Analog and Digital Communication

Explain the radio frequency spectrum used in communication system

In the radio communication system, the frequencies ranging from a few kilohertz to many gigahertz all are being used for various purposes.

Let us discuss the applications of various frequency bands.

The frequencies most commonly used in early days were from about 300 kHz to 3 MHz and were called as medium frequencies(MF).

The frequencies in the range 30 kHz to 300 kHz are known as the low frequencies (LF).

The frequencies in the range 3 kHz to 30 kHz are called as very low frequencies (VLF).

On the higher frequency side, high frequencies (HF) will cover the frequency range from 3 MHz to 30 MHz.

Then very high frequency (VHF) from 30 MHz TO 300 MHz and so on .

Following table presents the details of entire usable frequency spectrum and its applications.

S.No	Frequency Band	Wavelength	Applications
1.	30 Hz – 300 Hz. Extremely low frequencies(ELF)	10^4 km to 10^3 km	Power transmission
2.	300 Hz – 3 kHz. Voice frequencies (VF)	10^3 km to 100 km	Audio applications
3.	3 kHz – 30 kHz. Very low frequencies (VLF)	100 km to 10 km	Submarine communications. Navy, Military communications
4.	30 kHz – 300 kHz . Low frequencies (LF)	10 km to 1 km. Long waves.	Aeronautical and marine navigation, these frequencies act as sub carriers.
5.	300 kHz – 30 MHz. Medium frequencies (MF)	1 km to 100 m. Medium waves	AM radio broadcast, Marine and aeronautical communications
6.	3 MHz – 30 MHz. High Frequencies (HF)	100 m TO 10 m. Short waves	Shortwave transmission, Amateur and CB communication.
7.	30 MHz – 300 MHz. Very high frequencies (VHF)	10 m to 1 m	T.V. broadcasting. F.M. broadcasting

7.	30 MHz – 300 MHz. Very high frequencies (VHF)	10 m to 1 m	T.V. broadcasting. F.M. broadcasting
8.	300 MHz – 3 GHz. Ultra high frequencies (UHF)	1 m to 10 cm. Microwaves	UHF T.V. Channels, Cellular phones, Military applications
9.	3 GHz – 30 GHz (SHF)	10-1 m to 10-2 m	Satellite communications and Radar
10.	30 -300 GHz (EHF)	10-2 m to 10-3 m	Satellites and specialized Radars.

The radio frequency (RF) spectrum

Explain the Difference Between Analog and Digital Communications

The difference between analog and digital communication system is explained in the table below :

Analog Communication	Digital Communication
Transmitted modulated signal is analog in nature.	Transmitted signal is digital i.e. train of digital pulses.
Amplitude, frequency or phase variations in the transmitted signal represent the information or message.	Amplitude, width or position of the transmitted pulses is constant. The message is transmitted in the form of code words.
Noise immunity is poor for AM, but improved for FM and PM.	Noise immunity is excellent.
It is not possible to separate out noise and signal. Therefore, repeaters cannot be used.	It is possible to separate signal from noise. Therefore, repeaters can be used.
Coding is not possible.	Coding techniques can be used to detect and correct the errors.
Bandwidth required is lower than that for the digital modulation method.	Due to higher bit rates, higher channel bandwidth is required.
FDM is used for multiplexing.	TDM is used for multiplexing.
Not suitable for transmission of secret information in military applications.	Due to coding techniques, it is suitable for military applications.
Analog modulation systems are AM, FM, PM, PAM, AWM, etc.	Digital modulation systems are PCM, DM, ADM, DPCM, etc.

Explain the need for modulation in a communication system

Modulation

In the modulation process, two signals are used namely the modulating signal and the carrier .

The modulating signal is nothing but the baseband signal or information signal while the carrier is a high frequency sinusoidal signal .

In the modulation process, some parameter of the carrier wave (such as amplitude, frequency or phase) is varied in accordance with the modulating signal . This modulated signal is then transmitted by the transmitter .

The receiver demodultes the received modulated signal and gets the original information signal back .

Thus, demodulation is exactly opposite to modulation .

In the process of modulation the carrier wave actually acts as carrier which carries the information signal from the transmitter to receiver .

Need of Modulation

You may be ask, when the baseband signal can be transmitted directly why to use the modulation ?

The answer is that the baseband transmission has many limitations which can be overcome using modulation . It is explained below .

In the process of modulation, the baseband signal is translated i.e., shifted from low frequency to high frequency . This frequency shift is proportional to the frequency of carrier.

Advantages of Modulation

1. Reduction in the height of antenna
2. Avoids mixing of signals
3. Increases the range of communication
4. Multiplexing is possible
5. Improves quality of reception

We will discuss each of these advantages in detail below .

1. Reduction in the height of antenna

For the transmission of radio signals, the antenna height must be multiple of $\lambda/4$,where λ is the wavelength .
$\lambda = c/f$

where c : is the velocity of light

f: is the frequency of the signal to be transmitted

The minimum antenna height required to transmit a baseband signal of f = 10 kHz is calculated as follows :

The antenna of this height is practically impossible to install .

$$Minimum\ antenna\ height = \frac{\lambda}{4} = \frac{c}{4f} = \frac{3 \times 10^8}{4 \times 10 \times 10^3} = 7500\ meters\ i.e.\ 7.5\ km$$

Now, let us consider a modulated signal at f = 1 MHz . The minimum antenna height is given by,

$$Minimum\ antenna\ height = \frac{\lambda}{4} = \frac{c}{4f} = \frac{3 \times 10^8}{4 \times 10 \times 10^6} = 75\ meters$$

This antenna can be easily installed practically . Thus, modulation reduces the height of the antenna .

2. Avoids mixing of signals

If the baseband sound signals are transmitted without using the modulation by more than one transmitter, then all the signals will be in the same frequency range i.e. 0 to 20 kHz . Therefore, all the signals get mixed together and a receiver can not separate them from each other .

Hence, if each baseband sound signal is used to modulate a different carrier then they will occupy different slots in the frequency domain (different channels). Thus, modulation avoids mixing of signals .

3. Increase the Range of Communication

The frequency of baseband signal is low, and the low frequency signals can not travel long distance when they are transmitted . They get heavily attenuated .

The attenuation reduces with increase in frequency of the transmitted signal, and they travel longer distance

The modulation process increases the frequency of the signal to be transmitted . Therefore, it increases the range of communication.

4. Multiplexing is possible

Multiplexing is a process in which two or more signals can be transmitted over the same communication channel simultaneously .

This is possible only with modulation.

The multiplexing allows the same channel to be used by many signals . Hence, many TV channels can use the same frequency range, without getting mixed with each other or different frequency signals can be transmitted at the same time .

5. Improves Quality of Reception

With frequency modulation (FM) and the digital communication techniques such as PCM, the effect of noise is reduced to a great extent . This improves quality of reception .

Block Diagram of Communication System with Detailed Explanation

Communication System

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

OR

Communication is simply the basic process of exchanging information.

The electronics equipments which are used for communication purpose, are called communication equipments. Different communication equipments when assembled together form a communication system. Typical example of communication system are line telephony and line telegraphy, radio telephony and radio telegraphy, radio broadcasting, point-to-point communication and mobile communication, computer communication, radar communication, television broadcasting, radio telemetry, radio aids to navigation, radio aids to aircraft landing etc.

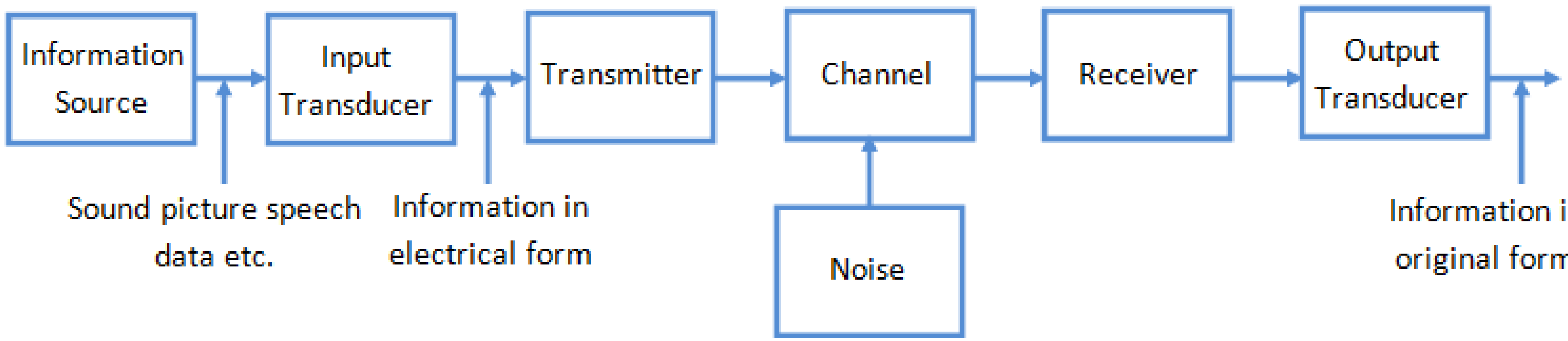
The Communication Process

In the most fundamental sense, communication involves the transmission of information from one point to another through a succession of process as listed below :

1. The generation of a thought pattern or image in the mind of an originator.
2. The description of that image, with a certain measure of precision, by a set of oral visual symbols.
3. The encoding of these symbols in a form that is suitable for transmission over a physical medium of interest.
4. The transmission of the encoded symbols to the desired destination.
5. The decoding and reproduction of the original symbols.
6. The recreation of the original thought pattern or image, with a definable degradation in quality, in the mind of recipient.

Block Diagram of Communication System

Fig. shows the block diagram of a general communication system, in which the different functional elements are represented by blocks.



Fig

The essential components of a communication system are information source, input transducer, transmitter, communication channel, receiver and destination.
Now, we shall discuss the functioning of these blocks.

(i) Information Source

As we know, a communication system serves to communicate a message or information. This information originates in the information source.

In general, there can be various messages in the form of words, group of words, code, symbols, sound signal etc. However, out of these messages, only the desired message is selected and communicated.

Therefore, we can say that the function of information source is to produce required message which has to be transmitted.

(ii) 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. In a case when the message produced by the information source is not electrical in nature, an input transducer is used to convert it into a time-varying electrical signal.

For example, in case of radio-broadcasting, a microphone converts the information or massage which is in the form of sound waves into corresponding electrical signal.

(iii) Transmitter

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

For example in radio broadcasting the electrical signal obtained from sound signal, is processed to restrict its range of audio frequencies (up to 5 kHz in amplitude modulation radio broadcast) and is often amplified.

In wire telephony, no real processing is needed. However, in long-distance radio communication, signal amplification is necessary before modulation.

Modulation is the main function of the transmitter. In modulation, the message signal is superimposed upon the high-frequency carrier signal.

In short, we can say that inside the transmitter, signal processing such as restriction of range of audio frequencies, amplification and modulation of are achieved.

All these processing of the message signal are done just to ease the transmission of the signal through the channel.

(iv) The Channel and the Noise

The term channel means the medium through which the message travels from the transmitter to the receiver. In other words, we can say that the function of the channel is to provide a physical connection between the transmitter and the receiver.

There are two types of channels, namely point-to-point channels and broadcast channels.

Example 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.

In case of microwave links, the transmitted signal is radiated as an electromagnetic wave in free space. Microwave links are used in long distance telephone transmission.

An optical fibre is a low-loss, well-controlled, guided optical medium. Optical fibres are used in optical communications.

Although these three channels operate differently, they all provide a physical medium for the transmission of signals from one point to another point. Therefore, for these channels, the term point-to-point is used.

On the other hand, the broadcast channel provides a capability where several receiving stations can be reached simultaneously from a single transmitter.

An example of a broadcast channel is a satellite in geostationary orbit, which covers about one third of the earth's surface.

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

(v) 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 or detection. Demodulation is the reverse process of modulation carried out in transmitter.

(vi) Destination

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

AMPLITUDE MODULATION

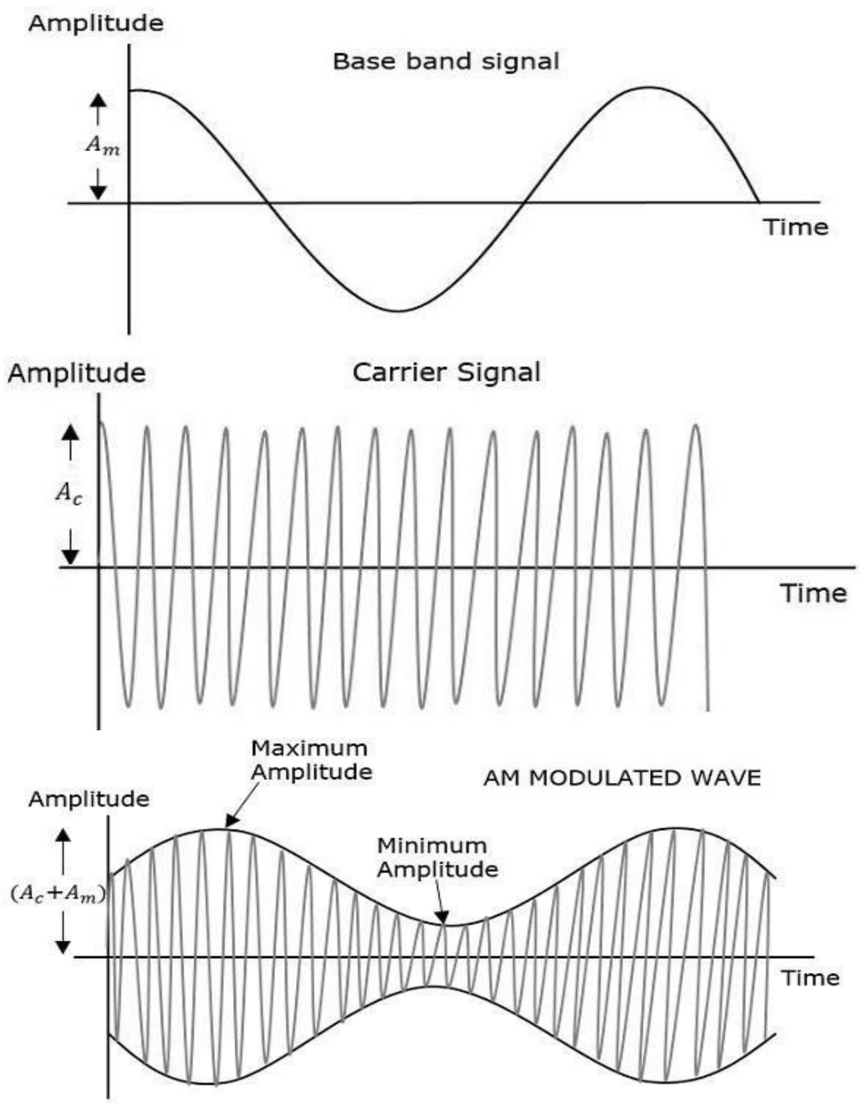
AMPLITUDE MODULATION & DERIVE THE EXPRESSION FOR AMPLITUDE MODULATION SIGNAL, POWER RELATION IN AM WAVE & FIND MODULATION INDEX

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)= $A_m Cosw_m t$

Carrier Signal, C(t)= $A_c Cosw_c t$

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

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

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

$= {}_c(1 + m_a\, Cosw_m t)sw_c t \qquad (\frac{A_m}{A_c} = m_a)$

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

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

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

Bandwidth-

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

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

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

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

$= w_c + w_m - w_c + w_m$

$= 2w_m$

<u>MODULATION INDEX</u>

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

$=A_c + A_m - A_c + A_m$

$= 2A_m$

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

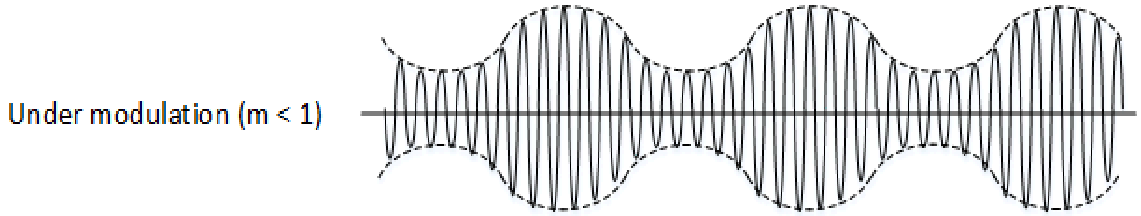
=> $A_m = \frac{A_{max} - A_{min}}{2}$

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

There are three cases of m_a.

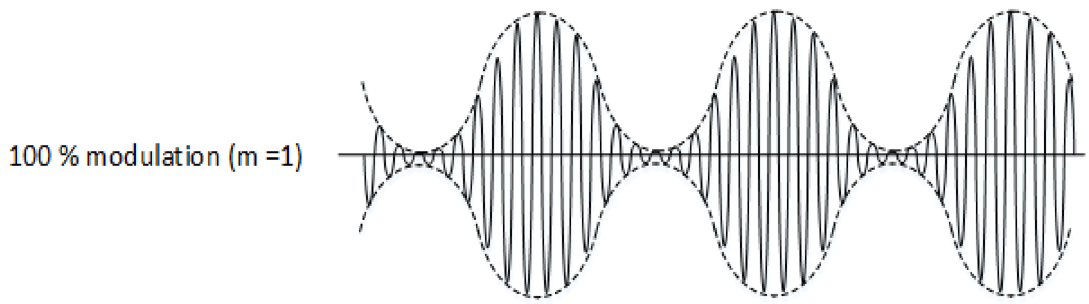
<u>Case-I</u>

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



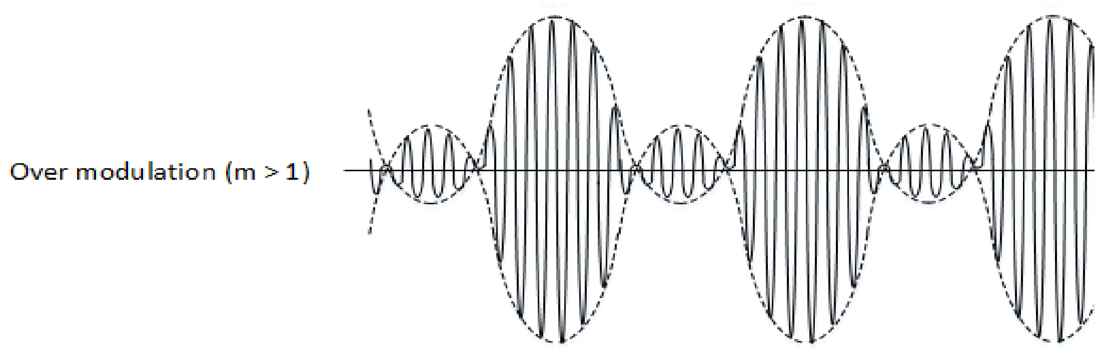
Case-II

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



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 signalof 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:-

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

$= P_c + P_s$

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

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

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

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

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

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

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

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

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

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

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

$P_t = P_c + P_s$

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

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

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

Current Relation-

P= $I^2 t$

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

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

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

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

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.

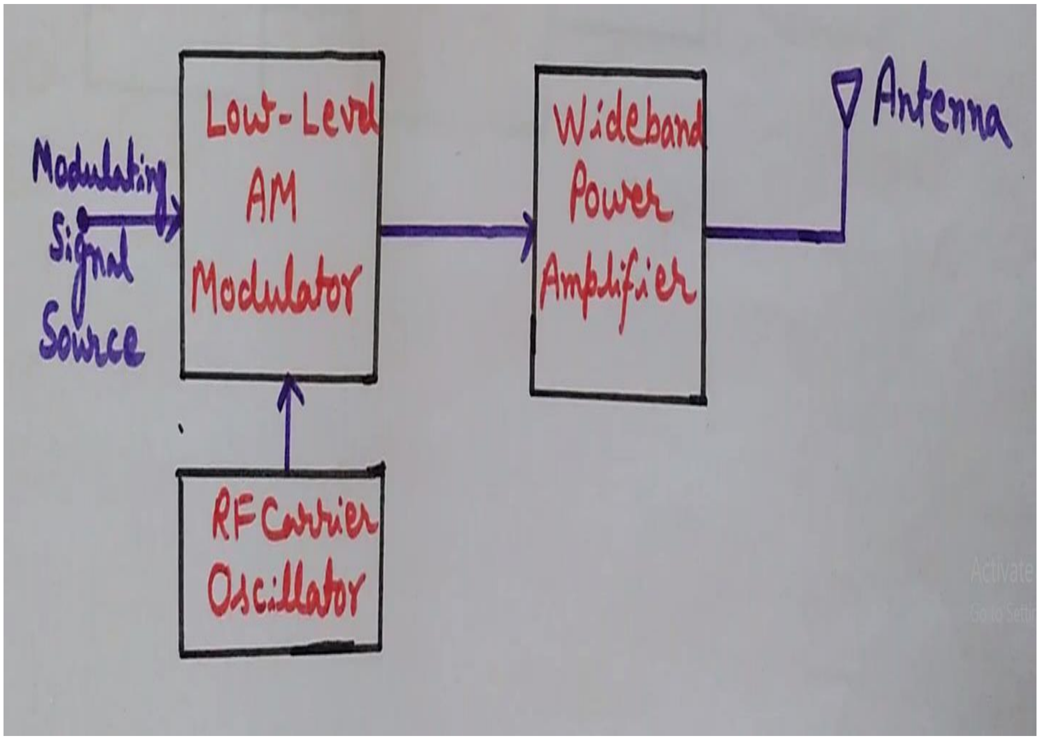
GENERATION OF AM WAVES-

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:-

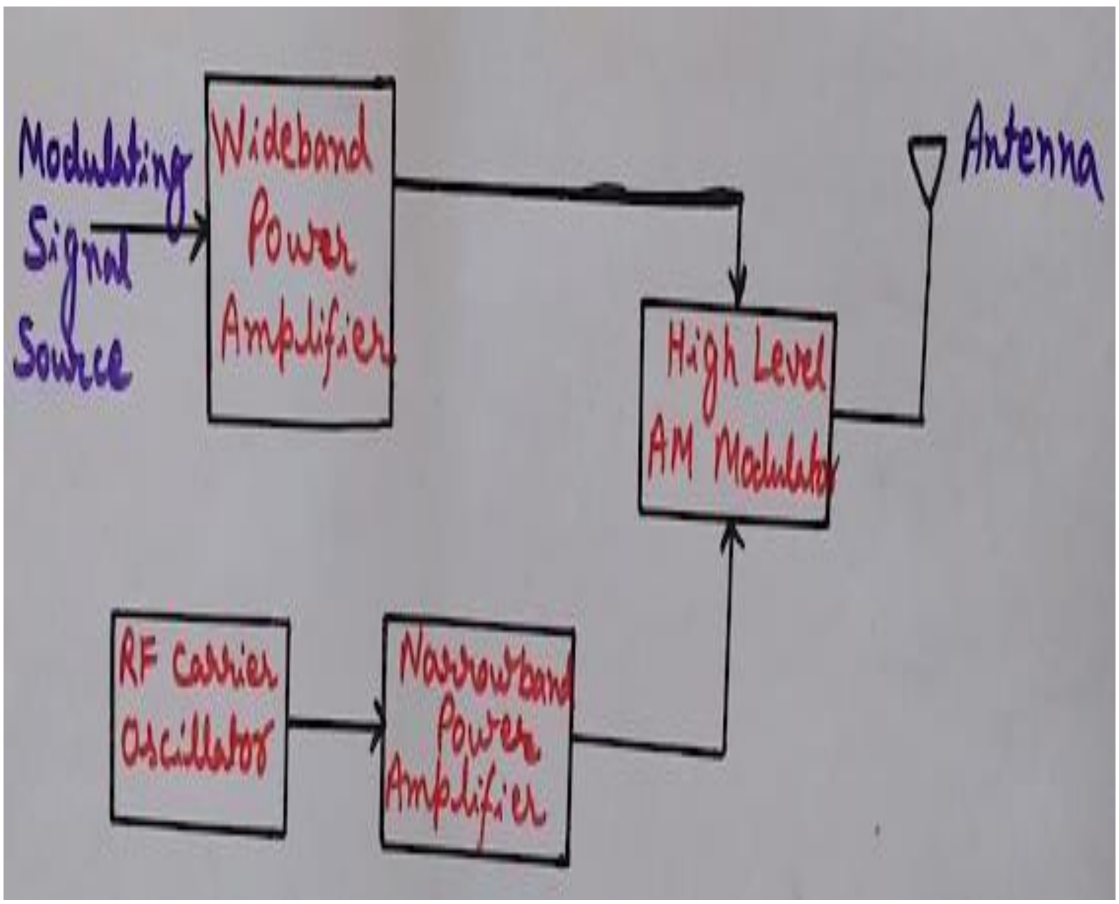
In a low level amplitude modulation system, the modulation is done at low power level. At low power levels, a very small power is associated with the carrier signal and the modulation signal. Because of this the output power of modulation is low. Therefore the power amplifiers are required to boost the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

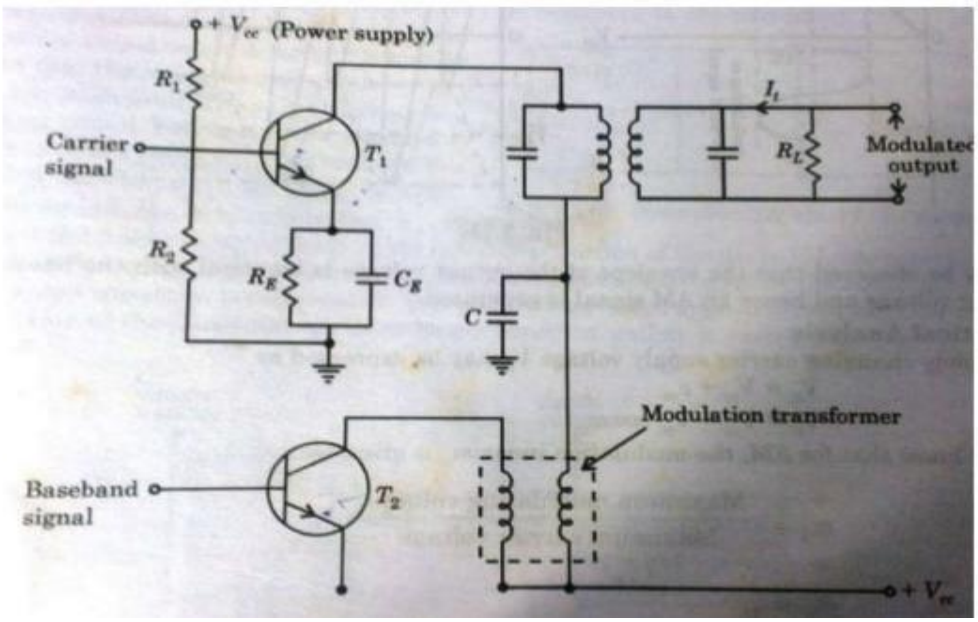
COLLECTOR MODULATION (LINEAR LEVEL AM MODULATION)-

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 voltage V_{cc}.

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

Where $V_m = V_m cos v_m t$

V_{cc}= amplitude of the carrier signal

Carrier signal represented as

$V_c = V_{cc} cos v_c t$

$The\ modulated\ signal\ is$

$V_o = (V_{cc} + V_m cos v_m t) cos v_c t$

$=> V_o = V_{cc} (1 + \frac{V_m}{V_{CC}} cos v_m t)\ cos v_c t$

$=> V_o = (1 + m_a cos v_m t) cos v_c t$

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

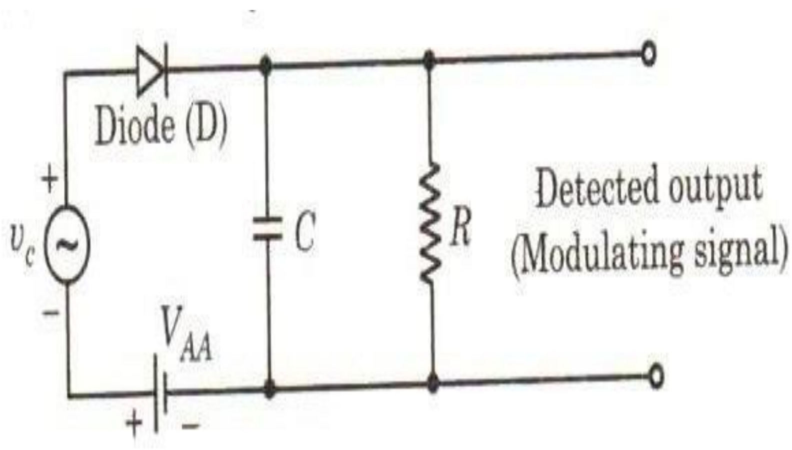
(3) PLL AM detector

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.

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

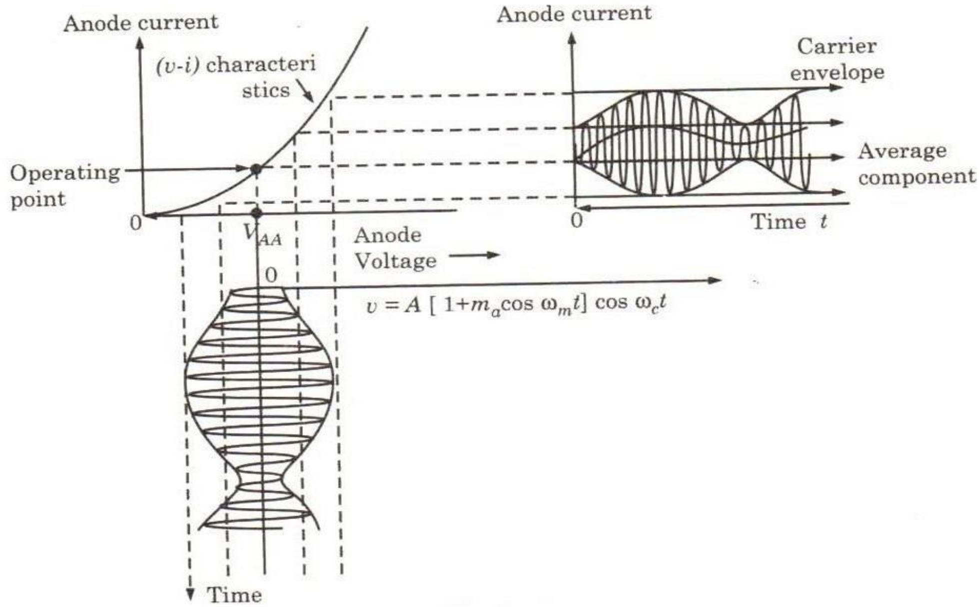
$v=A\ (1+m_a \cos \omega_m t) \cos \omega_c t$

Substituting, the value of v, we get

$I = a\ [A\ (1+m_a \cos \omega_m t) \cos \omega_c t] + b\ [A\ (1+m_a \cos \omega_m t) \cos \omega_c t]\ 2$

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.

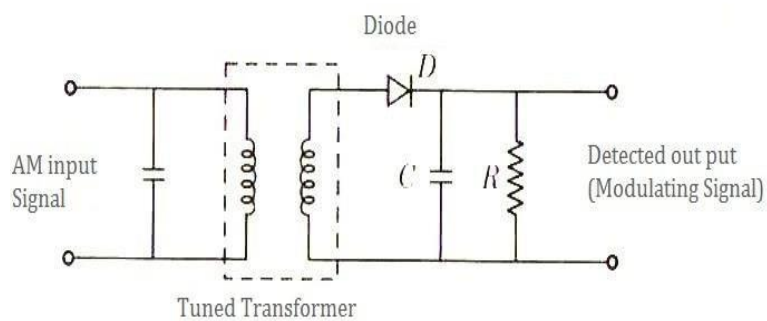
Hence this diode current I containing all these frequencies terms is passed through a low pass filter, which allows to pass the frequency below or up to modulating frequency ω_m and rejects the other higher frequency components. Therefore, the modulating signal with frequency ω_m is recovered from the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

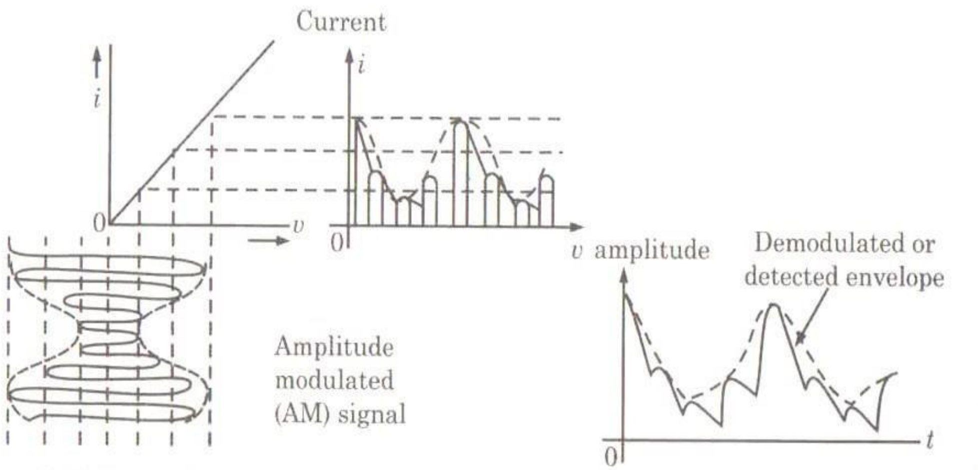


A Linear Diode Detecter

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
r = 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.

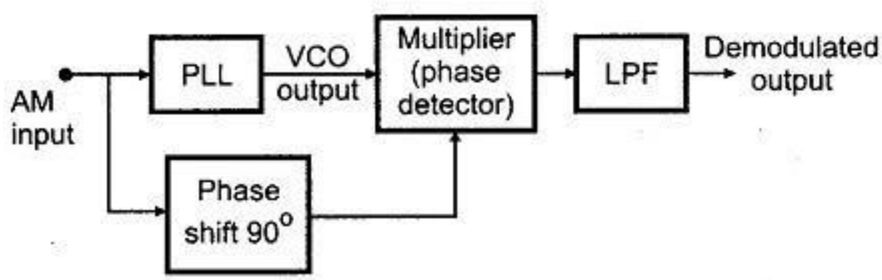


(a) Characteristics of linear diode detector

(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.

DSB-SC SIGNAL AND SSB

SIGNALDSB-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,

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

$Put\ m_a = 1$

$P_t = (1 + 1/2)$

=> $P_t = 3/2\, P_c$

=> $P_c = 2/3\, P_t$

=> $P_c = 0.67 P_t$

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

METHODS OF GENERATING & DETECTION OF SSB-SC SIGNALSSB-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.

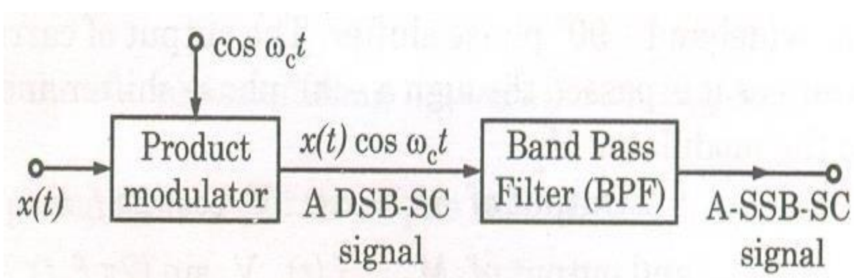
Generation-

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

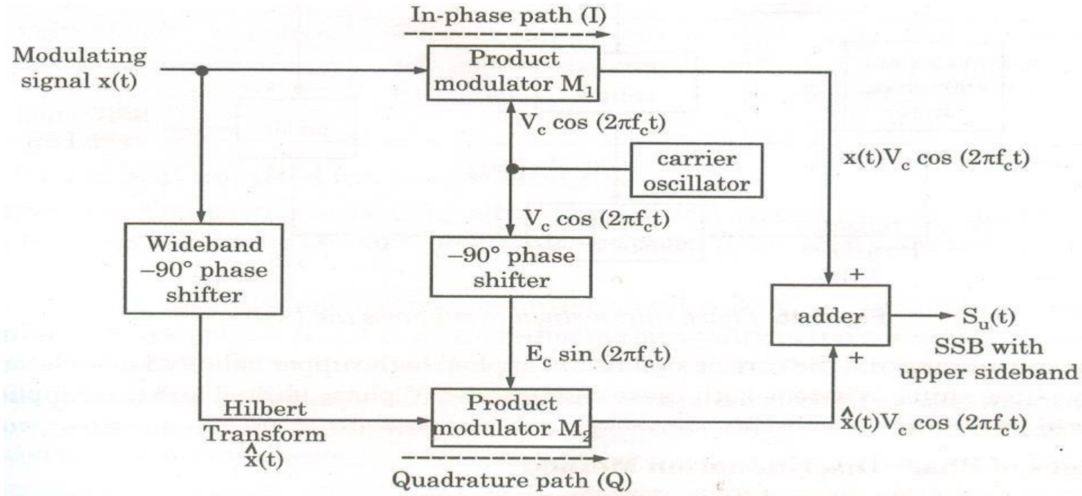
Limitations-

- 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.

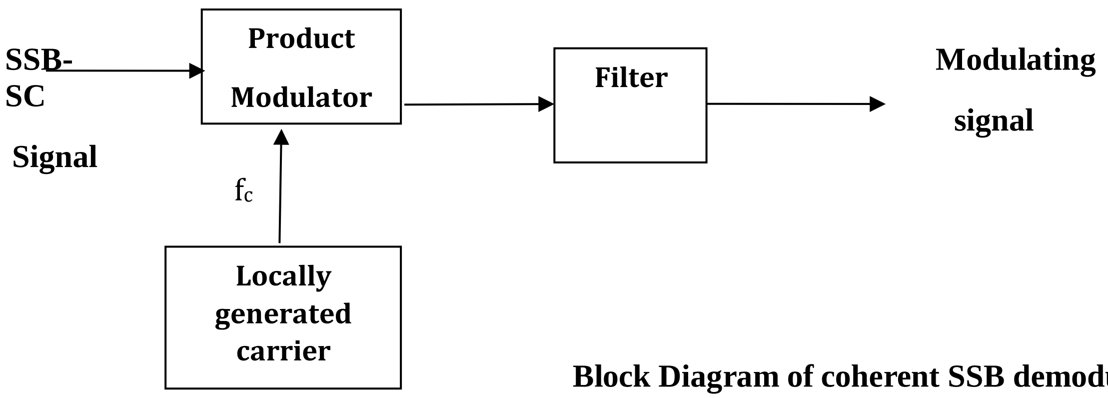
$$e_d(t) = S(t)_{SSB} \times \cos\omega ct$$

$$= [x(t) \cos\omega_c t \pm x_n(t) \sin\omega_c t] \cos\omega_c t$$

$= 1/2\ x(t) + ½\ [x(t)\ Cos2\omega_c t \pm x_n(t)\ Sin2\omega_c t]$

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 $e_o(t)$

$= 1/2\ x(t)$



Block Diagram of coherent SSB demodulator

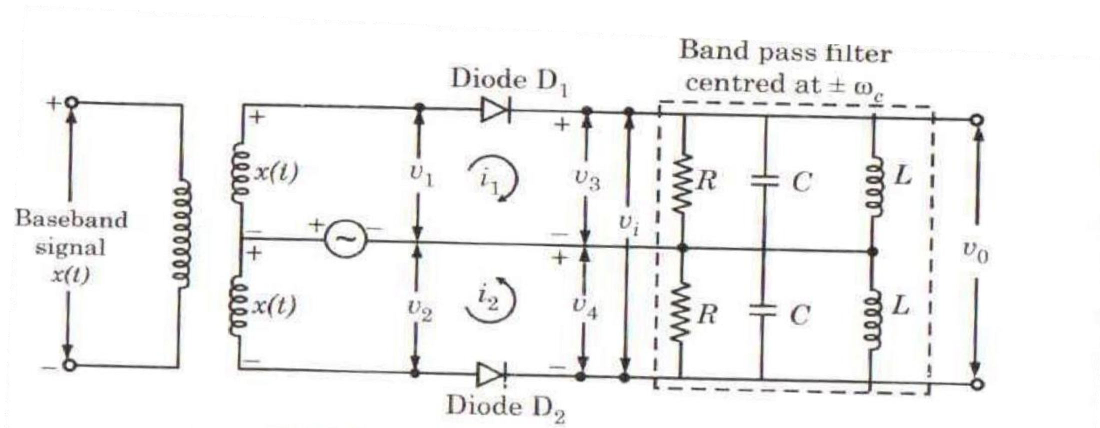
<u>METHODS OF GENERATION OF DSB-SC SIGNAL</u><u>GENERATION OF DSB-SC SIGNAL</u>-

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

<u>Balance Modulator :-</u>

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 withthe carrier signal Cos $\omega_c t$.

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 isthe 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 filterexpressed as $v_i = i_1R - i_2R$.

After substituting the values of i_1 & i_2 we get

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

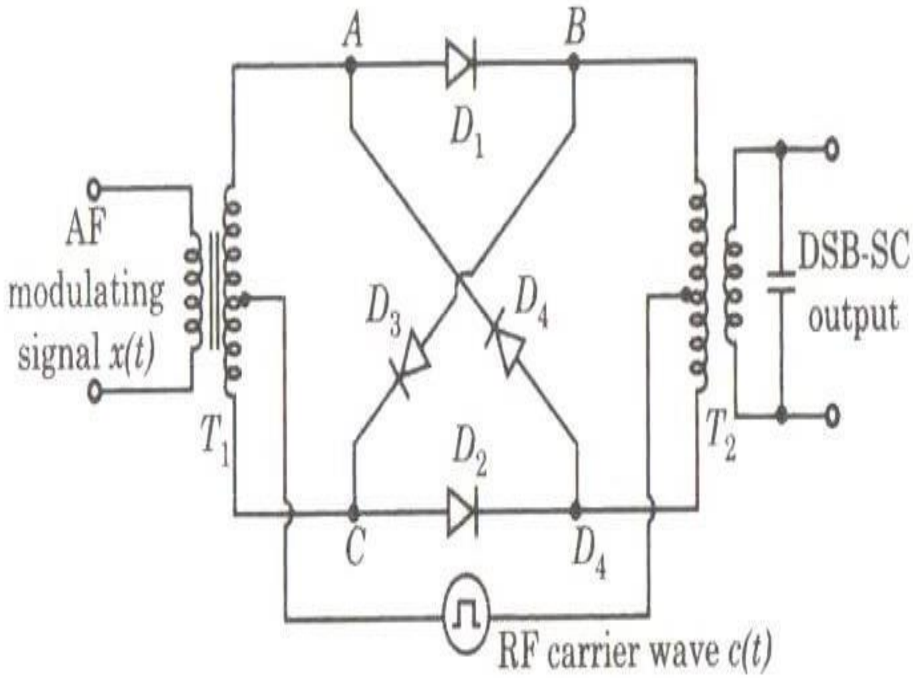
A band pass filter is that type of filter which allows to pass a band of frequencies. Here the band pass filter is centred around ± , it will pass a narrow bandof frequencies cantered at $\pm\omega_c$.

The output of the BPF is

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

<u>Ring Modulator-</u>

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 squarewave carrier signal c(t) of frequency f_c applied through a centre tapped transformer.



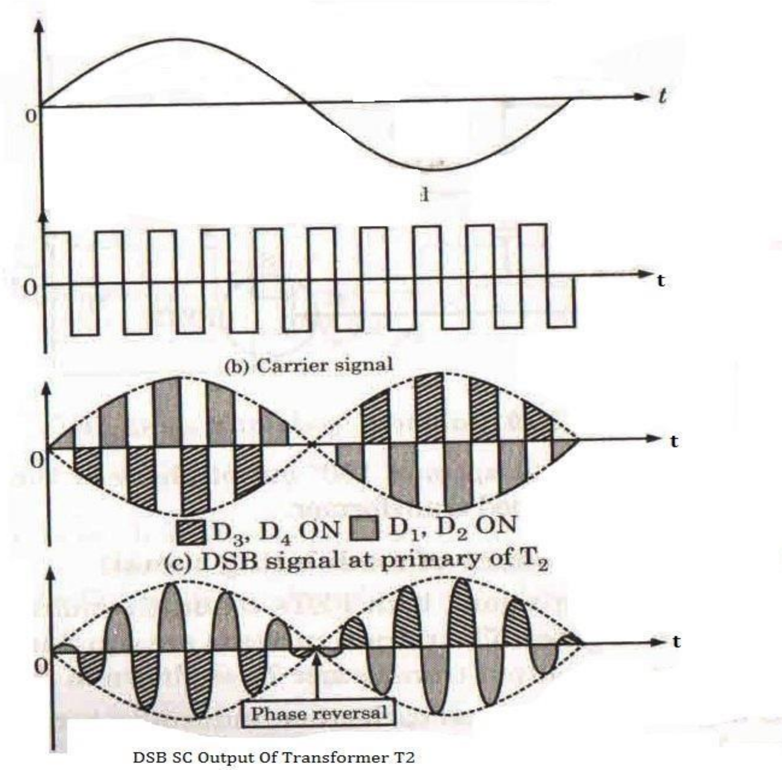
A diode ring modulator.

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 impedance. 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 $-f_m \leq f \leq$. 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 .

DETECTION OF DSB-SC SIGNAL-

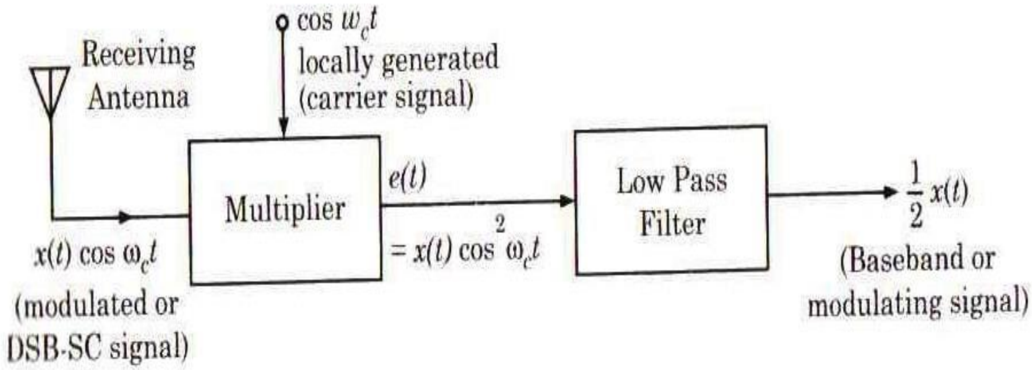
The DSB-SC signal may be demodulated by following two methods.

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) isrecovered from the modulated signal. This can be achieved by simply retranslating the baseband or modulating signal from a higher spectrum, cantered at $\pm\omega c$, 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.

<u>Working principle</u>-

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

<u>VESTIGIAL SIDE BAND MODULATION:</u>

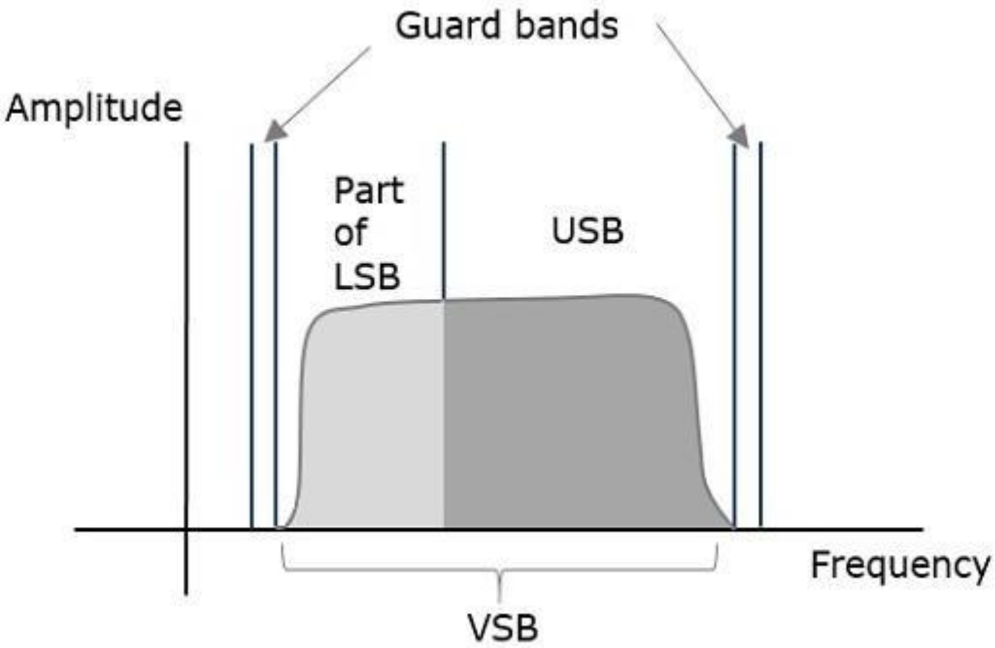
In case of SSB modulation, when a sideband is passed through the filters, the band pass filter may not work perfectly in practice. As a result of which, some of the information mayget lost.

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 plottedas 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 −

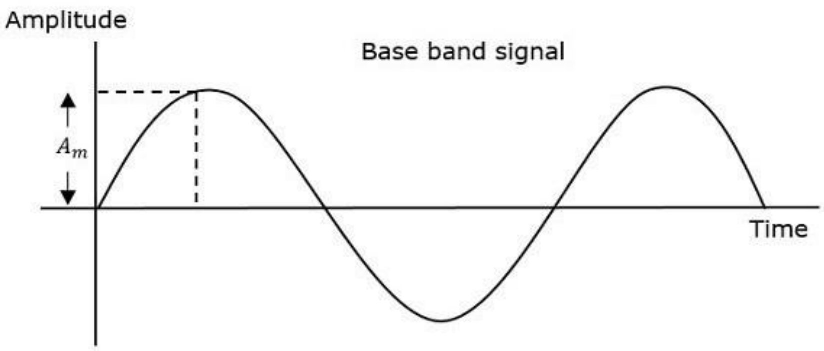
- Bandwidth when compared to SSB is greater.
- Demodulation is complex.

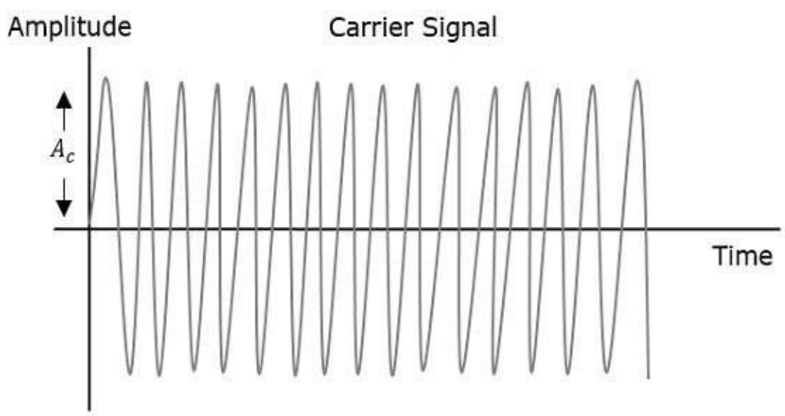
BASIC PRINCIPLE OF FREQUENCY MODULATION & FREQUENCY SPECTRUM OF FM SIGNAL

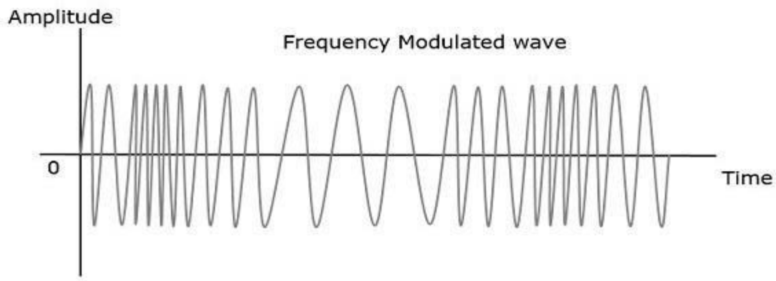
FREQUENCY MODULATION

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

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

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

We have

$$\cos(\omega_c t + \beta \sin \omega_m) = \cos \omega_c t \, cos\, (\beta \sin \omega_m t) - \sin \omega_c t \, sin\, (\beta \sin \omega_m t) \ldots\ldots\ldots\ldots (2)$$

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

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

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

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

The functions (β) occur often in the solution of engineering problem. They areknown 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

$$(t) = J_0(\beta)\cos\omega_c - J_1(\beta)[cos(\omega_c - \omega_m)t - \cos(\omega_c + \omega_m)\,t]$$

$$+ J_2(\beta)[cos(\omega_c - 2\omega_m)t + \cos(\omega_c + 2\omega_m)\,t]$$

$$- J_3(\beta)[cos(\omega_c - 3\omega_m)t - \cos(\omega_c + 3\omega_m)\,t] + \cdots \ldots$$

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.

EXPRESSION FOR FREQUENCY MODULATED SIGNAL & MODULATION INDEXEXPRESSION 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 (t) is

$$w_i = \frac{d\theta_i(t)}{dt}$$

□ $2\pi f_i = \frac{d\theta_i(t)}{dt}$

□ $(t) = 2\pi\int f_i\ dt$

Substitute f_i value in the equation

$(t) = 2\pi\int(f_c + K_f m(t))\ dt$

□ $(t) = 2\pi f_c t + 2\pi K_f \int(t)\ dt$

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

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

If the modulating signal is m(t)= $A_m\cos(2\pi f_m t)$, then the equation of FM wave will beS(t)= $A_c Cos(2\pi f_c t + 2\pi K_f \int A_m\, cos\,(2\pi f_m t\,)\, dt$

□ S(t)= $A_c(2\pi f_c t + 2\pi K_f \frac{1}{2\pi f_m} A_m\, Sin\,(2\pi f_m t\,))$

□ 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 istermed as frequency deviation. It is denoted by Δf.

$$\Delta f = k_f A_m$$

<u>MODULATION INDEX</u>-

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

Modulation index, β = frequency deviation/modulation frequency

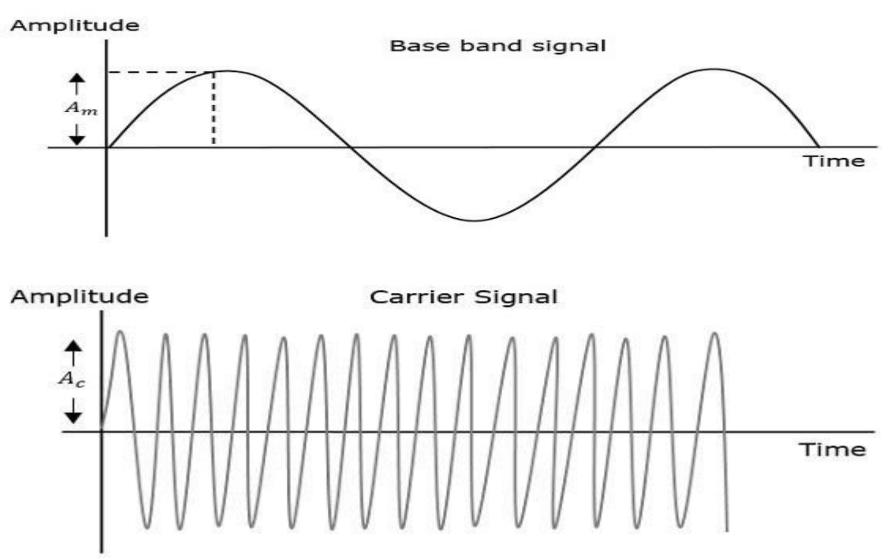
Or $\beta = \frac{\Delta f}{f_m}$

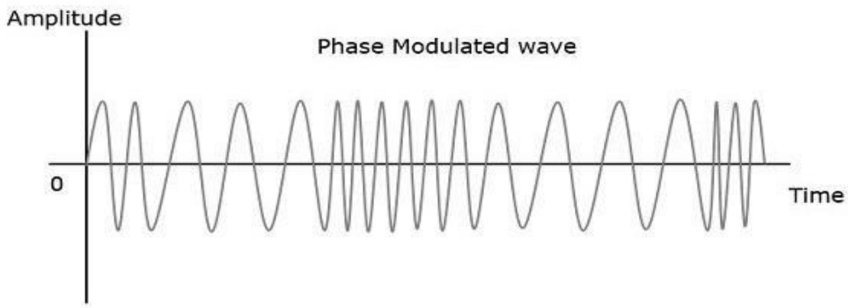
This modulation index may be greater than unity.

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

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 phasemodulating 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

DIFFERENCE OF FM & PM-

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.

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 = 2 . 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 iscovered 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[Δ_f +]. 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.

FM GENERATION-

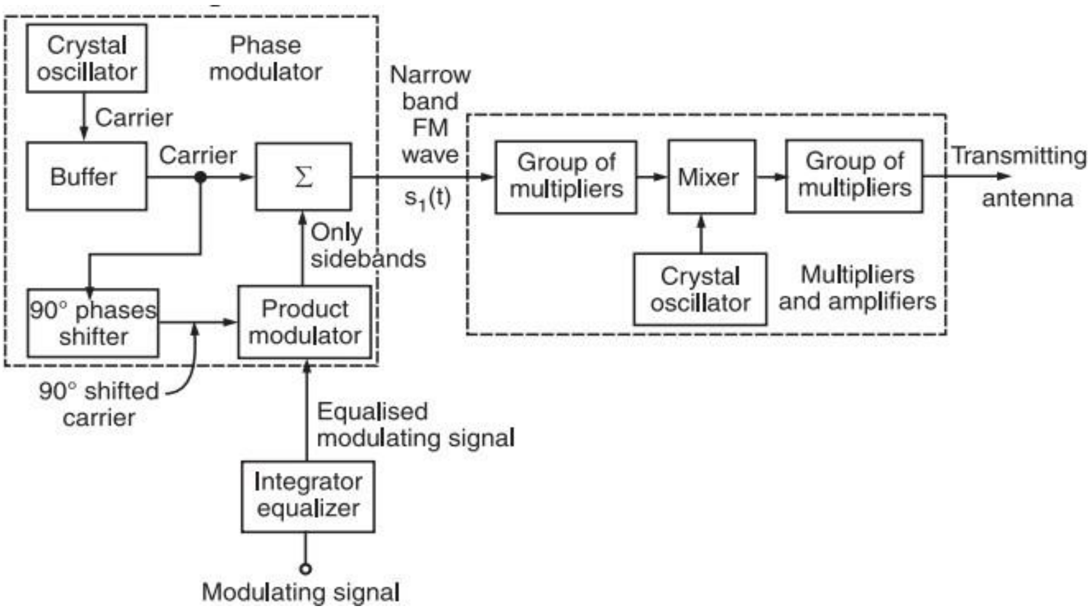
The FM modulator circuits used for generating FM signals may be put into two categories as under.

(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.

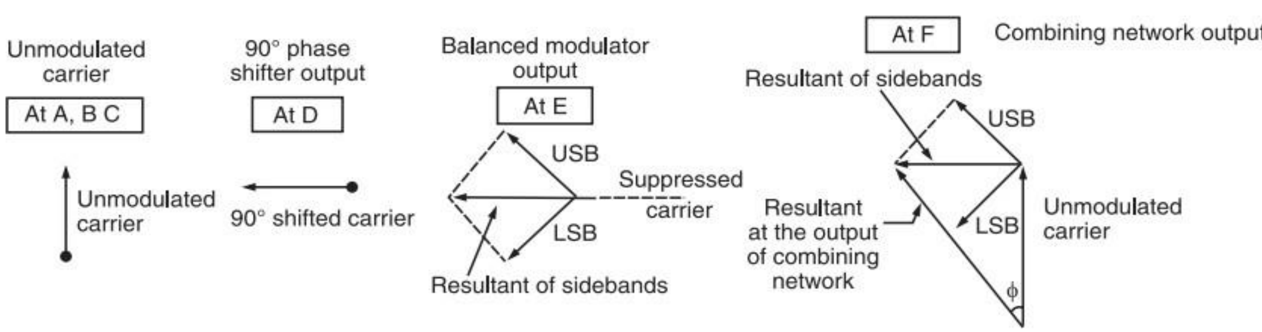


- The modulating signal x(t) is passed through an integrator before applying it to the phase modulator.
- The crystal oscillator produces a stable unmodulated carrier which is applied to the 90° phase shifter as well as the combining network through a buffer.

- 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 ∅ made by the resultant with unmodulated carrier. The angle ∅ 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, ∅ is small and for high modulation index, ∅ increases.

FM DEMODULATOR-

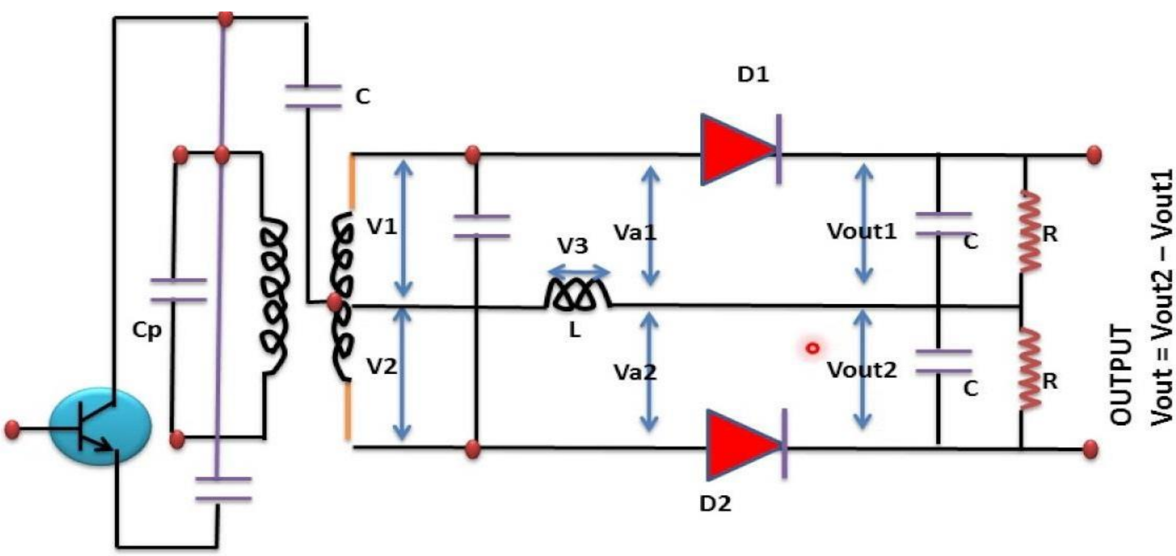
The demodulation process of FM waves is exactly opposite to that of the frequency modulation. After demodulation, we get the original modulating signal at the demodulation output.

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:

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

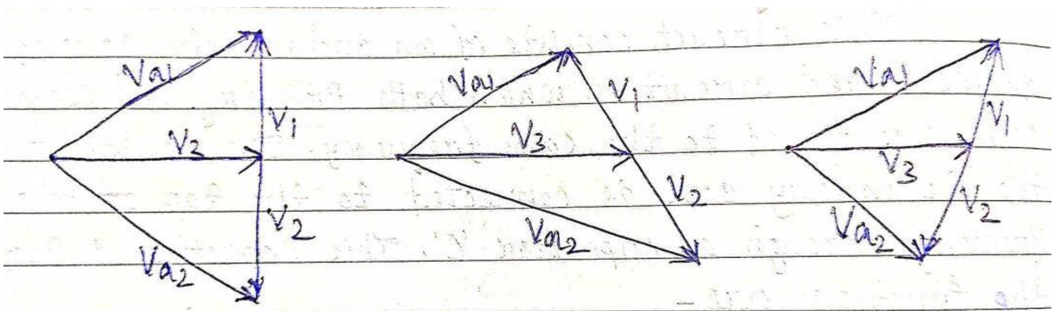
(iv) It couples the signal frequency from primary to center tapping of the secondary.

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

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

Advantages:

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

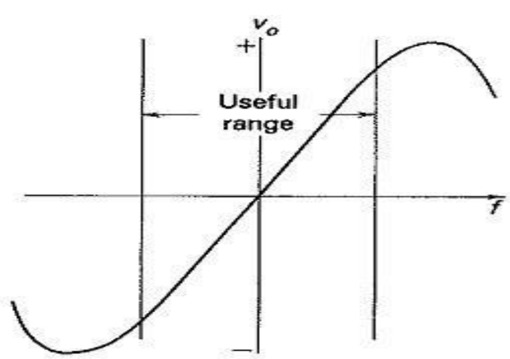
It does not provide amplitude limiting. So in the presence of noise or any other spurious amplitude variations, the demodulator output responds to them and produce errors.

RATIO DETECTOR-

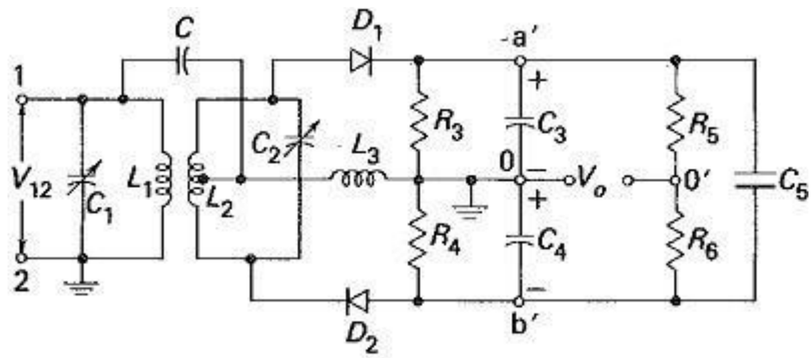
The RATIO DETECTOR uses a double-tuned transformer to convert the instantaneous frequency variations of the fm input signal to instantaneous amplitude variations. These amplitude variations are then rectified to provide a dc output voltage which varies in amplitude and polarity with the input signal 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.

Explain Delta Modulation in detail withsuitable diagram.

Delta Modulation

In PCM the signaling rate and transmission channel bandwidth are quite large since it transmits all the bits which are used to code a sample. To overcome this problem, Delta modulation is used.

Working Principle

Delta modulation transmits only one bit per sample. Here, the present sample value is compared with the previous sample value and this result whether the amplitude is increased or decreased is transmitted.

Input signal x(t) is approximated to step signal by the delta modulator. This step size is kept fixed.

The difference between the input signal x(t) and staircase approximated signal is confined to two levels, i.e., $+\Delta$ and $-\Delta$.

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

Fig.1 shows the analog signal x(t) and its staircase approximated signal by the delta modulator.

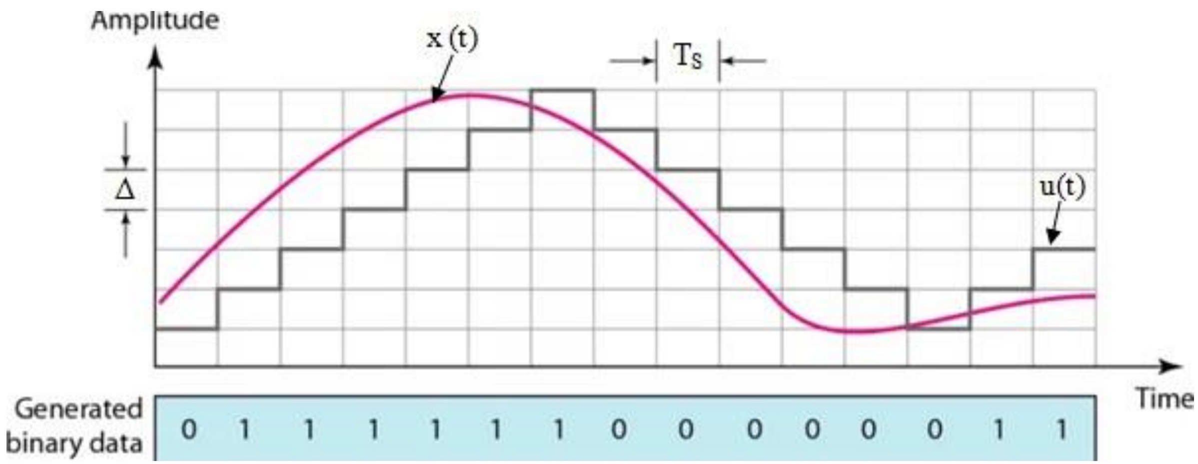


Fig.1. Delta Modulation Waveform

MATHEMATICAL EXPRESSIONS

The error between the sampled value of x(t) and last approximated sample is given as:

$$e(nT_s) = x(nT_s) - \hat{x}\,(nT_s)$$

Where $e(nT_s)$ = error at present sample
$x(nT_s)$ = sampled signal of x(t)

$\hat{x}\ (nT_s)$ = last sample approximation of the staircase waveform

If we assume $u(nT_s)$ as the present sample approximation of staircase output, then

$$u[(n-1)T_s) = \hat{x}\ (nT_s)$$

$$= \text{last sample approximation of staircase waveform}$$

Let us define a quantity $b(nT_s)$ in such a way that,

$$b(nT_s) = \Delta\, sgn[e(nT_s)]$$

This means that depending on the sign of error $e(nT_s)$, the sign of step size Δ is decided. In other words we can write

$$b(nT_s) = \begin{cases} +\Delta & if\ x(nT_s) \geq \hat{x}\,(nT_s) \\ -\Delta & if\ x(nT_s)\hat{x}\ < (nT_s) \end{cases}$$

Also if $\quad b(nT_s) = +\Delta$ then a binary '1' is transmitted
and if $\quad b(nT_s) = -\Delta$ then a binary '0' is transmitted
Here $\quad T_s$ = sampling interval.

TRANSMITTER

Fig. 2 (a) shows the transmitter . It is also known as Delta modulator

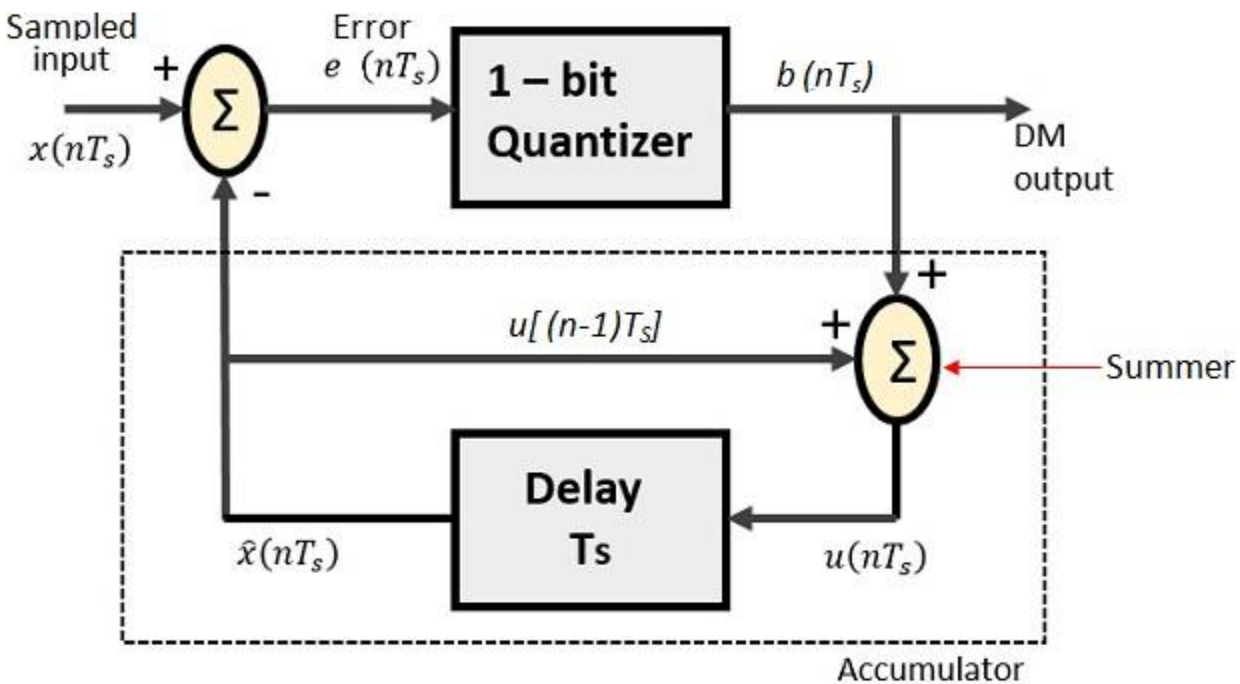


Fig.2 (a) Delta Modulation Transmitter

It consists of a 1-bit quantizer and a delay circuit along with two summer circuits.

The summer in the accumulator adds quantizer output ($\pm\Delta$) with the previous sample approximation . This gives present sample approximation. i.e.,

$$u(nT_s) = u((nT_s - T_s) + [\pm\Delta]$$

$$or \qquad u(nT_s) = u[(n-1)T_s] + b(nT_s)$$

The previous sample approximation $u[(n-1)T_s]$ is restored by delaying one sample period T_s .

The samples input signal $x(nT_s)$ and staircase approximated signal $\hat{x}(nT_s)$ are subtracted to get error signal $e(nT_s)$.
Thus, depending on the sign of $e(nT_s)$, one bit quantizer generates an output of $+\Delta$ or $-\Delta$.
If the step size is $+\Delta$, then binary ‘1’ is transmitted and if it is $-\Delta$, then binary ‘0’ is transmitted .

RECEIVER

At the receiver end also known as delta demodulator, as shown in fig.2 (b) , it comprises of a low pass filter(LPF), a summer, and a delay circuit. The predictor circuit is eliminated here and hence no assumed input is given to the demodulator.

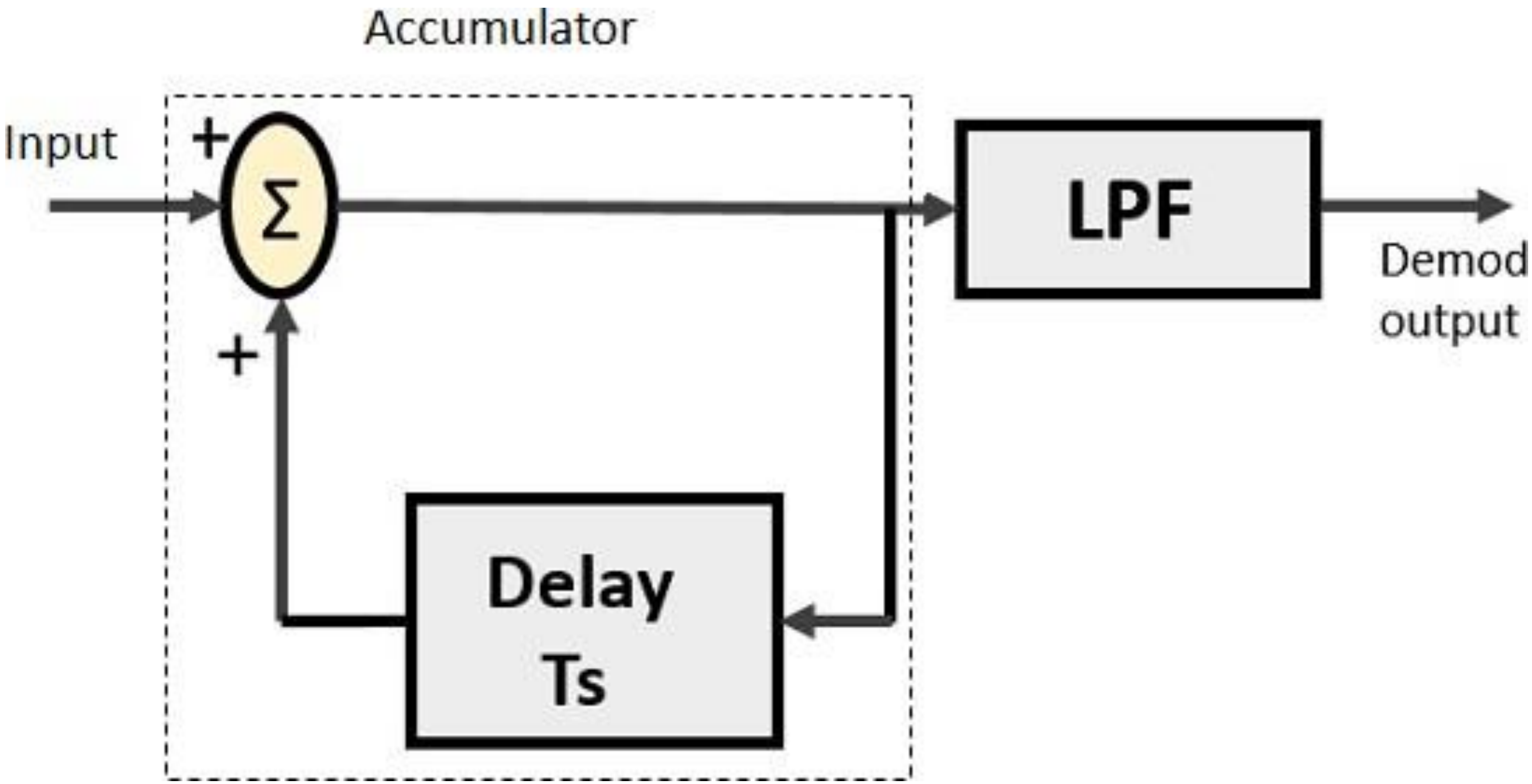


Fig.2 (b) Delta Modulation Receiver

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.

Also, the low pass filter smoothens the staircase signal to reconstruct the original message signal x(t) .

ADVANTAGES AND DISADVANTAGES OF DELTAMODULATION

Advantages of Delta Modulation

The delta modulation has certain advantages over PCM as under :

1. Since, the delta modulation transmits only one bit for one sample, therefore the signaling rate and transmission channel bandwidth is quite small for delta modulation compared to PCM .
2. The transmitter and receiver implementation is very much simple for delta modulation. There is no analog to digital converter required in delta modulation.

DISADVANTAGES OF DELTA MODULATION

The delta modulation has two major drawbacks as under :

1. Slope overload distortion
2. Granular or idle noise

Now, we will discuss these two drawbacks in detail.

1. Slope Overload Distortion

This distortion arises because of large dynamic range of the input signal.

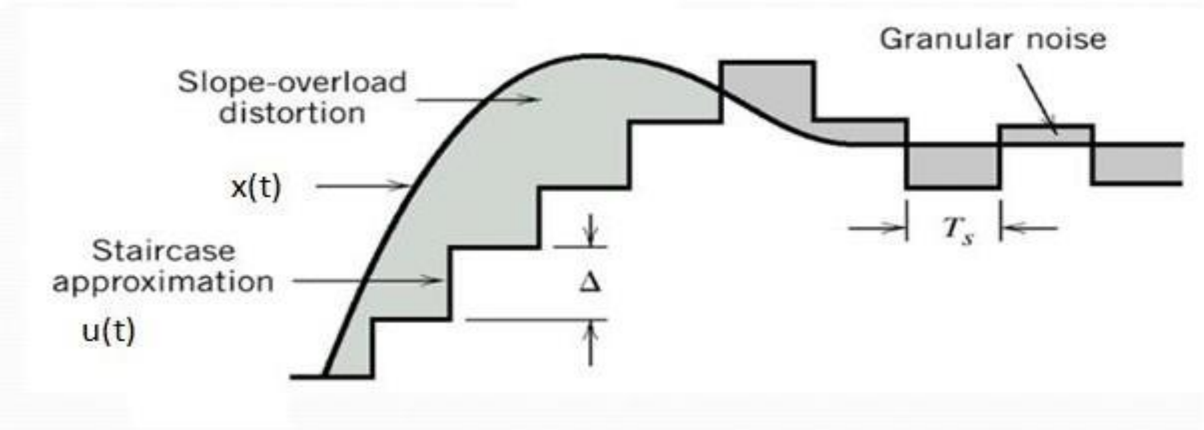


Fig.1 : Quantization

We can observe from fig.1 , the rate of rise of input signal x(t) is so high that the staircase signal can not approximate it, the step size ‘Δ’ becomes too small for staircase signal u(t) to follow the step segment of x(t).

Hence, there is a large error between the staircase approximated signal and the original input signal x(t).

This error or noise is known as slope overload distortion .
To reduce this error, the step size must be increased when slope of signal x(t) is high.

Since, the step size of delta modulator remain fixed, its maximum or minimum slopes occur along straight lines. Therefore, this modulator is known as Linear Delta Modulator (LDM) .

2. GRANULAR OR IDLE NOISE

Granular or Idle noise occurs when the step size is too large compared to small variation in the input signal.

This means that for very small variations in the input signal, the staircase signal is changed by large amount (Δ) because of large step size.

Fig.1 shows that when the input signal is almost flat , the staircase signal u(t) keeps on oscillating by ±Δ around the signal.

The error between the input and approximated signal is called granular noise.
The solution to this problem is to make the step size small .

EXPLAIN DIFFERENTIAL PULSE CODE MODULATION

DIFFERENTIAL PULSE CODE MODULATION

It may be observed that the samples of a signal are highly correlated with each other. This is due to the fact that any signal does not change fast. Which means , its value from present sample to next sample does not vary by a large amount.

The adjacent samples of the signal carry the same information with a little difference.

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

REDUNDANT INFORMATION IN PCM

Fig.1 shows a continuous time signal x(t) by dotted line. This signal is sampled by flat top sampling at intervals T_s , $2T_s$, $3T_s$ ….. nT_s .

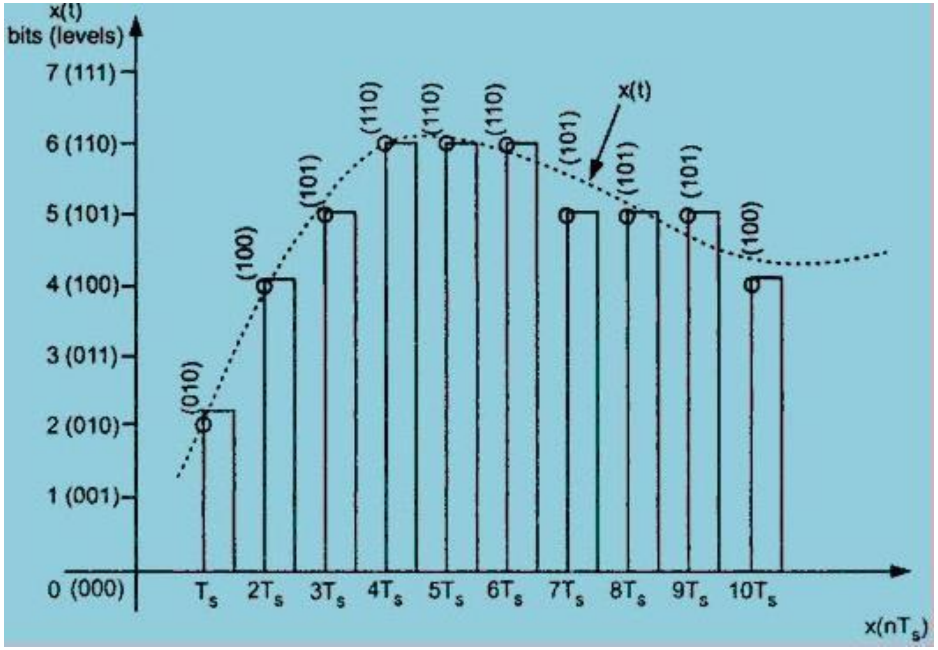


Fig.1 : Illustration of redundant information in PCM

The sampling frequency is selected to be higher than nyquist rate.

The samples are encoded by using 3 bit (7 levels) PCM.

The sample is quantized to the nearest digital level as shown by small circles in fig.1 .

The encoded binary value of each sample is written on the top of the samples.

We can observe from fig.1 that the samples taken at $4T_s$, $5T_s$ and $6T_s$ are encoded to same value of (110). This information can be carried only by one sample.

But three smaples are carrying the same information means that it is redundant .

We consider another example of samples taken at $9T_s$ and $10T_s$. The difference between these samples only due to last bit and first two bits are redundant, as they do not change.
If this redundancy is reduced, then overall bit rate will decrease and number of bits required to transmit one sample will also be reduced.

This type of digital pulse modulation technique is called as Differential Code Modulation (DPCM).

WORKING PRINCIPLE

The differential pulse code modulation works on the principle of prediction. The value of the present sample is predicted from the past samples.

The prediction may not be exact but it is very close to the actual sample value.

Fig.2 shows the transmitter of DPCM system.

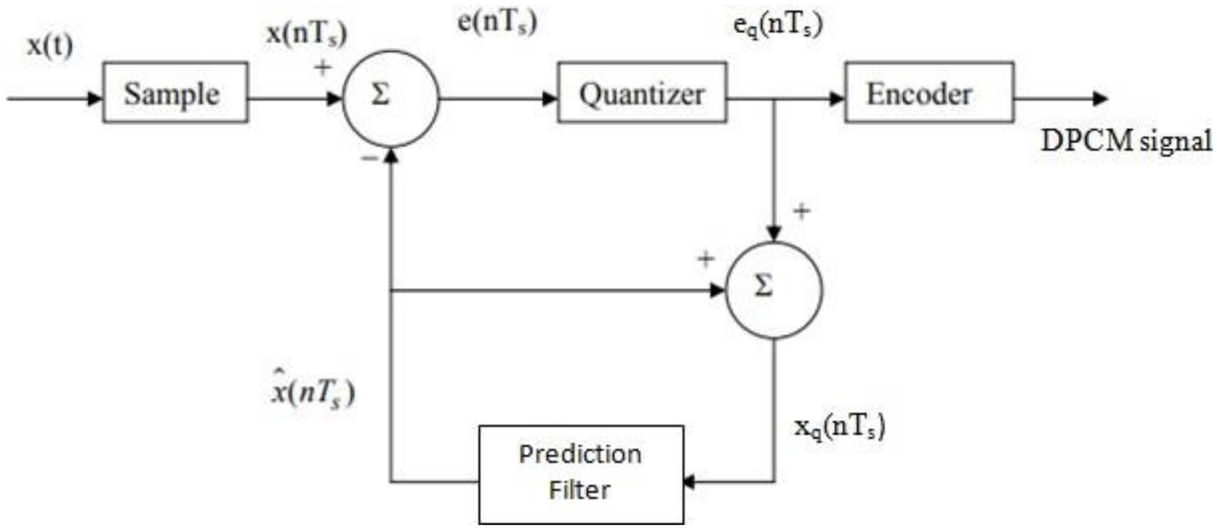


Fig.2 : A Differential pulse code modulation

The sampled signal is denoted by $x(nT_s)$ and predicted signal is denoted by $x\hat{}(nT_s)$.
The comparator finds out the difference between the actual sample value $x(nT_s)$ and predicted sample value $x\hat{}(nT_s)$.
This is known as prediction error and it is denoted by $e(nT_s)$.
It can be defined as ,

$e(nT_s) = x(nT_s) – x\hat{}(nT_s)$…................................ (1)
The predicted value is produced by using a prediction filter.

The quantizer output signal gap $e_q(nT_s)$ and previous prediction is added and given as input to the prediction filter.This signal is called $x_q(nT_s)$.
This makes the prediction more and more close to the actual sampled signal.

We can observe that the quantized error signal $e_q(nT_s)$ is very small and can be encoded by using small number of bits.
Thus number of bits per sample are reduced in DPCM.

The quantizer output can be written as ,

$e_q(nT_s) = e(nT_s) + q(nT_s)$… .... (2)
Here, $q(nT_s)$ is the quantization error.
As shown in fig.2, the prediction filter input $x_q(nT_s)$ is obtained by sum $\hat{x}(nT_s)$ and quantizer output. i.e.,
$x_q(nT_s) = \hat{x}(nT_s) + e_q(nT_s)$… .... (3)
Substituting the value of $e_q(nT_s)$ from eq.(2) in the above eq. (3) , we get,
$x_q(nT_s) = \hat{x}(nT_s) + e(nT_s) + q(nT_s)$............................ (4)
eq.(1) is written as,

$e(nT_s) = x(nT_s) – \hat{x}(nT_s)$
$\therefore \quad e(nT_s) + \hat{x}(nT_s) = x(nT_s)$
Therefore, substituing the value of $e(nT_s) + \hat{x}(nT_s)$ from the above equation into eq. (4), we get,
$x_q(nT_s) = x(nT_s) + q(nT_s)$..............................(5)

RECEPTION OF DPCM SIGNAL

Fig.3 shows the block diagram of DPCM receiver.

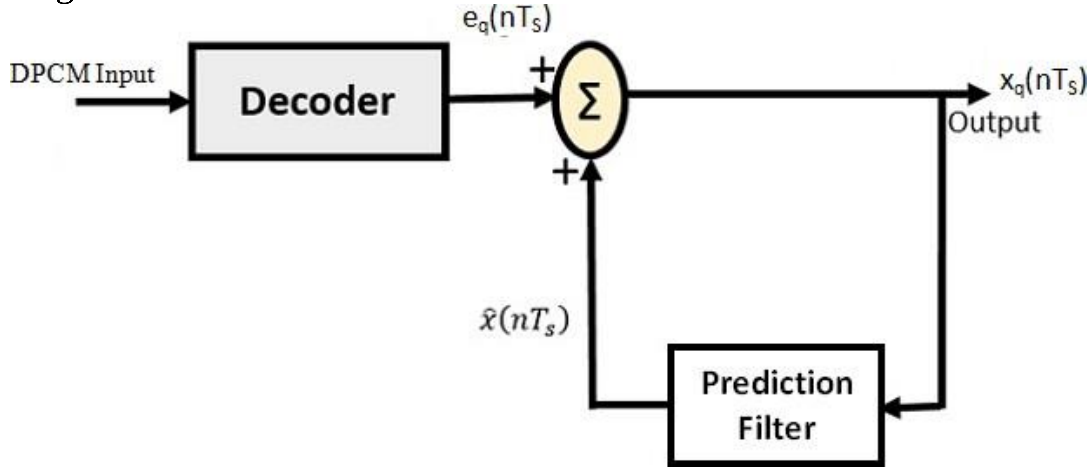


Fig.3 : DPCM Receiver

The decoder first reconstructs the quantized error signal from incoming binary signal.

The prediction filter output and quantized error signals are summed up to give the quantized version of the original signal.

Thus the signal at the receiver differs from actual signal by quantization error $q(nT_s)$, which is introduced permanently in the reconstructed signal.

ADVANTAGES OF DPCM

1. As the difference between $x(nT_s)$ and $x\hat{}(nT_s)$ is being encoded and transmitted by the DPCM technique, a small difference voltage is to be quantized and encoded.
2. This will require less number of quantization levels and hence less number of bits to represent them.
3. Thus signaling rate and bandwidth of a DPCM system will be less than that of PCM.

COMPARISON BETWEEN PCM, DM, ADM ANDDPCM

In this article we will compare Pulse Code Modulation (PCM), Delta Modulation (DM), Adaptive Delta Modulation (ADM) and Differential Pulse Code Modulation.

We have already discussed all these modulation techniques in our previous articles.

Comparison between all these modulation techniques is shown in the table below.

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

CONVERSION OF ANALOG SIGNALS TO DIGITALSIGNALS

In communication systems, sometimes it happens that we are available with an analog signal, and we have to transmit a digital signal for that particular application.

In such cases, we have to convert the analog signal to digital signal. That means that we have to convert a continuous time signal in the form of digits.

To see how a signal can be converted from analog signal to digital form, let us consider an analog signal x(t) as shown in fig.1(a).

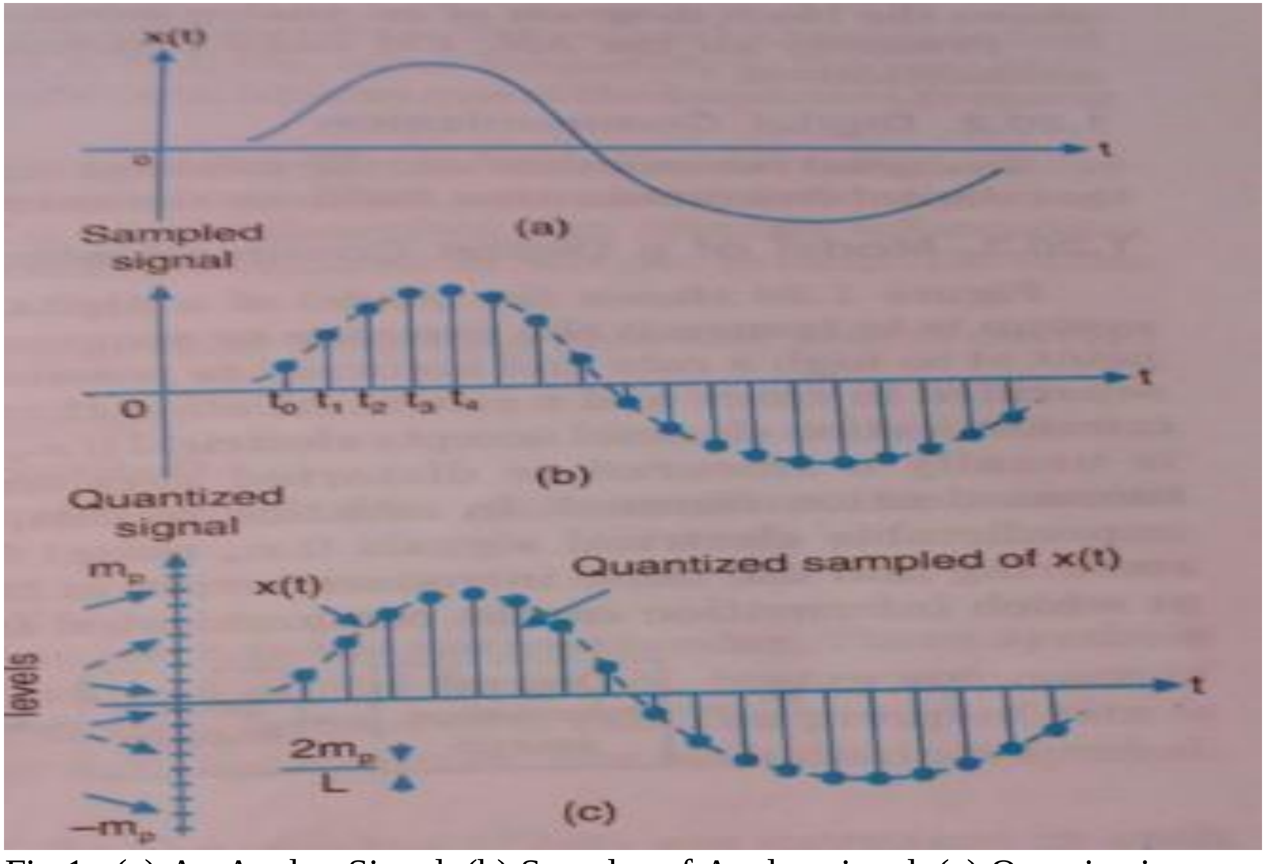


Fig.1 : (a) An Analog Signal, (b) Samples of Analog signal, (c) Quantization

First of all , we get sample of this signal according to the sampling theorem.

For this purpose, we mark the time-instants t_0, t_1 ,t_2 and so on , at equal time-intervals along the time axis. At each of these time-instants , the magnitude of the signal is measured and thus samples of the signal are taken. Fig.1(b) shows a representation of the signal of fig.1(a) in terms of its samples.

Now, we can say that the signal in fig.1(b) is defined only at the sampling instants.
This means that, it no longer is a continuous function of time, but rather, it is a discrete-time signal.

However, since the magnitude of each sample can take any value in a continuous range, the signal in fig.1(b) is still an analog signal.

This difficulty is neatly resolved by a process known as quantization. In quantization, the total amplitude range which the signal may occupy is divided into a number of standard levels.

As shown in fig.1(c), amplitudes of the signal x(t) lie in the range ($-m_p$, m_p) which is partitioned into L intervals, each of magnitude $\Delta v = 2m_p / L$.
Now, each sample is approximated or rounded off to the nearest quantized level as shown in fig.1(c) .

Since each sample is now approximated to one of the L numbers, therefore, the information is digitized.

The quantized signal is an approximation of the original one. We can improve the accuracy of the quantized signal to any desired degree simply by increasing the number of levels L .

COMPARISON OF PCM AND ANALOG MODULATION

COMPAIRING PCM AND ANALOG MODULATION

In this tutorial we shall compare PCM and Analog Modulation in detail.

The threshold effect in PCM is similar to a property of analog modulation methods such as FM or PPM.

The property is that , these systems tend to reduce the wideband noise above the threshold levels.

The PCM also provides the wideband noise reduction if it is operated above its threshold which is given by:

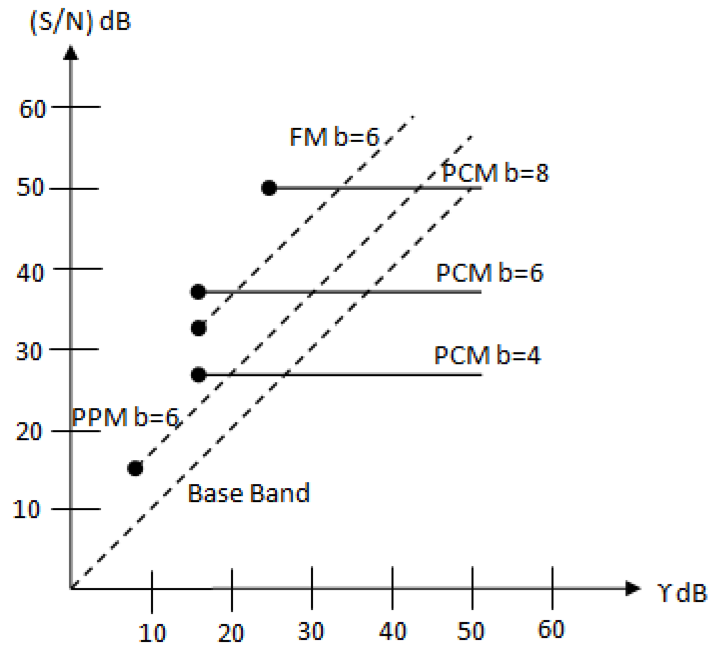
$$(\frac{S}{N})_D = 3q^2 S_x$$

Where $q=2^v$ for binary PCM and $q= M^v$ for M-ary PCM.
We assume that the sampling frequency is close to the Nyquist rate and bandwidth BW = Nf_m Hz.
Then, $q=M^v= M^b$ where b = BW/ f_m is known as the bandwidth ratio.
Therefore, we have,

$$(\frac{S}{N})_D = 3 \times (M^V)^2 S_x = 3 \times M^{2V} S_x$$

Here, $(S/N)_D$ = Signal to noise ratio at the destination
S_x = Signal power at the destination
Here, it may be noted that the signal to noise ratio $(S/N)_D$ is proportional to M^{2b} which is much higher than the $(S/N)_D$ of the wideband FM which is proportional to only b or b^2.
Hence, PCM performs better than FM .

Fig.1 shows the performance of various modulation types as a function of γ .



Comparison of PCM and analog communication

Fig.1

All the curves in fig.1 have been plotted for S_x= 1/2. The dots indicate the threshold points. The PCM curves have been drawn for M = 2 and v = b.

CONCLUSIONS

Some of the important observations from fig.1 may be listed as under:

1. For PCM if b is constant , then increase in γ beyond the threshold value γth (corresponding to the threshold point) does not increase $(S/N)_D$ at all . Let us observe the flat PCM curves in fig.1. hence, PCM must be operated just above the threshold.
2. Near threshold, the PCM does offer some advantages over FM and PPM, with thesame values of b and $(S/N)_D$.
3. However, this advantage is gained at the expense of more complicated and expensive circuitry.
4. The $(S/N)_D$ for FM and PPM increases linearly with increase in the value of γ and becomes better than that of PCM for higher values of γ .

BENEFITS OF PCM

Fig.1 reveals the following benefits of using the PCM :

1. PCM allows the use of regenerative repeaters. This improves its noise performance.
2. PCM allows the transmission of analog signals in the form of digital signals.

PCM IS NOT USED FOR RADIO BROADCASTING

In Radio Broadcasting, a relatively large signal to noise ratio (typically of the order of 60 dB) is required.

To get this level of $(S/N)_D$, the PCM with $b>8$ is needed.
However, we can obtain the same performance with an FM system with $b = 6$ and with much simpler transmitter and receiver circuits.

Therefore, higher bandwidth requirement and complicated circuitry are the drawbacks of PCM which does not allow its use fr the radio, TV broadcasting applications .

Mapping of course / module to the Programme Learning Outcomes												
Learning Outcome of the course	Programme Outcomes											
	01	02	03	04	05	06	07	08	09	10	11	12

ANALOG TRANSMISSION

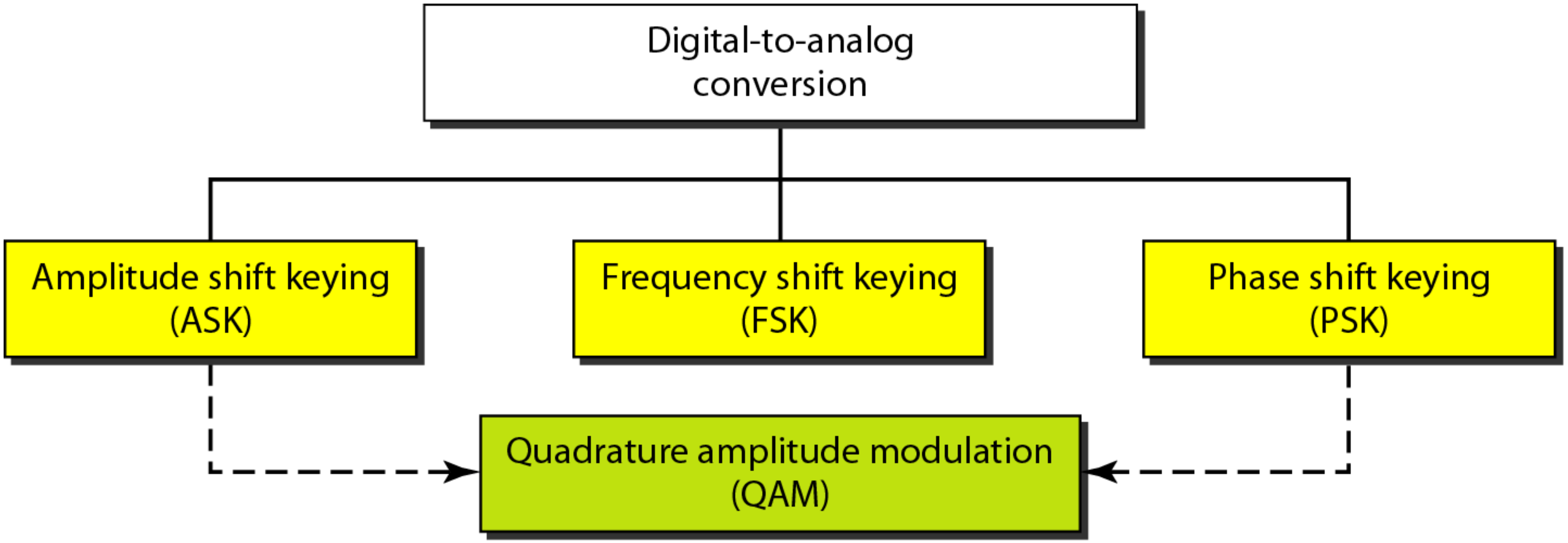
● Three mechanisms of *modulating* digital datainto an analog signal by *altering* any of the three *characteristics* of analog signal:

→ ASK : Amplitude shift keying

→ FSK : Frequency shift keying

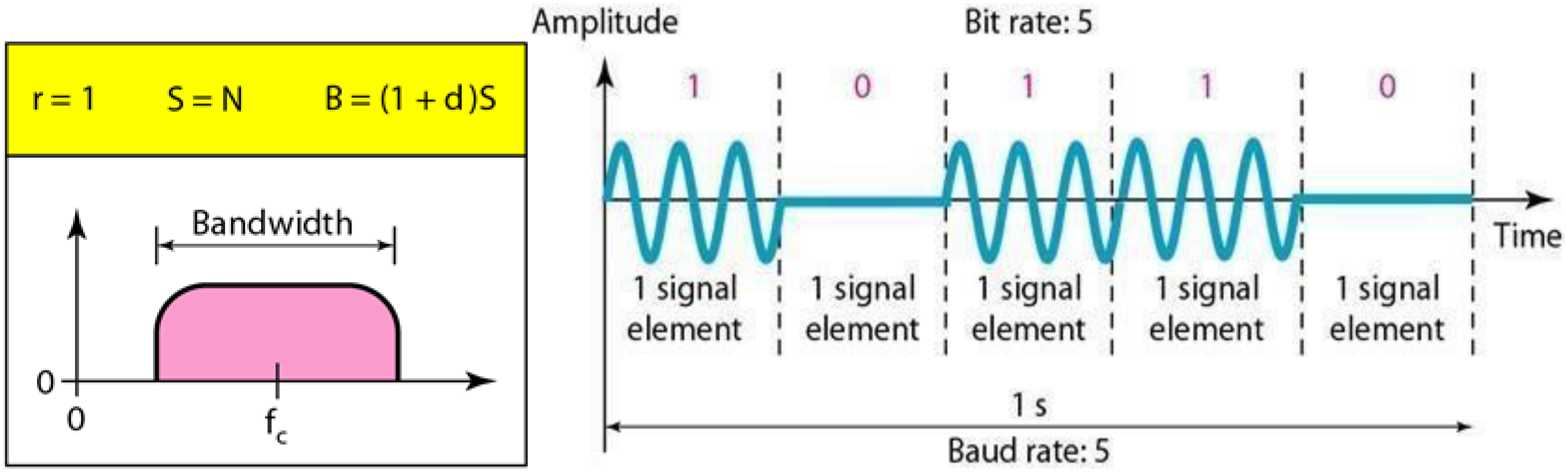
→ PSK : Phase shift keying

TYPES OF ANALOG TRANSMISSION



AMPLITUDE SHIFT KEYING •

modulation produces aperiodic composite signal, with continuous setof frequencies bandwidth is proportional to the signal (baud) rate



Bandwidth is divided into two with two carrier frequencies, as The figure shows the positions of two carrier frequencies and the bandwidths.

The available bandwidth for each direction is now 50 kHz, which leaves us with a data rate of 25 kbps in each direction.

MERITS AND DEMERITS

•Values represented by different amplitudes ofcarrier

•Usually, one amplitude is zero – i.e. presence andabsence of carrier is used

•Susceptible to sudden gain changes

•Inefficient

•Typically used up to 1200bps on voice grade lines

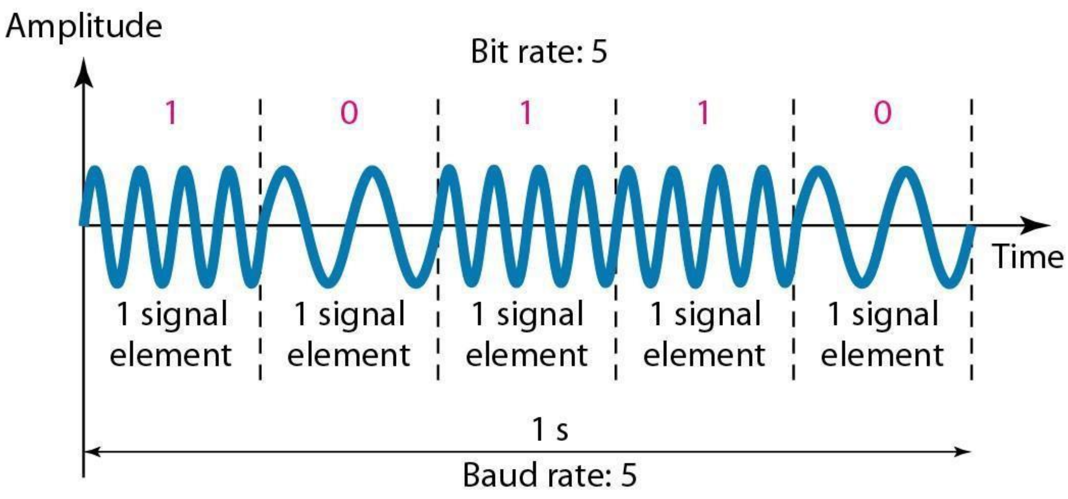
•Used over optical fiber

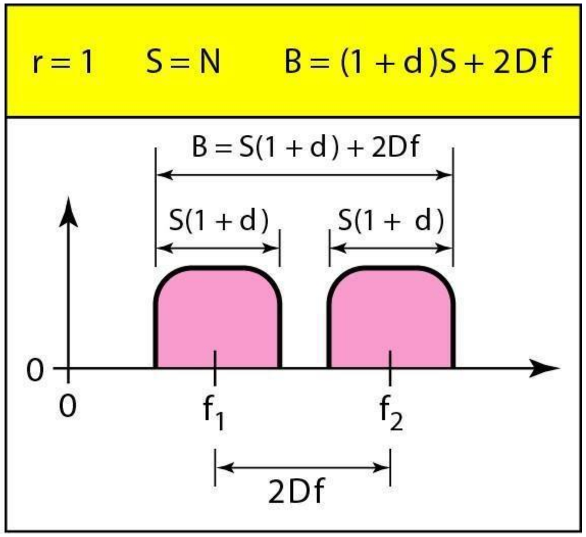
FREQUENCY SHIFT KEYING

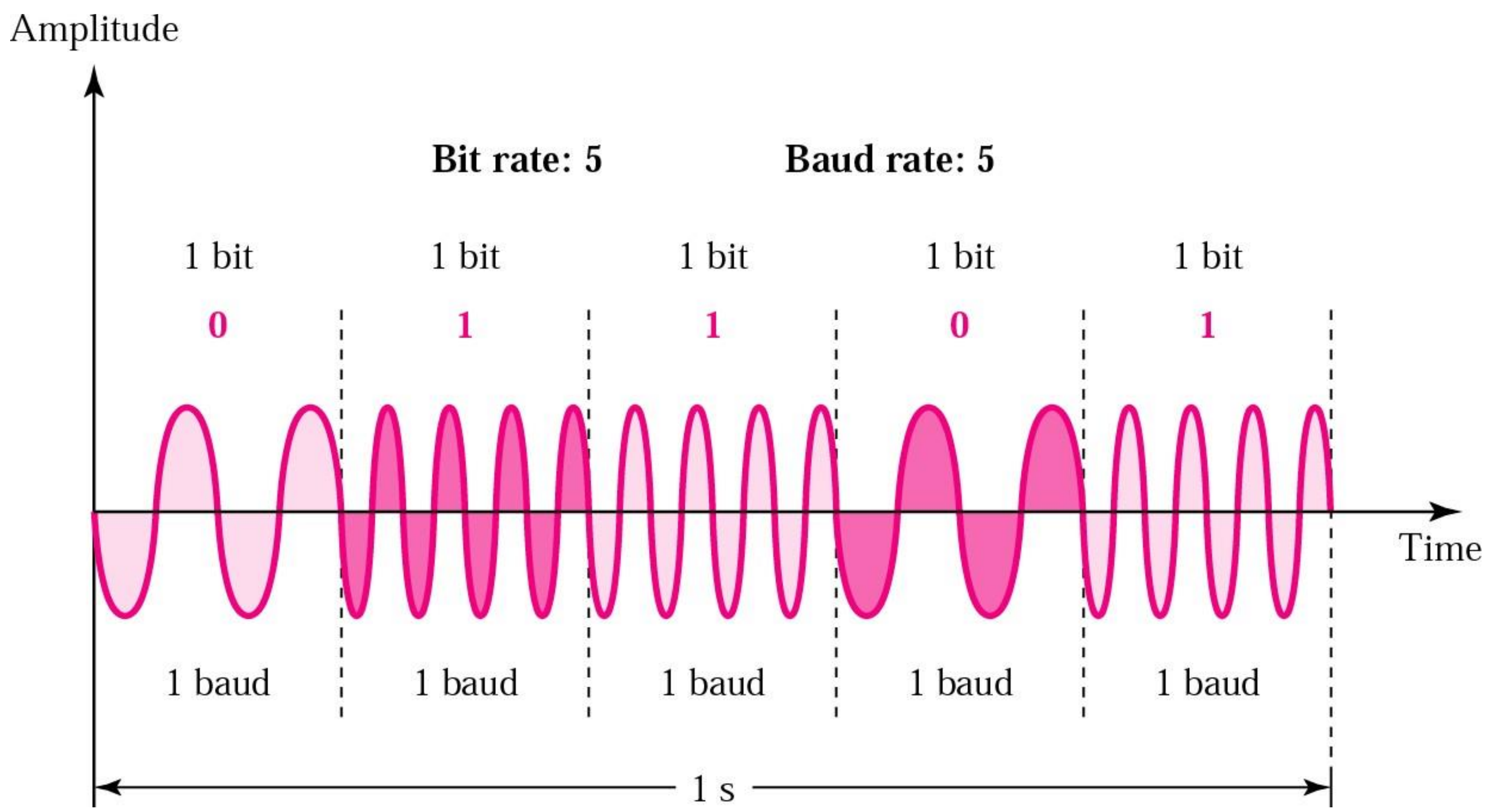
• *frequency* of the carrier signal is *varied* to represent data frequency of the modulated signal is constant for the duration of one signal element and changes for the next signal element if the data element changes amplitude Amplitude and Phase remain constant for all signal elements

BINARY FSK

- implemented using *two* carrier frequencies:
- F1,(space frequency) data elements 0
- f2, (mark frequency) data elements 1
- Both f1 and f2 are 2Δf apart






Amplitude
Bit rate: 5
Baud rate: 5
1 bit
1 bit
1 bit
1 bit
1 bit
0
1
1
0
1
Time
1 baud
1 baud
1 baud
1 baud
1 baud
1 s

MERITS AND DEMERITS

•Values represented by different frequencies (nearcarrier)

•Less susceptible to error than ASK

•Typically used up to 1200bps on voice grade lines

•High frequency radio

•Even higher frequency on LANs using co-ax

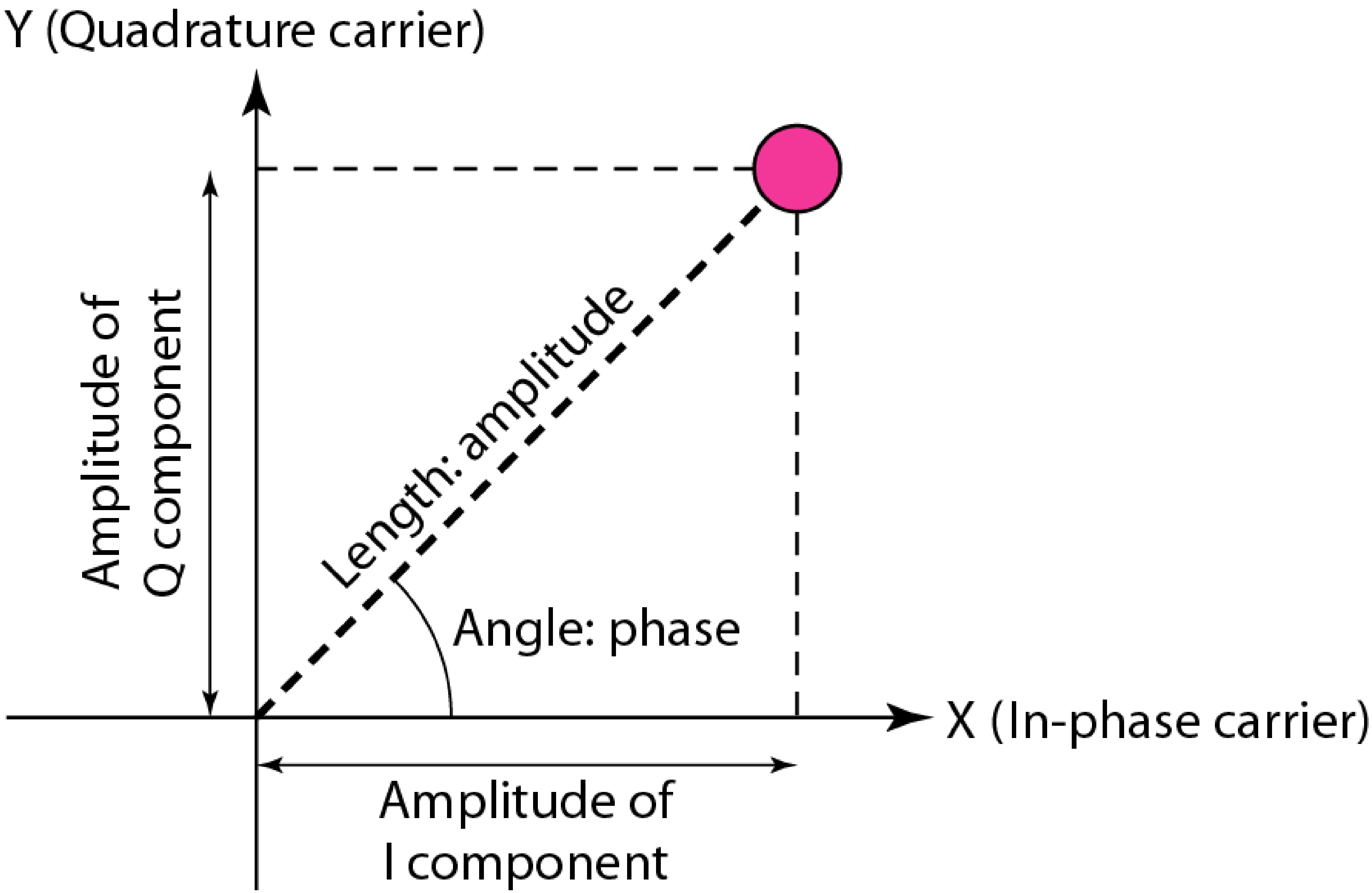
•Used in cordless and paging system

PHASE SHIFT KEYING

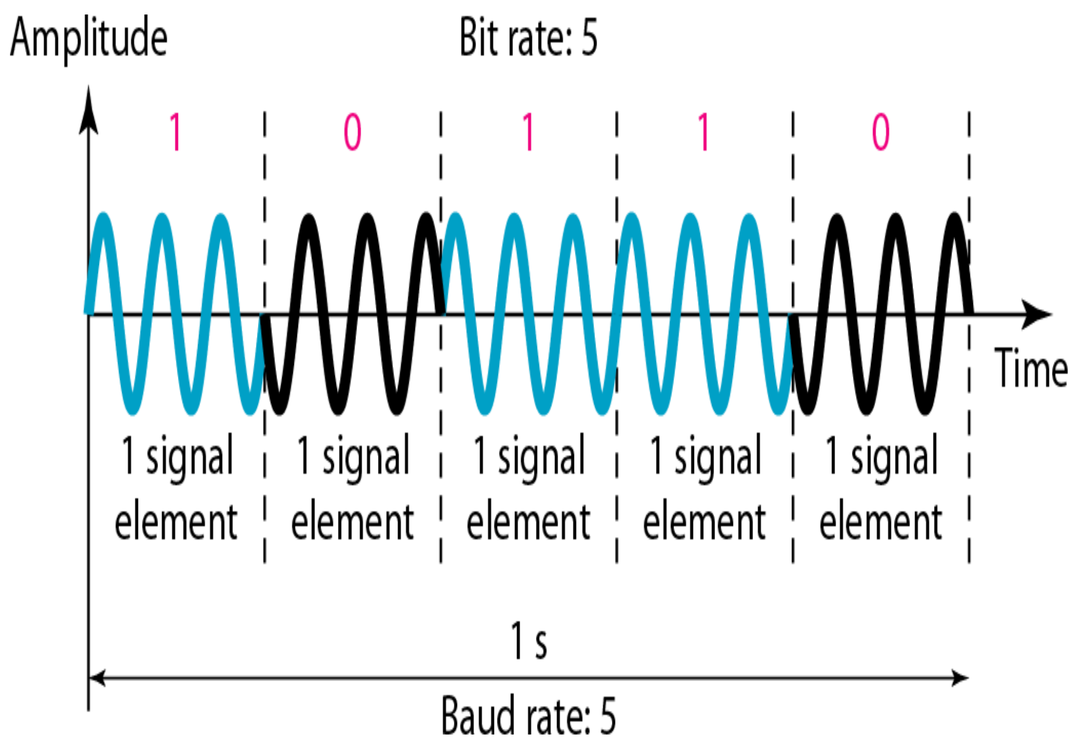
• *Phase* of the carrier signal is *varied* to represent two or more different signal elements amplitude and frequency remain constant

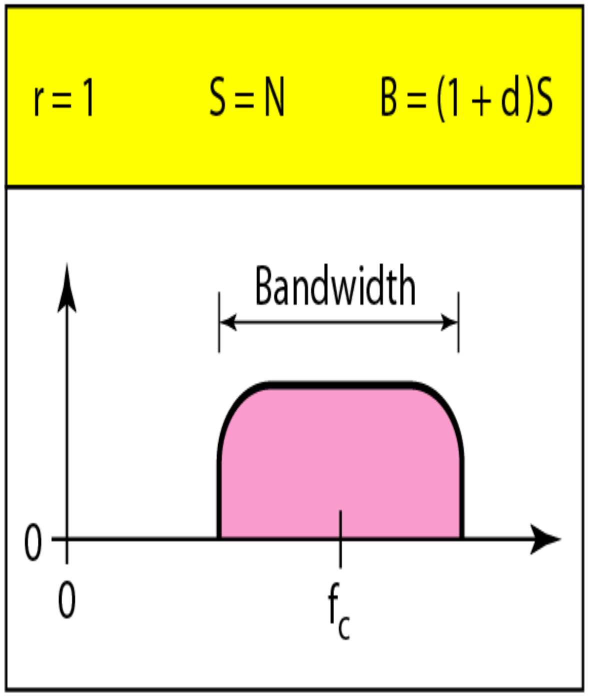
CONSTELLATION DIAGRAM

• Helps defining the amplitude and phase of a signal element signal element type is represented as a dot the bit or combination of bits it carries is written next to the dot diagram has two axes X-axis → related to the in-phase carrier Y-axis → related to the quadrature carrier

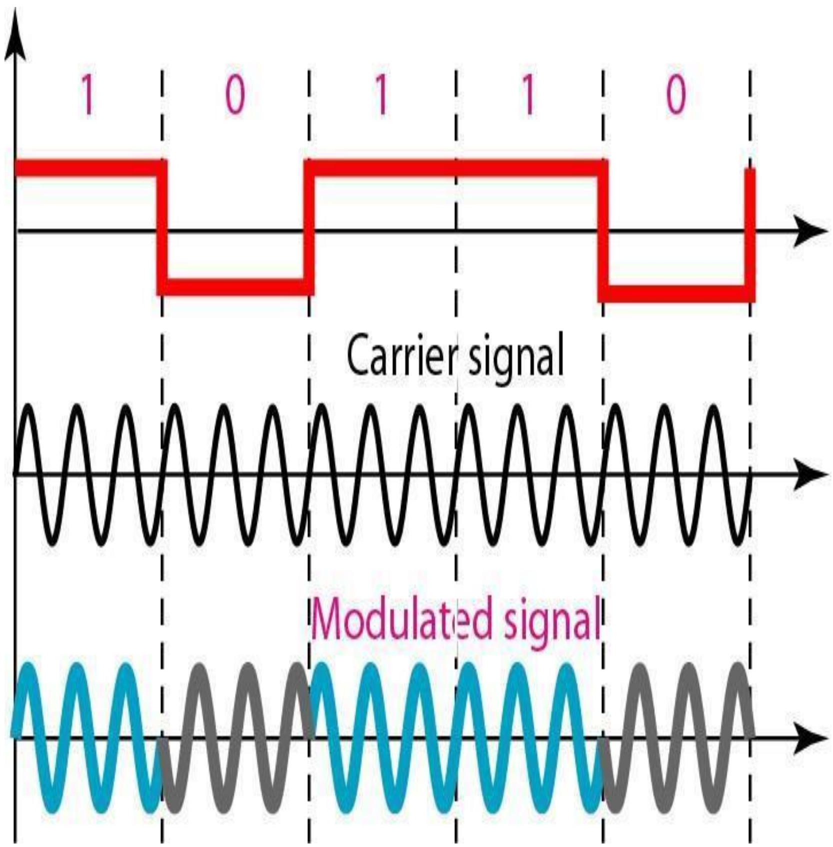

Y (Quadrature carrier)
Amplitude of
Q component
Length: amplitude
Angle: phase
X (In-phase carrier)
Amplitude of
I component

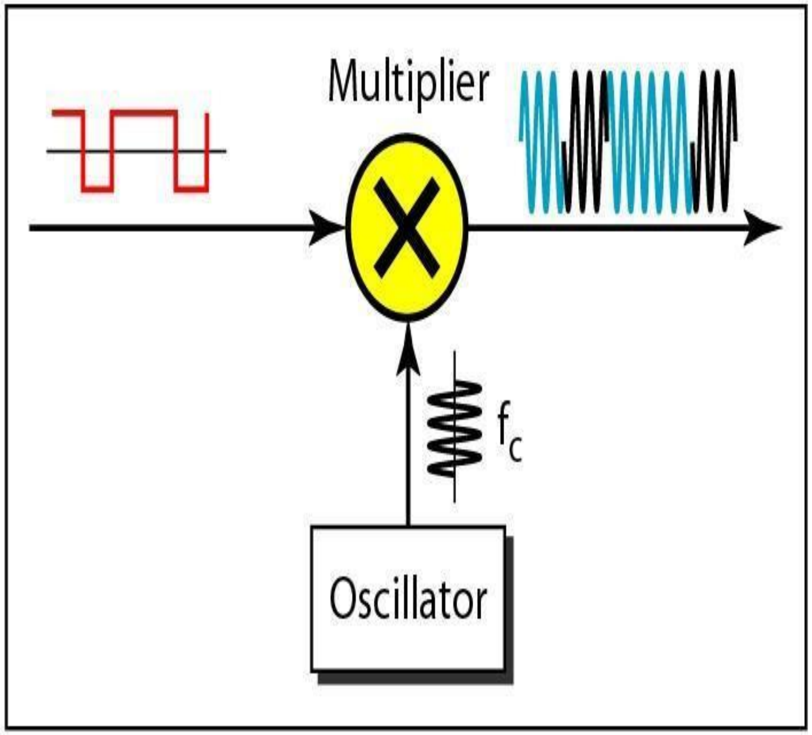
phase 0o → 1 bit ; phase 180o → 0 bit bandwidth requirement isthe same as that of ASK





Binary PSK : implementation



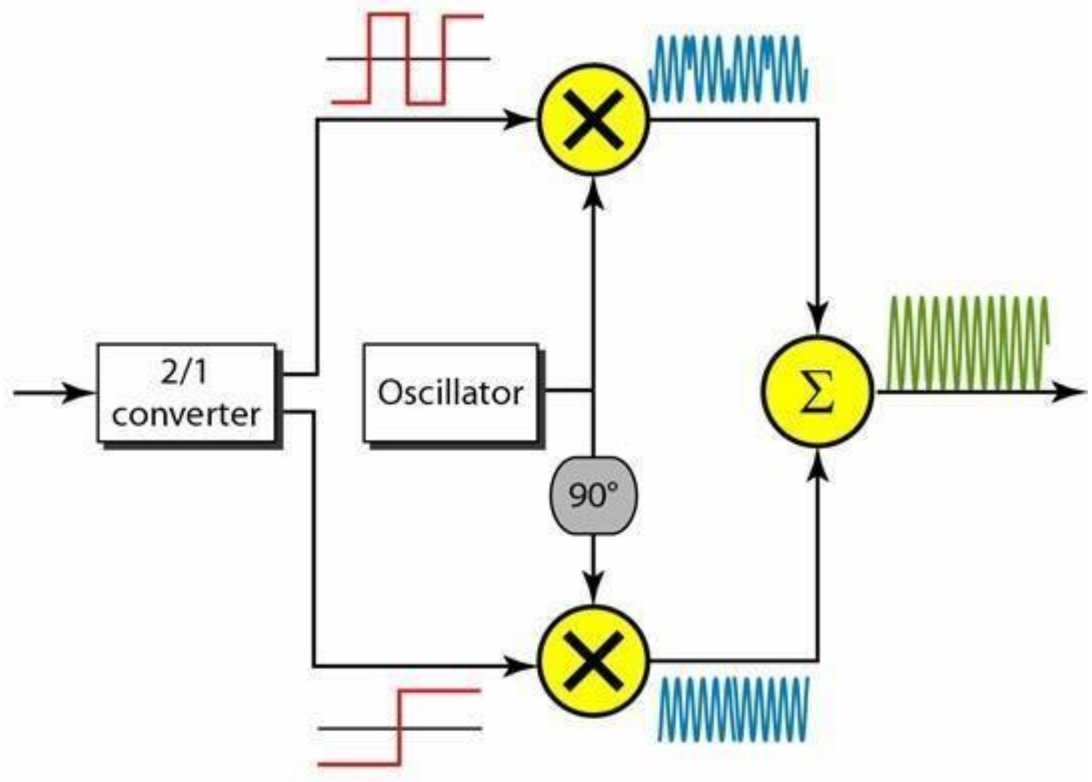


QUADRATURE PSK

- Use of *two bits* at a time in each signal element → decrease of baud rate →reduction of required bandwidth

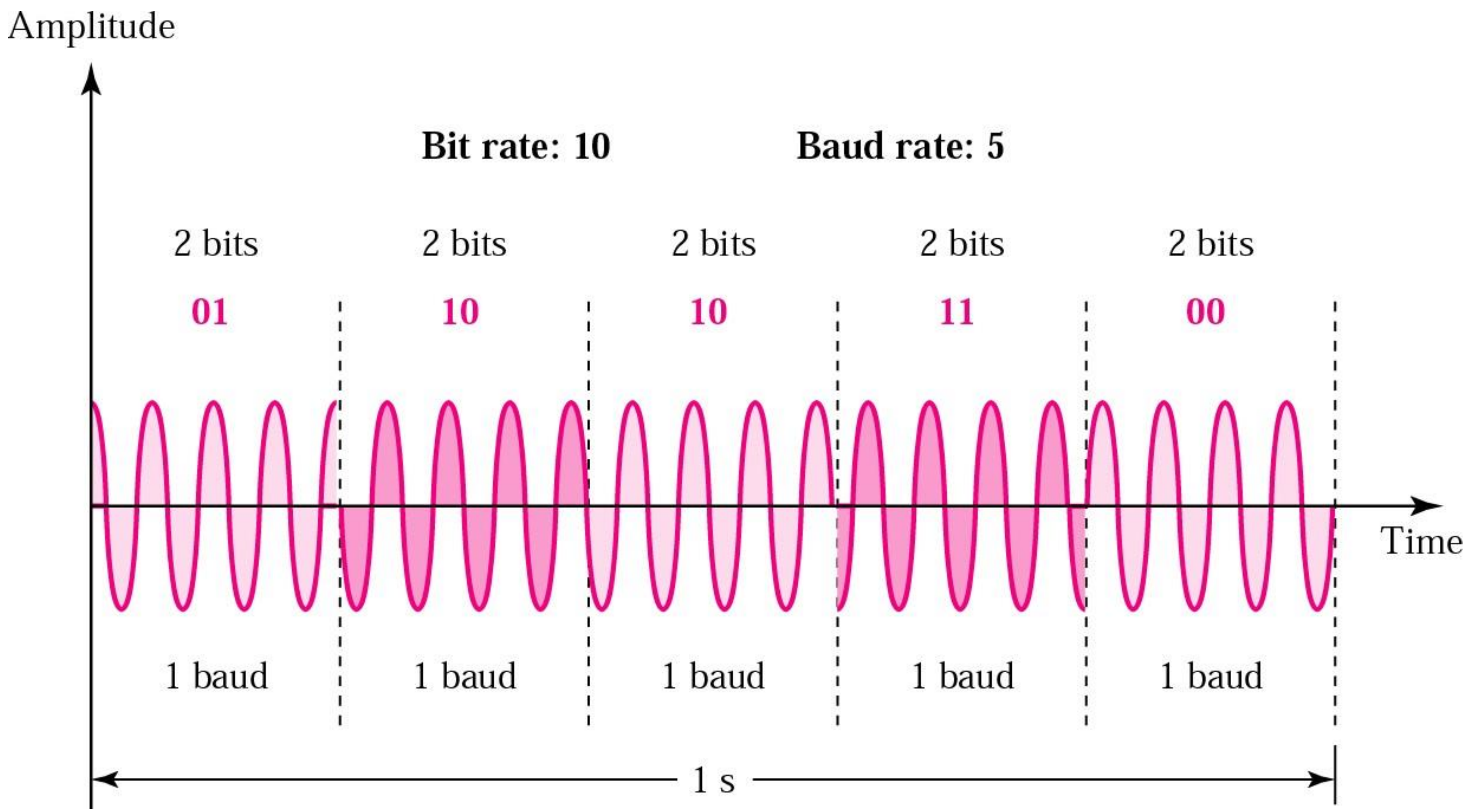
- Uses two separate BPSK modulations :one in-phase and the other out-of-phase (quadrature)

Quadrature PSK: implementation



serial to parallel converter sends one bit to one modulatorand the next bit to the other modulator

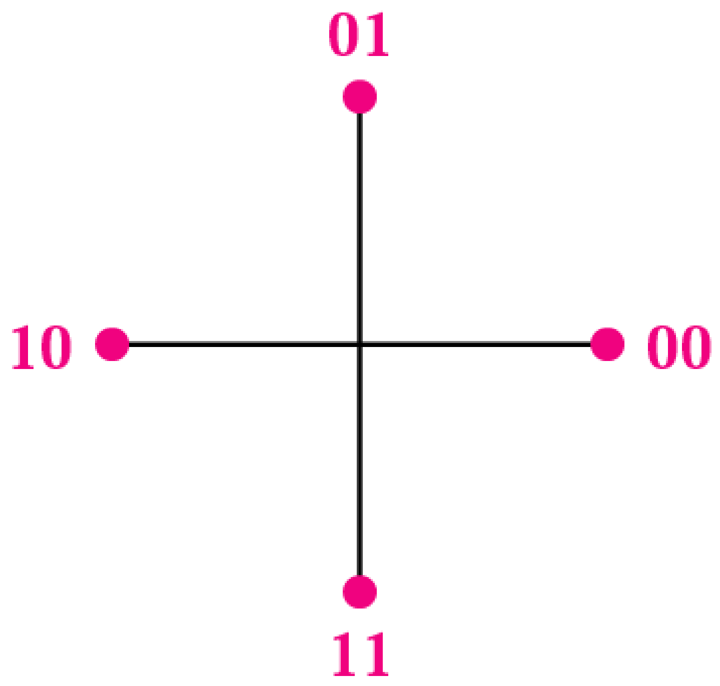
Quadrature PSK



- **4-PSK characteristics**

Dibit	Phase
00	0
01	90
10	180
11	270

Dibit
(2 bits)



Constellation diagram