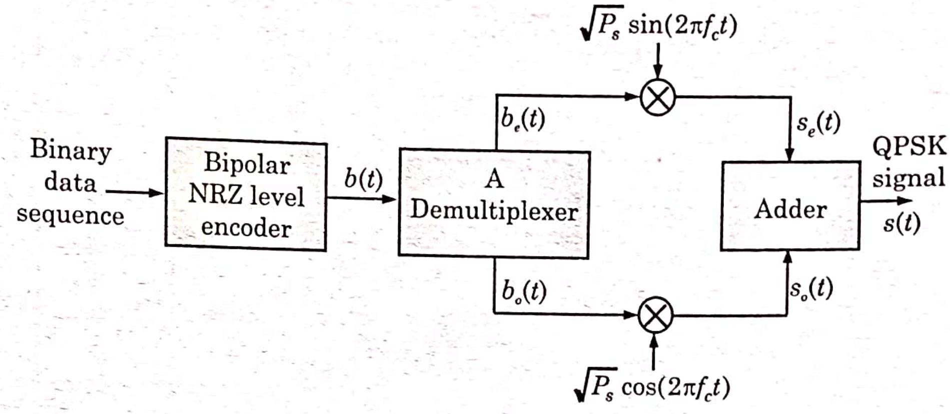


Fig. 14.24 *Generation of QPSK.*

It may be observed that the first even bit occurs after the first odd bit. Hence, even numbered bit sequence $b_e(t)$ starts with the delay of one bit period due to first odd bit. Thus, first symbol of $b_e(t)$ is delayed by one bit period 'T_b' with respect to first symbol of $b_0(t)$. This delay of T_b is known as offset. This shows that the change in levels of $b_e(t)$ and $b_0(t)$ cannot occur at the same time due to offset or staggering.

Also, the bit steam $b_e(t)$ modulates carrier $\sqrt{P_s}\cos(2\pi f_c t)$ and $b_0(t)$ modulates $\sqrt{P_s}\sin(2\pi f_c t)$. These modulators are the balanced modulators. The two carriers $\sqrt{P_s}\cos(2\pi f_c t)$ and $\sqrt{P_s}\sin(2\pi f_c t)$ have been shown in figure 14.25(*d*) and (*e*). There carriers are also known as **quadrature carriers.**

> **DO YOU KNOW?**
>
> In QPSK, each symbol represents two bits and the bit rate is twic the baud rate. This is called a dibit system.

The two modulated signals can be written as,

$$s_e(t) = b_e(t)\sqrt{P_s}\sin(2\pi f_c t) \qquad ...(14.50)$$

and

$$s_0(t) = b_0(t)\sqrt{P_s}\cos(2\pi f_c t) \qquad ...(14.51)$$

Hence, $s_e(t)$ and $s_0(t)$ are basically BPSK signals. The only difference is that $T = 2T_b$ here. The value of $b_e(t)$ and $b_0(t)$ would be $+ 1V$ or $- 1V$. Figure 14.25 (*f*) and (*g*) shows the waveforms of $s_e(t)$ and $s_0(t)$. The adder in figure 14.24 adds these two signals $b_e(t)$ and $b_0(t)$.

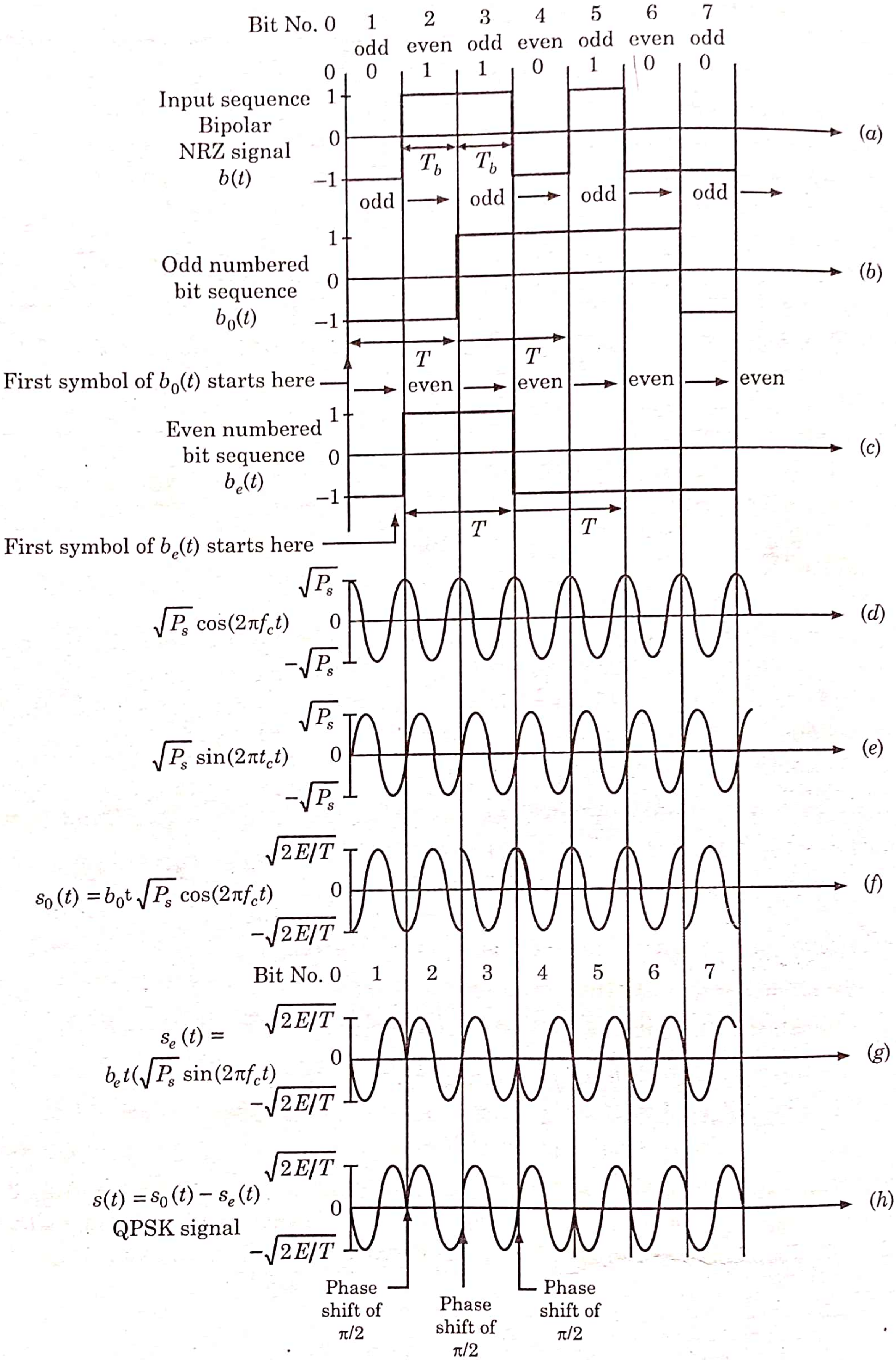


Fig. 14.25 *QPSK waveforms, (a) Input sequence and its corresponding NRZ waveform, (b) Odd numbered bit sequence and its corresponding waveform (c) Even numbered bit sequence and its NRZ waveform (d) Basis function $f_1(t)$ (e) Basis function $f_2(t)$ (f) Binary PSK waveform for odd numbered channel (g) Binary PSK waveform for even numbered channel (h) Final QPSK waveform.*

The output of the adder is QPSK signal and it is given by,

$$s(t) = s_0(t) + s_e(t)$$

or

$$s(t) = b_0(t)\sqrt{P_s}\cos(2\pi f_c t) + b_e(t)\sqrt{P_s}\sin(2\pi f_c t) \qquad \ldots(14.52)$$

Figure 14.25(h) shows the QPSK signal represented by equation (14.52). In QPSK signal in figure 14.25(h), if there is any phase change, it occurs at minimum duration of T_b. This is because the two signals $s_e(t)$ and $s_0(t)$ have an offset of 'T_b'. Due to this offset, the phase shift in QPSK signal is $\frac{\pi}{2}$.

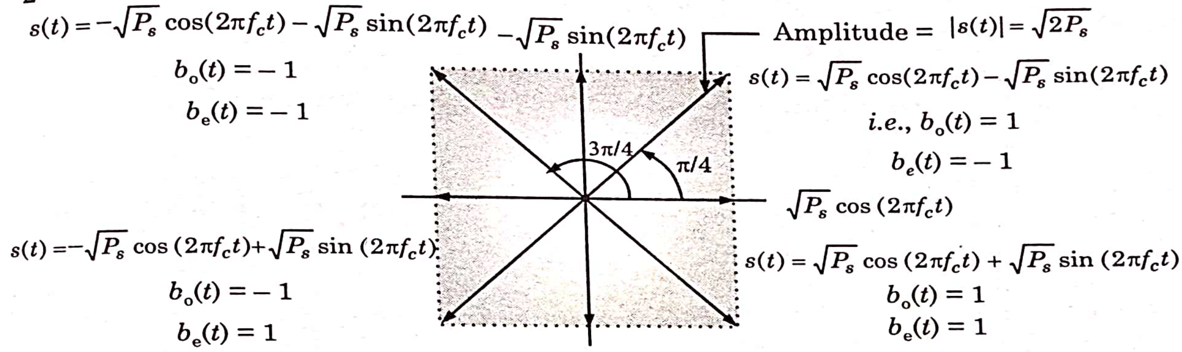


Fig. 14.26 *The Phasor diagram of QPSK signal.*

14.13.1. Reception of QPSK (i.e. Detection of QPSK)

Figure 14.27 shows the QPSK receiver. This is synchronous reception. Hence, the coherent carrier is to be recovered from the received signal $s(t)$. The received signal $s(t)$ is first raised to its 4th power, *i.e.*, $s^4(t)$. After that, it is allowed to pass through a bandpass filter (BPF) which is centred around $4f_c$. The output of the bandpass filter is a coherent carrier of frequency $4f_c$. This is divided by 4 and it provides two coherent quadrature carriers, *i.e.*, $\cos(2\pi f_c t)$ and $\sin(2\pi f_c t)$. These coherent carriers are applied to two synchronous demodulators. These synchronous demodulators consist of multiplier and an integrator.*

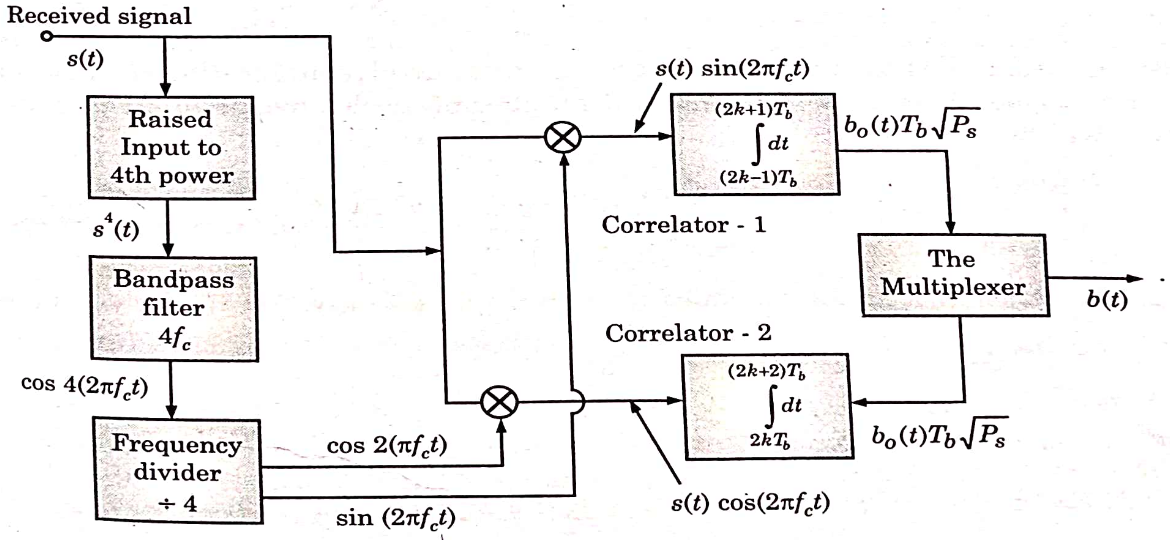


Fig. 14.27 *Reception of QPSK.*

*The

(ii) Because of reduced bandwidth, the information ...

14.14 MINIMUM SHIFT KEYING (MSK)

(MDU, Rohtak, Sem., Examination 2003-2004)

We have discussed QPSK technique in last article. The bandwidth requirement of QPSK is high. Filters or other methods can overcome these problems, but they have other side effects. For example filters alter the amplitude of the waveform.

MSK overcomes these problems. In MSK, the output waveform is continuous in phase hence there are no abrupt changes in amplitude. The sidelobes of MSK are very small hence bandpass filtering is not required to avoid interchannel interference. Figure 14.31 shows the waveform of MSK. The binary bit sequence is shown at the top. Figure 14.31(*a*) shows the corresponding NRZ

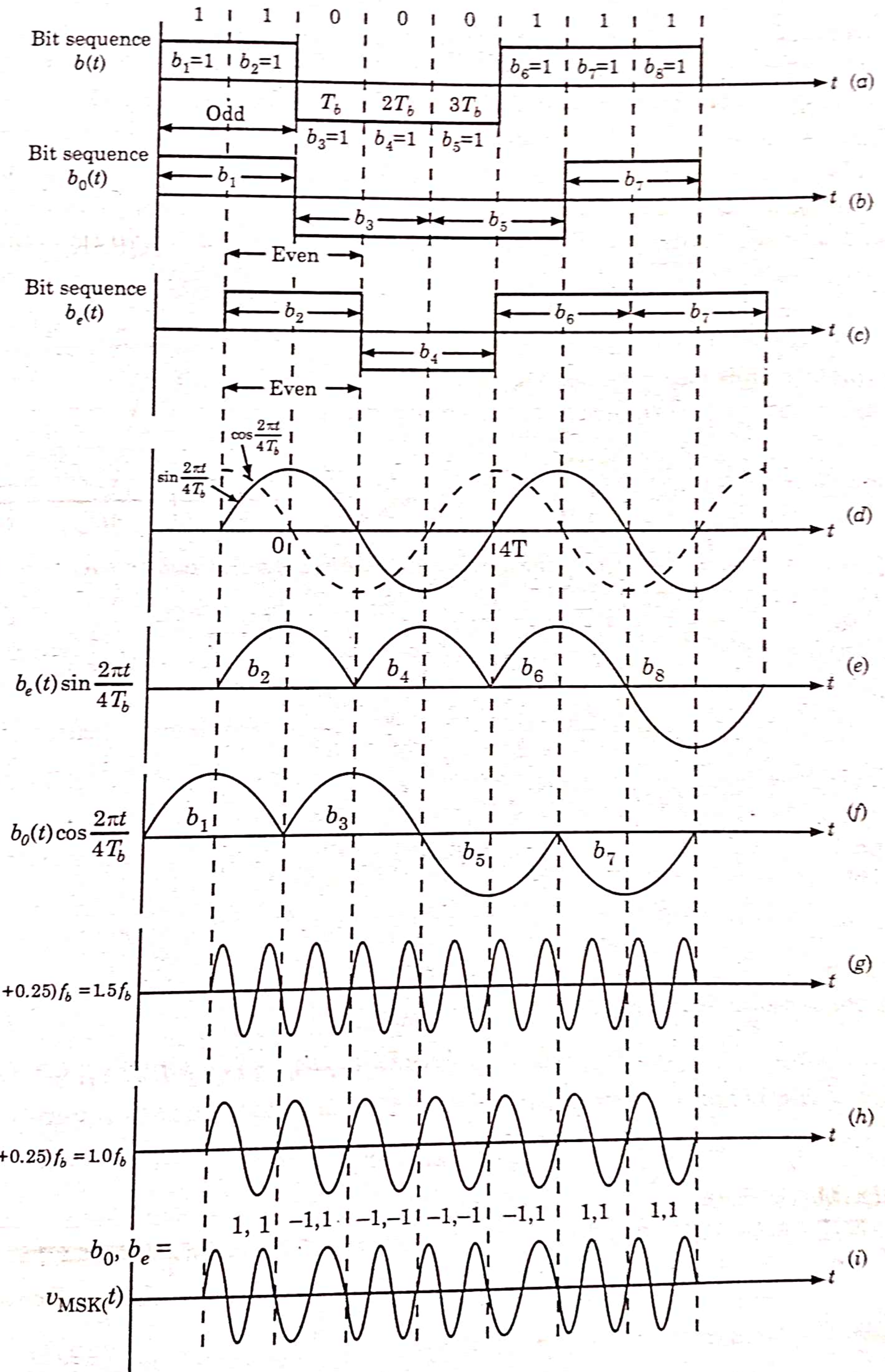


Fig. 14.31 *(a) Bipolar NRZ waveform representing bit sequence (b) Odd bit sequence waveforms $b_0(t)$ (c) Even bit sequence waveform $b_e(t)$ (d) Waveforms of frequency $f_b/4$ used for smoothing of $b_e(t)$ and $b_o(t)$ (e) Modulating waveform of even sequence (f) Modulating waveform of odd sequence (g) Waveform of frequency f_H (h) Waveform of frequency f_L (i) MSK waveform.*

14.14.3. Generation of MSK

Figure 14.34 shows the block diagram of MSK transmitter. The two sinusoidal signals $\sin(2\pi f_c t)$ and $\cos(2\pi t/4T_b)$ are mixed (*i.e.*, multiplied). The bandpass filters then pass only sum and

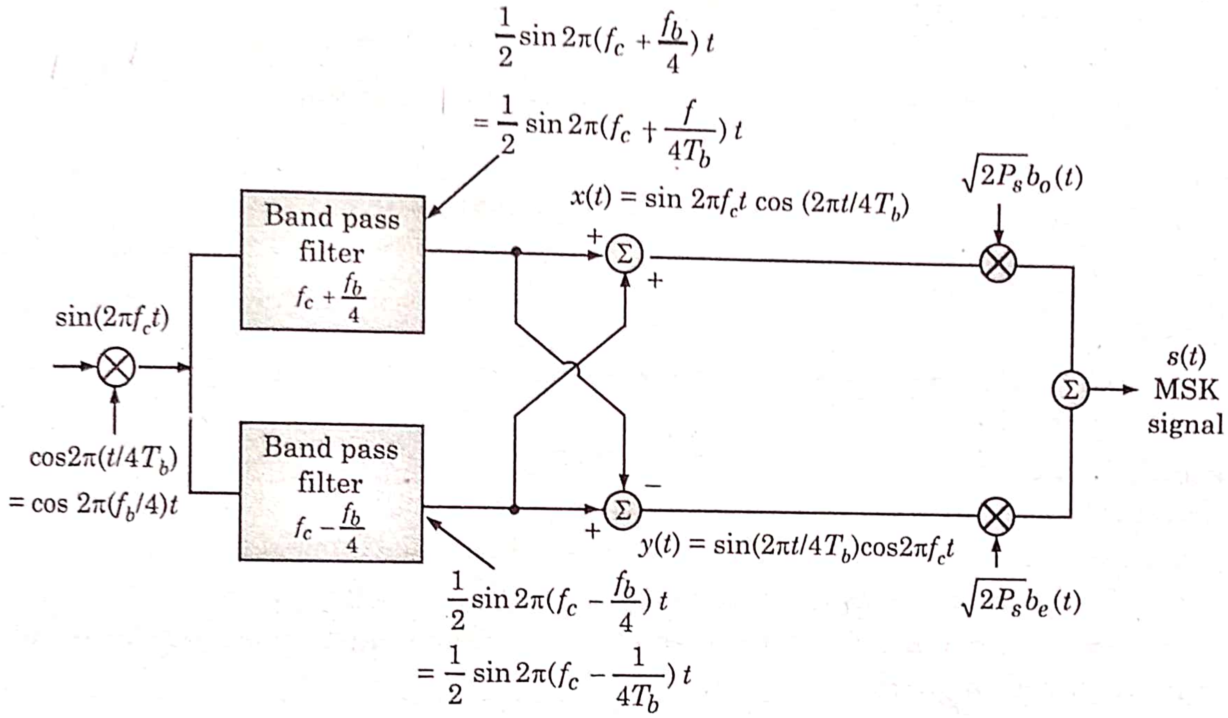


Fig. 14.34 *MSK transmitter block diagram.*

difference components $f_c + \frac{f_b}{4}$ and $f_c - \frac{f_b}{4}$. The outputs of bandpass filters (BPFs) are then added and subtracted such that two signals $x(t)$ and $y(t)$ are generated. Signal $x(t)$ is multiplied by $\sqrt{2P_s}\, b_0(t)$ and $y(t)$ is multiplied by $\sqrt{2P_s}\, b_e(t)$. The outputs of the multipliers are then added to give final MSK signal. Thus the block diagram of figure 14.34 is the step to step implementation of equation (14.73).

14.14.4. Reception of MSK (i.e. Detection of MSK)

Figure 14.35 shows the block diagram of MSK receiver. MSK uses synchronous detection. The signals $x(t)$ and $y(t)$ are multiplied with the received MSK signal. Here $x(t)$ and $y(t)$ have same values as shown in transmitter block diagram of figure 14.35. The outputs of the multipliers are $b_e(t)$ and $b_0(t)$. The integrators integrate over the period of $2T_b$. For the upper correlator, the sampling switch samples output of integrator at $t = (2k+1)T_b$. Then the decision device decides whether $b_0(t)$ is + 1 or – 1. Similarly, lower correlator output is $b_e(t)$. The outputs of two decision devices are staggered by T_b. The switch S_3 operates at $t = kT_b$ and simply multiplexes the two correlator outputs.

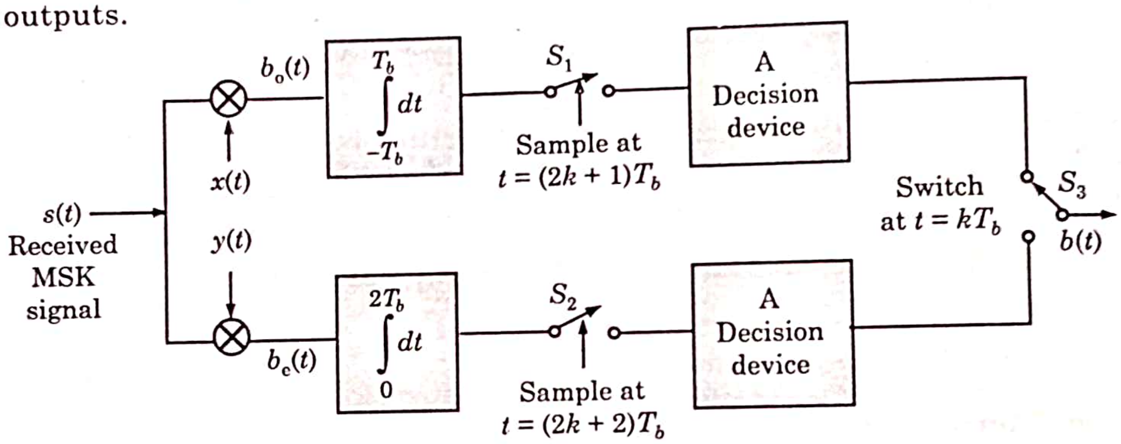


Fig. 14.35 *MSK receiver block diagram.*

14.14.5. Advantages and Drawbacks of MSK as Compared to QPSK

(JNTU, Hyderabad, Sem. Exam. 2005-06)

From the discussion of MSK, we can now compare the advantages of MSK over QPSK.

Advantages:

1. The MSK baseband waveforms are smoother compared to QPSK.
2. MSK signal have continuous phase in all the cases, whereas QPSK has abrupt phase shift of $\frac{\pi}{2}$ or π.
3. MSK waveform does not have amplitude variations, whereas QPSK signals have abrupt amplitude variations.
4. The main lobe of MSK is wider than that of QPSK. Main lobe of MSK contains around 99% of signal energy whereas QPSK main lobe contains around 90% signal energy.
5. Side lobes of MSK are smaller compared to that of QPSK. Hence, interchannel interference because of side lobes is significantly large in QPSK.
6. To avoid interchannel interference due to sidelobes, QPSK needs bandpass filtering, whereas it is not required in MSK.

8. Bandpass filtering changes the amplitude waveform of QPSK because of abrupt changes in phase. This problem does not exist in MSK.

The distance between signal points is same in QPSK as well as in MSK. Hence, the probability of error is also same. However, there are some drawback of MSK.

(ii) Drawbacks

1. The bandwidth requirement of MSK is 1.5 f_b, whereas it is f_b in QPSK. Actually, this cannot be said serious drawback of MSK. Because power to bandwidth ratio of MSK is more. In fact, 99% of signal power can be transmitted within the bandwidth of 1.2 f_b in MSK. While QPSK needs around 8 f_b to transmit the same power.
2. The generation and detection of MSK is slightly complex. Because of incorrect synchronization, phase jitter can be present in MSK. This degrades the performance of MSK.

14.15 GAUSSIAN MINIMUM SHIFT KEYING (i.e., GMSK)

(GGSIPU, Delhi, Sem. Exam., 2006-07)

Like Minimum shift keying (MSK), Gaussian MSK (GMSK) yields a constant amplitude and continuous phase RF carrier signal. It only differs in use of a Gaussian baseband pulse shape in place of square pulse shape for MSK. Because, the Gaussian pulse rises and decays asymptotically with respect to a zero response level, it has a much more constrained bandwidth. A typical GMSK system has been shown in figure 14.36 along with an unfiltered MSK system. The unfiltered MSK is generated by direct FSK modulation of a carrier with a baseband signal which is scaled in amplitude to produce a modulation index of 0.5. This value of modulation index produces a difference of 180° phase shift for the two data values. However, in GMSK, there is ISI (Inter symbol interference) which is a bandwidth limiting factor. GMSK is employed in GSM digital cellular radios and cellular digital packet data (CDPD) applications.

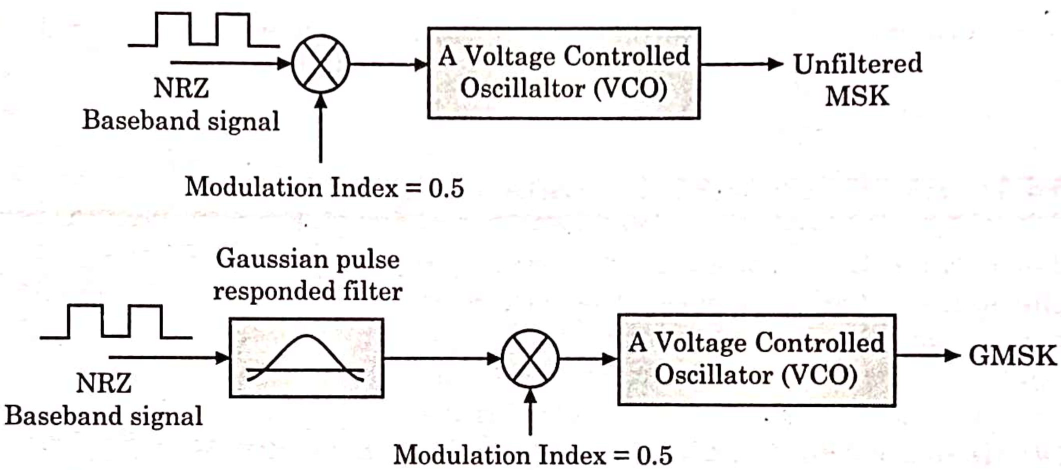


Fig. 14.36 *GMSK as unfiltered MSK.*

Gaussian Minimum Shift Keying (GMSK) is a modification of MSK. A filter used to reduce the bandwidth of a baseband pulse train prior to modulation is called a pre-modulation filter. The Gaussian pre-modulation filter smooths the phase trajectory of the MSK signal and hence limiting the instantaneous frequency variations. The result is an FM modulated signal with a much narrower bandwidth. This bandwidth reduction does not come for free since the pre-modulation filter smears the individual pulses in pulse train. As a consequence of this smearing in time, adjacent pulses interfere with each other generating what is commonly called inter-symbol interference or ISI. In the applications, where GMSK is used, the trade-off between power efficiency and bandwidth efficiency is well worth the cost.

Bit Error Rate (BER) for GMSK is given by

$$P_e = Q\left(\sqrt{\frac{2\alpha E_b}{N_0}}\right) \quad ...(14.10$$

where α is a constant related to BT_b.

Table. 14.8. GMSK Parameter α Related to BT_b.

S.No.	*The value of BT_b*	*The values of α*
1.	0.25	0.68
2.	∞	0.85

It may be noted that the case where $BT_b \to \infty$ corresponds to MSK (*i.e.* the filter is all pass f a fixed symbol interval T_g).

Recall that the probability of error for plain MSK is given by

$$P_e \cong Q\left(\sqrt{\frac{2E_b}{N_0}}\right) \quad ...(14.10$$

Here, we can conclude that $P_e^{GMSK} > P_e^{MSK}$. This arises from the trade off between pow and bandwidth efficiency. GMSK achieves a better bandwidth efficiency than MSK at the exper of power efficiency.

11.9 A PCM-TDM SYSTEM : T1 CARRIER SYSTEM

(Calicut University,Kerala, Sem. Exam., 2006-07)

1. Block Diagram

When a large number of PCM signals are to be transmitted over a common channel, multiplexing of these PCM signals is required. Figure 11.17 shows the basic time division multiplexing scheme, called as the ***T*1-digital system** or **T1 carrier system**. This system is used to convey multiple signals over telephone lines using wideband coaxial cable.

2. Working Operation of the *T*1 Carrier System*

The working operation of the PCM-TDM system shown in figure 11.17 can be explained in the form of few points as under:

(*i*) This system has been designed to accommodate 24 voice channels marked S_1 to S_{24}. Each signal is bandlimited to 3.3 kHz, and the sampling is done at a standard rate of 8 kHz. This is higher than the Nyquist rate. The sampling is done by the commutator switch SW_1.

(*ii*) These voice signals are selected one by one and connected to a PCM transmitter by the commutator switch SW_1.

(*iii*) Each sampled signal is then applied to the PCM transmitter which converts it into a digital signal by the process of A to D conversion and companding, as explained earlier.

(*iv*) The resulting digital waveform is transmitted over a co-axial cable.

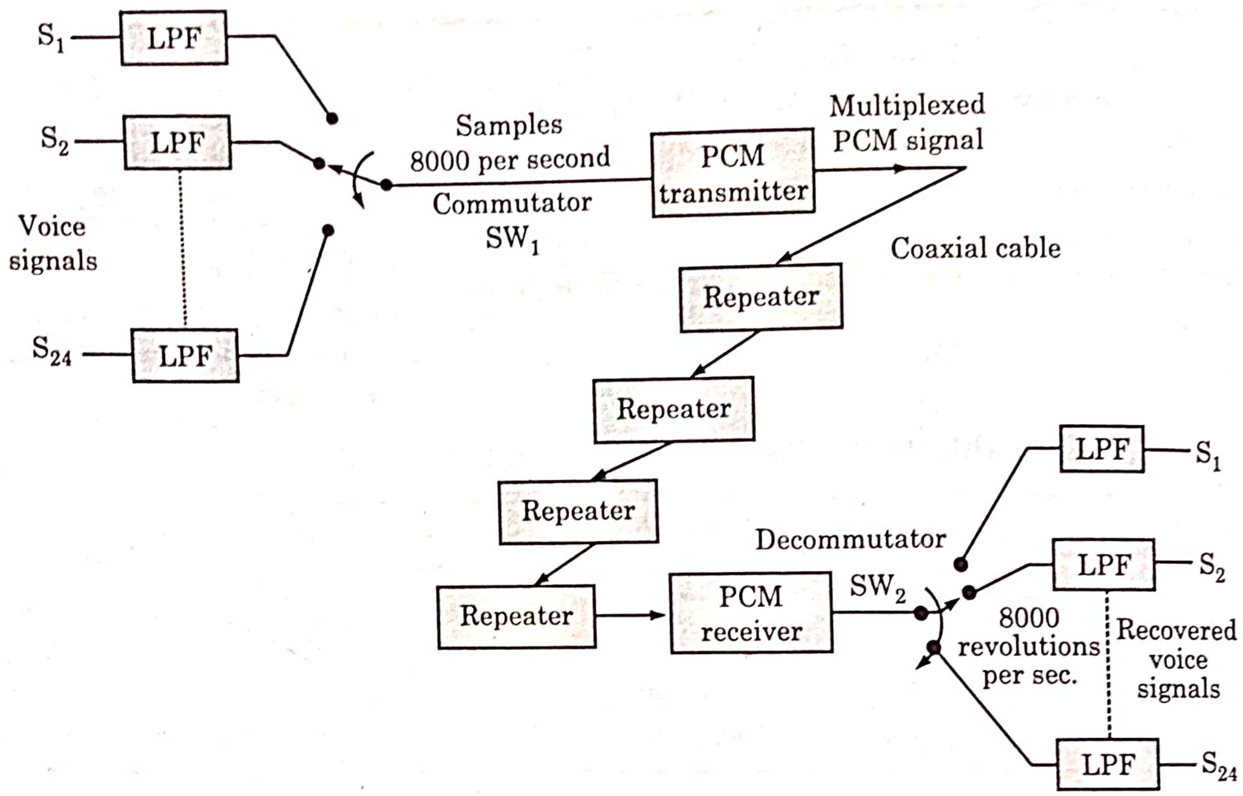


Fig. 11.17 *Block diagram of a basic PCM-TDM system or T1 carrier system.*

(*v*) Periodically, after every 6000 ft, the PCM-TDM signal is regenerated by amplifiers called *Repeaters.* They eliminate the distortion introduced by the channel and remove the superimposed noise and regenerate a clean PCM-TDM signal at their output. This ensures that the received signal is free from the distortions and noise.

(*vi*) At the destination, the signal is companded, decoded and demultiplexed, using a PCM receiver. The PCM receiver output is connected to different low pass filters via the decommutator switch SW_2.

(*vii*) Synchronization between the transmitter and receiver commutators SW_1 and SW_2 is essential in order to ensure proper communication.

6.6 Spread Spectrum

Spread Spectrum refers to a system originally developed for military applications, to provide secure communications by spreading the signal over a large frequency band.
Figure 1 represents a narrow band signal in the frequency domain. These narrowband signals are easily jammed by any other signal in the same band. Likewise, the signal can also be intercepted since the frequency band is fixed and narrow (i.e. easy to detect).

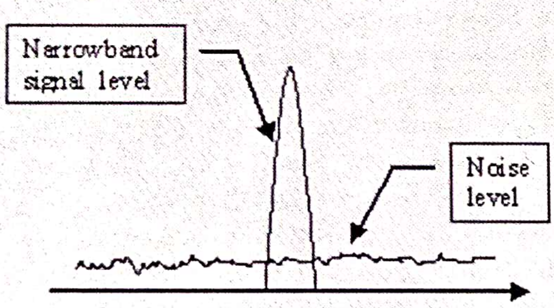


Figure 1: Narrow band signal, relatively easy to jam or intercepted.

The idea behind spread spectrum is to use more bandwidth than the original message while maintaining the same signal power. A spread spectrum signal does not have a clearly distinguishable peak in the spectrum. This makes the signal more difficult to distinguish from noise and therefore more difficult to jam or intercept. This concept is illustrated in Figure 3.

This document will explore basics concepts of spread spectrum for the remaining of the introduction and then it will explore the supporting concepts of the most used technique in spread spectrum systems. The last section will give the reader some insight of more advance topics but will not deeply explore them. We encourage the reader to seek the references for advance knowledge of spread spectrum systems.

General Block Diagram

We present now the block diagram of a typical communication system with the difference that the modulator/demodulator has as input the spreading generator. This piece will be explored in following sections.

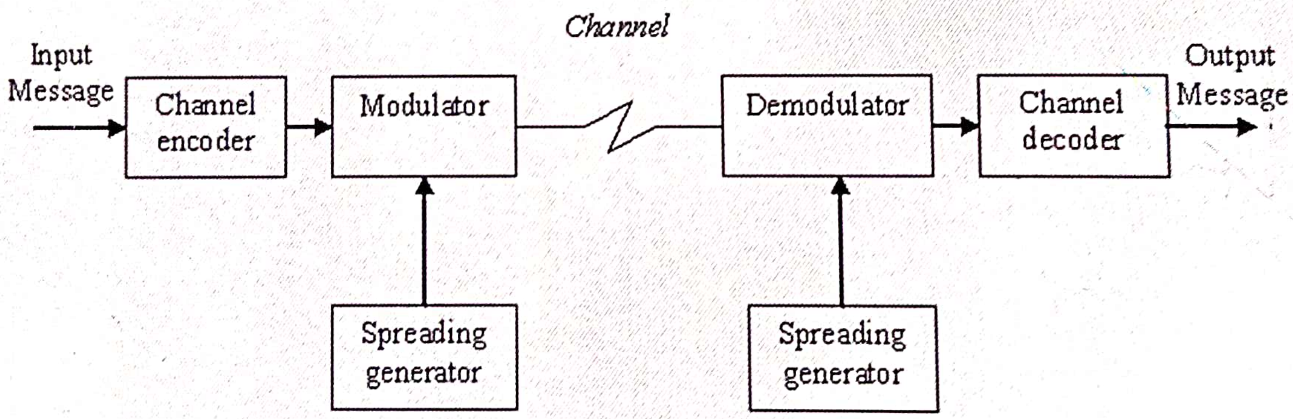


Figure 2: Block diagram of the spread spectrum communication system

There are two predominant techniques to spread the spectrum:

1) Frequency hoping (FH), which makes the narrow band signal jump in random narrow bands within a large bandwidth.

2) Direct sequence (DS) which introduces rapid phase transition to the data to make it larger in bandwidth.

We will focus on Direct Sequence Spread Spectrum technique since it is the mostly used in the industry (CDMA, UMTS, 802.11, GPS).

What are applications of spread spectrum?

Current applications of spread spectrum technology include **wireless LANs (local area networks), bar code scanners, and microphones**. This technology improves the efficiency and effectiveness of business processes, many of which are finding that wireless communications are requisite for success.

6.8 Define bit, Baud, Symbol & Channel Capacity formula (Shannon Theorems) :—

Bit Rate ⇒

→ Information transfer rate ⟶ bit rate.

→ Digital Signal represent by '0' or '1'.

∴ Bit rate is the number of bits that are sent per unit of time from source to destination.

Example : If 12 bits of information are sent every second, then

→ Information transfer rate = 12 bits/1s = 12 bps (in 1 sec, 12 bits can tx)

→ Unit of information transfer rate : bits/second or bps

Symbol Rate ⇒

Symbol rate (Baud rate) is the number of symbol changes, waveform changes, or signalling events, across the transmission medium per time unit using a digitally modulated signal or a line code.

Example :

If 1 bit = 1 symbol, then the symbol rate would be the same as the bit rate.

If 2 bit = " , " half of the bit rate.

Symbol rate = bit rate / the number of bits to represent a symbol.

Unit of symbol rate : Symbol/second or sps

Bit Rate Vs Symbol Rate

1 0 1 0 0 1 1 0 1 1 0 1

Bit rate = 12 bps

for Example, 1 symbol = 4 bits

1 0 1 0 | 0 1 1 0 | 1 1 0 1

Symbol Rate = 3 sps

for this case, 12 bps = 3 sps

Channel Capacity formula (Shannon Theorems) :-

$$C = B \log_2 \left[1 + \frac{S}{N}\right]$$

Where,

B = Bandwidth of channel

S = Signal Power

N = Noise Power

Proof :- Received signal = Signal Power (S) + Noise Power (N) and it's mean square value is $\sqrt{S+N}$.

→ Noise Power is N and it's mean square value is $\sqrt{N}$.

→ So Number of levels can be separated without error is

$$m = \frac{\sqrt{N+S}}{\sqrt{N}} = \sqrt{1 + \frac{S}{N}}$$

→ So digital Information is

$$I = \log_2 m$$

$$= \log_2 \sqrt{1 + S/N}$$

$$= \frac{1}{2} \log_2 \left(1 + \frac{S}{N}\right)$$

→ IF channel transmitter K pulses per second then channel capacity is

$$C = IK = \frac{K}{2} \log_2 \left(1 + \frac{S}{N}\right)$$

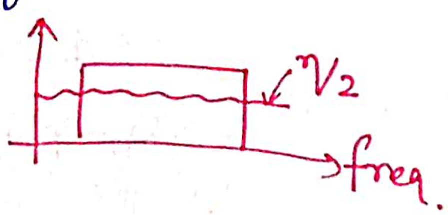
→ Nyquist Bandwidth is $K = 2B$

$$\boxed{C = B \log_2 \left(1 + \frac{S}{N}\right)}$$

→ Practically If we increase B then noise N will also increase. So capacity of channel can not be infinite.

If $\eta/2$ is power density then

$$N = \eta B$$



So channel Capacity is

$$C = B \log_2\left(1 + \frac{S}{\eta B}\right)$$

$$= \frac{\eta B}{S}\left(\frac{S}{\eta}\right)\log_2\left(1 + \frac{S}{\eta B}\right)$$

$$= \frac{S}{\eta}\left[\frac{\log_2\left(1 + \frac{S}{\eta B}\right)}{\left(\frac{S}{\eta B}\right)}\right]$$

→ for $\lim_{x \to 0} \frac{\log_2(1+x)}{(1/x)} = \log_2 e = 1.44$

$$C = \frac{S}{\eta}\log_2 e = \boxed{1.44\,\frac{S}{\eta}}$$

What is modem and its applications?

A modem **transmits data by modulating one or more carrier wave signals to encode digital information, while the receiver demodulates the signal to recreate the original digital information**. The goal is to produce a signal that can be transmitted easily and decoded reliably.

Types of Modem

- Cable Modems. Cable modems help in establishing communication between computer and ISP over landline connection. ...
- Telephone Modems. These modems are network devices that allow data communication between two computers over voice-grade telephone lines. ...
- Dial modems. ...
- Satellite Modems. ...
- Digital Subscriber Line (DSL)

31-Oct-2022