Government polytechnic kendrapara

DEPARTMENT

OF

ELECTRONICS AND TELECOMMUNICATION ENGINEERING

LECTURE NOTES

Semester : 5th

Subject: ANALOG & DIGITAL COMMUNICATION (TH-3)

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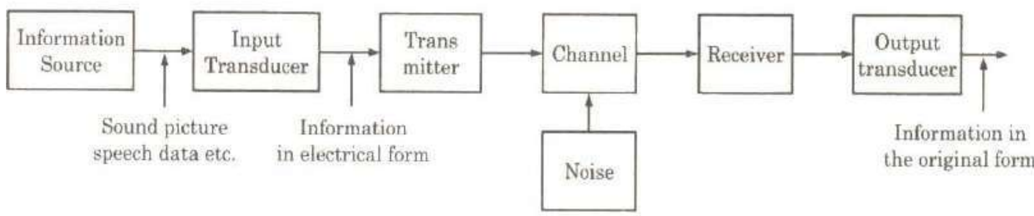
UNIT-1: ELEMENTS OF COMMUNICATION SYSTEMS

1.1 ELEMENTS OF COMMUNICATION SYSTEM-

Communication involves the transmission of information from one point to another.



Block diagram of a communication system

INFORMATION SOURCE-

Communication system serves to communicate a message or information. This message originates in the information source. There can be various messages in the form of words, groups of words, code, symbols, sound signals etc.

INPUT TRANSDUCER-

A transducer is a device which converts one form of energy into another form. The message from the information source may or may not be electrical in nature. 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.

TRANSMITTER-

The function of the transmitter is to process the electrical signal from different aspects. Inside the transmitter, signal processing such as restriction of range of audio freqiuencies, amplification and modulation are achieved.

THE CHANNEL AND THE NOISE-

There are two types of channels, namely point to point channels and broadcast channels. Examples of point to point channels are wire lines, microwave links and optical fibers. Wire lines operate by guided electromagnetic waves and they are used for local telephone transmission. Microwave links are used in long distance telephone transmission. Optical fibers are used in optical communication. On the other hand the broadcast channels provide a capability where several receiving stations can be reached simultaneously from a single transmitter. During the process of transmission and reception the signal gets distorted due to noise introduced in the system. Noise is an unwanted signal which tends to interfere with the required signal. Noise may interfere with signal at any point in a communication system.

RECEIVER-

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

DESTINATION-

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

1.2 SOURCE OF INFORMATION & COMMUNICATION CHANNELS

SOURCE OF INFORMATION-

Some of the important source of information in the communication environment given below-

(i) Speech (iii) Picture

(ii) Music (iv) Computer data

A source of information is basically a signal which carries the information.

Signal-

A signal may be defined as the single valued function of time. Time plays the role of an independent variable. This means that at every instant of time, the signal has a unique value.

The signals may be classified as:

(i) Speech

Speech involves transfer of information from the speaker to the listener. Such a transfer of information takes place in following three stages:

(1) Production

(2) Propagation and

(3) Perception

(ii) **Music signal**-

It is originated from the instruments such as the piano, violin, flute etc. Music signal has following two possible structure:

(1) Melodic structure

(2) Harmonic structure

(iii) Picture

The picture can be either static or dynamic. Examples of static picture is the picture sent by fax machine and that of a dynamic picture is the one produced on T.V.

(iv) Computer Data

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

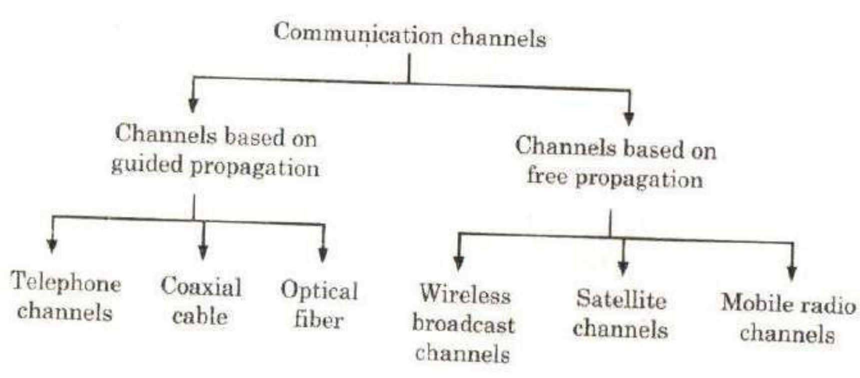
COMMUNICATION CHANNEL-

The medium over which the information is passed from the transmitter to the receiver is called as a communication channel. Depending on the mode of transmission, the communication channels classified in to two categories.

(i) Guided medium

(ii) Unguided medium

The classification of channels has been shown below:

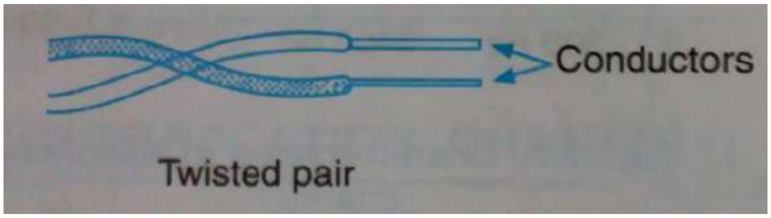


Classification of communication channels

Guided Medium-

Telephone Channels-

It is designed for providing service to voice signals such as telephones. The telephone channels are also used for the worldwide internet connection. Therefore, telephone channel is the best possible option for the data transmission over long distances.

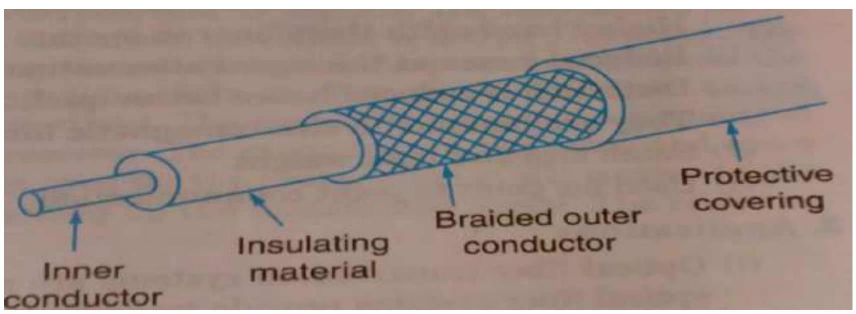


The telephone channels are built using the twisted pair of wires. A twisted pair consists of two insulated conductor twisted together in the spiral form. It can be shielded or un-shielded.

The un-shielded twisted pair cables are very cheap and easy to install. However, they are badly affected by the noise interference.

The noise immunity can be improved by using shielded twisted pair cable.

Co-axial cable-

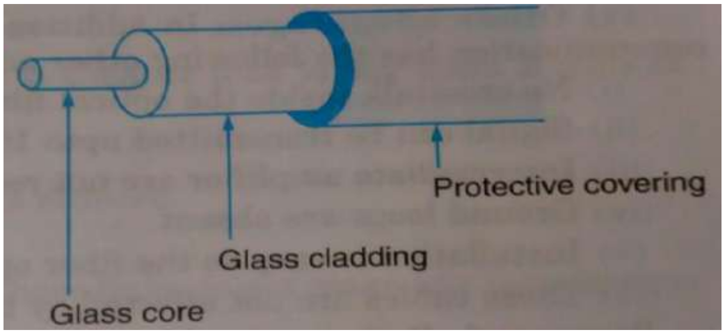


It consists of two concentric conductors separated by a dielectric material. The external conductor is metallic braid and used for the purpose of shielding. The co-axial cable may contain one or more co-axial pairs.

One more application is Ethernet LAN using the co-axial cable.

The co-axial cable is used for its large bandwidth and high noise immunity.

Optical fiber cables-



It consists of an inner glass core surrounded by a glass cladding which has a lower refractive index.

Digital signals are transmitted in the form of intensity-modulated light signal which is trapped in the glass core.

Light is launched into the fiber using a light source such as a LED or laser.

It is detected on the other side using a photo detector such as a phototransistor.

Unguided Medium-

Wireless Broadcast channel-

These channels are used for the transmission of radio and TV signals.

The information signal which represents the speech, music etc. modulates a carrier frequency.

The carrier frequency is different for every transmitting station.

A transmitting antenna radiates the modulated signal in the form of electromagnetic radiation into the free space.

These waves are radiated in all directions or in some specified directions.

The transmitting antenna is mounted on a tower or a hall in order to reach the farther receiver.

The ground wave, sky wave and space wave are three types of propagation techniques used for the propagation of EM waves.

At the receiving end, the receiving antenna is used for picking up the transmitted signal.

Satellite Channels-

Satellite microwave systems transmits signals between directional parabolic antennas.

They use low gigahertz frequencies and line of sight communication.

These systems use satellites which are in the geostationary orbit (36000 km above the earth).

The satellites act as repeaters with receiving antennas, transponder and transmitting antenna.

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

In this method 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.

Mobile Radio Channels-

In mobile communication, the sender and the receiver both are allowed to move with respect to each other.

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

The signals taking different paths will have to travel different path lengths. So, they have different 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.

1.3 CLASSIFICATION OF COMMUNICATION SYSTEM-

- Depending upon the message signal, communication system may be classified as-

i) Analog communication system

ii) Digital communication system

Analog communication system-

It is a 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 is an analog signal.

Digital communication system-

It is a type of communication in which the message signal to be transmitted is digital in nature. This means that digital communication involves the transmission of information in digital form.

- Based on communication channel used, communication system may be classified as-

  i) Line communication system

  ii) Radio communication system

Line communication system-

There is a physical link present between the transmitter and receiver in line communication system.

Ex- Landline telephony is purely a line communication system

Radio communication system-

In this system there is no link present between transmitter and receiver.

Ex.- Radio broadcast

- Based on transmission mode, communication system may be classified as-

  i) Simplex communication system

  ii) Duplex communication system

Simplex communication system-

In this communication system, one way transmission is used. Ex.- TV transmission

Duplex communication system-

In this communication system, two way transmission is used. Ex.- Telephony system.

1.4 MODULATION PROCESS, NEED OF MODULATION AND CLASSIFY MODULATION PROCESS

MODULATION:

Modulation is the process in which parameters (i.e, Amplitude, Phase and Frequency) of the carrier signal changes according to the message signal or modulating signal.

NEED OF MODULATION:

1) Practicality of Antenna-

The voice frequencies in the band of 20HZ to 20KHZ, For the efficient transmission and reception of radio frequency signals. The antenna length 'L' required in terms of wavelength

$$L = \frac{\lambda}{4}$$

$$= \frac{C}{4f} \qquad (C = 3*10^8 \text{ m/sec})$$

Where λ = wavelength
C = Speed of light
f = Frequency

2) To remove Interference
3) Reduction of noise
4) Multiplexing-

- Simultaneously transmission of multiple message over a single channel is known as multiplexing.
- If it transmits without modulation, the different message signal over a single channel will interfere with one another.
- Multiplexing helps in transmitting numbers of message signal simultaneously over a single channel & therefore a number of channel needed will be less.

CLASSIFY MODULATION PROCESS-

Continuous wave modulation-

When the carrier wave is continuous in nature, the modulation process is known as continuous wave modulation or analog modulation.

Pulse modulation-

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.

ANALOG AND DIGITAL SIGNALS-

- The analog signal is that type of signal which varies smoothly and continuously with time.

- This means that analog signals are defined for every value of time and they take on continuous values in a given time interval.

- An alternative form of signal representation is that of a sequence of numbers, each number representing the signal magnitude at an instant of time. The resulting signal is called a digital signal.

1.5 CONVERSION OF ANALOG SIGNALS TO DIGITAL SIGNALS-

There are three steps for conversion process.

1. The signal is sampled.
2. The sampled signal is quantized.
3. The quantized signal is digitally coded.

Sampling

Sampling generally is done with a Sample-And-Hold circuit. To be able to reconstruct the signal we must consider the Sampling Theorem which says that a sampling frequency twice of the highest frequency. In a simple way sampling can be defined as the process of taking samples from the continuous time function x(t) and for the signal to reconstruct we must consider the sampling theorem which states that the sampling frequency must be always greater than or equal to twice the highest frequency.

Quantization

Quantization is the process of taking a continuous voltage signal and mapping it to a discrete number of voltage levels. The number of voltage levels affects the quantization noise that occurs. Since digital computers are binary in nature, the number of quantization levels is usually a power of 2, i.e.,

$N=2^n$

where n is the number of quantization bits.

Encoding

Encoding is the process of converting the quantized signals into a digital representation. This encoding is performed by giving each quantization level a unique label. For instance, if four bits are used, the lowest level may be (in binary) 0000, and the next highest level 0001, etc.

1.6 BASIC CONCEPT OF SIGNALS & SIGNALS CLASSIFICATION

BASIC CONCEPT OF SIGNALS-

A signal is a physical quantity which varies with respect to time, space and contains some information from source to destination. The term signal includes audio, video, speech, image etc.

The signals are functions of one or more variables and the systems respond to an input signal by producing an output signal.

CLASSIFICATION OF SIGNAL-

Signals are classified into the following categories:

1) Continuous Time and Discrete Time Signals
2) Deterministic and Non-deterministic Signals
3) Even and Odd Signals
4) Periodic and Aperiodic Signals
5) Energy and Power Signals
6) Real and Imaginary Signals

Continuous Time and Discrete Time Signals-

Continuous signal-

A signal is said to be continuous when it is defined for all instants of time.

Discrete signal-

A signal is said to be discrete when it is defined at only discrete instants of time.

Deterministic and Non-deterministic Signals-

Deterministic signal-

A signal is said to be deterministic if there is no uncertainty with respect to its value at any instant of time. Or, signals which can be defined exactly by a mathematical formula are known as deterministic signals.

Non-Deterministic signal-

A signal is said to be non-deterministic if there is uncertainty with respect to its value at some instant of time. Non-deterministic signals are random in nature hence they are called random signals. Random signals cannot be described by a mathematical equation.

Even and Odd Signals-

Even signals-

A signal is said to be even when it satisfies the condition $x(t) = x(-t)$.

Odd signals-

A signal is said to be odd when it satisfies the condition $x(t) = -x(-t)$.

Periodic and Aperiodic Signals-

Periodic signals-

A signal is said to be periodic if it satisfies the condition $x(t) = x(t + T)$ or $x(n) = x(n + N)$.

Where

T = fundamental time period,

1/T = f = fundamental frequency.

Aperiodic signals-

The above signal will repeat for every time interval $T0$ hence it is periodic with period $T0$.

Energy and Power Signals-

Energy signals-

A signal is said to be energy signal when it has finite energy.
Power signals-
A signal is said to be power signal when it has finite power.

Real and complex signals-

Real signals-
A signal x(t) is a real signal if its value is a real number.
Complex signals-
A signal x(t) is a complex signal if its value is a complex number.

1.7 BANDWIDTH LIMITATION-

While designing a communication system, an engineer generally faces several limitations. These are:

1) Noise limitation
2) Bandwidth limitation
3) Equipment limitation

Noise Limitation-

The noise may be defined as an unwanted form of energy which tend to interfere with the transmission and reception of the desired signals in a communication system. The noise cannot be eliminated completely. However the effect of noise on desired signals can be minimized.

The noise limits our ability to identify the intended or desired message correctly.

Bandwidth Limitation-

The bandwidth limitation is another major limitation in a communication system. The frequency range needed for a particular transmission is known as bandwidth.

This band of frequencies or bandwidth for a particular transmission is also called channel and it is always allocated by some international regulatory agencies. This type of regulation is essential to avoid interference among the signals having same frequency.

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

Equipment Limitation-

The noise and bandwidth limitation dictate theoretical limit may not be realized in a practical system due to equipment limitations.

UNIT-2: AMPLITUDE (LINEAR) MODULATION SYSTEM

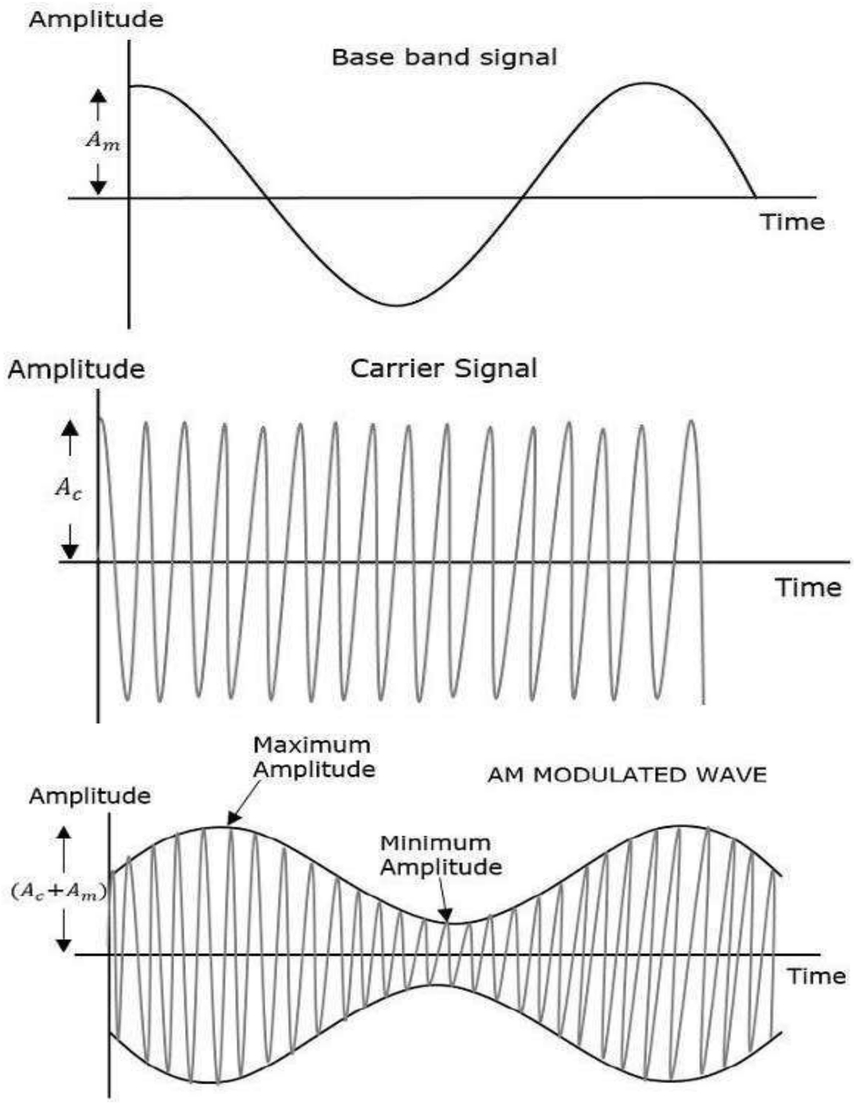
2.1 AMPLITUDE MODULATION & DERIVE THE EXPRESSION FOR AMPLITUDE MODULATION SIGNAL, POWER RELATION IN AM WAVE & FIND MODULATION INDEX

AMPLITUDE MODULATION AND DERIVE THE EXPRESSION FOR AMPLITUDE MODULATED SIGNAL:-

Amplitude modulation may be defined as a system in which the maximum amplitude of the carrier wave is proportional to the instantaneous value of the modulating signal.

A continuous-wave goes on continuously without any intervals and it is the baseband message signal, which contains the information. This wave has to be modulated.

According to the standard definition, "The amplitude of the carrier signal varies in accordance with the instantaneous amplitude of the modulating signal." Which means, the amplitude of the carrier signal containing no information varies as per the amplitude of the signal containing information, at each instant. This can be well explained by the following figures.



The first figure shows the modulating wave, which is the message signal. The next one is the carrier wave, which is a high frequency signal and contains no information. While, the last one is the resultant modulated wave.

It can be observed that the positive and negative peaks of the carrier wave, are interconnected with an imaginary line. This line helps recreating the exact shape of the modulating signal. This imaginary line on the carrier wave is called as **Envelope**. It is the same as that of the message signal.

Mathematical Expressions

Modulating signal- x(t)= $A_m Cos w_m t$

Carrier Signal, C(t)= $A_c Cos w_c t$

Where, A_m *and* A_c are the amplitude of the modulating signal and the carrier signal respectively.

w_m *and* w_c are the frequency of the modulating signal and the carrier signal.

Modulated Signal-

m(t)= $[A_c + x(t)]Cosw_c t$

= $(A_c + A_m Cosw_m t)Cosw_c t$

= $A_c\left(1 + \frac{A_m}{A_c} Cos\, w_m t\right) Cosw_c t$

= $A_c(1 + m_a Cosw_m t)Cosw_c t$ $\quad (\frac{A_m}{A_c} = m_a)$

= $A_c Cosw_c t + A_c m_a Cosw_m t\, Cosw_c t$

= $A_c Cosw_c t + \frac{A_c m_a}{2}[Cos\,(w_c - w_m)t + Cos\,(w_c + w_m)t]$

=$A_c Cosw_c t + \frac{A_c m_a}{2} Cos\,(w_c - w_m)t + \frac{A_c m_a}{2} Cos\,(w_c + w_m)t$

Bandwidth-

$$f_{max} = w_c + w_m$$
$$f_{min} = w_c - w_m$$

Bandwidth= $f_{max} - f_{min}$

= $(w_c + w_m) - (w_c - w_m)$

=$w_c + w_m - w_c + w_m$

= $2w_m$

MODULATION INDEX

Modulation Index-

It is the ratio of amplitude of modulating signal to the amplitude of the carrier signal. It is denoted as m_a.

$$m_a = \frac{A_m}{A_c} = \frac{Maximum\ Amplitude\ of\ modulating\ signal}{Amplitude\ of\ the\ carrier\ signal}$$

Modulation index means how much energy of the carrier wave used during modulation.

Maximum amplitude= $A_{max} = A_c + A_m$

Minimum amplitude= $A_{min} = A_c - A_m$

When, $A_{max} + A_{min}$

=$A_c + A_m + A_c - A_m$

$= 2A_c$

$=> A_{max} + A_{min} = 2A_c$

$=> A_c = \frac{A_{max} + A_{min}}{2}$

When, $A_{max} - A_{min}$

$= A_c + A_m - A_c + A_m$

$= 2A_m$

$=> A_{max} - A_{min} = 2A_m$

$=> A_m = \frac{A_{max} - A_{min}}{2}$

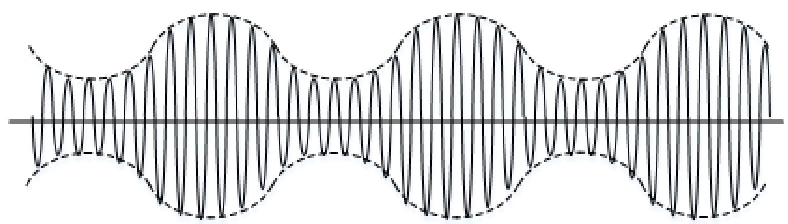
$$m_a = \frac{A_m}{A_c} = \frac{\frac{A_{max} - A_{min}}{2}}{\frac{A_{max} + A_{min}}{2}} = \frac{A_{max} - A_{min}}{A_{max} + A_{min}}$$

There are three cases of m_a.

Case-I

When $A_m < A_c, m_a < 1$. This condition is known as under modulation.

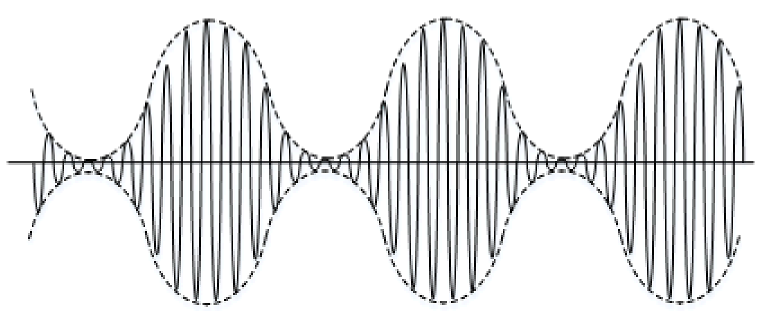
Under modulation (m < 1)



Case-II

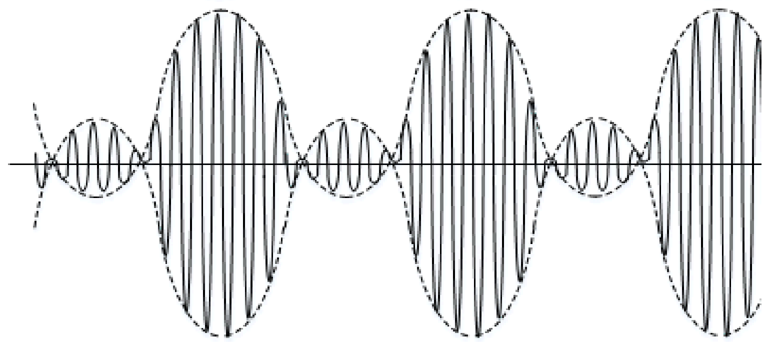
When $A_m = A_c, m_a = 1$. This condition is known as 100% modulation.

100 % modulation (m =1)



Case-III

When $A_m > A_c, m_a > 1$. This condition is known as over modulation.



$Questions -$

1. Carrier wave of frequency 1 MHz with peak voltage of 20V used to modulate a signal of frequency 1 KHz with peak voltage of 10V.
   Find out the following-
   i) μ
   ii) Frequencies of modulated signal
   iii) Bandwidth
2. A modulating signal is x(t)= 10 cos (2π x $10^3 t$) and carrier signal is c(t)= 50 cos (2π x $10^5 t$). Find out the percentage of modulation.
3. What is the modulation index value if $V_{max} = 5.9\ V\ and\ V_{min} = 1.2V$.

POWER RELATION IN AM WAVE:-

$$P_t(total\ power) = P_c + P_{USB} + P_{LSB}$$

$$= P_c + P_s$$

$$P = \frac{V^2}{R}$$

$$= \frac{(V_{rms})^2}{R}$$

$$= \frac{(V_m/\sqrt{2})^2}{R}$$

$$P_c = \frac{(A_c/\sqrt{2})^2}{R}$$

$$= \frac{{A_c}^2}{2R}$$

$$P_{USB} = \frac{(V_m/\sqrt{2})^2}{R}$$

$$=\frac{(\frac{m_a A_c}{2\sqrt{2}})^2}{R}$$

$$P_{USB} = \frac{m_a^2 A_c^2}{8R} = P_{LSB}$$

$$P_S = P_{USB} + P_{LSB}$$

$$=\frac{m_a^2 A_c^2}{8R} + \frac{m_a^2 A_c^2}{8R}$$

$$=2\frac{m_a^2 A_c^2}{8R} = \frac{m_a^2 A_c^2}{4R}$$

$$P_t = P_c + P_s$$

$$= \frac{{A_c}^2}{2R} + \frac{m_a^2 A_c^2}{4R}$$

$$= \frac{{A_c}^2}{2R}[1 + \frac{m_a^2}{2}]$$

$$P_t = P_c[1 + \frac{m_a^2}{2}]$$

Current Relation-

P= $I^2 t$

$$P_t = P_c[1 + \frac{m_a^2}{2}]$$

$$=> I_t^2 R = I_c^2 R[1 + \frac{m_a^2}{2}]$$

$$=> I_t^2 = I_c^2[1 + \frac{m_a^2}{2}]$$

$$=> I_t = \sqrt{I_c^2[1 + \frac{m_a^2}{2}]}$$

Questions-

1) A 800watt carrier is modulated to a depth of 50%. Find the total power in the AM wave.
2) An AM broadcast radio transmitter radiates 10Kwatt of power of modulation percentage is 60%. Calculate how much of this is carrier power.

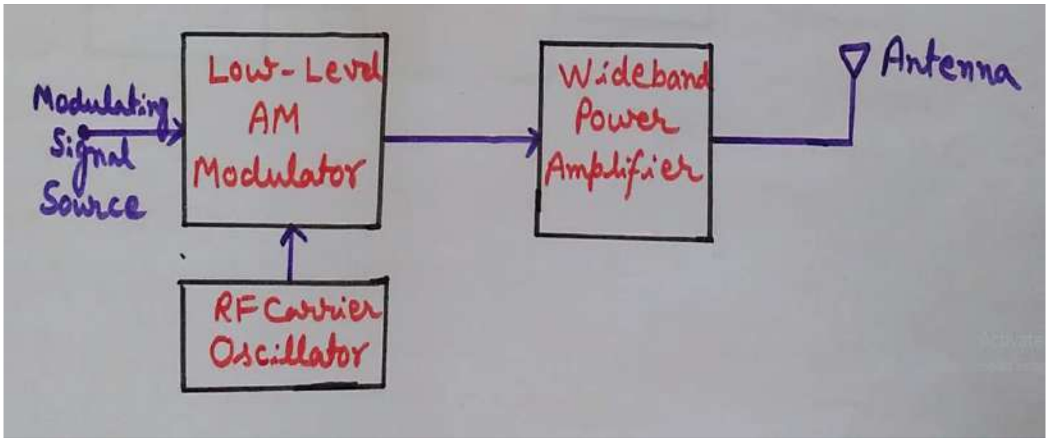
2.2 GENERATION OF AM WAVES-

The device which is used to generate an amplitude modulation (AM) wave is known as amplitude modulator. The methods as amplitude modulator Generation may be broadly classified as following:-

1) Low level AM Modulation.
2) High level AM Modulation.

1) Low Level Amplitude Modulation:-

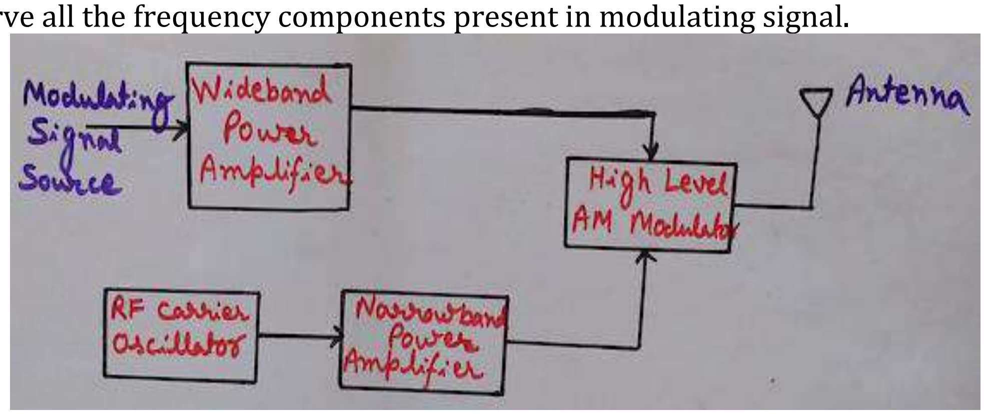
In a low level amplitude modulation system, the modulation is done at low power level. At low power levels, a very small power is associated with the carrier signal and the modulation signal. Because of this the output power of modulation is low. Therefore the power amplifiers are required to boost the amplitude modulated signals up to the desired output level.



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.
Square-law diode modulation and switching modulation are examples of low-level modulation.

2) High level Amplitude 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 base band signal and the carrier signal must be at high power levels. In block diagram of figure 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. The collector modulation method is the example of high level modulation.

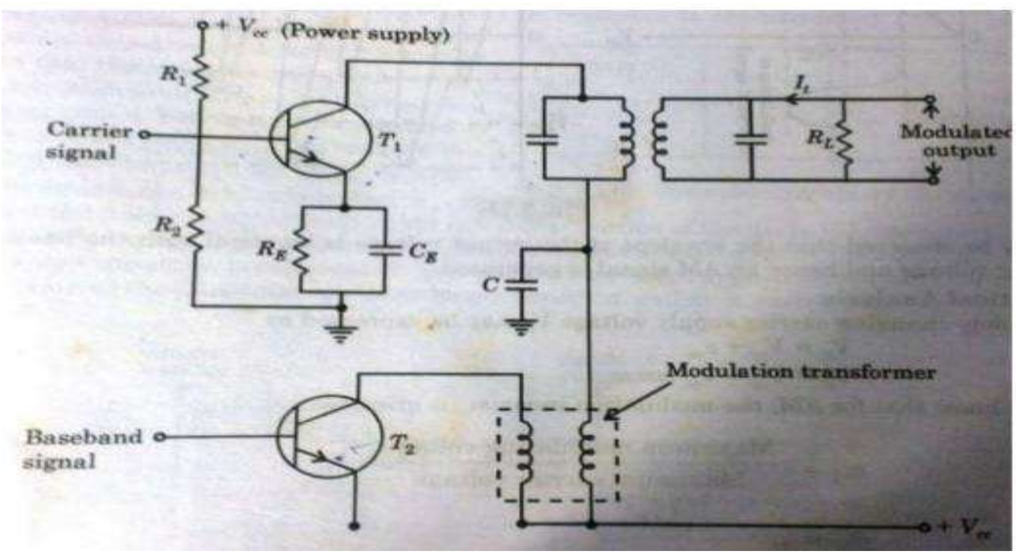
COLLECTOR MODULATION (LINEAR LEVEL AM MODULATION)-

Collector modulator is a linear modulator.

The circuit consists of two transistors T1 and T2. The transistor T1 makes a radio frequency class-C amplifier. At the base of the T1 carrier signal is applied.

The transistor T2 makes a class B amplifier, which is used to amplify the modulating signal appears across the modulation transformer. For biasing purpose voltage divider circuit is used.

A capacitor is used to isolate the modulation transformer from the high frequency carrier signal. Here double tuned circuit is used for better performance. The resonance frequency of tank circuit is equal to the carrier frequency.



Operation-

As we know class C amplifier gives 80% efficiency but more distortion. But here a high frequency carrier signal is used. So, distortion is less.

A linear relationship exists between the output tank current (I_t) and the variable supply voltage V_c.

During absence of modulating signal, the output voltage will be an exact replica of the input voltage waveform.

So, if R_L is the resistance of the output tank circuit at resonance, then the magnitude of the magnitude of the output voltage is

$$R_L I_t \cong V_{cc}$$

But, if a modulating signal voltage appears across the modulating transformer, this signal will be added to the carrier supply voltageV_{cc}.

So, $V_c = V_{cc} + V_m$

Where $V_m = V_m cos\omega_m t$

V_{cc}= amplitude of the carrier signal

Carrier signal represented as

$V_c = V_{cc} cos\omega_c t$

$The\ modulated\ signal\ is$

$V_o = (V_{cc} + V_m cos\omega_m t) cos\omega_c t$

$=> V_o = V_{cc}\left(1 + \frac{V_m}{V_{CC}} cos\omega_m t\right) cos\omega_c t$

$=> V_o = V_{cc}(1 + m_a cos\omega_m t) cos\omega_c t$

DEMODULATION OF AM WAVE:-

The process of extracting a modulating signal from the modulated signal is called demodulation. The devices used for demodulation are called demodulators.

Types of detector (1) square-law detectors

(2) Envelope detectors

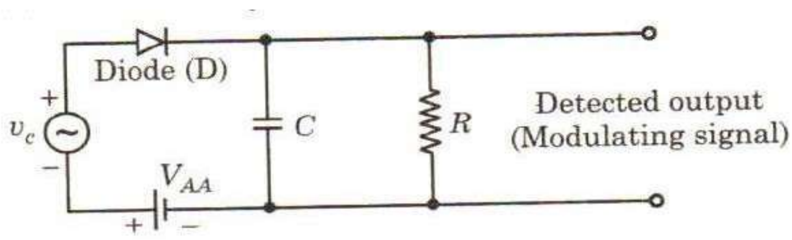
(3) PLL AM detector

AM signal with large carrier are detected by using the envelope detector uses the circuit which extracts the envelope of the am wave but detected by using square-low detectors.

2.3 DEMODULATION OF AM WAVES

SQUARE-LAW DETECTORS/LINEAR DIODE DETECTOR:-

The Square-Law Detector ckt is used for detecting modulated signal of small magnitude, so that operating region may be restricted to the non –linear portion of the v-characteristics of the device it may be observed that the circuit is very similar to the square law modulator. The only difference is that in square low modulator the filter used is a band pass filter where in a square law detector, a low pass filter is used.



Basic circuit of square law diode detector.

In the circuit, the dc supply voltage V_{AA} is used to get the fixed operating point in the non-linear portion of the diode V-I characteristics. Since, the operation is limited to the non-linear region of the diode characteristics, the lower half portion of the modulated wave form is compressed. This produces envelope applied distortion. Due to this the average value of the diode –current is no longer constant, rather it varies with time.

This distorted output diode current is expressed by

$I = av + bv^2$

v=is the i/p modulated voltage

AM wave is expressed as

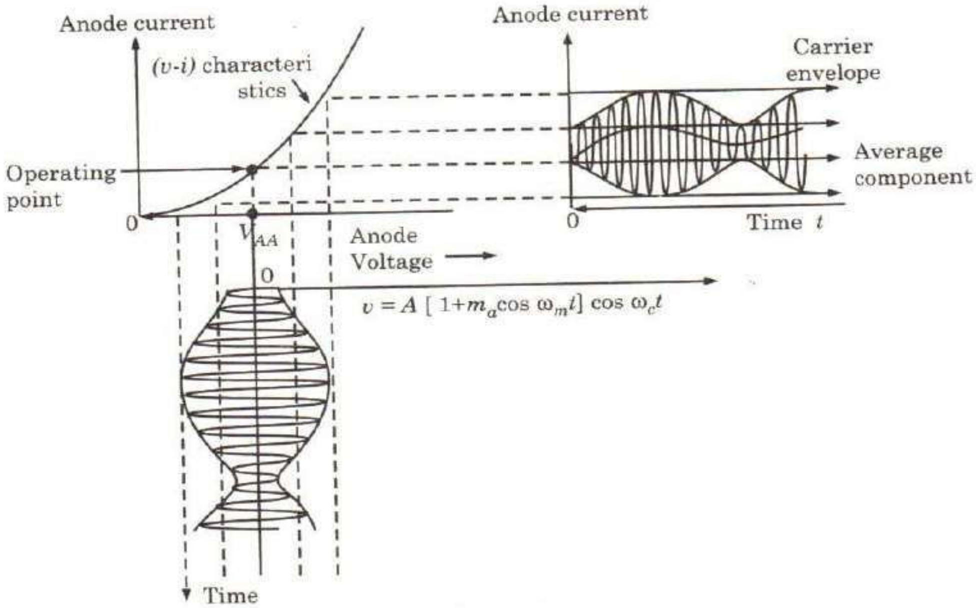
$v = A\,(1 + m_a \operatorname{Cos} \omega_m t) \operatorname{Cos} \omega_c t$

Substituting, the value of v, we get

$I = a\,[A\,(1 + m_a \operatorname{Cos} \omega_m t) \operatorname{Cos} \omega_c t] + b\,[A\,(1 + m_a \operatorname{Cos} \omega_m t) \operatorname{Cos} \omega_c t]\ 2$

If above expression is expanded, then we get terms of frequencies like $2\omega_c$, $2(\omega_c \pm \omega_m)$, ω_m & $2\omega_m$ besides the input frequency terms.

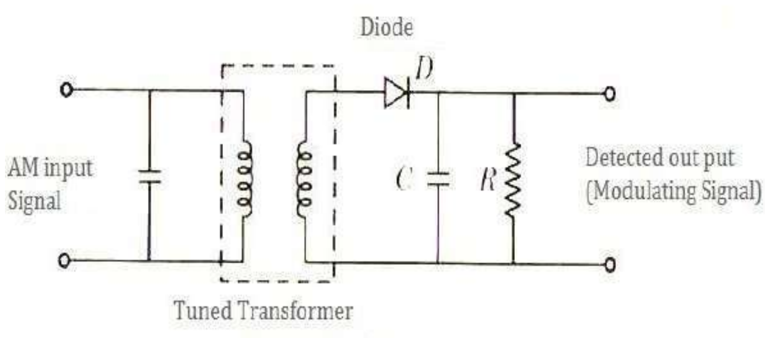
Hence this diode current I containing all these frequencies terms is passed through a low pass filter, which allows to pass the frequency below or up to modulating frequency ω_m and rejects the other higher frequency components. Therefore, the modulating signal with frequency ω_m is recovered from the input modulated signal.



ENVELOPE DETECTOR:-

A diode operating in a linear region of its V-I characteristics can extract the envelope of an AM wave. This type of detector is known as envelope detector. Envelope detector is most popular in commercial receiver circuits. Since it is very simple and is not expensive.

In the input portion of the ckt, the tuned transformer provides perfect tuning at the desired carrier frequency. RC network is the time-constant network. If the magnitude of the modulated signal at the input of the detector is 1 volt or more, the operation takes place in the linear portion of the V-I characteristics of diode.

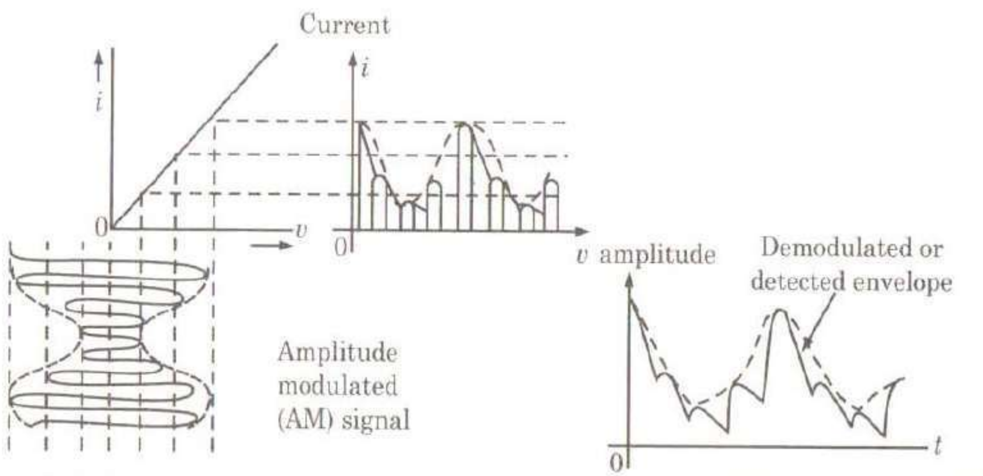


A Linear Diode Detecter

Operation:-

First, let us assume that the capacitor is absent in the ckt. In this case, the detector ckt will work as a half-wave rectifier. Therefore, the output waveform would be a half rectified modulated signal. Now let us consider that the capacitor is introduced in the circuit. For the +ve half cycle b, the diode conducts and the capacitor is charged to the peak value of the carrier voltage. However, for a –ve half cycle, the diode is reverse biased and does not conduct. This means that the input carrier voltage is disconnected from the RC circuit. Therefore the capacitor starts discharging through the resistance are with a time constant τ = RC is suitably chosen, the voltage across the capacitor C will not fall appreciably during the small period of –ve half cycle, and by that time the next +ve cycle appear. The +ve cycle again charges the capacitor C to the peak value of the voltage and thus this process repeats again and again.

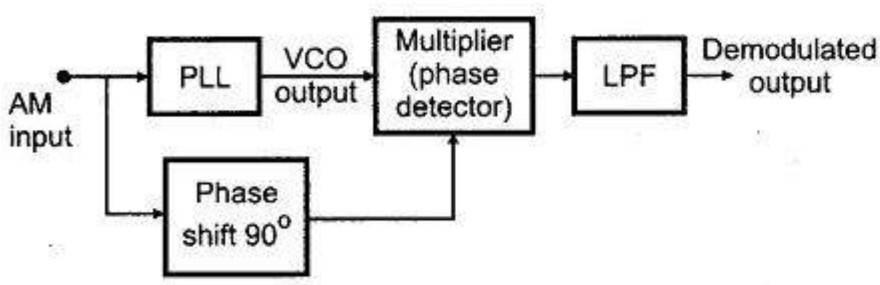
Hence the output voltage across the capacitor C is spiky modulating signal. However spikes are introduced because of charging and discharging of the capacitor C.



(a) Characteristics of linear diode detector (b) Detected output

AM Demodulator using Phase locked loop

A PLL can be used to demodulate AM signals.



- The PLL is locked to the carrier frequency of the incoming AM signal. Once locked the output frequency of VCO is same as the carrier frequency, but it is in unmodulated form.
- The modulated signal with 90° phase shift and the unmodulated carrier from output of PLL are fed to the multiplier. Since VCO output is always 90° out of phase with the incoming AM signal under the locked condition, both the signals applied to the multiplier are in same phase.
- Therefore, the output of the multiplier contains both the sum and the difference signals. The low pass filter connected at the output of the multiplier rejects high frequency components gives demodulated output.
- As PLL follows the input frequencies with high accuracy, a PLL AM detector exhibits a high degree of selectivity and noise immunity which is not possible with conventional peak detector type AM modulators.

2.4 DSB-SC SIGNAL AND SSB SIGNAL

DSB-SC

For 100% modulation about 67% of the total power is required for transmitting the carrier which does not contain any information. Hence, if the carrier is suppressed, only the sidebands remain and in this way a saving of two-third power may be achieved at 100% modulation. This type of suppression of carriers does not affect baseband signal. The resulting signal is DSB-SC signal.

As we know,

$$P_t = (1 + \frac{{m_a}^2}{2})P_c$$

$$Put\ m_a = 1$$

$$P_t = \left(1 + {}^{1}\!/_{2}\right)P_c$$

$$=> \ P_t = {}^{3}\!/_{2}\,P_c$$

$$=> \ P_c = {}^{2}\!/_{3}\,P_t$$

$$=> P_c = 0.67P_t$$

$So, P_c\ is\ 67\%\ of\ the\ transmitted\ power.$

2.5 METHODS OF GENERATING & DETECTION OF SSB-SC SIGNAL

SSB-SC

- Amplitude modulation and double-sideband suppressed carrier modulation are wasteful of bandwidth. Since then both need a transmission bandwidth equal to twice the message signal bandwidth.
- In either case one half of the transmission bandwidth is occupied by the upper sideband of the modulated signal whereas the other half is occupied by the lower sideband. As far as the transmission of information is concerned, only one sideband is necessary.
- Thus if the carrier and one of the two side bands are suppressed at the transmitter, no information is lost. Modulation of this type which provides a single sideband with supressed carrier is known as single sideband supressed carrier system. Thus, SSB-SC system reduces the transmission bandwidth by half.

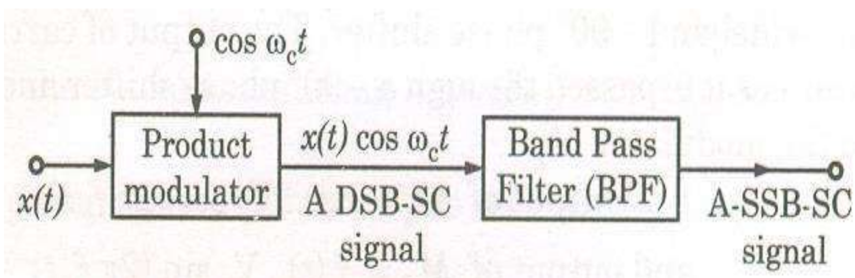
Generation-

SSB-SC signals may be generated by two methods

(i) Frequency discrimination
(ii) Phase discrimination

FREQUENCY DISCRIMINATION METHOD-

In a frequency discrimination method, a DSB-SC signal is generated by using an ordinary product modulator or balance modulator. After this, from the DSB-SC signal one of the two sidebands is filtered out by a suitable band pass filter.



Frequency Discrimination Method for SSB SC Generation

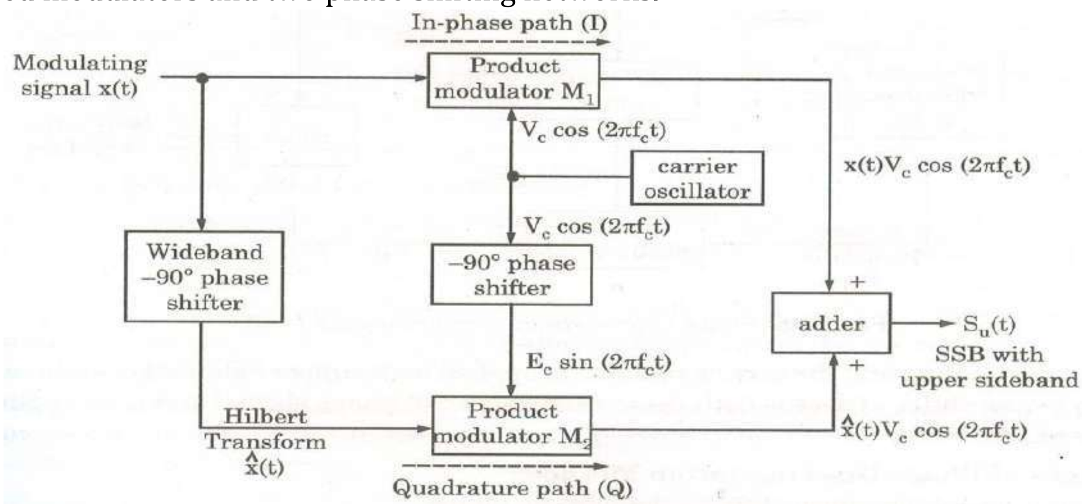
Limitations-

- The frequency discrimination method is useful only if the base band signal is restricted at its lower edge due to which the upper and lower sidebands are non-overlapping.

- The design of the band pass filter becomes difficult if the carrier frequency is quite higher than the bandwidth of the baseband signal.

PHASE-SHIFT METHOD-

The phase shift method avoids filter. This method makes use of the two balanced modulators and two phase shifting networks.



Phase Discrimination Method for SSB- SC signal

One of the modulators M_1 receives the carrier voltage shifted by 90° and the modulating voltage, whereas another balanced modulator M_2 receives the modulating voltage shifted by 90° and the carrier voltage. Both balanced modulators produce an output consisting only of sidebands. The two lower sidebands are out of phase and when combined together in the adder, they cancel each other. The upper sidebands are in phase and they added in the adder producing SSB in which the lower sideband has been cancelled.

DEMODULATION-

The baseband signal x(t) can be recovered from the SSB-SC signal by using the synchronous detection technique. With the help of synchronous detection method the spectrum of an SSB-SC signal centred about $\omega=\pm\omega c$, is retranslated to the baseband spectrum which is centered about $\omega=0$. The process of synchronous detection involves multiplication of the received SSB-SC signal with locally generated carrier. The generated carrier should have exactly the same frequency as that of the suppressed carrier. The product modulator multiplies the two signals at its input and the product signal is passed through a low pass filter with a bandwidth equal to fm. At the output of the filter, we get the modulating signal back.

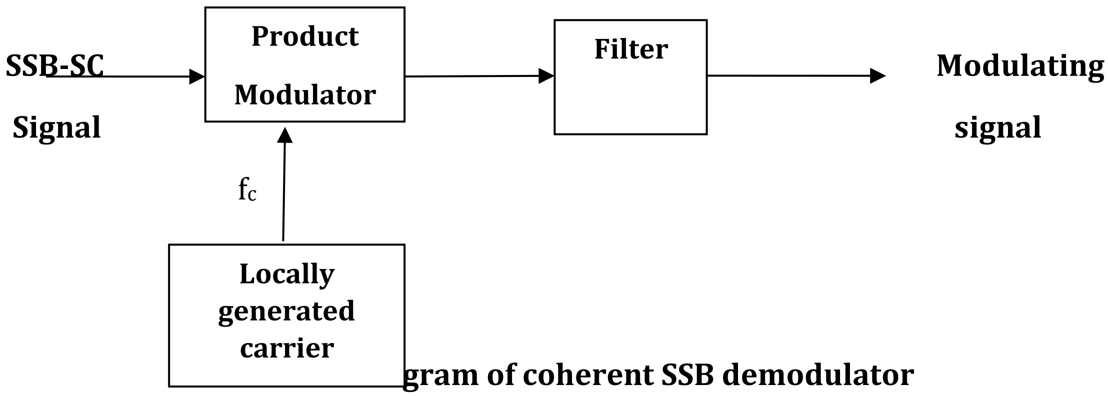
$e_d(t) = S(t)_{SSB} \times \cos\omega ct$

$= [x(t) \, \text{Cos}\omega_c t \pm x_n(t) \, \text{Sin}\omega_c t] \cos\omega_c t$

$= 1/2\, x(t) + \frac{1}{2} [x(t) \cos 2\omega_c t \pm x_n(t) \sin 2\omega_c t]$

When $e_d(t)$ is passed through a low pass filter, then the terms cantered about $\pm 2\omega_c$ are filtered out and we get, at the output of detector, signal e_o which is given as

$$e_o(t) = 1/2 \;\; x(t)$$



gram of coherent SSB demodulator

2.6 METHODS OF GENERATION OF DSB-SC SIGNAL

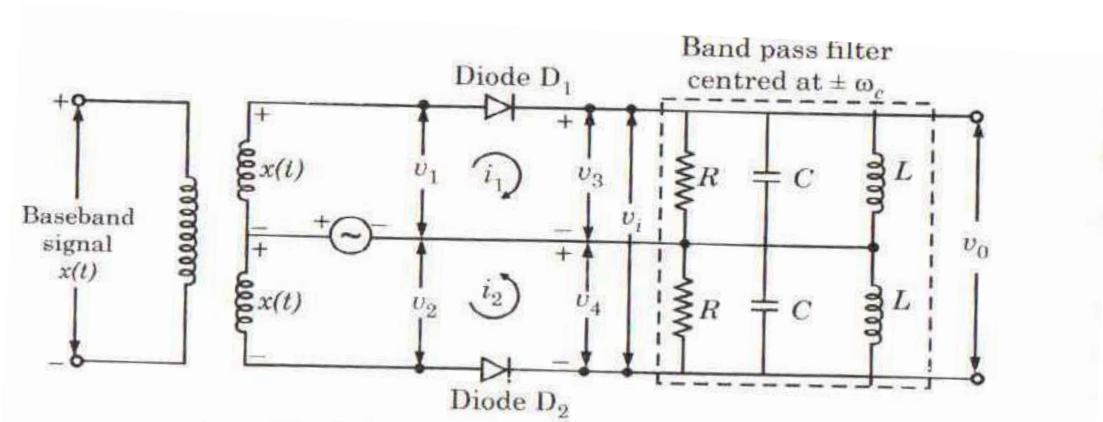
GENERATION OF DSB-SC SIGNAL-

A circuit used to achieve the generation of a DSB-SC signal is called a product modulator. There are two types of product modulator.

1. Balanced Modulator
2. Ring Modulator

Balance Modulator :-

A non-linear resistance or a non –linear device may be used to produce amplitude modulation i.e, one carrier and two sidebands. However a DSB-SC signal contains only two sidebands. Thus if two nom- linear devices such as diodes , transistors etc. are connected in a balanced mode so as to suppress the carriers of each other , then only sidebands are left i.e. a DSB- SC signal is generated.



Balanced Modulator using Diodes.

Therefore a Balanced Modulator may be defined as a circuit in which two nonlinear devices are connected in a balanced mode to produce a DSB-SC signal. A modulating signal x(t) is applied to the diodes through a center-tapped transformer with the carrier signal Cos ω_ct.

A non- linear VI relationship is given as,

i= av + bv^2 where v is the input voltage applied across a non- linear device and i is the current through the non-linear device.

For diode D_1, $i_1 = av_1 + bv_1^2$

Similarly, For diode D_2, $i_2 = av_2 + bv_2^2$

$$v_1 = \cos\omega_c t + x(t)$$

$$v_2 = \cos\omega_c t - x(t)$$

Done to currents i_1 and i_2 the net voltage v_i at the input of band pass filter expressed as $v_i = i_1R - i_2R$.

After substituting the values of i_1 & i_2 we get

$$v_i = 2R[ax(t) + 2bx(t)\cos\omega_c t]$$

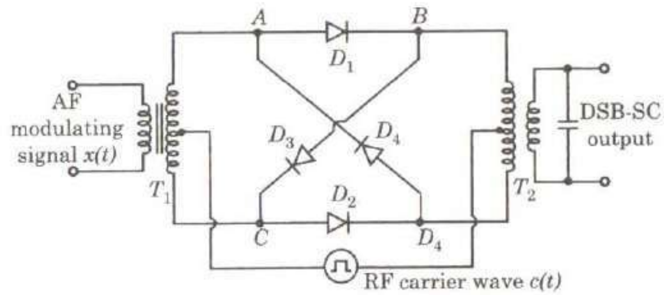
A band pass filter is that type of filter which allows to pass a band of frequencies. Here the band pass filter is centred around $\pm\omega_c$, it will pass a narrow band of frequencies cantered at $\pm\omega_c$.

The output of the BPF is

$$v_o = 4bR\, x(t) \cos\omega_c t$$

Ring Modulator-

Ring Modulator is another product Modulator, which is used to generate DSB-SC Signal. In a ring modulator circuit, four diodes are connected in the form of ring in which all four diodes point in same manner. All the four diodes in ring are controlled by a square wave carrier signal c(t) of frequency f_c applied through a centre tapped transformer.



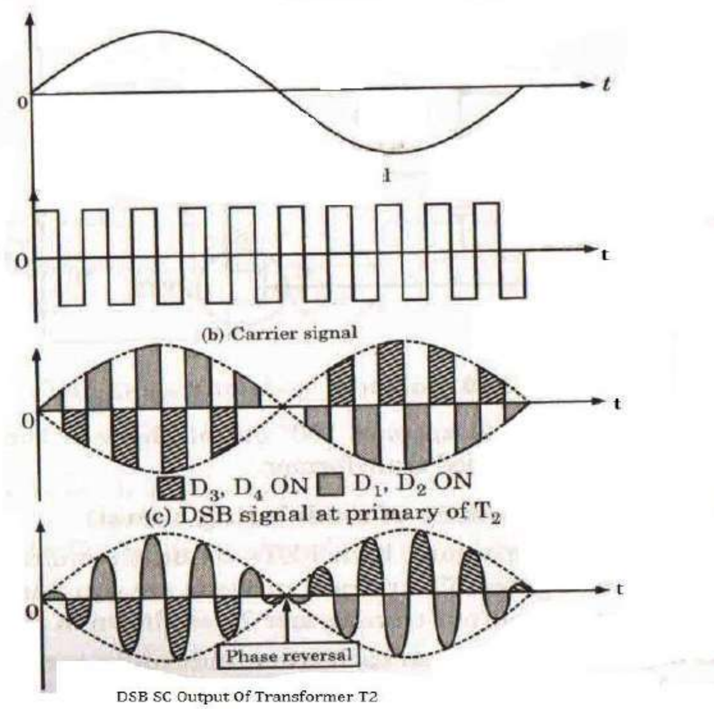
A diode ring modulator.

In case, when diodes are ideal and transformer are perfectly balanced, the two outer diodes are switched on if the carrier signal is positive whereas the two inner diodes are switched off and thus presenting very high impedence. Under this condition, the modulator multiplies the modulating signal x(t) by +1.

When carrier signal is –ve , he situation becomes reversed . In this case the modulator multiplies the modulating signal by -1.

C(t) = $4/\pi \sum_{n=1}^{\infty} \frac{(-1)^{n=1}}{2n-1} \{\cos[2\pi f_c t(2n-1)]\}$

We have S(t) = x(t) C(t)



Waveform of Lattice type Balanced Modulator

A Ring modulator is also known as a double balanced modulator. The modulating signal is band limited to $-f_m \le f \le f_m$. The desired sideband around the carrier frequency f_c may be selected using band pass filter having centre frequency ω_c and bandwidth $2f_m$. To avoid overlapping of side bands f_c is greater than f_m .

2.7 DETECTION OF DSB-SC SIGNAL-

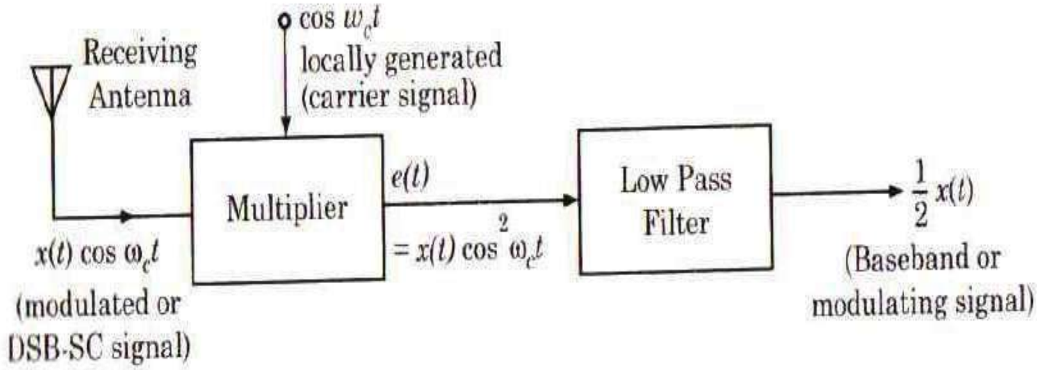
The DSB-SC signal may be demodulated by following two methods.

1. Synchronous detection method
2. Using envelope detector after carrier reinsertion.

Synchronous detection Method-

DSB-SC signal is transmitted from the transmitter and it reaches the receiver through a transmission medium. At the receiver end, the original modulating signal x(t) is recovered from the modulated signal. This can be achieved by simply retranslating the baseband or modulating signal from a higher spectrum, cantered at ±ωc, to the original spectrum. This process is called demodulation or detection. Hence, the original or baseband signal is recovered from the modulated signal by the detection process.

A method of DSB-SC detection is known as synchronous detection.



Synchronous detection method.

Working principle-

In synchronous detection method, the received modulated or DSB-SC signal is first multiplied with a locally generated carrier signal Cos ω_ct and then passed through a low pass filter. At the output of a low pass filter, the original modulating signal is recovered.

2.8 VESTIGIAL SIDE BAND MODULATION:

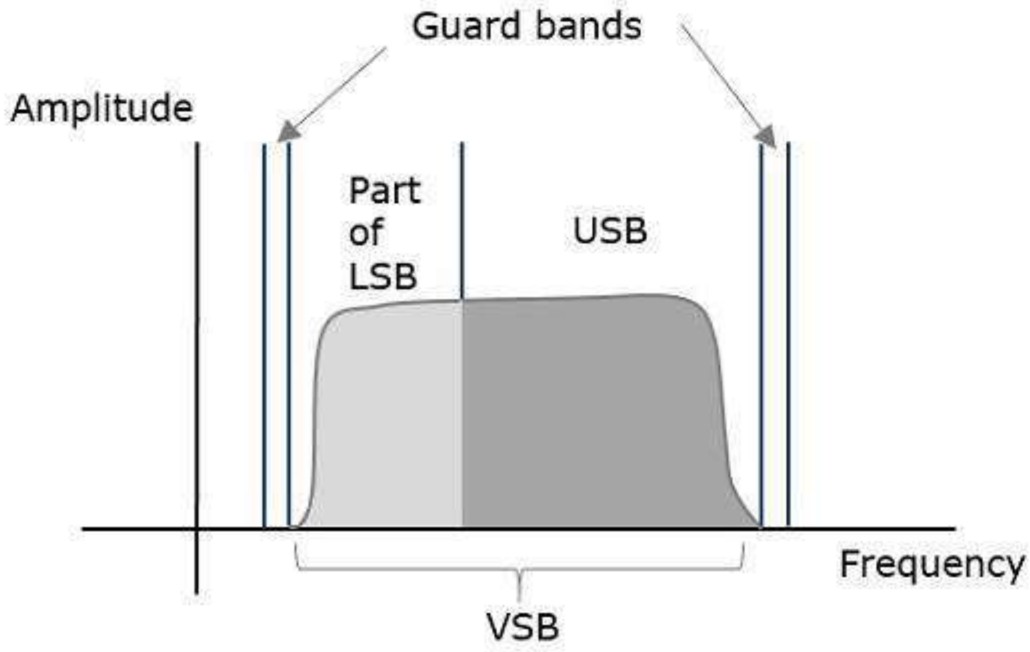
In case of SSB modulation, when a sideband is passed through the filters, the band pass filter may not work perfectly in practice. As a result of which, some of the information may get lost.

Hence to avoid this loss, a technique is chosen, which is a compromise between **DSB-SC** and **SSB**, called as **Vestigial Sideband (VSB)** technique. The word vestige which means "a part" from which the name is derived.

Vestigial Sideband

Both of the sidebands are not required for the transmission, as it is a waste. But a single band if transmitted, leads to loss of information. Hence, this technique has evolved.

Vestigial Sideband Modulation or **VSB Modulation** is the process where a part of the signal called as **vestige** is modulated, along with one sideband. A VSB signal can be plotted as shown in the following figure.



VSB Modulation

Along with the upper sideband, a part of the lower sideband is also being transmitted in this technique. A guard band of very small width is laid on either side of VSB in order to avoid the interferences. VSB modulation is mostly used in television transmissions.

VSB Modulation – Advantages

Following are the advantages of VSB –

- Highly efficient.
- Reduction in bandwidth.
- Filter design is easy as high accuracy is not needed.
- The transmission of low frequency components is possible, without difficulty.
- Possesses good phase characteristics.

VSB Modulation – Disadvantages

Following are the disadvantages of VSB –

- Bandwidth when compared to SSB is greater.
- Demodulation is complex.

UNIT-3: ANGLE MODULATION SYSTEMS

3.1 CONCEPT OF ANGLE MODULATION & ITS TYPES

Angle modulation is the process in which the frequency or the phase of the carrier signal varies according to the message signal.

The standard equation of the angle modulated wave is

$$S\ (t) = A_c Cos\theta_i(t)$$

Where A_c is the amplitude of the modulated wave, which is the same as the amplitude of the carrier signal.

$\theta_i(t)$ is the angle of the modulated wave

TYPES

Angle modulation is further divided into frequency modulation and phase modulation.

Frequency Modulation- it is the process of varying the frequency of the carrier signal linearly with the message signal.

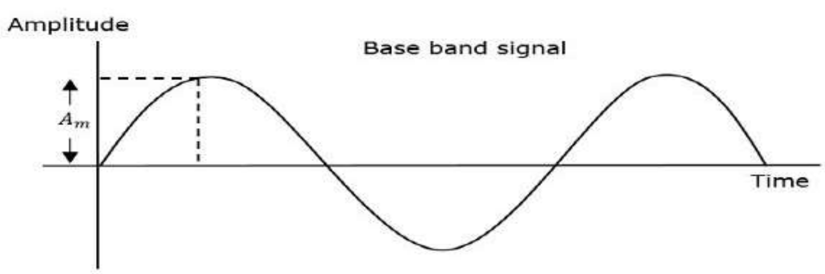
Phase Modulation- It is the process of varying the phase of the carrier signal linearly with the message signal.

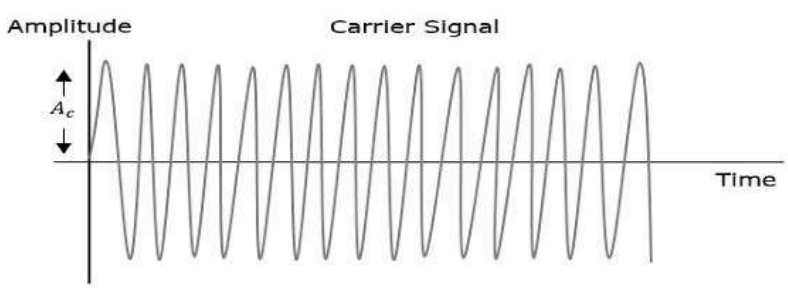
3.2 BASIC PRINCIPLE OF FREQUENCY MODULATION & FREQUENCY SPECTRUM OF FM SIGNAL

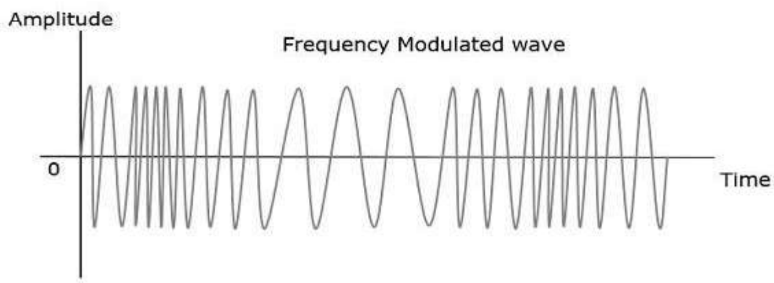
FREQUENCY MODULATION

In frequency modulation, the frequency of the carrier signal varies in accordance with the instantaneous amplitude of the modulating signal. Here amplitude and phase of the carrier signal remains constant.

The frequency of the modulated wave increases, when the amplitude of the modulating signal increases and the frequency of the modulated wave decreases, when the amplitude of the modulating signal decreases.







FREQUENCY SPECTRUM OF FM SIGNAL-

The frequency spectrum of the signal

$$v(t) = \cos(\omega_c t + \beta \sin \omega_m) \qquad \text{.......(1)}$$

which is the signal with the amplitude arbitrarily set at unity.

We have

$$\cos(\omega_c t + \beta \sin \omega_m) = \cos \omega_c t \, cos \, (\beta \sin \omega_m t) - \sin \omega_c t \, sin \, (\beta \sin \omega_m t) \quad \text{...........(2)}$$

Consider now the expression $cos\ (\beta \sin \omega_m t)$ which appears as a factor on the right hand side. It is an even, periodic function having an angular frequency ω_m. Therefore, it is possible to expand this expression in a Fourier series in which $\omega_m/2\pi$ is the fundamental frequency. The coefficients are functions of β, and, since function is even, the coefficients of the odd harmonics are zero. The result is

$$\cos \omega_c t \, cos \, (\beta \sin \omega_m t) = J_0(\beta) + 2J_2(\beta) \cos 2\omega_m \mathrm{t} + 2J_4(\beta) \cos 4\omega_m \mathrm{t} + \cdots \ldots \ldots + 2\, J_{2n}(\beta) \cos 2n\omega_m t + \cdots ..$$

While for $\sin \omega_m t$, which is an odd function, we find the expansion contains only odd harmonics and is given by

$$sin\ (\beta \sin \omega_m t) = 2J_1(\beta) \sin \omega_m t + 2J_3(\beta) \sin 3\omega_m \mathrm{t} + \cdots \ldots \ldots \ldots \ldots + 2\, J_{2n-1}(\beta) \sin\ (2n-1)\omega_m t + \cdots ..$$

The functions $J_n(\beta)$ occur often in the solution of engineering problem. They are known as Bessel functions of the first kind and of order n.

Putting the results given and using the identities

$$cos\ A. cos\ B = ½\, cos\ (A - B) + ½\, cos\ (A - B)$$

$$sin\ A. sin\ B = ½\, cos\ (A - B) - ½\, cos\ (A - B)$$

We find v (t) becomes

$$v(t) = J_0(\beta)\cos\omega_c - J_1(\beta)[cos(\omega_c - \omega_m)t - \cos(\omega_c + \omega_m)\,t]$$
$$+ J_2(\beta)[cos(\omega_c - 2\omega_m)t + \cos(\omega_c + 2\omega_m)\,t]$$
$$- J_3(\beta)[cos(\omega_c - 3\omega_m)t - \cos(\omega_c + 3\omega_m)\,t] + \cdots \ldots$$

Observe that the spectrum is composed of a carrier with an amplitude and a set of sidebands spaced symmetrically on either side of the carrier at frequency separations of ω_m, $2\omega_m$, $3\omega_m$, etc.

3.3 EXPRESSION FOR FREQUENCY MODULATED SIGNAL & MODULATION INDEX

EXPRESSION FOR FREQUENCY MODULATED SIGNAL-

The equation for instantaneous frequency f_i in FM modulation is:

$$f_i = f_c + K_f m(t)$$

Where f_c is the carrier frequency

K_f is the frequency sensitivity

m(t) is the message signal

The relationship between angular frequency w_i and angle $\theta_i(t)$ is

$$w_i = \frac{d\theta_i(t)}{dt}$$
$$\Rightarrow 2\pi f_i = \frac{d\theta_i(t)}{dt}$$
$$\Rightarrow \theta_i(t) = 2\pi\int f_i \, \mathrm{dt}$$

Substitute f_i value in the equation

$$\theta_i(t) = 2\pi\int (f_c + K_f m(t)) \; \mathrm{dt}$$
$$\Rightarrow \theta_i(t) = 2\pi f_c t + 2\pi K_f \int m(t)\; dt$$

Substitute the $\theta_i(t)$ value in the standard equation of angle modulated wave.

$$\mathrm{S(t)}= A_c Cos(2\pi f_c t + 2\pi K_f \int m(t)\, dt$$

If the modulating signal is m(t)= $A_m \cos(2\pi f_m t)$, then the equation of FM wave will be

$$\mathrm{S(t)}= A_c Cos(2\pi f_c t + 2\pi K_f \int A_m\, cos\,(2\pi f_m t\,)\, dt$$
$$\Rightarrow \mathrm{S(t)}= A_c Cos(2\pi f_c t + 2\pi K_f \frac{1}{2\pi f_m} A_m\, Sin\,(2\pi f_m t\,))$$
$$\Rightarrow \mathrm{S(t)}= A_c Cos(2\pi f_c t + \beta\, Sin\,(2\pi f_m t\,))$$

Where β= modulation index

$$= \frac{k_f A_m}{f_m} = \frac{\Delta f}{f_m}$$

The difference between FM modulated frequency and normal carrier frequency is termed as frequency deviation. It is denoted by Δf.

$$\Delta f = k_f A_m$$

MODULATION INDEX-

The modulation index is defined as the ratio of frequency deviation to the modulating frequency.

Modulation index, β = frequency deviation/modulation frequency

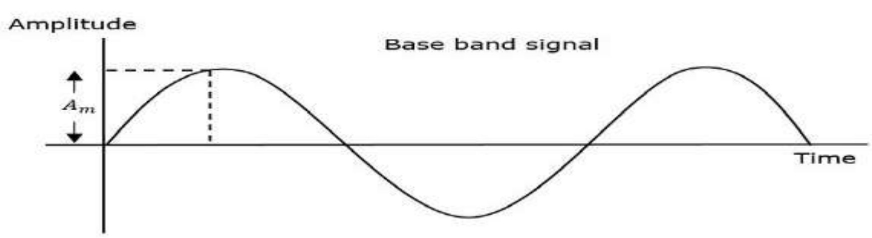
Or $\beta = \frac{\Delta f}{f_m}$

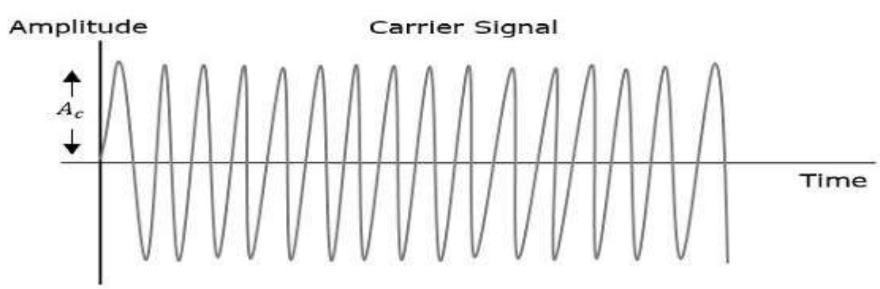
This modulation index may be greater than unity.

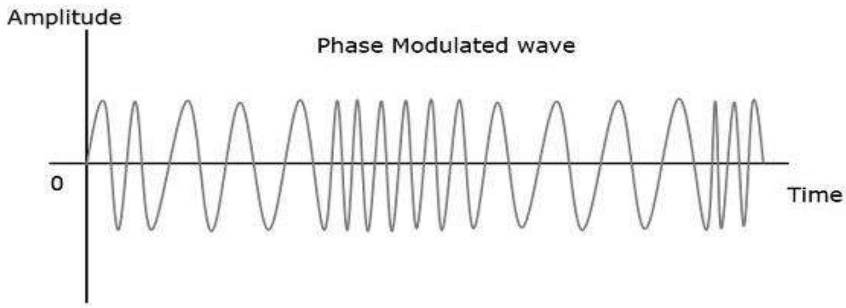
3.4 PHASE MODULATION & DIFFERENCE OF FM & PM-

PHASE MODULATION-

In phase modulation, the phase of the carrier signal varies in accordance with the instantaneous amplitude of the modulating signal. Here with the change in phase, the frequency of the signal also shows variation. Thus it can be said that while phase modulating any signal, the phase as well as the frequency of the carrier signal shows variation.







The figure shows a sinusoidal message signal that is to be transmitted from one end to another, a carrier signal which is to be phase modulated and the last figure represents the phase modulated signal. Here it is clear from the above figure that when the amplitude of the sinusoidal signal starts to increase and reaches the maximum value then the phase lead of the carrier signal gets increased. Due to this a compression in the carrier signal is noticed.

However, when the amplitude of the modulating signal starts falling and attains a minimum value, then the phase lag of the carrier wave occurs. Due to this, the frequency of the signal gets increased.

The equation for instantaneous phase Φ_i in phase modulation is

$$\Phi_i = K_P m(t)$$

Where, K_P is the phase sensitivity

m(t) is the modulating signal

The standard equation of angle modulated wave is

S(t)= $A_c Cos\ (2\pi f_c t + \Phi_i)$

Substitute, Φ_i value in the above equation

S(t)= $A_c Cos\ (2\pi f_c t + K_P m(t))$

If the modulating signal, m(t)= $A_m\ cos\ (2\pi f_m t)$ then the equation of PM wave will be

S(t)= $A_c Cos\ (2\pi f_c t + K_P A_m\ cos\ (2\pi f_m t\ ))$

=> S(t)= $A_c Cos\ (2\pi f_c t + \beta\ cos\ (2\pi f_m t\ ))$
Where, β= modulation index

<u>DIFFERENCE OF FM & PM-</u>

S.No.	FM	PM
1.	$s(t) = V_c \sin[\omega_c t + m_f \sin \omega_m t]$	$s(t) = V_c \sin[\omega_c t + m_p \sin \omega_m t]$
2.	Frequency deviation is proportional to modulating voltage.	Phase deviation is proportional to the modulating voltage.
3.	Associated with the change in f_c, there is some phase change.	Associated with the changes in phase, there is some change in f_c.
4.	m_f is proportional to the modulating voltage as well as the modulating frequency f_m.	m_p is proportional only to the modulating voltage.
5.	It is possible to receive FM on a PM receiver.	It is possible to receive PM on a FM receiver.
6.	Noise immunity is better than AM and PM.	Noise immunity is better than AM but worse than FM.
7.	Amplitude of the FM wave is constant.	Amplitude of the PM wave is constant.
8.	Signal to noise ratio is better than that of PM.	Signal to noise ratio is inferior to that in FM.
9.	FM is widely used.	PM is used in some mobile systems.
10.	In FM, the frequency deviation is proportional to the modulating voltage only.	In PM, the frequency deviation is proportional to both the modulating voltage and modulating frequency.

3.5 COMPARE BETWEEN AM AND FM MODULATION-

AM-

(i) Amplitude of AM wave will change with the modulating voltage.
(ii) Transmitted power is dependent on the modulation index.
(iii) Carrier power and one sideband power are useless.
(iv) AM receivers are not immune to noise.
(v) Frequency deviation feature is absent in AM.
(vi) Bandwidth = $2f_m$. It is not dependent on the modulation index.
(vii) Bandwidth is much less than FM.
(viii) Ground wave and sky wave propagation is used. Therefore larger area is covered than FM.
(ix) Not possible to operate more channels on the same frequency.
(x) AM equipment are less complex.
(xi) Number of sidebands in AM will be constant and equal to 2.
(xii) The information is contained in the amplitude variation of the carrier.

FM-

(i) Amplitude of FM wave is constant. It is independent of the modulation index.
(ii) Transmitted power remains constant. It is independent of mf.
(iii) All the transmitted power is useful.
(iv) FM receivers are immune to noise.
(v) It is possible to decrease noise further by increasing deviation.
(vi) Bandwidth = $2[\Delta_f + f_m]$. The bandwidth depends on modulation index.
(vii) Bandwidth is large. Hence, wide channel is required.
(viii) Space wave is used for propagation. So, radius of transmission is limited to line of sight.
(ix) It is possible to operate several transmitters on same frequency.
(x) FM transmission and reception equipment are more complex.
(xi) The number of sidebands having significant amplitudes depends on modulation index mf.
(xii) The information is contained in the frequency variation of the carrier.

3.6 FM GENERATION-

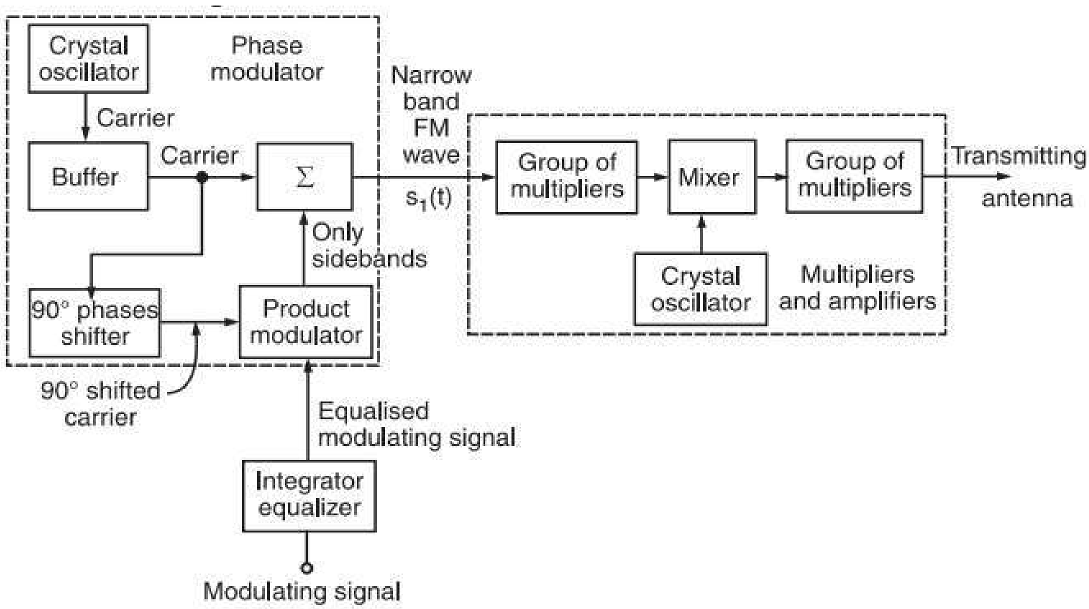
The FM modulator circuits used for generating FM signals may be put into two categories as under.

(i) The direct method or parameter variation method
(ii) The indirect method or the Armstrong method

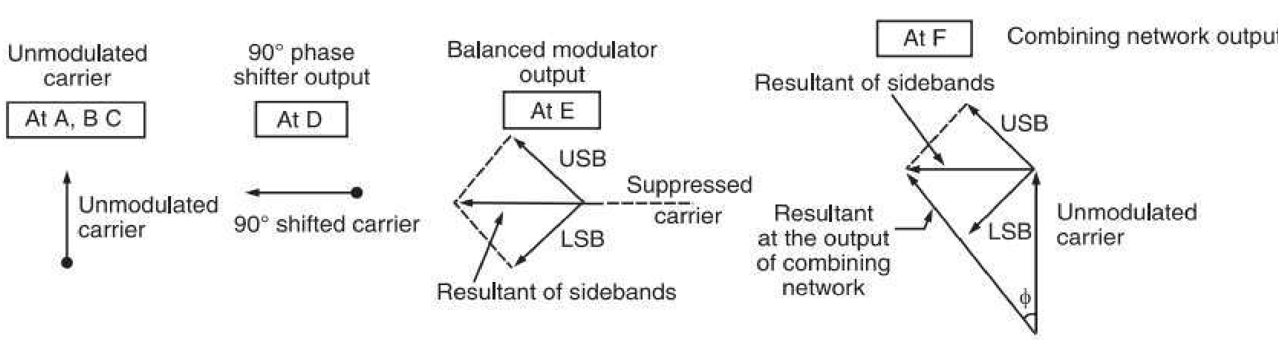
INDIRECT METHOD OR THE ARMSTRONG METHOD:

In direct methods of generation of FM, LC oscillators are used. The crystal oscillator cannot be used. The LC oscillators are not stable enough for the broadcast purpose. Thus, the direct methods cannot be used for the broadcast application. In order to overcome the limitation of direct method, we use indirect method of FM generation called as the Armstrong method.

In this method, the FM is obtained through phase modulation. A crystal oscillator can be used hence the frequency stability is very high and this method is widely used in practice. The Armstrong method uses the phase modulator to generate a frequency modulated wave. Crystal oscillator produce stable frequency upto 1MHz.



- The modulating signal x(t) is passed through an integrator before applying it to the phase modulator.
- The crystal oscillator produces a stable unmodulated carrier which is applied to the 90° phase shifter as well as the combining network through a buffer.
- The 90° phase shifter produces a 90° phase shifted carrier. It is applied to the balanced modulator along with the modulating signal. Thus, the carrier used for modulation is 90° shifted with respect to the original carrier.
- At the output of the product modulator, we get DSB SC signal i.e., AM signal without carrier. This signal consists of only two sidebands with their resultant in phase with the 90° shifted carrier.
- The two sidebands and the original carrier without any phase shift are applied to a combining network (∑). At the output of the combining network, we get the resultant of vector addition of the carrier and two sidebands



- As the amplitude of modulating signal increases the modulation index will increase and the amplitude of sidebands will also increase. Hence the amplitude of their resultant increases.
- This will increase the angle Ø made by the resultant with unmodulated carrier. The angle Ø deceases with reduction in modulation index. Thus the resultant at the output of the combining network is phase modulated.
- The FM signal produced at the output of phase modulator has a low carrier frequency and low modulation index. They are increased to an high value with the help of frequency multiplies, mixer and amplifier.
- For low modulation index, Ø is small and for high modulation index, Ø increases.

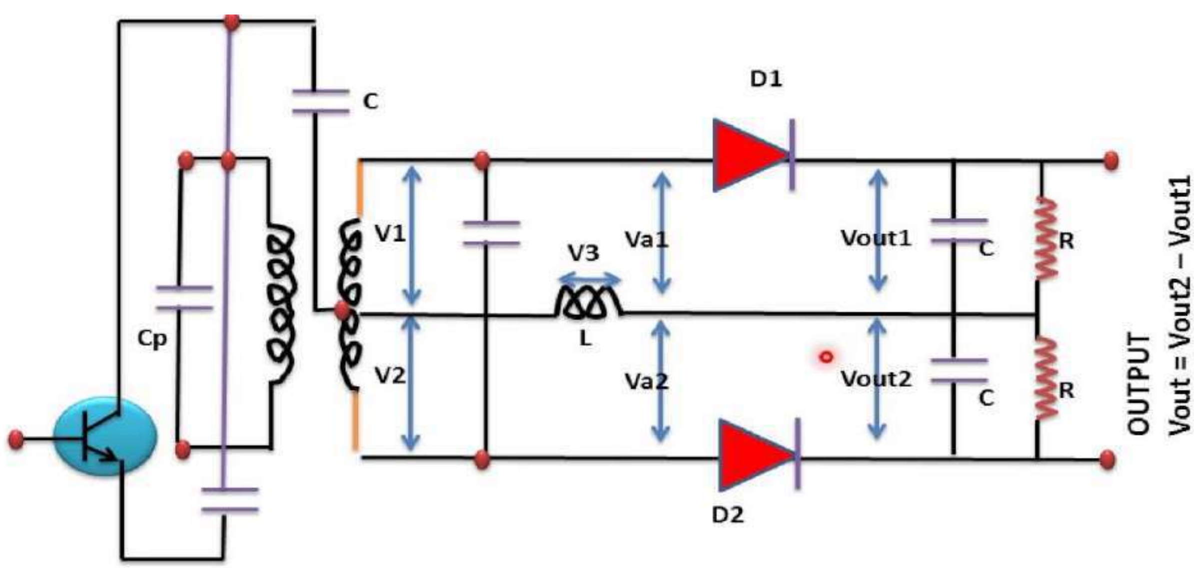
3.7 FM DEMODULATOR-

The demodulation process of FM waves is exactly opposite to that of the frequency modulation. After demodulation, we get the original modulating signal at the demodulation output.

METHODS OF FM DEMODULATION-

FORSTER SEELY DETECTOR-

The circuit diagram of phase discriminator or Foster Seeley Discriminator is given below



This circuit consists of an inductively coupled double tuned circuit in which both primary and secondary coils are tuned to the same frequency. The center of the secondary coil is connected to the top of the primary through a capacitor C. this capacitor performs the functions are:

(i) It blocks the D.C. from primary to secondary.

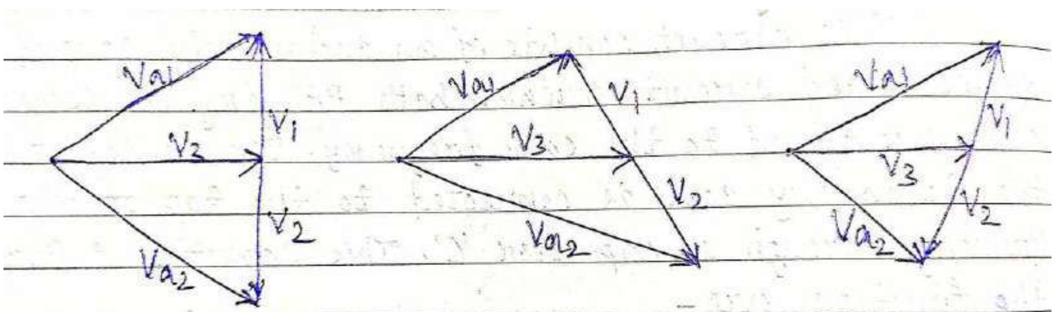
(ii) It couples the signal frequency from primary to center tapping of the secondary.

The primary voltage V_3 appears across the inductor in primary side. Nearly entire voltage V_3 appears across inductor L except a small drop across the capacitor C.

The center tapping of the secondary coil has an equal and opposite voltage across each half winding. Hence V_1 and V_2 are equal in magnitude but opposite in phase. The radio frequency voltages V_{a1} and V_{a2} applied to the diodes D_1 *and* D_2 are expressed as

$$V_{a1} = V_3 + V_1$$

$$V_{a2} = V_3 + V_2$$



Voltages V_{a1} and V_{a2} depend upon the phasor relations between V_1, V_2 *and* V_3. The phase position of V_1 *and* V_2 relative to V_3 would depend upon the tuned secondary coil at the resonance or off the resonance.

- At resonance- when an input voltage has a frequency equal to the resonant frequency of the tuned secondary, V_3 is in phase quadrature with V_1 and V_2. The resultant voltage V_{a1} *and* V_{a2} are equal in magnitude.
- Off resonance- when an input signal frequency is above the resonant frequency the phase difference between V_3 and V_1 is 45°. Because V_2 is in phase opposition of V_1 the phase difference between V_3 and V_2 is 135°. The phasor diagram reveals that V_{a1} is reduced where as V_{a2} is increased. The situation is reversed when the input voltage has a frequency below the resonant frequency. Hence the amplitude of the voltage V_{a1} *and* V_{a2} will vary.

The voltage V_{a1} *and* V_{a2} are separately rectified by diodes D_1 and D_2 respectively to produce V_{out1} *and* V_{out2}. The output voltage V_o is

$$V_o = |V_{out2}| - |V_{out1}|$$

<u>Advantages</u>:

1.It is more easy to align than the balanced slope detector as there are only two tuned circuits and both are to be tuned at the same frequency f_c.

2. Linearity is better. This is because the operation of the circuit is dependent more on the primary to secondary relationship which is very much linear.

Drawbacks

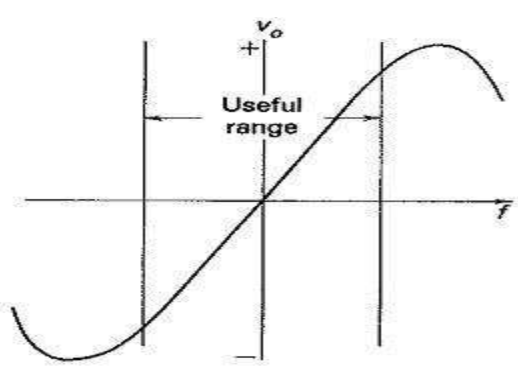
It does not provide amplitude limiting. So in the presence of noise or any other spurious amplitude variations, the demodulator output responds to them and produce errors.

RATIO DETECTOR-

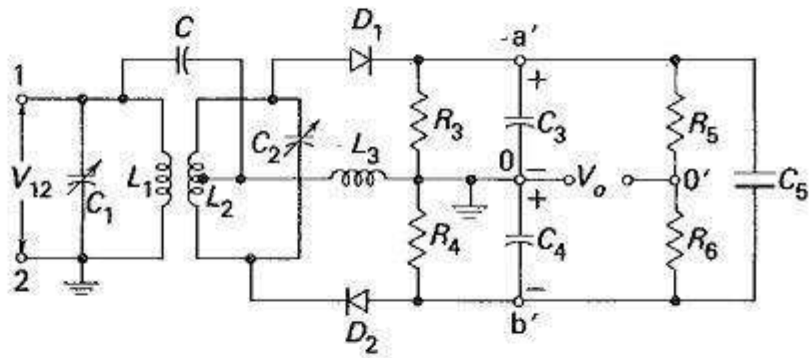
The RATIO DETECTOR uses a double-tuned transformer to convert the instantaneous frequency variations of the fm input signal to instantaneous amplitude variations. These amplitude variations are then rectified to provide a dc output voltage which varies in amplitude and polarity with the input signal frequency. This detector demodulates fm signals and suppresses amplitude noise without the need of limiter stages.

In the Foster-Seeley discriminator, changes in the magnitude of the input signal will give rise to amplitude changes in the resulting output voltage. This makes prior limiting necessary. It is possible to modify the discriminator circuit to provide limiting, so that the amplitude limiter may be dispensed with. A circuit so modified is called a Ratio Detector Circuit.

As we now, the sum V_{ao} + V_{bo} remains constant, although the difference varies because of changes in input frequency. This assumption is not completely true. Deviation from this ideal does not result in undue distortion in the Ratio Detector Circuit, although some distortion is undoubtedly introduced. It follows that any variations in the magnitude of this sum voltage can be considered spurious here. Their suppression will lead to a discriminator which is unaffected by the amplitude of the incoming signal. It will therefore not react to noise amplitude or spurious amplitude modulation.



It now remains to ensure that the sum voltage is kept constant. Unfortunately, this cannot be accomplished in the phase discriminator, and the circuit must be modified.. This is used to show how the circuit is derived from the discriminator and to explain its operation. It is seen that three important changes have been made: one of the diodes has been reversed, a large capacitor (C_5) has been placed across what used to be the output, and the output now is taken from elsewhere.



Operation:

With diode D_2 reversed, o is now positive with respect to b', so that $V_{a'b'}$ is now a sum voltage, rather than the difference it was in the discriminator. It is now possible to connect a large capacitor between a' and b' to keep this sum voltage constant. Once C_5 has been connected, it is obvious that $V_{a'b'}$ is no longer the output voltage; thus the output voltage is now taken between o and o′. It is now necessary to ground one of these two points, and o happens to be the more convenient, as will be seen when dealing with practical Ratio Detector Circuit. Bearing in mind that in practice $R_5 = R_6$, V_o is calculated as follows:

$$V_o = V_{b'o'} - V_{b'o} = \frac{V_{a'b'}}{2} - V_{b'o} = \frac{V_{a'o} + V_{b'o}}{2} - V_{b'o}$$

$$= \frac{V_{a'o} - V_{b'o}}{2}$$

The above equation shows the ratio detector output voltage is equal to half the difference between the output voltages from the individual diodes. Thus (as in the phase discriminator) the output voltage is proportional to the difference between the individual output voltages. The Ratio Detector Circuit therefore behaves identically to the discriminator for input frequency changes.

Unit-4: AM & FM TRANSMITTER & RECEIVER

4.1. CLASSIFICATION OF RADIO RECEIVERS-

Radio receivers are classified in two ways.

(A) Depending upon the applications, the radio receivers are classified as:

1) Amplitude Modulation broadcast receivers: These receivers are used to receive the broadcast of speech or music transmitted from amplitude modulation broadcast transmitters which operate on long wave, medium wave or short wave bands
2) Frequency Modulation broadcast receivers: These receivers are used to receive the broadcast programs from FM broadcast transmitters which operate in VHF or UHF bands.
3) Communication receivers: Communication receivers are used for reception of telegraph and short wave telephone signals. This means that communication receivers are used for various purposes other than broadcast services.
4) Television receivers: Television receivers are used to receive television broadcast in VHF or in UHF bands.
5) Radar receivers: Radar receivers are used to receive radar signals.

(B) Depending upon the fundamental aspects, the radio receivers are classified as:

1) Tuned Radio Frequency receivers
2) Super Heterodyne receiver

4.2 DEFINATION-

FIDELITY-

The fidelity of a receiver is its ability to accurately reproduce in its output, the signal that appears at its input.

SENSITIVITY-

The sensitivity of a radio receiver is its ability to amplify weak signal.

SELECTIVITY-

The ability of a device to respond to a particular frequency without interference from others. The selectivity of a receiver is its ability to reject unwanted signal and accept the desired signal.

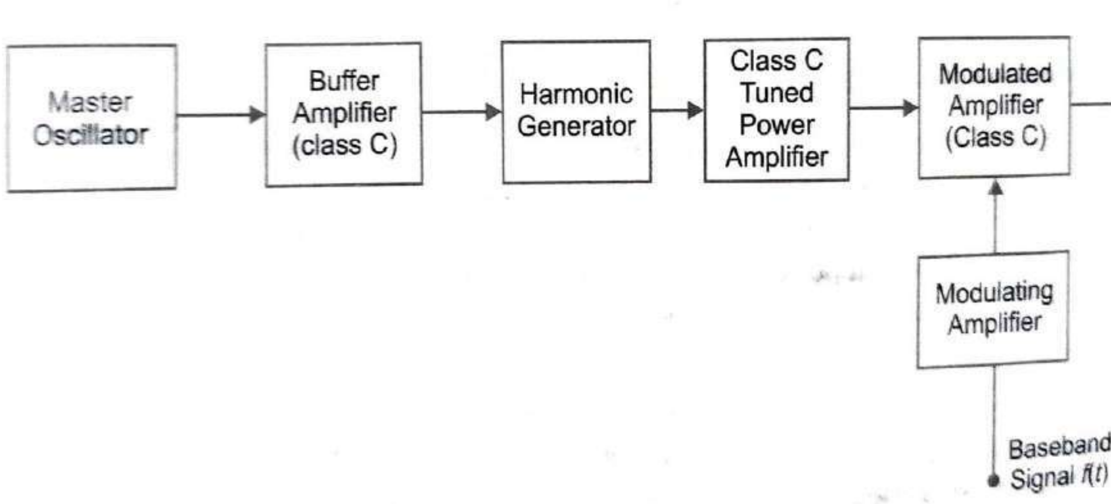
NOISE FIGURE-

It is defined as the ratio of signal to noise ratio at the input to that at the output.

$$NF = \frac{(S/N)_i}{(S/N)_o}$$

Noise figure is the measures of degradation of the signal to noise ratio caused by components in a signal chain. It is a number by which the performance of an amplifier or a radio receiver can be specified with lower values indicating better performance.

4.3 AM TRANSMITTER-



Master Oscillator-

It generates the carrier signal, which lies in the RF range. As we know the frequency of the carrier is always very high. But it very difficult to generate high frequencies with good frequency stability. The master oscillator generates a sub multiple with the required carrier frequency.

This submultiple frequency is multiplied by the frequency multiplier stage to get the required carrier frequency. A crystal oscillator can be used in this stage to generate a low frequency carrier with the best frequency stability. The frequency multiplier stage then increase the frequency of the carrier to its required value.

Buffer Amplifier-

A buffer amplifier is one that provides electrical impedance transformation from one circuit to another and helps to prevent the signal source from being affected by whatever currents that the load may be produced with. The signal is buffered from load currents.

It first matches the output impedance of the carrier oscillator with the input impedance of the frequency multiplier. It then isolates the carrier oscillator and frequency multiplier. This is required so that the multiplier does not draw a large current from the carrier oscillator. If this occurs the frequency of the carrier oscillator will not remain stable.

Frequency Multiplier-

Frequency multiplier is an electronic circuit that generates an output signal whose output frequency is a multiple of its input frequency. The submultiple frequency of the carrier signal generated by the carrier oscillator is applied to the frequency multiplier through the buffer amplifier. This stage is also called harmonic generator.

The frequency multiplier generates high carrier frequency. The frequency multiplier is a tuned circuit that can be tuned to the required carrier frequency that is to be transmitted.

Class-C tuned power amplifier-

A class-C power amplifier gives high power current pulses of the carrier signal at its output.

Modulated Amplifier (Class-C)-

The modulating audio signal and the carrier signal after power amplification are applied to this modulating stage. The modulation takes place at this stage. This signal finally passed to the antenna.

Antenna-

The output of the modulated class-C power amplifier feeds the signal to the transmitting antenna. To transfer maximum power from the output stage to the antenna it is necessary that the impedance of the two sections match. For this a matching network is required. The matching network consists of L & C components.

4.4 CONCEPT OF FREQUENCY CONVERSION, RF AMPLIFIER & IF AMPLIFIER, TUNING, S/N RATIO-

CONCEPT OF FREQUENCY CONVERSION-

It is the process of converting the carrier frequency of a received signal from its original value to the intermediate frequency value in a super heterodyne receiver.

In radio reception using the super heterodyne principle the incoming signal is changed in frequency by converter stage of the receiver to a new and lower frequency known as the intermediate frequency.

RF AMPLIFIER-

The antenna not only provides very low amplitude input signals but it picks up all available transmissions at the same time.

The receiver circuits generate noise signals, which are added to the wanted signals. We hear this as a 'background hiss' and are particularly noticeable if the receiver is tuned between stations or if a weak station is being received. The RF amplifier is the first stage of amplification. It has to amplify the incoming signal above the level of the internally generated noise and also to start the process of selecting the wanted station and rejecting the unwanted ones.

IF AMPLIFIER-

The IF Amplifier consists of two stages of amplification and provides the main signal amplification and selectivity. Operating at a fixed IF frequency means that the design of the amplifiers can be simplified. If it were not for the fixed frequency, all the amplifiers may need to be tunable across the whole range of incoming RF frequencies and it would be difficult to arrange for all the amplifiers to keep in step as they are re-tuned.

The radio must select the wanted transmission and reject all the others. To do this the band pass of all the stages must carefully controlled. Each IF stage does not necessarily have the same band pass characteristics. The overall response is important. Again, this is something which is much more easily achieved without the added complication of making them tunable. At the final output from the IF amplifiers, we have a 455 KHz wave which is amplitude modulated by the wanted audio information. The selectivity of the IF amplifiers has removed the unwanted components generated by the mixing process.

TUNING-

In super heterodyne receiver the front end RF tuning circuit is required to remove the image signal. LC tuned circuit is used. The tuning of this tracks that of the local oscillator, so that frequency of both sections change simultaneously.

Now a days the tuning is normally carried out using varactor diodes that are driven by a voltage that is programmed from the microprocessor that controls the operation of the radio. This controls the frequency synthesizer used as the local oscillator.

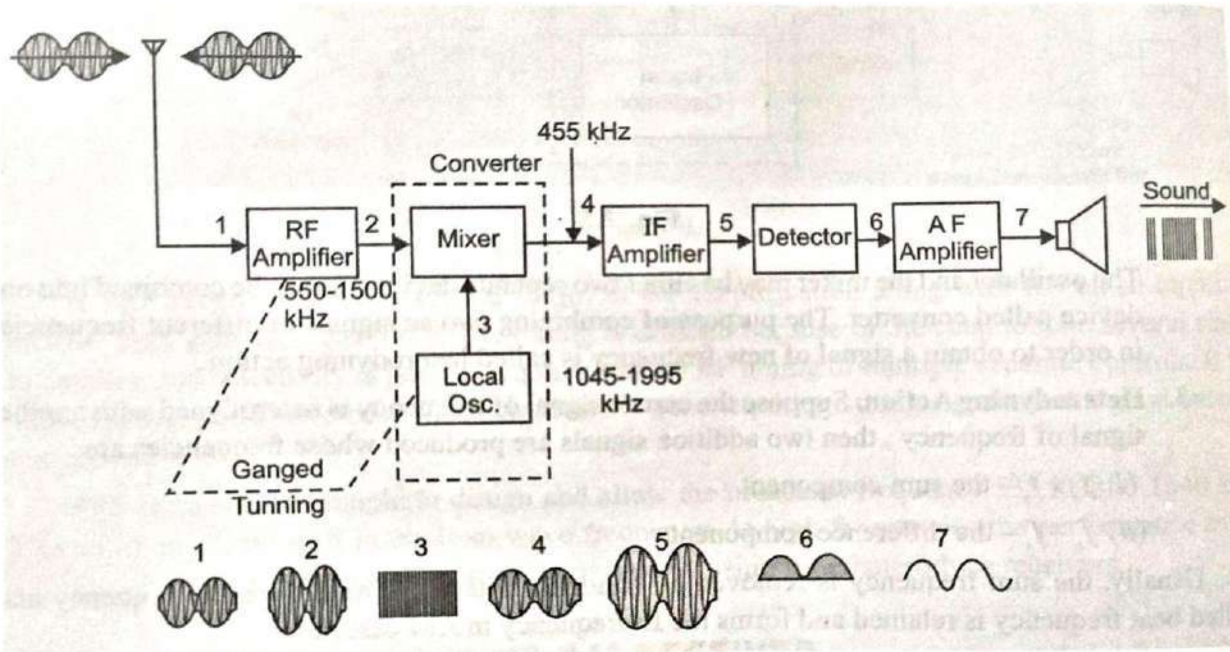
S/N RATIO-

4.5 SUPER HETERODYNE RECEIVER-

It is difficult to design amplifiers which give uniform high gain over a wide range of radio frequencies. However it is possible to design amplifiers which can provide high gain and uniform amplification over a narrow band of lower frequencies called intermediate frequencies.

Hence it is necessary to convert the modulated RF signal into modulated IF signal by using a frequency converter. For this we use super heterodyne receiver.

The word heterodyne stands for mixing. Here we have mixed the incoming signal frequency with the local oscillator frequency. Therefore this receiver is called super heterodyne receiver



RF Amplifier-

RF low noise amplifiers are designed to increase the desired RF signal amplitude without adding distortion or noise.

Mixer-

Mixer circuit is designed to combine two radio frequencies. It has two input one from RF amplifier and other from local oscillator.

When mixer combined the two signal we get, $f_o + f_s$ $(Sum) and$ $f_o - f_s$ $(Difference)$. The sum frequency is removed by band pass filtering. The difference frequency given to the input of IF amplifier.

The difference frequency is always maintained at 455KHz.

Local Oscillator-

The local oscillator is an RF oscillator whose frequency of oscillation can be controlled by varying the capacitance of its capacitor. The frequency of local oscillator always maintained higher than incoming signal.

IF Amplifier-

The 455 KHz output of the mixer is then passed on to IF amplifier. The IF amplifier amplify the signal and the output of IF is demodulated by a detector which provides the audio signal.

A.F Amplifier-

The audio signal is amplified by the audio frequency amplifier whose output is fed to a loud speaker which reproduces the original signal.

Ganged Tuning-

If the incoming signal changes then the local oscillator frequency also changes to maintain the difference frequency. For this purpose the tuning capacitor of the oscillator ganged with the capacitor of the input circuit i.e, RF amplifier. So that the difference in the frequency of the incoming signal and oscillator frequency is always constant i.e, 455 KHz.

Example- if the f_s= 1500KHz

f_o = 1955 KHz

Then $f_o - f_s$= 455 KHz

Or if the f_s= 1345 KHz

f_o = 1800 KHz

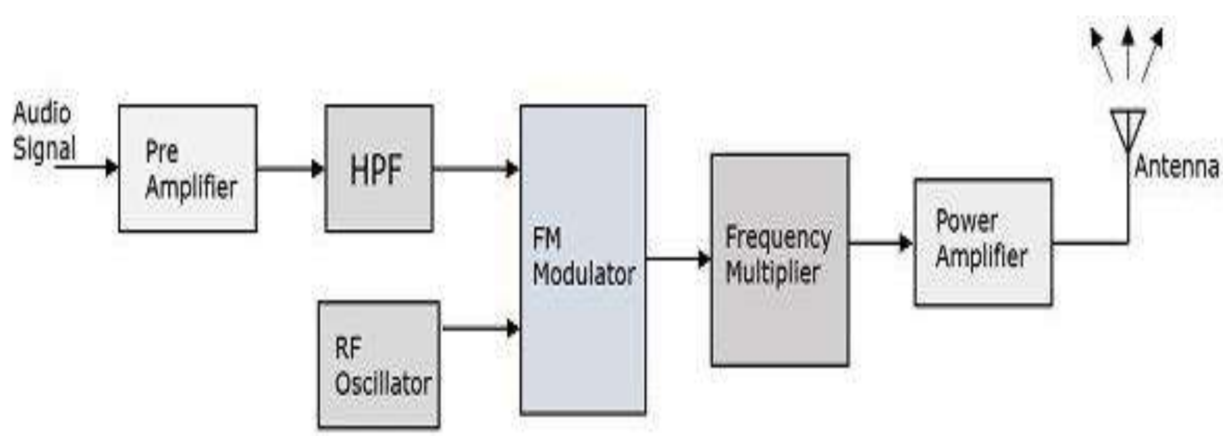
Then $f_o - f_s$= 455 KHz

4.6 FM TRANSMITTER & RECEIVER

FM TRANSMITTER-

The FM transmitter is a low power transmitter and it uses FM waves for transmitting the sound.

FM transmitter circuit can produce the radio frequency waves which are transmitted through the antenna.



Audio Signal-

Here a microphone is used as a source of an audio signal. The microphone is a transducer which can convert the sound energy into an electrical energy.

Pre Amplifier-

The audio pre amplifier is used to amplify the audio signal coming from the microphone.

HFA-

The output of Pre Amplifier is then applied to HPF which acts as a pre emphasis network. Pre emphasis increases the magnitude of the higher signal frequency, thereby improving the signal to noise ratio.

FM Modulator-

The modulator circuit is the main part of an FM transmitter circuit. It converts the audio signal which is to be transmitted. The modulator circuit takes two signal as input, one is the signal coming from the high pass filter and another is carrier signal from RF oscillator.

The modulator circuit modulates the RF signal according to the audio signal and produces the modulated RF signal as the output.

RF Oscillator-

An RF oscillator produces signals in the radio frequency range of about 100khz to 100Ghz.

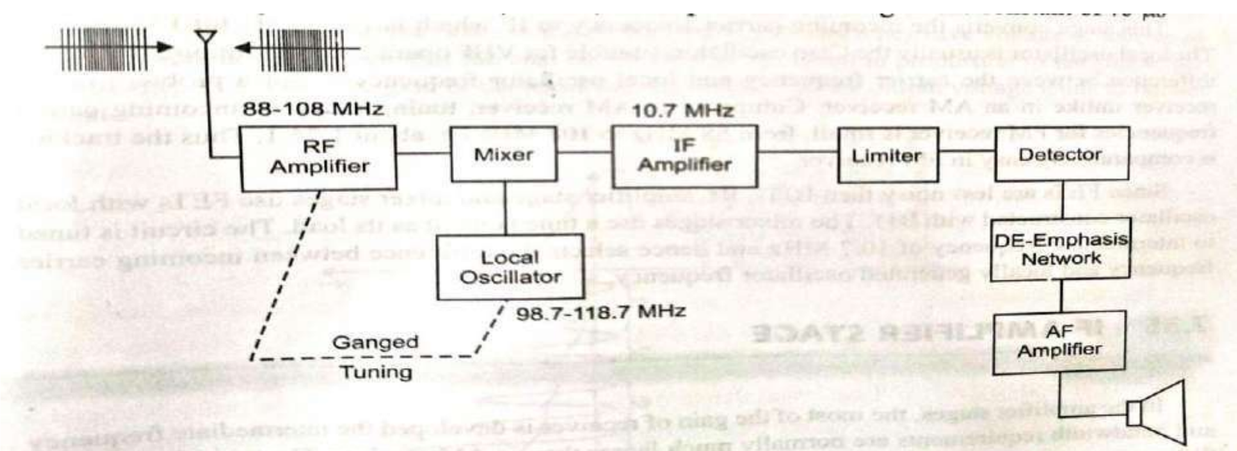
Frequency Multiplier-

Several stages of frequency multiplier are used to increase the operating frequency. Even then, the power of the signal is not enough to transmit. The RF amplifier circuit amplifies the signal coming from frequency multiplier.

RF amplifier-

RF amplifier is used to amplify the signal which can be easily transmitted for a long distance. Then the amplified radio signal is fed to the antenna for the transmission.

4.6.2 FM RECEIVER-



RF amplifier-

The RF amplifier amplifies the received signal intercepted by the antenna. The amplifier is a low noise amplifier.

Mixer-

The amplified signal is then applied to the mixer stage. The second input of the mixer comes from the local oscillator.

IF Amplifier-

The output of the mixer is a difference signal of incoming signal and carrier signal i.e, known as intermediate frequency of 10.7MHz. This signal is then amplified by IF amplifier.

Limiter-

The output of the IF amplifier is applied to the limiter circuit. The limiter removes the noise in the received signal and gives a constant amplitude signal.

This is very important in FM receivers because at amplitude variation in the received signal will result in unfaithful reproduction of the audio signals. Limiter is a sort of clipping circuit.

Detector-

The output of the limiter is now applied to the detector, which recovers the modulating signal. However, this signal is still not the original modulating signal. Before applying it to the audio amplifier stage, it is applied to de-emphasized network.

The de-emphasis network reduces the amplitude of high frequencies in the audio signal to bring them back to their original amplitude as these are increased by the pre-emphasis network at the transmitting station.

AF Amplifier-

The output of the de-emphasized stage is the audio signal, which is then applied to the audio amplifier stages and finally to the speaker.

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

5.1 CONCEPT OF SAMPLING THEOREM, NYQUIST RATE & ALISING

SAMPLING THEOREM

A continuous time signal is first converted to discrete-time signal by sampling process. The sufficient number of samples of the signal must be taken so that the original signal represented in its samples completely. Also, it should be possible to recover or reconstruct the original signal completely from its samples. The number of samples to be taken depends on maximum signal frequency present in the signal.

The statement of sampling theorem can be given in two parts as:

(i) A band-limited signal of finite energy, which has no frequency component higher than f_m Hz, is completely described by its sample values at uniform intervals less than or equal to $1/2f_m$ second apart.

(ii) A band-limited signal of finite energy, which has no frequency components higher than f_m Hz, may be completely recovered from the knowledge of its samples taken at the rate of $2f_m$ samples per second.

Combining the two parts, the sampling theorem may be stated as under:

"A continuous time signal may be completely represented in its samples and recovered back if the sampling frequency is f_s ^$\geq 2f_m$. Here f_s is the sampling frequency and fm is the maximum frequency present in the signal".

NYQUIST RATE

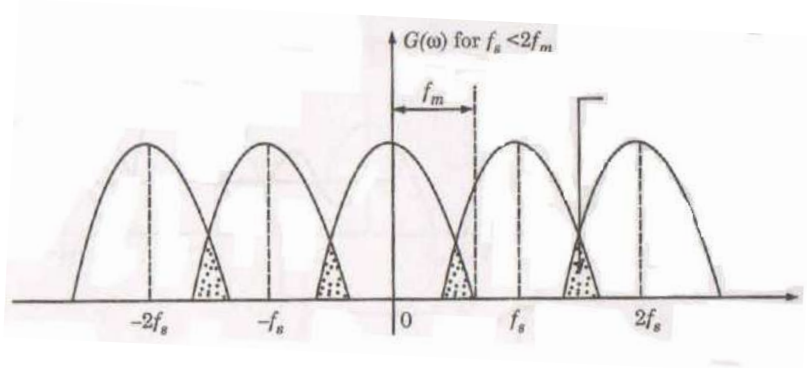
When the sampling rate becomes exactly equal to $2f_m$ samples per sec, then it is called Nyquist rate. Nyquist rate is also called the minimum sampling rate.

It is given by $f_s = 2f_m$

Nyquist Interval $T_s = ½ f_m$ sec

ALIASING:

When a continuous time band limited signal is sampled at a rate lower than Nyquist rate $f_s < 2f_m$, then the cycles of the spectrum G(w) of the sampled signal g(t) overlap with each other.



Spectrum of the sampled signal for the case $f_s < 2f_m$

5.2 SAMPLING TECHNIQUES

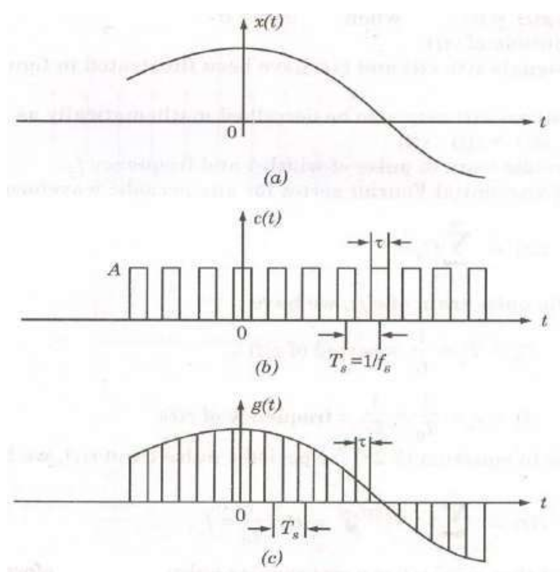
CLASSIFY SAMPLING

There are 3 types of sampling techniques.

(i) Instantaneous sampling
(ii) Natural sampling
(iii) Flat top sampling

- **Natural sampling:**

In this sampling the top of the pulses are curved according to the modulating signal.

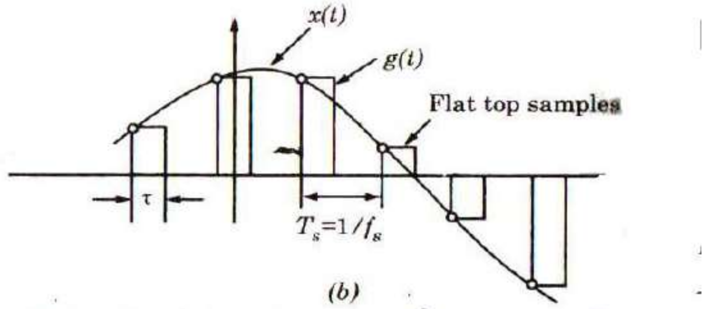


(a) Continuous time signal x(t).
(b) Sampling function waveform i.e. periodic pulse train
(c) Naturally sampled signal waveform g(t).

Natural Sampling

- **Flat- top Sampling:**

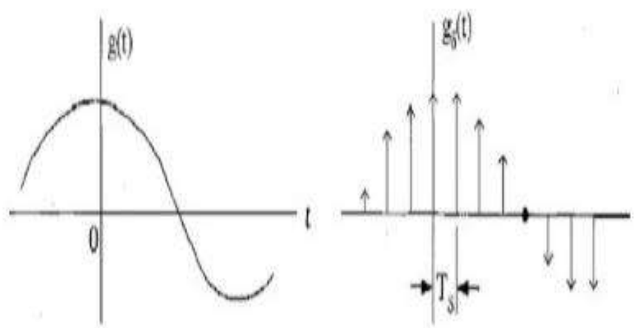
  In this sampling the top of the pulses are flat.



Flat-top sampling

- **instantaneous sampling:**

  In this sampling the modulating signal is multiplied with samples of unit strength. This form of modulation is known as impulse modulation. The main disadvantage of this modulation is very difficult to generate.

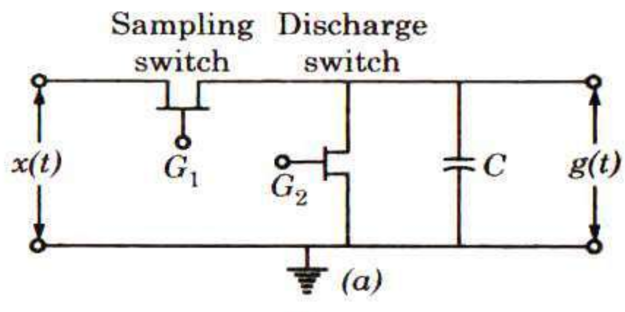


Instantaneous sampling

<u>5.3 ANALOG PULSE MODULATION</u>

<u>GENERATION OF PAM</u>-

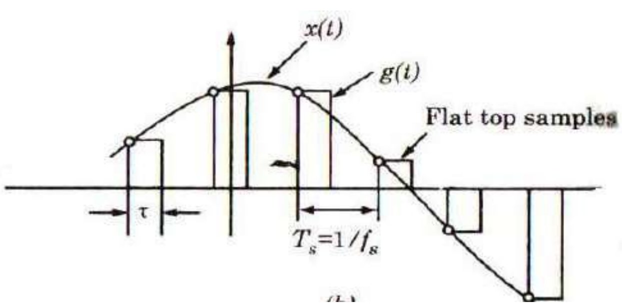
Pulse amplitude modulation may be defined as that type of modulation in which the amplitudes of regularly spaced rectangular pulses vary according to instantaneous value of the modulating signal. The pulses in a PAM signal may be of flat top type or natural type or ideal type. Out of these three pulse amplitude modulation methods, the flat top PAM is most popular and is widely used. The reason for using flat top PAM is that during the transmission, the noise interferes with the top of the transmitted pulses and this noise can be easily removed if the PAM pulses has flat top.

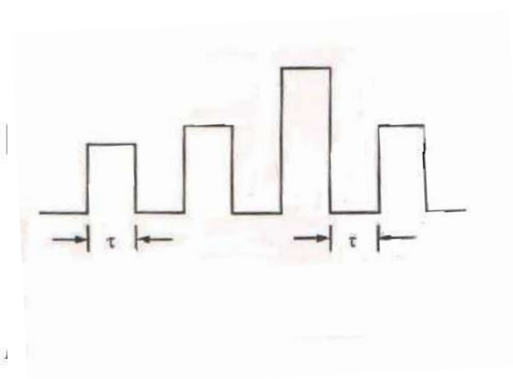


Sample and hold circuit generation Flat top sampled PAM

A sample and hold circuit is used to produce flat top sampled PAM. The working principle of this circuit is quite easy. The sample and hold circuit consists of two field effect transistor switches and a capacitor. The sampling switch is closed for a short duration by a short pulse applied to the gate G1 of the transistor. During this period, the capacitor C is quickly charge up to a voltage equal to the instantaneous sample value of the incoming signal x(t). Now the sampling switch is opened and the capacitor C holds the charge. The discharge switch is then closed by a pulse applied to gate G2 of the other transistor. Due to this, the capacitor ‘C’ is discharged to zero volts. The discharge switch is then opened and thus capacitor has no voltage.

Hence, the output of the sample and hold circuit consists of a sequence of flat top samples.



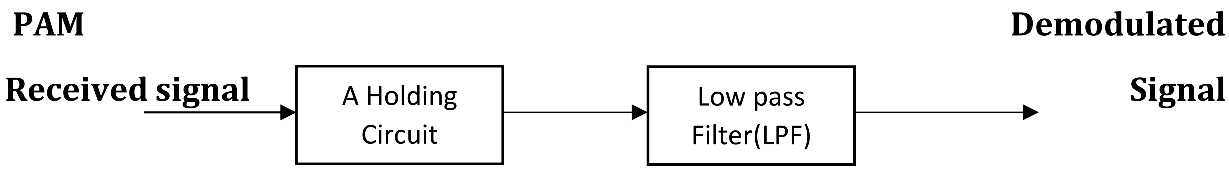


(a) (b)

(a) & (b) Illustration of maximum frequency PAM Signal

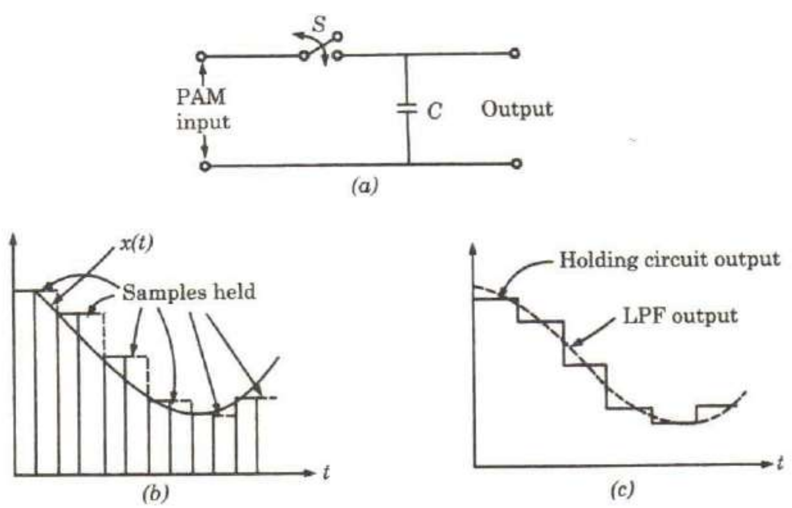
DETECTION OF PAM-

Demodulation is the reverse process of modulation in which the modulating signal is recovered back from a modulated signal. For pulse amplitude modulated signals, the demodulation is done using a holding circuit.



A Block diagram of PAM Demodulator

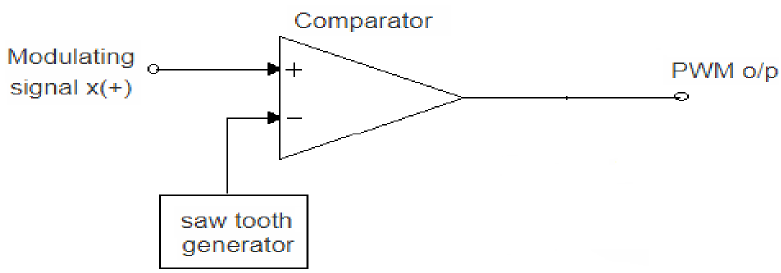
In this method, the received PAM signal is allowed to pass through a holding circuit and a low pass filter. In the holding circuit the switch s is closed after the arrival of the pulse and it is opened at the end of the pulses. In this way, the capacitor C is charged to the pulse amplitude value and it holds this value during the interval between the two pulses. After this the holding circuit output is smoothened in low pass filter. It may be observed that some kind of distortion is introduced due to the holding circuit. Here we use a zero order holding circuit. This zero order holding circuit considers only the previous sample to decide the value between the two pulses.



(a) A zero-order holding circuit (b) the output of holding circuit (c) the output of a Low Pass filter (LPF)

PAM signal generator generating modulating Signal

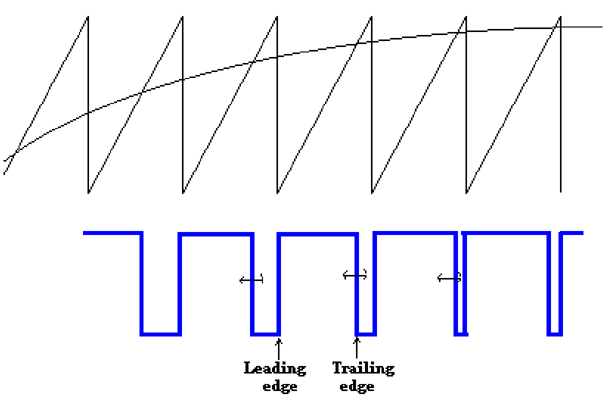
GENERATION OF PWM-

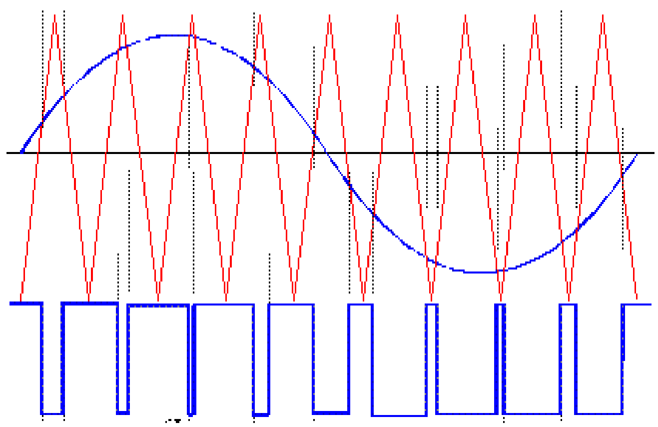


The modulating signal x(t) is applied to the non-inverting input of the comparator. The saw tooth generator generates the saw tooth signal. The saw tooth signal also known as sampling signal applied to the inverting input of comparator.

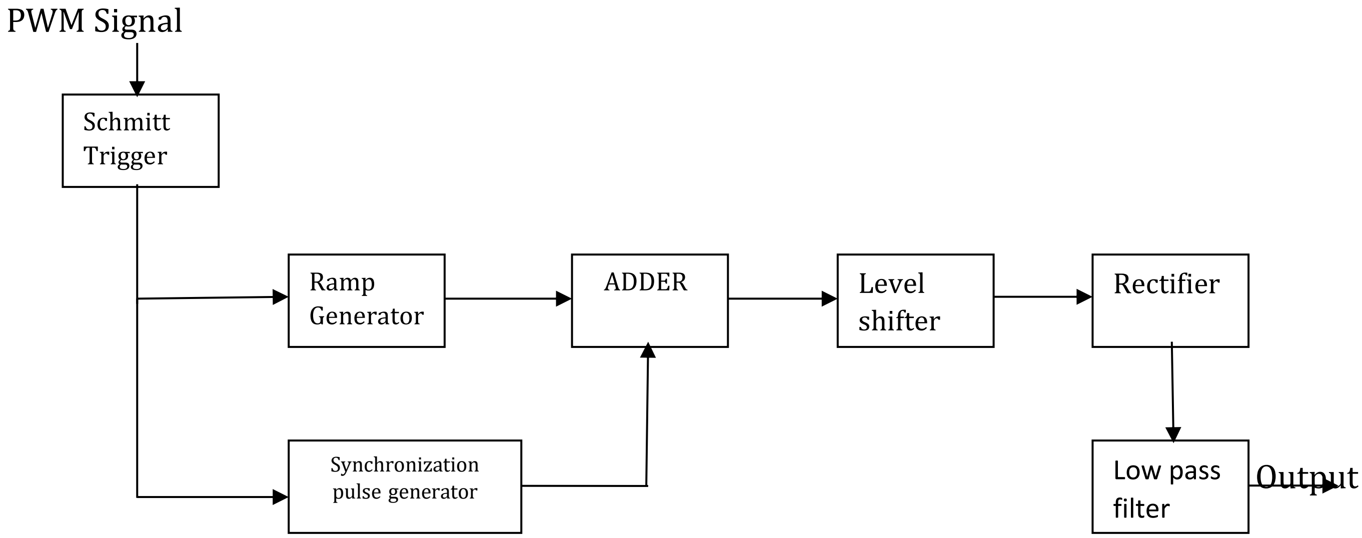
The output of the comparator is high only when instantaneous value of x(t) is higher than that of saw tooth waveform. Hence, the leading edge of PDM signal will be fixed and trailing edge will be modulated. When saw tooth waveform voltage is greater than voltage of x(t) at that instant, the output of comparator remains zero.

If the saw tooth waveform is reversed, then trailing edge will be fixed and leading edge will be modulated. If saw tooth waveform is replaced by a triangular waveform then both leading and trailing edges will be modulated. The pulse duration modulation or pulse width modulation signal is nothing but output of the comparator.





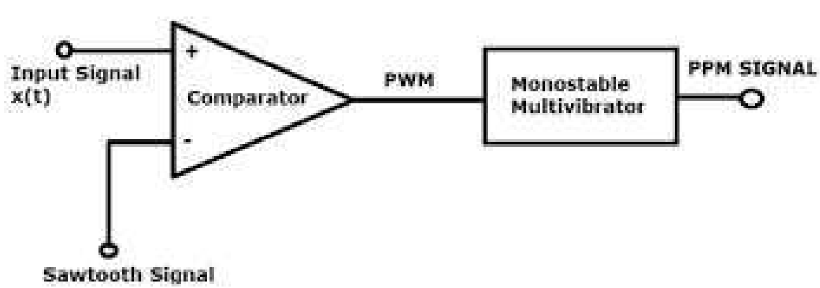
DEMODULATION OF PWM-



PWM Detector

The received PWM signal is applied to the Schmitt trigger circuit. This Schmitt trigger circuit removes the noise in the PWM waveform. The regenerated PWM is then applied to the ramp generator and the synchronization pulse detector. The ramp generator produces ramps for the duration of pulses such that height of ramps are proportional to the widths of PWM pulses. The maximum ramp voltage is retained till the next pulse. On the other hand, synchronous pulse detector produces reference pulses with constant amplitude and pulse width. These pulses are delayed by specific amount of delay. The delayed reference pulses and the output of ramp generator is added with the help of adder. The output of adder is given to the level shifter. Here negative offset shifts the waveform. Then the negative part of the waveform is clipped by rectifier. Finally, the output of rectifier is passed through low pass filter to recover the modulating signal.

GENERATION OF PPM:-



Generation of PPM Signal

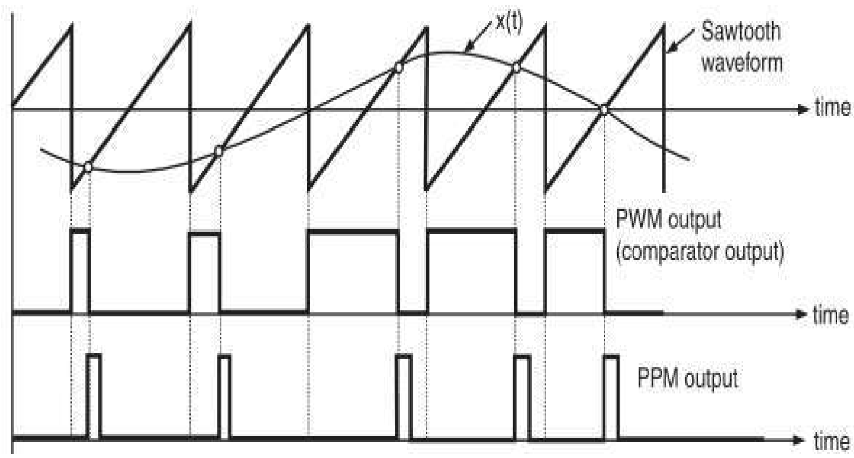
The modulating signal x(t) is applied to the non-inverting input of the comparator. The saw tooth generator generates the saw tooth signal of frequency f_s. The saw tooth signal is also known as sampling signal is applied to the inverting input of the comparator.

The output of the comparator is high only when instantaneous value of x(t) is higher than that of saw tooth waveform.

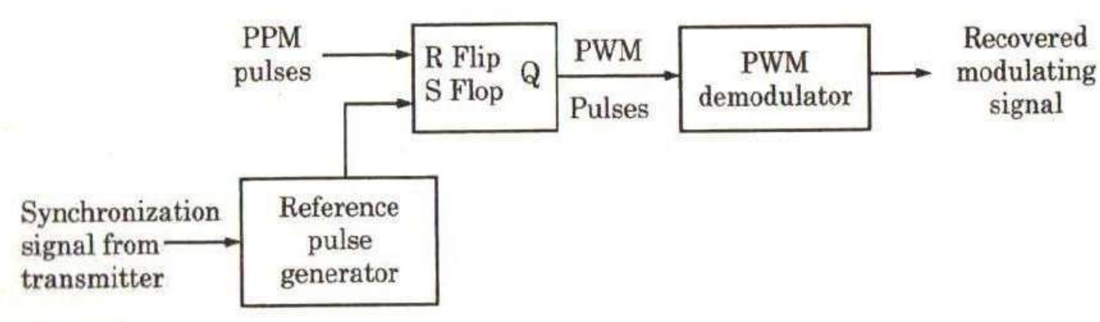
When saw tooth waveform voltage is greater than voltage of x(t) at that instant, the output of comparator remains zero. The pulse duration modulation signal is nothing but output of the comparator. To generate pulse position modulation, PDM signal is used as the trigger input to one monostable multivibrator.

The monostable output remains zero until it is triggered. The monostable is triggered on the falling edge of PDM. The output of monostable then switches to +ve saturation level 'A'. This voltage remains high for the fixed period then goes low.

The amplitude of all PPM and PDM pulses is same. Therefore nonlinear amplitude distortion as well as noise interference does not affect the detection at the receivers.



DETECTION OF PPM:-



PPM demodulator

Flip flop is set or turned ON when the reference pulse arrives. This reference pulse is generated by reference pulse generator of the receiver with the synchronization signal from the transmitter. The flip flop circuit is reset or turned

OFF at the leading edge of position modulated pulse. This repeats and we get PWM pulses at the output of the flip flop.

COMPARISON OF PAM, PWM AND PPM-

Sl. No.	Basis for Comparison	PAM	PWM	PPM
1	Varying parameter	Amplitude	Width	Position
2	Immunity towards noise	Low	High	High
3	Need of synchronization pulse	Not exist	Not exist	Exist
4	Signal to noise ratio	Low	Moderate	Comparatively high
5	Transmission power	Variable	Variable	Constant
6	Bandwidth dependency	On pulse width	On rise time of pulse	On rise time of pulse
7	Synchronization between Transmitter and Receiver	Not needed	Not needed	Needed
8	Similarity of implementation	Similar to AM	Similar to FM	Similar to PM
9	Bandwidth requirement	Low	High	High

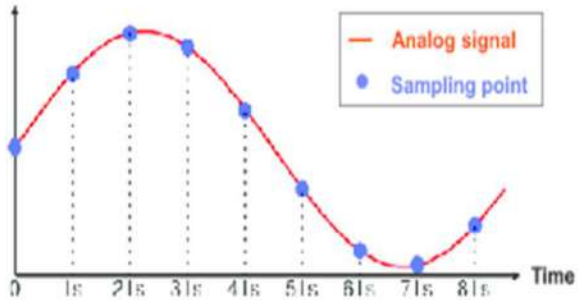
5.4 QUANTIZATION OF SIGNAL & QUANTIZATION ERROR-

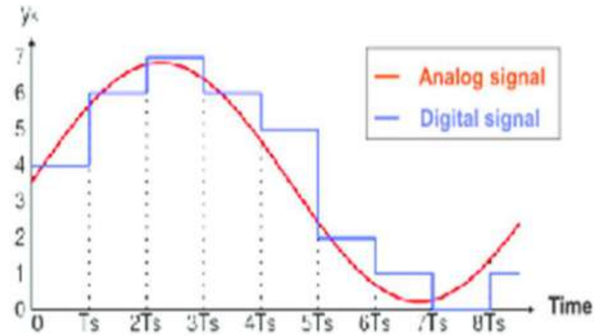
QUANTIZATION-

Quantization refers to the process of approximating the continuous set of values with a finite set of values. The input to a quantizer is the original data and the output is always one among a finite number of levels.

The analog and digital converters perform this type of function to create a series of digital values out of the analog signal. The quantizing of an analog signal is done by discretizing the signal with a number of quantization levels.

Samples taken are assigned numeric values that the digital circuit can use in a process called quantization. When a sample is quantized, the instantaneous value of its analog amplitude has to be rounded off to the nearest available digital value. This rounding off process is called approximation.





A quantizer can be specified by its input partitions and output levels. If the input range is divided into levels of equal spacing, then the quantizer is termed as a uniform quantizer and if not it is termed as a non-uniform quantizer.

A uniform quantizer can be specified easily by its lower bound and the step size. Also implementing a uniform quantizer is easier than a non-uniform quantizer. There are two types of uniform quantizer.

1) Mid rise type
2) Mid tread type

The **mid-rise type** is so called because the origin lies in the middle of a raising part of the stair case. The **mid tread type** is so called because the origin lies in the middle of a tread of stair case

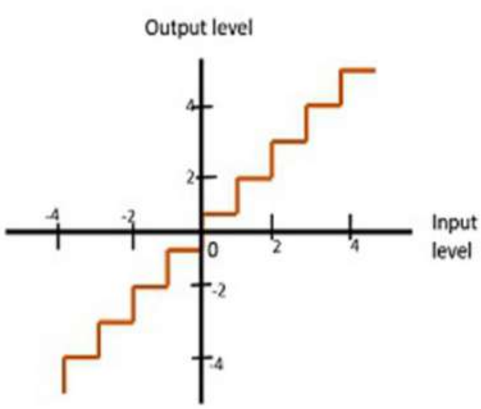


Fig 1 : Mid-Rise type Uniform Quantization

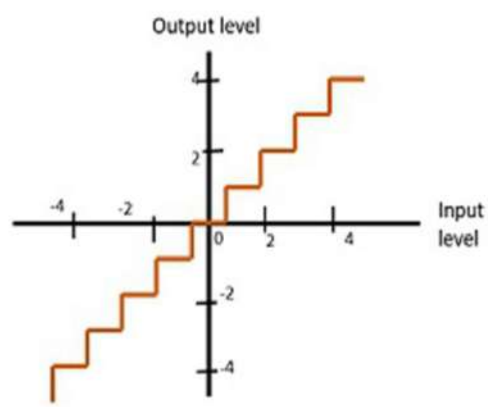


Fig 2 : Mid-Tread type Uniform Quantization

Quantization error-

For any system, during its functioning there is always a difference in the values of its input and output. The processing of the system results in an error, which is the difference of those values. The difference between an input value and its quantized value is called a quantization error.

Quantization Noise-

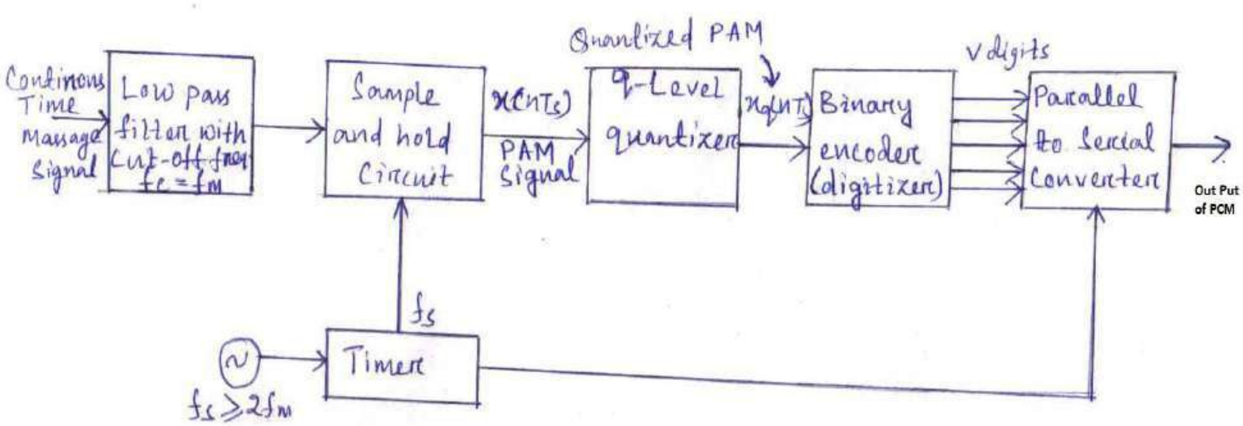
It is a type of quantization error, which usually occurs in an analog audio signal, while quantizing it to digital. For example, in music, the signals keep changing continuously where regularity is not found in errors. Such errors create a wideband noise called as quantization.

5.5 PULSE CODE MODULATION-

Pulse code modulation is known as a digital pulse modulation technique. The pulse code modulation is quite complex compared to the analog pulse modulation techniques.

A PCM system consists of 3 main parts i.e, transmitter, transmission path and receiver. The essential operations in the transmitter of a PCM system are sampling, quantizing and encoding. Sampling is the operation in which an analog signal is sampled according to the sampling theorem resulting in a discrete time signal. The quantizing and encoding operations are usually performed in same circuit which is known as an analog to digital converter. Also the essential operations in the receiver are regeneration of impaired signals, decoding and demodulation of the train of quantized samples.

PCM GENERATION TRANSMITTER-



In PCM generator the signal x(t) is first passed through the low pass filter of cut off frequency f_m Hz. This low pass filter blocks all the frequency components which are lying above f_m Hz.

This means the signal x(t) is band limited to f_m Hz. The sample and hold circuit then samples this signal at the rate of f_s. Sampling frequency f_s is selected sufficiently nyquist rate to avoid aliasing i.e,

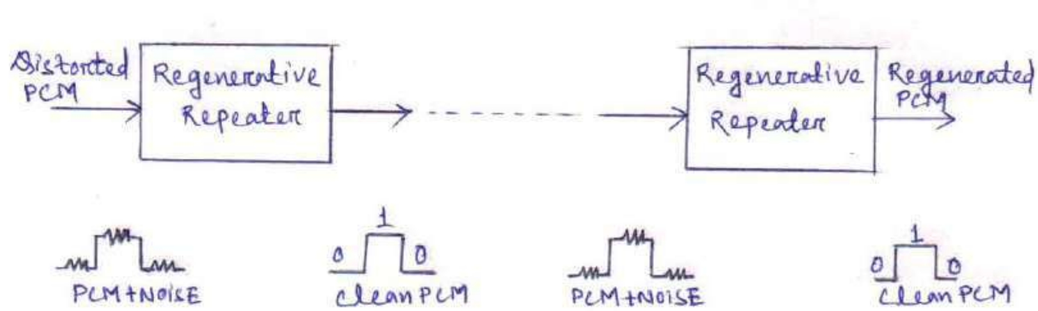
$$f_s \geq 2f_m$$

The output of sample and hold circuit is denoted by $x(nT_s)$. This signal $x(nT_s)$ is discrete in time and continuous in amplitude. A Q-level quantizer compares input $x(nT_s)$ with its fixed digital levels. It assigns any one of the digital level to $x(nT_s)$ with its fixed digital level. It then assigns any one of the digital level to $x(nT_s)$ which results in minimum distortion or error. This error is called quantization error. Thus output of quantizer is a digital level called $x_q(nT_s)$.

Now the quantized signal level $x_q(nT_s)$is given to binary encoder. This encoder converts input signal to 'v' digits binary word. Thus $x_q(nT_s)$ is converted to 'v' binary bits. This encoder is also known as digitizer.

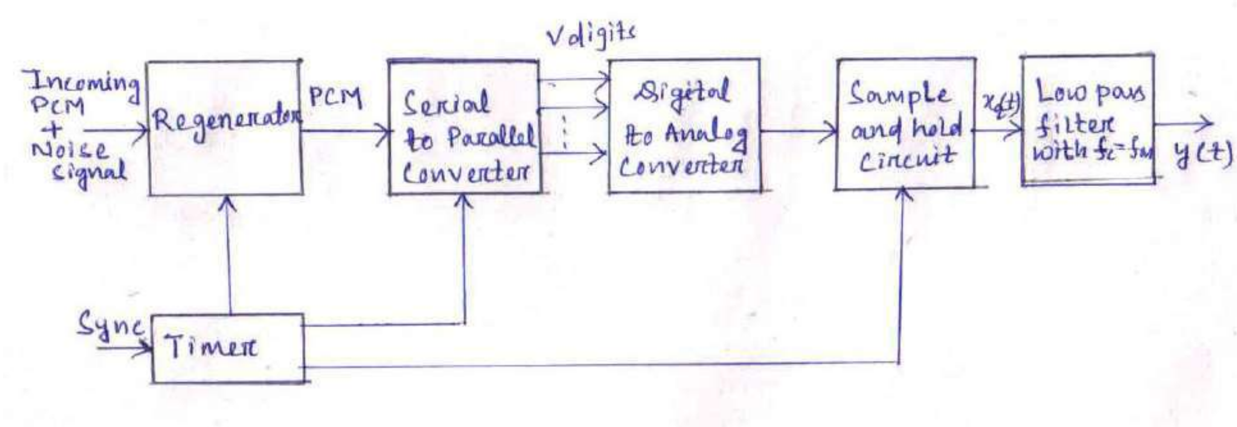
Also an oscillator generates the clocks for sample and hold circuit and parallel to serial converter. In the pulse code modulation generator, sample and hold, quantizer and encoder combinely form an anlog to digital converter.

<u>PCM TRANSMISSION PATH</u>-



The path between the PCM transmitter and PCM receiver over which the PCM signal travel, is called as PCM transmission path. The most important feature of PCM system lies in its ability to control the effects of distortion and noise when the PCM wave travels on the channel. PCM accomplishes this capacity by means of using a chain of regenerative repeaters. Such repeaters are spaced close enough to each other on the transmission path. The regenerative performs three basic operations namely equalization, timing and decision making. Hence each repeater actually reproduces the clean noise free PCM signal from the PCM signal distorted by the performance of PCM in presence of noise.

DEMODULATION OF PCM-



In the above diagram regenerator at the start of PCM receiver reshapes the pulse and removes the noise. This signal is then converted to parallel digital words for each sample.

Now, the digital word is converted to its analog value denoted as $x_q(t)$ with the help of a sample and hold circuit. This signal, at the output of sample and hold circuit, is allowed to pass through a low pass reconstruction filter to get the appropriate original massage signal denoted as y(t).

5.6 COMPANDING IN PCM & VOCODER

COMPANDING-

Companding refers to a technique for compressing and expanding an analog or digital signal. It is a combination of the words compressing and expanding. In this process a voice signal is compressed, then changed from analog to digital in the transmitter and in receiver converted back from digital to analog before it is expanded again.

Companding helps to improve SNR of weak signals. As we know in non-uniform quantization, the step size varies according to the signal level. If the signal level is low then step size will be small. So, the step size will be low for weak signal. Thus the quantization noise will also be low.

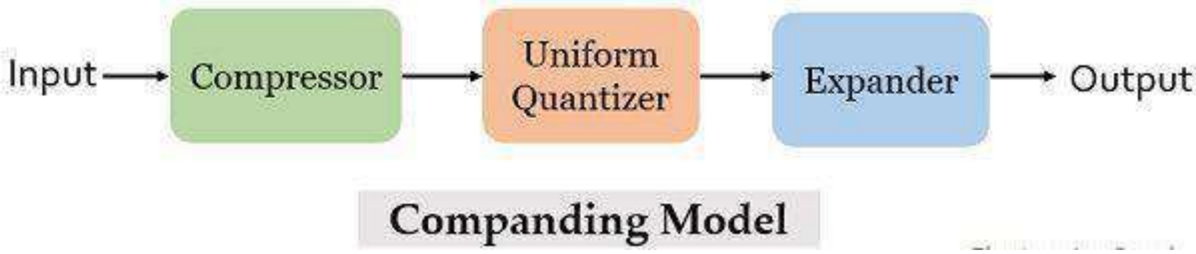
Quantization noise is given by:

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

For uniform quantization Δ is constant. So, for weak signal in uniform quantization the quantization noise increases. But in non-uniform quantization the step size deceases for weak signal. So, the quantization noise decreases. This will improve SNR.

Model of Companding-

This model consists of a compressor, a uniform quantizer and an expander.



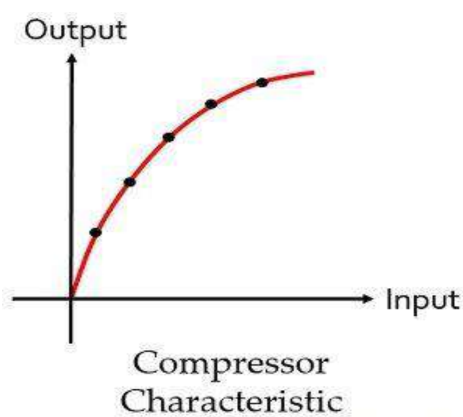
The signal is first given to the compressor. The compressor unit amplifies the weak signal in order to increase the signal level. While if the input signal is a strong signal then compressor attenuates that signal before providing it to the uniform quantizer.

This is done in order to have an appropriate signal level as the input to the uniform quantizer. High amplitude signal needs more bandwidth and also is more likely to distort. The output of the compressor is provided to uniform quantizer where the quantization of the applied signal is performed. At the receiver end the output of the uniform quantizer is fed to the expander.

It performs the reverse of the process executed by the compressor. This unit when receives a low value signal then it attenuates it, while if a strong signal is present then the expander amplifies it. This is done in order to get the originally transmitted signal at the output.

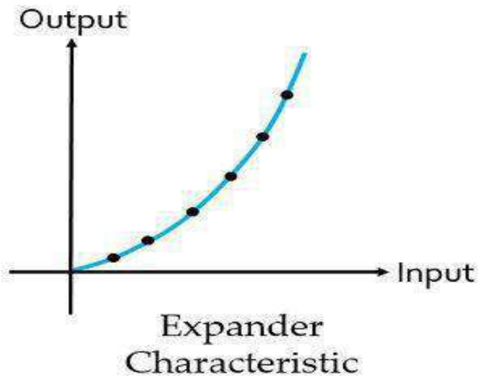
Compressor Characteristic-

The graph clearly represent that the compressor provides amplification to weak signal and attenuates the high input signal.

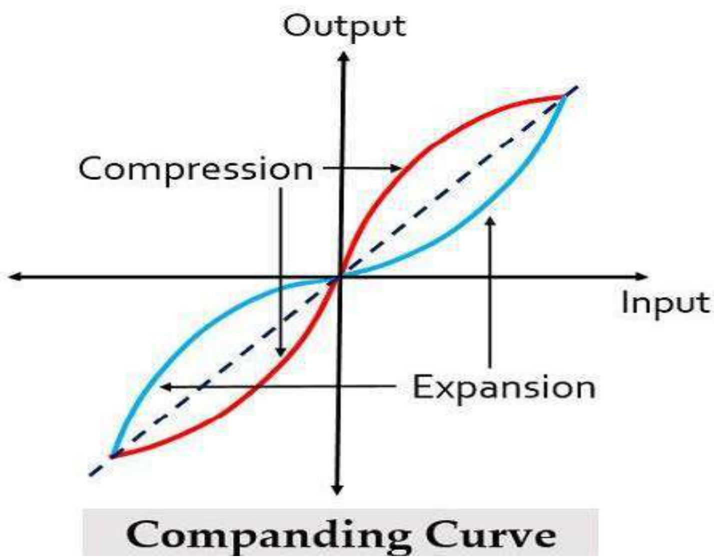


Expander Characteristic-

Expander performs reverse operation of the compander. So, it is clear from the figure that expander provides amplification to strong signal and attenuates the low input signal.



The below one is the compander characterteristics. This is the combined curve of compressor and expander. The dotted line represents the linear characteristic of the compander indicating that the originally transmitted signal is recovered at the receiver.



For digital audio signals, companding is used in pulse code modulation. The process involves decreasing the number of bits used to record the strongest signals. In the digital file format, companding improves the signal to noise ratio at reduced bit rates. For example, a 16 bit PCM signal may be converted to an eight bit signal.

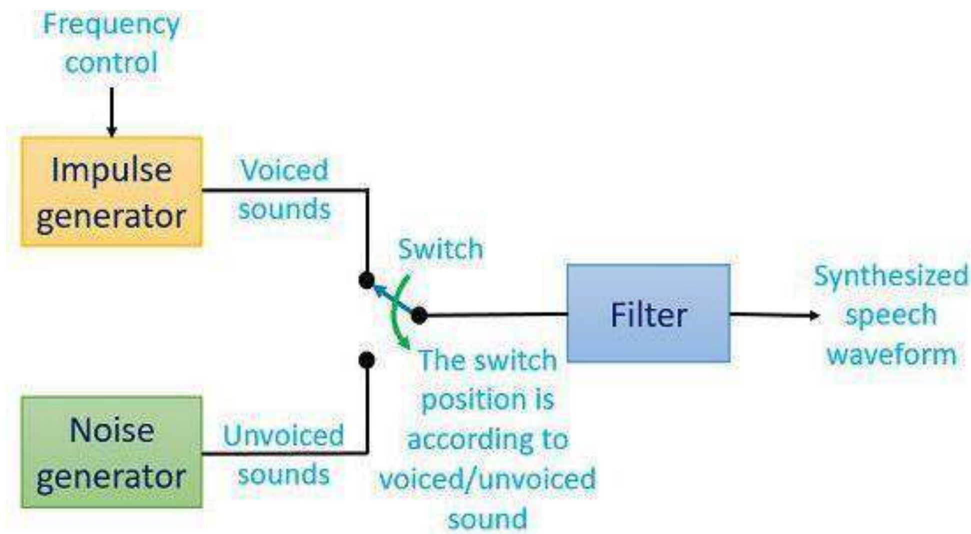
VOCODER-

Vocoder is an audio processor that is used to transmit speech or voice signal in the form of digital data. The full form of vocoder is voice coder. Vocoders are used for digital coding of speech and voice simulation.

Vocoder operates on the principle of formants. Formants are the meaningful components of a speech that is generated due to the human voice. Whenever a speech signal is transmitted, it is not needed to transmit the precise waveform. We can simply transmit the information by which one can reconstruct that particular waveform.

Vocoders are used for voice synthesis. The vocoder takes two signals and creates a third signal using the spectral information of the two input signals.

Speech model of vocoder-



A voice model is used to stimulate voice. As speech contains a sequence of voiced and unvoiced sounds. Voice sounds are the sounds generated by vibrations of the vocal cords. Unvoiced sounds are generated by expelling air through lips and teeth.

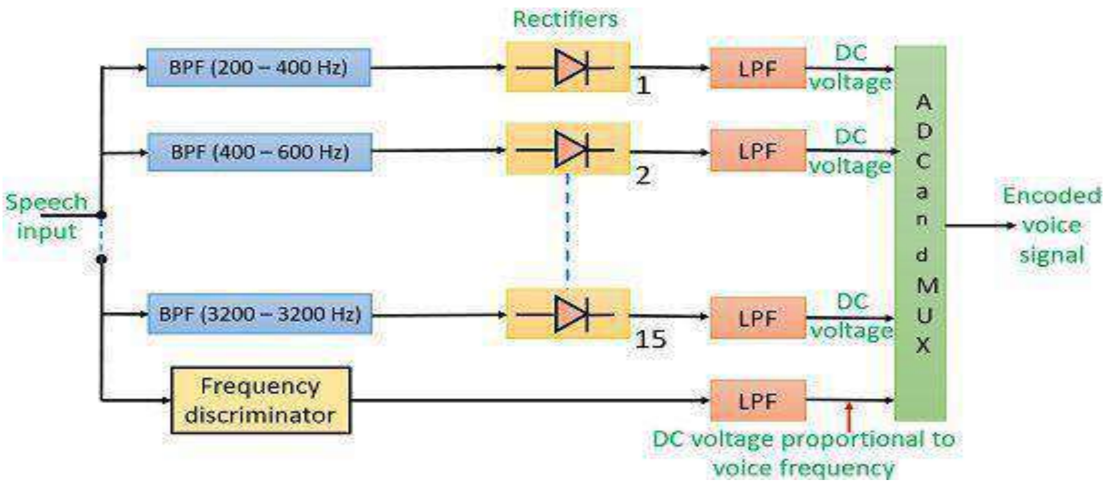
Voiced sounds are simulated by the impulse generator, the frequency of which is equal to the fundamental frequency of vocal cords. The noise source present in the circuit is used to simulate the unvoiced sounds. The position of the switch helps in determining whether the sound is voiced or unvoiced. Then the selected signal is passed through a filter that simulates the effect of mouth, throat and nasal passage of speaker.

The filter unit then filters the input in such a way so as the required letter is pronounced. LPC is extensively used in case of speech and music application. Full form of LPC is Linear Predictive coding. This technique is used to predict the estimate future values. So, we can say, by analyzing two previous samples, it predicts the outcome. Vocoder is comprised of voice encoder and decoder.

Voice Encoder-

The frequency spectrum of the speech signal is divided into 15 frequency ranges by using 15 band pass filter each having bandwidth range of 200Hz. The output of BPF acts as input for the rectifier unit. The signal is rectified and filtered so as to produce a dc voltage. This generated dc voltage is proportional to the amplitude of AC signal present at the output of the filter.

The input of the frequency discriminator is the speech signal. Frequency discriminator unit is followed by a low pass filter of 20 Hz. This low pass filter generates a dc voltage proportional to the voice frequency. The output at all the LPF's is DC voltage which is sampled, multiplexed and A/D converted. So, we have a digital equivalent of the speech signal at the output of the encoder. This encoded voice signal consists of frequency component from 200 Hz to 3200 Hz.

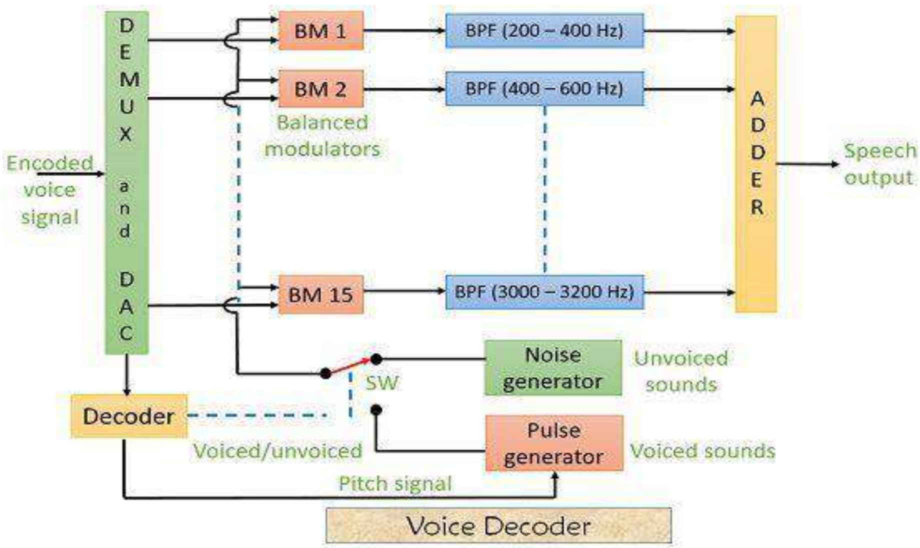


Voice Decoder-

The de-multiplexed and DAC section convert the received encoded signal back to its analog form. Here a balanced modulator- filter combination is used in correspondence to rectifier- filter combination at the encoder. The carrier to this balanced modulator is either the output of noise generator or pulse generator.

But this depends on the position of the switch. The switch position is decided by the decoder because when the voiced signal is received, the switch connects the pulse generator output to the input of all the balanced modulator. When an unvoiced signal is received the switch connects noise generator output to the input of all the balanced modulator.

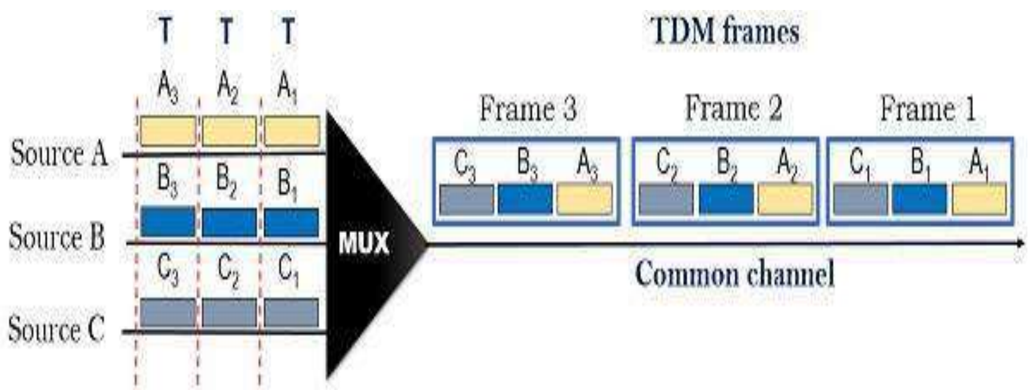
The adder will thus add up all the analog signal and produce voice or speech output. Speech transmission using vocoder is helpful technique but it has a disadvantage. This is because it leads to degradation in speech quality.



5.7 TDM-

Multiplexing is a technique in which multiple data signals can be transmitted over a single communication channel. In which multiplexing technique data signals are transmitted over a single channel in different timeslots is known as Time Division Multiplexing.

One may need to differentiate between the various signals for proper data transmission. So, in TDM the complete signal gets transmitted by occupying different time slots.

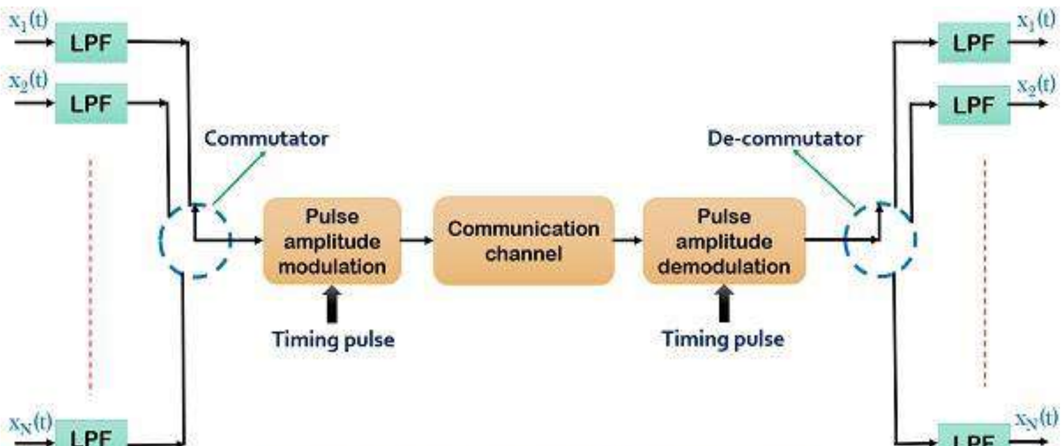


In the above diagram source A, B and C wants to transmit data through a common medium. Thus the signal from the 3 sources is divided into multiple segments each having their fixed time slots.

In this diagram we can see each signal divided into 3 segments. So by taking one segment from each source the multiplexer forms one frame. So, here 3 frames are formed. As these segments are entirely different from each other thus the chances of unnecessary signal mixing can be eliminated. When a frame gets transmitted over the particular time slot, the next frame uses the same channel to get transmitted and the process is repeated until the completion of the transmission.

Here, we have taken the example of 3 different sources, but one can perform multiplexing of n source signals. Both analog and digital signals can be multiplexed using TDM, but its processing technique allows the multiplexing of digital signals.

TDM System-



Block diagram of PAM/TDM system

This is a diagram of TDM system using PAM technique. Here at the beginning, the system consists of multiple LPF depending on the number of data inputs. These LPF are basically anti-aliasing filters that eliminate the aliasing of the data input signal.

The output of the LPF is then fed to the commutator. As per the rotation of the commutator the samples of the data inputs are collected by it. Here f_s is the rate of rotation of the commutator. Commutator perform the function of multiplexing. After that modulation is done and transmitted.

At the receiving end, a de-commutator is placed that is synchronized with the commutator at the transmitting end. This de-commutator separates the time division multiplexed signal at the receiving end. The commutator and de-commutator must have same rotational speed in order to get proper de-multiplexed signal at the receiving end.

After that signals given to the LPF and original data is recovered. In TDM

$$f_s \geq 2f_m$$

Thus the time duration in between successive sample is given as,

$$T_s = {}^{1}/_{f_s}$$

We have considered that one can use this for ‘n’ input channels, then one sample is collected from each of the ‘n’ signals. Each frame will provide us with N samples and the spacing between the two is given as ${T_s}/{N}$.

Pulse frequency means the no. of pulses per second is also known as signaling rate denoted as ‘r’.

r= $1/_{Spacing\ between\ 2\ samples}$

$= \frac{1}{T_s/N}$

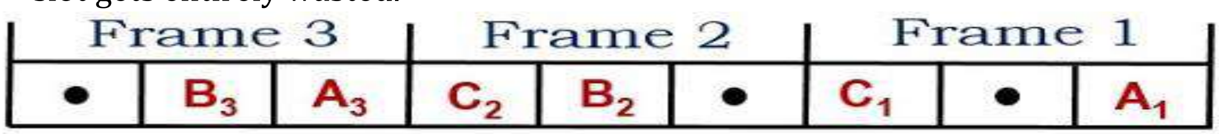
$= N/T_s = \frac{N}{1/f_s} = \mathrm{N}f_s$

The technique of time division multiplexing can be implemented in two ways.

1) Synchronous TDM
2) Asynchronous TDM

1) Synchronous TDM-

In this technique, the time slots are assigned at the beginning. This leads to the wastage of the capacity. As in the absence of any data unit, that particular time slot gets entirely wasted.



Synchronous TDM

2) Asynchronous TDM-

It eliminates the drawback of wastage of time slot present in synchronous TDM. Here a particular frame is transmitted by the transmitting end only when it gets completely filled by the data units. It exhibits higher efficiency than that of synchronous TDM. It allocates time slots as per demand

Application-

TDM mainly used in digital communication system i.e, in cellular radio and in satellite communication system.

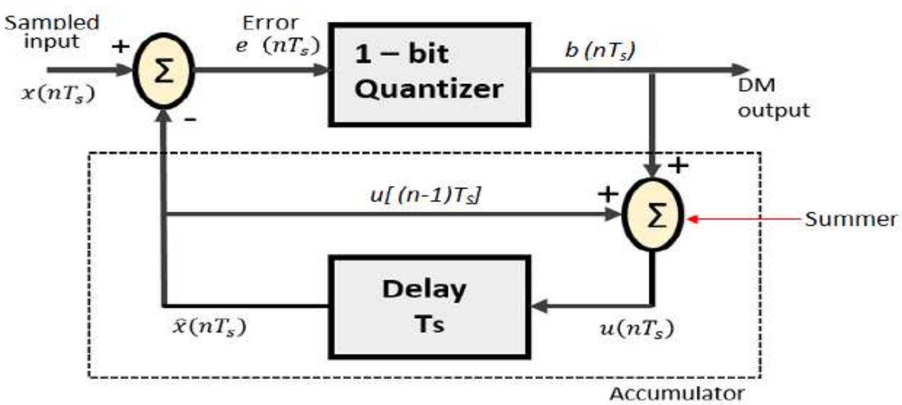
5.8 DELTA MODULATION-

The sampling rate of a signal should be higher than the Nyquist rate, to achieve better sampling. If this sampling interval in DPCM is reduced considerably, the sample to sample amplitude difference is 1 bit quantization, then the step size will be very small i.e, Δ.

The type of modulation, where the sampling rate is much higher and in which the step size after quantization is of a smaller value Δ, such a modulation is termed as delta modulation. Delta modulation is a simplified form of DPCM technique, also viewed as 1 bit DPCM scheme. As the sampling interval is reduced the signal correlation will be higher.

Transmitter-

- 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., +Δ and -Δ.
- Now, if the difference is positive, then approximated signal is increased by one step, i.e., 'Δ'. If the difference is negative, then approximated signal 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.



The error between the sampled value of x(t) and last approximated sample is given as e(nTs)= x(nTs)- x^(nTs).
e(nTs)= error at present sample
x(nTs)= sampled signal of x(t)
x^(nTs)= last sample approximation of the stair case waveform

if we assume u(nTs) as the present sample approximation of staircase output, then u(n-1)Ts= x^(nTs)

Depending on the sign of error e(nTs), the sign of step size Δ is decided. In other words, we can write

b(nTs)= +Δ if x(nTs)≥ x^(nTs)
-Δ if x(nTs)< x^(nTs)

Also if $b(nTs) = +\Delta$ then a binary '1' is transmitted and if $b(nTs) = -\Delta$ then a binary '0' is transmitted.

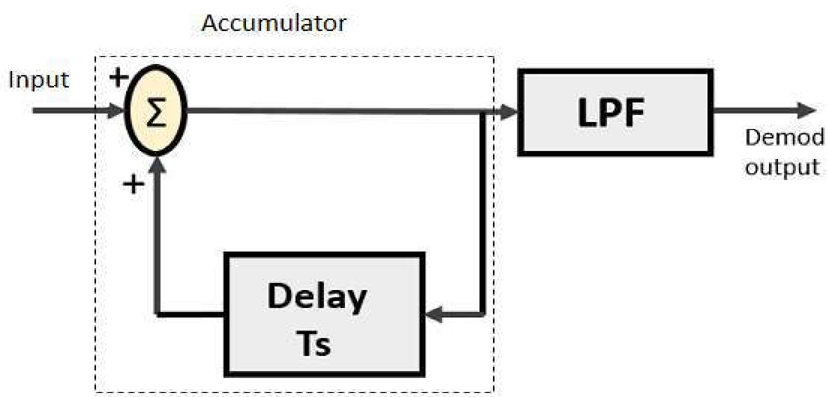
The summer in the accumulator adds quantizer output with the previous sample approximation. This gives present sample approximation i.e,

$$u(nTs) = u(nTs - Ts) + \pm\Delta$$

$u(nTs) = u(n-1)Ts + b(nTs)$

The previous sample approximation $u(n-1)Ts$ is restored by delaying one sample period Ts. The sampled input signal $x(nTs)$ and staircase approximation signal $\hat{x}(nTs)$ are subtracted to get error signal $e(nTs)$.

Receiver-



- The delta demodulator comprises of a low pass filter, a summer, and a delay circuit. The predictor circuit is eliminated here and hence no assumed input is given to the demodulator.
- 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 the input is binary '1' then it adds $+\Delta$ step to the previous output (which is delayed).
- If the input is binary '0' then one step 'Δ' is subtracted from the delayed signal.
- The low pass filter smoothens the staircase signal to reconstruct the original message signal x (t).

Advantages and disadvantages-

Advantages-

1) It transmits only one bit for one sample.
2) The transmitter and receiver implementation is very much simple.

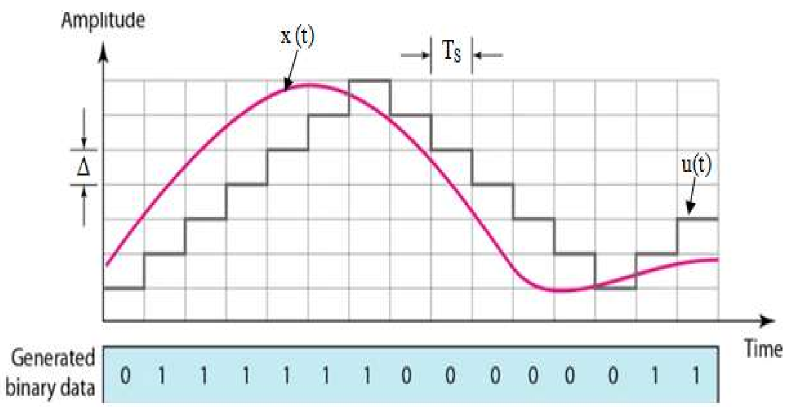
Disadvantages-

1) Slope overload distortion-

When 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). Here, there is a large error between the staircase approximated signal and the original input signal x(t). this error is known as slope overload distortion. To reduce this error, the step size must be increased when slope of signal x(t) is high.

2) Granular noise-

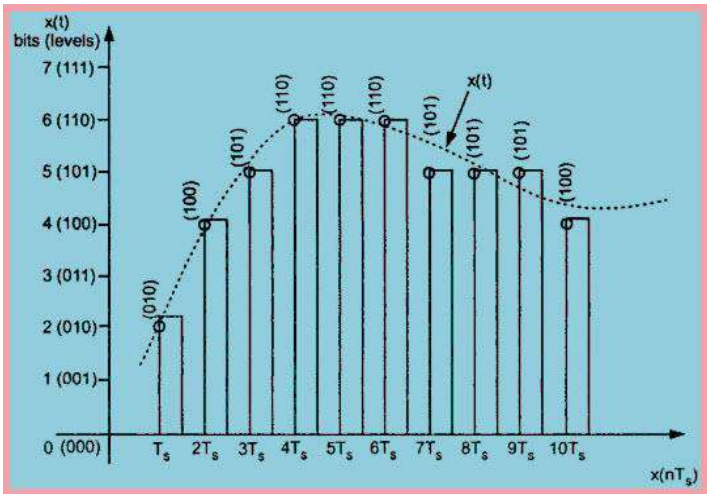
Granular noise occurs when the step size is too large compared to small variations in the input signal. This means that for very small variations in the input signal, the staircase signal is changed by large amount because of large step size. The error between the input and approximated signal is called granular noise. The solution to this problem is to make step size small.



5.9 GENERATION & DEMODULATION OF DPCM-

We can observe that samples of a signal are highly correlated with each other. This is due to the fact that any signal does not change fast means values from present sample to next sample does not vary by a large amount. The adjacent samples of the signal carry the same information or with a little difference.

When these samples are encoded by a PCM system, the resulting encoded signal contains some redundant information.



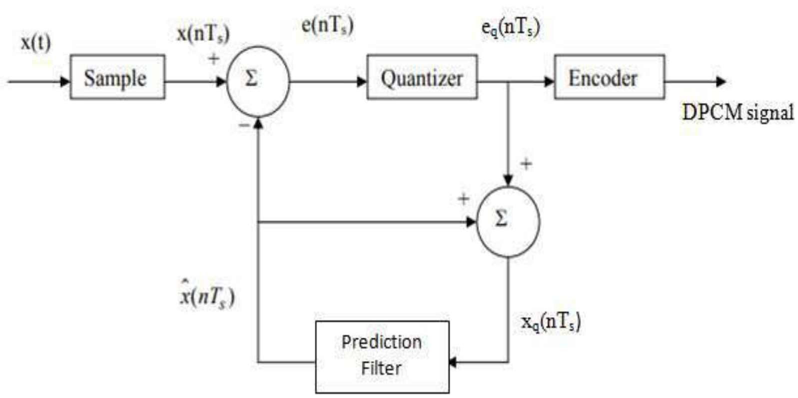
In this diagram dotted line represent the continuous time signal x(t). This signal is sampled by flat top sampling at intervals T_s, $2T_s$, $3T_s$ ……. nT_s. The sampling frequency is selected higher than nyquist rate. Here the samples are encoded by using 3 bit. The sample is quantized to the nearest digital level.

Here we can observe that the samples taken at $4T_s$, $5T_s$ *and* $6T_s$ *are* encoded to same value i.e, 110. This information can be carried only by one sample. But three samples are carrying the same information means that it is redundant.

If this redundancy is reduced, then overall bit rate will decrease. This type of digital pulse modulation is called as Differential Pulse Code Modulation (DPCM).

The DPCM 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 value.

DPCM Transmitter-



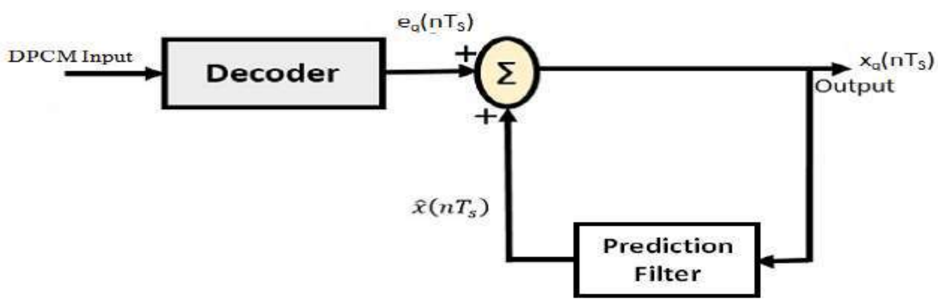
- The sampled signal is denoted by x(nTs) and the predicted signal is indicated by x^(nTs). The comparator finds out the difference between the actual sample value x(nTs) and the predicted value x^(nTs). This is called signal error and it is denoted as e(nTs).

$e(nTs)= x(nTs)- x^{\wedge}(nTs)$

- Here the predicted value x^(nTs) is produced by using a prediction filter.
- The quantizer output signal eq(nTs) and the previous prediction is added and given as input to the prediction filter, this signal is denoted by xq(nTs). This makes the prediction closer to the actually sampled signal. The quantized error signal eq(nTs) is very small and can be encoded by using a small number of bits. Thus the number of bits per sample is reduced in DPCM.

DPCM Receiver-

- In order to reconstruct the received digital signal, the DPCM receiver consists of a decoder and prediction filter.
- The input given to the decoder is processed and that output is summed up with the output of the predictor, to obtain better output. That means here first of all the decoder will reconstruct the quantized form of the original signal.
- Therefore the signal at the receiver differs from the actual signal by quantization error q(nTs), which is introduced permanently in the reconstructed signal.



5.10 COMPARISON BETWEEN PCM, DM, ADM AND DPCM-

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	It uses only one bit for one sample	Bits can be more than one but are less than PCM.
2.	Levels and step size	The number of levels depends on number of bits. Level size is fixed.	Step size is kept fixed and cannot be varied.	According to the signal variation, step size varies.	Number of levels is fixed.
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 distortion and quantization noise is present.
4.	Transmission bandwidth	Highest bandwidth is required since numbers of bits are high.	Lowest bandwidth is required.	Lowest bandwidth is required.	Bandwidth required is less than PCM.
5.	Feedback	There is no feedback in transmitter or receiver.	Feedback exists in transmitter.	Feedback exists.	Feedback exists.
6.	Complexity of Implementation	System is complex.	Simple	Simple	Simple

UNIT-6: DIGITAL MODULATION TECHNIQUES

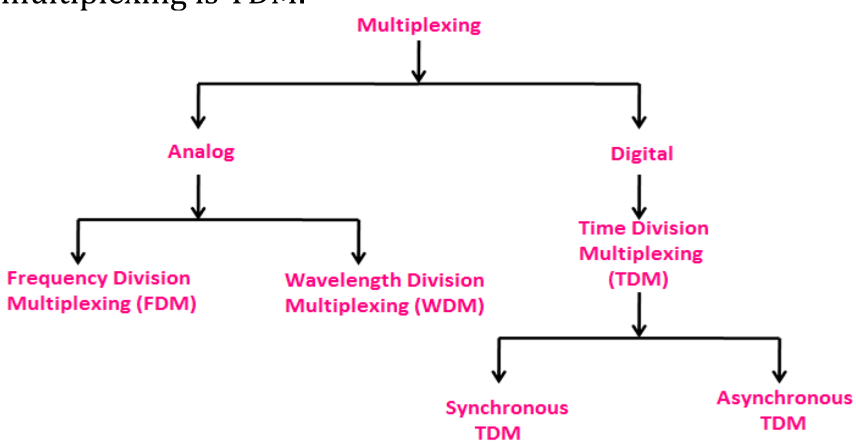
6.1 CONCEPT OF MULTIPLEXING

MULTIPLEXING-

Multiplexing is a technique which combines multiple signals into one signal, suitable for transmission over a communication channel such as coaxial cable or optical fiber. Multiplexing is the process of simultaneously transmitting two or more individual signals over a single communication channel. Multiplexing is done by using a device called Multiplexer or MUX. The multiplexer combines n input lines to generate one output line.

Classification-

The multiplexing techniques can be broadly classified into two categories i.e, analog and digital. Analog multiplexing can be either FDM or WDM and digital multiplexing is TDM.



In FDM many signals are transmitted simultaneously where each signal occupies a different frequency slot within a common bandwidth. WDM is a technology which multiplexes a number of optical carrier signals onto a single optical fiber by using different wavelengths of laser light. In TDM, the signals are not transmitted at a time, instead they are transmitted in different time slots.

FREQUENCY DIVISION MULTIPLEXING-

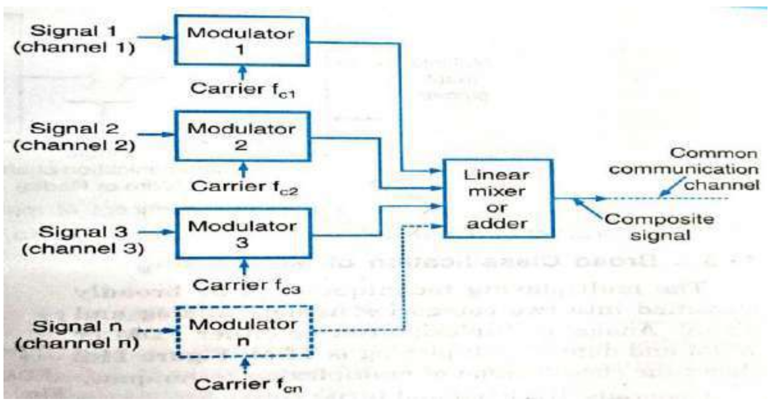
The operation of FDM is based on sharing the available bandwidth of a communication channel among the signals to be transmitted. This means that many signals are transmitted simultaneously with each signal occupying a different frequency slot within a common bandwidth. Each signal to be transmitted modulates a different carrier.

FDM transmitter-

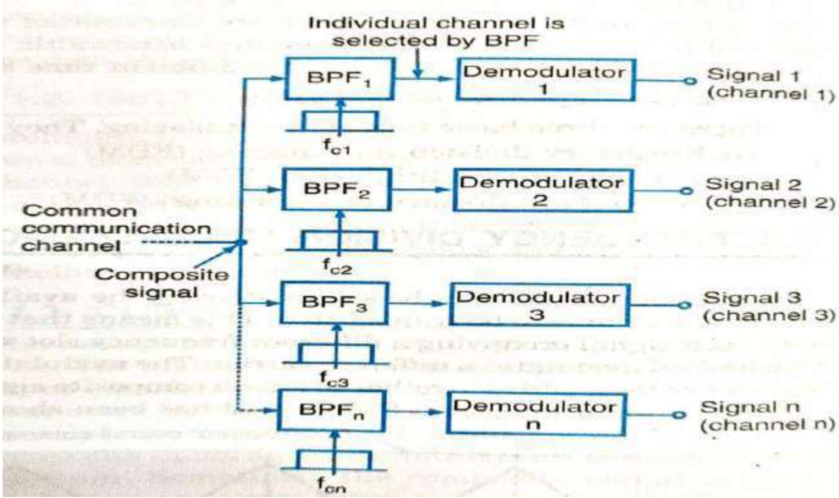
The signals which are to be multiplexed will modulate a separate carrier. Each signal modulates a separate carrier. The modulator outputs will contain the sidebands of the corresponding signals. The modulator outputs are added together in a linear mixer or adder.

The linear mixer is different from the normal mixers. Here the sum and difference frequency components are not produced. But only the algebraic addition of the modulated outputs will take place.

The composite signal at the output of mixer is transmitted over the single communication channel.



FDM Receiver-



The composite signal is applied to a group of band pass filters. Each BPF has a center frequency corresponding to one of the carriers. The BPFs have an adequate bandwidth to pass all the channel information without any distortion. Each filter will pass only its channel and reject all other channels. The channel demodulator then removes the carrier and recovers the original signal back.

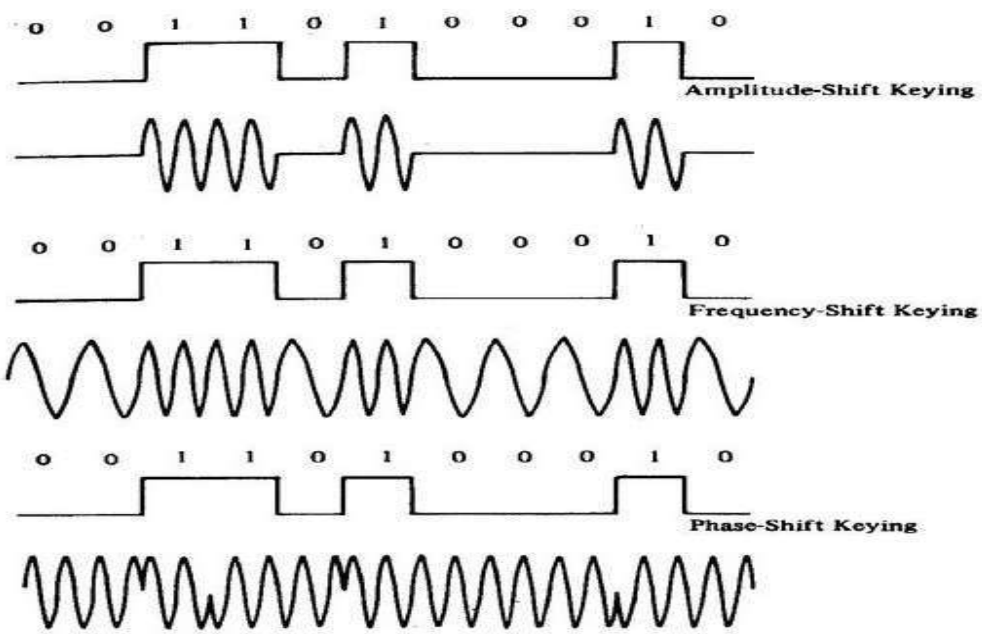
DIGITAL MODULATION FORMATS-

When we have to transmit a digital signal over a long distance, we need continuous wave modulation. For this purpose, the transmission medium can be in form of radio, cable or other type of channel.

Due to this process, there is some deviation in carrier frequency'f_c'. This deviation is known as the bandwidth of the channel. Such type of transmission is known as band pass transmission and the communication channel is known as band pass channel. The word band pass is used since the range of frequencies does not start from zero Hz to f_m Hz. The range of frequency from zero Hz to f_m Hz is known as low pass signal and such channel is known as low pass channel.

The modulating digital signal modulates some parameters like frequency, phase or amplitude of the carrier. If an amplitude of the carrier is changes depending on the input digital signal, then it is called ASK (Amplitude Shift Keying). This process is quite similar to analog amplitude modulation.

The FSK and PSK has constant amplitude envelope. Because of constant amplitude of FSK and PSK, the effect of non-linearity, noise interference is minimum on signal detection. However, these effects are more in ASK. Therefore, FSK and PSK are preferred over ASK.



In these waveforms, a single feature of the carrier undergoes modulation. 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.

The choice is made in favour of a scheme which possesses as many of the following design characteristics as possible.

1) Maximum data rate.
2) Minimum probability of symbol error.
3) Minimum transmitted power.
4) Maximum channel bandwidth.
5) Maximum resistance to interfering signals.
6) Minimum circuit complexity.

ADVANTAGES OF DIGITAL COMMUNICATION SYSTEM-

- The effect of distortion, noise, and interference is much less in digital signals as they are less affected.
- Digital circuits are more reliable.
- Digital circuits are easy to design and cheaper than analog circuits.
- The hardware implementation in digital circuits, is more flexible than analog.
- The occurrence of cross-talk is very rare in digital communication.

6.2 DIGITAL MODULATION TECHNIQUES-

Digital Modulation techniques can be coherent or non-coherent.

1) Coherent Digital Modulation Technique-

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.

2) Non- Coherent Digital Modulation Technique-

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.

6.3 GENERATION AND DETECTION OF BINARY ASK, FSK, PSK, QPSK, QAM, MSK, GMSK

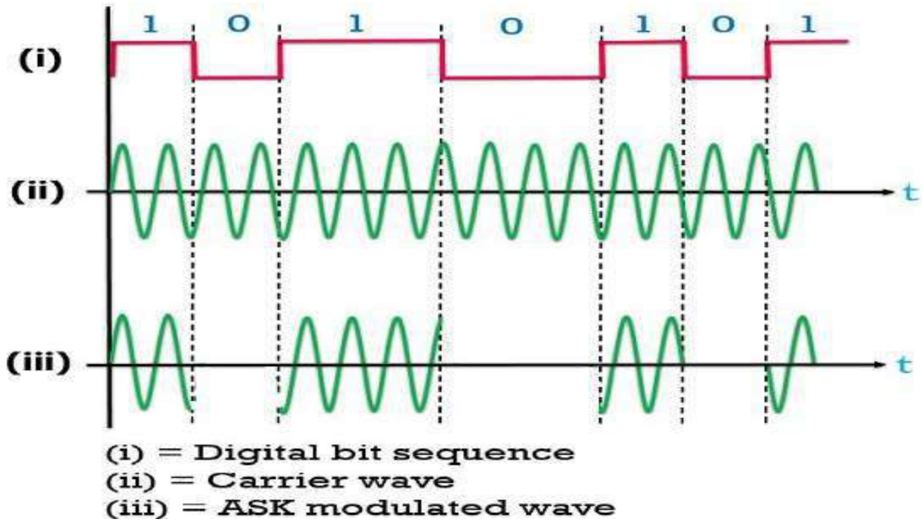
AMPLITUDE SHIFT KEYING-

- Amplitude Shift Keying is the simplest digital modulation technique.
- In this method, there is only one unit energy carrier and it is switched ON and OFF depending upon the input binary sequence.
- The ASK waveform may be represented as-

S(t)= $\sqrt{2P_s}\cos(2\pi f_c t)$ (To transmit 1)

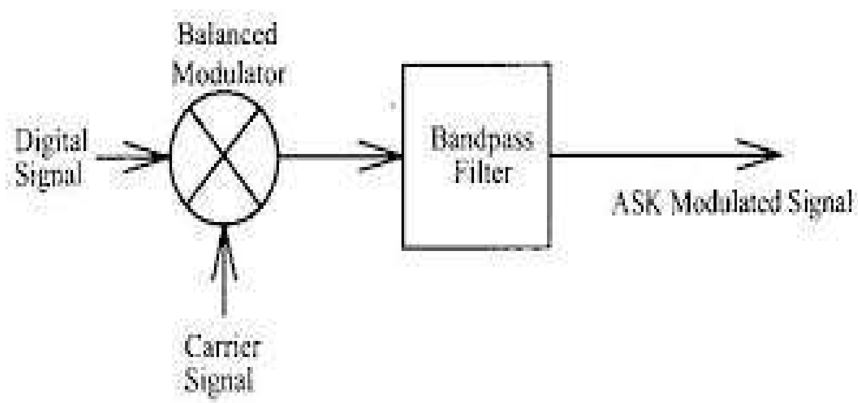
To transmit symbol ‘0’, the signal S(t)=0 i.e, no signal is transmitted.

Hence the ASK waveform looks like an ON-OFF of the signal. Therefore, it is also known as the ON-OFF keying.



The first figure shows a message signal represented in the forms of the bit stream. The second one is a carrier wave whose amplitude is to be varied according to the digital message signal. The last one shows the resultant ASK waveform which is amplitude modulated signal.

GENERATION-



ASK signal may be generated by simply applying the incoming binary data and the sinusoidal carrier to the two inputs of a balanced modulator. The resulting output will be applied to the band pass filter. The band-limiting filter, shapes the pulse depending upon the amplitude and phase characteristics of the band-limiting filter or the pulse-shaping filter. At the output of band pass filter we get the ASK modulated signal.

DEMODULATION-

Detection or demodulation is the process of recovering original message signal from the modulated waveform.

There are 2 types of detection technique.

1) Coherent Detection
2) Non- coherent Detection

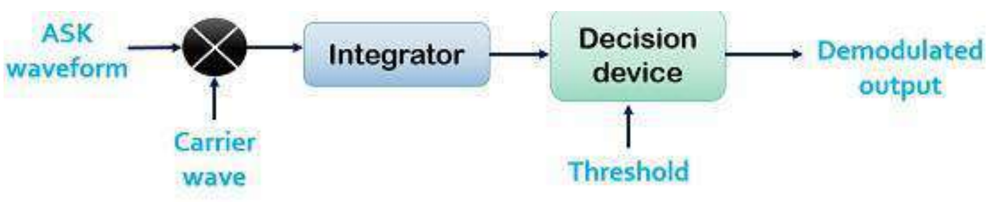
Coherent ASK demodulator-

The demodulation of binary ASK waveform can be achieved with the help of coherent detector. 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 sine carrier which is generated with the help of a local oscillator. The output of the product modulator is given to the input of the integrator.

The integrator operates on the output of the product modulator 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.

Then 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.

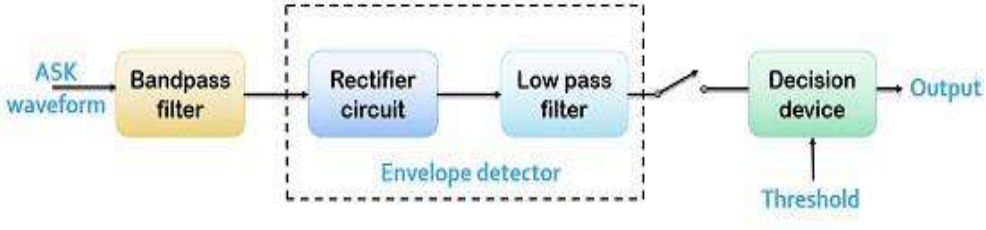


Coherent ASK detector

Non coherent Demodulator-

A non-coherent ASK detection technique composed of a band pass filter and envelope detector along with a decision device. As it does not require a synchronized carrier thus the method makes use of the rectifier circuit for the rectification of the signal. After which the signal is fed to the low pass filter.

The output of which is then provided to a decision device that compares the signal value with the present threshold value in a similar manner as done in the coherent detection. Thus generates the equivalent output, which is the original digital bit stream.



Non-coherent ASK detector

BINARY PHASE SHIFT KEYING (BPSK):-

In binary phase shift keying, binary symbol '1' and '0' modulate the phase of the carrier. Let us assume that the carrier is given as,

$$S(t)= A \operatorname{Cos}(2\pi f_c t)$$

Here 'A' represents peak value of sinusoidal carrier. The power dissipated would be

$$P= \tfrac{1}{2} A^2$$

$$A=\sqrt{(2P)}$$

For symbol '1'

$$S_1(t) = \sqrt{(2P)} \operatorname{Cos}(2\pi f_c t)$$

If next symbol is '0', then we have,

for symbol '0'

$$S_2(t) = \sqrt{(2P)}\operatorname{Cos}(2\pi f_c t)$$

Because $\operatorname{Cos}(\theta+\pi)= - \operatorname{Cos} \theta$, therefore the above equation can be written as,

$$S_2(t) = -\sqrt{(2P)}\operatorname{Cos}(2\pi f_c t)$$

We can define BPSK signal combinely as,

$$S(t)=b(t) \sqrt{(2P)} \operatorname{Cos}(2\pi f_c t)$$

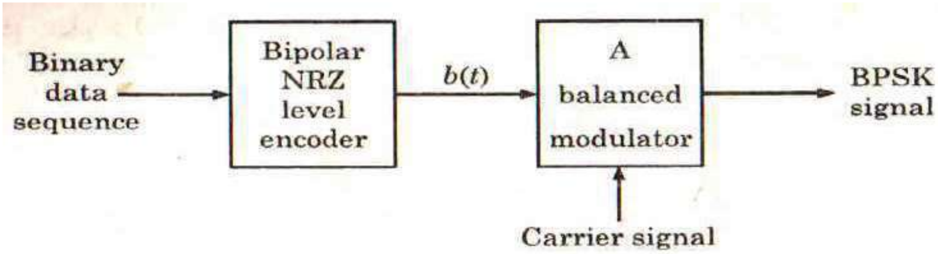
b(t)= +1 when binary '1' is to be transmitted

-1 when binary '0' is to be transmitted

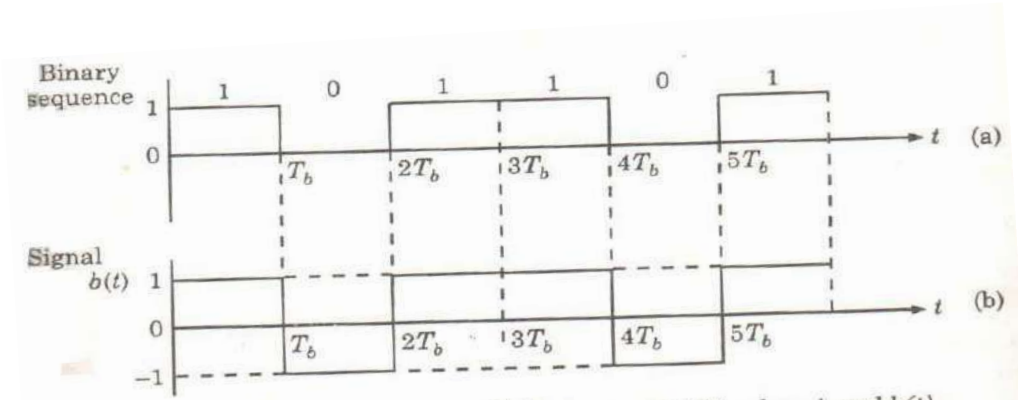
GENERATION OF BPSK SIGNAL:-

The BPSK signal may be generated by applying carrier signal to a balanced modulator. Here, the baseband signal b(t) is applied as a modulating signal to the balanced modulator.

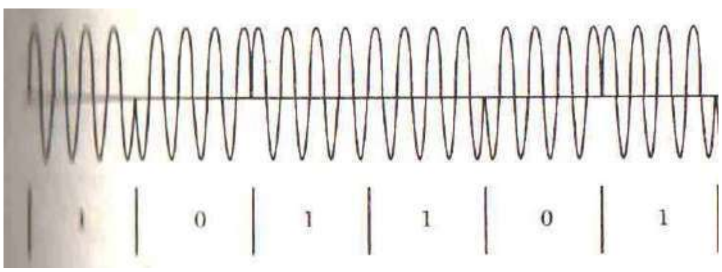
A NRZ level encoder converts the binary data sequence into bipolar NRZ signal.



Generation of BPSK



(a) Binary sequence, (b) Its equivalent bipolar signal b(t).



DEMODULATION OF BPSK :-

The transmitted BPSK signal is given as

$$S(t)=b(t)\sqrt{(2P)}\,\mathrm{Cos}(2\pi f_c t)$$

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 this phase shift be θ. Therefore, the signal at the input of the receiver can be written as

$$S(t)=b(t)\sqrt{(2P)}\,\mathrm{Cos}(2\pi f_c t+\theta)$$

Now, from this received signal, a carrier is separated because this is coherent detection. The received signal is allowed to pass through a square law device, we get a signal which is given as,

$\text{Cos}^2 (2\pi f_c t + \theta)$

Again, we know that

$\text{Cos}^2 \theta = (1+ \text{Cos } 2\theta)/2$

Therefore, we have

$\text{Cos}^2 (2\pi f_c t + \theta) = (1+ \text{Cos } 2(2\pi f_c t +\theta))/2$

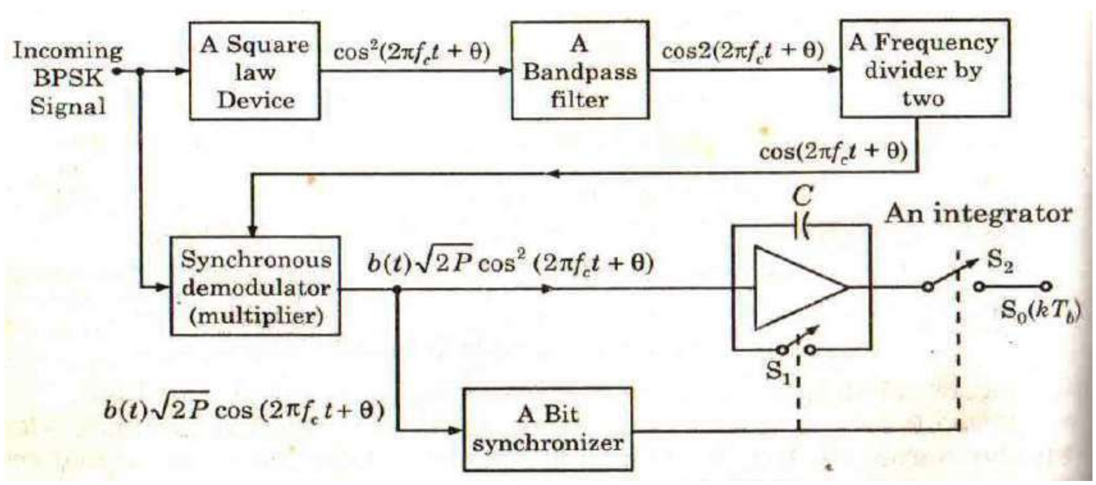
$= \frac{1}{2} + \frac{1}{2}(\text{Cos } 2(2\pi f_c t +\theta))$

Here, ½ represents a DC level. This signal is then allowed to pass through a band pass filter whose pass band is centered around $2f_c$. Band pass filter removes the DC level of ½ and at the output, we obtain

$\text{Cos } 2(2\pi f_c t +\theta)$

Then this signal 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,

$\text{Cos } (2\pi f_c t +\theta)$



Reception of Baseband Signal in BPSK

The synchronous demodulator multiplies the input signal. The output of multiplier is,

$b(t)\sqrt{(2P)}\,\text{Cos}(2\pi f_c t +\theta) \times \text{Cos}(2\pi f_c t +\theta) = b(t)\sqrt{(2P)}\,\text{Cos}^2(2\pi f_c t +\theta)$

$= b(t)\sqrt{(2P)} \times 1/2[1+ \text{Cos } 2(2\pi f_c t +\theta)]$

$$= b(t)\sqrt{(P/2)}[1+ \text{Cos } 2(2\pi f_c t + \theta)]$$

This signal is then applied to the bit synchronizer and integrator. The integrator integrates the signal over one bit period. The bit synchronizer takes care of starting and ending times of a bit. At the end of bit duration Tb, the bit synchronizer closes switch S2 temporarily. This connects the output of an integrator to the decision device. The synchronizer then opens switch S2 and switch S1 is closed temporarily. The integrator then integrates next bit.

In the K^{th} bit interval we can write output signal as under

$$s_o(KT_b) = b(KT_b)\sqrt{(\tfrac{P}{2})} \int_{(K-1)T_b}^{KT_b} [1 + \text{Cos } 2(2\pi f_c\ t + \theta)]\ dt$$

$$s_o(KT_b) = b(KT_b)\sqrt{(\tfrac{P}{2})}[\int_{(K-1)T_b}^{KT_b} [1dt + \int_{(K-1)T_b}^{KT_b} \text{Cos } 2(2\pi f_c\ t + \theta)]\ dt\ ]$$

Where $\int_{(K-1)T_b}^{KT_b} \text{Cos } 2(2\pi f_c\ t + \theta)]\ dt = 0$

$$s_o(KT_b) = b(KT_b)\sqrt{(\tfrac{P}{2})} \int_{(K-1)T_b}^{KT_b} 1dt$$

$$= b(KT_b)\sqrt{(P/2)}\ [t]_{(K-1)Tb}^{KTb}$$

$$= b(KT_b)\sqrt{(P/2)}\{KT_b - (K-1)T_b\}$$

$$= b(KT_b)\sqrt{(P/2)}\ T_b$$

This signal is then applied to a decision device which decides whether transmitted symbol was zero or one.

<u>BANDWIDTH</u>

The minimum bandwidth of BPSK signal is equal to twice of the highest frequency contained in baseband signal. BW = $2f_b$

BINARY FREQUENCY SHIFT KEYING (BFSK)-

In binary frequency shift keying, the frequency of the carrier is shifted according to the binary symbol. However the phase of the carrier is unaffected. We have two different frequency signals according to binary symbols.

If $b(t)$= '1', then $S_H(t) = \sqrt{(2P_s)}$ Cos $(2\pi f_c t + \Omega)t$

$b(t)$= '0', then $S_L(t) = \sqrt{(2P_s)}$ Cos $(2\pi f_c t - \Omega)t$

the equation combinely written as

$S(t) = \sqrt{(2P_s)}$ Cos $[(2\pi f_c t + d(t)\,\Omega)t]$

Hence if symbol '1' is to be transmitted the carrier frequency will be $f_c + \Omega/2\pi$

Hence if symbol '1' is to be transmitted the carrier frequency will be $f_c - \Omega/2\pi$

Therefore, we have

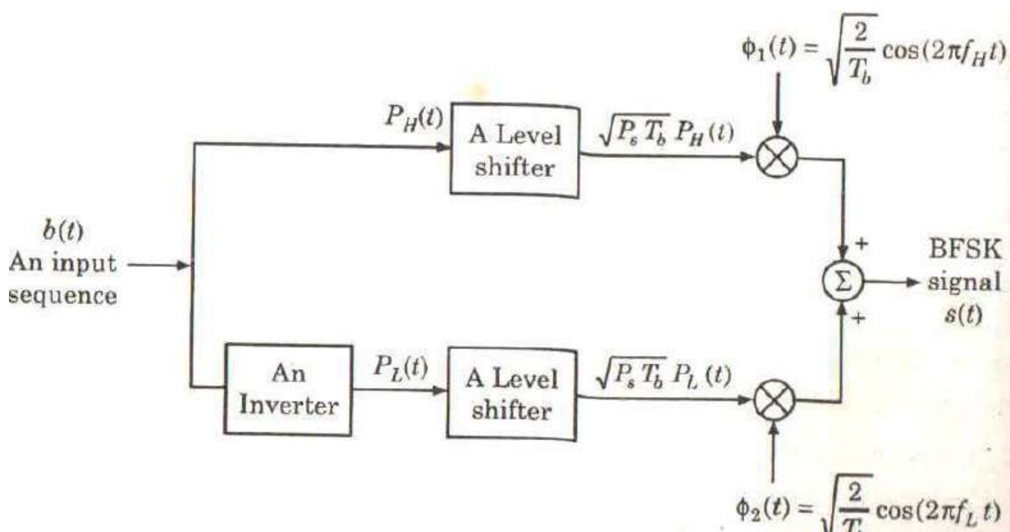
$f_H = f_c + \Omega/2\pi$ for symbol '1'

$f_L = f_c - \Omega/2\pi$ for symbol '1'

GENERATION OF BFSK:-

The 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 input is '+1') or zero (if input is zero). Further, there are product modulators after level shifter. The two carrier signals $\Phi_1(t)$ & $\Phi_2(t)$ are used. $\Phi_1(t)$ & $\Phi_2(t)$ are orthogonal to each other. The carrier signal multiplied with the output of the level shifter in product modulator.

The adder then adds the two signals from product modulator.

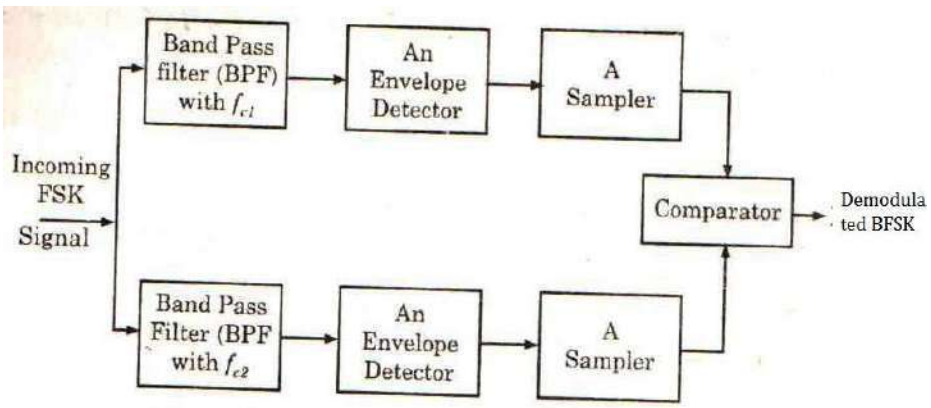


Bolck diagram for BFSK generation.

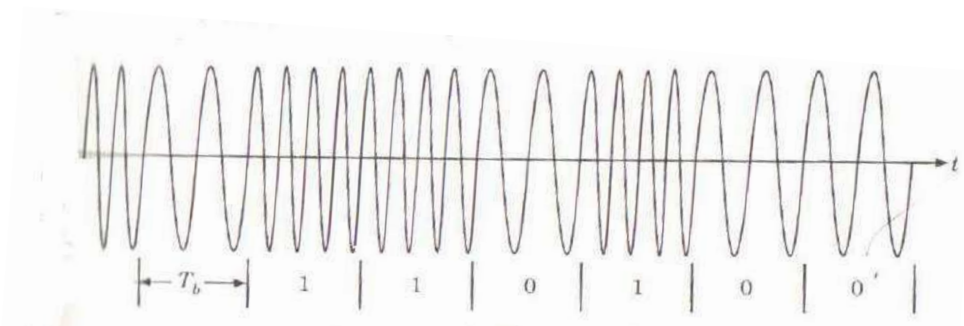
DETECTION OF BFSK-

This receiver contains two band pass filters, one with center frequency f_{C1} and other with center frequency f_{C2}.

Because f_{C1}- f_{C2}= $2f_b$, the outputs of filters do not overlap each other. The band pass filters pass their corresponding main lobes without much distortion. The outputs of filters are applied to envelop detectors. The outputs of detectors are compared by the comparator is used, then the output of comparator is the bit sequence b(t).



Demodulation of BFSK Signal



The BFSK signal.

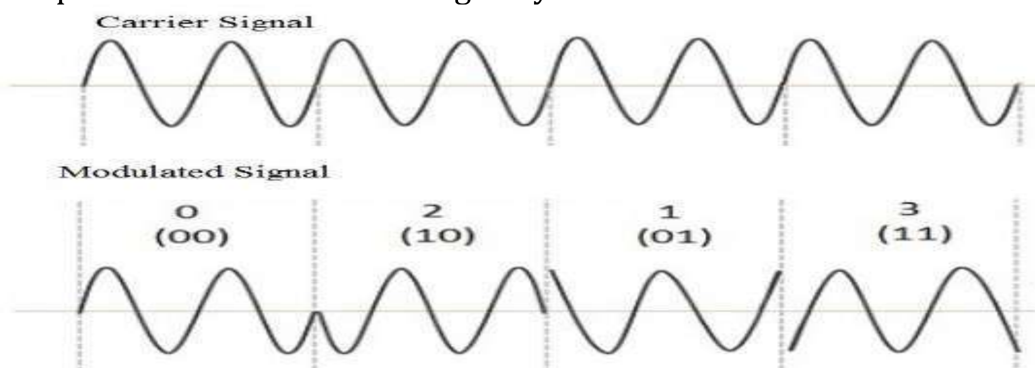
QUADRATURE PHASE SHIFT KEYING (QPSK)-

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. If two or more bits are combined in some symbols, then the bit rate will be reduced. This reduces the transmission channel bandwidth.

Quadrature Phase Shift Keying (QPSK) is a form of phase modulation technique, in which two information bits (combined as one symbol) are modulated at once, selecting one of the four possible carrier phase shift states.

Symbol	Phase
00	0
01	90
10	180
11	270

In case of BPSK, when symbol changes the level, the phase of the carrier is changed by 180°, there were only two symbols in BPSK, the phase shift occurs in two levels only. But in QPSK, two successive bits are combined. 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 90° .



QPSK Transmitter-

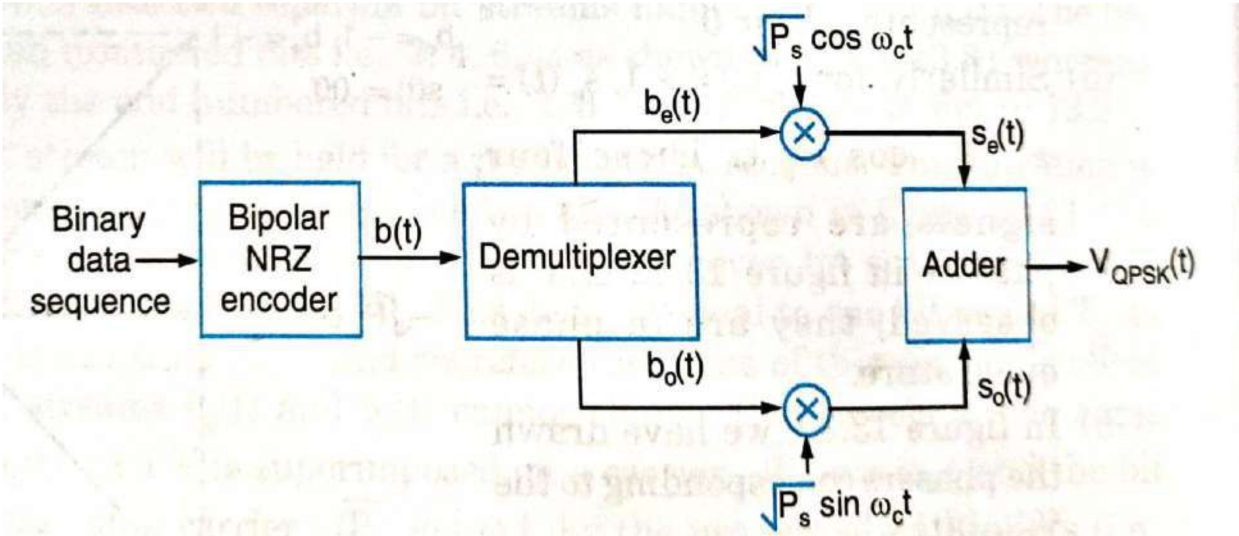
``

The input binary sequence is first converted into a bipolar NRZ signal, b (t). The value of b (t) = +1V for a logic 1 input and b (t) = -1V when the binary input is equal to 0. The de-multiplexer will divide b (t) into two separate bit streams named $b_o(t)$ and$b_e(t)$.

The bit stream $b_e(t)$ consists of only the even numbered bits whereas the $b_o(t)$ bit stream consists of only the odd numbered bits. The bit streams $b_e(t)$ is superimposed on a carrier $\sqrt{P_s}$Cos $w_c t$ and the bit stream $b_o(t)$ is superimposed on a carrier $\sqrt{P_s}$Sin $w_c t$, by the use of two multipliers, to generate two and $s_e(t)$ and $s_o(t)$ respectively. These two signals are basically BPSK signals. These signals are then added together to generate the QPSK output signal $V_{QPSK}(t)$ or s(t).

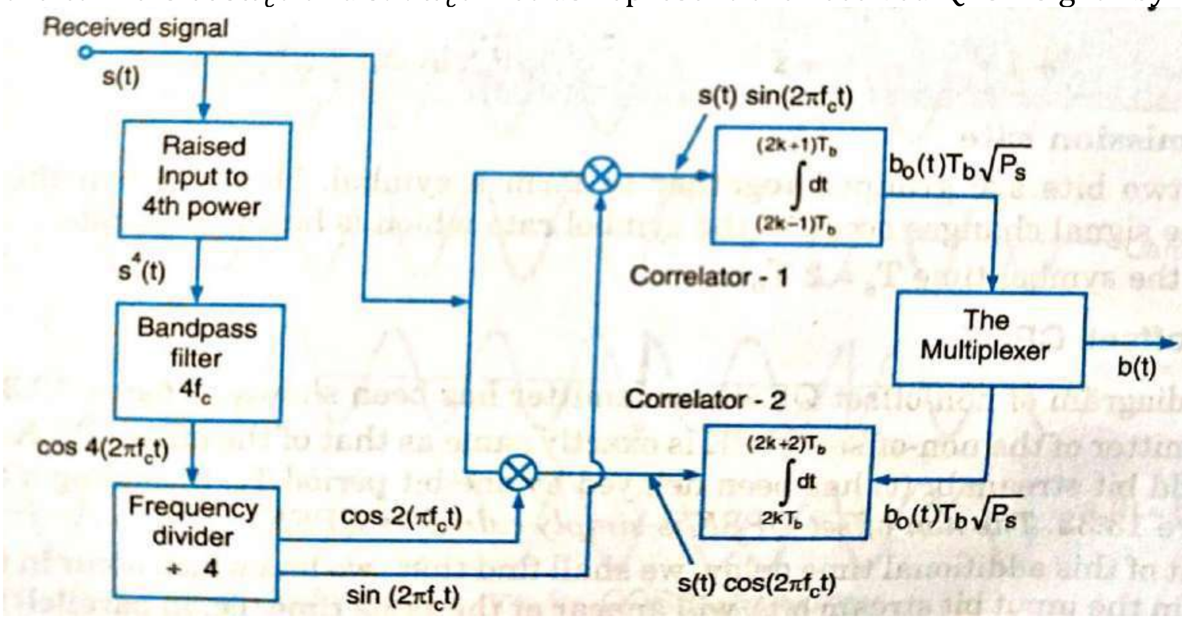
$$V_{QPSK}(t) = s_o(t) + s_e(t)$$

$$= b_o(t)\sqrt{P_s}\, Sinw_c t + b_e(t)\sqrt{P_s}\, cosw_c t$$



QPSK Receiver-

It is a synchronous detection technique. Therefore it is necessary to locally generate the carriers $cosw_c t$ and $sin\ w_c t$. Let us represent the received QPSK signal by s(t).



The received QPSK signal s(t) is raised to fourth power i.e, s^4(t). This signal is then filtered by using a band pass filter with a center frequency of $4w_c$(t). The output of band pass filter is cos $4w_c$(t). A frequency divider which divides the frequency at the filter output by 4 generates the two carrier signals $cosw_c t$ and $sin\ w_c t$. The incoming signal s(t) is applied to two synchronous demodulators consisting of a multiplier followed by an integrator. Each integrator integrates over a two bit interval of duration T_s= $2T_b$. One synchronous demodulator uses $cosw_c t$ as carrier signal and the other one uses $sin\ w_c t$ as a carrier signal.

The input to the upper integrator-

S(t) * $\sin w_c t = b_o(t)\sqrt{P_s} sin^2 w_c t + b_e(t)\sqrt{P_s} \sin w_c t \; cos w_c t$

This signal is applied to the upper integrator. This integrator will integrate its input signal over a symbol period of $T_s = 2T_b$.

Upper Integrator output= $\int_{(2k-1)T_b}^{(2k+1)T_b} s(t) \sin w_c t \; dt$

= $b_o(t)\sqrt{P_s}\int_{(2k-1)T_b}^{(2k+1)T_b} sin^2 w_c t \; dt + b_e(t)\sqrt{P_s}\int_{(2k-1)T_b}^{(2k+1)T_b} sin w_c t \; cos w_c t \; dt$

= ½ $b_o(t)\sqrt{P_s}\int_{(2k-1)T_b}^{(2k+1)T_b}[1 - cos\; 2w_c t]dt + 1/2 \; b_e(t)\sqrt{P_s}\int_{(2k-1)T_b}^{(2k+1)T_b} sin 2w_c t \; dt$

= ½ $b_o(t)\sqrt{P_s}\int_{(2k-1)T_b}^{(2k+1)T_b} 1 \; dt - \frac{1}{2} b_o(t)\sqrt{P_s}\int_{(2k-1)T_b}^{(2k+1)T_b} cos 2w_c t \; dt + \frac{1}{2} b_e(t)\sqrt{P_s}\int_{(2k-1)T_b}^{(2k+1)T_b} sin 2w_c \text{t dt}$

The value of second and the third term is zero.

Hence, integrator output-

½ $b_0(t)\sqrt{P_s}\int_{(2k-1)T_b}^{(2k+1)T_b} 1 \; dt$

= ½ $b_o(t)\sqrt{P_s}[t]_{(2k-1)T_b}^{(2k+1)T_b}$

= ½ $b_o(t)\sqrt{P_s}[(2\text{k}+1)T_b - (2k-1)T_b]$

= ½ $b_o(t)\sqrt{P_s}[2\text{k}T_b + T_b - 2kT_b + T_b]$

= ½ $b_o(t)\sqrt{P_s}\; 2T_b$

=$b_o(t)\sqrt{P_s}\; T_b$

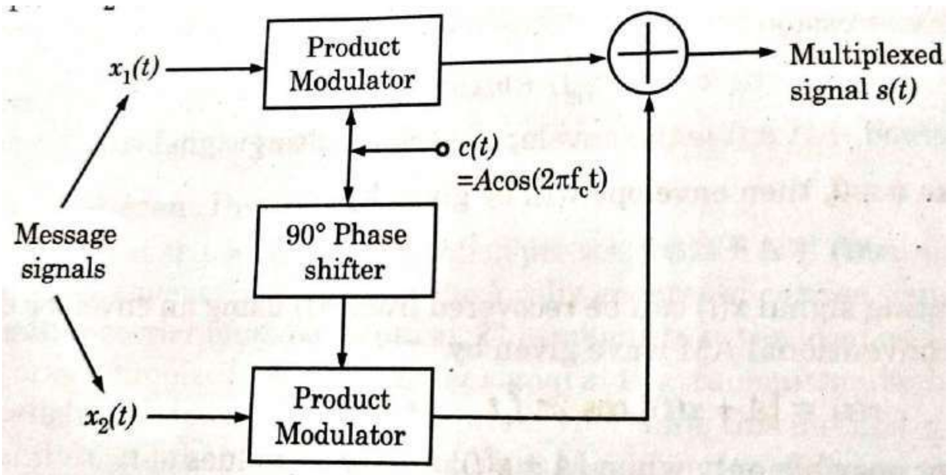
Similarly we can prove the output of the lower integrator is given by $b_e(t)\sqrt{P_s}T_b$. Thus at the output of the two integrators, we obtain the bit streams $b_o(t) and \; b_e(t)$. Both the bit streams are multiplexed and at the output we get the original bit stream b(t).

QUADRATURE AMPLITUDE MODULATION (QAM)-

QAM modulation techniques can modulate two individual signals and transmitted to the receiver. Due to use of two input signals the channel bandwidth increases. QAM can able to transmit two message signals over the same channel. This QAM technique is also known as “Quadrature Carrier Multiplexing”.

In QAM the phase and amplitude of the carrier signal is varied based on the information in a digital signal.

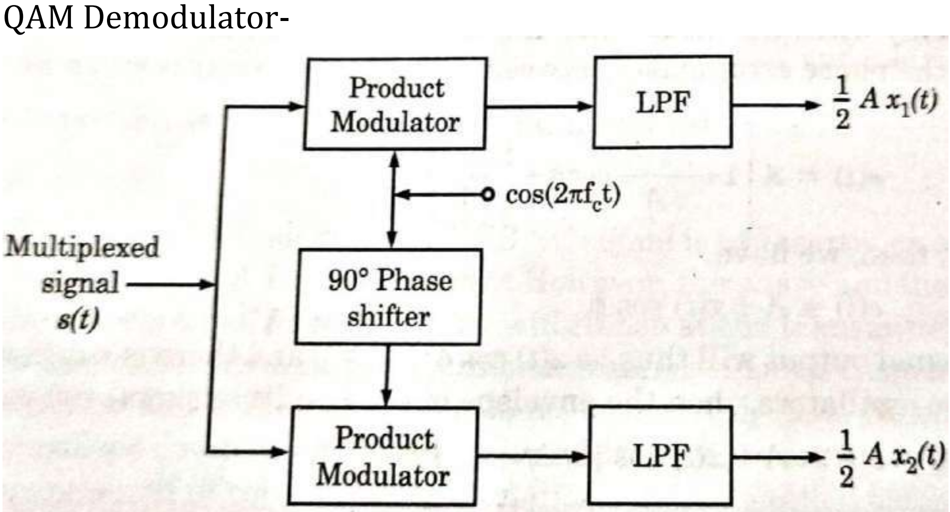
QAM modulator-



In the QAM transmitter the upper product modulator and local oscillator are called the in-phase (I) channel. The lower product modulator and 90° phase shifted local oscillator output using 90° phase shift network are called a quadrature (Q) channel. Both output signals of the in phase channel and quadrature channel are summed. So the resultant output will be QAM signal.

The resultant overall signal consisting of the combination of both I and Q carriers contains both amplitude and phase variation.

QAM Demodulator-



The QAM signal received by the demodulator. The signal s(t) applied to the one input of the two product modulator. To the other input of the upper product modulator carrier signal $i.e,$ Cos W_ct.is given and to the other input of the lower product modulator 90° phase shifted carrier i.e, Sin W_ct is given.

Output of upper product modulator is $y_1'(t)$.

$$y_1'(t) = S(t)\,.\;\text{Cos } W_c t$$

$$= [x_1(t)\ A\ \text{Cos } W_c t + x_2(t) A\ \text{Sin } W_c t]\ \text{Cos } W_c t$$

$$= x_1(t)\ A\ \text{Cos}^2\ W_c t + x_2(t) A\ \text{Sin } W_c t\ .\ \text{Cos } W_c t$$

$= x_1(t)\, A\, [\frac{1+\text{Cos } 2\, W_c t}{2}] + \frac{x_2(t)A}{2} \text{Sin } 2\, W_c t$

$= \frac{x_1(t)A}{2} + \frac{x_1(t)A}{2} \text{Cos } 2\, W_c t + \frac{x_2(t)A}{2} \text{Sin } 2\, W_c t$

Output of lower product modulator is $y_2'(t)$.

$y_2'(t) = S(t)\,.\ \ \text{Sin } W_c t$

$= [x_1(t)\, A\, \text{Cos } W_c t + x_2(t)A\, \text{Sin } W_c t]\, \text{Sin } W_c t$

$= x_1(t)\, A\text{Sin } W_c t\, .\, \text{Cos } W_c t + x_2(t)A\text{Sin}^2\, W_c t$

$= \frac{x_1(t)A}{2} \text{Sin } 2\, W_c t + x_2(t)\, A\, [\frac{1+\text{Cos } 2\, W_c t}{2}]$

$= \frac{x_1(t)A}{2} \text{Sin } 2\, W_c t + \frac{x_2(t)A}{2} - \frac{x_2(t)A}{2} \text{Cos } 2\, W_c t$

LPF eliminate the high frequency components and at the output of the upper LPF we get,

$$y_1'(t) = \frac{x_1(t)A}{2}$$

At the output of the lower LPF we get,

$$y_2'(t) = \frac{x_2(t)A}{2}$$

MINIMUM SHIFT KEYING (MSK)-

Minimum Shift Keying (MSK), is a type of continuous phase frequency shift keying. MSK encodes each bit as a half sinusoidal signal. This results in a constant envelope signal, which reduces problem caused by non-linear distortion.

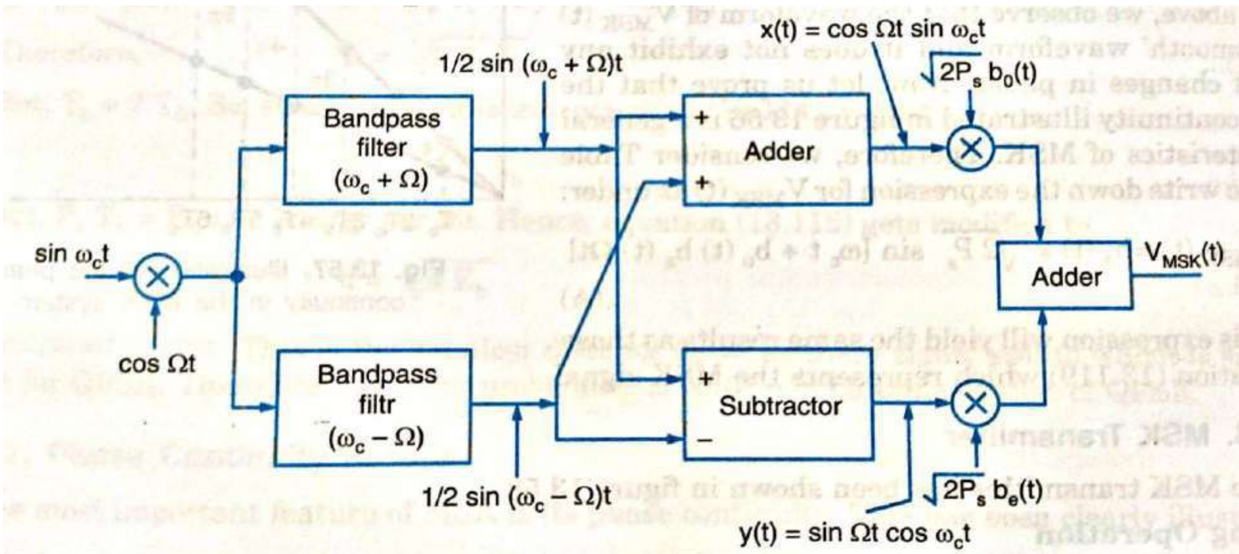
In MSK the difference between the higher and lower frequency is identical to half the bit rate. Thus the maximum deviation is $0.5 f_m$ where f_m is the maximum modulating frequency 0.5 is the modulation index.

This is the smallest FSK modulation index that can be chosen such that the waveforms for 0 and 1 are orthogonal. Continuous Phase Frequency Shift Keying (CPFSK) is a commonly used variation of frequency shift keying. A standard FSK does not have continuous phase, as the modulated waveform switches between two sine wave with different frequencies.

The phase of a CPFSK is in fact continuous, this characteristic is desirable for signals that are to be transmitted over a bandlimited channel.

GENERATION OF MSK-

The carrier signal $\text{Sin} w_c t$ is multiplied with the $cos\Omega t$ in a balanced modulator.



Output of the multiplier- Sin$w_c tcos\Omega t$

= ½ sin$(w_c + \Omega)\ t + \frac{1}{2}sin(\ w_c - \Omega)t$

The multiplier output contains the sum and difference components of w_c and Ω. This is applied to two band pass filters with center frequencies at $(w_c + \Omega)$ and $(w_c - \Omega)$ respectively.

Output of BPF1= ½ sin$(w_c + \Omega)$

Output of BPF2= ½ sin$(w_c - \Omega)$

Both these outputs are applied to an adder and a subtractor. At the output of these adder and subtractor, we get the signals x(t) and y(t) respectively which are given by,

X(t) = ½ sin$(w_c + \Omega)\ t + \frac{1}{2} sin(\ w_c - \Omega)t$ = Sin$w_c tcos\Omega t$

Y(t)= ½ sin$(w_c + \Omega)\ t - \frac{1}{2} sin(\ w_c - \Omega)t$

= ½ [sin$w_c t \cos \Omega t + cos\ w_c t \sin \Omega t - \sin w_c t \cos \Omega t + \cos w_c t\ sin\Omega t$]

= cos $w_c t \sin \Omega t$

Here, x(t) is then multiplied with $\sqrt{2P_s} b_o(t)$ while y(t) is multiplied with $\sqrt{2P_s} b_e(t)$. The outputs of these multipliers are then added together to produce the MSK signal given by,

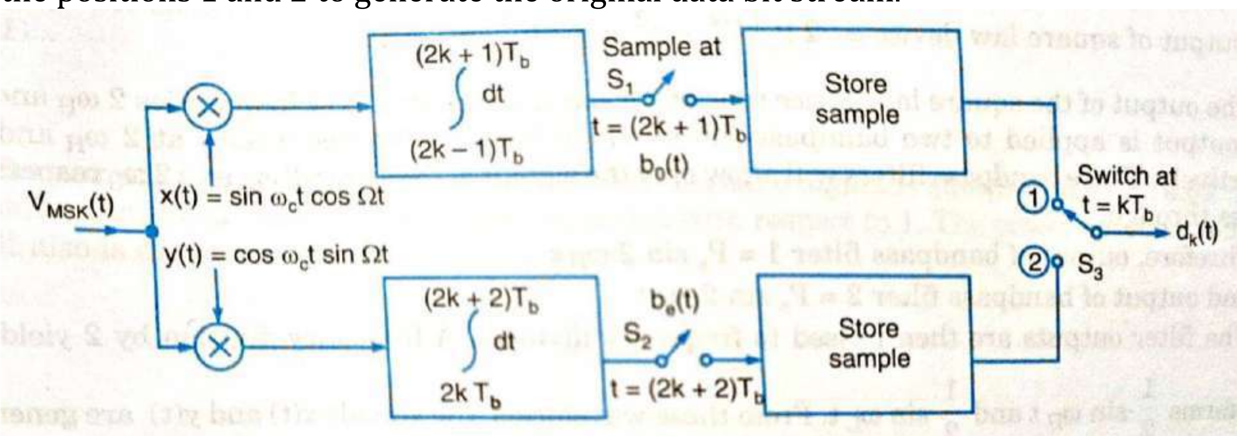
$$V_{MSK}(t) = \sqrt{2P_s} b_o(t) sinw_c(t) \cos \Omega t + \sqrt{2P_s} b_e(t) cosw_c t \sin \Omega t$$

= $\sqrt{2P_s} b_o(t) \frac{1}{2}$ [sin$(w_c + \Omega)\ t +$ sin$(\ w_c - \Omega)t] + \sqrt{2P_s} b_e(t)$ ½ [sin$(w_c + \Omega)t -$ sin$(\ w_c - \Omega)t]$

$$= \sqrt{2P_s}\,[\frac{b_o(t)+ b_e(t)}{2}]\sin(w_c + \Omega)t + \sqrt{2P_s}\,[\frac{b_o(t)- b_e(t)}{2}]\sin(w_c - \Omega)t$$

RECEPTION OF MSK-

This is a synchronous type of detection. The signals x(t) and y(t) are regenerated at the receiver. Then the incoming MSK signal is multiplied by the signals x(t) and y(t) in the two balanced modulators. The bit $b_o(t)$ and $b_e(t)$ is determined from the multiplier integrator chain. Both the integrators will integrate over the symbol duration of $2T_b$ seconds. At the end of each integration interval, the integrator outputs are sampled and stored. The switch S_3 at the output will then switch between the positions 1 and 2 to generate the original data bit stream.



GAUSSIAN MINIMUM SHIFT KEYING (GMSK)-

GMSK is the derivative of MSK modulation. MSK is derived from offset QPSK modulation. GMSK is probably most widely associated with the 2G GSM mobile communication system where it proved to be an effective form of modulation.

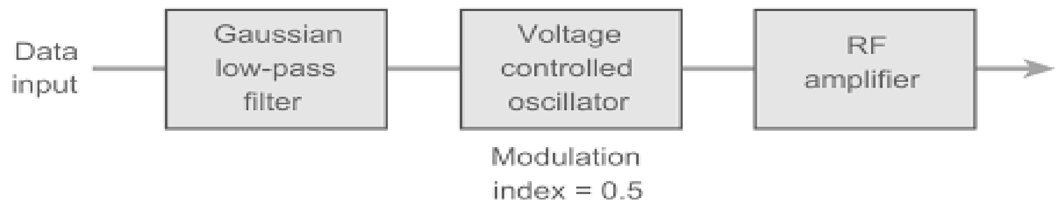
GMSK modulation is based on MSK which is a form of continuous phase frequency shift keying (CPFSK). One of the problems with standard forms of PSK is that sidebands extend out from the carrier. To overcome this, MSK and its derivative GMSK can be used.

Here there are no phase discontinuities because the frequency changes occur at the carrier zero crossing points.

GMSK Modulator-

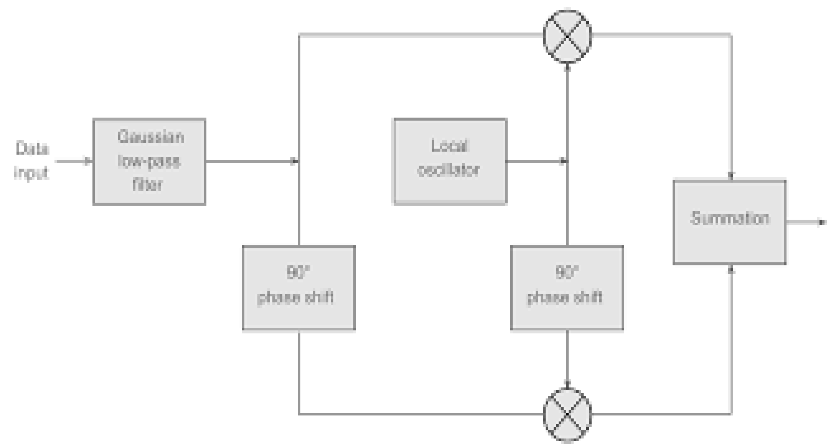
There are two ways in which GMSK modulation can be generated.

In the first method the modulating signal filtered using a Gaussian filter and then apply this to a frequency modulator, where the modulation index is set to 0.5.



This method is very simple and straight forward but it has the draw back that the modulation index must exactly equal to 0.5. A Gaussian filter is a linear filter. It's usually used to blur the image or to reduce noise.

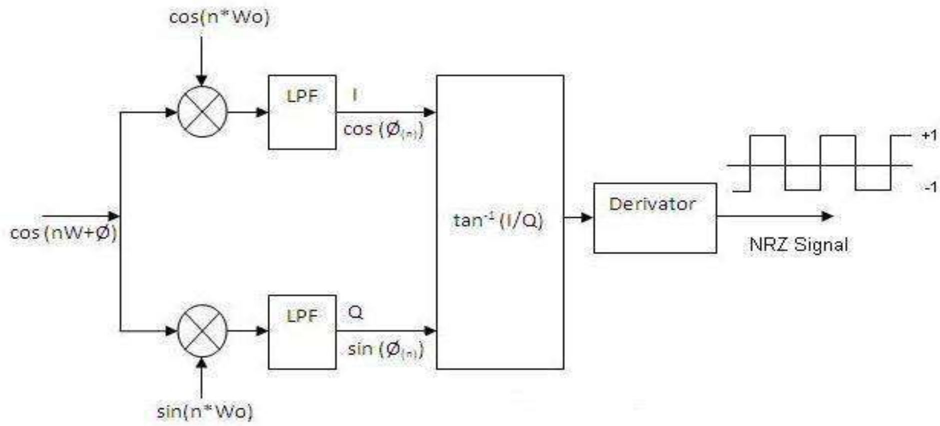
A second method is more widely used. Here a quadrature modulator is used. The term quadrature means that the phase of a signal is in quadrature or 90° to another one.



The quadrature modulator uses one signal that is said to be in phase and another that is in quadrature to this. This type of modulator is called I-Q modulator. Using this type of modulator the modulation index can be maintained at exactly 0.5 without the need for any setting or adjustments. This makes it much easier to use and capable of providing the required level of performance without the need for adjustments.

GMSK Demodulator-

For demodulation the technique can be used in reverse. GMSK demodulator derives back Ø using tan inverse function, which is applied to derivator block to obtain NRZ signal back.



Before doing this mixing and low pass filtering is done to obtain I and Q components from two chains.

6.4 T1 CARRIER SYSTEM-

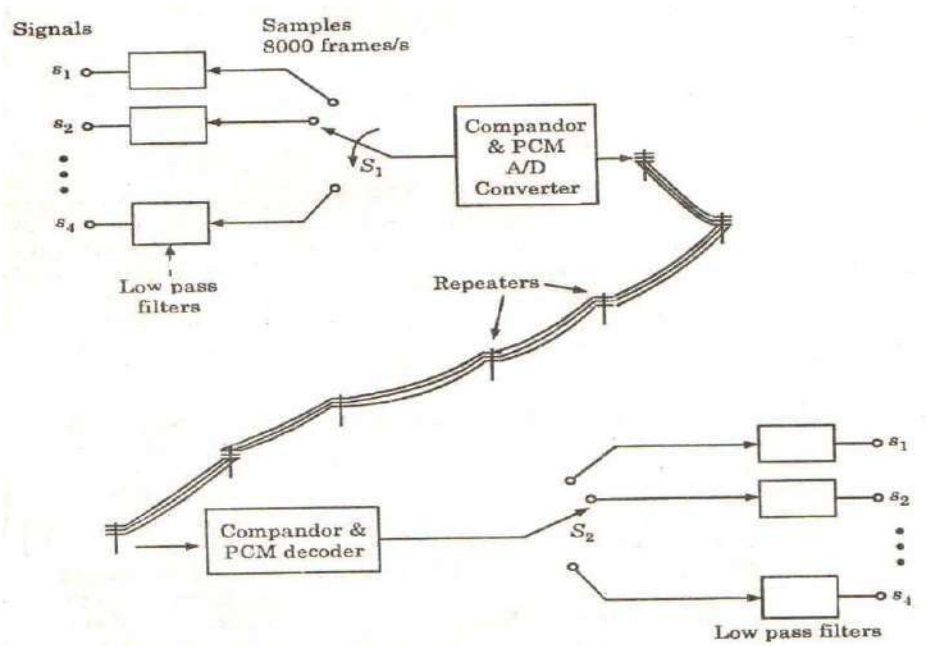
A T1 carrier is a dedicated telephone connection or a time division multiplexed digital transmission service that supports a data rate of 1.544Mbps. A T1 line generally includes 24 separate channels, each of which is able to support 64Kbps. Every 64 Kbps channel is often customized to transport voice or data traffic.

The T1 carrier systems are totally digital and use time division multiplexing (TDM) and Pulse Code Modulation. T1 is the method that is conventionally used by the telephone companies for transporting digitized telephone communication.

T1 is fully digital, it eradicates the chance of crosstalk which is happen in analog carrier network. A T1 physically includes two twisted pairs of copper wire. The pairs use a full duplex configuration in which one pair sends information while the other receives information.

Operation-

The basic time division multiplexing scheme called the T- carrier system, which is used to convey multiple signals over telephone lines using wideband coaxial cable. It accommodates 24 analog signals which are referred as S1 to S24. Each signal is band limited to approximately 3.3khz and is sampled at the rate of 8khz. .



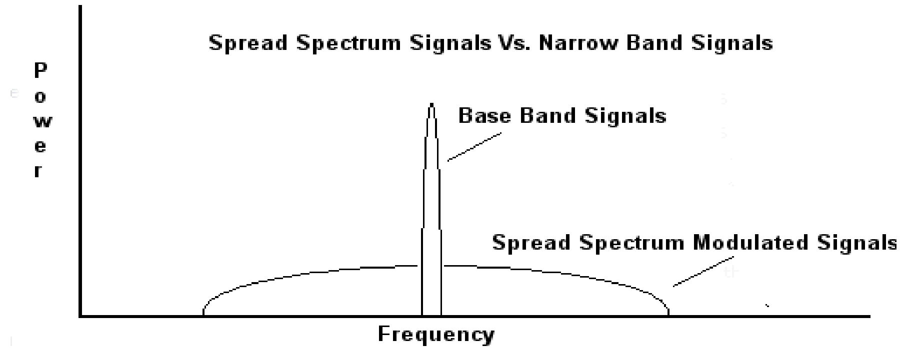
Each of the time division multiplexed signals is next A/D converted and compounded. The resulting digital waveform is transmitted over a coaxial cable, the cable serving to minimize signal distortion and serving also to suppress signal distortion due to noises from external sources. Periodically, at approximately 6000ft intervals the signal is regenerated by amplifiers known as repeaters and then sent towards its destination. The repeater eliminates from each bit the effect of the distortion introduced by the channel.

Also the repeater removes from each bit any superimposed noise and thus, even having received a distorted and noisy signal, it retransmits a distortion less and noise free signal which was originally sent.

At the destination, the signal is companded, decoded and de-multiplexed and thus making available the 24 original signals individually.

6.5 SPREAD SPECTRUM-

Spread spectrum is a type of signal modulation, in which the signal frequency is spread over a very large bandwidth signal. The large bandwidth signal is known as pseudo noise sequence or PN code or chipping code.



The main advantage of spread spectrum communication technique is to prevent interference whether it is intentional or unintentional. An intruder with no official access is not able to crack them. Hence, these techniques were used initially for military purposes.

Spread spectrum was also known as noise modulation. This was because the spread spectrum signals look like a noise to an ordinary FM/AM receivers due to spreading of signals over a large bandwidth.

There are three types of spread spectrum.

- Frequency Hopped Spread Spectrum (FH-SS)
- Direct Sequence Spread Spectrum (DS-SS)
- Time-hopping spread spectrum (TH-SS)

6.6 WORKING OPERATION OF SPREAD SPECTRUM MODULATION TECHNIQUES-

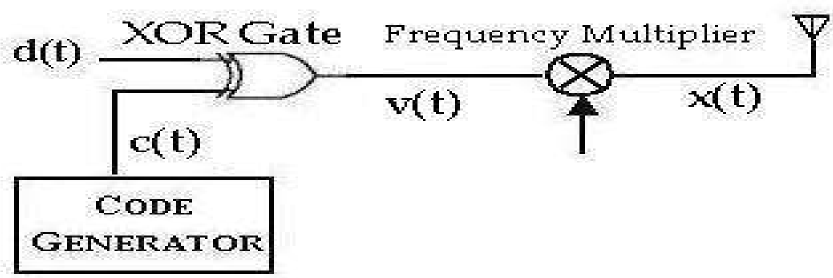
Direct Sequence Spread Spectrum (DS-SS)-

In DS-SS, the base band signals are called bits and code bits are called chips. The base band signal bandwidth is multiplies several times by the spreading signals. The chip rate is much higher than the bit rate.

The spreading signal sequence is unique for the transmitter and the same chip sequence is used at the receiver to reconstruct the signal.

DS-SS Transmitter-

In the DS-SS transmitter, a code generator is a pseudo random generator that generates a known pseudo noise code sequence. The code has finite length and repeats periodically.

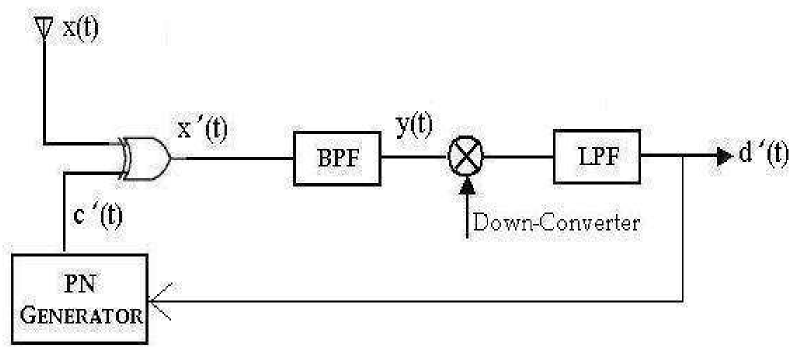


In this figure, each data bit is coded with 8 chips. Higher the number of chips per bit, higher will be the processing gain.

Processing gain=10log[$\frac{R_c}{R_b}$]

Higher processing gain, results in greater immunity to noise and interfering signals.

DS-SS Receiver-



The receiver consists of a PN generator that feeds the matching chip sequence as used in transmitter, to an XOR gate to reproduce the original bit sequence.

The PN generator is driven by an error signal from the output of the LPF. An error voltage at the output of the LPF provides necessary correction to the PN generator.

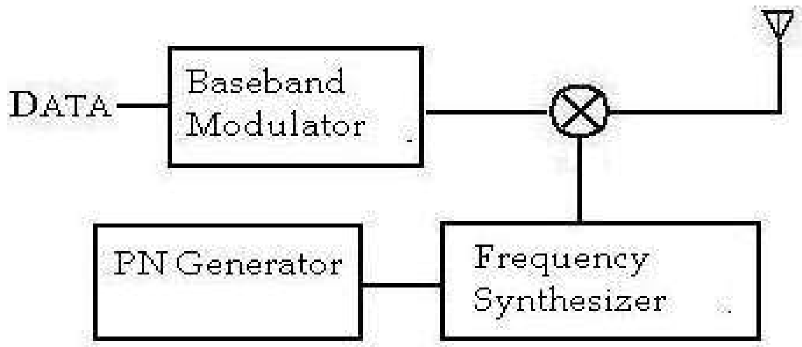
FREQUENCY HOPPED SPREAD SPECTRUM (FH-SS)-

In FH-SS, users are made to change the frequencies from one to another in a specified time interval, hence called as frequency hopping. The frequency of the data are hopped from one to another in order to provide a secure transmission.

The amount of time spent on each frequency hop is called as Dwell time. As per the change of code the carrier frequency will change. Hopping classified into two types i.e, slow hopping and fast hopping.

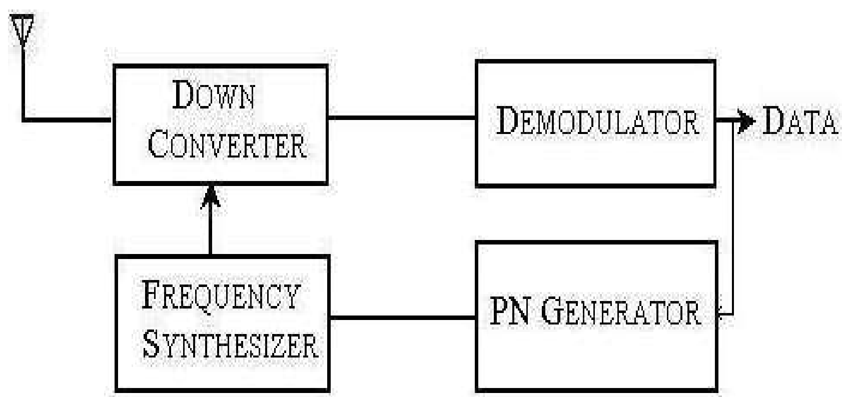
In fast hopping $T_b > T_c$ and in slow hopping $T_b < T_c$. Where T_b is the bit period and T_c is the chip period. We always prefer fast hopping.

FH-SS Transmitter-



The transmitter consists of a base band modulator followed by frequency synthesizer. The frequency synthesizer is driven by a PN generator. A PN generator may be built internal to the synthesizer. The synthesizer hopping different frequencies at different point of time.

FH-SS Receiver-



A FH-SS receiver consists of a down converter followed by a demodulator. A synthesizer driven by a matching PN generator is used to down convert the received signals.

<u>6.7 DEFINATION</u>

<u>BAUD:</u>- In telecommunication and electronics, **baud** (unit symbol Bd) is the unit for symbol rate or modulation rate in symbols per second or pulses per second. It is the number of distinct symbol changes (signaling events) made to the transmission medium per second in a digitally modulated signal or a line code.

<u>BIT: -</u> A **bit** is the basic unit of information in computing and digital communication. A bit can have only one of two values, and may therefore be physically implemented with a two-state device. These values are most commonly represented as either a 0or1. The term *bit* is a called of **binary digit.**

SYMBOL: - A **symbol** is an object that represents, stands for, or suggests an idea, visual image, belief, action, or material entity. Symbols take the form of words, sounds, gestures, or visual images and are used to convey ideas and beliefs.

SHANNON THEOREM:

Shannon showed that error-free communication is possible on a noisy channel provided that the data rate is less than the channel capacity. Shannon capacity (data rate) equation is the basis for spread spectrum systems, which typically operate at a very low SNR, but use a very large bandwidth in order to provide an acceptable data rate per user.

CHANNEL CAPACITY "C":

Channel capacity "C" (error free bps) is directly proportional to the bandwidth "B" and is proportional to the log of SNR.

$$C = B \times \log_2(1+S/N)$$

Where
C is the channel capacity in bits per second (bps), which is the maximum data rate for a theoretical bit error rate (BER)
B is the required bandwidth in Hz
S/N is the signal to noise ratio
[Note: C which represents the amount of information allowed by communication channel, also represent the desired performance. S/N ratio expresses the environmental conditions such as obstacles, presence of jammers, interferences, etc.]

In Shannon formula by changing the log base from 2 to e (the Napierian number) and noting that e ln = log Therefore:

$$C/B = (1 / \ln 2) \times \ln(1 + S/N) = 1.443 \times \ln(1 + S/N)$$

6.8 APPLICATIONS OF DIFFERENT MODULATION SCHEME-

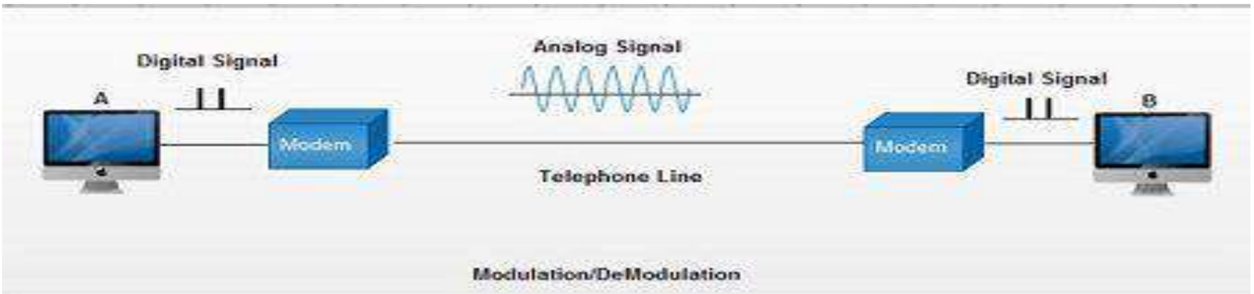
Some of the important applications of digital modulation techniques are-

1. Voice grade MODEMS
2. Digital Radio
3. Digital communications by satellite

6.9 TYPES OF MODEM & ITS APPLICATION

MODEM-

Modem is the combination of modulator- demodulator. Modem is used for transmitting and receiving data over a communication channel such as twisted pair, telephone lines, coaxial cables and fiber optics. Modem convert the digital data to analog format, so that it can be transmitted over the analog telephone line.



At the source, modulation techniques are used to convert digital data into analog form for transmission across the channel. At the destination, the received analog signal is converted to digital data using demodulation.

Modems can be of several types and they can be categorized in a number of ways.

• Categorization is usually based on the following basic modem features:

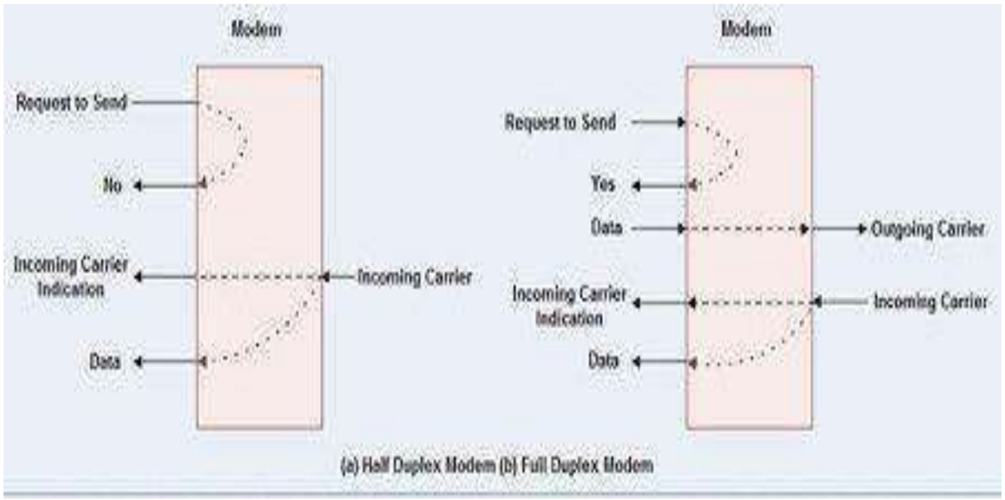
1. Directional capacity: half duplex modem and full duplex modem.
2. Connection to the line: 2-wire modem and 4-wire modem.
3. Transmission mode: asynchronous modem and synchronous modem.
4. Place of connection: External Modem, Internal Modem and Wireless Modem

Half duplex modem-

- A half-duplex modem permits transmission in one direction at a time. If a carrier is detected on the line by the modem, it gives an indication of the incoming carrier to the Data Terminal Equipment (DTE), it can be a terminal or a computer, through a control signal of its digital interface.
- As long as the carrier is not received, the modem does not give permission to the Data Terminal Equipment to transmit data.

Full duplex modem-

- A full duplex modem allows simultaneous transmission in both direction. Therefore, there are two carriers on the line, one outgoing and the other incoming.



2-wire modem-

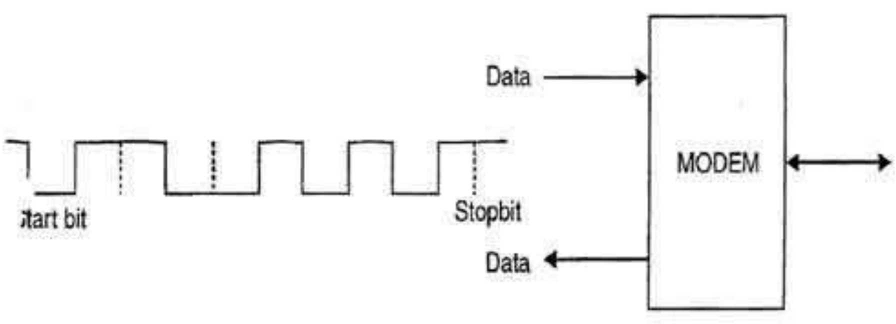
- 2 wire modems use the same pair of wires for outgoing and incoming carriers.
- A 2 wire connection is usually cheaper than a 4 wire connection as only one pair of wires is used.
- The data connection established through telephone exchange is also a 2 wire connection.

4-wire modem-

- In a 4 wire connection, one pair of wires is used for the outgoing carrier and the other pair is used for incoming carrier

Asynchronous modem-

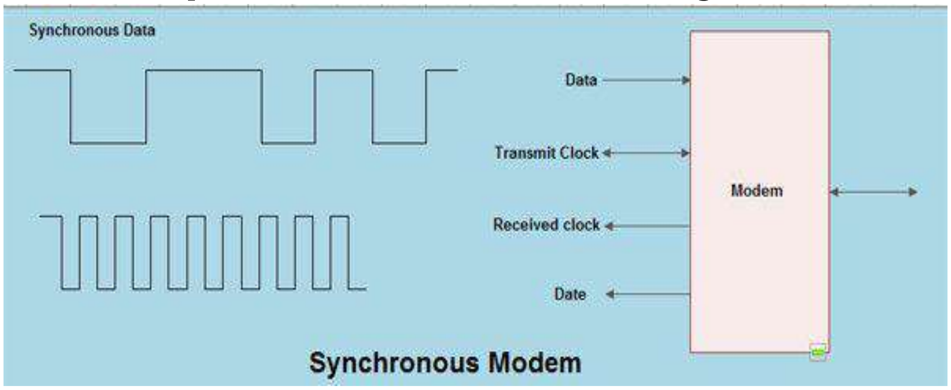
- Asynchronous modems can handle data bytes with start and stop bits.
- There is no separate timing signal or clock between the modem and the Data Terminal Equipment (DTE).
- The internal timing pulses are synchronized repeatedly to the leading edge of the start pulse.



Asynchronous modem

Synchronous modem-

- Synchronous modems can handle a continuous stream of data bits but requires a clock pulse.
- The data bits are always synchronized to the clock signal.
- There are separate clocks for the data bits being transmitted and received.



External Modem-

- An external modem is a removable device which is used for communication purpose. This type of modem is externally connected with a computer with a common port or USB port and has some lights which indicate the processing status.
- The external modem may be dial up modem or broadband modems.
- It is connected to the telephone wall jack by another cable. External power is supplied to it. It is very easy to setup.

Internal modem-

- An internal modem is a type of modem that is installed on the slots of the mother board. This modem is an expansion card.
- An internal modem is a circuit board that can be attached inside the system through expansion slots.
- It cannot be moved easily from one PC to another. Its set up is difficult.

Wireless modem-

- The wireless modem transmits data through the air instead of cable. It is also called a radio frequency modem.
- These are designed to work with cellular technology and wireless LAN.

Application-

- Internet Access- faster downloads, efficient use of graphics and audio intensive applications, multimedia application, interactive games etc.
- Computer Networking- local area network, wide area network, intranets, extranets.
- Entertainment- Music on demand, video on demand, HDTV

- Home Shopping- Online shopping facilities, Virtual auctions
- Remote/Distance Learning- Virtual classrooms, Virtual laboratories.
- Medical Purposes- Immediate access to patients or health care facilities, transmission of medical images.
- Remote Professional Services- Legal services, health care and real estate.
- Telecommuting- video conferencing, interactive meeting and collaboration.