# ELECTRONICS & TELECOMMUNICATION ENGG.

Semester-4th

Subject -**Communication Engg.**


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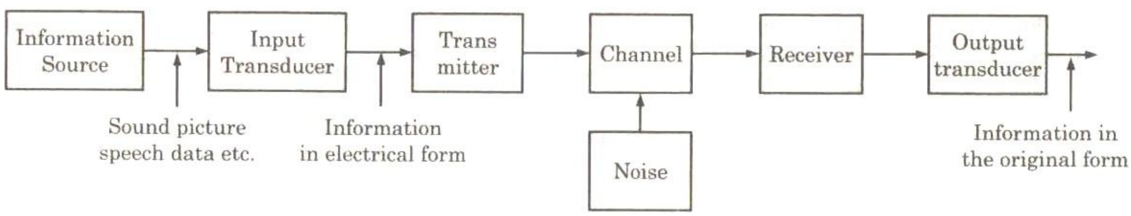
# ELEMENTS OF COMMUNICATION SYSTEM

## ELEMENTS OF COMMUNICATION SYSTEM-

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




*Block diagram of a communication system*

**Fig- 1.1**

## INFORMATION SOURCE-

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

## INPUT TRANSDUCER-

A transducer is a device which converts one form of energy into another form. The message from the information source may or may not be electrical in nature. When the message produced by the information source is not electrical in nature, an input transducer is used to convert it into a time varying electrical signal.

## TRANSMITTER-

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

## THE CHANNEL AND THE NOISE-

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

**RECEIVER-**

The main function of the receiver is to reproduce the message signal in electrical form from the distorted received signal. This reproduction of the original signal is accomplished by a process known as the demodulation. Demodulation is the reverse process of modulation carried out in transmitter.

**DESTINATION-**

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

## SOURCE OF INFORMATION-

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

(i) Speech (iii) Picture

(ii) Music (iv) Computer data

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

## Signal-

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

The signals may be classified as:

### (i) Speech

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

(1) Production

(2) Propagation and

(3) Perception

### (ii) Music signal-

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

(1) Melodic structure

(2) Harmonic structure

**(iii) Picture**

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

**(iv) Computer Data**

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

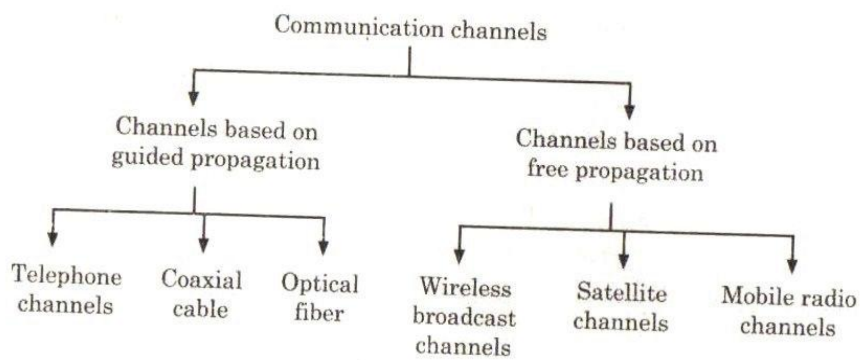
## COMMUNICATION CHANNEL-

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

(i) channels based on guided propagation

(ii) channels based on free propagation

The classification of channels has been shown below:




*Classification of communication channels*

**Fig- 1.3**

Some of the important characteristics of a channel are:

i. power required to achieve the desired s/n ratio

ii. bandwidth of the channel

iii. amplitude and phase response of channel

iv. type of channel (linear or nonlinear)

v. effects of external interference on the channel

## CLASSIFICATION OF COMMUNICATION SYSTEM-

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

(i) Analog communication system

(ii) Digital communication system

### Analog communication system

Analog communication is that type of communication in which the message or information signal to be transmitted is analog in nature. This means that in analog communication the modulating signal is an analog signal.

### Digital communication system

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

## ANALOG AND DIGITAL SIGNALS-

### Analog signal-

It is a continuous signal for which the time varying feature of the signal is a representation of some other time varying quantity i.e, analogous to another time varying signal. For example, in an analog audio signal, the instantaneous voltage of the signal varies continuously.

### Digital signal-

A digital signal is a physical signal that is a representation of a sequence of discrete values, a signal that is generated by means of a digital modulation, to be transferred between modems. The signals that are discrete in time and quantized in amplitude are digital signals.

**TRANSFORMATION POWER AND CHANNEL BANDWIDTH:**

Transformation Power: Communication system consisting of sub-systems and each sub-system is consisting of different functional blocks. Each functional blocks consisting of different parameters like input and output devices which performs a specific signal processing and information transmission. The power required for transmission of a particular signal is known as transformation power.

**Bandwidth**: The frequency range or band of frequency needed for a particular given transmission is known as Bandwidth.

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# AMPLITUDE MODULATION SYSTEM

## NEED OF MODULATION-

1) Practicality of Antenna

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

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

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

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

## CLASSIFICATION OF MODULATION:-

There are three types of Modulation.

1) Amplitude Modulation
2) Frequency Modulation
3) Phase Modulation

## AMPLITUDE MODULATION:-

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

Sinusoidal Carrier wave

$$C(t) = A \cos \omega_c t$$

Where $A$= Maximum Amplitude

$\omega_c$ = Carrier Frequency

Let x(t) denotes the modulating or baseband signal, then according to amplitude modulation, the maximum amplitude A of the carrier will have to be made proportional to the instantaneous amplitude of modulating signal x(t). Amplitude modulated signal expressed as:-

S (t) = x (t) cos$\omega_c$t + $A$cos$\omega_c$t

S (t) = [$A$+x (t)] cos$\omega_c$t

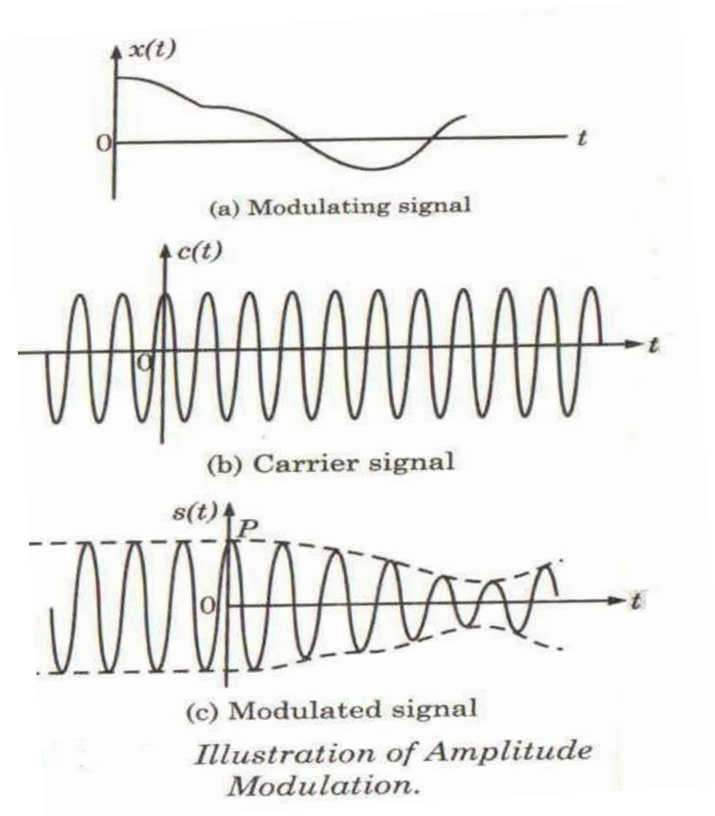



**Fig 2.1**

## MODULATION INDEX:-

$$M_a = \frac{|X(t)|\ max}{Maximum\ \ carrier\ \ amplitude}$$

$$= \frac{|X(t)|\ max}{A}$$

This is also known as depth of modulation degree of modulation or modulation factor over modulation if $m_a$>1 the baseband signal is not preserved in the envelope. It means the baseband signal recovered from the envelope will be distorted .this type of distortion is called envelop distortion.

## POWER RELATION IN AM WAVE:-

Carrier power,

$P_c = \overline{(Acos\ \omega_c t})2\ =\ A^2/2$ ……………..(1)

Sideband power

$P_s = ½\ \overline{x^2(t)}$

$=½(\overline{V_m Cos\ \omega_m t})^2$

$= 1/2(\frac{V_m^2}{2})$

$=\frac{V_m^2}{4}$

Total Modulated power

$P_t = P_c + P_s$

$= A^2/2 + \frac{V_m^2}{4}$

$=A^2/2[1 + 1/2(\frac{V_m^2}{2})\ ]$

$= A^2/2[1 + \frac{M_a^2}{2}\ ]$

$P_c = A^2/2$ …………..(1)

$P_t = A^2/2\ [1 + \frac{M_a^2}{2}\ ]$ …………….(2)

$I_t = I_c\sqrt{[1 + \frac{M_a^2}{2}]}$ …………….(3)

## DEMODULATION OF AM WAVE:-

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

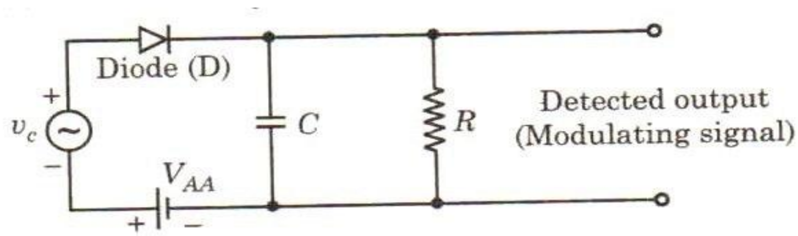
Types of detector (1) square-law detectors

(2) Envelopedetectors

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

## SQUARE-LAW DETECTORS/LINEAR DIODE DETECTOR:-

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




*Basic circuit of square law diode detector.*

**Fig. 2.2**

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

This distorted output diode current is expressed by

$I = av + bv^2$

v=is the i/p modulated voltage

AM wave is expressed as

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

Substituting, the value of v, we get

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

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

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

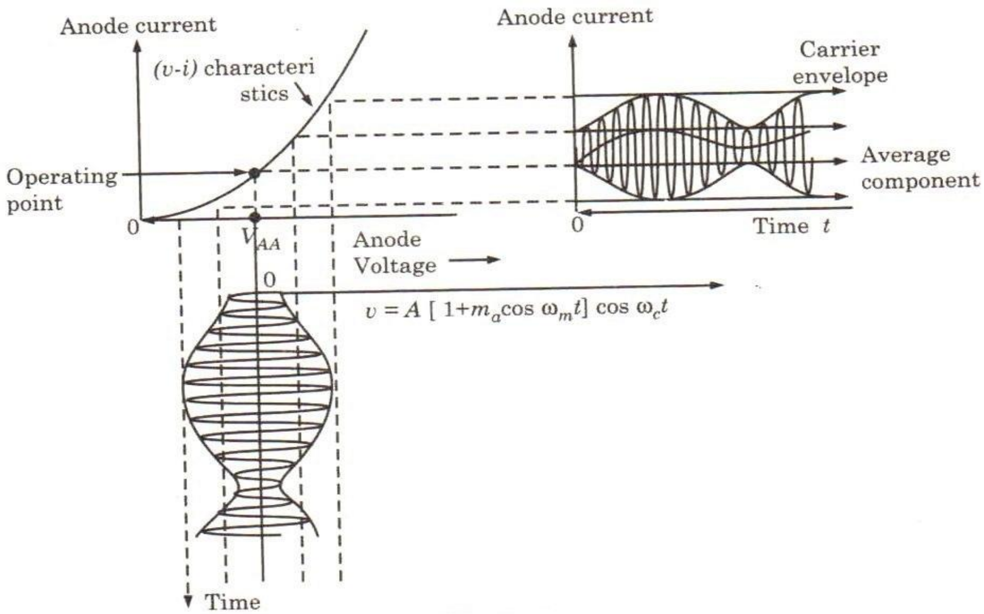



**Fig 2.3**

## ENVELOPE DETECTOR:-

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

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

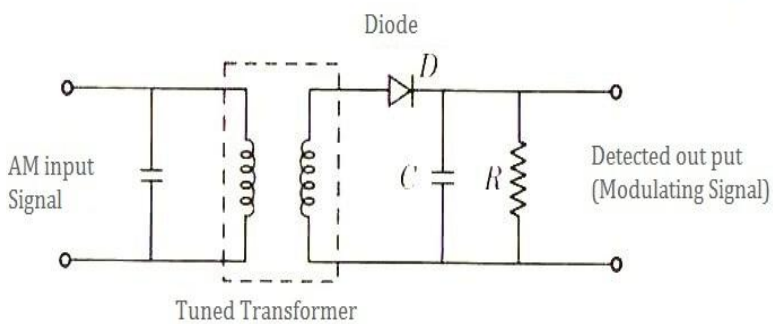



**Fig. 2.4**

**Operation**:-

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

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

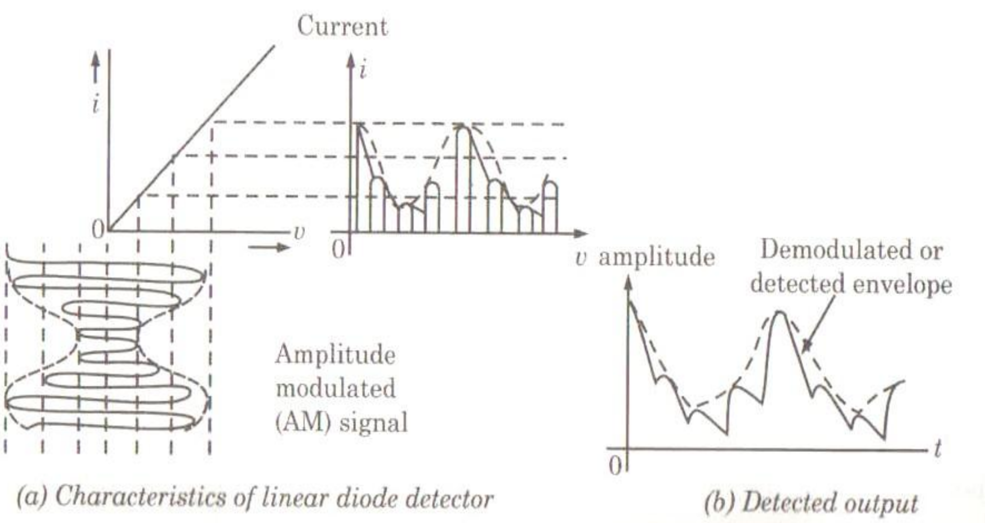



**Fig. 2.5**

## AM Demodulator using Phase locked loop

Demodulated

PM Output~ $\theta_i$

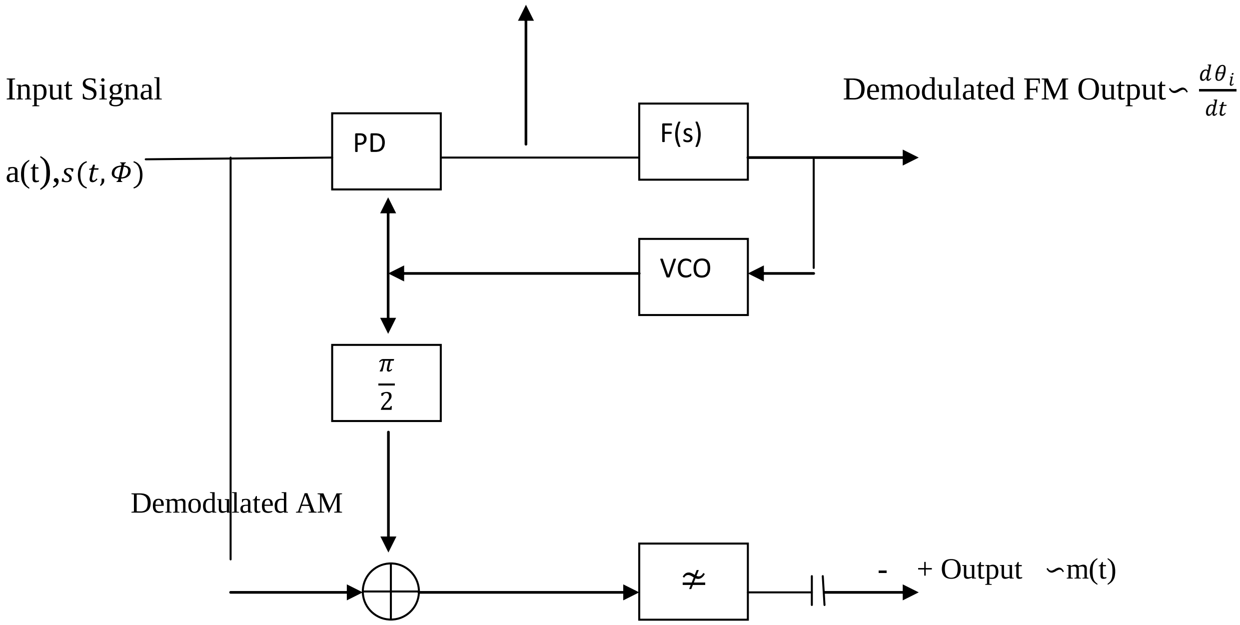



**Fig. 2.6 AM demodulation by A PLL**

Let the input signal be amplitude modulated

$x(t) = [1 + m(t)]\sqrt{2}A\sin(\omega_i t + \theta_{i0})$ …………….(1)

where$m(t)$ carries the information, and A, $\omega_i t$ and $\theta_{i0}$ are constants.

The PLL demodulator contains a carrier recovery circuit (see the PLL in Fig. 5) and an AM demodulator (see the analog multiplier and low-pass filter in Fig. 5). Since the PLL needs an input signal to be tracked continuously, the spectrum of the AM signal must contain a carrier component.

The carrier is recovered by the PLL, its VCO output is

$r(t,\hat{\varphi}) = \sqrt{2}V_o\cos(\omega_i t + \theta_{i0})$ ……….........(2)

This signal is multiplied by the AM input signal. The low-pass filter selects the difference-frequency output of multiplier and the DC blocking capacitor removes its DC component. The demodulated signal is obtained from Eqs. (1) and (2)

$AV_o m(t)$ ……………..(3)

where$AV_o$ is the gain of the AM demodulator.

## DSB-SC

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

## SSB-SC

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

Generation-

SSB-SC signals may be generated by two methods

(i) Frequency discrimination
(ii) Phase discrimination

### FREQUENCY DISCRIMINATION METHOD-

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

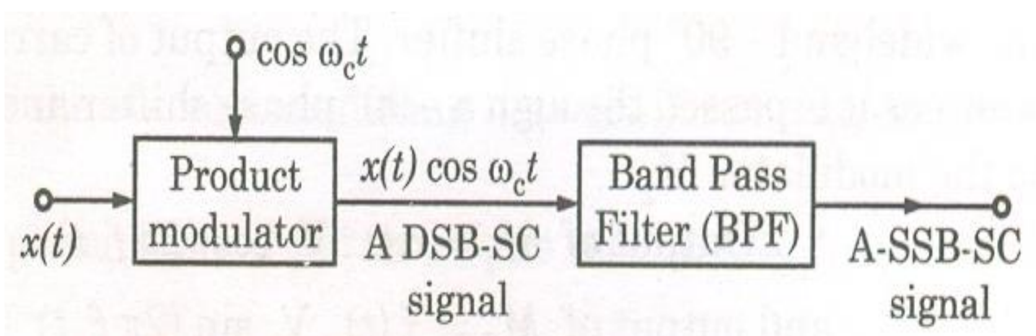



**Fig. 2.7 Frequency Discrimination Method for SSB SC Generation**

**Limitations**-

- The frequency discrimination method is useful only if the base band signal is restricted at its lower edge due to which the upper and lower sidebands are non-overlapping.
- The design of the band pass filter becomes difficult if the carrier frequency is quite higher than the bandwidth of the baseband signal.

## PHASE-SHIFT METHOD-

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

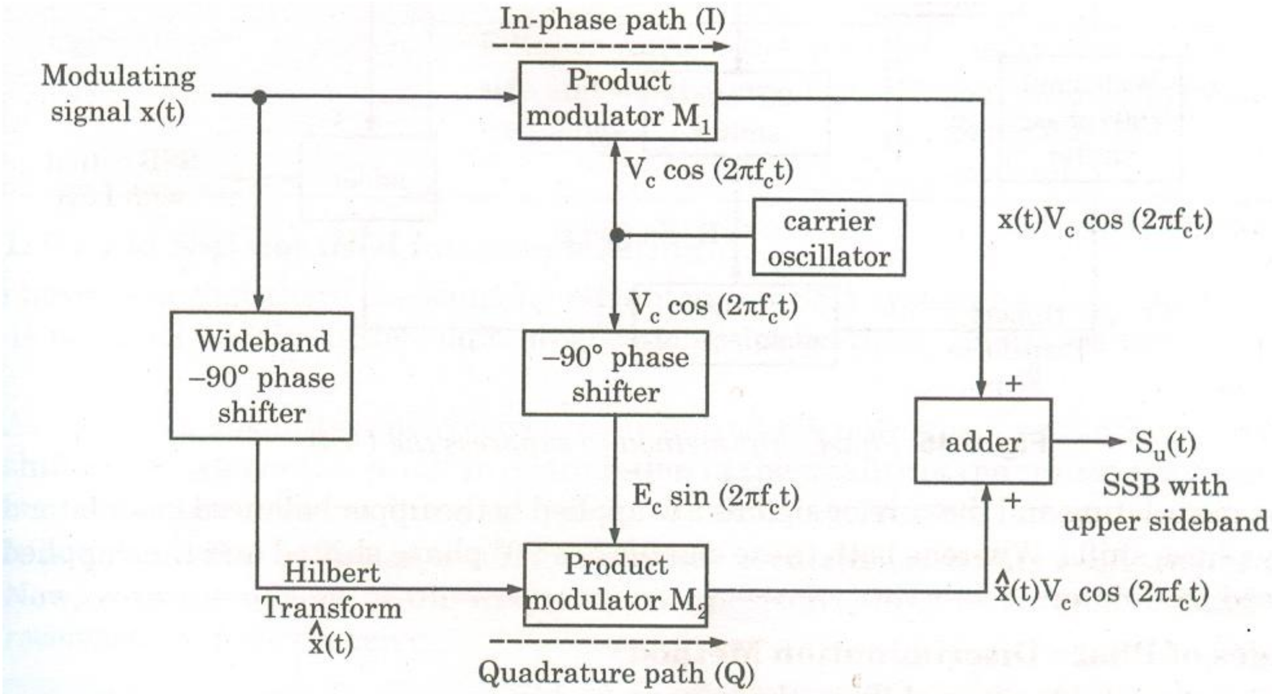



**Fig.2.8 Phase Discrimination Method for SSB- SC signal**

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

## DEMODULATION-

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

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

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

$=1/2\ x(t)+ ½\ [x(t)\ \mathrm{Cos}2\omega_c t \pm x_n(t)\ \mathrm{Sin}2\omega_c t]$

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

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

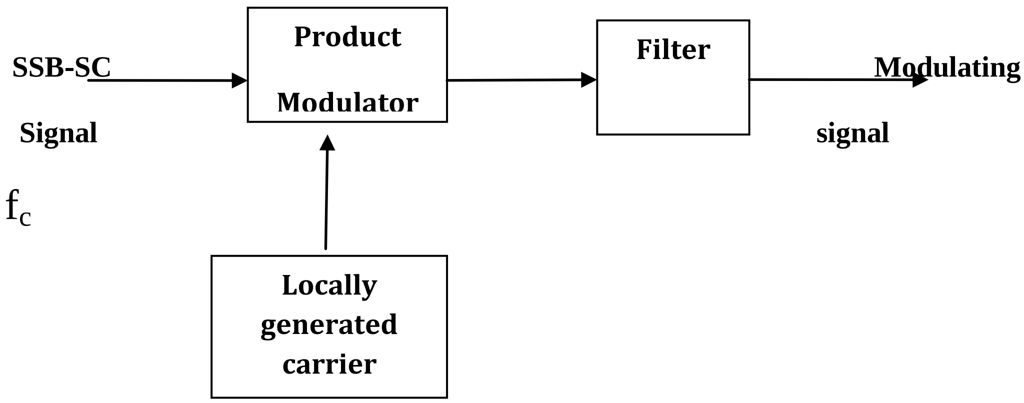



**Fig. 2.9 Block diagram of coherent SSB demodulator**

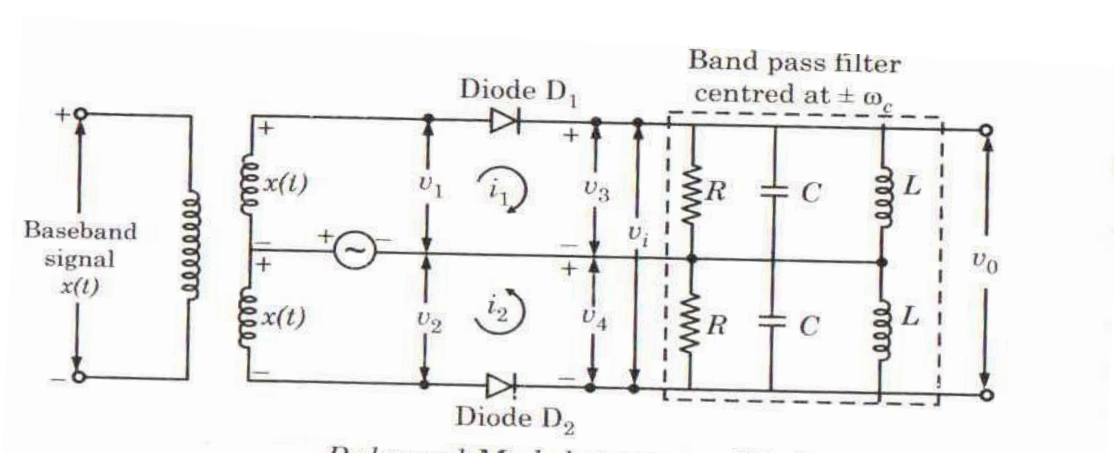
## GENERATION OF DSB-SC SIGNAL-

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

1. Balanced Modulator
2. Ring Modulator

### Balance Modulator :-

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




*Balanced Modulator using Diodes.*

**Fig. 2.10**

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

A non- linear VI relationship is given as,

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

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

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

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

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

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

After substituting the values of $i_1$ & $i_2$ we get

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

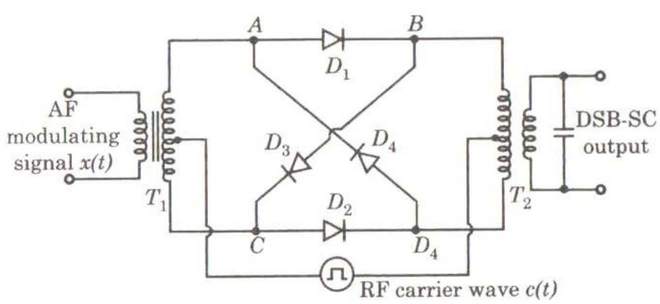
A band pass filter is that type of filter which allows to pass a band of frequencies. Here the band pass filter is centred around $\pm\omega_c$ , it will pass a narrow band of frequencies cantered at $\pm\omega_c$.

The output of the BPF is

$v_o = 4bRx(t)\cos \omega_c t$

## Ring Modulator-

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



*A diode ring modulator.*

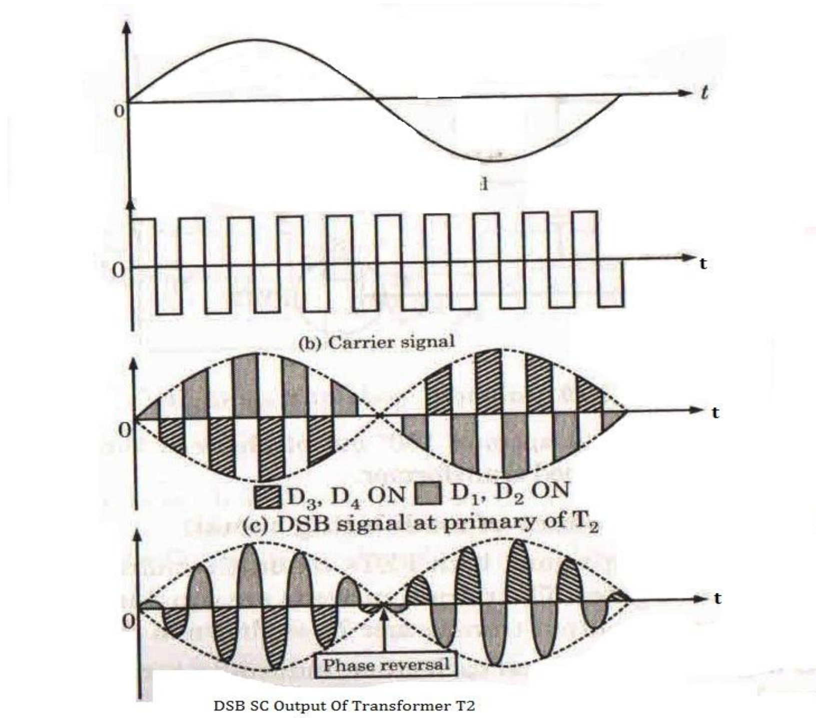
**Fig. 2.11**

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

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

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

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



**Waveform of Lattice type Balanced Modulator**

**Fig. 2.12**

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

## DETECTION OF DSB-SC SIGNAL-

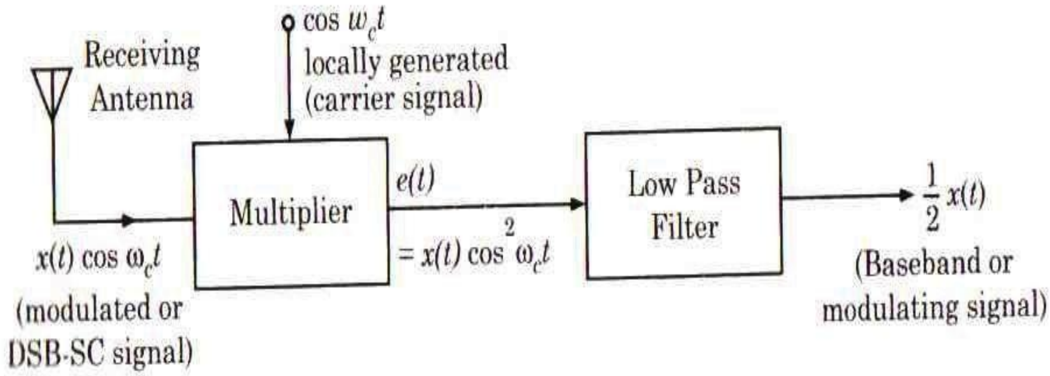
The DSB-SC signal may be demodulated by following two methods.

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

**Synchronous detection Method**-

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

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




*Synchronous detection method.*

**Fig 2.13**

**Working principle**-

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

## MULTIPLEXING-

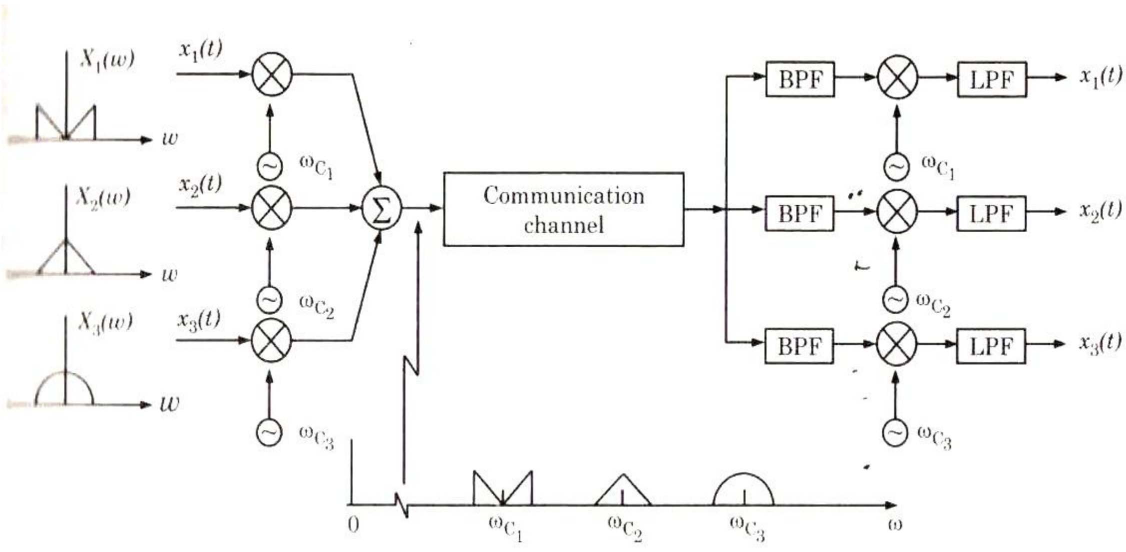
Simultaneous transmission of multiple messages over a single channel is known as multiplexing. Several message signals are combined into a composite signal for transmission over a common channel. In order to transmit a number of these signals over the same channel, the signals must be kept apart so that they do not interfere with each other, and hence they can be separated easily at receiver.

Multiplexing is of two types-

(i) Frequency division multiplexing (FDM)
(ii) Time division multiplexing (TDM)

## FREQUENCY DIVISION MULTIPLEXING-

The spectra of the message signals and the sum of the modulated carriers are indicated in the below figure. Any type of modulation can be used in FDM as long as the carrier spacing is sufficient to avoid spectral overlapping however, the most widely used method of modulation is SSB modulation. At the receiving end of the channel the three modulated signals are separated by band pass filters and then demodulated.




*Frequency-division multiplexing*

**Fig. 2.14**

FDM is used in telephone system, telemetry, commercial broadcast, Television and communication networks. Commercial AM broadcast stations use carrier frequency spaced 10 kHz apart in the frequency range from 540 to 1640 kHz. This separation is not sufficient to avoid spectral overlap for AM with a reasonably high fidelity (50 Hz to 15 kHz) audio signal. Therefore, AM stations on adjacent carrier frequencies are placed geographically far apart to minimize interference. Commercial FM broadcast uses carrier frequencies spaced 200 kHz apart. In long distance telephone system, up to 600 or more voice signals are transmitted over a coaxial cable or microwave links by using SSB modulation with carrier frequencies spaced 4 kHz apart. The composite signal formed by spacing several signals in frequency may be modulated by using another carrier frequency. In this case, the first carrier frequencies are often called sub-carriers.

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# FREQUENCY MODULATION SYSTEMS

## FREQUENCY MODULATION

Frequency modulation is that type of angle modulation in which the instantaneous frequency $\omega_i$is varied linearly with a massage signal about an unmodulated carrier frequency $\omega_c$.This means that the instantaneous value of the angular frequency $\omega_i$will be equal to the carrier frequency $\omega_c$ plus a time varying component proportional to the baseband signal x(t).

## EXPRESSION FOR FREQUENCY MODULATED SIGNAL-

We know that the instantaneous frequency is given by

$\omega_i = \omega_c + k_f$ .x(t)

Where $k_f$ is proportionality constant and is known as the frequency sensitivity of the modulator. This is expressed in Hz /volt.

Now, let the expression for unmodulated carrier signal be

$$c(t)= A\ cos\ (\omega_c t + \theta_0)$$

or $$c(t)= A \cos \phi$$

where, $\varphi = \omega_c t + \theta_0$

'φ' is the total phase angle of the unmodulated carrier.

Let φ be the instantaneous phase angle of the modulated signal.

On frequency modulation amplitude A must remain constant and only angle 'φ' will change. Hence, the expression for frequency modulated wave will be

$$s(t) = A \cos \varphi_i$$

where $\varphi_i$= instantaneous phase angle

$$\varphi = \omega_c t + \theta_0$$

on differentiation, we get

$$\frac{d\varphi}{dt} = \omega_c$$

Or $$\varphi = \int \omega_c\ dt$$

Based on above equation, we may write the expression for instantaneous phase angle $\varphi_i$ as

$$\varphi_i = \int \omega_i\ dt$$

Where,$\omega_i$= instantaneous frequency of frequency modulated wave

Putting the value of $\omega_i$ in above equation, we get

$\varphi_i = \int [\omega_c + k_f . x(t)] dt \; = \; \omega_c t + k_f \int x(t) dt$

Putting this value of$\varphi_i$, we get the expression for frequency modulated wave will be

s(t) = $A \, cos[\omega_c t + k_f \int x(t) dt]$

If the phase angle of the unmodulated carrier is taken at t=0, then the limit of integration will be 0 to t.

In this case the expression for FM wave will be

$$s(t) = A \, cos[\omega_c t + k_f \int_0^t x(t) dt]$$

## MODULATION INDEX-

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

Modulation index, $m_f$ = frequency deviation/modulation frequency

Or $m_f = \frac{\Delta\omega}{\omega_m}$

This modulation index may be greater than unity.

## FREQUENCY SPECTRUM OF FM SIGNAL-

The frequency spectrum of the signal

v(t) = $\cos(\omega_c \, t + \beta \sin \omega_m)$ …….(1)

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

We have

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

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

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

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

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

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

Putting the results given and using the identities

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

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

We find v(t) becomes

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

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

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

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

## PHASE MODULATION-

Phase modulation is that type of angle modulation in which the phase angle φ is varied linearly with a baseband or modulating signal x(t) about an unmodulated phase angle $\omega_c$t + $\theta_0$ . This means that in phase modulation, the instantaneous value of the phase angle is equal to the phase angle to the phase angle of the unmodulated carrier plus a time varying component which is proportional to modulating signal x(t).

Unmodulated carrier signal is expressed as

$$c(t)= A \cos [\omega_c t + \theta_0 ]$$

Or $c(t) = A \cos \varphi$

Where $\varphi = \omega_c t + \theta_0$

Neglecting $\theta_0$, we get total phase angle of unmodulated carrier is

$$\varphi = \omega_c t$$

Now according to phase modulation, this phase angle 'φ'is varied linearly with a baseband or modulating signal x(t).

Let the instantaneous value of phase angle be denoted by$\varphi_i$. Therefore,

$$\varphi_i = \omega_c t + k_p.x(t)$$

Where is the proportionality constant and is known as phase sensitivity of the modulator. This is expressed in radians/volts.

Since, the expression for unmodulated carrier wave is

$$c(t) = A \cos \varphi$$

Therefore, the expression for phase modulatedwave will be

$$s(t) = A \cos \varphi_i$$

Putting the value of φ in equation 4.11 from equation 4.10, we get

$$s(t) = A \cos [\omega_c t + k_p . x(t)]$$

Which is the required mathematical expression for a phase modulated wave.

## COMPARE BETWEEN AM AND FM MODULATION-

### AM-

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

### FM-

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

## FM GENERATION-

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

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

## PARAMETER VARIATION METHOD-

In direct method or parameter variation method, the baseband or modulating signal directly modulates the carrier. The carrier signal is generated with the help of an oscillator circuit. This oscillator circuit uses a parallel tuned L-C circuit. Thus the frequency of oscillation of the carrier generation is governed by the expression

$$\omega_c = \frac{1}{\sqrt{LC}}$$

We can make the carrier frequency $\omega_c$ to vary in accordance with the baseband or modulating signal x(t)if L or C is varied according to x(t). An oscillator circuit whose frequency is controlled by a modulating voltage is called voltage controlled oscillator (VCO). The frequency of VCO is varied according to the modulating signal simply by putting a shunt voltage variable capacitor with its tuned circuit. This voltage capacitor is called varactor or varicap. This type of property is exhibited by reverse biased semiconductor diodes. The capacitance of BJT and FET is varied by the Miller-effect. This Miller capacitance may be utilized for frequency modulation. The electron tubes may also provide variable reactance which is proportional to modulating or baseband signal. This type of tubes are called reactance tubes and may be used for FM generation.

The inductance L of the tuned circuit may also be varied in accordance with the baseband or modulating signal x(t).

## Varactor Diode Method for FM Generation:

The Varactor diode is a semiconductor diode whose junction capacitance changes with d.c. bias voltage. This varactor diode is connected in shunt with the tuned circuit of the carrier oscillator.

In varactor diode FM generation arrangement, the capacitor C is made much smaller than the varactor diode capacitance $C_d$so that the radio frequency voltage from oscillator across the diode is small as compared to reverse bias d.c. voltage across the varactor diode. In addition to this the reactance of the capacitor C at the highest modulating frequency is made large enough compared to resistor R. so that the shunting of the baseband or modulating signal through the tuned ckt. may be checked.

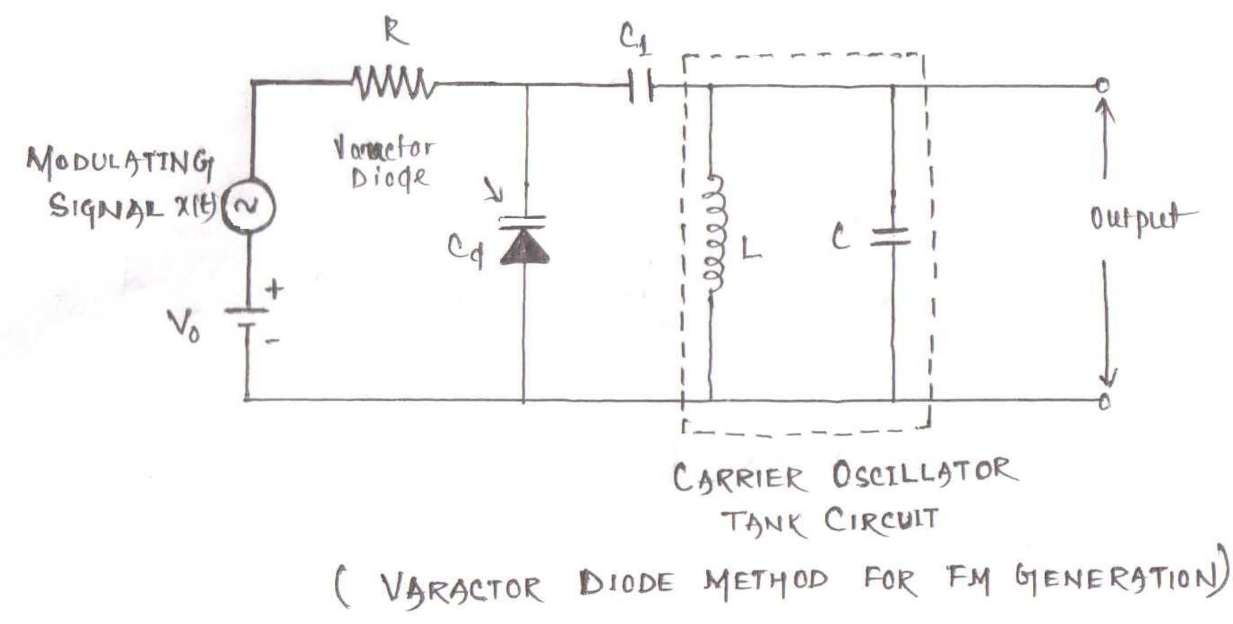



**Fig. 3.1**

**Mathematical analysis**:-

The Capacitance $C_d$ of the varactor diode is expressed as,

$C_d = \frac{K}{\sqrt{V_D}} = K(V_D)^{-1/2}$……………(1)

Here, $V_D$ is the total varactor diode and is given by

$V_D = V_0 + x(t)$ …………….(2)

Also, K is a constant of proportionality. The oscillation frequency is given as,

$$\omega_c = 1/\sqrt{LC}$$

Now, the total capacitance of the oscillator tank ckt will be $C_0 + C_d$ and thus the instantaneous frequency of oscillation

$\omega_i = 1/\sqrt{L_0(C_0 + C_d)}$ ……………..(3)

In above equation, substituting the value of $C_d$ from equation, we have

$\omega_i = 1/\sqrt{L_0(C_0 + K{V_D}^{-1/2})}$ ……..(4)

We conclude that the instantaneous frequency $\omega_i$ of FM signal depends upon $V_D$ which intern depends upon the value of the modulating signal x(t). Thus the instantaneous oscillator frequency $\omega_i$also depends upon the baseband signal x(t) and hence frequency modulation is generated.

## INDIRECT METHOD OR THE ARMSTRONG METHOD:

A phase-modulated waveform in which the modulating waveform is $m(t)$ is the modulating waveform and cos [$\omega_c$t + $(t)$ ]. If the modulation is narrowband [$|m(t)| \ll 1$] , then we may use the approximation

$\cos[\omega_c t + m(t)] \cong \cos \omega_c t - m(t) \sin \omega_c t$ ……................(1)

The term $m(t)$ and $\sin \omega_c t$ is a DSB-SC waveform in which $m(t)$is the modulating waveform and $\sin \omega_c t$ is the carrier. We note that the carrier of the FM waveform, that is$\cos \omega_c t$, and the carrier of the DSB-SC waveform are quadrature. We may note in passing that if the two carriers are in phase, the result is an AM signal since

$\cos \omega_c t + m(t) \cos \omega_c t \cong [1 + m(t)] \cos \omega_c t$ …………………………….(2)

A technique used in commercial FM system s to generate NBFM. Here balanced modulator is employed to generate the DSB-FC signal using $\sin \omega_c t$ as the carrier of the modulator. This carrier is then shifted in phase by $90^0$ and, when added to balanced modulator output, thereby forms an NBFM signal. However the signal so generated will be phase-modulated rather than frequency-modulated. If we desire that the frequency rather than phase be proportional to the modulation $m(t)$, then,we need merely integrate the modulating signal before application to the modulator.

If the system is to yield an output signal whose phase deviation is directly proportional to the amplitude modulating signal,then the phase deviation must be kept small. That such is the case is readily to be seen. If we neglect the small second –order correction in the carrier amplitude and assume it to be of unit magnitude, we have tan φ= $\Delta_1$. Since, however, $\Delta_1$ (=$\beta$ sin $\omega_m$t) is proportional to the modulating signal, we actually require that φ = $\Delta_1$. In order that we may replace tan φ by φ, we require that all times φ≪1. In this case $\beta$ ≪1, and then φ = $\beta$ sin $\omega_m$t.

The restriction that $\beta$ ≪1 imposes a similar constraint on the allowable frequency deviation

$\Delta f$ = ($\beta\omega_m/2\pi$) when the system is adapted for use as a frequency modulation system by the addition of an integrator.

## FM DEMODULATOR-

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

## Foster Seeley Discriminator-

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

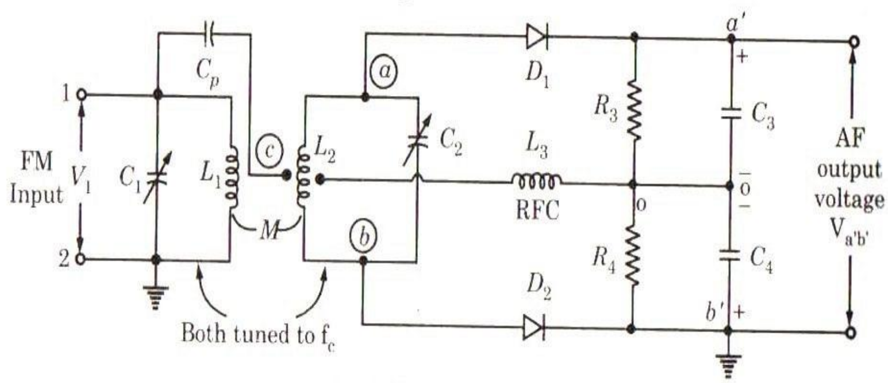
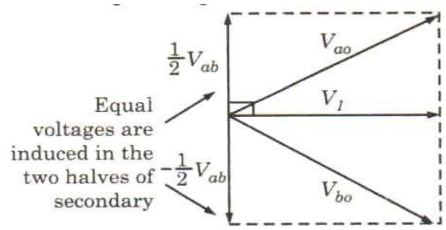



**Fig3.2Frequency Discriminator**

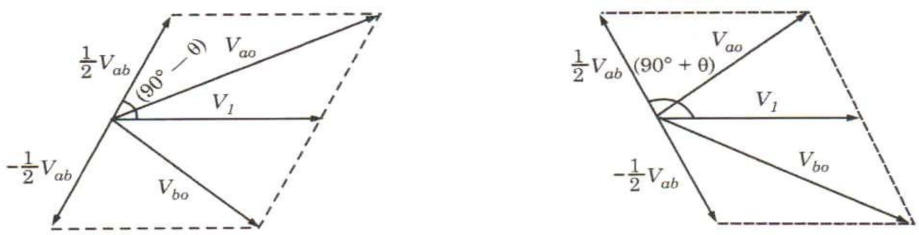
This circuit consists of an inductively coupled double tuned circuit in which both primary and secondary coils are tuned to the same frequency. The center of the secondary coil is connected to the top of the primary through a capacitor C. this capacitor performs the functions are:

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

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




*Relation between primary and secondary.*




(a) Secondary equivalent circuit (b) Phasor diagram for $f_{in} = f_c$

**Fig3.3 Phasor diagram**

Even though the primary and secondary tuned circuits are tuned to the same center frequency, the voltages applied to the two diodes D1 and D2 are not constant. They may vary depending on the frequency of the input signal. This is due to the change in phase shift between the primary and secondary windings depending on the input frequency. The result of this is as follows:

(i) At $f_{in} = f_c$, the individual output voltages of the two diodes will be equal and opposite. The output voltage is zero as,

$$v_0$$

(ii) For $f_{in} > f_c$, the phase shift between the primary and secondary windings is such that the output of D1 is higher than D2. Hence, the output voltage will be positive.

(iii) For $f_{in} < f_c$, the phase shift between the primary and secondary windings is such that output of D2 ids higher than that of D1 making the output voltage negative.

This is clearly observe in the above Phase diagrams.

**Advantages:**

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

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

**Drawbacks**

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

*******

# AM & FM RECEIVER

## SELECTIVITY-

It is a measure of the performance of radio receiver to respond only to respond only to the radio signal, it is tuned to and reject other signals nearby in frequency such as another broadcast on an adjacent channel.

## SENSITIVITY-

The sensitivity of an electronic device, such as a communication system receiver, or detection device, is the minimum magnitude of input signal required to produce a specified output signal. Receiver sensitivity indicates how faint an input signal can be successfully received by the receiver.

## FIDELITY-

It is the degree to which output of a system, such as an amplifier or radio, accurately reproduces the characteristics of the input signal.

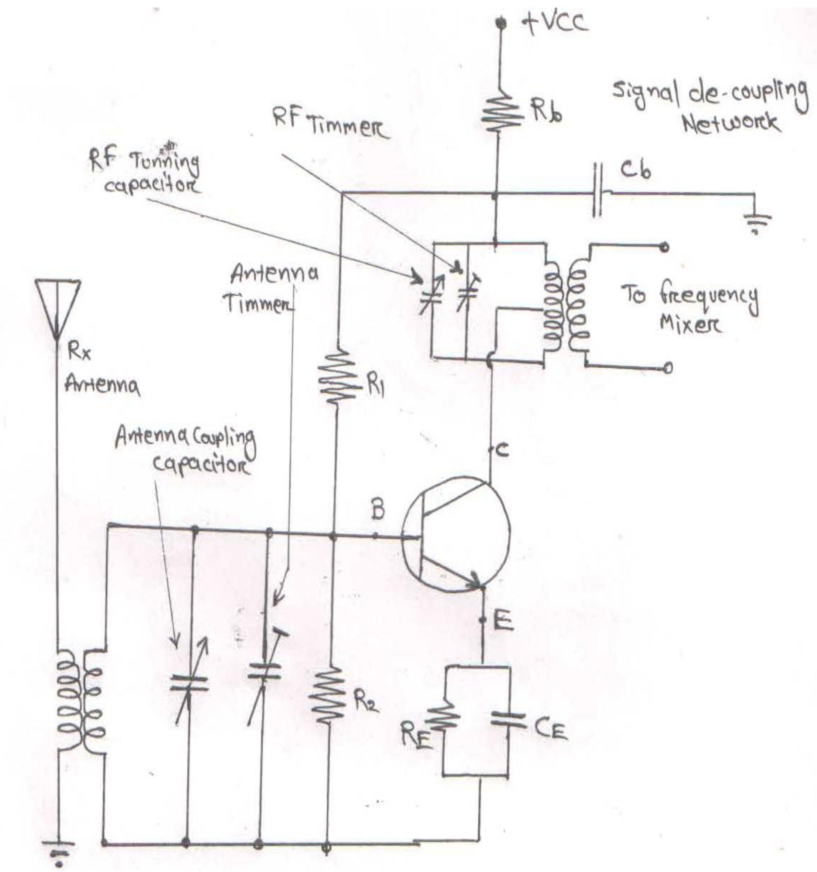
## NOISE FIGURE-

Noise figure is a measure of degradation of the signal to noise ratio, caused by components in a radio frequency signal. It is defined as the ratio of the signal to noise power ratio at the input to signal to noise power ratio at the output.

$F = S_i/N_i$

## R.F AMPLIFIER MIXER USING TRANSISTOR AND I.F. AMPLIFIER-

R.F. Amplifier is a small signal tuned amplifier with tuned circuits both in the input side and the output side. Both these input and output tuned circuits are tuned to the desired incoming carrier frequency. Accordingly, the tuned circuits select the desired carrier frequency and reject all undesired frequencies including the image frequency. Hence the R.F. Amplifier provides image frequency rejection. Also the gain provided by the R.F. amplifier will result in improved signal/ noise ratio in the output of the receiver. This is due to the fact that the incoming weak signal is raised to a higher the output the receiver. This is due to the fact that the incoming weak signal is raised to a higher level with the help of RF amplifier before it is fed at the input of the mixer stage which contributes most of the noise generated in the receiver. However, if the incoming weak signal is fed directly to the frequency mixer, signal/noise ratio at the output of the mixer stage is quite poor and hence any amount of subsequent amplification cannot improve S/N ration. Thus the one important function of the RF amplifier is to improve S/N ratio.



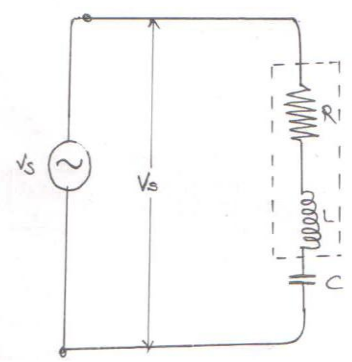




**Fig. 4.1** **A Series Resonance Circuit** **Fig. 4.2** **Circuit Diagram of RF Amplifier**

The circuit diagram of one stage RF amplifier using an NPN transistor. It is a small signal amplifier using parallel tuned circuit as the load impedance. This parallel output tuned circuit is tuned to the incoming desired signal frequency. The output from the receiving antenna is the transformer coupled to the base of the transistor. The secondary coil of the input tuned circuit is tuned to the incoming desired signal frequency with the help of ganged tuning capacitor. In fact the tuning capacitors i.e. variable air capacitors in the input side and the output side of the RF amplifier are ganged together. In addition to this, small trimmer capacitors are connected in shunt with this tuning capacitors for the purpose of RF alignment.

A self-biased is provided with the help of the resistors $R_1$, $R_2$, $R_e$- $C_e$ assembly. A decoupling network consisting of resistor $R_b$ and capacitor $C_b$ is placed in the collector supply lead.

The amplified RF signal developed across the collector tuned circuit is coupled through a step down transformer to the input of the frequency mixer. This step down transformer provides the impedance matching between the high impedance of the RF amplifier collector circuit and the low impedance of the base to emitter circuit of the following stage.

Also the collector is connected to a suitable point on the primary of the output transformer so that load impedance to the collector is optimum.

## IMAGE SIGNAL SELECTION OF I.F. AND ALIGNMENT OF RECEIVER

A super heterodyne receiver suffers from a major drawback known as Image Frequency problem. This problem of image frequency is inherent to a super heterodyne receiver and arises because of the use of heterodyne principle. In fact, the frequency conversion process carried out by the local oscillator and the mixer often allow an undesired frequency in addition to the desired incoming frequency.

In standard broadcast receiver, the local oscillator frequency is always made higher than the incoming signal frequency. It is kept equal to the signal frequency plus the intermediate frequency (I.F).

Mathematically,

$$f_0 = f_s + f_i$$

where $f_0$ = local oscillator frequency

$f_s$ = desired incoming frequency

$f_i$ = intermediate frequency

From above equation

$$f_0 = f_s - f_i$$

Hence, the intermediate frequency is the difference between the local oscillator frequency and the signal frequency.

Now, if a frequency $f_{si}$ = manages to reach the mixer, such that

$$f_{si} = f_0 + f_i$$

then this frequency $f_{si}$ would also produce $f_i$ when it is mixed with $f_0$. This undesired or spurious intermediate frequency signal will also be amplified by the I.F. stage and thus would cause interference. This has the effect sources or stations being received simultaneous. This situation is obviously undesirable.

The term $f_{si}$ is known as the image frequency and is defined as the signal frequency plustwice the intermediate is obviously undesirable.

Putting the value of $f_0$ in equation (2) from eq. (1), we get

$$f_{si} = f_0 + f_i$$

$$f_{si} = f_0 + f_i + f_i$$

or $$f_{si} = f_0 + 2f_i$$

Thus this spurious frequency signal cannot be distinguished by the I.F. stage and hence would be treated in the same manner as the desired frequency signal.

The rejection of an image frequency signal by a single tuned circuit may be defined as the ration of the gain at the signal frequency to the gain at the image frequency. This is given as

$$\alpha = \sqrt{(1 + Q^2\rho^2)}$$

here $\rho = \frac{f_{si}}{f_s} - \frac{f_s}{f_{si}}$

Q = quality factor of the tuned circuit in a loaded condition

If the receiver has an RF stage, then there is only a single tuned circuit and the rejection will be calculated using equation (4). However if the receiver has an RF stage then there are two tuned circuits both tuned to $f_s$. The total or overall rejection will be the product of the two.

The image frequency rejection of the receiver depends upon the front end selectively of the receiver. The rejection of image frequency must be achieved before the I.F. stage. Once an undesired or spurious frequency enters the first I.F. amplifier, it would become impossible to remove it from the desired signal.

It may be observed that if $\frac{f_{si}}{f_s}$ is large as is the case for AM broadcast band the use of an RF stage is not necessary for good image frequency rejection. However, it would become essential above about 3MHz

## FM RECEIVER—

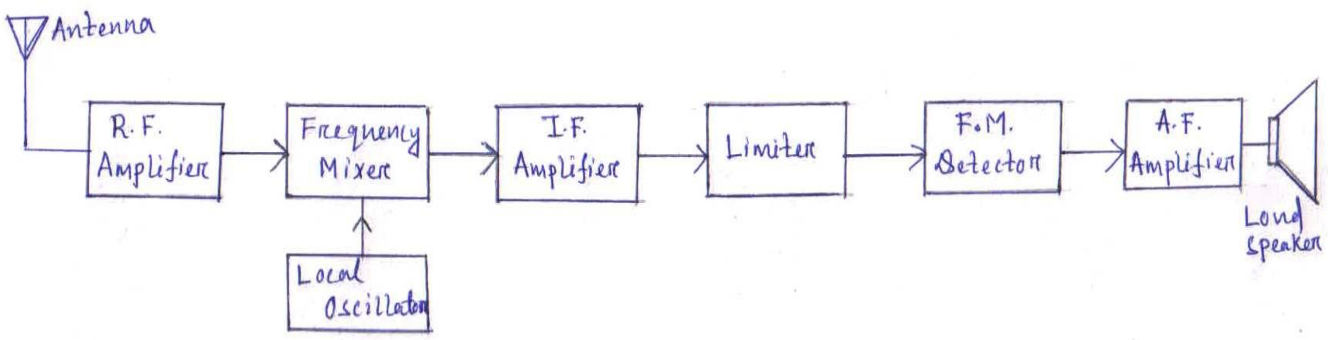



**Fig 4.3 FM Receiver**

***R.F. Amplifier-***

It is used to raise the signal level appreciably before the signal is fed to the mixer. But in F.M. broadcast, the signal bandwidth is large. Hence the RF amplifier must be designed to handle this large bandwidth.

***Frequency mixer*—**

It performs the usual function of mixing the signal frequency voltage and local oscillator voltage to produce the difference voltage, which is known as intermediate frequency voltage. The intermediate frequency used in FM receiver is higher than that in AM receivers.

***IF Amplifier*—**

A multistage IF amplifier is used to provide large gain.

***Limiter*—**

The IF amplifier is followed by a limiter which limits the IF voltage to a predetermined level and thus removes all amplitude variations which may be incidentally caused due to changes in the transmission path.

***FM detector*—**

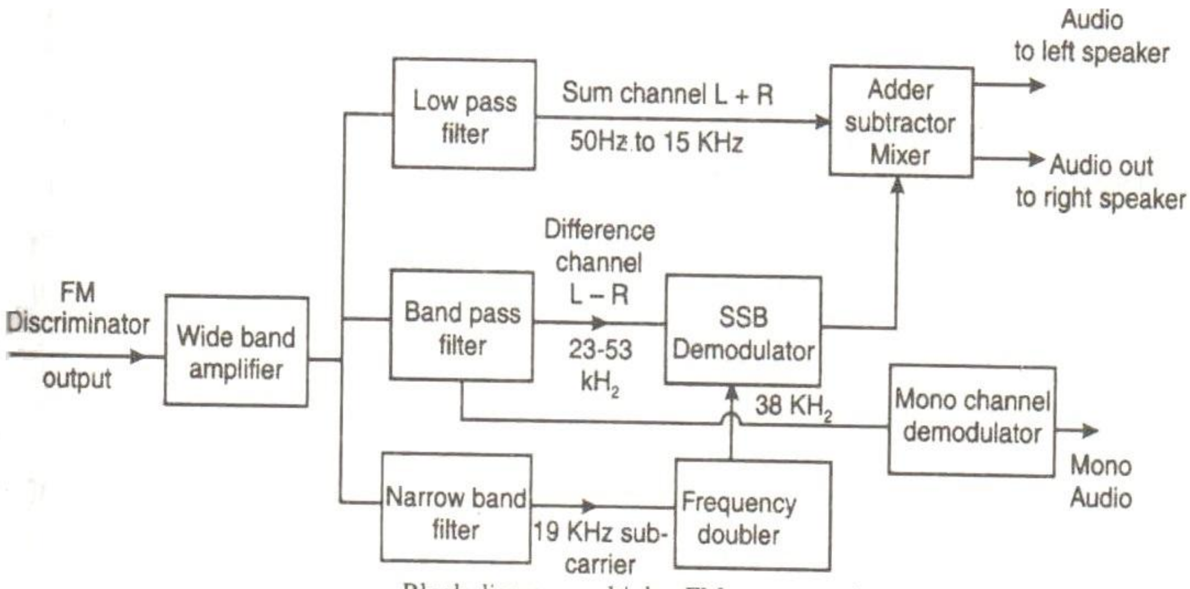
This extracts the original signal from the frequency modulated signal.

***Audio amplifier*—**

The output of the FM detector is fed to an audio frequency amplifier. It amplifies the signal to a required level, until it is capable to drive the speaker. The output of the audio frequency amplifier is then fed to the loudspeaker.

## STEREO PHONICS FM RECEIVER—

Stereo FM receiver, receives the two channels, these are separated and we hear the stereo effect. The stereo receiver reverses the process obtained in the stereo transmitter. The L+R signal confined in the 0 to 15 kHz frequency range is directly to a low pass filter. The L-R signal contained in the 23 to 53 kHz frequency range is extracted by an appropriate band pass filter. The 19 kHz carrier is extracted by an appropriate filter and converted to 38 kHz in a frequency in a frequency doubler. The 38 kHz carrier is mixed with L-R signal in an AM demodulator. Here it functions as the carrier for the L-R double side bands. The output of the AM demodulator is the L-R audio signal. This output is combined with the L+R audio and mixed in a adder and subtractor circuit produces the two channels output for stereo effect. The outputs of this stage are the original L & R channel information.



Block diagram multiplex FM stereo receiver.

**Figure 4.4**

## SUPER HETERODYNE RADIO RECEIVER—

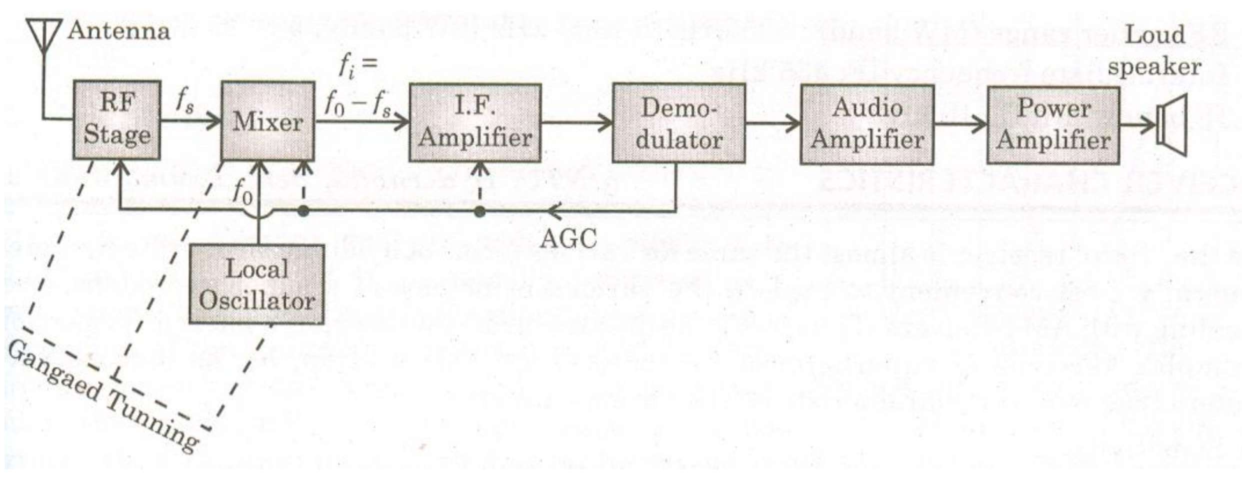



**Fig 4.5 Block Diagram of Super-heterodyne radio Receiver**

In super heterodyne receiver, the incoming RF signal frequency is combined with the local oscillator signal frequency through a mixer and is converted into a signal of lower fixed frequency. This lower fixed frequency is known as intermediate frequency. However, the intermediate frequency signal contains the same modulation as the original signal. This intermediate frequency signal is amplified and demodulated to reproduce the original signal.

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

Thus, in a super heterodyne receiver, a constant frequency difference is maintained between the local oscillator signal frequency and incoming RF signals frequency through capacitance tuning in which the capacitances are ganged together and operated by common control knob.

After the IF amplifier, the signal is applied at the input of demodulator which extracts the original modulating signal (i.e, audio signal). This audio signal is amplified by an audio amplifier to get a particular voltage level. This amplified audio signal is further amplified by a power amplifier to get a specified power level so that it may activate the loudspeaker. The loudspeaker is a transducer which converts this audio electrical signal into sound signal and thus the original signal is reproduced.

******

# ANALOG TO DIGITAL CONVERSION & PULSE MODULATION SYSTEM

## SAMPLING THEOREM

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

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

(i) A band-limited signal of finite energy, which has no frequency component higher than $f_m$ Hz, is completely described by its sample values at uniform intervals less than or equal to $1/2f_m$ second apart.
(ii) A band-limited signal of finite energy, which has no frequency components higher than $f_m$ Hz, may be completely recovered from the knowledge of its samples taken at the rate of $2f_m$ samples per second.

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

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

## NYQUIST RATE

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

It is given by $f_s= 2f_m$

Nyquist Interval$T_s$= ½ $f_m$ sec

## ALIASING:

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

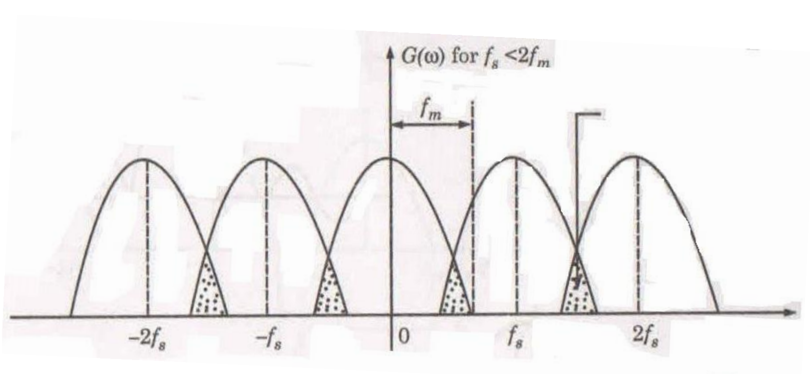



**Fig 5.1 Spectrum of the sampled signal for the case $f_s< 2f_m$**

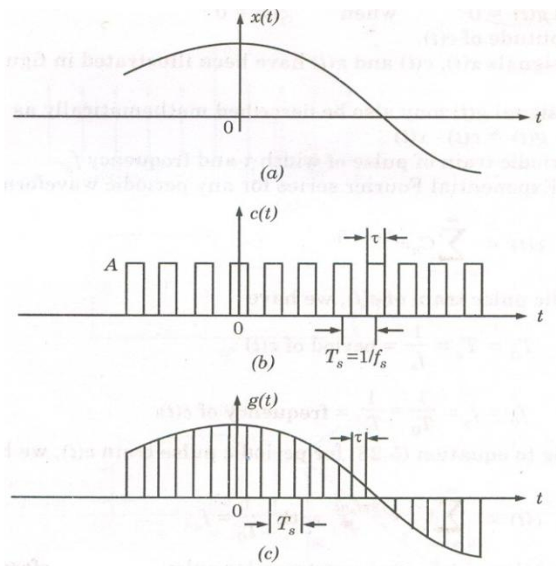
# <u>CLASSIFY SAMPLING</u>

There are 3 types of sampling techniques.

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

- **<u>Natural sampling</u>:**

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




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

**Fig. 5.2 Natural Sampling**

- **Flat- top Sampling:**

In this sampling the top of the pulses are flat.

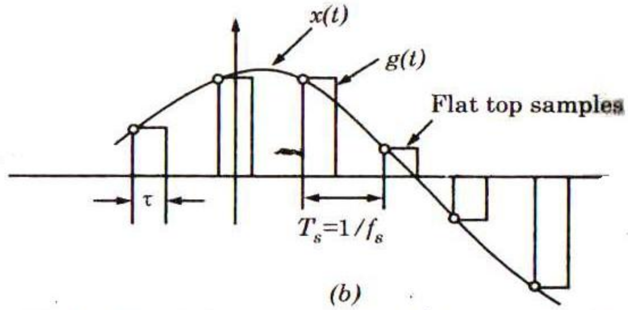



**Fig. 5.3 Flat-top sampling**

- **instantaneous sampling:**

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

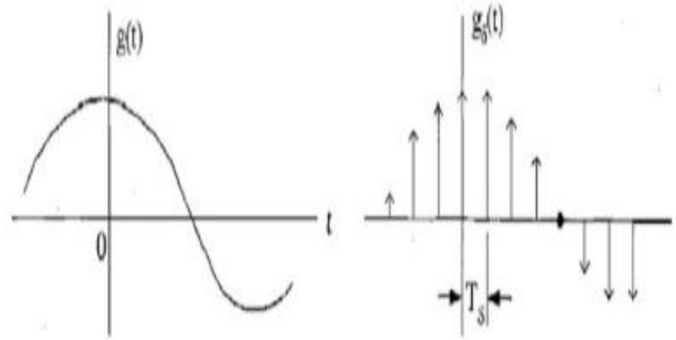



**Fig. 5.4 Instantaneous sampling**

## GENERATION OF PAM-

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

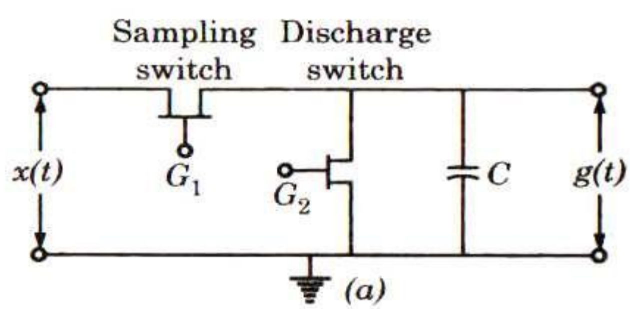
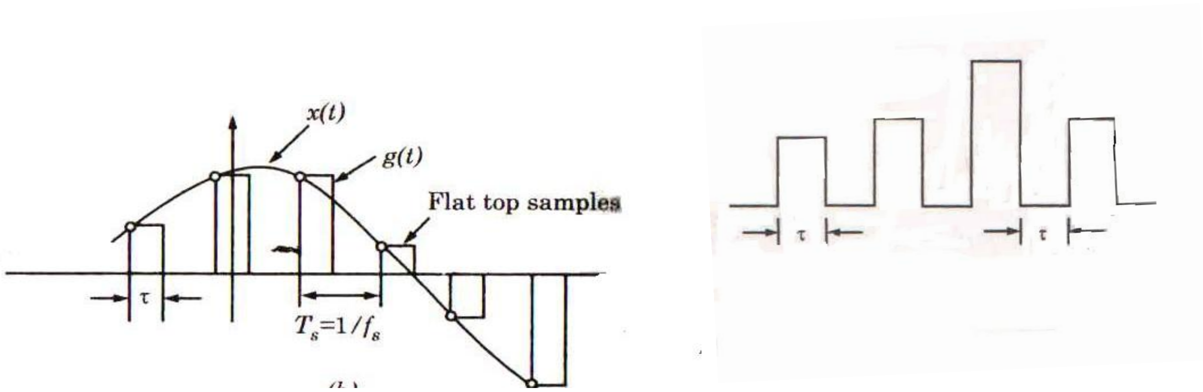



**Fig 5.5 Sample and hold circuit generation Flat top sampled PAM**

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

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




(a) (b)

**Fig5.6(a) & (b) Illustration of maximum frequency PAM Signal**

## DETECTION OF PAM-

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

**PAM Demodulated**

**Received signalSignal**

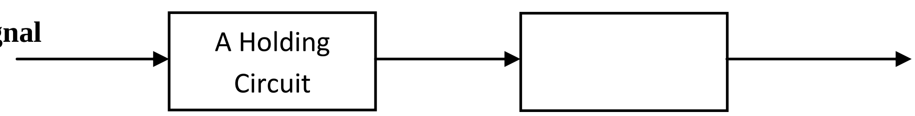



**Fig 5.7A Block diagram of PAM Demodulator**

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

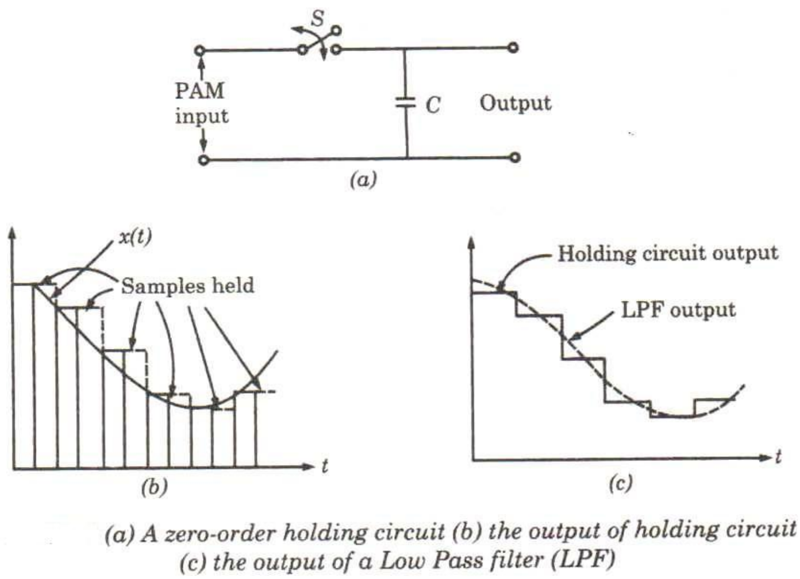



**Fig5.8PAM signal generator generating modulating Signal**

## GENERATION OF PWM -

It is basically a monostablemultivibrator with a modulating input signal applied at the control voltage input. Internally, the control voltage is adjusted to the 2/3 Vcc. Externally applied modulating signal changes the control voltage, and hence the threshold voltage level. As a result, the time period required to charge the capacitor up to threshold voltage level changes, giving pulse modulated signal at the output.

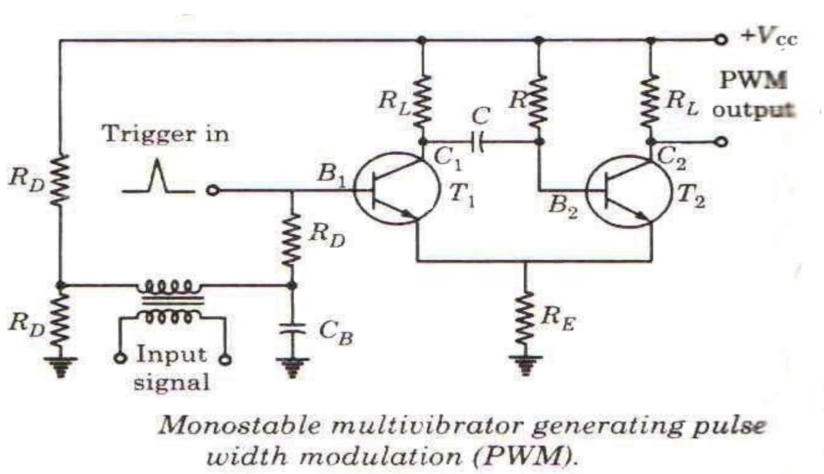



**Fig 5.9**

The stable state for above circuit is achieved when T1 is OFF and T2 is ON. The positive going trigger pulse at B1 switches T1 ON. Because of this, the voltage at C1 falls as T1 begins to draw the collector current. As a result, voltage at B2 also falls and T2 is switched OFF, C begins to charge up to the collector supply voltage through resistor R. After a time determined by the supply voltage and the RC time constant of the charging network, the base of the T2 becomes sufficiently positive to switch T2 ON. The transistor T1 is simultaneously switched OFF by regenerative action and stays OFF until the arrival of the next trigger pulse. To make T2 ON, the base of the T2 must be slightly more positive than the voltage across resistor Re. This voltage depends on the emitter current Ie which is controlled by the signal voltage applied at the base of transistor T1. Therefore, the changing voltage necessary to turn OFF transistor T2 is decided by the signal voltage. If signal voltage is maximum, the voltage that capacitor should charge to turn ON T2 is also maximum. Therefore, at maximum signal voltage, capacitor has to charge to maximum voltage requiring maximum time to charge. This gives us maximum us maximum pulse width at maximum input signal voltage. At minimum signal voltage, capacitor has to charge for minimum voltage and we get minimum pulse width at the output.

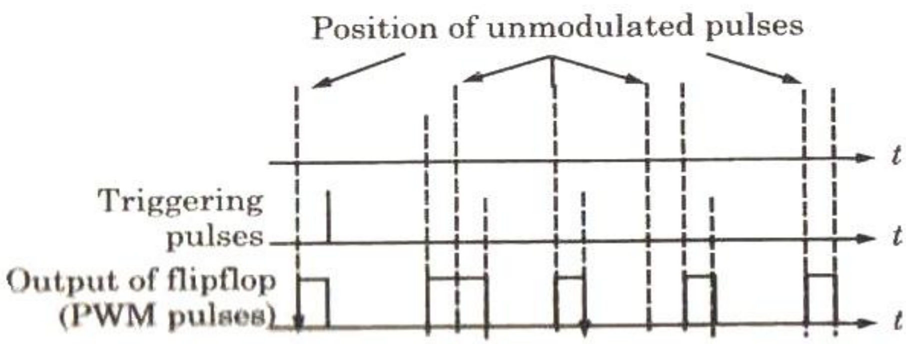



**Fig 5.10**

## DEMODULATION OF PWM-

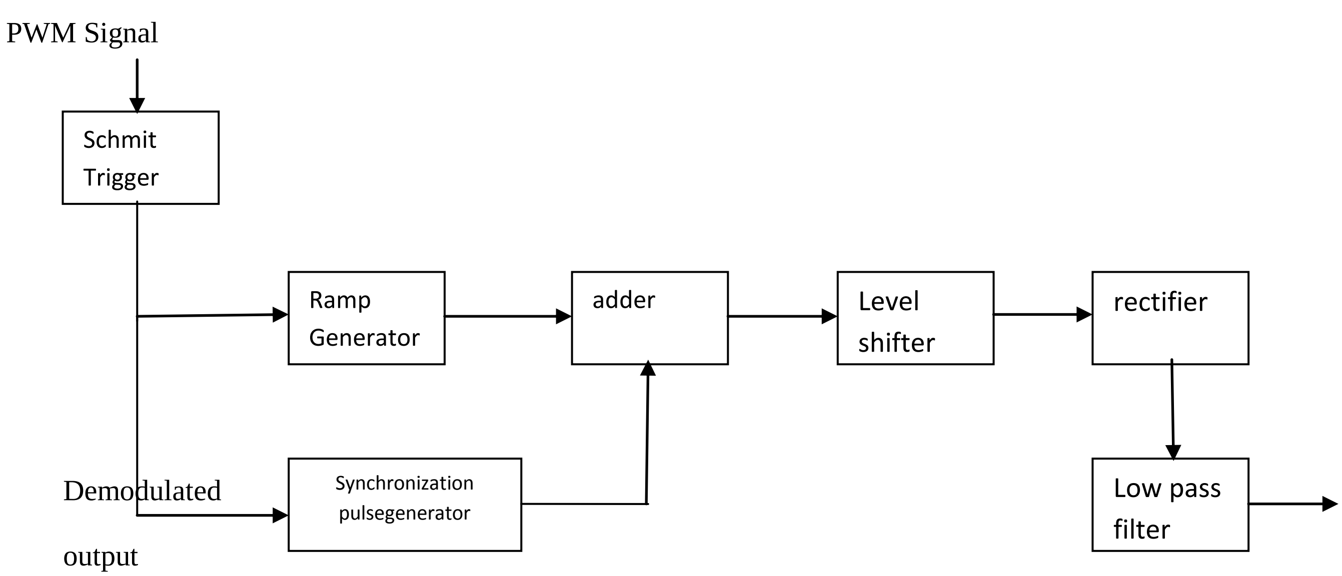



**Fig. 5.11 PWM Detector**

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

## GENERATION OF PPM SIGNAL-

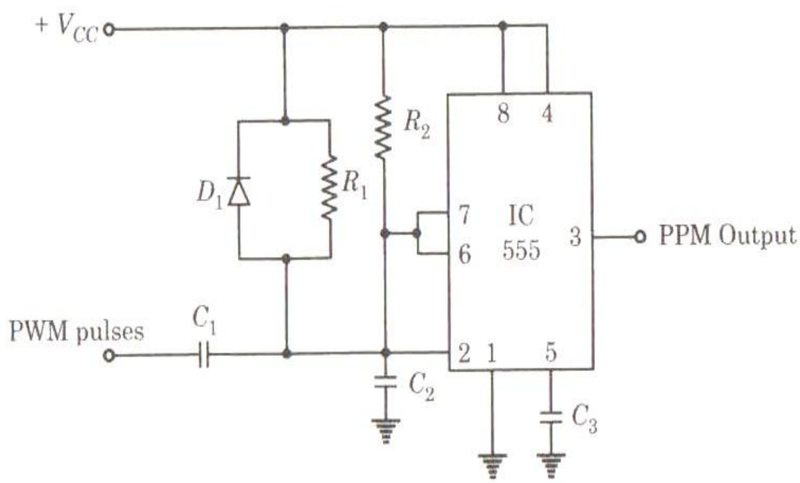



**Fig 5.12 PPM Generator**

It consists of differentiator and a monostablemultivibrator. The input to the differentiator is a PWM waveform. The differentiator generates positive and negative spikes corresponding to leading and trailing edges of the PWM waveform. Diode D1 is used to bypass the positive spikes. The negative spikes are used to the trigger monostablemultivibrator. The monostablemultivibrator then generates the pulses of same width and amplitude with reference to trigger to give pulse position modulated waveform.

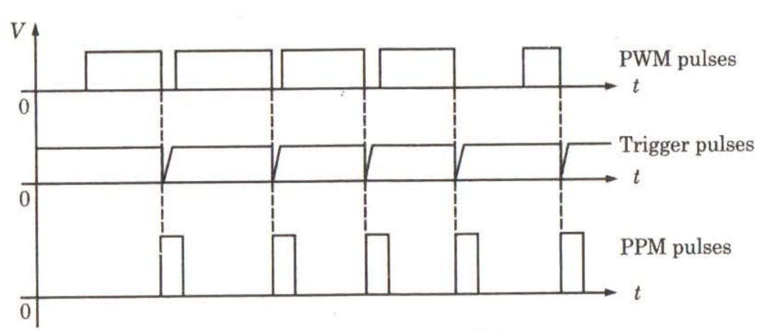



**Fig5.13 PPM generated Waveform**

## DETECTION OF PPM:-

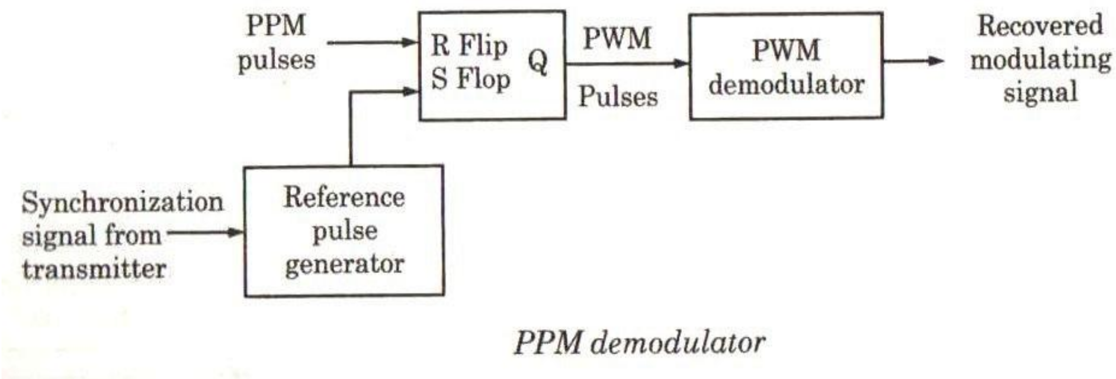



**Fig 5.14**

Flip flop is set or turned ON when the reference pulse arrives. This reference pulse is generated by reference pulse generator of the receiver with the synchronization signal from the transmitter. The flip flop circuit is reset or turned OFF at the leading edge of position modulated pulse. This repeats and we get PWM pulses at the output of the flip flop.

## QUANTIZATION:-

The process of representing continuous amplitude level to a constant finite amplitude level is quantization. Quantization makes the range of a signal discrete, so that the quantized signal takes on only a discrete, usually finite, set of values. Unlike sampling quantization is generally irreversible and results in loss of information. It therefore introduces distortion into the quantized signal that cannot be eliminated.

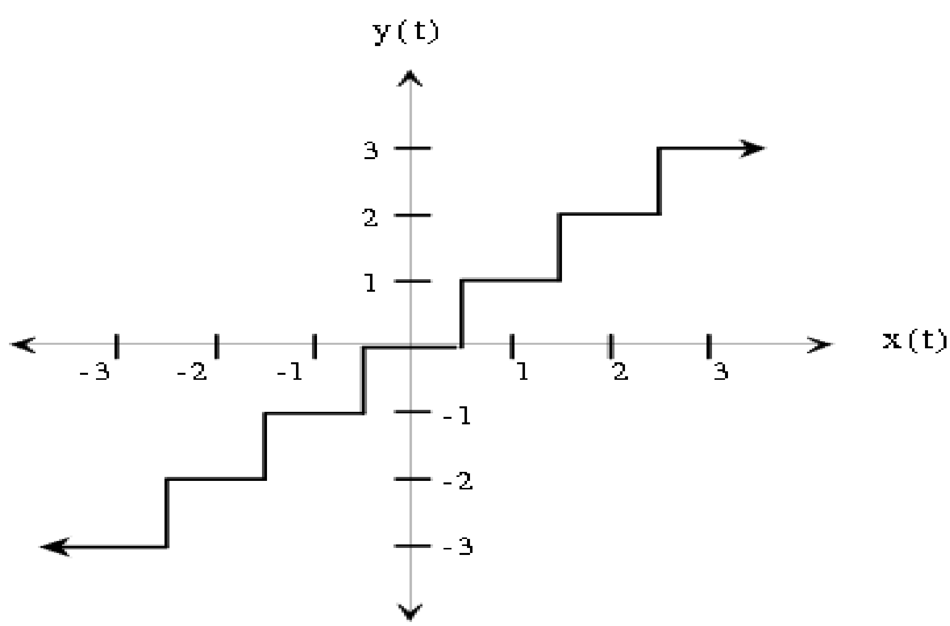



**Fig 5.15A Zero Memory Quantizer figure**

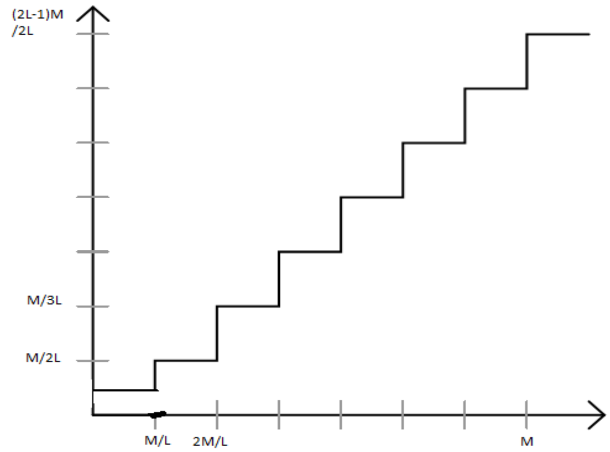



**Fig 5.16A uniform Quantizer figure**

## QUANTIZATION NOISE & QUANTIZATION ERROR:

The difference between the baseband signal m(t) and quantized signal $m_q(t)$ is known as quantization noise. Quantization error is nothing but the error due to quantization noise.

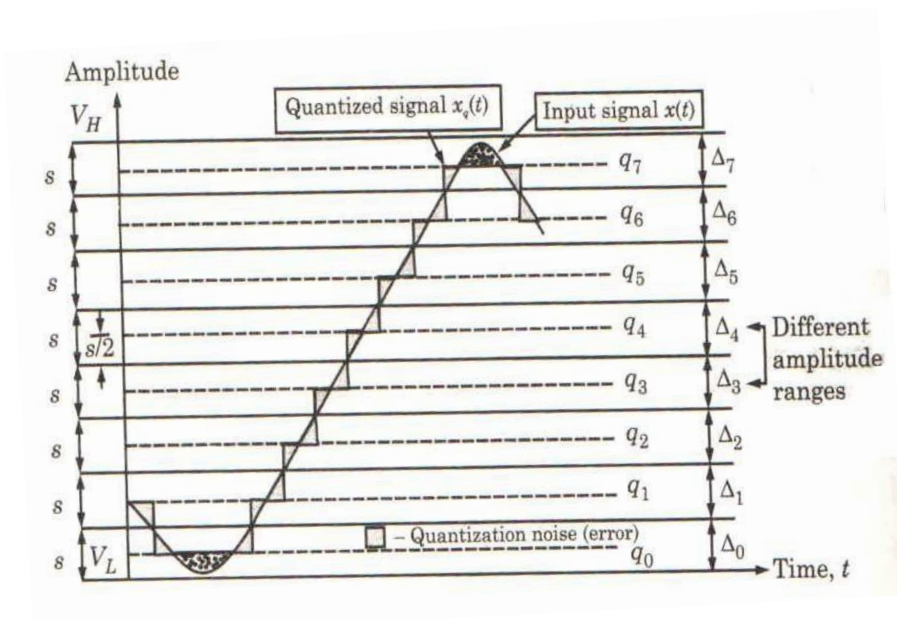



**Fig.5.17Illustration of Quantization Process**

$x_q(t)$ represents the quantized version of x(t). $x_q(t)$ is obtained at the output of the quantizer. When x(t) is in the range $\Delta_0$, then corresponding to any value of x(t), the quantizer output will be equal to $q_0$. Thus in each range from $\Delta_0$ and $\Delta_7$, the signal x(t) is rounded off to the nearest quantization level and the quantized signal is produced . The quantized signal $x_q(t)$ is thus an approximation of x(t). The difference between them is called as Quantization Error or Quantization noise.

$$\varepsilon = x_q(t) - x(t)$$

# PULSE CODE MODULATION-

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

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

## PCM GENERATION TRANSMITTER-

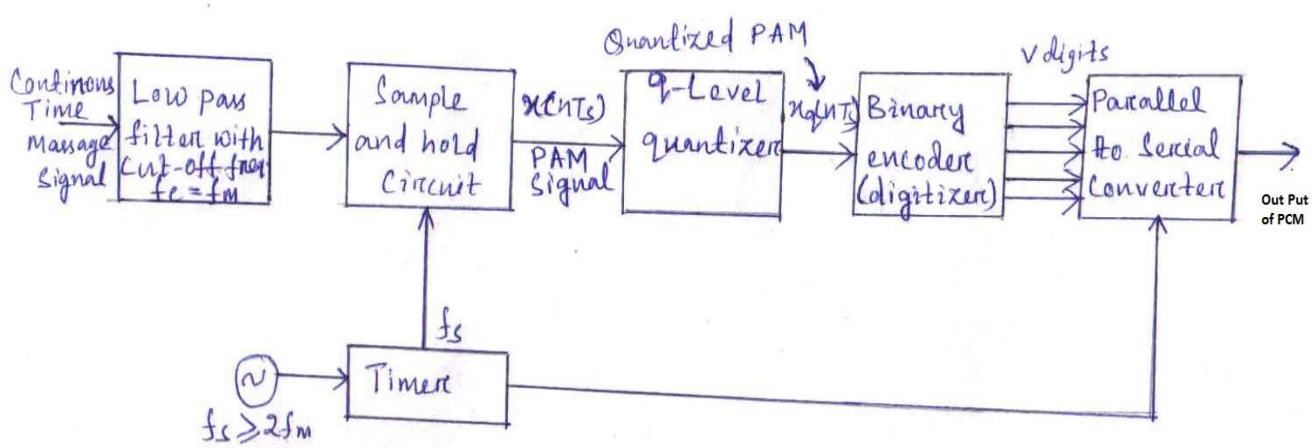



**Fig 5.18 PCM Generator**

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

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

$f_s \geq 2f_m$

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

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

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

## PCM TRANSMISSION PATH-

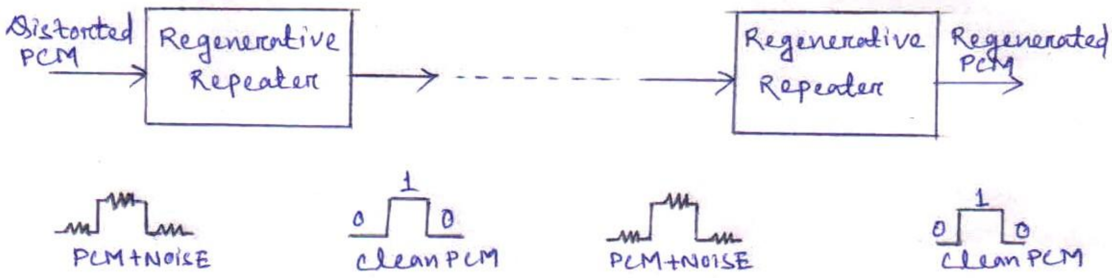



**Fig. 5.19PCM transmission path**

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

## DEMODULATION OF PCM-

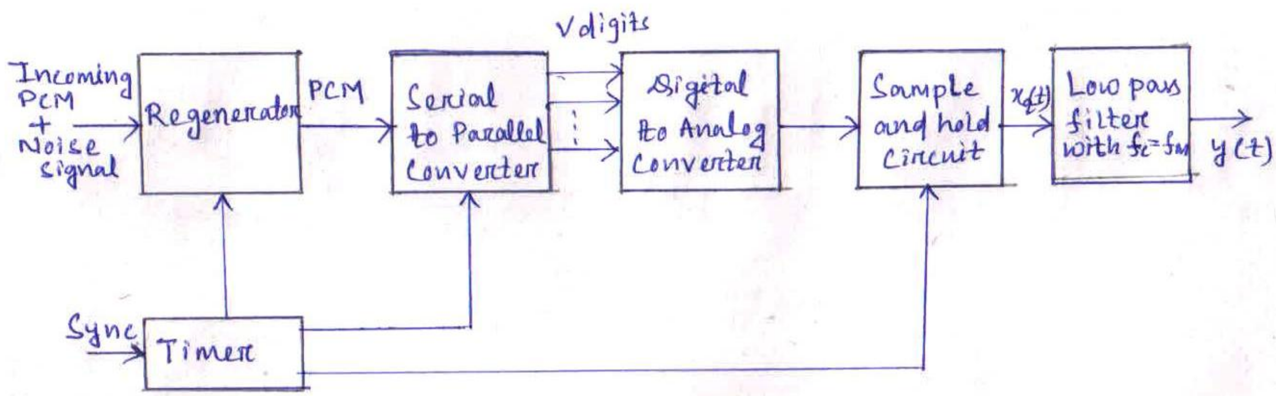



**Fig. 5.20 PCM Receiver**

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

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

## COMPANDING IN PCM-

When the steps are uniform in size, the small amplitude signals would have a poorer signal to quantization noise ratio than the large amplitude signals, since in both the cases the denominator is the same whereas the numerator order is quite small for small amplitude signals and large for large amplitude signals. Since we have to use a fixed number of quantization levels, the only way to have a uniform signal to quantization noise ratio is to adjust the step size in such a manner that the ratio remains constant. This means that the step size must be small for small amplitude signals and large for large amplitude signals.

The effect of an adaptive step size may be achieved in a more feasible way by distorting the signal before the quantization process. An inverse distortion has to be introduced at the receiver to make the overall transmission distortionless.

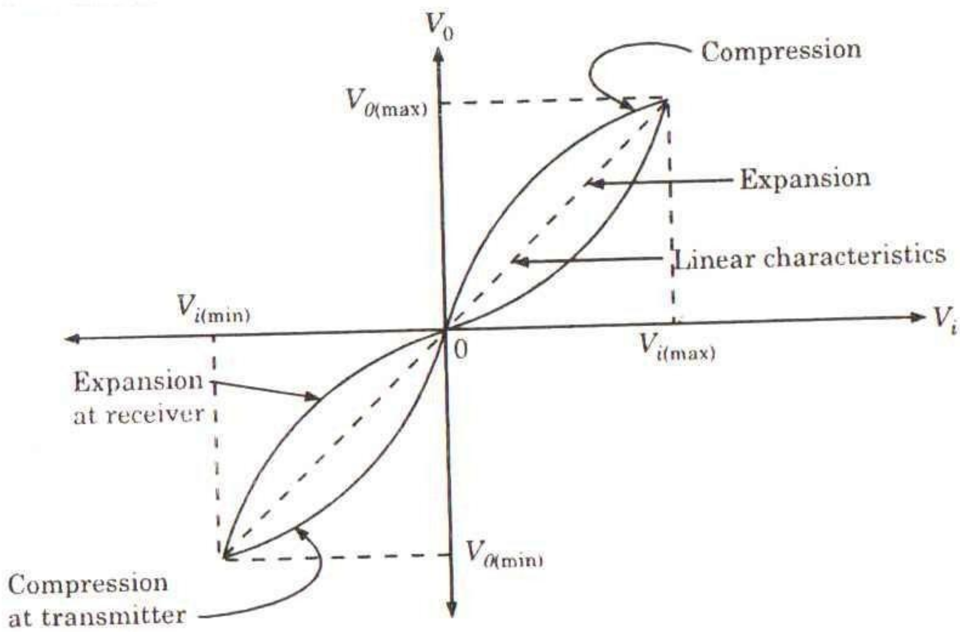



**Fig. 5.21Companding in PCM**

Therefore, the signal amplified at low signal levels and attenuated at high signal levels. After this process, uniform quantization is used. This is equivalent to more step size at low signal levels and small step size at high signal levels. At the receiver a reverse process is done. This means that the signal is attenuated at low signal levels and amplified at high levels to get original signal. Thus the compression of signal at transmitter and expansion at receiver is combinely known as companding**.**

## VOCODER

The source coders employed are called VOCODERS (voice coders) and they operate at a significantly lower bit rate than even ADM. VOCODER bit rates are the range 1.2 to 2.4 kb/s. The resulting reproduced voice has a synthetic – sounding and a somewhat artificial quality. As a result VOCODERS are employed for special application where it acceptable to trade speech quality for the advantage of low bit rate. To transmit speech we need not transmit the pre size waveform generated by the speaker. Rather we can transmit information from which a waveform can be reconstructed at the receiver which is only similar to, rather than identical to, the wave form generated by the speaker.

Application are found in military communication, operated recorded messages, etc.

## T-carrier system-

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

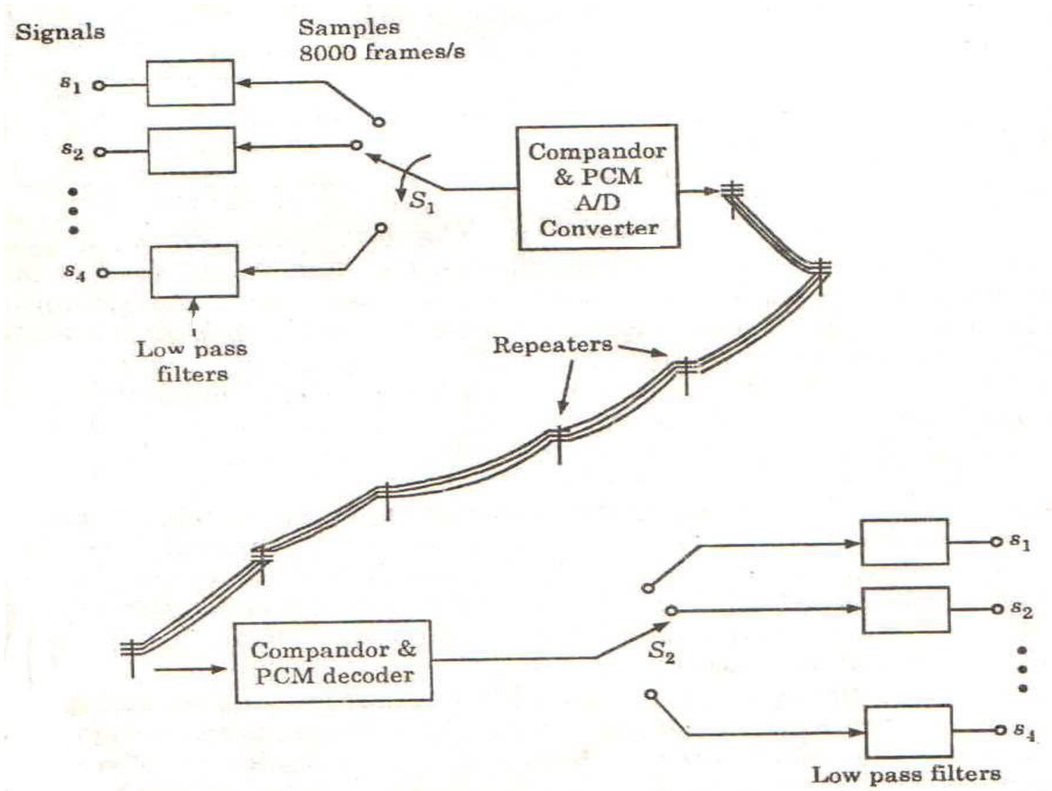



**Fig 5.22 A $T_1$ carrier system**

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

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

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

# TIME DIVISION MULTIPLEXING-

In case of Time Division Multiplexing (TDM), the complete channel bandwidth is allotted to one user for a fixed time slot. As an example, if there are ten users, then every user can be given the time slot of one second. Thus, complete channel can be used by each user for one second time in every ten seconds. This technique is suitable for digital signals.

## Operation-

In Time Division Multiplexed system, different time intervals rather than frequencies are allotted to different signals. During these intervals, these signals are sampled and transmitted. Thus, this system transmits information intermittently rather than continuously. Continuously varying analogue signals have to be sampled at proper intervals for transmission and the receiver must recognize these samples for TDM system to properly.

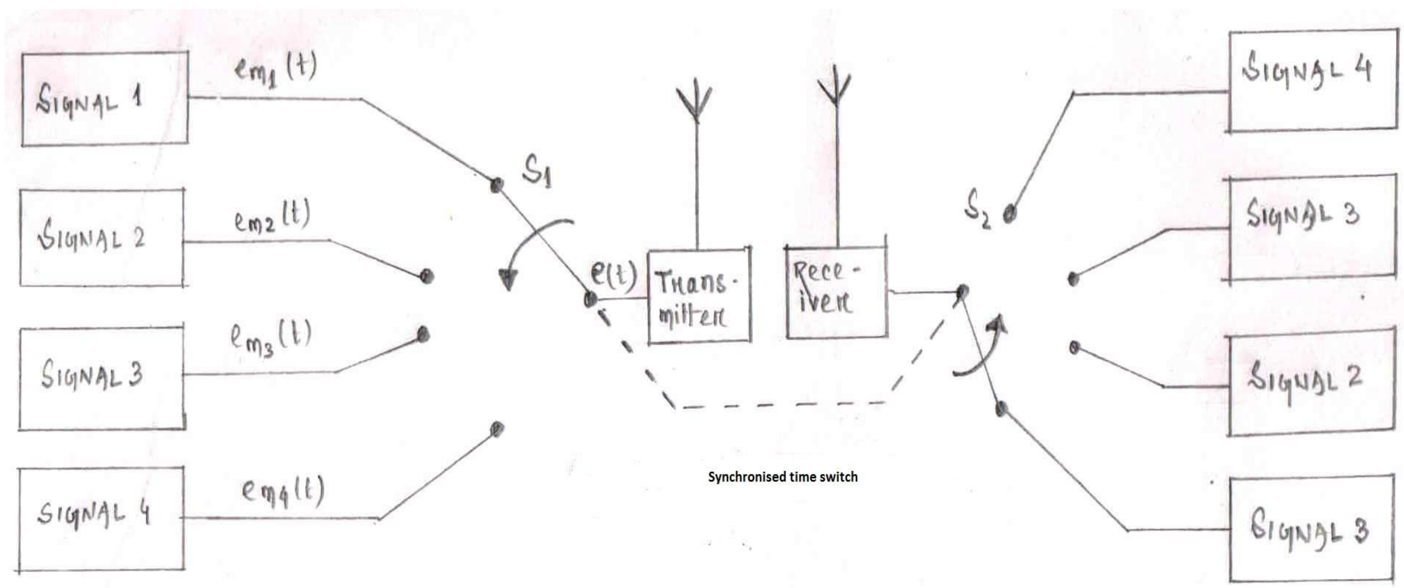



**Fig. 5.23 Time Division Multiplexing**

Each signal source is switched in for a fixed time interval by a time switch $S_1$. During this time, the connected signal modulates the carrier of the transmitter. The switch then moves to the next position connecting the second signal to the transmitter. The process is repeated by the time switch which must rotate continuously at a uniform speed for proper operation of the system. The time for which a signal is connected to the transmitter and the time gap between the instants when the first signal is disconnected and second is connected to the transmitter is very important. The time for which a signal remains connected to the transmitter and the frequency at which the switch rotates are important and related to the highest frequency in the signal. The relationship between them is governed by the sampling theorem.

## DELTA MODULATIONGENERATION : -

Signaling rate and transmission channel bandwidth are quite large in PCM. To overcome this problem, Delta modulation is used.

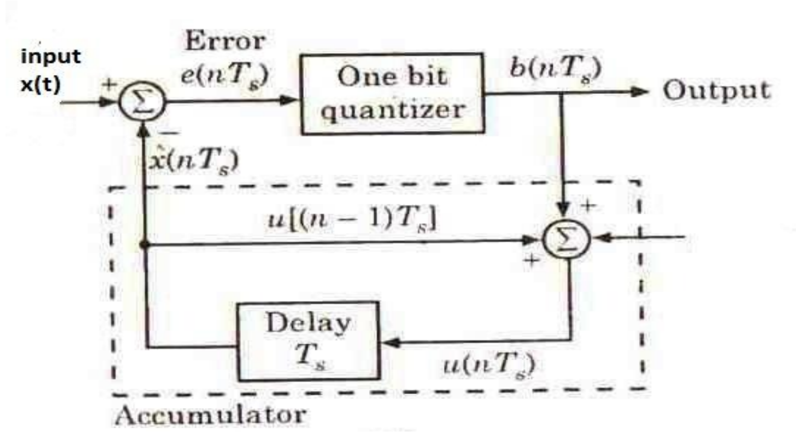



**Fig.5.24 Delta Modulation Transmitter**

Delta modulation transmit s only one bit per sample. Here the present sample value is compared with the previous sample value this result whether the amplitude is increased or decreased is transmitted input signal x(t) is approximated to step signal by the delta modulation . This step size is kept fixed .The difference between the input signal x(t) and staircase approximated signal is confined to 2level s i.e. , +Δ and – Δ. If the difference is +ve, then approximated signal is reduced by 'Δ'. If the difference is –ve, then approximated signal is reduced by 'Δ'.

When the step is reduced, '0' is transmitted and if the step is increased, '1' is transmitted. Hence, for each sample, only one binary bit is transmitted.

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

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

$e(nT_s)$ = error at present sample

$x(nT_s)$ = sampled signal of x(t)

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

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

Then $u[(n-1)T_s] = \hat{x}(nT_s)$ ( last sample approximation of the staircase waveform )

Depending on the sign of error $e(nT_s)$, the quantizer generates an $d_p$ of +Δ and – Δ. Sign of step size Δ is decided. In other words , we can write

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

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

Here $T_s$= sampling interval

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

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

$u(nT_s) = u((n-1)T_s - T_s) + b(nT_s)$

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

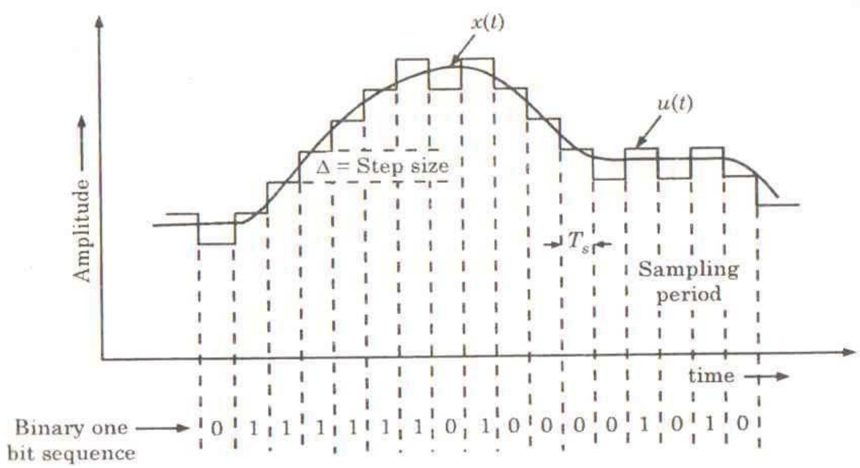



**Fig5.25Delta Modulation Waveform**

## Delta Modulation Receiver:

At the receiver end, the accumulator and LPF are used. The accumulator generates the staircase approximated signal output and is delayed by one sampling period $T_s$. It is then added to the input signal. If input is binary '1' then it adds $+\Delta$ step to the previous output. If input is binary '0' then one step to '$\Delta$' is subtracted from the delayed signal. Also the LPF has the cutoff frequency equal to highest frequency in x(t). This LPF smoothens the staircase signal to reconstruct original message signal x(t).

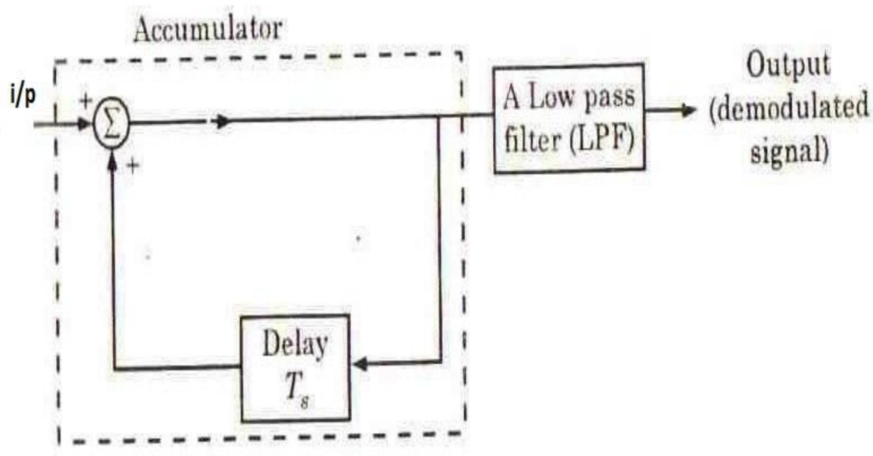



**Fig5.26 Delta Modulation receiver**

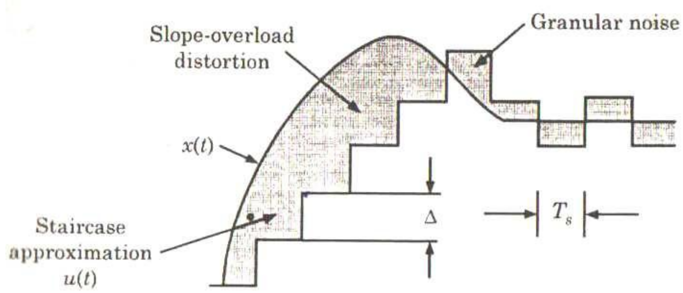



**Fig. 5.27 Quantization error in delta modulation**

**Advantages**

1. Since the Delta modulation transmits only one bit for one sample, therefore the signaling rate and transmission channel bandwidth is quite small for delta modulation compared to PCM.
2. The Transmitter & Receiver implementation is very much simple for delta modulation.

**Disadvantages**

1. Slope overload distortion.
2. Granular or Idle noise

## ADAPTIVE DELTA MODULATIONGENERATION :

To overcome the quantization errors due to slope overload and granular noise, the step size is made adaptive to variations in the input signal x(t). If the input signal is varying fast, the step size is increased & if the input signal is varying slow the step signal is reduced. This method is known as Adaptive Delta modulation.

In the transmitter, the logic for step size control is added in the diagram. The step size increases or decreases according to a specified rule depending on one bit quantizer output. As an example if one bit quantizer output is high, then step size may be doubled for next sample. If one bit quantizer output is next sample, then step size may be reduced by one step.

In the receiver of adaptive delta modulation, there are two portions. The first portion produces the step size from each incoming bit. Exactly the same process is followed as that in transmitter. The previous input and present input decides the step sizes. It is then applied to an accumulator which builds up staircase waveform. The Low Pass Filter then smoothens out the staircase waveform to reconstruct the original signal.

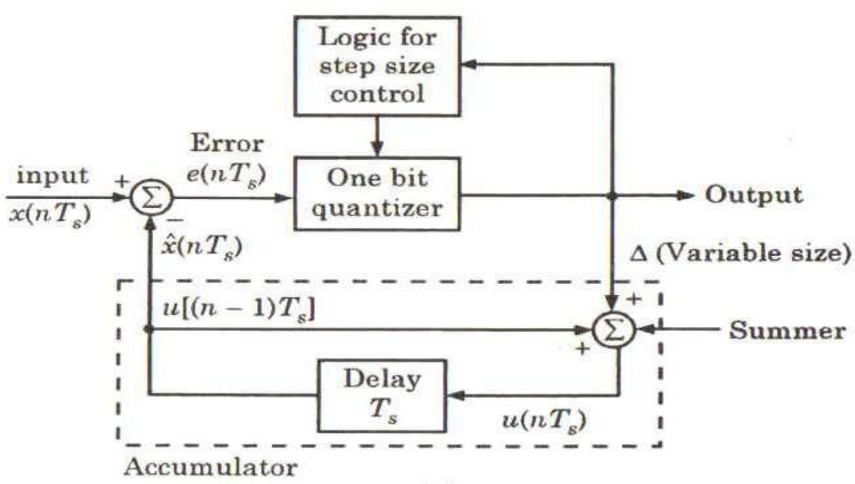



**Fig. 5.28 Adaptive Delta Modulation Transmitter**

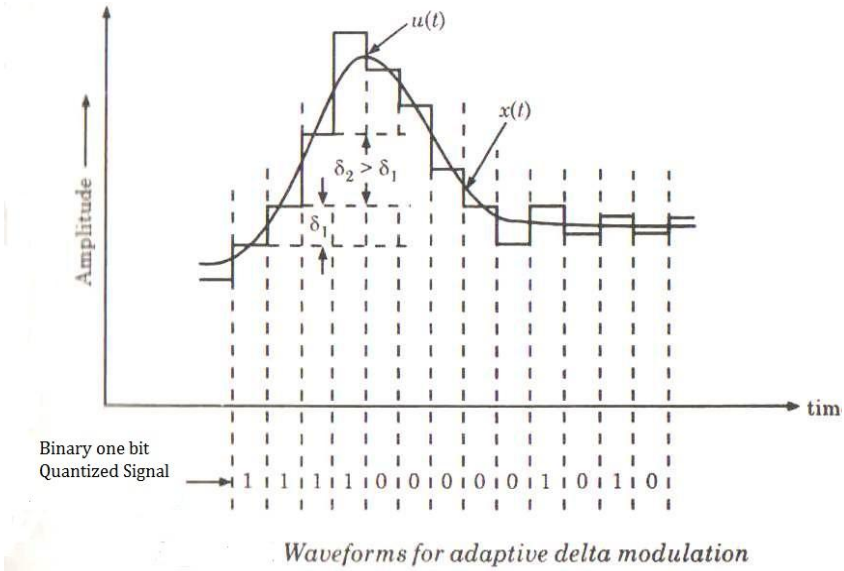



**Fig. 5.29**

## ADAPTIVE DELTA MODULATION RECEIVER

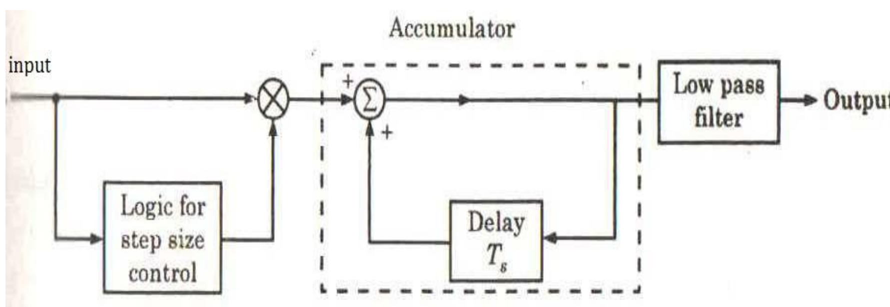



**Fig. 5.30Adaptive Delta Modulation Receiver**

In the receiver of adaptive delta modulator, there are two portions. The first portion produces the step size from each incoming bit. Exactly the same process is followed as that in transmitter. The previous input and present input decides the step size. It is then applied to an accumulator which builds up staircase waveform. The low pass filter then smoothens out the staircase waveform to reconstruct the original signal.

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# DIGITAL MODULATION TECHNIQUES

## DIGITAL COMMUNICATION-

Digital communications become important with the expansion of the use of computers and data processing, and have continued to develop into major industry providing the interconnection of computer peripherals and transmission of data between distant sites. Data communication depends on digital electronics. So this is also known as digital communication.

## CLASSIFY DIGITAL MODULATION TECHNIQUES –

Digital modulation techniques can be coherent or non - coherent.

### COHERENT DIGITAL MODULATION TECHNIQUES-

Coherent digital modulation techniques are those techniques which employ coherent detection. In coherent detection, the local carrier generated at the receiver is phase locked with the carrier at the transmitter. Thus, the detection is done by correlating received noisy signal and locally generated carrier. The coherent detection is a synchronous detection.

### NON COHERENT DIGITAL MODULATION TECHNIQUES-

Non coherent digital modulation techniques are those techniques in which the detection process does not need receiver carrier to be phase locked with transmitter carrier.

The advantage of such type of system is that the system becomes simple. But the drawback of such a system is that the error probability increases.

### ADVANTAGES OF DIGITAL COMMUNICATION SYSTEM-

(1)The digital communication systems are simpler and cheaper.

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

(3)Using data encryption, only permitted receivers may be allowed to detect the transmitted data.

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

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

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

## GENERATION AND DETECTION OF ASK –

### AMPLITUDE SHIFT KEYING (ASK):-

Amplitude shift keying is the simplest digital modulation technique. In this method, there is only one unit energy carrier and it is switched on or off depending upon the input binary sequence. The ASK waveform may be represented as

$S(t) = \sqrt{(2P_s)}\mathrm{Cos}(2\pi f_c t)$ (to transmitted to 1)

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

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

## GENERATION OF ASK SIGNAL:-

ASK signal may be generated by simply applying the incoming binary data and the sinusoidal carrier to the two inputs of a product modulator. The resulting output will be the ASK waveform.

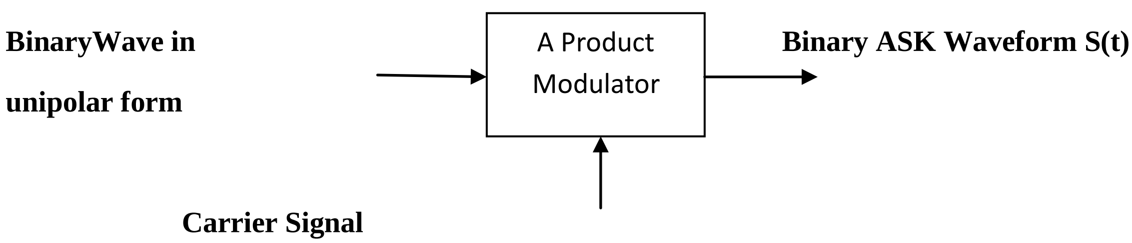



**Fig6.1 Generation of Binary ASK Signal**

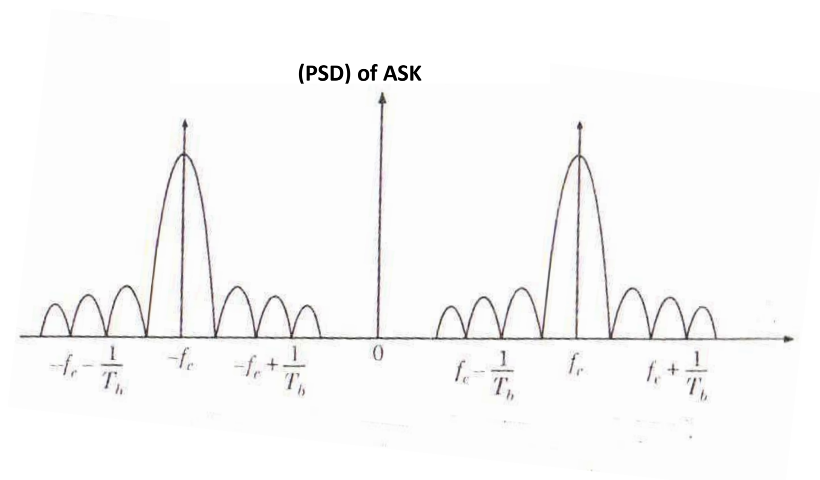



**Fig 6.2 Power Spectral Density of ASK Signal**

## DEMODULATION OF ASK:-

The demodulation of binary ASK waveform can be achieved with the help of coherent detector. It consists of a product modulator which is followed by an integrator and a decision making device. The incoming ASK signal is applied to one input of the product modulator. The other input of the product modulator is supplied with a sinusoidal carrier which is generated with the help of a local oscillator. The output of the product modulator goes to input of the integrator. The integrator operates on the output of the product modulator for successive bit intervals and essentially performs a low pass filtering action. The output of the integrator goes to the input of a decision making device.

Then the decision making device compares the output of the integrator with a preset threshold. It makes a decision in favors of symbol ‘1’ when the threshold is exceeded and in favors of symbol ‘0’ otherwise.

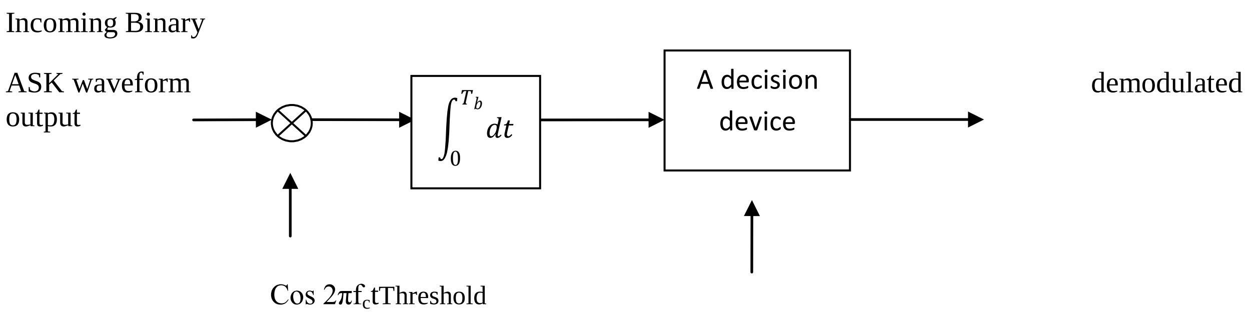



**Fig. 6.3 Coherent detection of binary ASK signal**

## GENERATION AND DETECTION OF BPSK-

### BINARY PHASE SHIFT KEYING (BPSK):-

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

$S(t)= A \cos(2\pi f_c t)$

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

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

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

For symbol ‘1’

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

If next symbol is ‘0’, then we have,

for symbol ’0’

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

Because $\cos(\theta+\pi)= - \cos \theta$, therefore the above equation can be written as,

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

We can define BPSK signal combinely as,

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

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

-1 when binary ‘0’ is to be transmitted

## GENERATION OF BPSK SIGNAL :-

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

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

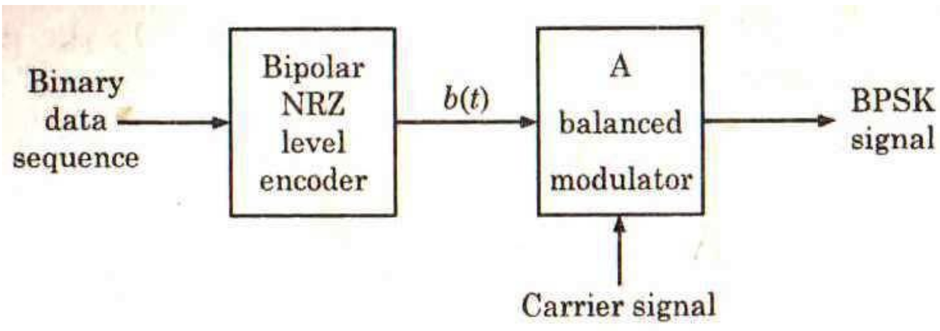



**Fig.6.4 Generation of BPSK**

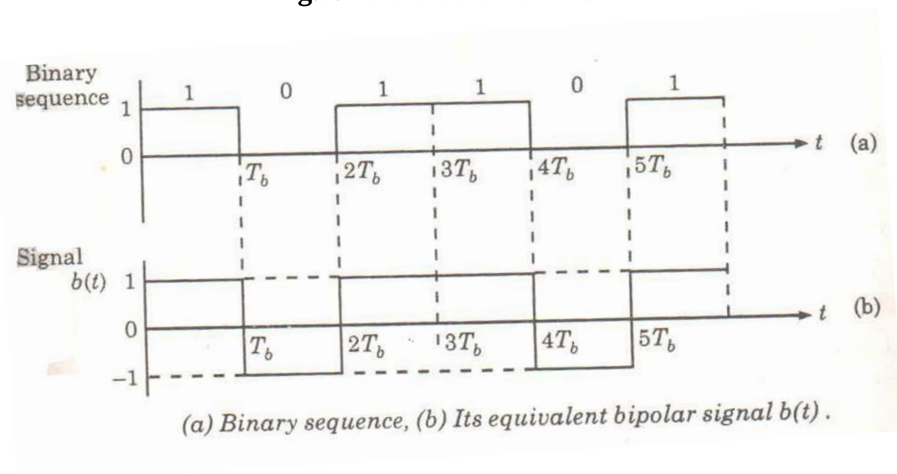



**Fig. 6.5**

## DEMODULATION OFBPSK :-

The transmitted BPSK signal is given as

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

This signal undergoes the phase change depending upon the time delay from transmitter end to receiver end. This phase change is usually a fixed phase shift in the transmitted signal.

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

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

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

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

Again, we know that

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

Therefore, we have

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

$$= ½ + ½(\mathrm{Cos}\ 2(2\pi f_c t + \theta))$$

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

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

Then this signal is passed through a frequency divider by two. Thus, at the output of frequency divider we get a carrier signal whose frequency is $f_c$ i.e,

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

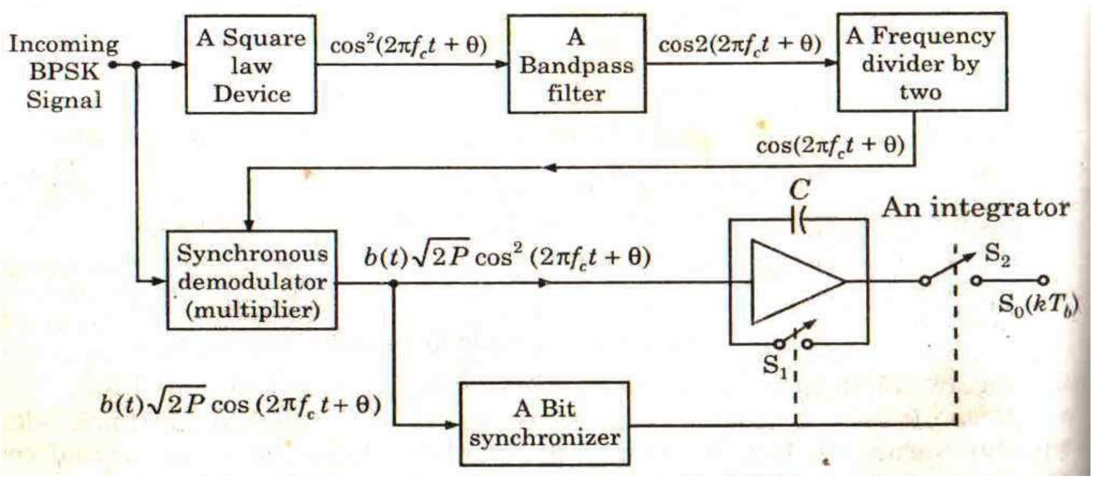



**Fig.6.6 Reception of Baseband Signal in BPSK**

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

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

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

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

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

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

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

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

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

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

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

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

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

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

## BANDWIDTH

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

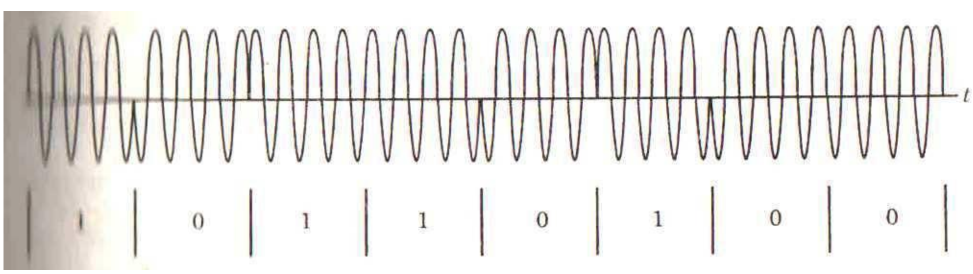



**Fig.6.7 Waveform generation Baseband Signal in BPSK**

# GENERATION AND DETECTION OF BFSK-

## BINARY FREQUENCY SHIFT KEYING (BFSK)

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

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

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

the equation combinely written as

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

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

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

Therefore, we have

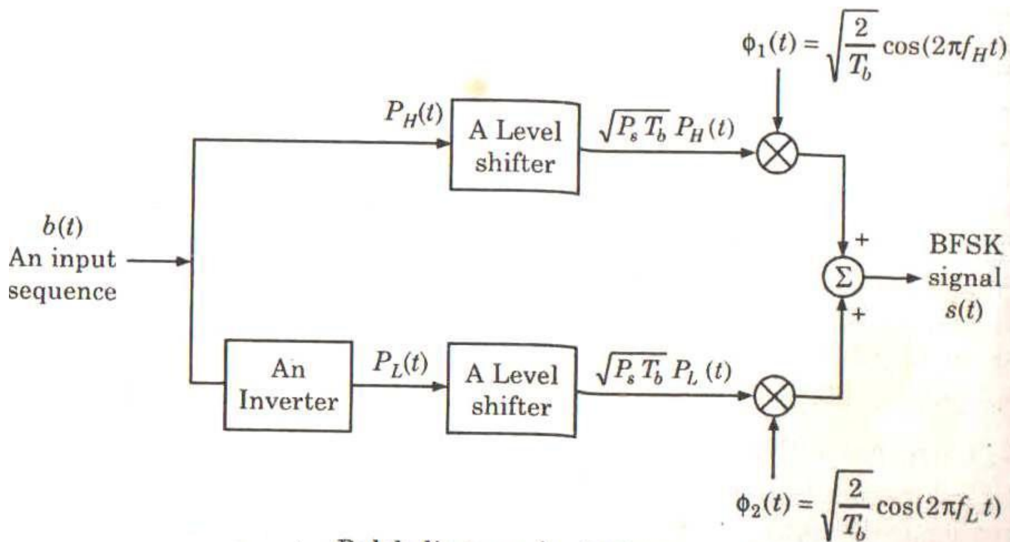
$f_H = f_c + \Omega/2\pi$ for symbol '1'

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

## GENERATION OF BFSK:-

The input sequence b(t) is same as $P_H(t)$. An inverter is added after b(t) to get $P_L(t)$. The level shifter $P_H(t)$ and $P_L(t)$ are unipolar signals. The level shifter converts the '+1' level to $\sqrt{(P_sT_b)}$ . Zero level is unaffected. Thus the output of the level shifters will be either $\sqrt{(P_sT_b)}$ (if input is '+1') or zero (if input is zero). Further, there are product modulators after level shifter. The two carrier signals $\Phi_1(t)$ & $\Phi_2(t)$ are used. $\Phi_1(t)$ & $\Phi_2(t)$ are orthogonal to each other. The carrier signal multiplied with the output of the level shifter in product modulator.

The adder then adds the two signals from product modulator.




*Bolck diagram for BFSK generation.*

**Fig.6.8**

## DETECTION OF BFSK-

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

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

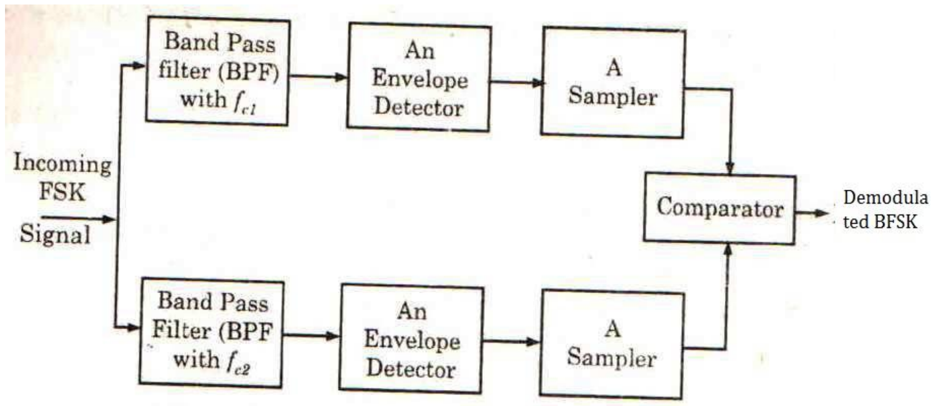



**Fig . 6.9 Demodulation of BFSK Signal**

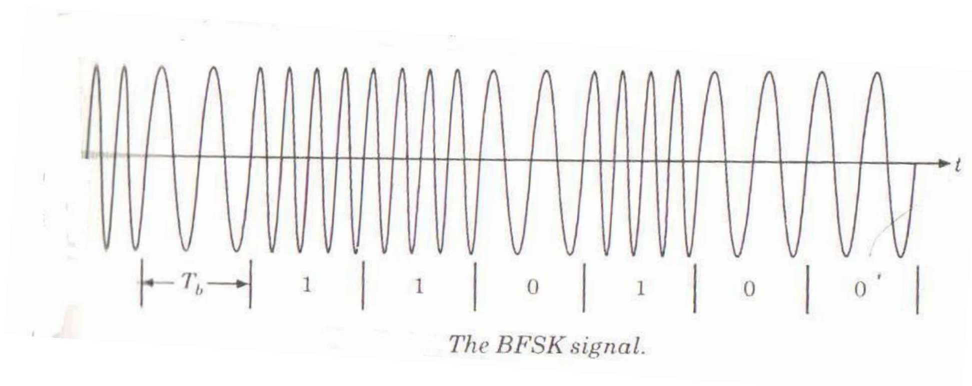



**Fig.6.10 Waveform of BFSK Signal**

## GENERATION AND DETECTION OF DPSK-

### Differential phase shift keying (DPSK)

The differential phase shift keying is the non-coherent version of the PSK. DPSK does not need a synchronous carrier at the demodulator.

## Generation of DPSK

The digital information content of the binary data is encoded in terms of signal transitions. As an example, the symbol '0' may be used to represent transition in a given binary sequence and symbol '1' indicate no transition. This new signaling technique that combines differential encoding with phase shift keying is known as differential phase shift keying.

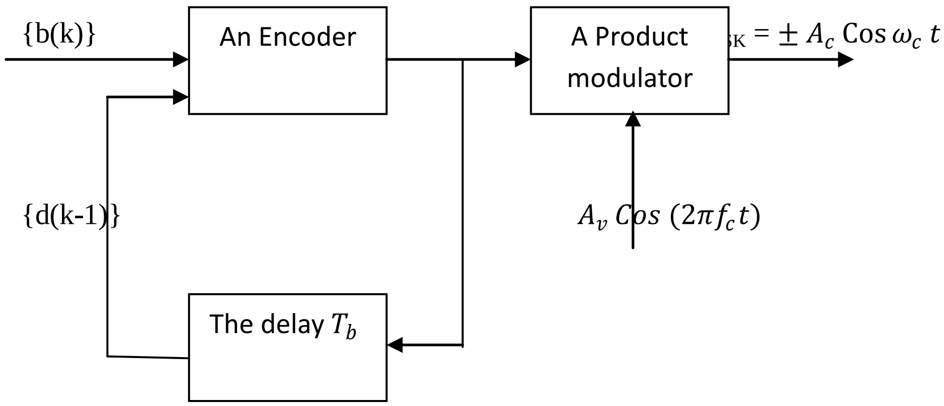



**Fig. 6.11 Illustration of The Scheme for the generation of DPSK SIgnal**

The data stream b(t) is applied to input of the encoder. To another input of the encoder delayed version of the encoder output is applied. The output of the encoder is applied to one input of the product modulator. To the other input of this product modulator a sinusoidal carrier of fixed amplitude and frequency is applied.

## <u>Detection DPSK</u>-

The received DPSK signal is applied to one input of the multiplier. To the other of the multiplier, a delayed version of the received DPSK signal by the time interval Tb is applied. The output of the difference is proportional to Cos(ϕ), where ϕ is the difference between the carrier phase angle of the received DPSK signal and its delayed version , measured in the same bit interval. The phase difference between the two sequences for each bit interval is used to determine the sign of the phase comparator output. When ϕ=0, the integrator output is positive whereas when ϕ=π, the integrator output is negative. By comparing the integrator output with a decision level of zero volt, the decision device can reconstruct the binary sequence by assigning a symbol 0 for negative output and symbol 1 for positive output. In the absence of noise the receiver can reconstruct the transmitted binary data exactly.

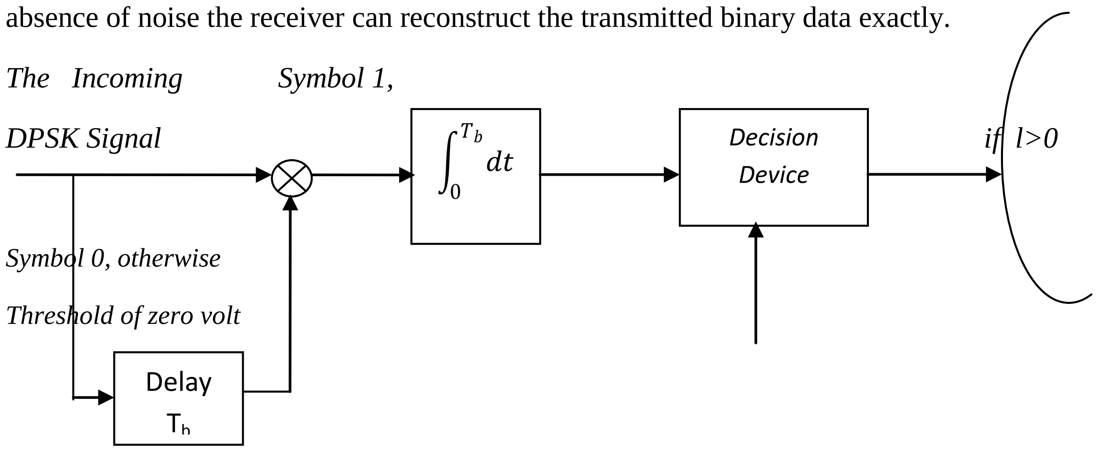



**Fig.6.12** ***Receiver for the detection of DPSK signals***

**Advantages**:-

DPSK does not need carrier at the receiver end. This means that the complicated circuitry for generation of local carrier is not required. The bandwidth requirement of DPSK is reduced as compared to that of BPSK.

## QPSK-

In communication systems, we have two main resources. These are the transmission power and the channel bandwidth. The channel bandwidth depends upon the bit rate or signaling rate $f_b$. If two or more bits are combined in some symbols, then the signaling rate will be reduced. This reduces the transmission channel bandwidth.

In quadrature phase shift keying, two successive bits in the data sequence are grouped together. This reduces the bits rate or signaling rate and thus reduces the bandwidth of the channel.

In case of BPSK, when symbol changes the level, the phase of the carrier is changed by 180. Because, there were only two symbols in BPSK, the phase shift occurs in two levels only. However, in QPSK, two successive bits are combined. This combination of two bits forms four distinct symbols. When the symbol is changed to next symbol, then phase of the carrier is changed by 45.

## GENERATION OF QPSK TRANSMITTER-

The toggle flip-flop is driven by a clock waveform whose period is the bit time $T_b$. The toggle flip-flop generates an odd clock waveform and an even waveform. The active edge of one of the clock and the active edge of the other are separated by the bit time $T_b$. The bit stream b(t) is applied as the data input to both type-D flip-flops, one driven by the odd and one driven by the even clock waveform.

The output bit stream $b_e(t)$ is superimposed on a carrier $\sqrt{P_s}$ Cos $\omega_o t$ and the bit stream $b_o(t)$ is superimposed on a carrier $\sqrt{P_s}$ Sin $\omega_o t$ by the use of two multipliers to generate two signals $S_e(t)$ and $S_o(t)$. These signals are then added to generate the transmitted output signal S(t) which is

$S(t) = \sqrt{P_s}\, b_o(t) \text{ Sin } \omega_o t + \sqrt{P_s}\, b_e(t) \text{ Cos } \omega_o t$

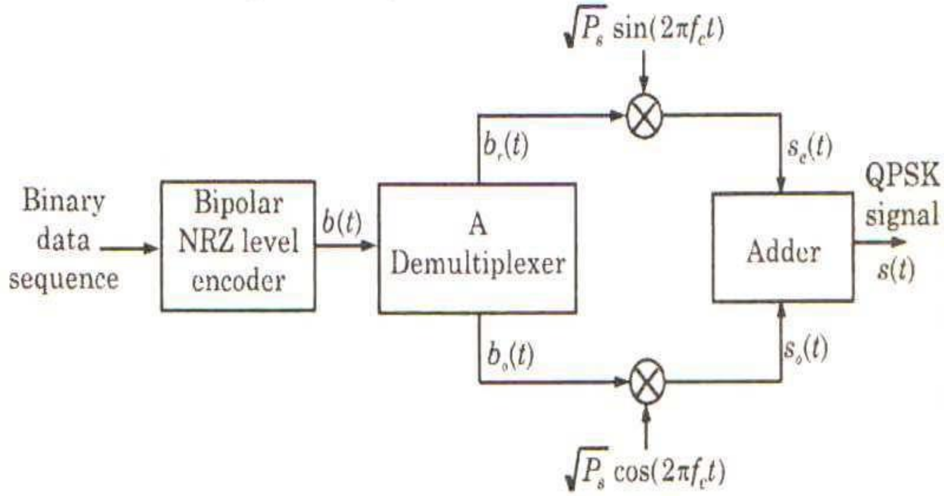



**Fig. 6.13 Generation of QPSK Signal**

## DETECTION OF QPSK-

The incoming signal be raised to the fourth power after which filtering recovers a waveform at four times the carrier frequency and finally frequency division by four regenerates the carrier.

The incoming signal is also applied to two synchronous demodulators consisting of a multiplier followed by an integrator. The integrator integrates over a two bit interval of duration $T_s=2T_b$. One demodulator uses the carrier Cos $\omega_o t$ and the other one uses the carrier Sin $\omega_o t$ .The integrator output is sampled. Samples are taken alternatively from one and other integrator output at the end of each bit time $T_b$ and these samples are held in the latch for the bit time $T_b$. Each individual integrator output is sampled at intervals $2T_b$. The latch output is the recovered bit stream b(t).

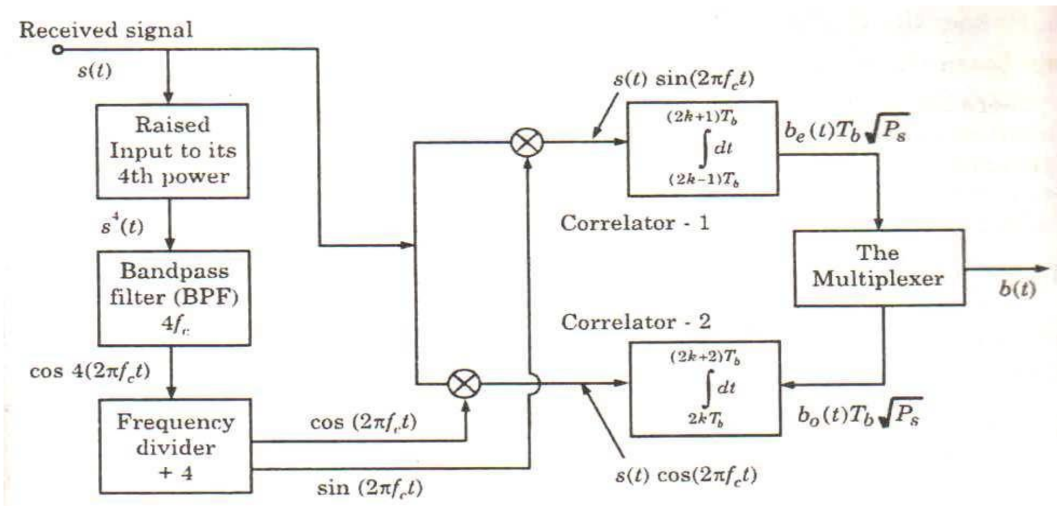



**Fig.6.13 Reception of QPSK Signal**

**Advantages:**

I. For the same bit error rate, the bandwidth required by QPSK is reduced to half of compared to BPSK.
II. Because of reduced bandwidth the information transmission rate of QPSK is higher.
III. Variation in QPSK amplitude is not much. Hence carrier power almost remain constant.

## GENERATION OF MSK:

Two input sinusoidal waves one of frequency $f_c = n_c/\,4T_b$ for some fixed integer $n_c$ and the other of frequency $1/\,4T_b$ are first applied to the modulator. This produces two phase coherent sinusoidal waves at frequencies $f_1$ and $f_2$ which are related to the carrier frequency $f_c$ and bitrate $R_b$ by

$$f_1 = f_c + \frac{h}{2T_b} \quad or \quad f_c + \frac{h}{2R_b}$$

$f_2 = f_c - \frac{h}{2T_b} \quad or \quad f_c - \frac{h}{2R_b}$ for $h = \frac{I}{2}$

These two sinusoidal waves are separated from each other by two narrow band filters one centred at $f_1$ and the other at $f_2$. The resulting filter output are next linearly combined to produce the pair of basic functions $\varphi_1(t)$ and $\varphi_2(t)$ . Finally $\varphi_1(t)$ and $\varphi_2(t)$ are multiplied with two binary waves $a_1(t)$ and $a_2(t)$ both of which have a bit rate equal to $1/\ 2T_b$ . These two binary waves are extracted from the incoming binary sequence.

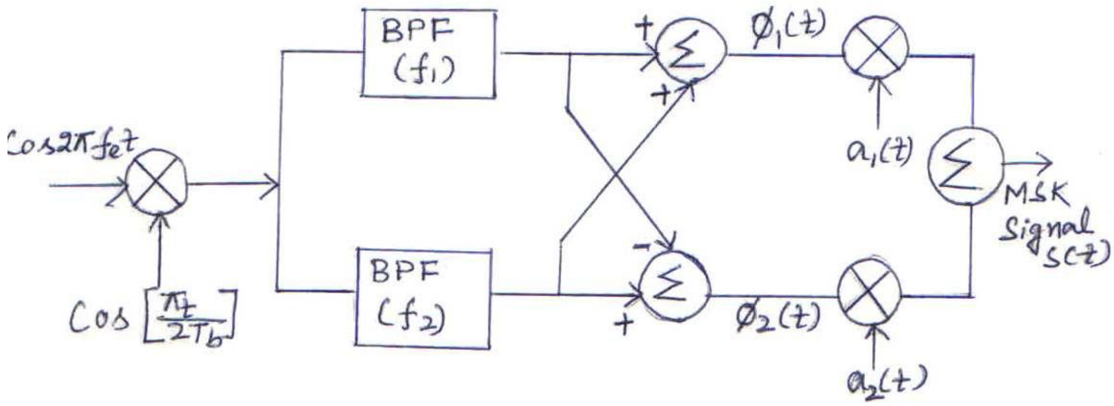



**Fig. 6.14 Transmitter of MSK Signal**

## DETECTION OF MSK :

The received signal $x(t)$ is correlated with locally generated replicas of the coherent reference signals $\varphi_1(t)$ and $\varphi_2(t)$ . The integration in the Q- channel is delayed by $T_b$ seconds with respect to the I-Channel.

The resulting in-phase and quadrature channel correlator outputs $x_1$ and $x_2$ are compared with a threshold of zero. To estimate the phase $\theta(0)$ and $\theta(T_b)$. Finally these phase decision s are interleaved so as to reconstruct the original input binary sequence with minimum average probability of symbol error in an AGWN channel.

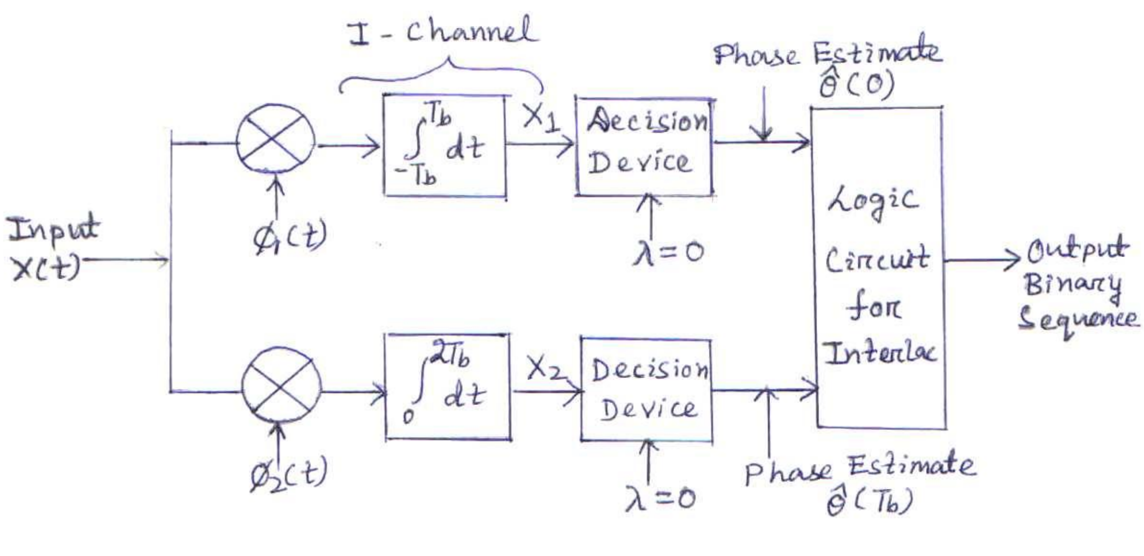



**Fig. 6.15 MSK Receiver**

## QAM TRANSMITTER:

The following diagram shows the ideal structure of a QAM transmitter, with a carrier frequency $f_0$and the frequency response of the transmitter's filter $H_t$.

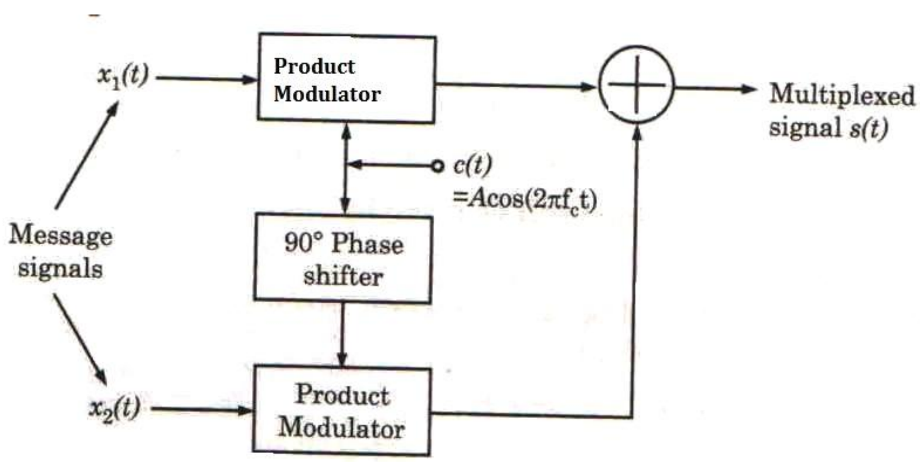



**Figure 6.16 Illustration of QAM system Transmitter**

First the flow of bits to be transmitted is split into two equal parts: this process generates two independent signals to be transmitted. They are encoded separately just like they were in an amplitude shift keying modulator. Then one channel (the one "in phase") is multiplied by a cosine, while the other ("in quadrature") is multiplied by a sine. This way there is a phase of $90^0$ between them. They are simply added one to the other and sent through the real channel.

The sent signal can be expressed in the form:

$$s(t) = \sum_{n=-\infty}^{\infty} [V_c[n].h_t\,(t - nT_c)\cos(2\pi f_0 t) - V_s[n].h_t(t - nT_s)\sin(2\pi f_0 t)$$

Where $V_c[n]$ and $V_s[n]$ are the voltages applied in response to the nth symbol to the cosine and sine waves respectively.

## QAM RECEIVER:

The receiver performs the inverse process of the transmitter. It's ideal structure is shown in the figure below with $H_r$the receiver filter's frequency response:

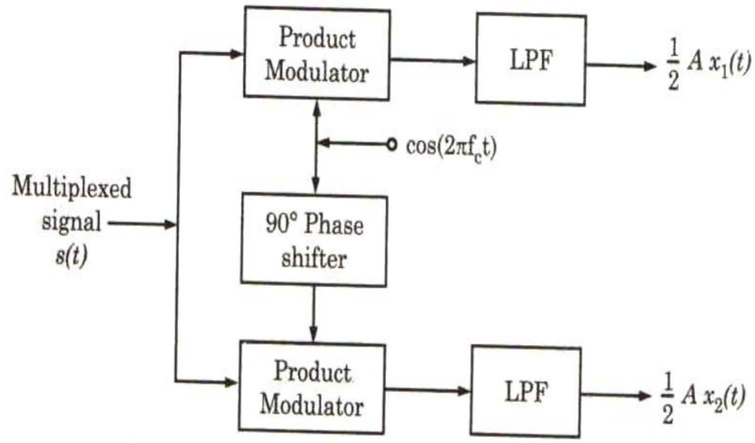



**Fig6.17 QAM System Receiver**

Multiplying by a low-pass-filter it is possible to extract the component in phase (or in quadrature). Then there is only an ASK demodulator and the two flows of data are merged back.

There is an unknown phase delay between the transmitter and receiver that must be compensated by synchronization of the receiver's local oscillator, i.e. the sine and cosine functions in the above fig. . In mobile application there will often be an offset in the relative frequency as well, due to the possible presence of a Doppler shift proportional to the relative velocity of the transmitter and receiver. Both the phase and frequency variations introduced by the channel must be compensated by properly tuning the sine and cosine components, which requires a phase reference, and is typically accomplished using a phase locked loop.

In any application, the low pass filter in the receiver $H_r$ filter will be implemented as a single combine filter. Here they are shown as separate just to be clearer.

## GENERATION AND DETECTION OF GMSK-

## GAUSSIAN MINIMUM SHIFT KEYING(GMSK)

Gaussian minimum shift keying is a continuous phase frequency shift keying modulation scheme. It is similar to standard minimum shift keying, however the digital data stream is first shaped with a Gaussian filter before being applied to frequency modulator. This has the advantage of reducing sideband power, which in turn reduces out of band interference between signal carries in adjacent frequency channels.

## GENERATION OF GMSK-

The GMSK signal is generated by filtering the input binary NRZ symbols with a Gaussian filter with BT=O.5 and frequency modulating the filter output onto a sinusoidal carrier signal with a modulation index of 0.5. *B* is the one sided3 *dB* bandwidth of the filter, *T* is the data symbol period. The Gaussian filter pulse response is truncated to time duration of 2T about its peak.

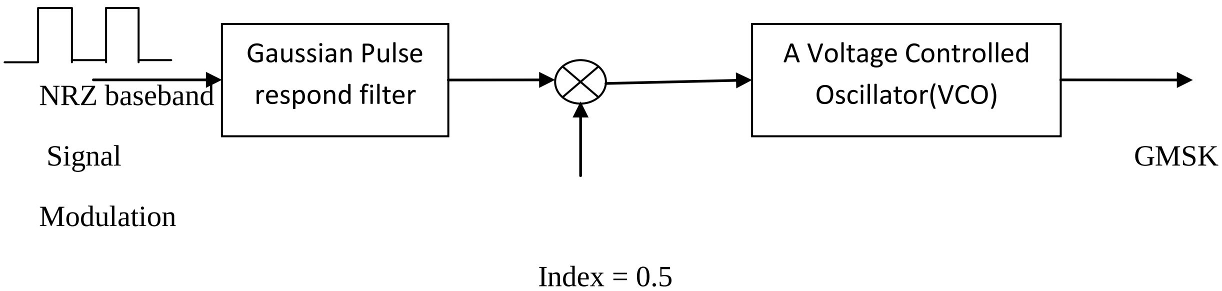



**Fig.6.18 GMSK Generation**

## DETECTION OF GMSK-

The receiver considered here is basically a serial MSK type of receiver. The received rf signal is first down-converted to base-band with a phase error 0, it is then filtered by the filter $JZ_o(-t)$ and sampled at symbol rate with a timing error r. From the filter output sample sequence, the data decision variable is formed by taking the in-phase component at odd data symbol times and the quadrature component at even symbol times. Data decisions are made simply by taking the sign of the data decision variables.

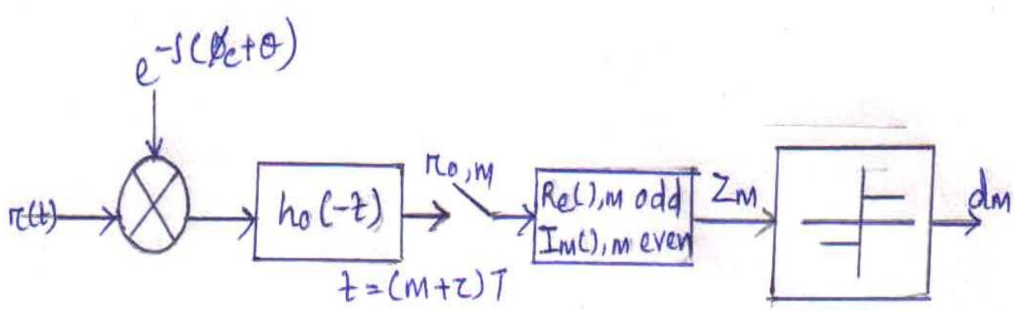



**Fig 6.69 GMSK Receiver**

## SPREAD SPECTRUM

### Spread Spectrum:

The term 'spread spectrum' describes a modulation technique that makes the sacrifice of bandwidth in order to gain signal-to-noise (S/N) performance. Basically, in a spread-spectrum system, the transmitted signal is spread over a frequency much wider than the minimum bandwidth required to send the signal.

## APPLICATION OF SPREAD SPECTRUM:

Typical applications for spread-spectrum radio are:
1. Cellular/PCS base station interconnect
2. Last-mile obstacle avoidance

3. Private networks
4. Railroads and transportation
5. Utilities like electricity, oil, gas and water
6. Banks, hospitals, universities and corporations
7. Disaster recovery and special event PSTN extensions
8. TELCO bypass
9. Rural telephony
10. Video conferencing
11. LAN/WAN/Internet connection

## SPREAD SPECTRUM MODULATION:

Spread Spectrum modulation is of 2 type,

1. Direct sequence Spread Spectrum
2. Frequency Hopped Spread Spectrum

## DIRECT SEQUENCE SPREAD SPECTRUM:

Direct-sequence systems, the encoding signal is used to modulate a carrier, usually by phase shift keying (e.g., biphase or quadriphase) at the code rate.

To generate a spread spectrum signal one requires:

1. A modulated signal somewhere in the RF spectrum
2. A PN sequence to spread it

These two are combined as shown in Figure 1.

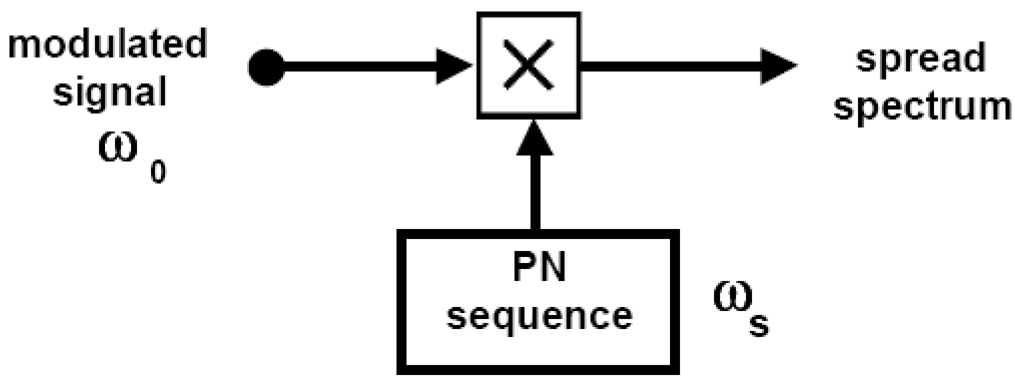



**Fig. 6.20 Basis of Direct Spread Spectrum Modulator**

There are two bandwidths involved here: that of the modulated signal, and the spreading sequence. The first will be very much less than the second. The output spread spectrum signal will be spread either side of the original RF carrier ($\omega_0$) by an amount equal to the bandwidth of the PN sequence. Most of the energy of the sequence will lie in the range DC to $\omega_s$, where $\omega_s$ is the sequence clock. The longer the sequence the more spectral components will lie in this range. It is necessary and usual that $\omega_0 >> \omega_s$, although in the experiment to follow the difference will not be large.

The modulated signal can be of any type, but typically digitally-derived, such as binary phase shift keyed - BPSK. In this case the arrangement of Figure 6.20 can be expanded to that of Figure 6.21. A digital message is preferred in an operational spread spectrum system, since it makes the task of the eavesdropper even more difficult.

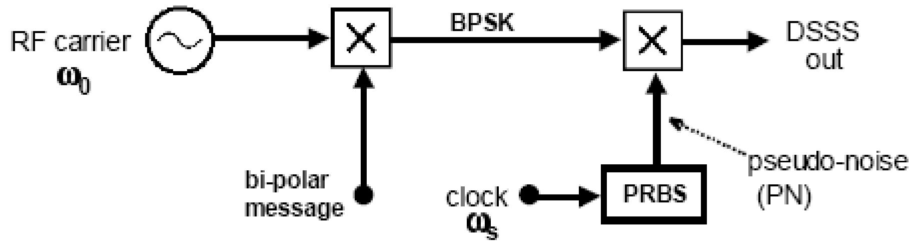



**Fig. 6.21 Direct Spread Spectrum modulator**

The arrangement of Figure 6.21 can be simplified by noting that, if the clock of the bipolar message is a sub-multiple of the clock of the PN sequence, then the modulator sum of the message and the PN sequence can be used to multiply the RF carrier, generating a DSSS signal with a single multiplier. Such a simplification will not be implemented in this experiment.

## A DS-SS DEMODULATOR:

A demodulator for the DSSS of Figure6.20 is shown in block form in Figure 6.22.

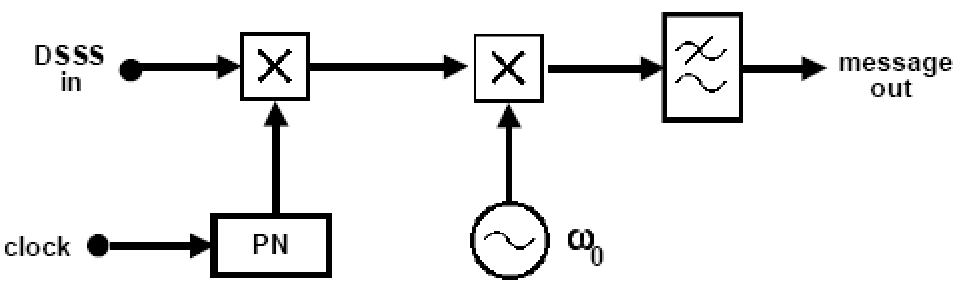



**Fig. 6.22 demodulator for the DSSS**

The input multiplier performs the de-spreading of the received signal, and the second multiplier translates the modulated signal down to baseband. The filter output would probably require further processing - not shown - to ‘clean up’ the waveform to binary format. The PN sequence at the receiver acts as a ‘key’ to the transmission. It must not only have the same clock and bit pattern; it must be *aligned* properly with the sequence at the transmitter

## FREQUENCY HOPPING SPREAD SPECTRUM (FH-SS)

Frequency hopping spread spectrum (FHSS) is a method of transmitting radio signals by shifting carriers across numerous channels with pseudorandom sequence which is already known to the sender and receiver.
Frequency hopping spread spectrum is defined in the 2.4 GHz band and operates in around 79 frequencies ranging from 2.402 GHz to 2.480 GHz. Every frequency is GFSK modulated with channel width of 1MHz and rates defined as 1 Mbps and 2 Mbps respectively.

Frequency hopping spread spectrum is a transmission technology used in wireless networks and a technique to generate spread spectrum by hopping the carrier frequency. FHSS uses narrow band signal which is less than 1 MHz, In this method data signal is modulated with a narrowband carrier signal that "hops" in random and hopping happens in pseudo-random "predictable" sequence in a regular time from frequency to frequency which is synchronized at both ends. Using FHSS technology improves privacy, it is a powerful solution to avoid interference and multi path fading (distortion), it decreases narrowband interference, increases signal capacity, improve the signal to noise ratio, efficiency of bandwidth is high and difficult to intercept also this transmission can share a frequency band with many types of conventional transmissions with minimal interference. For frequency hopping a mechanism must be defined to transmit data in a clear channel and to avoid the congested channels. Frequency hopping is the periodic change of transmission frequency and hopping happens over a frequency bandwidth which consists of numbers of channels. Channel which is used as a hopped channel is instantaneous bandwidth while the hopping spectrum is called total hopping bandwidth.

## SHANNON THEOREM:

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

### CHANNEL CAPACITY "C":

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

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

Where

C is the channel capacity in bits per second (bps), which is the maximum data rate for a theoretical bit error rate (BER)

B is the required bandwidth in Hz

S/N is the signal to noise ratio

[Note: C which represents the amount of information allowed by communication channel, also represent the desired performance. S/N ratio expresses the environmental conditions such as obstacles, presence of jammers, interferences, etc.]

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

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

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

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

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

## MODEM

### Modem:-

Modem is a contraction of the term MOdulator&DEModulator. Both function are included in modem. When used in the transmitting mode, the modem accepts digital data and converts it to analog signals for use in modulating a carrier signal. At the receive end of the system, the carrier is demodulated to recover the data.

There are two types of MODEM,

1. The Hard-wired Modem
2. The acoustically coupled data set.

## THE HARD-WIRED MODEM:

AThe Hard-wired Modem connects directly to the communication ckt.  connects directly to the communication ckt. in semi-permanent way. Such modems may be self –contained devices which connects to terminals and business machine, or they may be incorporated in the business machines.

The one limitation of the Hard-wired Modem is that it precludes mobility since, being hard-wired, the equipment must remain connected to the circuit terminals.

## THE ACOUSTICALLY COUPLED DATA SET:

The acoustically coupled data set modem solves the mobility problem. A standard telephone handset can be placed in the foam cups of an acoustic coupler, and the transmitter and receiver sounds will be conveyed to and from the telephone channel by transmit and receive elements of the acoustic coupler. The modem components of the acoustic coupler form an interface with the business machine.

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