CHAPTER-1

Elements Of Communication System

* What is communication?

Communication is simply the act of transferring information from one Place/Person /group to another.

Elements of communication -

From the formation of to the delivery of a message all the component which helps a communication Process to be Completed are known as elements of Communication The different elements of communication.

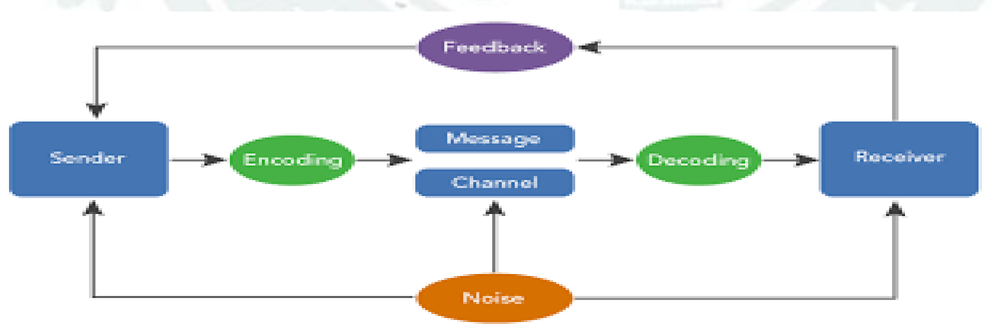
1.Sender

2. Massage

3-Channel

4. Receiver

5. Feed Back



Sender- In the process of communication the sender is one who forms and encodes a message and transmits it to the receivers.

Message: Message can be a any information (idea, thoughts, feelings of sender) It can be either verbal or non-verbal.

Encoding- Encoding is simply putting the message in may be an appropriate form which verbal or non-verbal.

Channel- The link between sender and receiver is known as channel. Channel refers to the way or mode the Message flows through. or is transmitted

Receiver- Receiver who is the person or group. the message is produced for! He may be reader listen or a viewer.

Decoding: The receiver after receiving the message interprets it and tries to understand it in the best possible manner. It is known as decoding.

Feed back: After understanding, the massage when b receiver response to it. It is known Communication channels as feedback.

<u>Communication channels</u>-

A communication channel is a path that Connects the I sender and receiver for transmission of information over a network.

There are two types of communication Channel

1-Guided media

2- Unguided media

<u>Guided media-</u>

It is defined as the physical medium through which the signals are transmitted.

It is also known as bounded media.

Guided media are three types

1. Twisted pair cable

2. Coaxial cable

3 . Optical fiber

Twisted Pair cable -

Twisted Pair cabling is a type of wiring in which two conductors of a single circuit are twisted together for the purposes of improving electromagnetic compatibility.

Coaxial cable: -

Coaxial cable, or coax is a type of electrical cable consisting of an inner conductor surrounded by a Concentric conducting shield, with the two separated by a dielectric; many coaxial Cables also have a Protective outer Sheath or Jacket.

Optical Fiber –

A fiber optic cable also known as an optical fiber. It is an assembly Similar to an electrical cable but containing that one or more optical fibers are used to carry light.

Unguided media:

Unguided transmission Media are methods that allow the transmission of data without the use of physical means to define the path it takes.

Unguided media provide a means for transmitting electromagnetic waves but do not guide them. It is also known as wireless transmission.

Ex-Through There ahi air, vacuum and sea water.

There are three types of unguided media

1. Radio waves.

2. Micro waves

3. Infrared

Radio Waves

Radio waves are the electromagnetic waves that are transmitted in all the direction of free space.

*Radio waves are unidirectional, i.e. the Signals are propagated in all the direction. The range in frequencies of radio waves is from 3 KHz to 1 KHz.

*In the case of radio Waves, the sent by sending and receiving antenna are not aligned, i.e. the wave the sending antenna can be received by any receiving antenna.

Ex-FM Radio, Television, cordless Phones.

Application of Radio waves-

*Radio Casting when wave is useful for multicasting when there is one sender and many receivers.

Advantages of Radio Transmission

*Radio transmission is mainly used for wide area networks and mobile Phones.

* Radio waves cover can large area, and penetrate.

*Radio transmission provides a higher transmission Rate

Micho waves.:-

There two types

* Terrestrial microwave

*Satellite microwave

Terrestrial microwave

Terrestrial microwave transmission is a technology that transmits the focused beam a radio signal from one ground based microwave transmission antenna to another.

*Micro waves are the electromagnetic waves frequency in the range from 1 GHz to 1000 GHz.

* Microwaves are unidirectional as the sending and receiving antenna is to be aligned, i.e. the waves sent by are the sending antennas are narrowly focused.

* In this case, antennas are mounted on the towers to send beam antenna to another which is a KM away.

* It works on the line of sight i transmission. i.e., the antennas mounted on the towers are the direct sight of each other.

Characteristics of microwave:-

*Frequency Range - The frequency range of terrestrial microwave is from 4-6 GHz to 21-23 GHz.

*Band width - It supports the bandwidth from 1 to 10. Mbps.

* Short Distance: It is inexpensive for Short distance..

* Long Distance: It is expensive as it requires a higher tower for a longer distance.

* Attenuation: Attenuation means loss of Signal; It is effected by environmental conditions and antenna size.

Advantages of microwave-

* Microwave transmission is cheaper than using cables.

* It is free from land acquisition as it does not require any land for the installation of cables.

* Communication over any land Oceans can be achieved by using microwave transmission.

Disadvantages of microwave :

A signal can be moved out of phase by using microwave transmission.

*A microwave transmission is susceptible to weather Condition. This means that any environmental changes such as rain, wind can distort the signal.

* Allocation of bandwidth is limited in the Case of the signal microwave transmission.

Satellite microwave Communication:

* A Satellite is a physical object that revolves around the earth at known height.

* Satellite Communication is more reliable nowadays as it offers more flexibility than cable and fiber optic systems.

* We can communicate with any point on the globe by using satellite communication.

How Does satellite works

The satellite accepts the signal that is transmitted from the earth station and it amplifies signal. The amplified signal is transmitted to another the earth station.

Advantages of satellite microwave communication-

*The coverage is area of a satellite microwave more than the terrestrial microwave.

* The transmission cost of the satellite is independent of the distance from the centre of the coverage area.

* Satellite Communication is used in mobile and wireless communication applications.

* It is easy to install.

* It is used in a wide variety of application such as weather forecasting radio /TV signal broadcasting, mobile communication etc..

Disadvantages of Satellite microwave communication-

* Satellite designing and development requires more time and higher cost.

*The Satellite needs to be monitored and controlled on regular Periods so that it remains in orbit.

*The life of the satellite is about 12-15 years. Due to this reason, another launch of the satellite has to be planned before it becomes non-functional.

Infrared-

* An infrared transmission is a wireless technology used for communication over short ranges.

*The frequency of the infrared in the range from 300GHz to 400GHz.

*It used for short-range Communication such as data transfer between two Cell Phones, TV remote operation, data transfer between computer and cell Phone resides in the same closed area.

Characteristics of Infrared-

*It supports high bandwidth and hence the data rate will be very high.

* Infrared wave cannot penetrate the walls.

* Infrared Communication provides better security with minimum interference.

*Infrared communication is unreliable outside the building because the sun rays will interfere with the infrared waves.

Modulation -

The Process converting low frequency range to high frequency range of signal is called modulation.

Signal-

Electromagnetic field used to convey data from one place to another.

Base band signal

A base band signal is a signal that can include frequencies that are very near zero, by comparison with its highest Frequency.

Carrier signals-

Carrier signal is a waveform that is modulated with an information bearing signal for the purpose of conveying Information.

Modulating signal:

After modulation Process to when the base band signal and carrier signal mixed for transmitting the information. Then that signal is called modulating signal.

Need of modulation

* To reduce the interference.

* To reduce antenna size .

* To allow the multiplexing of the different signal.

* modulation is important to transfer the signals over large distance.

Different Types of modulation:

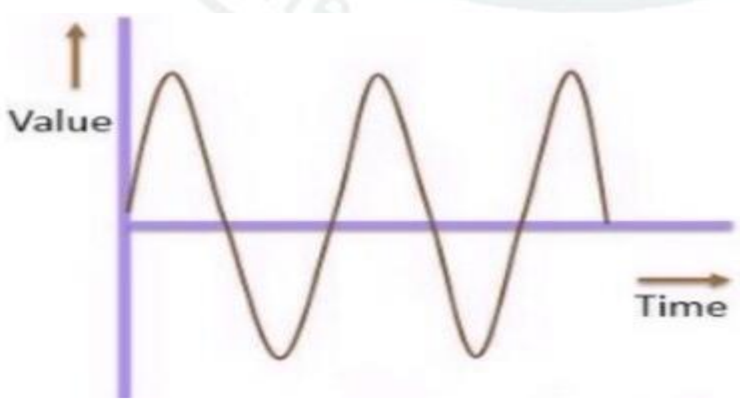
There are two types of modulation.

1.Analog modulation

2. Digital Modulation

Analog modulation-

In this modulation, a continuously varying sine wave used as a carrier that modulates the message signal or data signal.



There are three parameters can be altered to get modulation.

They are mainly amplitude, frequency, and Phase.

Types of analog modulation-

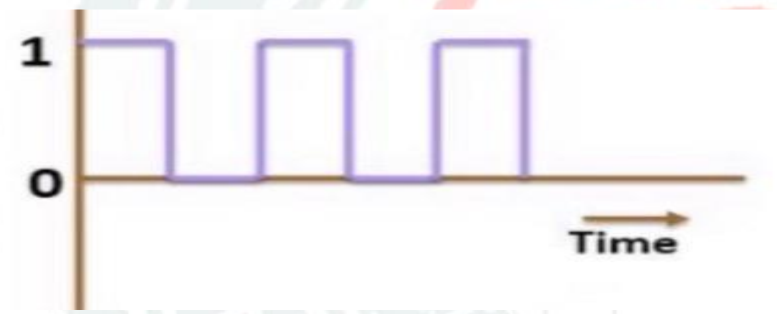
1. Amplitude Modulation

2.Frequency Modulation

3. Phase modulation

Digital Modulation

In digital modulation a message signal is converted from analog to digital message and then modulated by using a carrier wave.



Advantage of digital Modulation

* High noise immunity.

* Better quality and efficient Communication.

* Available bandwidth.

Basic concept of signals and its classification

Signal-

A signal is an electromagnetic or electrical current that carries data from one system or network to another.

Analog signal-

An analog signal is any continuous signal for which the time-varying feature of the signal is a representation of some other time varying quantity.

Advantages-

*Analog signals are easier to Process.

* Analog signals best suited for audio and video transmission .

* Analog signals are much higher density and can present more refined information.

* Analog signals useless bandwidth than digital signals.

* Analog signals provide a more accurate representation of change in physical Phenomena, such as sound, light temperature, position or pressure.

* Analog Communication systems are less sensitive in terms of electrical tolerance.

Disadvantages-

* Data transmission at long distances in may result in undesirable signal disturbances..

*Analog signals are subject to noise and distortion.

*Analog signals are generally lower quality signals than digital signals.

Digital signal –

A digital signal is a signal that represents data discrete as a sequence of values.

Advantages-

*Digital signal can convey information with less noise, distortion and interference.

* Digital circuits can be reproduced easily in mass quantities at comparatively low costs.

* Digital signal processing is more secure because digital information Can be easily encrypted and compressed.

* Digital signals can be easily stored on any magnetic media or optical media using semiconductor chips. * Digital signals can be transmitted over long distances.

Disadvantages-

* A higher bandwidth is required for digital communication.

*Digital communication systems and processing are typically more complex.

Analog digital conversion –

Sampling: Sample is a piece of data taken from the whole data which is continuous in the time domain. This discretization of analog signal is called Sampling.

Quantization: In quantization, the total amplitude range which the signal may occupy is divided into a number of standard levels.

Conversion of analog to Digital signal:

This means we have to convert analog (continuous time signal) in the form of digits.

* First of all, we jet sample of this signal according to sampling Theorem.

* We mark the time instants to t0, t1,t2 and so on, at equal time intervals a along the time axis.

* At each of these times, instants, the magnitude of the signal is measured and thus samples of the signal are taken.

* This means that it no longer is a continuous function of time, but rather it is a discrete - time signal.

* However, since the magnitude each sample can take any value in a continuous range, the signal is a an analog signal.

* The signal is defined only at the sampling instants.

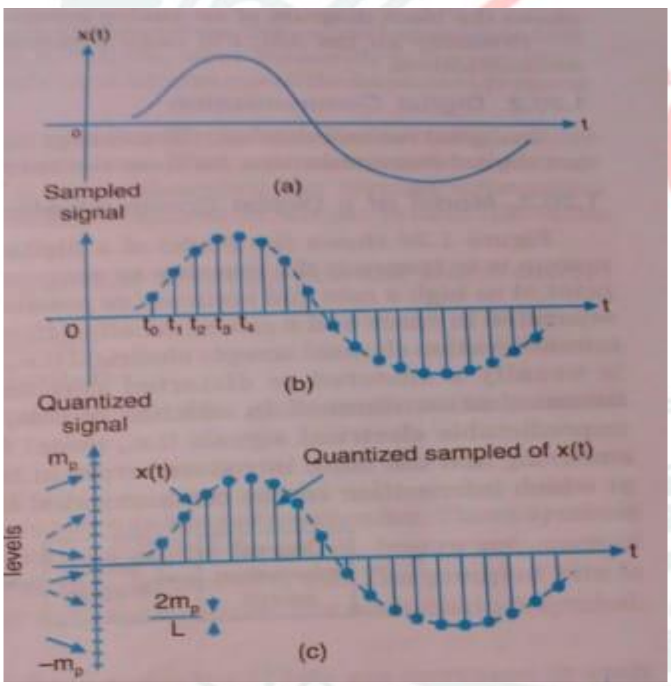
*This difficulty is neatly resolved by a process known as quantization.

* Amplitudes of the signal x (t) lie in the range (-mp, mp) which is partitioned into L intervals,

* 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.



Bandwidth Limitation -

Bandwidth- The Frequency range needed for a particular given transmission is known as bandwidth.

* The bandwidth Frequency for a particular transmission is always allocated by Some International regulation 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 band width of a communication system, the more is the information that can be transmitted.

* The human ear is contained in a frequency range up to 15 KH_2.

* AM radio the maximum modulating Frequency range up to 5KH_2 and hence maximum bandwidth of Am trans mission 10kHz.

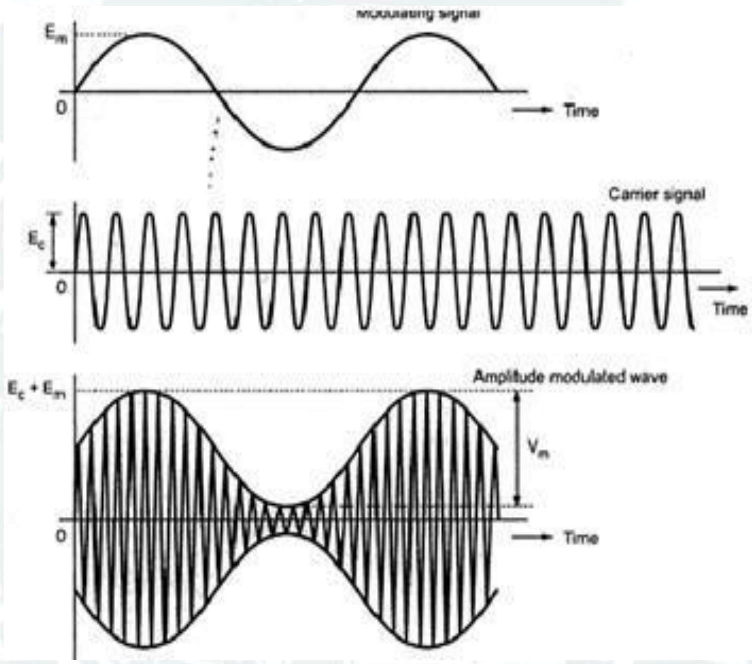
* The bandwidth allocated to a FM transmission is about 200 kHz

* This means FM system has a better fidelity than an Am system.

CHAPTER-2 Amplitude Modulation

Amplitude Modulation is the changing the amplitude of the carrier signal with respect tothe instantaneous change in message signal.

The amplitude modulated wave form, its envelope and itsfrequencyspectrum and bandwidth. Fig (a) Sinosoidal modulation signal (b)High frequency carrier (c)



AMsignal.

Let us represent the modulating signal by e_m and it is given as,

$$e_m = E_m \sin \omega_m t \quad \text{... (1.2.1)}$$

and carrier signal can be represented by e_c as,

$$e_c = E_c \sin \omega_c t \quad \text{... (1.2.2)}$$

Here E_m is maximum amplitude of modulating signal

E_c is maximum amplitude of carrier signal

ω_m is frequency of modulating signal

and ω_c is frequency of carrier signal.

Using the above mathematical expressions for modulating and carrier signals, we can create a new mathematical expression for the complete modulated wave. It is given as,

$$E_{AM} = E_c + e_m$$

$$= E_c + E_m \sin \omega_m t \quad \text{by putting } e_m \text{ from equation (1.2.1)}$$

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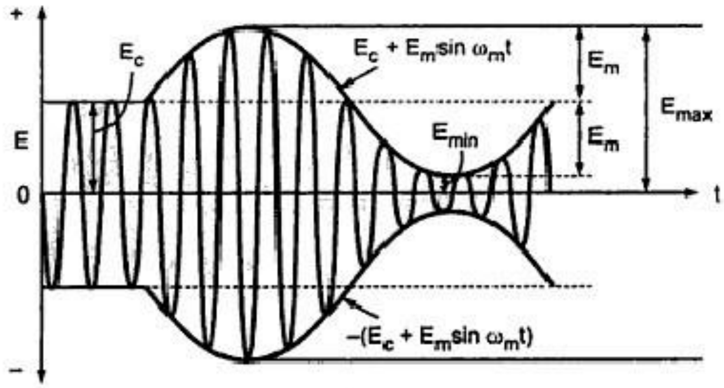

Fig. 1.2.2 AM wave

It is clear from the above signal that the modulating signal rides upon the carrier signal. From above figure we can write,

$$E_m = \frac{E_{max} - E_{min}}{2} \qquad \text{... (1.2.5)}$$

∴ The instantaneous value of the amplitude modulated wave can be given as,

$$e_{AM} = E_{AM} \sin\theta$$
$$= E_{AM} \sin\omega_c t$$

$$\therefore \quad \boxed{e_{AM} = (E_c + E_m \sin\omega_m t)\sin\omega_c t} \quad \text{... (1.2.3)}$$

This is an equation of AM wave.

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Modulation Index and Percent Modulation

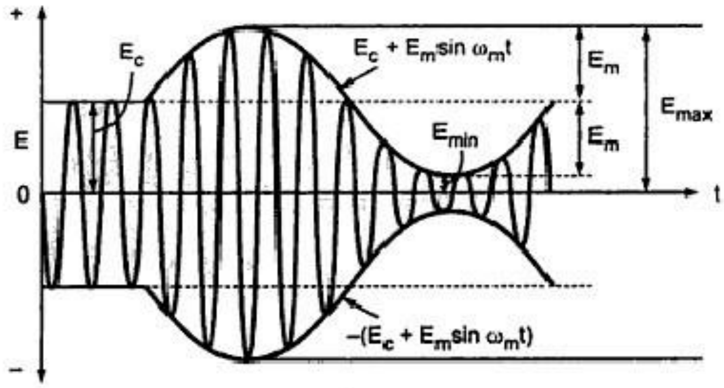
The ratio of maximum amplitude of modulating signal to maximum amplitude of carrier signal is called modulation index. i.e.,

$$\boxed{\text{Modulation index, } m = \frac{E_m}{E_c}} \quad \text{... (1.2.4)}$$

Value of E_m must be less than value of E_c to avoid any distortion in the modulated signal. Hence maximum value of modulation index will be equal to 1 when $E_m = E_c$. Minimum value will be zero. If modulation index is higher than 1, then it is called *over modulation*. Data is lost in such case. When modulation index is expressed in percentage, it is also called percentage modulation.

Calculation of modulation index from AM waveform :

Fig. 1.2.2 shows the AM waveform. This is also called time domain representation of AM signal.



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and

$$E_c = E_{max} - E_m \qquad \text{... (1.2.6)}$$

$$= E_{max} - \frac{E_{max} - E_{min}}{2} \text{ by putting for } E_m \text{ from equation (1.2.5)}$$

$$= \frac{E_{max} + E_{min}}{2} \qquad \text{... (1.2.7)}$$

Taking the ratio of equation (1.2.5) and above equation,

$$m = \frac{E_m}{E_c} = \frac{\frac{E_{max} - E_{min}}{2}}{\frac{E_{max} + E_{min}}{2}}$$

$$\therefore \qquad \boxed{m = \frac{E_{max} - E_{min}}{E_{max} + E_{min}}} \qquad \text{... (1.2.8)}$$

This equation gives the technique of calculating modulation index from AM wave.

Frequency Spectrum and Bandwidth

The modulated carrier has new signals at different frequencies, called side frequencies or sidebands. They occur above and below the carrier frequency.

i.e.

$$f_{USB} = f_c + f_m$$

$$f_{LSB} = f_c - f_m$$

Here f_c is carrier frequency and

f_m is modulating signal frequency

f_{LSB} is lower sideband frequency

Consider the expression of AM wave given by equation (1.2.3), i.e.,

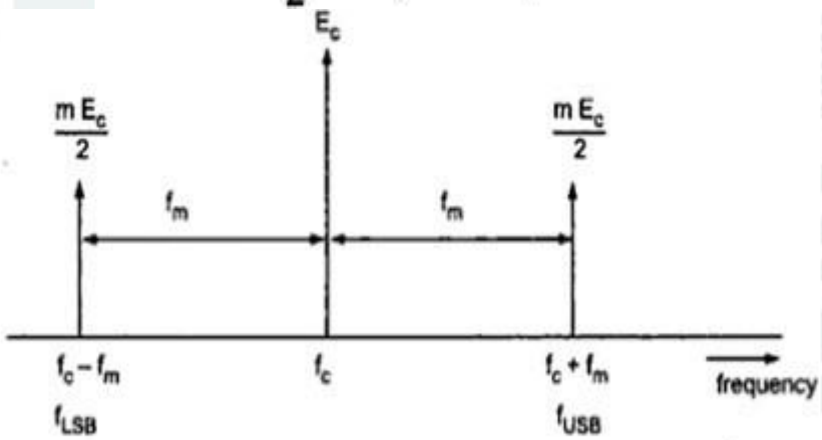
$$e_{AM} = (E_c + E_m \sin \omega_m t) \sin \omega_c t \qquad \text{... (1.2.9)}$$

We know that $m = \frac{E_m}{E_c}$ from equation (1.2.4). Hence we have $E_m = m E_c$. Putting this value of E_m in above equation we get,

$$\begin{aligned} e_{AM} &= (E_c + m E_c \sin \omega_m t) \sin \omega_c t \\ &= E_c (1 + m \sin \omega_m t) \sin \omega_c t \\ &= E_c \sin \omega_c t + m E_c \sin \omega_m t \sin \omega_c t \qquad \text{... (1.2.10)} \end{aligned}$$

We know that $\sin(A)\sin(B) = \frac{1}{2}\cos(A-B) - \frac{1}{2}\cos(A+B)$. Applying this result to last term in above equation we get,

$$\begin{aligned} e_{AM} &= E_c \sin \omega_c t + \frac{m E_c}{2} \cos(\omega_c - \omega_m) t \\ &- \frac{m E_c}{2} \cos(\omega_c + \omega_m) t \qquad \text{... (1.2.11)} \end{aligned}$$



Frequency domain Representation of AM Wave

<u>Amplitude Modulation of Power distribution</u>:

AM signal has three components : Unmodulated carrier, lower sideband and upper sideband. Hence total power of AM wave is the sum of carrier power P_c and powers in the two sidebands P_{USB} and P_{LSB}. i.e.,

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

$$= \frac{E_{carr}^2}{R} + \frac{E_{LSB}^2}{R} + \frac{E_{USB}^2}{R} \qquad \text{... (1.2.15)}$$

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$$= \frac{E_{carr}^2}{R} + \frac{E_{LSB}^2}{R} + \frac{E_{USB}^2}{R} \qquad \text{... (1.2.15)}$$

AM Power Distribution:

Here all the three voltages are rms values and R is characteristic impedence of antenna in which the power is dissipated. The carrier power is,

$$P_c = \frac{E_{carr}^2}{R} = \frac{\left(E_c / \sqrt{2}\right)^2}{R}$$

$$= \frac{E_c^2}{2R} \qquad \text{... (1.2.16)}$$

The power of upper and lower sidebands is same. i.e.,

$$P_{LSB} = P_{USB} = \frac{E_{SB}^2}{R}$$ Here E_{SB} is rms voltage of sidebands.

From equation (1.2.13) we know that the peak amplitude of both the sidebands is $\frac{mE_c}{2}$. Hence,

$$E_{SB} = \frac{mE_c / 2}{\sqrt{2}}$$

$$\therefore \quad P_{LSB} = P_{USB} = \left(\frac{mE_c / 2}{\sqrt{2}}\right)^2 \times \frac{1}{R}$$

This equation relates total power of AM wave to carrier power, Maximum Value of modulation index, m=1 to avoid distortion. At this value of modulation

$$= \frac{m^2 E_c^2}{8R} \quad \text{... (1.2.17)}$$

Hence the total power (equation 1.2.15) becomes,

$$P_{Total} = \frac{E_c^2}{2R} + \frac{m^2 E_c^2}{8R} + \frac{m^2 E_c^2}{8R}$$

$$\boxed{P_{Total} = P_c\left(1 + \frac{m^2}{2}\right)} \quad \text{... (1.2.19)}$$

$$\frac{P_{Total}}{P_c} = 1 + \frac{m^2}{2} \quad \text{... (1.2.20)}$$

index, Ptotal = 1.5 Pc. From the above equation we have

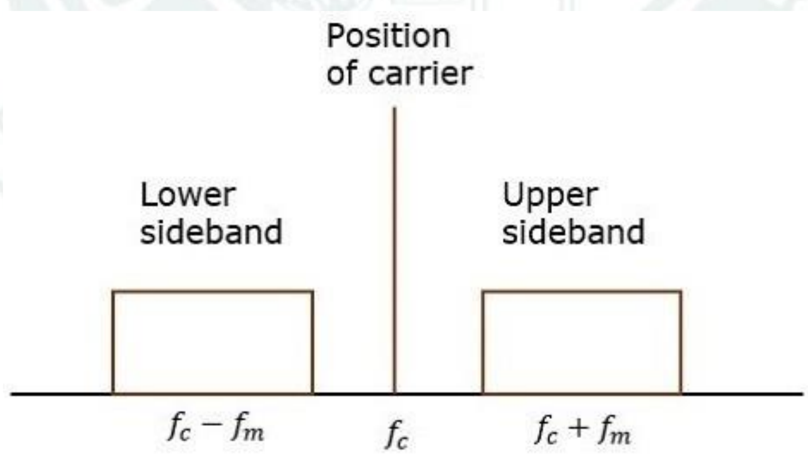
$$\frac{m^2}{2} = \frac{P_{Total}}{P_c} - 1$$

$$\therefore \quad m = \sqrt{2\left(\frac{P_{total}}{P_c} - 1\right)}$$

Double Side Band AM:

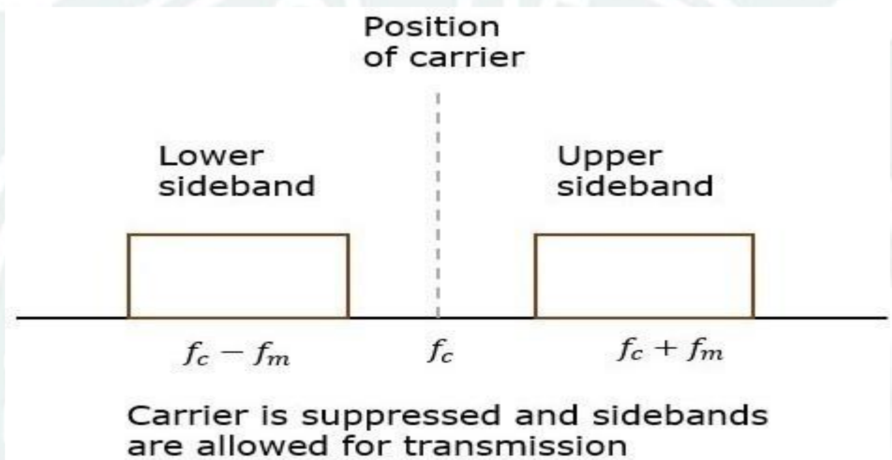
In the process of Amplitude Modulation, the modulated wave consists of the carrier wave and two sidebands. The modulated wave has the information only in the sidebands. **Sideband** is nothing but a band of frequencies, containing power, which are the lower and higher frequencies of the carrier frequency.

The transmission of a signal, which contains a carrier along with two sidebands can be termed as **Double Sideband Full Carrier** system or simply **DSBFC**. It is plotted as shown in the following figure.



However, such a transmission is inefficient. Because, two-thirds of the power is being wasted in the carrier, which carries no information.

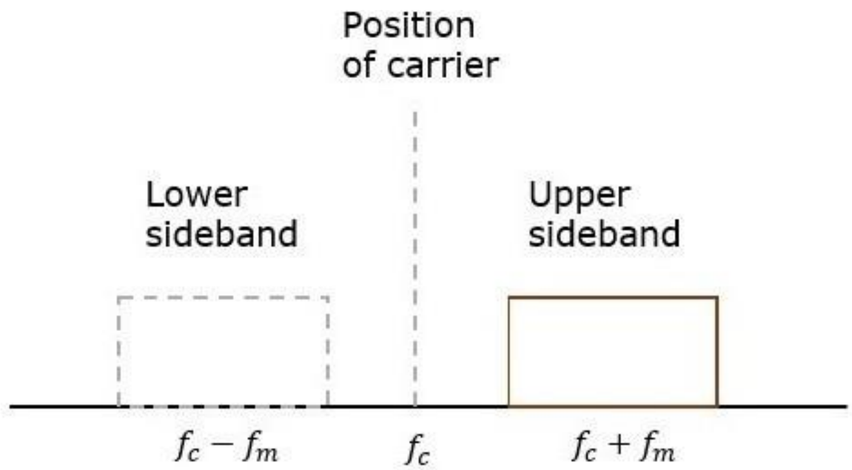
If this carrier is suppressed and the saved power is distributed to the two sidebands, then such a process is called as **Double Sideband Suppressed Carrier** system or simply **DSBSC**. It is plotted as shown in the following figure.



Single Sideband Suppressed Carrier

The DSBSC modulated signal has two sidebands. Since, the two sidebands carry the same information, there is no need to transmit both sidebands. We can eliminate one sideband.

The process of suppressing one of the sidebands along with the carrier and transmitting a single sideband is called as **Single Sideband Suppressed Carrier** system or simply **SSBSC**. It is plotted as shown in the following figure



Carrier and a sideband are suppressed and a single sideband is allowed for transmission

In the above figure, the carrier and the lower sideband are suppressed. Hence, the upper sideband is used for transmission. Similarly, we can suppress the carrier and the upper sideband while transmitting the lower sideband.

This SSBSC system, which transmits a single sideband has high power, as the power allotted for both the carrier and the other sideband is utilized in transmitting this Single Sideband.

VSBSC Modulation

In the previous chapters, we have discussed SSBSC modulation. SSBSC modulated signal has only one sideband frequency. Theoretically, we can get one sideband frequency component completely by using an ideal band pass filter. However, practically we may not get the entire sideband frequency component. Due to this, some information gets lost.

To avoid this loss, a technique is chosen, which is a compromise between DSBSC and SSBSC. This technique is known as **Vestigial Side Band Suppressed Carrier (VSBSC)** technique. The word "vestige" means "a part" from which, the name is derived.

VSBSC Modulation is the process, where a part of the signal called as vestige is modulated along with one sideband. The frequency spectrum of VSBSC wave is shown in the following figure.

Generation of DSB+C AM by Square Law Modulation

Square law diode modulation makes use of non-linear current-voltage characteristics of diode. This method is suited for low voltage levels as the current-voltage characteristic of diode is highly non-linear in the low voltage region. So the diode is biased to operate in this non-linear region for this application. A DC battery V_c is connected across the diode to get such a operating point on the characteristic. When the carrier and modulating signal are applied at the input of diode, different frequency terms appear at the output of the diode. These when applied across a tuned circuit tuned to carrier frequency and a narrow bandwidth just to allow the two pass-bands, the output has the carrier and the sidebands only which is essentially the DSB+C AM signal.

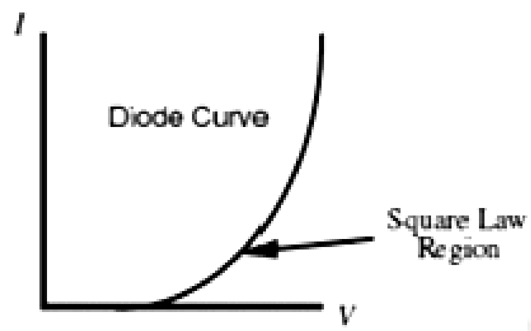


Figure 4 Current-voltage characteristic of diode

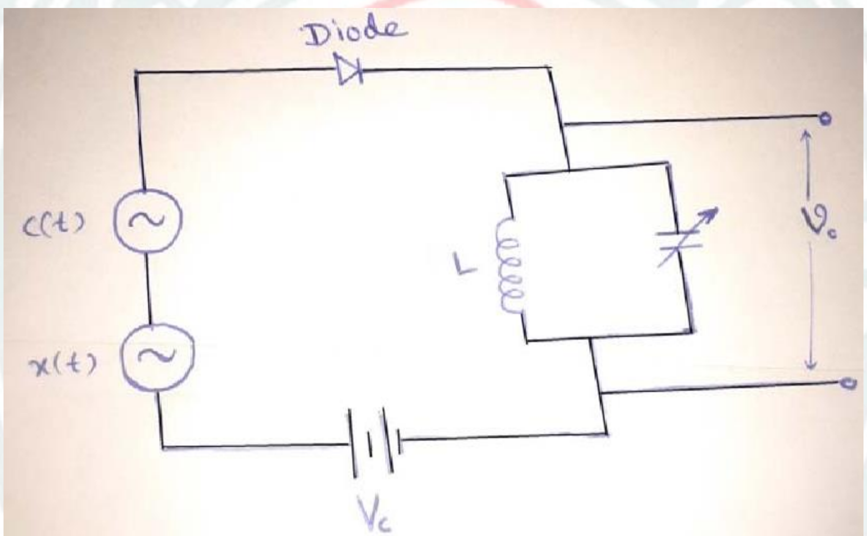


Figure 5 Square Law Diode Modulator

The non-linear current voltage relationship can be written in general as:

$i = av + bv^2$

I this application $v = c(t) + x(t)$

So

$$i = a[ACos(2\pi f_c t) + x(t)] + b[ACos(2\pi f_c t) + x(t)]^2$$

$$\Rightarrow i = a\,ACos(2\pi f_c t) + a\,x(t) + bA^2 Cos^2(2\pi f_c t) + b\,x^2(t) + 2bA\,x(t)\,Cos(2\pi f_c t)$$

$$\Rightarrow i = \boxed{a\,ACos(2\pi f_c t)} + a\,x(t) + \frac{bA^2}{2} Cos(2\pi(2f_c)t) + \frac{bA^2}{2} + b\,x^2(t) + \boxed{2bA\,x(t)\,Cos(2\pi f_c t)}$$

Out of the above frequency terms, only the boxed terms have the frequencies in the passband of the tuned circuit, and hence will be at the output of the tuned circuit. There is carrier frequency term and the sideband term which comprise essentially a DSB+C AM signal.

Demodulation of DSB+C by Square Law Detector

It can be used to detect modulated signals of small magnitude, so that the operating point may be chosen in the non-linear portion of the V-I characteristic of diode. A DC supply voltage is used to get a fixed operating point in the non-linear region of diode characteristics. The output diode current is hence

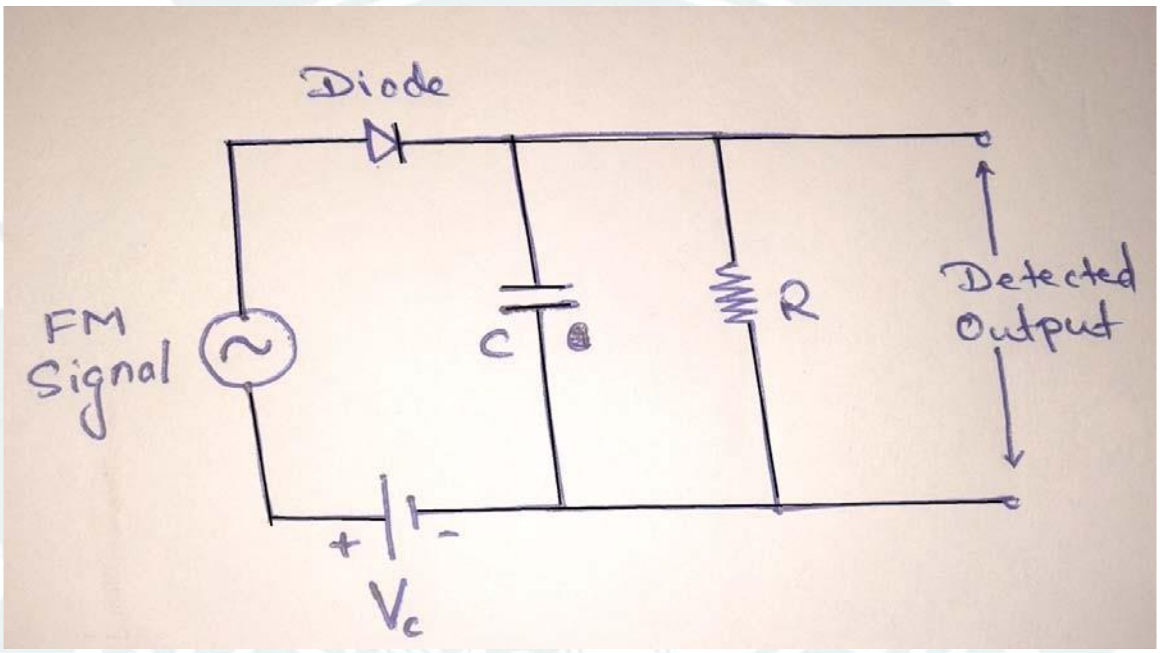


Figure 6 Square Law Detector

given by the non-linear expression:

$$i = av_{FM}(t) + bv^2_{FM}(t)$$

Where $v_{FM}(t) = [A + x(t)]\cos(2\pi f_c t)$

This current will have terms at baseband frequencies as well as spectral components at higher frequencies. The low pass filter comprised of the RC circuit is designed to have cut-off frequency as the highest modulating frequency of the band limited baseband signal. It will allow only the baseband frequency range, so the output of the filter will be the demodulated baseband signal.

Linear Diode Detector or Envelope Detector

This is essentially just a half-wave rectifier which charges a capacitor to a voltage to the peak voltage of the incoming AM waveform. When the input wave's amplitude increases, the capacitor voltage is increased via the rectifying diode quickly, due a very small RC time-constant (negligible *R*) of the charging path. When the input's amplitude falls, the capacitor voltage is reduced by being discharged by a 'bleed' resistor *R* which causes a considerable RC time constant in the discharge path making discharge process a slower one as compared to charging. The voltage across *C* does not fall appreciably during the small period of negative half-cycle, and by the time next positive half cycle appears. This cycle again charges the capacitor *C* to peak value of carrier voltage and thus this process repeats on. Hence the output voltage across capacitor *C* is a spiky envelope of the AM wave, which is same as the amplitude variation of the modulating signal.

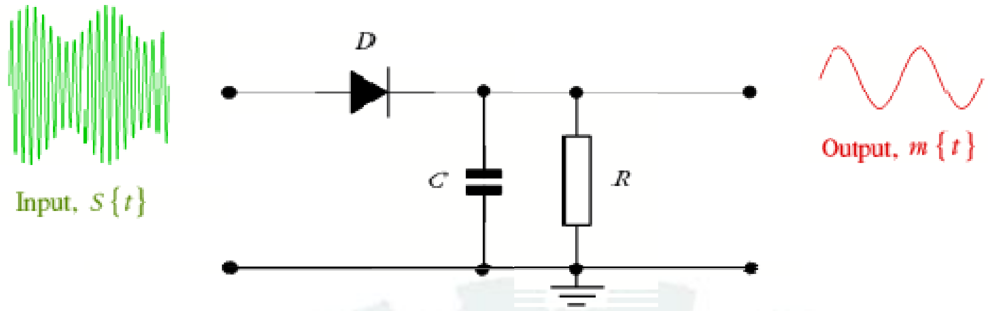


Figure 7 Envelope Detector

Double Sideband Suppressed Carrier(DSB-SC)

If the carrier is suppressed and only the sidebands are transmitted, this will be a way to saving transmitter power. This will not affect the information content of the AM signal as the carrier component of AM signal do not carry any information about the baseband signal variation. A DSB+CAM signal is given by:

$$s_{DSB+C}(t) = A\cos(2\pi f_c t) + x(t)\cos(2\pi f_c t)$$

So, the expression for DSB-SC where the carrier has been suppressed can be given as:

$$s_{DSB-SC}(t) = x(t)\cos(2\pi f_c t)$$

Therefore, a DSB-SC signal is obtained by simply multiply ng modulating signal *x(t)* with the carrier signal. This is accomplished by a **product modulator** or **mixer**.

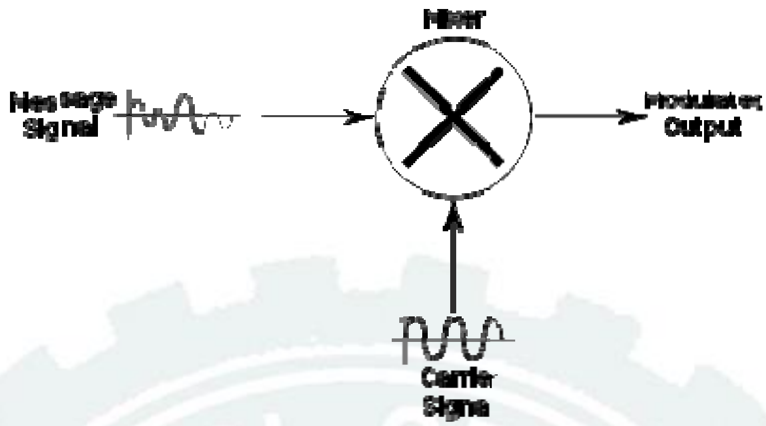


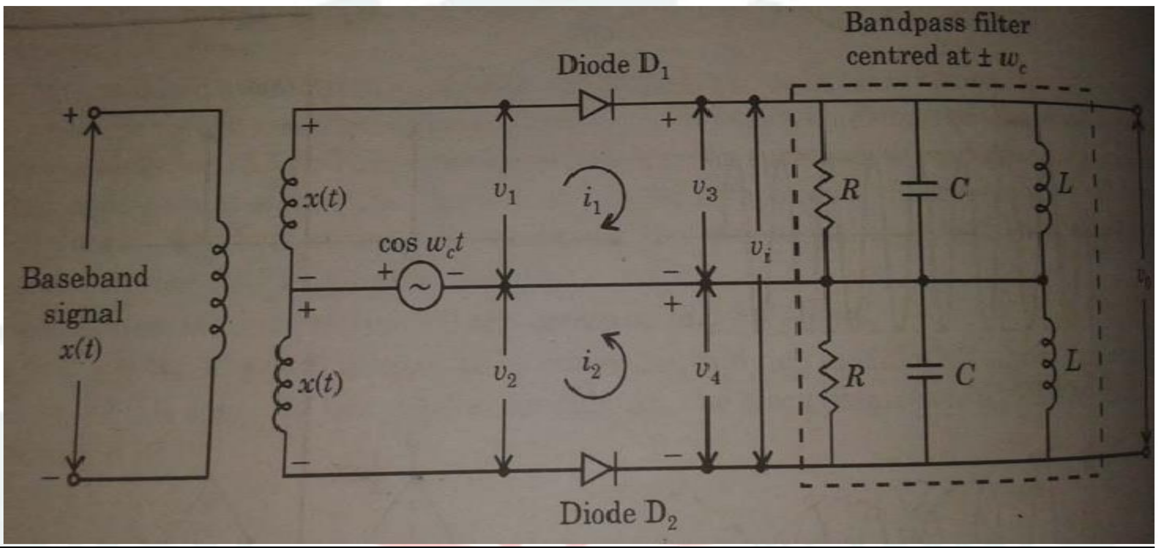
Figure 8 Product Modulator

Difference from the the DSB+C being only the absence of carrier component, and since DSBSC has still both the sidebands, spectral span of this DSBSC wave is still $f_c f_m$ to $f_c f_m$, hence has a bandwidth of $2 f_m$.

Generation of DSB-SC Signal

A circuit which can produce an output which is the product of two signals input to it is called a product modulator. Such an output when the inputs are the modulating signals and the carrier signal is a DSBSC signal. One such product modulator is a balanced modulator.

Balanced modulator:



$v_1 = Cos(2\pi f_c t) + x(t)$

$v_2 = Cos(2\pi f_c t) - x(t)$

For diode D_1, the nonlinear v-i relationship becomes:

$$i_1 = av_1 + bv_1^2 = a[Cos(2\pi f_c t) + x(t)] + b[Cos(2\pi f_c t) + x(t)]^2$$

Similarly, for diode D_2,

$$i_2 = av_2 + bv_2^2 = a[Cos(2\pi f_c t) - x(t)] + b[Cos(2\pi f_c t) - x(t)]^2$$

$$v_i = v_3 - v_4 = (i_1 - i_2)R$$

Now, $\therefore v_i = 2R[ax(t) + 2bx(t)\,Cos(2\pi f_c t)]$ (substituting for i_1 and i_2)

This voltage is input to the bandpass filter centre frequency f_c and bandwidth $2f_m$. Hence it allows the component corresponding to the second term of the v_i, which is our desired DSB-SC signal.

Demodulation of DSBSC signal

Synchronous Detection: DSB-SC signal is generated at the transmitter by frequency up-translating the baseband spectrum by the carrier frequency f_c. Hence the original baseband signal can be recovered by frequency down-translating the received modulated signal by the same amount. Recovery can be achieved by multiplying the received signal by synchronous carrier signal and then low-pass filtering.

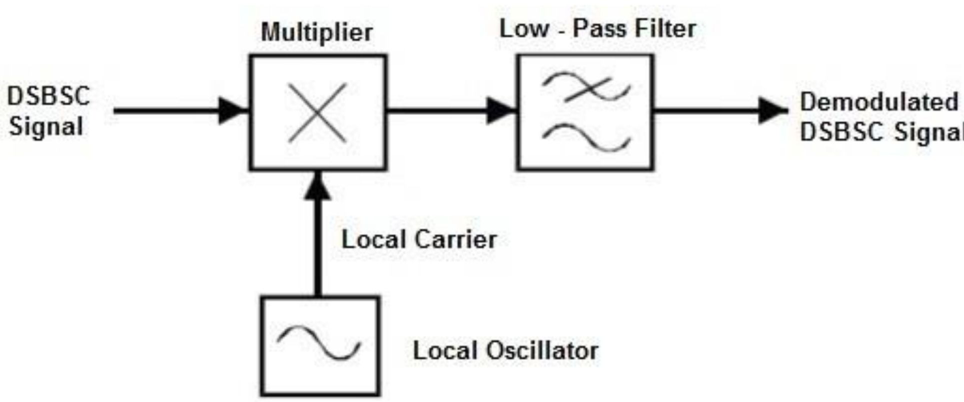


Figure 9 Synchronous Detection of DSBSC

Let the received DSB-SC signal is :

$$r(t) = x(t)\text{Cos}(2\pi f_C t)$$

So after carrier multiplication, the resulting signal:

$$e(t) = x(t)\,\text{Cos}(2\pi\, f_C\, t).\text{Cos}(2\pi\, f_C\, t)$$

$$\Rightarrow e(t) = x(t)\,\text{Cos}^2(2\pi\, f_c\, t)$$

$$\Rightarrow e(t) = \frac{1}{2}x(t)\left[1 + \text{Cos}(2\pi\,(2 f_c)\,t)\right]$$

$$\Rightarrow e(t) = \frac{1}{2}x(t) + \frac{1}{2}x(t)\,\text{Cos}(2\pi\,(2 f_c)\,t)$$

The low-pass filter having cut-off frequency f_m will only allow the baseband term $\frac{1}{2}x(t)$, which is in the pass-band of the filter and is the demodulated signal.

Single Sideband Suppressed Carrier (SSB-SC) Modulation

The lower and upper sidebands are uniquely related to each other by virtue of their symmetry about carrier frequency. If an amplitude and phase spectrum of either of the sidebands is known, the other sideband can be obtained from it. This means as far as the transmission of information is concerned, only one sideband is necessary. So bandwidth can be saved if only one of the sidebands is transmitted, and such a AM signal even without the carrier is called as Single Sideband Suppressed Carrier signal. It takes half as much bandwidth as DSB-SC or DSB+C modulation scheme.

For the case of single-tone baseband signal, the DSB-SC signal will have two sidebands :

The lower side-band: $Cos(2\pi(f_c - f_m)t) = Cos(2\pi f_m t)Cos(2\pi f_c t) + Sin(2\pi f_m t)Sin(2\pi f_c t)$

And the upper side-band: $Cos(2\pi(f_c + f_m)t) = Cos(2\pi f_m t)Cos(2\pi f_c t) - Sin(2\pi f_m t)Sin(2\pi f_c t)$

If any one of these sidebands is transmitted, it will be a SSB-SC waveform:

$$s(t)_{SSB} = Cos(2\pi f_m t)Cos(2\pi f_c t) \pm Sin(2\pi f_m t)Sin(2\pi f_c t)$$

Where (+) sign represents for the lower sideband, and (-) sign stands for the upper sideband. The modulating signal here is $x(t) = Cos(2\pi f_m t)$, so let $x_h(t) = Sin(2\pi f_m t)$ be the Hilbert Transform of $x(t)$. The Hilbert Transform is obtained by simply giving $\left(-\frac{\pi}{2}\right)$ to a signal. So the expression for SSB-SC signal can be written as:

$$s(t)_{SSB} = x(t)Cos(2\pi f_c t) \pm x_h(t)Sin(2\pi f_c t)$$

Where $x_h(t)$ is a signal obtained by shifting the phase of every component present in $x(t)$ by $\left(-\frac{\pi}{2}\right)$.

Generation of SSB-SC signal

<u>Frequency Discrimination Method</u>:

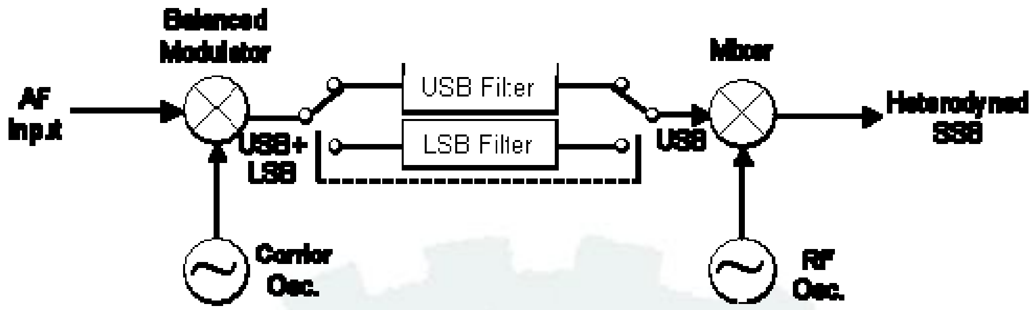


Figure 10 Frequency Discrimination Method of SSB-SC Generation

The filter method of SSB generation produces double sideband suppressed carrier signals (using a balanced modulator), one of which is then filtered to leave USB or LSB. It uses two filters that have different passband centre frequencies for USB and LSB respectively. The resultant SSB signal is then mixed (heterodyned) to shift its frequency higher.

Limitations:

I. This method can be used with practical filters only if the baseband signal is restricted at its lower edge due to which the upper and lower sidebands do not overlap with each other. Hence it is used for speech signal communication where lowest spectral component is 70 Hz and it may be taken as 300 Hz without affecting the intelligibility of the speech signal.
II. The design of band-pass filter becomes quite difficult if the carrier frequency is quite higher than the bandwidth of the baseband signal.

<u>Phase-Shift Method</u>:

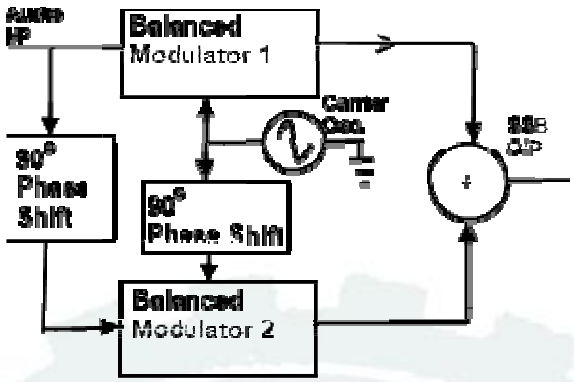


Figure 11 Phase shift method of SSB-SC generation

CHAPTER-3 ANGLE MODULATION SYSTEM

Concept of Angle Modulation

Angle Modulation may be defined as the process in which the total phase angle a carrier wave is varied in accordance with instantaneous value of the modulating or message signal while keeping amplitude and carrier. **Mathematical Expression:**

Let us consider that an unmodulated carrier signal is expressed as

$C(t) = A \cos (Wct + \emptyset_0)$.

A= Amplitude of carrier

ωc = Carrier frequency

$\emptyset_0$ = Same phase angle

substituting = $\omega ct + \emptyset_0 = \phi$

we get $C(t) = A \cos \emptyset$

where $\phi = (\omega ct + \emptyset_0)$

Types of Angle Modulation. –

We can very this phase angle ϕ in two ways and thus there are two types of are angle modulation.

1. Phase modulation (PM)

2. Frequency modulation (FM)

Phase modulation:

Phase modulation (PM) that type angle modulation in which the phase angle is varied linearly with a of baseband or modulating signal X(t) about an unmodulated phase angle (ω ct + $\emptyset_0$). This means that in phase modulation, the instantaneous value of the phase angle is equal to the phase angle of unmodulated Carrier (ω ct + $\emptyset_0$) plus a time-varying component which is proportional to the modulating signal x(t).

Frequency Madulation:

Frequency modulation is that type of angle modulation in which instantaneous frequency ω_i is varied linearly with a message or baseband signal x(t) the about an unmodulated carrier frequency ω_c. This means that the instantaneous value of the angular frequency ω_i will be equal to the carrier frequency ω_c plus a time-varying component proportional to the base band signal x(t).

Principle of Frequency Modulation

Frequency modulation uses the information signal, Vm (t) to vary the carrier frequency within some small range about its original value.

Here are the three signals in mathematical form:

Information- Vm(t)

Carrier – Vc(t) = Vc0 sin (2Pfc t +f)

FM - Vfm (t) = Vco Sin (2P [fc + (DF/ Vmo) vm (t)]t +f)

We have replaced the carrier frequency term with a time-varying frequency .

→We have also introduced a new term: Df the peak frequency deviation.

→In this form, you should be able to see that the carrier frequency term: fc + (DF/Vmo) Vm (t) now varies between the extremes of fc –Df and fc + Df.

The interpretation of Df becomes Clear: it is the farthest away the original frequency that the from the FM signal can be.

→We can also define a modulation index for FM, analogous to

Am:b =DF/fm,

where fm is the maximum modulating frequency used.

<u>Frequency spectrum</u>

→A spectrum represents the relative amounts of different frequency components in any signal.

→It's like the display on the graphic equalizer in your stereo which has leds showing the relative amount of bass, midrange and treble.

→ These correspond directly to increasing frequencies (treble being the high frequency components).

→It is a well-known fact of mathematics, that any function (signal) can be decomposed Components. into purely sinusoidal components.

→In technical terms, the sines and cosines form a complete. Set of function, also known as basis in the infinite-dimensional vector space of read-valued functions.

→Given that any signal can be thought to be made up of sinusoidal signals, the spectrum then represent the recipe card of how to make the signal from sinusoids.

Mathematical analysis of FM and PM

Let modulating signal represented as

$V_m(t)\ E_m \cos \omega_m t$

Carrier wave :

$V_c(t) = E_c \cos \omega_c t$

Angle modulated wave is given by

$m(t) = E_c \cos[\omega_c t + \varnothing\, t)]$ - - (ii)

where

E_m= Peak of modulating signal

E_c= Peak of carrier signal

ω_m= Angular frequency of modulating signal

ω_c = Angular frequency of carrier signal

$\varnothing(t)$ = Instantaneous Phase deviation,

= Instantaneous change in phase of carrier at given instant of is time is $\varnothing(t)$

d(t)=Instantaneous phase=Phase of the carrier at an instant of time $\omega_c + \varnothing(t)$

Modulation index of FM-

The modulation index of FM is defined as the ratio of the frequency deviation! of the carrier to the frequency of the modulating signal.

M_f-modulation index of FM= $\Delta f/f_m$

Side Band of FM Signal-

→Bandwidth is one of the main elements of FM signal. In Fm signal, the sidebands will extend either side bands will extend either side which will extend to infinity, however, the strength of them drop away.

→ Auspiciously, it is potential to restrict the BW of an FM signal without changing its value excessively.

→In FM it is not so simple.FM signal spectrum is quite complex and will have an infinite number of side bands as shown in figure.

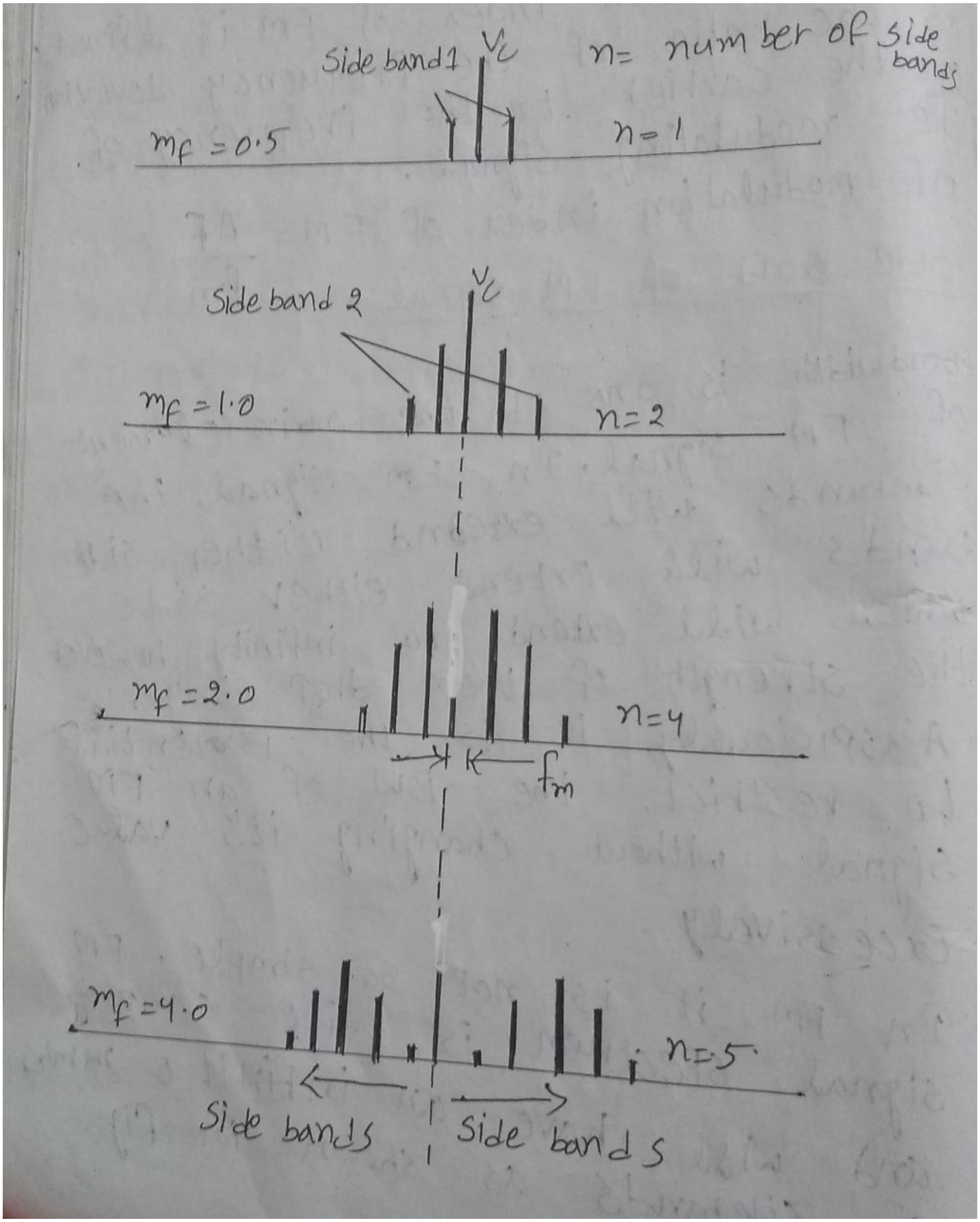
→ This fig. given an idea; how spectrum expands index increases. as the modulation index increases.

→Side bands are separated from the carrier fc ± fm, fc ± 2fm, fc± 3fm and so on.

→only the first few sidebands will contain the major share of the power and therefore only these few bands are considered to be significant sidebands.

->AS a rule of thumb, often termed as Carson's rule, 98% of the signal power in FM is contained within a bandwidth equal to the deviation frequency, plus the Modulation frequency doubled Carson's rule.

Bandwidth of FM BWFM = 2 [AF +fm] = 2 fm [mf +1]


Side band1
V_c
n= number of side bands
$m_f = 0.5$
n=1
Side band 2
V_c
$m_f = 1.0$
n=2
$m_f = 2.0$
n=4
f_m
$m_f = 4.0$
n=5
Side bands
Side bands

Difference Between AM and FM

S.No	Parameters	AM	FM
1.	Full form	Amplitude modulation	Frequency modulation
2.	Origin	AM method of audio transmission was successfully carried out in the mid-1870s.	FM radio was developed in the United States in the 1930s by Edwin Armstrong.
3.	Modulating differences	In AM, a radio wave is known as the "carrier" or "carrier wave" is modulated in amplitude by the signal that is to be transmitted.	In FM, a radio wave is known as the "carrier" or "carrier wave" is modulated in frequency by the signal that is to be transmitted.
4.	Constant parameters	The frequency and phase remain the same.	The amplitude and phase remain the same.
5.	Quality	AM has poorer sound quality, and a lower bandwidth but is cheaper and can be transmitted over long distances as it has a lower bandwidth that is why it can hold more stations available in any frequency range.	FM is less affected by interference, but FM signals are impacted by physical barriers. They have a better sound quality due to higher bandwidth.
6.	Frequency range	AM radio ranges from 535 to 1700 kHz or up to 1200 bits per second.	FM radio ranges in a higher spectrum from 88.1 to 108.1MHz. or up to 1200 to 2400 bits per second.
7.	Bandwidth BW	BW is much less than FM. BW = 2 . fm	BW is large. Hence a wide channel is required. BW = 2 x (δ + fm)

8.	Bandwidth requirements	Bandwidth is less than FM or PM and doesn't depend upon the modulation index. Bandwidth requirement is twice the highest modulating frequency. In AM radio broadcasting, if the modulating signal has a bandwidth of 15 kHz, then the bandwidth of an amplitude-modulated signal is 30 kHz.	Bandwidth requirement is greater and depends upon the modulating. .Bandwidth requirement is twice the sum of the modulating signal frequency and the frequency deviation. Let's say, if the frequency deviation is 75kHz and the modulating signal frequency is 15kHz, the bandwidth required is 180kHz.
9.	No of Sidebands	The number of sidebands are constant and equal to 2.	The number of sidebands having significant amplitude depends upon the modulation index

Generation of FM :

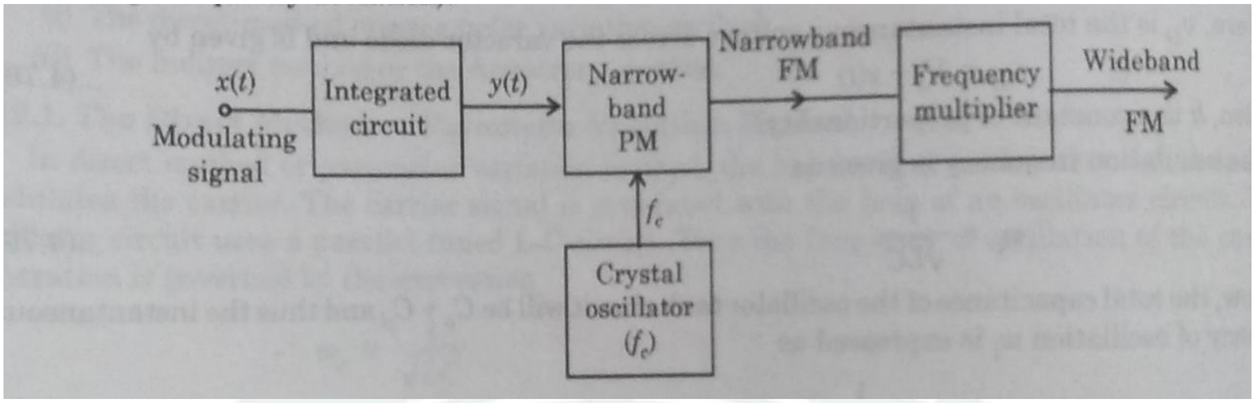
The FM modulator circuits used for generating FM signals may be put into two categories .

1. The direct method or parameter variation

2. The indirect Method or Armstrong Method.

<u>Armstrong Method</u>

In Armstrong method of FM generation we can get very high frequency stability since in this case the Crystal oscillator may be used as a Carrier frequency generator.



→ The working principle of Armstrong method is to generate a narrowband FM indirectly by utilizing the Phase modulation technique and then changing this narrowband FM into a wide band FM.

→Since in narrowband FM the modulation index is small, therefore the distortion is FM. very low in narrowband FM.

→ Here we prefer Phase modulation technique because it's generation

is easy.

→The multiplier circuit apart from multiplying the carrier frequency also increases the frequency deviation and hence the narrow band FM converted into wideband FM.

Demodulation of FM-

There are two type of FM detector.

1. Foster-Seely detector

2.Ratio detector

Foster-Seely Detector :

This used type of detector is most widely.

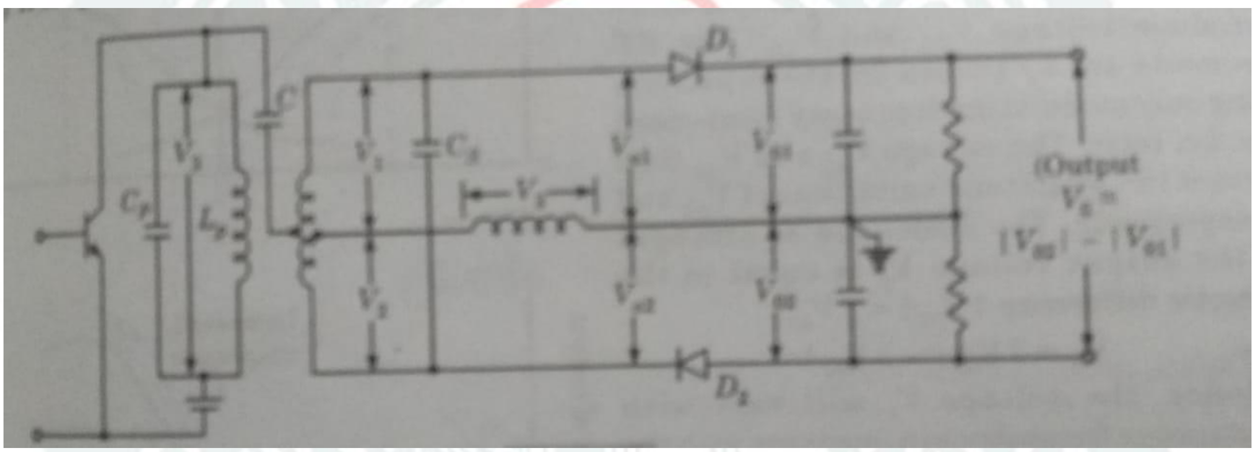
->This circuit consists an inductively coupled double-tuned circuit in which both Primary and secondary coils are tuned to the same frequency (intermediate frequency).

-> The Center of the Secondary coil is connected to the top of the primary through a capacitor C.

→ The capacitor & perform the Following function

It blocks the dc from primary to secondary

It couples the signal frequency from primary to Centre tapping of the secondary.



→ The Primary voltage V3 thus appears across the Indicator L. Nearly entire voltage V_3 appears across inductor L except a small drop across the Capacitor C. However by a suitable Choice of C and L, the drop across the capacitor c can be kept negligible.

→The centre-tapping of the secondary coil has an opposite voltage across each half winding. Hence V_1 and V_2 are equal in magnitude but opposite in phase .The radio frequency voltage Va1 and Va2 applied to the diode D1 and D2 are expressed as

Va1=v3+v1

Va2=V3-V2

→Voltage Va1 and Va2 depend upon the phasor relation between v1,v2 and v3.

→The phasor diagrams V1 and V2 are always equal and are in phase opposition.

→ However, the phase position of V1 and V2 relative to V3 would depend upon the tuned secondary coil at the resonance or off the resonance.

Chaper-4 AM&FM TRANSMITTER &RECEIVER

Classification Of Radio Receiver-

We two can classify the radio receiver in ways as under :

(1) Depending upon the applications, the radio receivers may be classified

Amplitude or modulation (AM) Broadcast Receiver These receivers are used to receive the broadcast of speech or music transmitted from amplitude modulation broadcast transmitter which operate on long wave, medium wave or short wave.

Frequency modulation (FM) Broadcast receiver-These receivers are used to receive the broadcast programmes from FM broadcast which operate transmitters. in VHF or UHF bands.

Communication Receiver-Communication receivers are used for reception telegraph and of short wave telephone Signals. This means that communication receivers are used for various purposes other than broadcast services.

Television Receiver-Television Receiver are used to receive television broadcast in VHF or in UHF bands.

Radar Receivers:- Radar Receivers are used to receive radar signad.

Depending upon the fundamental aspects, the radio receivers may also be classified as under

Tuned Radio Frequency Receiver (TRF)

Superhetoodyne Receiver.

Selectivity: The selectivity of a receiver may be defined as the ability to reject unwanted signals. It also expresses the attenuation that the receiver offers to signal at frequencies adjacent to the one to which it is tuned.

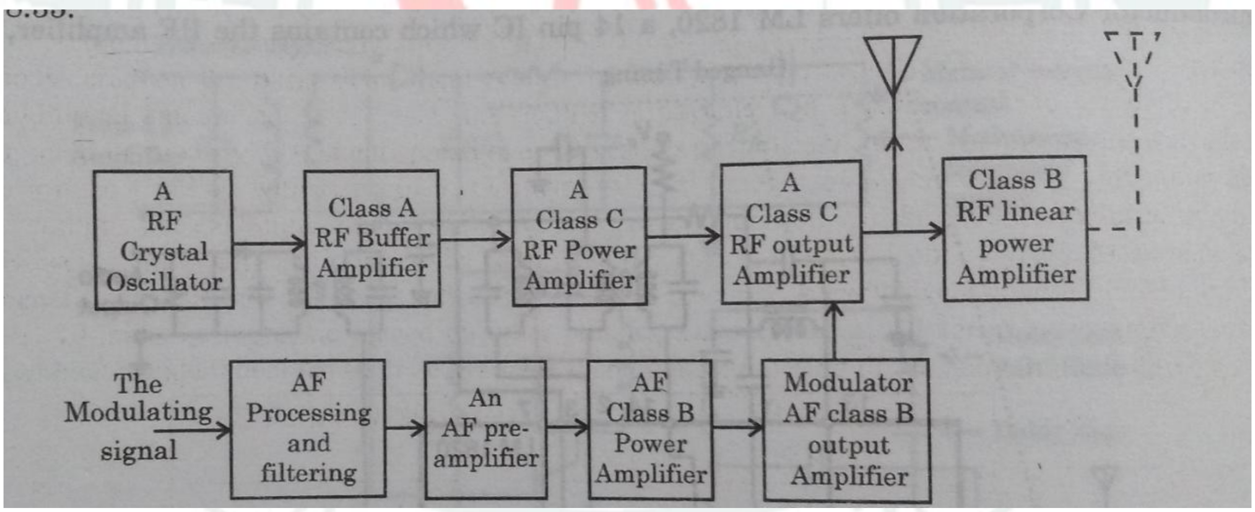
Sensitivity- The sensitivity of a Radio receiver may be to defined as its ability to amplify weak signals. It is generally defined in terms of the voltage which must be applied at receiver input terminal to provide a standard output power measured at the output terminal.

Fidelity - Fidelity is defined as the correspondence of the output signal to the input signal.

Noise Figure- Noise figure is a metric used to indicate the quality of amplifier. It is the decibel representation of the noise factor that determines the sensitivity of the amplifier is decreased to an incoming signal due to its internal noise.

It is defined as the ratio of total Output noise power to the output noise due to the input source.

AM TRANSMITTER-



Crystal oscillator-

RF crystal oscillator invariably makes use of a crystal controlled oscillator to ensure high accuracy of the carrier frequency as recommended by the FCC.

→ Then the output of crystal oscillator is connected to RF Buffer amplifier.

RF Buffer amplifier

→ The output of crystal oscillator is the input of RF buffer amplifier .

→Buffer amplifier provides a high impedance load for the oscillator to minimize drift.

→The output of RF Buffer amplifier is connected to RF Power.

RF Amplifier:

→ The of RF Buffer amplifier is the input of RF Power amplifier.

→ In RF Power amplifier Carrier signal is subsequently amplified.

→ The output of RF Power amplifier RF output amplifier.

RF output Ampilifier-

→ In RF output amplifier the modulating signal is amplified.

→ The modulating signal is processed before it is applied at the final stage.

AF Processing and Filtering :

->The modulating signal is filtered so as to occupy the Correct bandwidth of 10 KH_2 after modulation.

AF Pre-amplifier & Power Amplifier-

→The output of AF Processing and Filtering is the input of AF Pre-amplifier

→The modulating signal is then amplified by audio amplifier and Power amplifier.

→Where Audio frequency also amplified.

→Then it is connected to modulator Output amplifier Modulator output amplifier.

Modulator output amplifier-

→The modulating signal culminates in the modulator amplifier, which is the highest power audio amplifier.

→The difference between high level and low-level modulation depends upon the Point at which modulation takes place.

→For low-level modulator, the modulation is done at some stage before the final stage.

RF Amplifier

→RF amplifier is a small signal tuned amplifier with tuned circuits both in input side and the output side.

→Both these input and output circuits are tuned to desired incoming carrier frequency.

Frequency Conversion

→Frequency converter is also known as power changer, is a device that takes incoming low power and converts it to high Power output.

If Amplifier –

Intermediate frequency amplifier is used to gives a accurate frequency as band width requiration.

Tuning

Tuning is defined as to adjust to respond to waves of an particular frequency or to adjust the frequency of the output of a device to chosen frequency or range of frequencies.

Signal to Noise Ratio –

The ratio of signal power to the associated noise power signal to noise ratio.

In other words, signals to noise ratio is defined as the ratio of signal power to the noise Power at the same Point in the system.

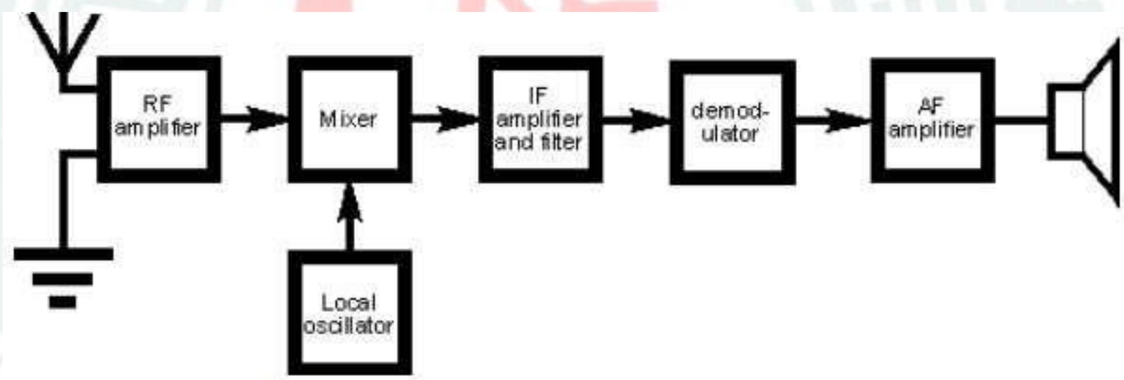
Signal to noise ratio is donated by S/N.

SUPERHETERODYNE RECEIVER:

A super heterodyne receiver(often shortened to superheat) uses frequency mixing to convert a received signal to a fixed intermediate frequency (IF) which can be more conveniently processed than the original radio carrier frequency.

Basic Super heterodyne Block Diagram and Functionality:

The basic block diagram of a basic super heterodyne receiver is shown below. This details the most basic form of the receiver and serves to illustrate the basic blocks and their function.



Block Diagram of a Basic Superheterodyne Radio Receiver

The way in which the receiver works can be seen by following the signal as is passes through the receiver.

Front end amplifier and tuning block: Signals enter the front end circuitry from the antenna. This circuit block performs two main functions:

Tuning: Broadband tuning is applied to the RF stage. The purpose of this is to reject the signals on the image frequency and accept those on the wanted frequency. It must also be able to track the local oscillator so that as the receiver is tuned, so the RF tuning remains on the required frequency. Typically the selectivity provided at this stage is not high. Its main purpose is to reject signals on the image frequency which is at a frequency equal to twice that of the IF away from the wanted frequency. As the tuning within this block provides all the rejection for the image response, it must be at a sufficiently sharp to reduce the image to an acceptable level. However the RF tuning may also help in preventing strong off-channel signals from entering the receiver and overloading elements of the receiver, in particular the mixer or possibly even the RF amplifier.

Amplification: In terms of amplification, the level is carefully chosen so that it does not overload the mixer when strong signals are present, but enables the signals to be amplified sufficiently to ensure a good signal to noise ratio is achieved. The amplifier must also be a low noise design. Any noise introduced in this block will be amplified later in the receiver.

Mixer / frequency translator block: The tuned and amplified signal then enters one port of the mixer. The local oscillator signal enters the other port. The performance of the mixer is crucial to many elements of the overall receiver performance. It should eb as linear as possible. If not, then spurious signals will be generated and these may appear as 'phantom' received signals.

Local oscillator: The local oscillator may consist of a variable frequency oscillator that can be tuned by altering the setting on a variable capacitor. Alternatively it may be a frequency synthesizer that will enable greater levels of stability and setting accuracy.

Intermediate frequency amplifier, IF block : Once the signals leave the mixer they enter the IF stages. These stages contain most of the amplification in the receiver as well as the filtering that enables signals on one frequency to be separated from those on the next. Filters may consist simply of LC tuned transformers providing inter-stage coupling, or they may be much higher performance ceramic or even crystal filters, dependent upon what is required.

Detector / demodulator stage: Once the signals have passed through the IF stages of the super heterodyne receiver, they need to be demodulated. Different demodulators are required for different types of transmission, and as a result some receivers may have a variety of demodulators that can be switched in to accommodate the different types of transmission that are to be encountered. Different demodulators used may include:

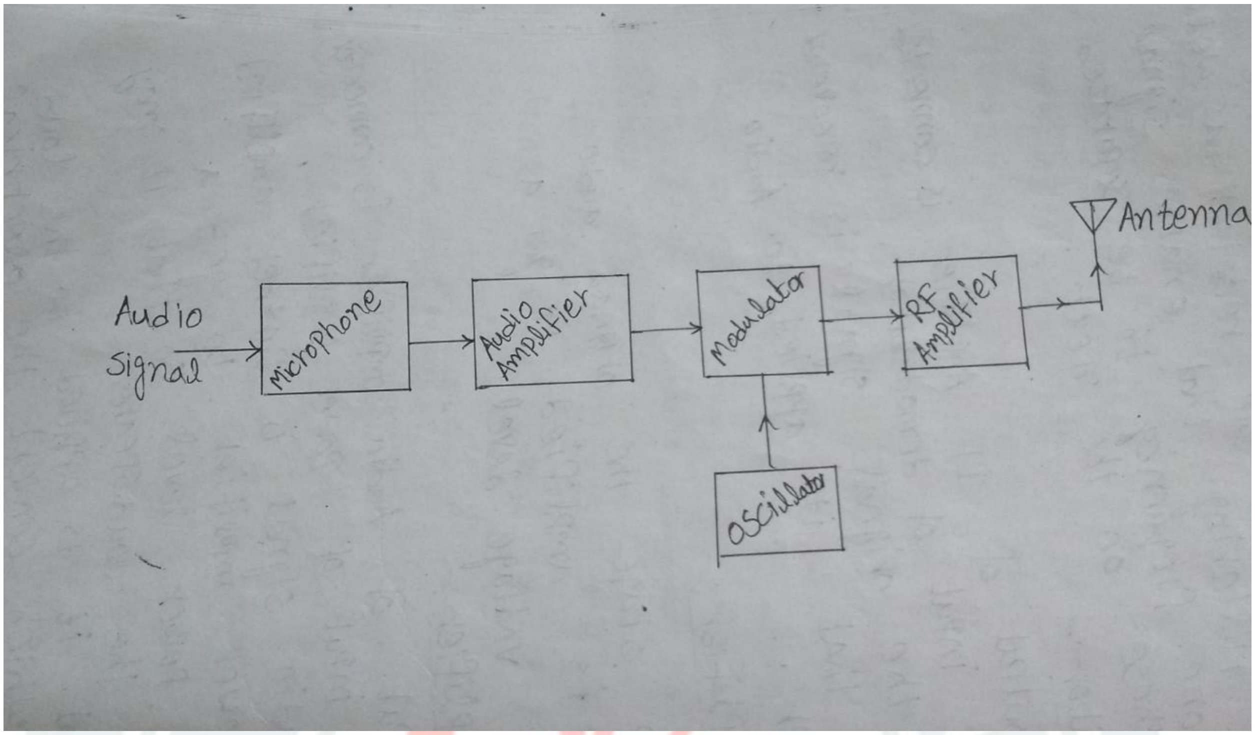
AM diode detector: This is the most basic form of detector and this circuit block would simple consist of a diode and possibly a small capacitor to remove any remaining RF.The detector is cheap and its performance is adequate, requiring a sufficient voltage to overcome the diode forward drop. It is also not particularly linear, and finally it is subject to the effects of selective fading that can be apparent, especially on the HF bands.

Synchronous AM detector: This form of AM detector block is used in where improved performance is needed. It mixes the incoming AM signal with another on the same frequency as the carrier. This second signal can be developed by passing the whole signal through a squaring amplifier. The advantages of the synchronous AM detector are that it provides a far more linear demodulation performance and it is far less subject to the problems of selective fading.

SSB product detector: The SSB product detector block consists of a mixer and a local oscillator, often termed a beat frequency oscillator, BFO or carrier insertion oscillator, CIO. This form of detector is used for Morse code transmissions where the BFO is used to create an audible tone in line with the on-off keying of the transmitted carrier. Without this the carrier without modulation is difficult to detect. For SSB, the CIO re-inserts the carrier to make the modulation comprehensible.

Audio amplifier: The output from the demodulator is the recovered audio. This is passed into the audio stages where they are amplified and presented to the headphones or loudspeaker.

FM TRANSMITTER-



The main function of FM transmitter is to transmit the sound using radio wave. Transmitter Converts the sound into radio wave then it transmit.
Microphone -

→The audio signal is applied to the Micro Phone.

→Microphone convert the sound energy into electrical . So microphone is energy a source of an audio signal.

→ The output of Microphone is applied to the Audio amplifier.

Audio Amplifier –

→ The output of microphone is the input of the audio amplifier.

→It amplify the audio signal. Convert it from weak signal to strong.

Modulator

→The output of audio amplifier is the input of modulator.

→ It converts the audio signal into a radio signal which is transmitted.

→The modulator takes two signal as input one is the audio signal coming from the audio amplifier and another is carrier signal.

Oscillator

→Oscillator is produce carrier signal.

→ The wave carrier signal is a pure radio wave.

→ The output of oscillator is applied to modulator

RF Amplifier

→ The output of modulator is applied to RF amplifier.

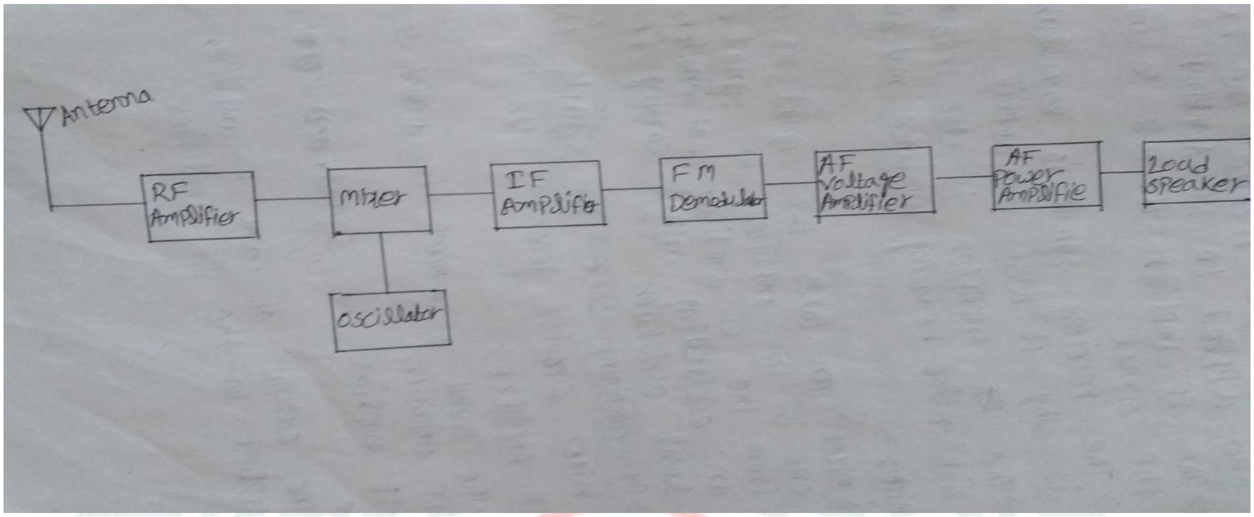
→The RF amplifier amplifies the modulated signal coming from the modulator.

→The modulated signal have very low amplitude be transmitted for a long distance So it cannot be transmitted for a long distance.

-> RF amplifier is 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 of signal.

FM RECEIVER-



RF Amplifier:

→ The antenna receives the radio signal and applied to RF Amplifier.

→The RF amplifier is used to amplify the signal.

→ The received signal is very weak that is why it needs to amplify.

→The RF amplifier is also responsible for noise-reducing, impedance matching.

Mixer

→ Mixer takes two signal as input, one is from the the amplified RF signal coming RF amplifier and another is an oscillating signal coming from the local oscillator.

→By combining those two signals the mixer circuit generates the IF signal.

IF Amplifier -

The output of a mixer is applied to the IF amplifier.

->It amplies the signal.

→The output of IF amplifier is applied to the IF amplifier.

Demodulator-

→The FM demodulator circuit recovers the actual modulated signal which comes from the transmitter from the radio signal.

→The output of demodulator is applied to voltage amplifier.

Voltage Amplifier and Power amplifier: -

→ After demodulation voltage amplifier and power amplifier amplifies the signal and we get the original signal.

→The output signal is applied to the loud speaker.

→ Loud to electrical speaker convert the audio signal audio signal.

CHAPTER-5 ANALOG TO DIGITAL CONVERSION

Sampling Theorem

Sampling of the signals is the fundamental operation in signal Processing. A Continuous time signal is first converted to discrete- signal by sampling process. The sufficient number of samples of the signal must be taken so that the original signal is represented in it's a sample 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. Sampling theorem gives the complete Idea about the sampling signals.

Nyquist Rate

When the sampling rate becomes exactly equal to 2fm sample Per Second, then it is called Nyquist rate. Nyquist rate is also called minimum sampling rate. It is given by

fs=2fm

similarly, maximum sampling inter vad called Nyquist interval.

It is given by Ts = 1/2fm seconds

Aliasing

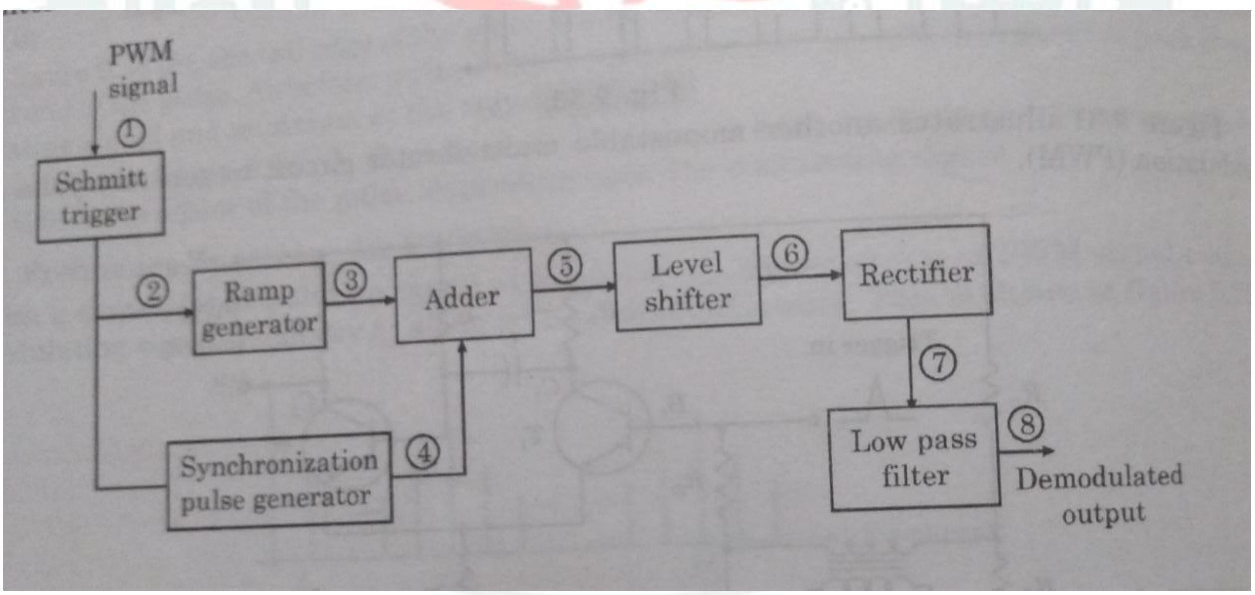
When a Continuous-time band limited signal is sampled at a rate lower than Nyquist rate fs<2fm then the successive cycle of the the spectrum of sampled signal overlap with each other.

→ Hence, the signal is under- sampled in this case (fs<2fm) and some amount of aliasing is produced in this under sampled process.

→ Aliasing is the phenomenon in which a high frequency component in the frequency Spectrum of the signal takes identity of a low-frequency component in the Spectrum of the sampled signal.

Demodulation of PWM Signal

The received PWM signal is applied to the Schmitt trigger circuit. This Schmitt trigger circuit remov the noise in the PWM waveform. The regenerated PWM is then applied to the ramp generator and the synchronization pulse generator. The ramp generator produces ramps for the duration pulses such that height of ramps are proportional to the widths of PWM pulses. The maxi ramp voltage is retained till the next pulse. On the other hand, synchronous pulse generator produs reference pulses with constant amplitude and pulse width. These pulses are delayed by specie 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.



Demodulation of PPM

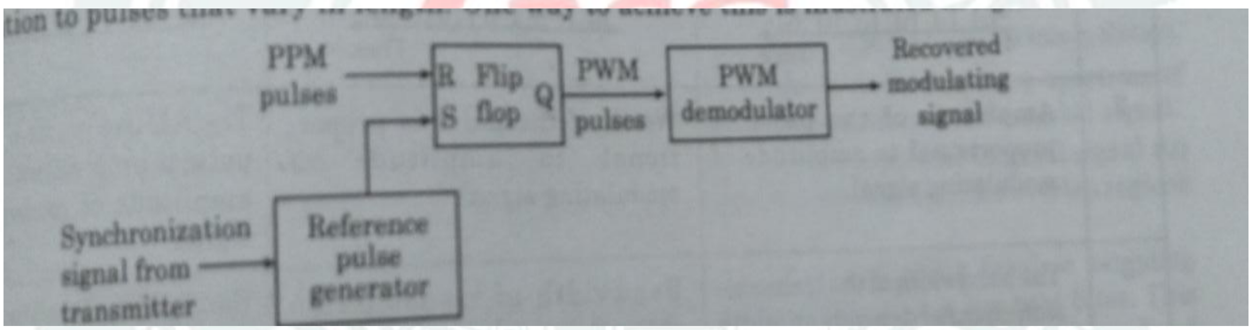
In case of pulse-position modulation, it is customary to convert the received pulses that vary in position to pulses that vary in length. One way to achieve this is illustrated.

Flip-flop circuit is set or turned 'ON' (giving high output) 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" (giving low output) at the leading edge of the position modulated pulse.

This repeats and we get PWM pulses at the output of the flip-flop.

The PWM pulses are then demodulated by PWM demodulator to get orginal modulating signal.



Comaparison between PAM,PWM,PPM

PAM	PWM	PPM
Amplitude is varied	Width is varied	Position is varied
Bandwidth depends on th width of the pulse	Bandwidth depends on the time of the pulse	Bandwidth depends on the time of the pulse
Instantaneous transmitter power varies with the amplitude of the pulses	Instantaneous transmitter powervaries with the amplit and width of the pulses	Instantaneous transmitter power remains constant wit the width of the pulses

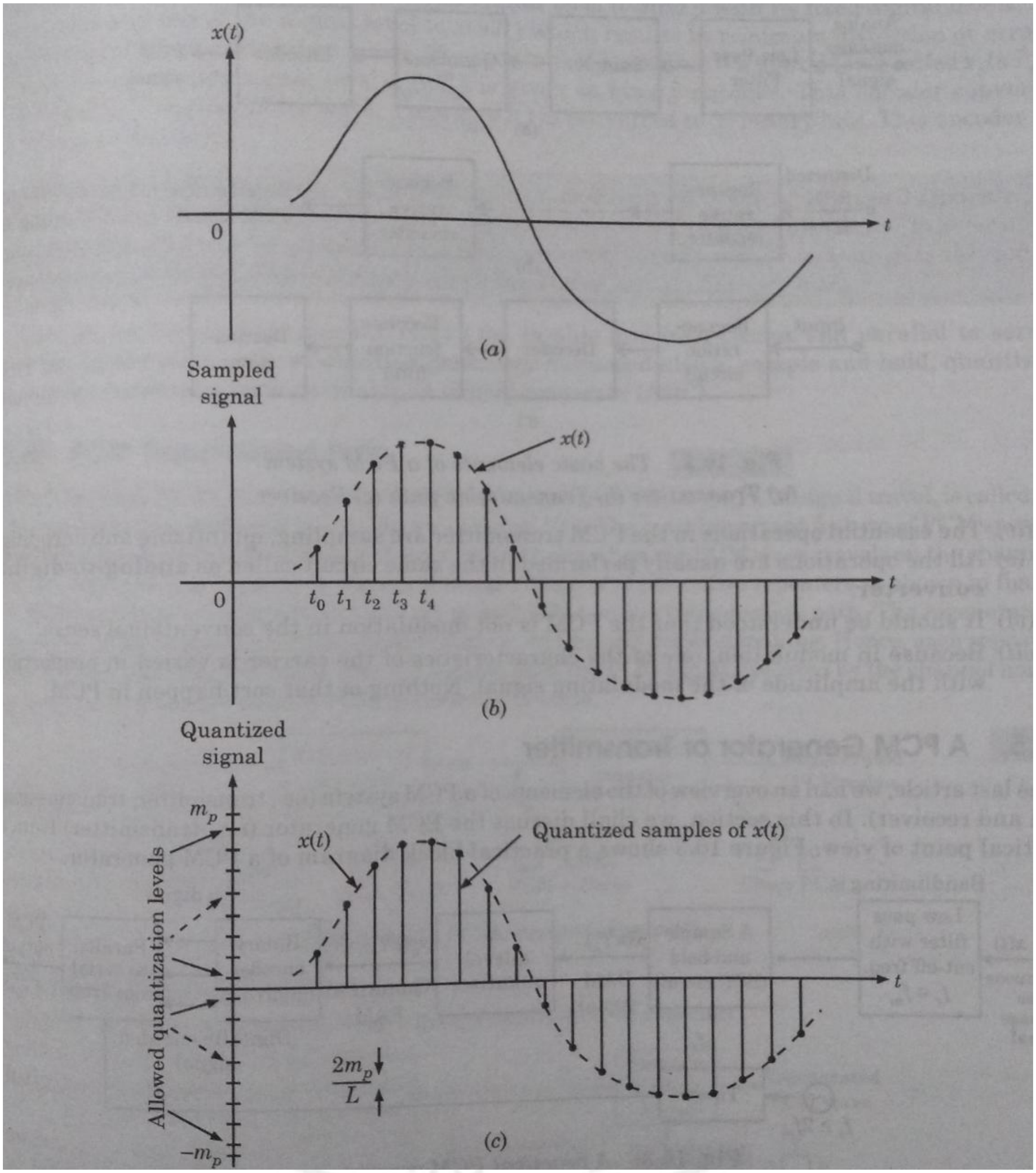
System complexity is hig	System complexity is low	System complexity is low
Noise interference is high	Noise interference is low	Noise interference is low
It is similar to amplitude modulation	It is similar to frequency modulation	It is similar to phase modulation

Concept of Quantization

In communication systems, sometimes it happens that we are available with analog signal, however, we have to transmit a digital signal for a particular application. In such cases, we have to convert a equal time analog signal into digital signal. This 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 to digital form, let us consider analog signal First of all, we get samples of this signal according to sampling theorem. For this purpose, we mark the time-instants t1,t2,t3, and so on,at equal time intevals 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. A representation of the signal of in terms of its samples.

Now, we can say that the signal is defined only at the sampling instants. This means that it no is a continuous function of time, but rather, it is a discrete-time signal However, since the magnitude of each smaple can take any value in a continuous range, the signal is still an analog signal.

This difficulty is neatly resolved by a process known as quantization.

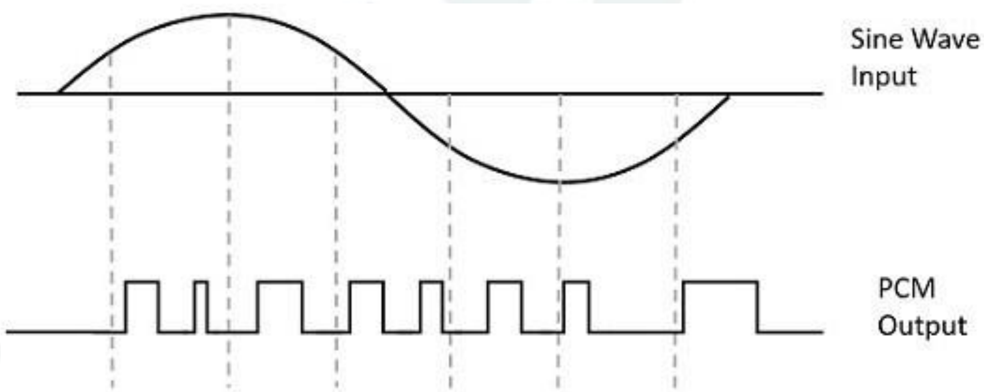


Quantization error-

A q-level quantizer compares the discrete-time input x(nT) with its fixed digital levels. It assigns any one of the digital level to x(nT) with its fixed digital levels. It then assigns any one of the digital level to x(nT) which results in minimum distortion or error. This error is called quantization error.

Pulse Code Modulation

A signal is Pulse Code modulated to convert its analog information into a binary sequence, i.e., 1s and 0s. The output of a **Pulse Code Modulation (PCM)** will resemble a binary sequence. The following figure shows an example of PCM output with respect to instantaneous values of a given sine wave.



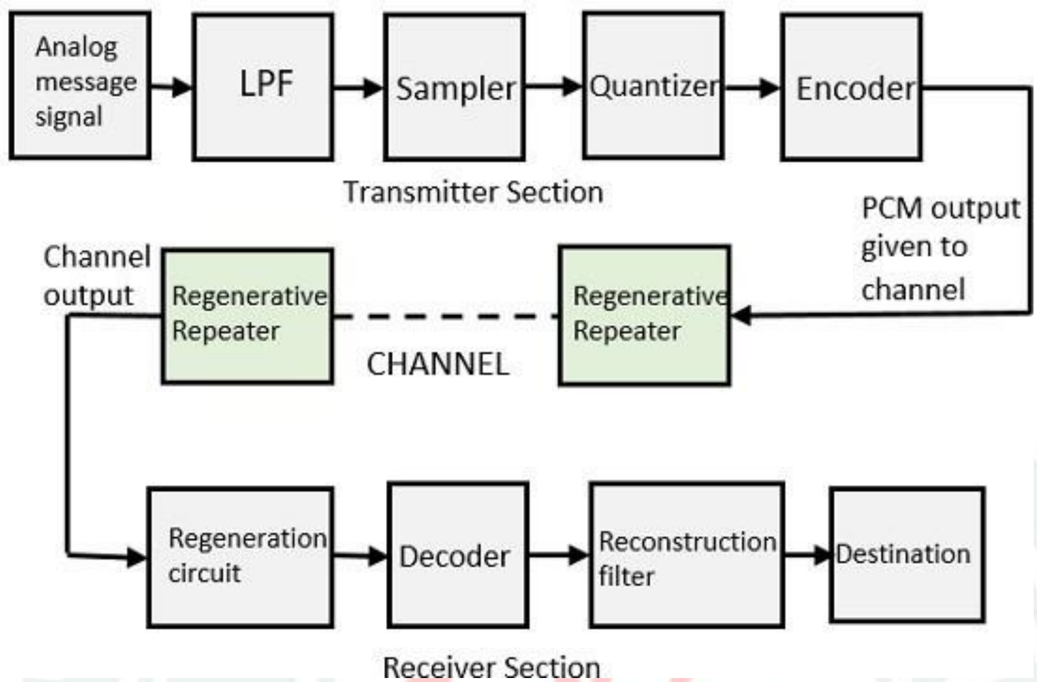
Instead of a pulse train, PCM produces a series of numbers or digits, and hence this process is called as digital. Each one of these digits, though in binary code, represent the approximate amplitude of the signal sample at that instant.

In Pulse Code Modulation, the message signal is represented by a sequence of coded pulses. This message signal is achieved by representing the signal in discrete form in both time and amplitude.

Basic Elements of PCM

The transmitter section of a Pulse Code Modulator circuit consists of **Sampling, Quantizing** and **Encoding**, which are performed in the **analog-to-digital converter** section. The low pass filter prior to sampling prevents aliasing of the message signal.

The basic operations in the receiver section are **regeneration of impaired signals, decoding,** and **reconstruction** of the quantized pulse train. The following figure is the block diagram of PCM which represents the basic elements of both the transmitter and the receiver sections.



Low Pass Filter (LPF)

This filter eliminates the high frequency components present in the input analog signal which is greater than the highest frequency of the message signal, to avoid aliasing of the message signal.

Sampler

This is the circuit which uses the technique that helps to collect the sample data at instantaneous values of the message signal, so as to reconstruct the original signal. The sampling rate must be greater than twice the highest frequency component **W** of the message signal, in accordance with the sampling theorem.

Quantizer

Quantizing is a process of reducing the excessive bits and confining the data. The sampled output when given to Quantizer, reduces the redundant bits and compresses the value.

Encoder

The digitization of analog signal is done by the encoder. It designates each quantized levelby a binary code. The sampling done here is the sample-and-hold process. These three sections will act as an analog to the digital converter. Encoding minimizes the bandwidth used.

Regenerative Repeater

The output of the channel has one regenerative repeater circuit to compensate the signal loss and reconstruct the signal. It also increases the strength of the signal.

Decoder

The decoder circuit decodes the pulse coded waveform to reproduce the original signal. This circuit acts as the **demodulator**.

Reconstruction Filter

After the digital-to-analog conversion is done by the regenerative circuit and the decoder, a low pass filter is employed, called as the reconstruction filter to get back the original signal.

Hence, the Pulse Code Modulator circuit digitizes the analog signal given, codes it, and samples it. It then transmits in an analog form. This whole process is repeated in a reverse pattern to obtain the original signal.

Companding

As a matter of fact, companding is nonuniform quantization. It is required to be implemented to improve the signal to quantization noise ratio of weak signals. We know that the quantization noise is given by

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

This shows that in the uniform quantization, once the step size is fixed, the quantization ise power remains constant. However, the signal power is not constant. It is proportional to the square of signal amplitude. Hence signal power will be small for weak signals, but quantization anise power is constant. Therefore, the signal to quantization noise for the weak signals is very poor. This will affect the quality of signal. The remedy is to use companding. Companding is a term derived from two words i.e., compression and expansion as under:

Companding =Compressing+Expanding

Time Division Multiplexing (TDM)

In TDM, the time frame is divided into slots. This technique is used to transmit a signal overa single communication channel, with allotting one slot for each message. Of all the types of TDM, the main ones are Synchronous and Asynchronous TDM.

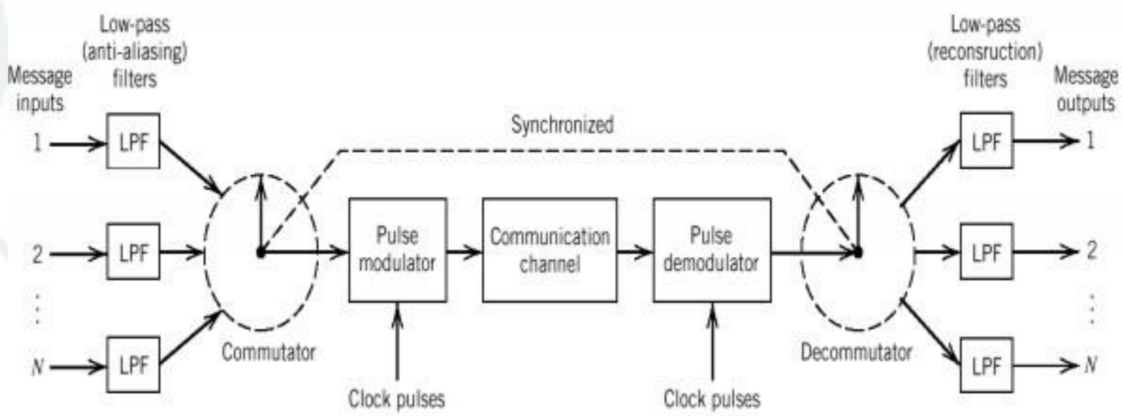
Synchronous TDM

In Synchronous TDM, the input is connected to a frame. If there are 'n' number of connections, then the frame is divided into 'n' time slots. One slot is allocated for each input line. In this technique, the sampling rate is common to all signals and hence same clock inputis given. The mux allocates the same slot to each device at all times.

Asynchronous TDM

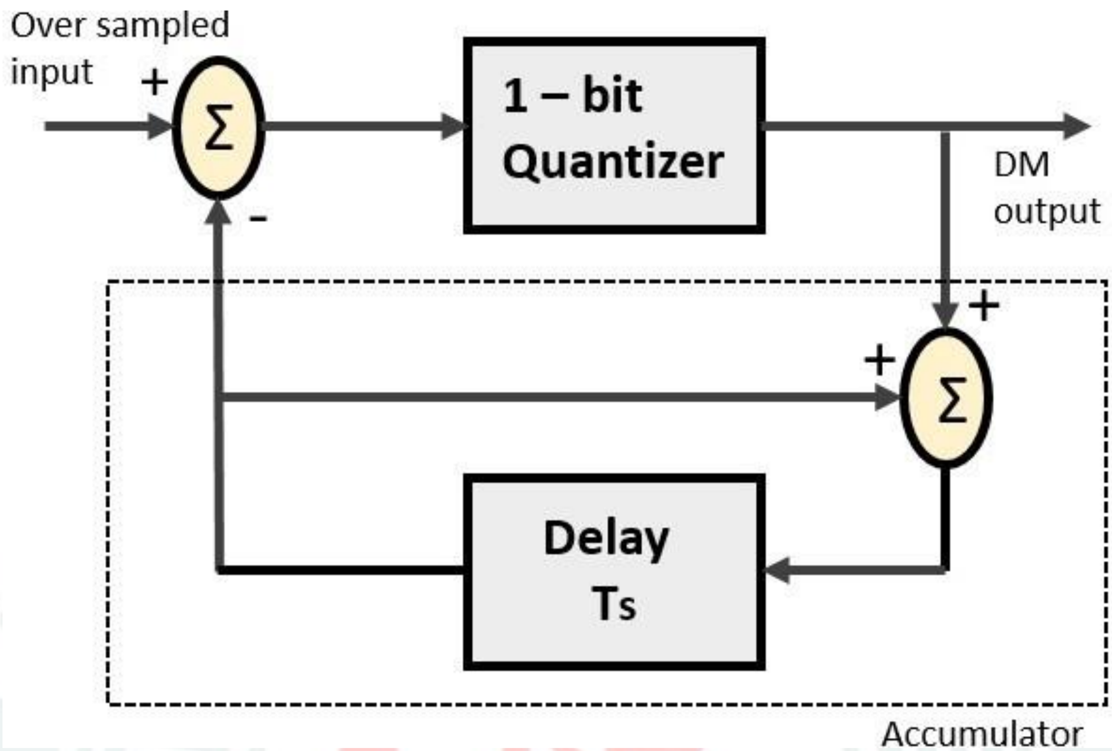
In Asynchronous TDM, the sampling rate is different for each of the signals and the clocksignal is also not in common. If the allotted device, for a time-slot, transmits nothing and sits idle, then that slot is allotted to another device, unlike synchronous.

Time Division Multiplexing (TDM)



Delta Modulator

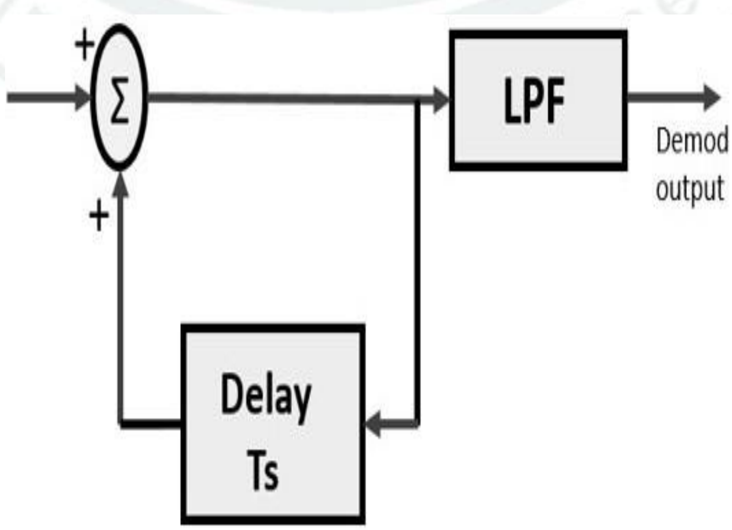
The **Delta Modulator** comprises of a 1-bit quantizer and a delay circuit along with twosummer circuits. Following is the block diagram of a delta modulator.



A stair-case approximated waveform will be the output of the delta modulator with the step-size as delta (**Δ**). The output quality of the waveform is moderate.

<u>Delta Demodulator</u>

The delta demodulator comprises of a low pass filter, a summer, and a delay circuit. Thepredictor circuit is eliminated here and hence no assumed input is given to the demodulator.



Low pass filter is used for many reasons, but the prominent one is noise elimination for out- of-band signals. The step-size error that may occur at the transmitter is called **granular noise**, which is eliminated here. If there is no noise present, then the modulator output equals the demodulator input.

Differential Pulse Code Modulation (DPCM)

For the signals which does not change rapidly from one sample to next sample, the PCM scheme is not preferred. When such highly correlated samples are encoded the resulting encoded signal contains redundant information. By removing this redundancy before encoding an efficient coded signal can be obtained. One of such scheme is the DPCM technique. By knowing the past behavior of a signal up to a certain point in time, it is possible to make some inference about the future values. The transmitter and receiver of the DPCM scheme is shown in the below

e(nTs) = x(nTs) - x^(nTs) ----- (3.31)

fig.respectively. Transmitter: Let x(t) be the signal to be sampled and x(nTs) be its samples. In this scheme the input to the quantizer is a signal.

where x^(nTs) is the prediction for unquantized sample x(nTs). This predicted value is produced by using a predictor whose input, consists of a quantized versions of the input signal x(nTs). The signal e(nTs) is called the prediction error.

By encoding the quantizer output, in this method, we obtain a modified version of the PCM called differential pulse code modulation (DPCM).

Quantizer output, v(nTs) = Q[e(nTs)]

= e(nTs) + q(nTs)---------------- (3.32)

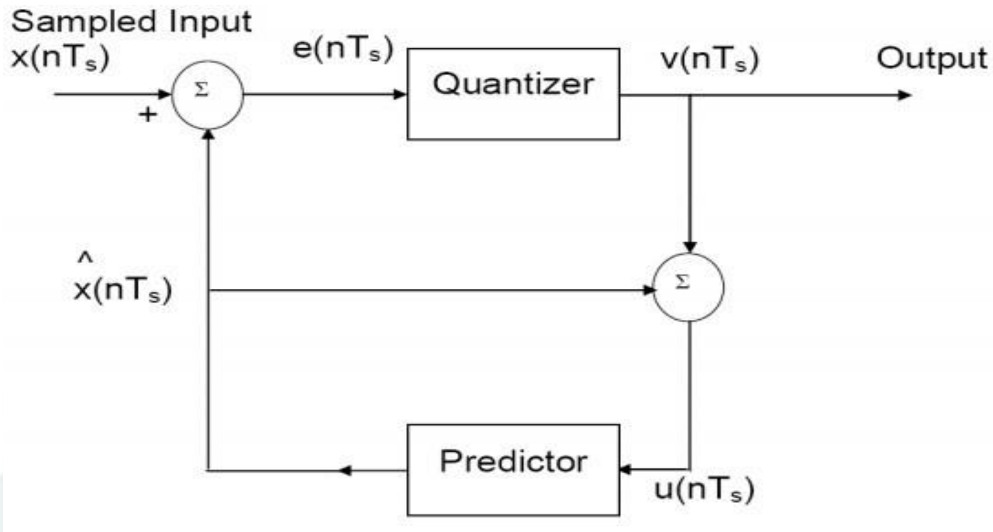
Predictor input is the sum of quantizer output and predictor output, u(nTs) = x^(nTs) + v(nTs) ----- (3.33)

Using 3.32 in 3.33,

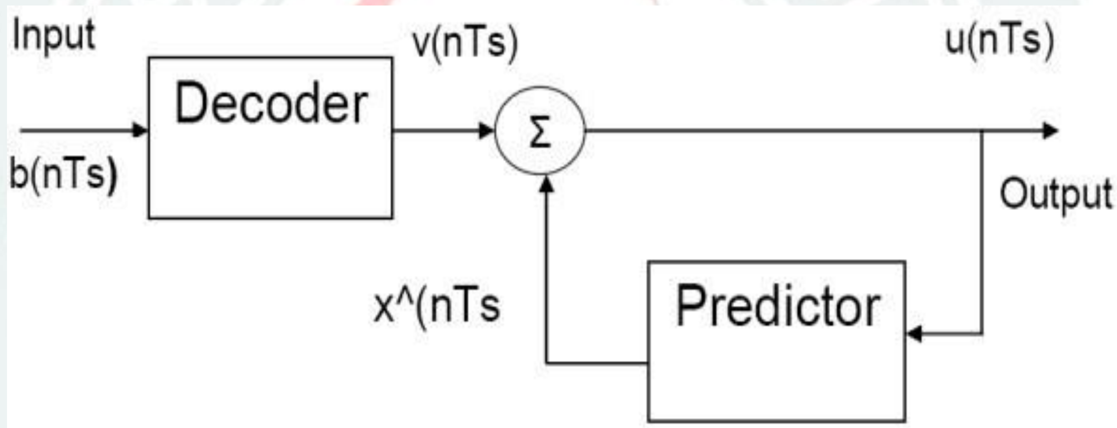
u(nTs) = x^(nTs) + e(nTs) + q(nTs) ---------- (3.34)

u(nTs) = x(nTs) + q(nTs) ------------ (3.35)

The receiver consists of a decoder to reconstruct the quantized error signal. The quantized version of the original input is reconstructed from the decoder output using the same predictor as used in the transmitter. In the absence of noise the encoded signal at the receiver input is identical to the encoded signal at the transmitter output. Correspondingly the receive output is equal to u(nTs), which differs from the input x(nts) only by the quantizing error q(nTs).



Block diagram of DPCM Transmitter



Block diagram of DPCM Receiver.

The sampling rate of a signal should be higher than the Nyquist rate, to achieve bettersampling. If this sampling interval in a Differential PCM (DPCM) is reduced considerably,the sample-to-sample amplitude difference is very small, as if the difference is **1-bit quantization**, then the step-size is very small i.e., **Δ** (delta).

Comparison PCM,DM,ADM&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.	Only one bit is used to encode one sample.	Bits can be more than one but are less than PCM.
2.	Levels and step size	The number of levels depend on number of bits. Level size is kept fixed.	Step size is kept fixed and cannot be varied.	According to the signal variation, step size varies (*i.e.* Adapted).	Here, Fixed number of levels are used.
3.	Quantization error and distortion	Quantization error depends on number of levels used.	Slope overload distor-tion and granular noise are present.	Quantization noise is present but other errors are absent.	Slope overload distor-tion and quantization noise is present.
4.	Transmission bandwidth	Highest bandwidth is required since number of bits are high	Lowest bandwidth is required.	Lowest bandwidth is required.	Bandwidth required is lower than PCM.
5.	Feedback	There is no feedback in transmitter or receiver.	Feedback exists in transmitter.	Feedback exists.	Here, Feedback exists.
6.	Complexity of implementation	System complex.	Simple.	Simple.	Simple

CHAPTER-6

DIGITALMODULATION TECHNIQUES

MULTIPLEXING

Multiplexing is the process of combining multiple signals into one signal, over a shared medium. If analog signals are multiplexed, it is Analog Multiplexing and if digital signals are multiplexed, that process is Digital Multiplexing.

Multiplexing and Demultiplexing

The process of multiplexing divides a communication channel into several number of logical channels, allotting each one for a different message signal or a data stream to be transferred. The device that does multiplexing can be simply called as a MUX while the one that reverses the process which is demultiplexing, is called as DEMUX.

Types of Multiplexers

Frequency Division Multiplexing (FDM)

In analog multiplexing, the most used technique is Frequency Division Multiplexing FDM. This technique uses various frequencies to combine streams of data, for sending them on a communication medium, as a single signal.

Example: A traditional television transmitter, which sends a number of channels through a single cable, uses FDM.

Time Division Multiplexing (TDM)

In TDM, the time frame is divided into slots. This technique is used to transmit a signal over a single communication channel, with allotting one slot for each message. Of all the types of TDM, the main ones are Synchronous and Asynchronous TDM.

DIGITAL MODULATION :

Digital Modulation provides more information capacity, high data security, quicker system availability with great quality communication. Hence, digital modulation techniques have a greater demand, for their capacity to convey larger amounts of data than analog modulation techniques.

There are many types of digital modulation techniques and also their combinations, depending upon the need. Of them all, we will discuss the prominent ones.

ASK – Amplitude Shift Keying

The amplitude of the resultant output depends upon the input data whether it should be a zero level or a variation of positive and negative, depending upon the carrier frequency.

FSK – Frequency Shift Keying

The frequency of the output signal will be either high or low, depending upon the input data applied.

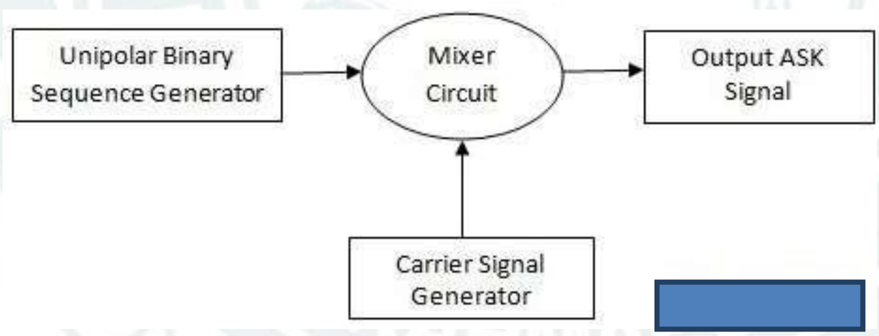
PSK – Phase Shift Keying

The phase of the output signal gets shifted depending upon the input. These are mainly of two types, namely Binary Phase Shift Keying BPSK and Quadrature Phase Shift Keying QPSK, according to the number of phase shifts. The other one is Differential Phase Shift Keying DPSK which changes the phase according to the previous value.

Amplitude Shift Keying (ASK):

In ASK, it requires two input signals, First input is binary sequence signal and the second input is carrier signal. Here the most important point we need to always consider the second input which is the carrier signal has the more amplitude/voltage range than the input binary sequence signal.

One more important point is to consider here, the carrier signal amplitude is should be greater than the input binary signal amplitude. Within carrier amplitude range we are going to modulate the binary input signal amplitude. If the carrier signal amplitude is less than the input binary signal voltage, then such a combination modulation process leads to over modulation and under modulation effects. So to achieve perfect modulation carrier single should have more amplitude range than input binary signal.



In amplitude shift keying theory, input binary signal amplitude varies according to the carrier signal voltage. In ASK, the input binary signal is multiplied with the carrier signal along with its time intervals. Between the first time interval of input binary signal multiplied with the first time interval of carrier signal voltage and the same process continues for all time intervals. If the input binary signal is logic HIGH for certain time interval, then the same should be delivered at the output ports with increment in voltage level. So the main aim of the amplitude shift keying modulation is to changing or improving the voltage characteristics of the input binary signal concerning the carrier signal.

ASK Demodulation Process

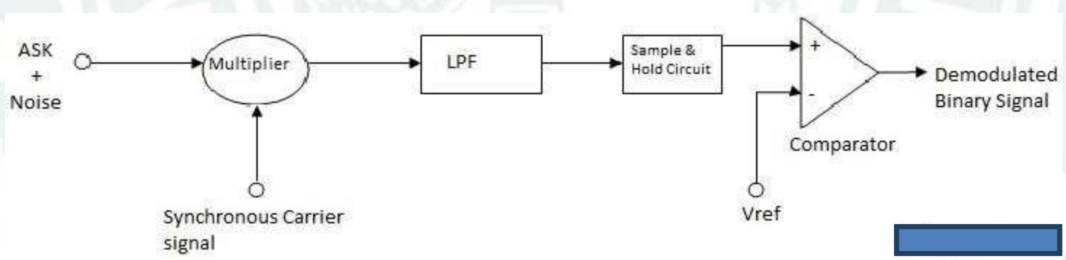
Demodulation is the process of reconstructing the original signal at the receiver level. And it is defined as, whatever the modulated signal received from the channel at the receiver side by implementing the proper demodulated techniques to recover/reproduce the original input signal at the output stage of the receiver.
ASK demodulation can be done in two ways. They are,
Coherent detection (Synchronous demodulation)

Noncoherent Detection (Asynchronous demodulation)

We will start the demodulation process with coherent detection which is also called assynchronous ASK detection.

1). Coherent ASK Detection

In this way of demodulation process, the carrier signal which we are using at the receiver stage is in the same phase with the carrier signal which we are using at the transmitter stage. It means the carrier signal at transmitter and receiver stages are the same values. This type of demodulation is called Synchronous ASK detection or coherent ASK detection.

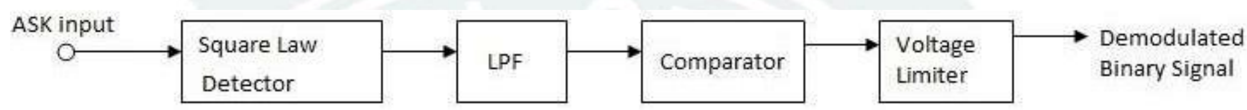


coherent-ask-detection-block-diagram

The receiver receives the ASK modulated waveform from the channel but here this modulated waveform is effected with noise signal because it is forwarded from the free space channel. So this, noise can be eliminated after the multiplier stage by the help of a low pass filter. Then it is forwarded from the sample and hold circuit for converting it into discrete signal form. Then at each interval, the discrete signal voltage is compared with the reference voltage (Vref) to reconstruct the original binary signal.

Non-coherent ASK Detection

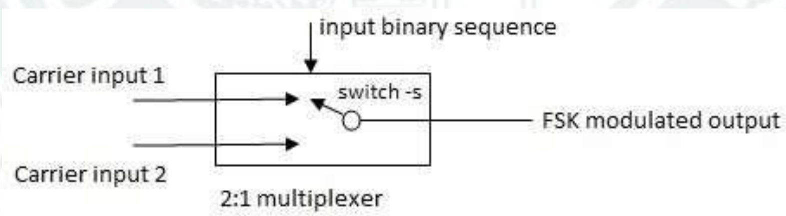
In this, the only difference is the carrier signal which is using at the transmitter side and receiver side are not in the same phase with each other. By this reason, this detection is called as Non-coherent ASK detection (Asynchronous ASK detection). This demodulation process can be completed by using with square law device. The output signal which is generating from the square-law device can be forwarded through a low pass filter to reconstruct the original binary signal.



non-coherent-ask-detection-block-diagram

Frequency Shift Keying Theory

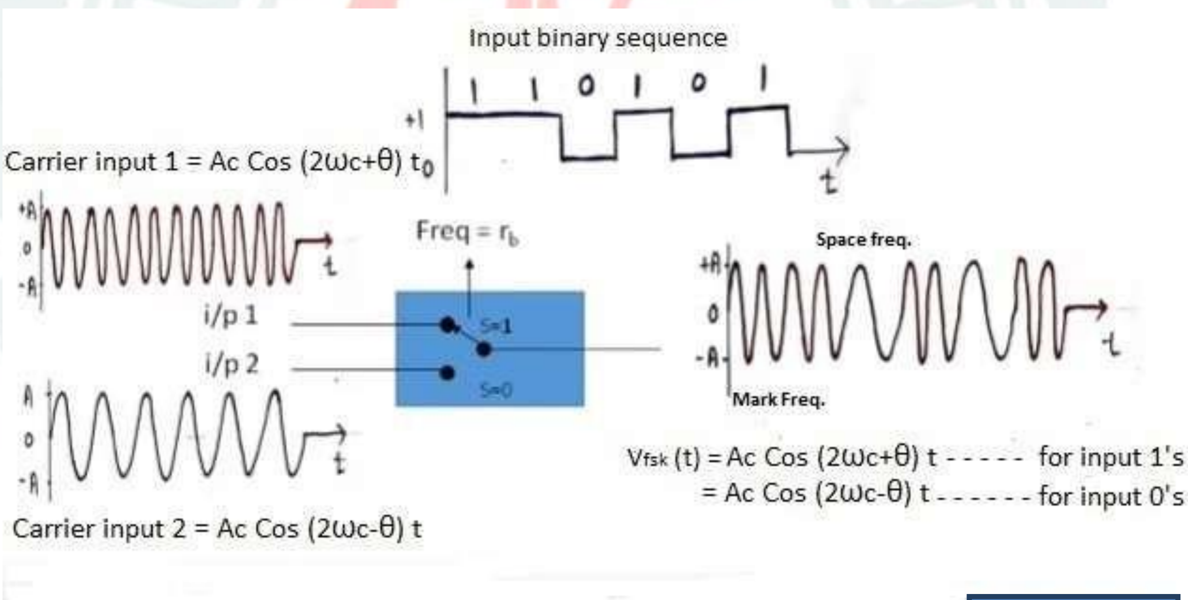
This frequency shift keying theory shows how the frequency characteristics of a binary signal changed according to the carrier signal. In FSK, the binary information can be transmitted through a carrier signal along with frequency changes. The below diagram shows the **frequency shift keying block diagram**.



FSK-block-diagram

In FSK, two carrier signals are used to produce FSK modulated waveforms. The reason behind this, FSK modulated signals are represented in terms of two different frequencies. The frequencies are called “mark frequency” and “space-frequency”. Mark frequency has represented logic 1 and space-frequency has represented the logic 0. There is only one difference between these two carrier signals, i.e. carrier input 1 having more frequency thanthe carrier input 2.

Carrier input 1 = Ac Cos (2ωc+θ) t Carrier
input 2 = Ac Cos (2ωc-θ) t
The switch (s) of the 2:1 multiplexer is having the important role to generate the FSK output. Here the switch is connected to carrier input 1 for all logic 1’s of the binary input sequence. And switch (s) is connected to carrier input 2 for all logic 0’s of the input binary sequence. So, the resultant FSK modulated waveforms have mark frequencies and space frequencies.



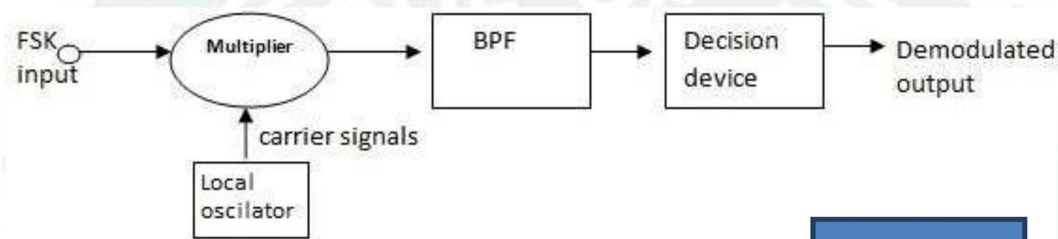
FSK-modulation-output-waveforms
Now we will see how the FSK modulated wave can be demodulated at the receiver side. Demodulation is defined as reconstructing the original signal from the modulated signal. This demodulation can be possible in two ways. They are

- Coherent FSK detection
- Non-coherent FSK detection

The only difference between the coherent and non-coherent way of detection is the phase of the carrier signal. If the carrier signal we are using at the transmitter side and receiver side are in the same phase while demodulation process i.e. called a coherent way of detection and it is also known as synchronous detection. If the carrier signals which we are using at transmitter and receiver side are not in the same phase then such modulation process known as Non- coherent detection. Another name for this detection is Asynchronous detection.

Coherent FSK Detection

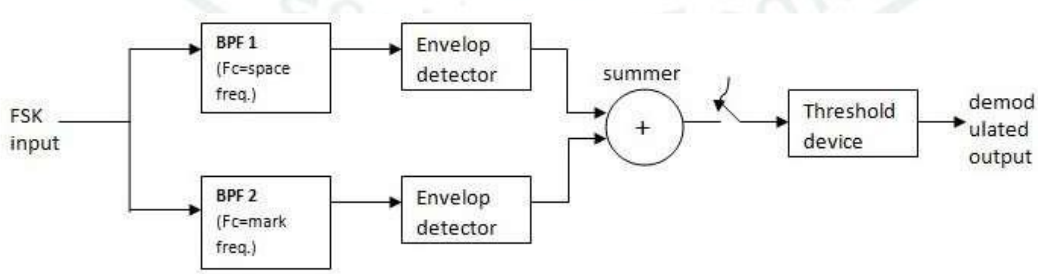
In this synchronous FSK detection, the modulated wave got affected by noise while reaching the receiver. So, this noise can be eliminated from using the bandpass filter (BPF). Here at multiplier stage, the noisy FSK modulated signal is multiplied with the carrier signal from the local oscillator device. Then the resultant signal passes from the BPF. Here this bandpass filter is assigned to cut off frequency which is equal to the binary input signal frequency. So the same frequencies can be allowed to the decision device. Here this decision device gives 0 and 1 for space and mark frequencies of the FSK modulated waveforms.



coherent-FSK-detection

Non-coherent FSK Detection

The modulated FSK signal is forwarded from the bandpass filter 1 and 2 with cut off frequencies equals to space and mark frequencies. So, the unwanted signal components can be eliminated from the BPF. And the modified FSK signals are applied as input to the two envelop detectors. This envelope detector is a circuit having a diode (D). Based upon the input to the envelope detector it delivers the output signal. This envelope detector used in the amplitude demodulation process. Based upon its input it generates the signal and then it is forwarded to the threshold device. This threshold device gives the logic 1 and 0 for the different

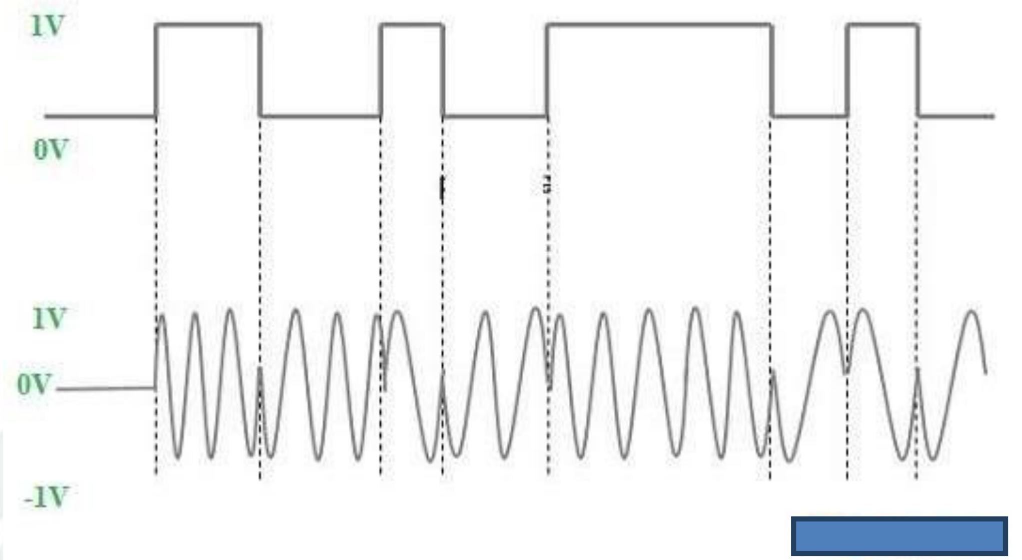


frequencies. This would be equal to the original binary input sequence. So, the FSK generation and detection can be done in this way.. In this FSK experiment, FSK can be generated by the 555 timer IC and detection can be possible by 565IC which is known as a phase-locked loop (PLL).

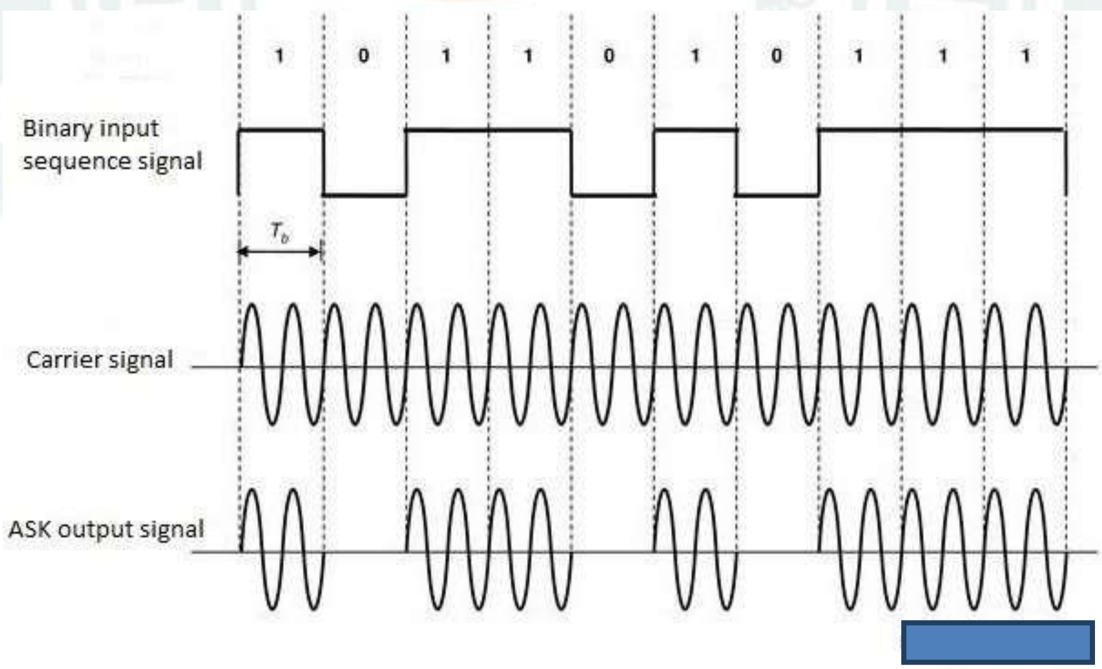
non-coherent-FSK-detection

What is Phase Shift Keying (PSK)?

The Phase Shift Keying is one kind of digital modulation method. This kind of method is used to transmit data by modulating otherwise changing the phase of the carrier signal which is known as a reference signal. The digital data can be represented with any kind of digital modulation method by using a limited number of separate signals. This kind of modulation method uses a limited number of phases where each phase can be assigned with binary digits. Generally, every phase encodes an equivalent number of bits. Every bits pattern forms the symbol that is denoted by the exact phase.
The PSK method can be represented by a convenient method namely constellations diagram. In this kind of communication, the points of the constellation can be selected are generally placed by uniform angular spacing in the region of circle. So that utmost phase separation can be offered among nearby points & therefore the best protection to corruption. These are arranged in a circle so that they can all be transmitted by similar energy.



phase-shift-keying



1). QPSK – Quadrature Phase-Shift Keying

The bit rate can be enhanced by adding more bits on one single segment. In this kind of PSK, the bitstream can be parallelized so that every two incoming bits can be split up & phase shift keying a carrier frequency. One carrier frequency can be phase-shifted with 90 degrees from the other within quadrature. Then the 2 phase-shift keying signals are added to generate one of four signal elements.

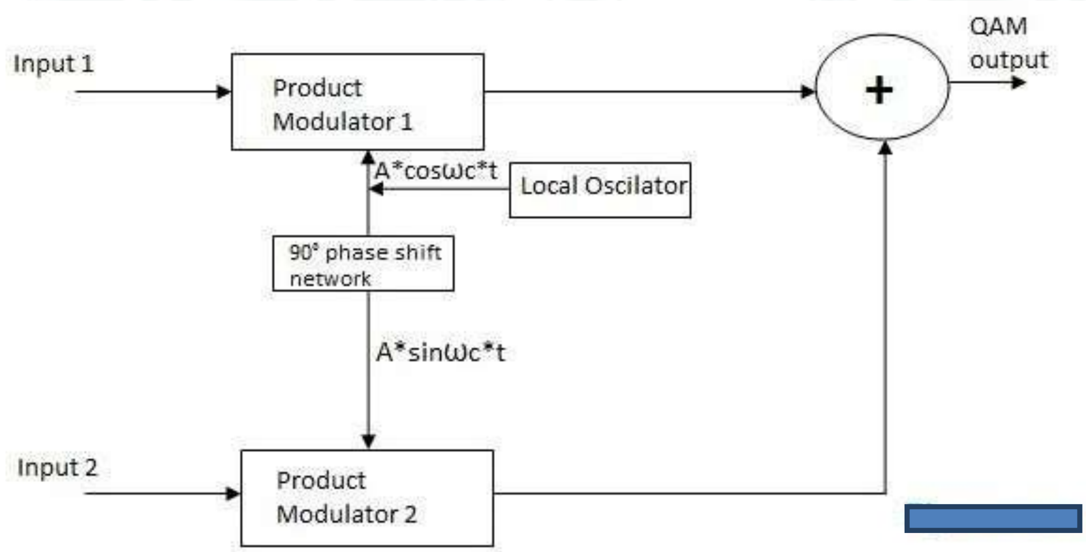
Quadrature Amplitude Modulation Definition

QAM can be defined as it is s a modulation technique that is used to combine two amplitude modulated waves into a single channel to increase the channel bandwidth.

Quadrature Amplitude Modulation Block Diagram

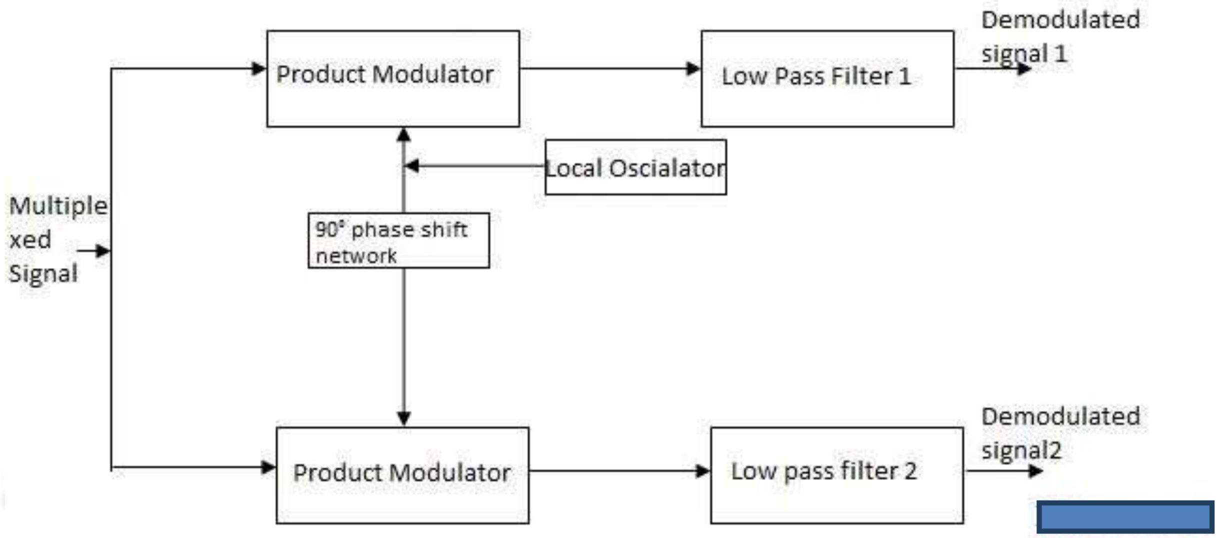
The below diagrams show the transmitter and receiver block diagram of the QAM scheme.

QAM Modulator



qam-modulator

QAM Demodulator



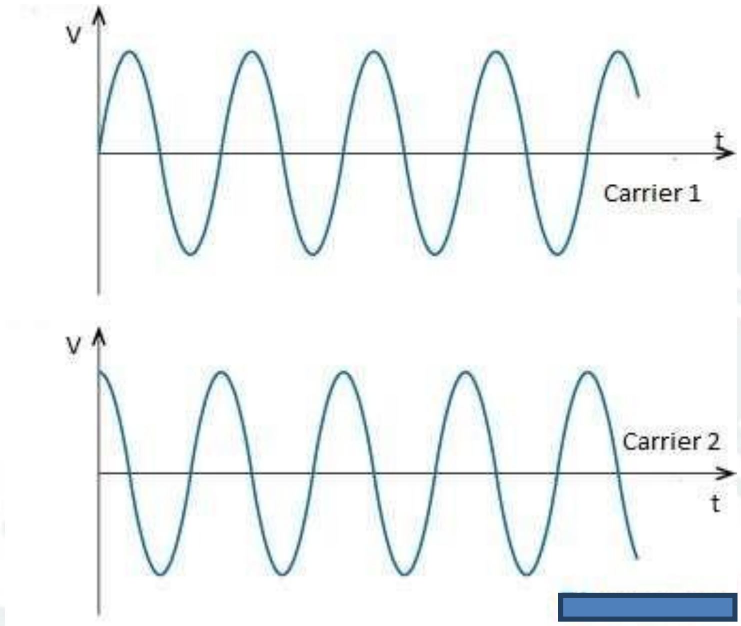
qam-demodulator

QAM Working Principle

"In the QAM transmitter, the above section i.e., product modulator1 and local oscillator are called the in-phase channel and product modulator2 and local oscillator are called a quadrature channel. Both output signals of the in-phase channel and quadrature channel are summed so the resultant output will be QAM."

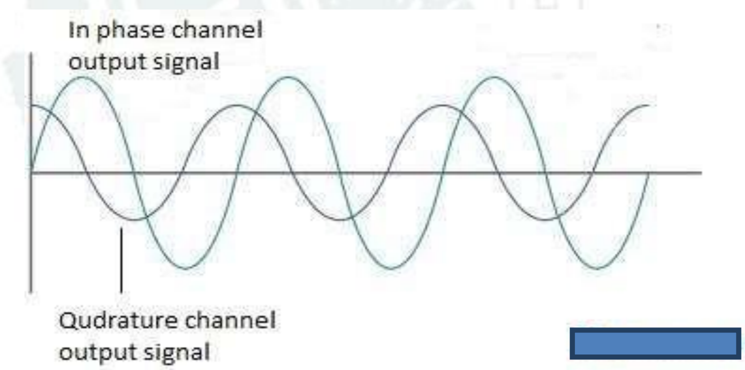
At the receiver level, the QAM signal is forwarded from the upper channel of receiver and lower channel, and the resultant signals of product modulators are forwarded from LPF1 and LPF2. These LPF's are fixed to the cut off frequencies of input 1 and input 2 signals. Then the filtered outputs are the recovered original signals.

The below waveforms are indicating the two different carrier signals of the QAM technique.



input-carriers-of-qam

The output waveforms of QAM is shown below.



quadrature-output-signal-waveform

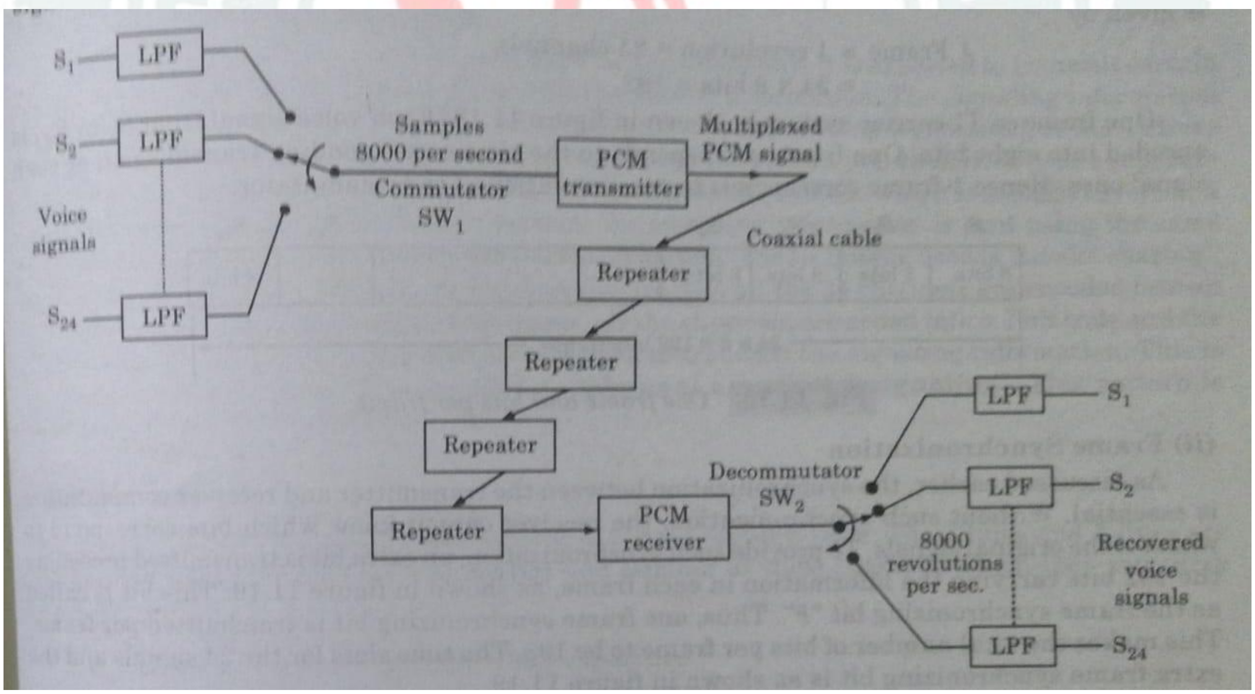
Minimum Shift Keying (MSK)

Minimum Shift Keying (MSK) is one of the most spectrally efficient modulation schemes available. Due to its constant envelope, it is resilient to non-linear distortion and was therefore chosen as the modulation technique for the GSM cell phone standard.

MSK is a special case of Continuous-Phase Frequency Shift Keying (CPFSK) which is a special case of a general class of modulation schemes known as Continuous-Phase Modulation (CPM). It is worth noting that CPM (and hence CPFSK) is a non-linear modulation and hence by extension MSK is a non-linear modulation as well. Nevertheless, it can also be cast as a linear modulation scheme, namely Offset Quadrature Phase Shift Keying (OQPSK), which is a special case of Phase Shift Keying (PSK). As a borderline case, these relationships are illustrated.

A PCM-TDM System (T1 Carrier System)

When a large number of PCM signals are to be transmitted over a common channel, multiplexing of these PCM signals is required. Figure shows the basic time division multiplexing scheme. called as the T1-digital system or TI carrier system. This system is used to convey multiple signals over telephone lines using wideband coaxial cable.



Working Operation of the TI Carrier System

The working operation of the PCM-TDM system shown in figure can be explained in the form of few points as under:

This system has been designed to accommodate 24 voice channels marked S_1 to S_2. Each signal is band limited to 3.3 kHz, and the sampling is done at a standard rate of 8 kHz. This is higher than the Nyquist rate. The sampling is done by the commutator witch SW_1.

These voices signals are selected one by one and connected to a PCM transmitter by the commutator switch SW_1.

Each sampled signal is the applied to the PCM transmitter which convert it into a digital signal by the process of A to D conversion and companding, as explained earlier.

The resulting digital waveform is transmitted over a co-axial cable.

Periodically, after every 6000 ft, the PCM-TDM signal is regenerated by amplifiers called "Repeaters. They eliminate the distortion introduced by the channel and remove the superimposed noise and regenerate a clean PCM-TDM signal at their output. This ensures that the received signal is free from the distortions and noise.

At the destination, the signal is companded, decoded and demultiplexed, using a PCM receiver. The PCM receiver output is connected to different low pass filters via the decommutator switch SW_2.

Synchronization between the transmitter and receiver commutators SW_1, and SW_2,is essential in order to ensure proper communication.

Spread Spectrum-

A collective class of signaling techniques is employed before transmitting a signal to provide a secure communication, known as the Spread Spectrum Modulation. The main advantage of spread spectrum communication technique is to prevent “interference” whether it is intentional or unintentional.

The signals modulated with these techniques are hard to interfere and cannot be jammed. An intruder with no official access is never allowed to crack them.

Hence, these techniques are used for military purposes. These spread spectrum signals transmit at low power density and has a wide spread of signals.

FHSS and DSSS / CDMA

Spread spectrum multiple access techniques uses signals which have a transmission bandwidth of a magnitude greater than the minimum required RF bandwidth.

These are of two types.

Frequency Hopped Spread Spectrum FHSS

Direct Sequence Spread Spectrum DSSS

Frequency Hopped Spread Spectrum FHSS

This is frequency hopping technique, where the users are made to change the frequencies of usage, from one to another in a specified time interval, hence called as frequency hopping. For example, a frequency was allotted to sender 1 for a particular period of time. Now, after a while, sender 1 hops to the other frequency and sender 2 uses the first frequency, which was previously used by sender 1. This is called as frequency reuse.

The frequencies 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.

Direct Sequence Spread Spectrum DSSS

Whenever a user wants to send data using this DSSS technique, each and every bit of the user data is multiplied by a secret code, called as chipping code. This chipping code is nothing but the spreading code which is multiplied with the original message and transmitted. The receiver uses the same code to retrieve the original message.

Bit-

The bit is the most basic unit of information in computing and digital communications. The name is a contraction of binary digit. The bit represents a logical state with one of two possible values.

Baud-

A unit of transmission speed equal to the number of times a signal changes state per second. For signals with only two possible states one baud is equivalent to one bit per second.

Channel Capacity Formula-

According to channel capacity equation, C = B log(1 + S/N)

C-capacity

B-bandwidth of channel

S-signal power,

N-noise power,

when B -> infinity (read B 'tends to' infinity),

capacity saturates to 1.44S/N.

Modem & Application of Modem

Modem stands for Modulator and Demodulator. It is a device that modulates signals to encode digital information for transmission and demodulates signals to decode the transmitted information.

A modem transmits data in bits per second (bps).

It is necessary for communication between digital devices and Analog devices.

Modem is necessary because it acts as a translator between the devices and rapidly transmits the information.

It converts the digital signal to Analog and vice versa to communicate between devices.

It encodes the signal and decodes at the other end and vice versa between the devices.

Tyepes of Modem-

Telephone modem

Digital Subscriber line

Cable modem

Satellite modem