# GOVERNMENT ARTS AND SCIENCE COLLEGE

**NAGERCOIL – 629 004**

**[ Affiliated to Manonmaniam Sundaranar University, Tirunelveli – 12 ]**

# DEPARTMENT OF PHYSICS

# COURSE MATERIAL

**NAME OF THE SUBJECT : COMMUNICATION ELECTRONICS**

**SUBJECT CODE : SEPH5C**

**YEAR : III B.Sc. PHYSICS**

**SEMESTER : V**


**STAFF IN-CHARGE : Dr. JAIN JOHN**
**ASSISTANT PROFESSOR**
**DEPARTMENT OF PHYSICS**
**Government Arts and Science College NAGERCOIL.**
**Email : stanjain@gmail.com**
**Mobile No.:8220769969**

# COMMUNICATION ELECTRONICS

Preamble: This course enable the students to understand various modulation and demodulation techniques used for communication. The paper needs a basic knowledge in electronics and mathematics and the learners are expected to come out with the ability to choose proper modulation techniques .

**UNIT I: AMPLITUDE MODULATION AND TRANSMISSION** Introduction – amplitude Modulation – AM envelop – AM frequency spectrum and bandwidth – Phasor representation of AM with carrier – coefficient of modulation or percentage modulation or modulation index – degrees of modulation – AM power distribution – AM Current relation and efficiency - modulation by complex information signal - doubleside band suppressed carrier AM - single side band suppressed carrier AM – Vestigal side band amplitude modulation – AM modulator circuits – emitter modulations or low power AM – collector modulator or medium and high power AM modulator - AM transmitters – Broadcast AM transmitters – Low level of AM transmitter – High level AM transmitter. (15L)

**UNIT II: AMPLITUDE MODULATION -RECEPTION** Comparison of AM system – Quadrature amplitude modulation – principles of AM detection – AM receivers – receiver parameters – Tuned radio frequency (TRF) receiver or straight receiver – principles of superhetrodyne –double frequency conversion AM receiver. (11L)



**UNIT III:ANGLE MODULATION – TRANSMISSION** Introduction – Frequency modulation – Phase modulation – Phase deviation and modulation index – Multitone modulation – Transmission band width of FM – conversion of PM to FM or frequency modulator – conversion of FM to PM / phase modulators – commercial broadcast

FM – phasor representation of an FM and PM – average power of an AM/FM wave – generation of FM – direct method of FM generation – reactance tube modulator – indirect method of FM wave generation – FM transmitters – indirect method – Comparison of AM and FM. (13L)

**UNIT IV:FM RECEPTION** FM detectors – Balanced slope detector – Foster seely discriminator – ratio detector – F'M super heterodyne receiver – FM noise suppression – threshold extension by FMFB technique. (11L)

**UNIT – V: DIGITAL MODULATION TECHNIQUES** Introduction – BFSK – Binary phase shift keying – Quadrature PSK – Differential PSK – Performance comparison of digital modulation schemes - M ary FSK – correlative coding – Duobinary encoding. (10L)

**Book For Study**

1.Principles Of Communication Engineering-Dr. K.S. Srinivasan, Second Edition : 2010.

**Book For Reference**

1.Electronic communication systems – George Kennedy & Bernard Davis, Tata Mcgraw Hills, 4th edition, 2008

2.Electronic communication Systems – Blake, Joseph J. Adams ki, Sun Yifeng, Delamer publication, 2nd edition, 2012 (Rupa Publication, India).

3.Fundamentals of Electrical engineering – Wayone tomasi K.Roy - University press.

## UNIT I: AMPLITUDE MODULATION AND TRANSMISSION

Introduction – amplitude Modulation – AM envelop – AM frequency spectrum and bandwidth – Phasor representation of AM with carrier – coefficient of modulation or percentage modulation or modulation index – degrees of modulation – AM power distribution – AM Current relation and efficiency - modulation by complex information signal - doubleside band suppressed carrier AM - single side band suppressed carrier AM – Vestigal side band amplitude modulation – AM modulator circuits – emitter modulations or low power AM – collector modulator or medium and high power AM modulator - AM transmitters – Broadcast AM transmitters – Low level of AM transmitter – High level AM transmitter

### Modulation

Modulation is the fundamental process of electronic communication. It allows voice, pictures and other informations to be transferred from one place to another. The modulation process is reversed at the receiver to recover the information (demodulation).

Modulation is defined as the process by which some characteristics of a high frequency wave such as amplitude, frequency or phase is altered in accordance with the instantaneous value of modulating signal. The modulating signal typically contains information to be trainsmitted.

If the amplitude of the carrier wave is varied in accordance with the signal voltage (low frequency), the result is amplitude modulated wave. The process is known as the *amplitude moduation*.

If the frequency of the carrier wave is varied in accordance with the signal voltage,. the result is frequency modulated wave. The process is known as the *frequency modulation*.

If the phase of the carrier wave is varied in accordance with the signal voltage, the result is phase modulated wave. The process is known as the *phase modulation*.

## Amplitude Modulation: Wave forms, side bands and power

In amplitude modulation the amplitude of high frequency carrier wave is made proportional to the instantaneous amplitude of modulating signal voltage.

Let the carrier wave voltage be represented by the equation

$$e_C = E_C \cos(\omega_C t + \theta)$$

Here $e_c$ denotes the instantaneous voltage of the carrier wave at the instant $t$. $E_c$ denotes the amplitude and $\omega_C$ the angular frequency of carrier and $\theta$ is the initial phase of the carrier wave.

For convenience, $\theta$ can be assumed to be zero

$$e_C = E_C \cos \omega_C t \qquad \text{... ... (1)}$$

Let the modulating signal voltage be represented by the equation

$$e_S = E_s \cos \omega_s t$$

where $e_s$ is the instantaneous signal voltage, $E_s$ its amplitude and $\omega_s$ its angular frequency.

As the signal modulates the amplitude of the carrier wave, one can write the varying voltage of the modulated wave as

$$e = (E_C + e_s) \cos \omega_C t$$

Putting the value of $e_s$, we have

$$e = (E_C + E_S \cos \omega_s t) \cos \omega_C t.$$

$$e = E_C \left(1 + \frac{E_S}{E_C} \cos \omega_S t\right) \cos \omega_c t$$

$$e = E_C \underline{(1 + m \cos \omega_S t)} \cos \omega_C t. \qquad \text{... ... (2)}$$

### *Modulation idex (m) and percentage Modulation*

The ratio $E_s$ of maximum amplitude of modulating signal to maximum amplitude of carrier signal $E_C$ is called modulation index ($m$).

$$\text{Modulation index, } m = \frac{E_S}{E_C}$$

The value of $E_S$ must be less than $E_C$ to avoid any distortion. The maximum value of $m = 1$ when $E_S = E_C$. The minimum value of $m$ is zero.

The ratio $m = \left(\frac{E_S}{E_C}\right)$ is known as the modulation index or degree of modulation. The quantity $m \times 100$ gives the percentage modulation ($M$) From equation (2), we find that

the amplitude of the modulated wave $= E_C (1 + m \cos \omega_S t)$

The maximum amplitude is for $\cos \omega t = 1$ and the minimum amplitude is for $\cos \omega t = -1$

Thus the maximum amplitude $E_{max} = E_C (1 + m)$ volt ... ... (3)

and the minimum amplitude $E_{min} = E_C (1 - m)$ volt ... ... (4)

For the modulated wave,

(3) / (4) gives $\frac{E_{max}}{E_{min}} = \frac{1 + m}{1 - m}$

$$\therefore m = \frac{E_{max} - E_{min}}{E_{max} + E_{min}}.$$

This gives an expression for the modulation index.

***Frequency spectrum and bandwidth***

Further, equation (2) can be written as

$$e = E_C \cos \omega_C t + m E_C \cos \omega_C t \cos \omega_S t. \quad .... \quad ... \quad (5)$$

We have the familiar relation

$\cos A \cos B = ½ [\cos (A+B) + \cos (A-B)]$

Using this relation,. equ.(5) can be written as

$$e = E_C \cos \omega_C t + \frac{m E_C}{2} \cos (\omega_C + \omega_S) t + \frac{m E_C}{2} \cos (\omega_C - \omega_S) t.$$

------------ (6)

Equation (6) gives the A.M spectrum or wave form of the amplitude modulated wave. It has three components namely:

1. $E_C \cos \omega_C t$, which is the same as the carrier wave. $(\omega_c = 2\pi f_c)$

2. $\left(\frac{m E_C}{2}\right) \cos(\omega_C + \omega_S)\, t$ which varies with an angular frequency $(\omega_C + \omega_S)$. This component is called the *upper side band*. Upper side band frequency (USB) = $f_c + f_s$.

3. $\left(\frac{m E_C}{2}\right) \cos(\omega_C - \omega_S)\, t$ which varies with an angular frequency $(\omega_C - \omega_S)$. This component is called the *lower side band*. Lower side band frequency (LSB) = $f_C - f_S$.

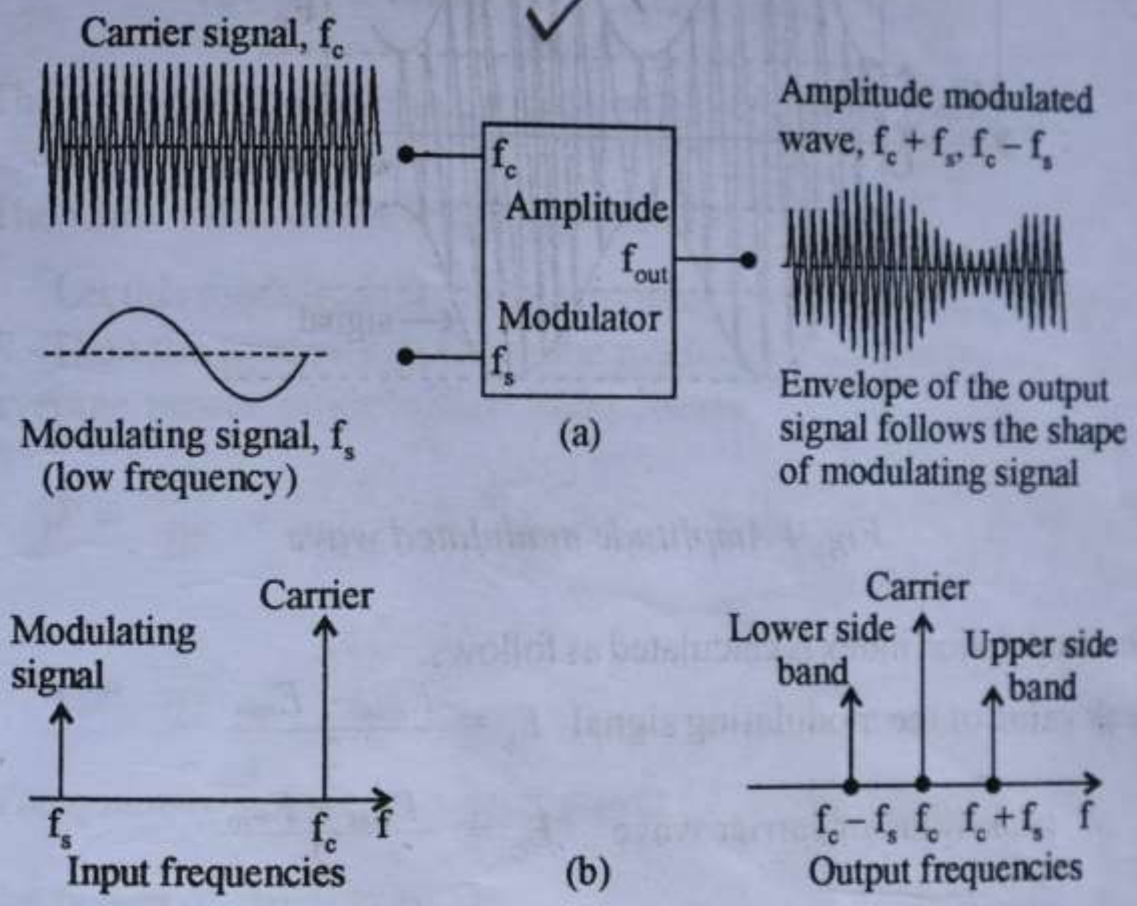



*Fig. 3 Amplitude modulation*

Bandwidth of A.M. wave is

$$BW = (f_C + f_S) - (f_C - f_S) = 2f_S.$$

The bandwidth of AM signal is twice the maximum frequency of modulating signal.

The amplitude modulated wave-form (A.M. spectrum) is shown in fig 4.

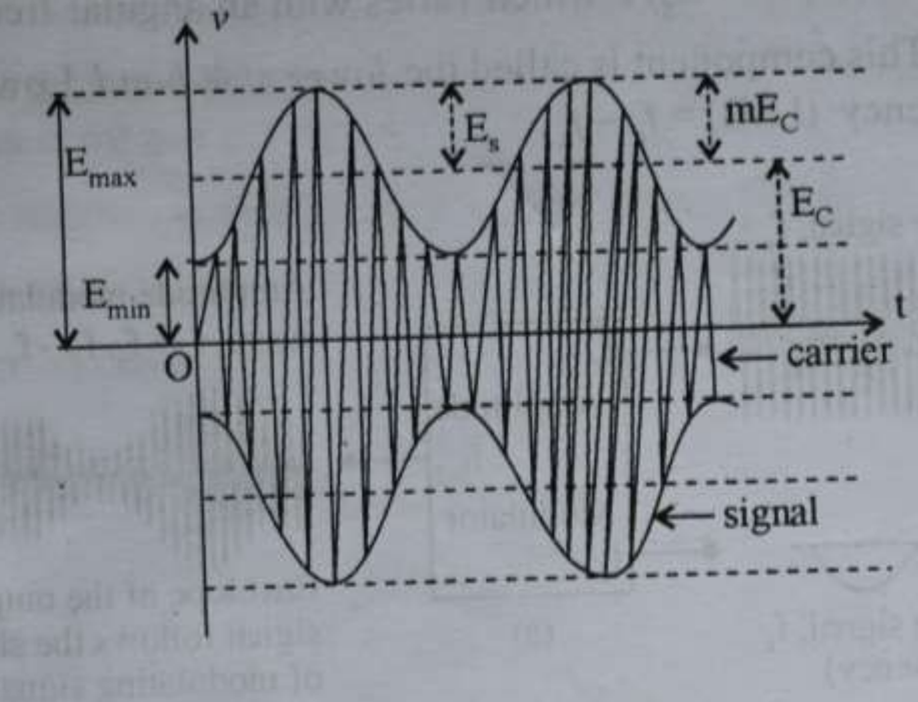



*Fig. 4 Amplitude modulated wave*

## Respresentation of AM: AM envelope

In amplitude modulation, the amplitude of a carrier signal is varied by the modulating voltage. The angular frequency $\omega_e$ of the carrier wave is greater than the angular frequency $\omega_s$ of the modulating signal. The carrier is a high-frequency (H.F) wave and the modulating signal is a audio - frequency wave. The amplitude modulation (AM) is defined as a system of modulation in which the amplitude of the carrier is made proportional to the instantaneous amplitude of the modulating signal.

The carrier voltage, $e_C = E_C \cos \omega_C t$ ............... 1

The modulating signal voltage, $e_S = E_S \cos \omega_S t$.

From the difinition of AM, it follows that the amplitude $E_c$ of the unmodulated carrier will have to be proportional to the instantaneous modulating voltage $E_S \cos \omega_S t$ when the carrier is modulated.

Modulation index $m = \frac{E_s}{E_c}$ ............ 2

Using eqn. (1) and (2), we can write an equation for the amplitude ( $A$ ) of amplitude - modulated voltage.

We have, $A = E_c + e_s$

$$A = E_c + E_s \cos \omega_s t = E_c + m E_c \cos \omega_s t \qquad m = \frac{E_s}{E_c}$$

$$= E_c + m E_c \cos \omega_s t$$

$$A = E_c(1 + m \cos \omega_s t)$$

The instantaneous voltage of the resulting amplitude - modulated wave is [using equation (1)]

$$e = A \cos \omega_c t = E_c(1 + m \cos \omega_s t) \cos \omega_c t$$

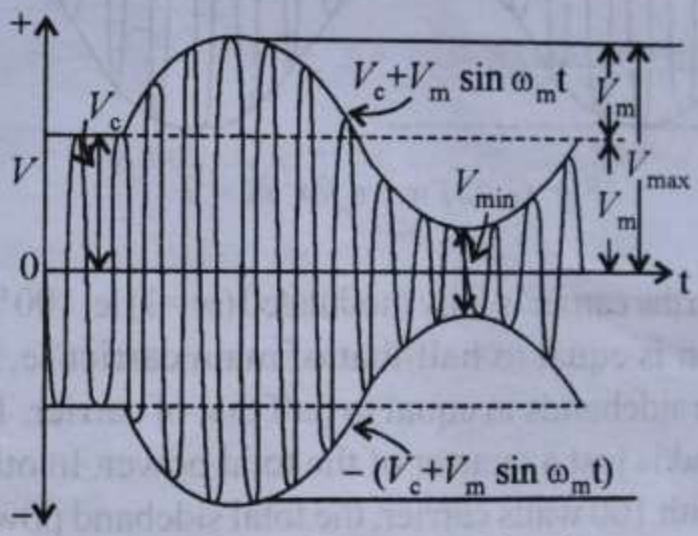



*Fig. 5. Amplitude - modulated wave.*

The appearance of the amplitude modulated wave is shown in figure. The top envelope of the AM wave is given by the relation

$$A = E_c + E_s \cos \omega_s t$$

Similarly, the maximum negative amplitude, or bottom envelope is given by $-A_1 = -(E_c + E_s \cos \omega_s t)$.

The modulated wave extends between these two limiting envelopes and has a repetition rate equal to the unmodulated carrier frequency.

## Phasor representation of AM with carrier

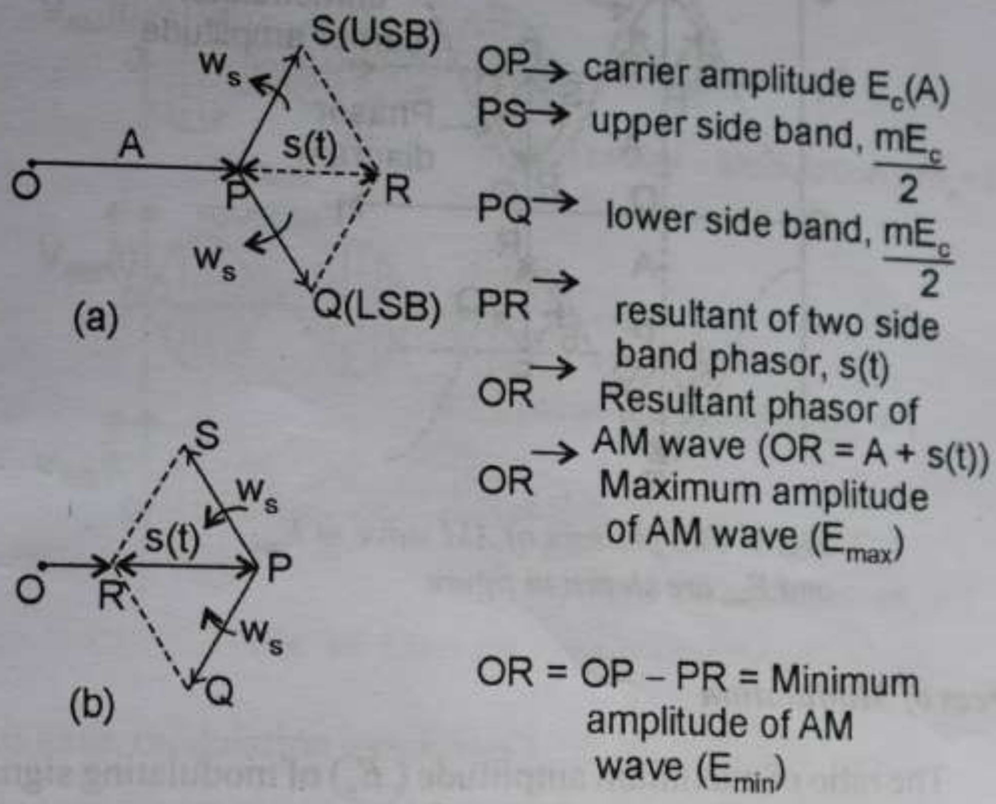



*Fig. 8. Phasor representation of AM*

Phasor representation of AM wave is the easy representation of AM wave with carrier. OP is the carrier wave phasor, taken as reference phasor.

The two side bands having a frequency $(\omega_c + \omega_s)$ and $(\omega_c - \omega_s)$ are represented by two phasors (PS and PQ), rotating in opposite directions with angular frequency of $\omega_s$. The resultant of sideband phasors is $s$(t) represented by PR.

The net or resultant phasor is $A + s(t)$ represented by OR. It depends on the position of the side band phasors and carrier wave phasor.

In AM the resultant of the two side band vectors is always *in phase* with carrier component.

***Degrees of Modulation***

The ratio of maximum amplitude ( $E_S$) of modulating signal to maximum amplitude ( $E_C$) of carrier signal is called modulation index or depth of modulation.

modulation index, $m = \frac{E_S}{E_C}$ .

The value of $E_S$ must be less than the value of $E_C$ to avoid any distortion in the modulated signal. ie, $m < 1$. Hence the maximum value of modulation index will be equal to 1 (ie, $m = 1$) when $E_S = E_C$ .

There are three degrees of modulation depending upon the amplitude of the message signal relative to carrier amplitude. (1) under modulation (2) critical modulation (3) over modulation.

*(1) Under modulation*

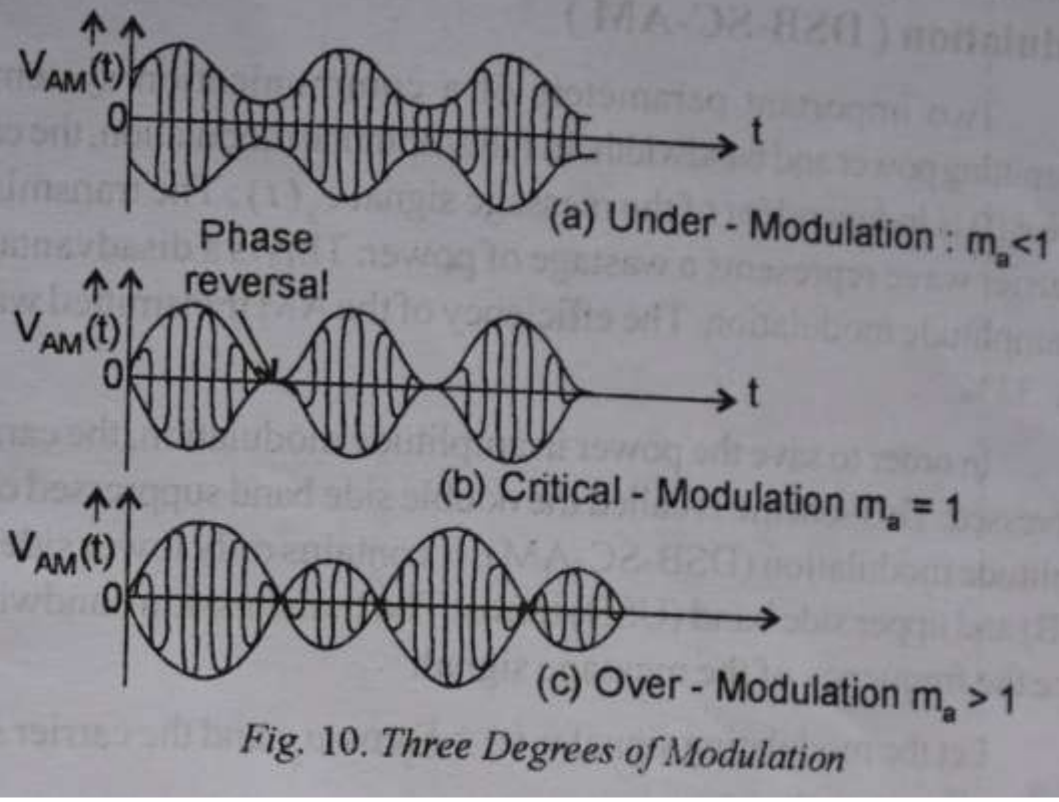



*Fig. 10. Three Degrees of Modulation*

In this case, modulation index $m < 1$. ie, $E_S < E_C$.

In this case the envelope of AM wave does not reach the zero amplitude axis. The message signal is fully preserved in the envelope of AM wave. This is known as under modulation. The envelope detector can recover the message signal without any distortion.

*(2) Critical modulation*

In this case $m = 1$, ie, $E_S = E_C$. The envelope of the modulated signal just touches the zero amplitude axis. The message signal remains preserved. This is known as critical modulation. The message signal can be recovered using an envelope detector.

*(3) Over modulation*

In this case $m > 1$. ie, $E_S > E_C$. The portion of envelope of the modulated signal (AM) crosses the zero axis. So both positive and negative extensions of modulating signal are cancelled or clipped out as shown in figure. The envelopes of message signal are not same. This is called invelope distortion. Hence the envelope detector provides distorted message signal.

## DOUBLESIDE BAND SUPRESSED CARRIER AMPLITUDE MODULATION (DSB-SC-AM)

In order to save the power in amplitude modulation, the carrier is suppressed. This scheme is called the double side band suppressed carrier amplitude modulation (DSB-SC-AM). It contains only lower side band (LSB) and upper side band (USB) terms. The transmission bandwidth is twice the frequency of the message signal.

Let the modulating signal $v_S(t) = E_S \cos \omega_S t$ and the carrier signal $v_C(t) = E_C \cos \omega_C t$.

A double side band suppressed - carrier AM signal ($e$) is obtained by multiplying the message signal $e_s$ with the carrier signal $e_c$.

Thus we have the amplitude modulated signal:

$$
\begin{aligned}
e &= e_C(t) \times e_s(t) \\
&= E_C \cos \omega_C t \, . \, E_S \cos \omega_S t \\
&= E_S E_C \cos \omega_C t \, . \cos \omega_S t
\end{aligned}
$$

$$e = \frac{E_S E_C}{2}\cos(\omega_c + \omega_s)t + \frac{E_S E_C}{2}\cos(\omega_c - \omega_s)t \quad \text{.........(1)}$$

In the modulated wave (eqn 1) the carrier signal is missing and only two side bands are present. Hence the signal is called as DSB-SC-AM. The frequency spectrum of DSB-SC-AM wave is given in the figure

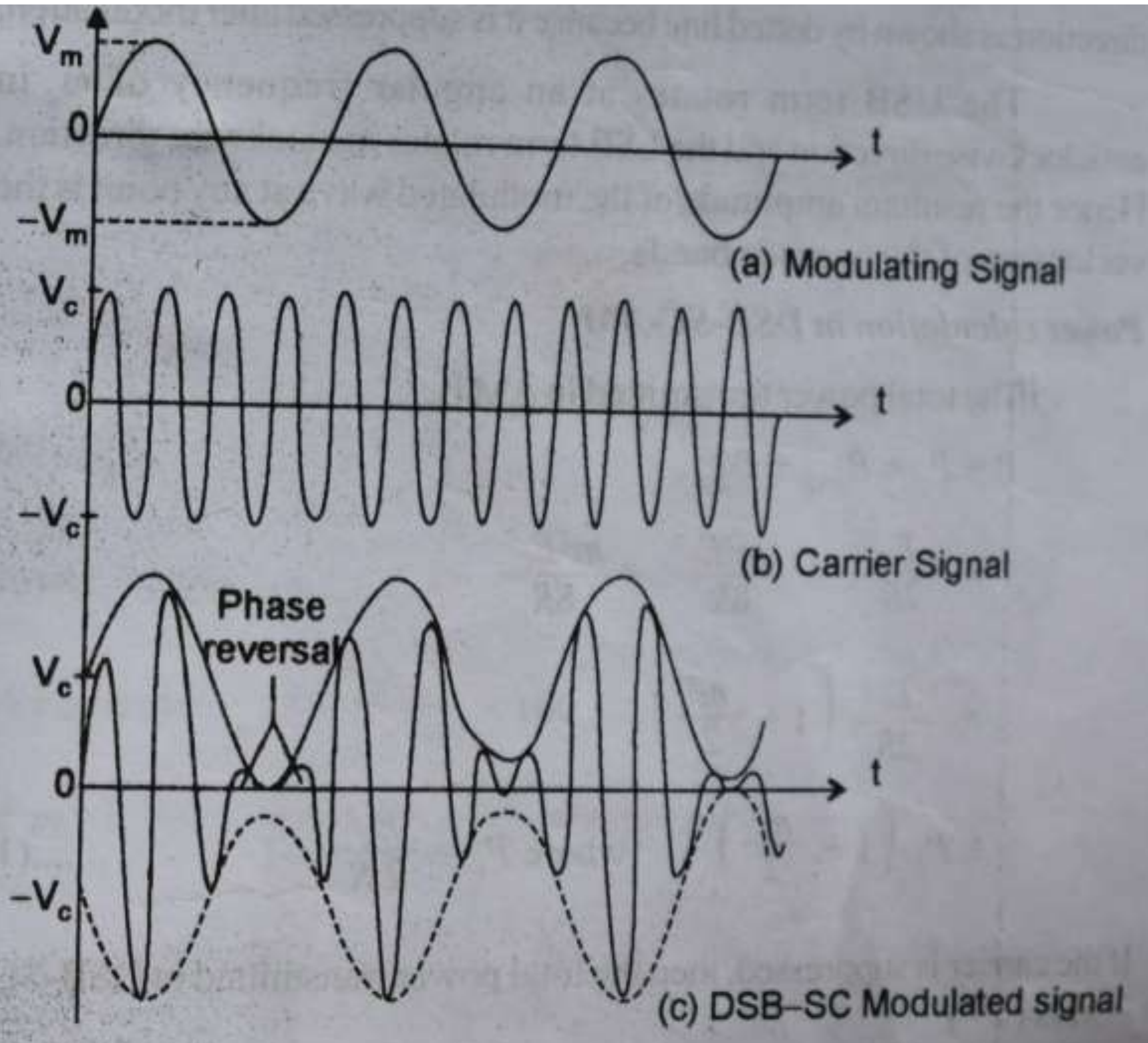



*Fig. 11B. Graphical Representation of DSB - SC - AM*

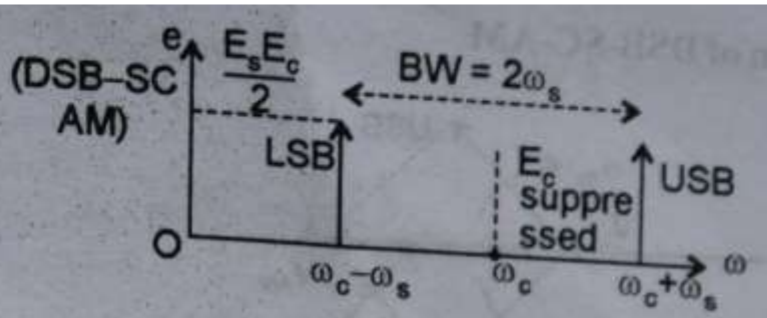



*Fig. 11A. Frequency spectrum of DSB - SC - AM*

## Single side band-suppressed carrier modulation (SSB-SC-AM)

In the amplitude modulated (AM) wave, the carrier wave conveys no information, but consumes two third of the power. The two side bands carry the same information. Thus only one side band is capable of carrying the full information. Thus saving of power is possible by eleminating one side band in addition to the carrier component. The transmission bandwidth can be cut into half if, one side band is suppressed along with the carrier. This scheme is known as SSB-SC-AM.

### Phase shift method to generate SSB-SC-AM

The block diagram of SSB-SC-AM is shown in figure.

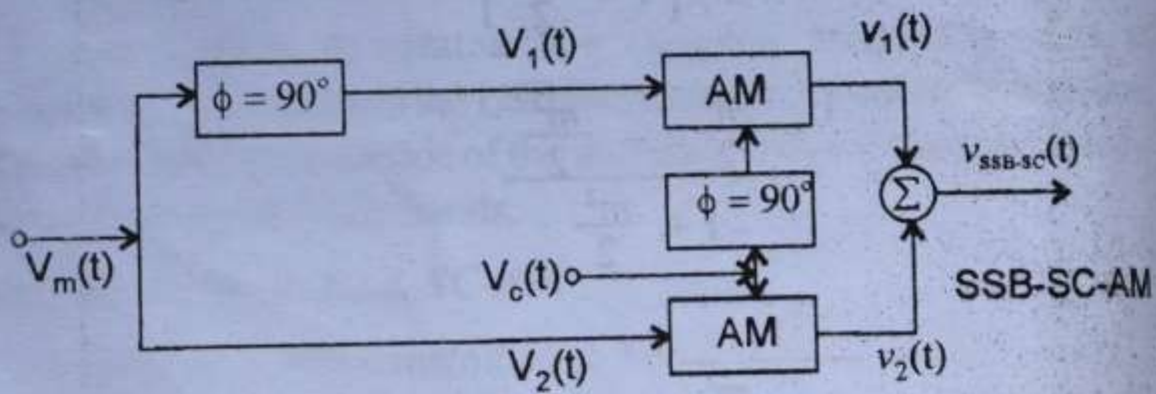



*Fig. 13. Block diagram of SSB-SC-AM*

The SSB-SC-AM can be obtained as follows.

The input signal $e_s(t)$ fed to the modulator 1 is 90° out of phase with that of the signal fed to the modulator 2.

$$\text{Let } V_1(t) = E_S \sin(\omega_S t + 90°) \times E_C \sin(\omega_C t + 90°)$$

$$V_1(t) = E_S \cos\omega_S t \,.\, E_C \cos\omega_C t \,.$$

$$V_2(t) = E_S \sin\omega_S t \,.\, E_C \sin\omega_C t \,.$$

$\therefore$ single side band modulated signal is obtained by summing the above two signals. The SSB-SC-AM signal is

$$e_m(t) = V_1(t) + V_2(t)$$

$$= E_S E_C \; [\cos \omega_s t \, . \cos \omega_c t + \sin \omega_s t . \sin \omega_c t] \qquad ..... (1)$$

We know that,

$$\sin A \; \sin B + \cos A \cos B = \frac{\cos (A-B)}{2}$$

$\therefore$ Equation (1) becomes,

$$e_m(t) = \frac{E_S E_c}{2} \; [\cos (\omega_c - \omega_s) \, t] \qquad ........(2)$$

This is the expression for SSB - SC-AM. It has only one side band $(\omega_C - \omega_S)$. The upper side band $(\omega_C + \omega_s)$ is suppressed.

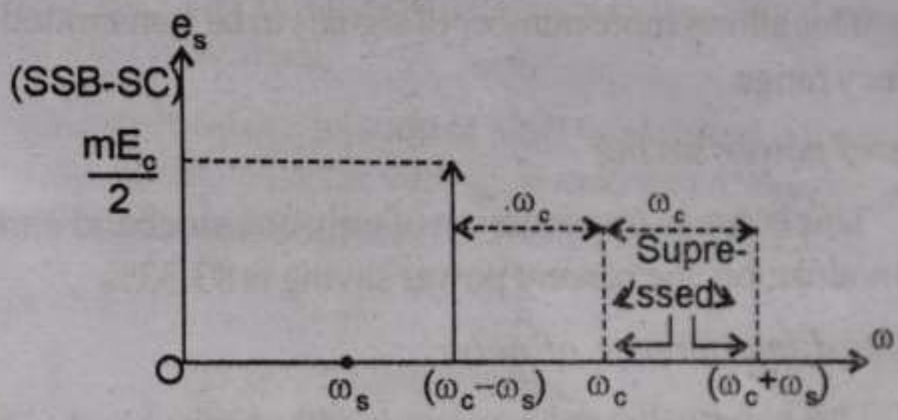



*Fig. 14. Frequency spectrum of SSB-SC-AM*

The frequency spectrum of SSC-SC-AM is shown in figure. It shows that only one side band (LSB) is present, the carrier and other (USB) side band signals are suppressed (dotted line). Thus the bandwidth required reduces from $2\omega_s$ to $\omega_s$. ie, bandwidth requirement is reduced to half compared to AM and DSB-SC-AM signals. The graphical representation and phasor diagram of SSB-SC-AM system is shown in figure.

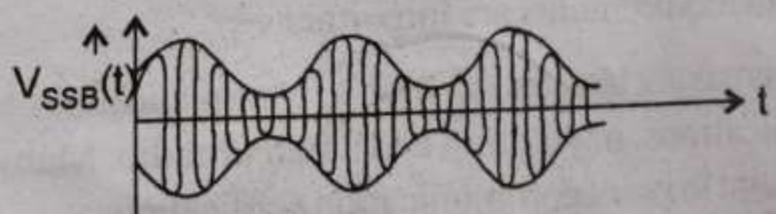



*Fig. 15(a). Graphical representation of SSB-SC-AM*

## Single side band-suppressed carrier modulation (SSB-SC-AM)

In the amplitude modulated (AM) wave, the carrier wave conveys no information, but consumes two third of the power. The two side bands carry the same information. Thus only one side band is capable of carrying the full information. Thus saving of power is possible by eleminating one side band in addition to the carrier component. The transmission bandwidth can be cut into half if, one side band is suppressed along with the carrier. This scheme is known as SSB-SC-AM.

### Phase shift method to generate SSB-SC-AM

The block diagram of SSB-SC-AM is shown in figure.

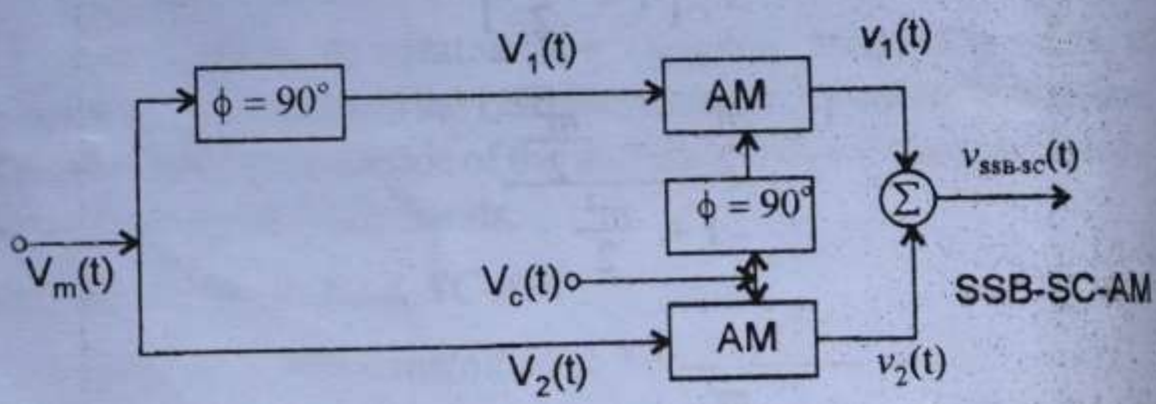



*Fig. 13. Block diagram of SSB-SC-AM*

The SSB-SC-AM can be obtained as follows.

The input signal $e_s(t)$ fed to the modulator 1 is 90° out of phase with that of the signal fed to the modulator 2.

$$\text{Let } V_1(t) = E_S \sin(\omega_S t + 90^\circ) \times E_C \sin(\omega_C t + 90^\circ)$$

$$V_1(t) = E_S \cos\omega_S t \, . \, E_C \cos\omega_C t \, .$$

$$V_2(t) = E_S \sin\omega_S t \, . \, E_C \sin\omega_C t \, .$$

$\therefore$ single side band modulated signal is obtained by summing the above two signals. The SSB-SC-AM signal is

$$e_m(t) = V_1(t) + V_2(t)$$

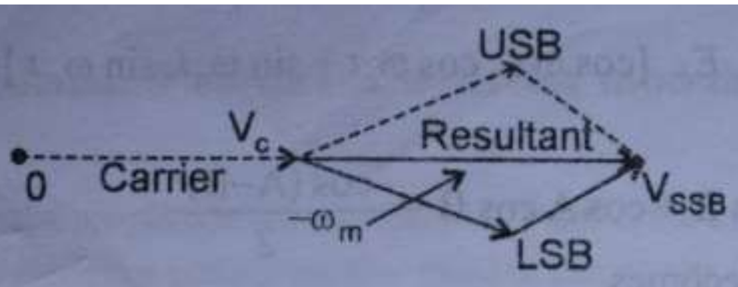



*Fig. 15(b). Phasor Diagram of SSB-SC-AM*

*Advantages of SSB-SC-AM*

The advantages of SSB-SC over DSB-FC signal are-

*1. Less bandwidth requirements*

This allows more number of signals to be transmitted in the same frequency range.

*2. Lots of power saving*

This is due to transmission of only one sideband component. At 100% modulation, the percent power saving is 83.33%

*3. Reduced interference of noise.*

This is due to the reduced bandwidth. As the bandwidth increases the amount of noise added to the signal will increase.

*Disadvantages of SSB-SC-AM*

1. The generation and reception of SSB signal is complicated.

2. The SSB transmitter and receiver need to have an excellent frequency stability. A slight change in frequency will hamper the quality of transmitted and received signal. Therefore, SSB is not generally used for the transmission of good quality music. It is used for speech transmission.

*Application of SSB*

1. SSB transmission is used in the applications where the power saving and low bandwidth requirements are important.

2. The application areas are land and air mobile communicatioin, telemetry, military communications, navigation and amateur radio. Many of these applications are point to point communication applications.

# Vestigial side band Transmission (VSB Modulation)

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

Hence to avoid this loss, a technique is chosen, which is a compromise between DSB-SC and SSB, called Vestigial side band (VSB) technique. The word vestige means 'a part or portion'.

*Vestigial side band*

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

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

*Frequency spectrum*

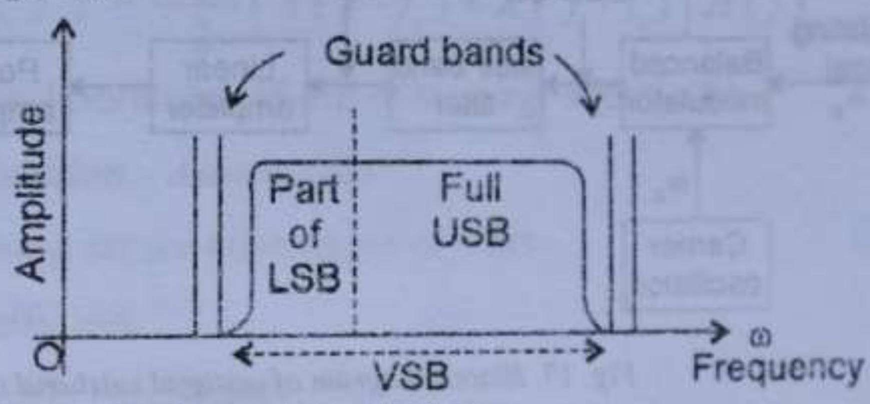



*Fig. 16. VSB Modulation*

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

*Transmission bandwidth*

The transmission bandwidth of VSB modulated wave is respresented as

$$B = (f_m + f_v)\ \text{Hz}$$

To generate a VSB signal, we have to first generate a DSB-SC signal and then pass it through a filter as shown in figure. This filter will pass the wanted sideband as it is along with a part of unwanted sideband.

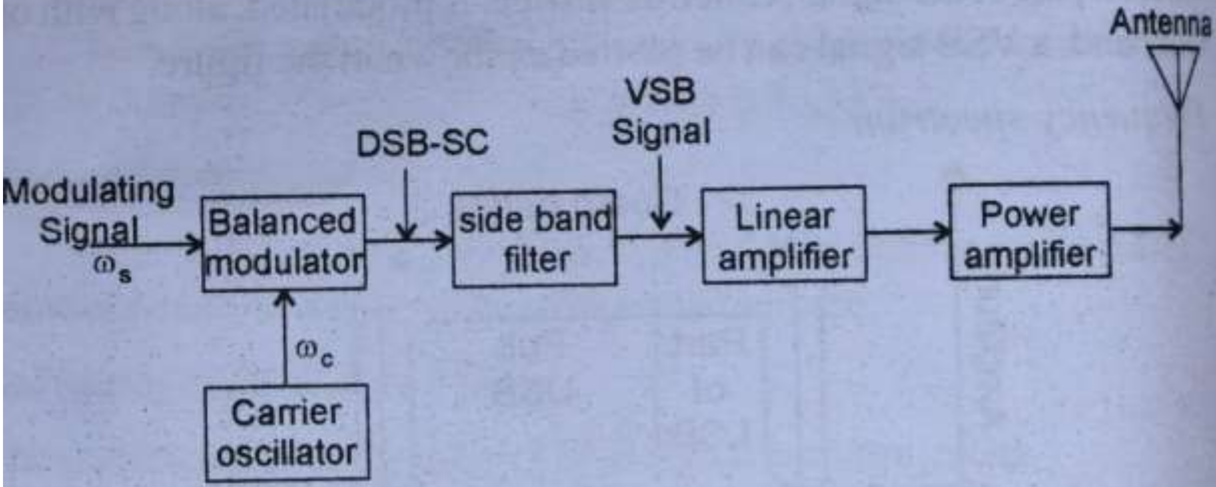



*Fig. 17. Blaock diagram of vestigial sideband transmission.*

## *Generation of VSB modulated wave*

The block diagram of a VSB modulator is shown in figure.

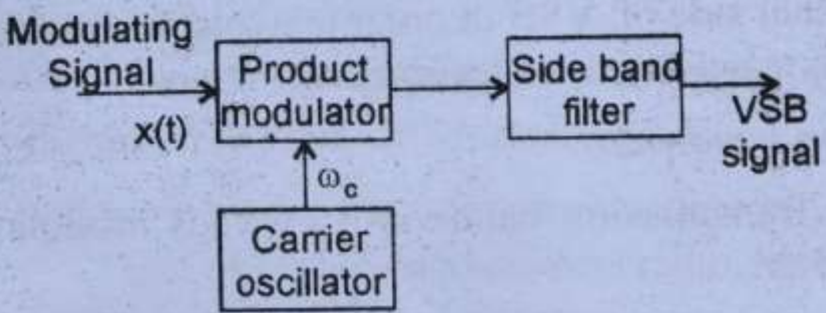



*Fig. 18. Generation of VSB modulated wave*

The modulating signal $x(t)$ is applied to a product modulator. The output of the carrier oscillator $c(t)$ is also applied to the other input of the product modulator. The output of the product modulator is then given by:

$$m(t) = x(t).\,c(t)$$

$$= x(t).\,V_c \cos(2\pi f_c t)$$

This represents a DSB–SC modulated wave.

*VSB Modulation - Advantages*

The following are the advantages of VSB -

1. Highly efficient.
2. Reduction in bandwidth
3. Filter design is easy as high accuracy is not needed.
4. Transmission of low frequency component is possible, without difficulty.
5. Possesses good phase characteristics.

*VSB modulation - Disadvantages*

The following are the disadvantages of VSB

1. Bandwidth when compared to SSB is greater.
2. Demodulation is complex.

**AM MODULATOR CIRCUITS**

**EMITTER MODULATIONS OR LOW POWER AM**

*Circuit diagram for low level AM modulator.*

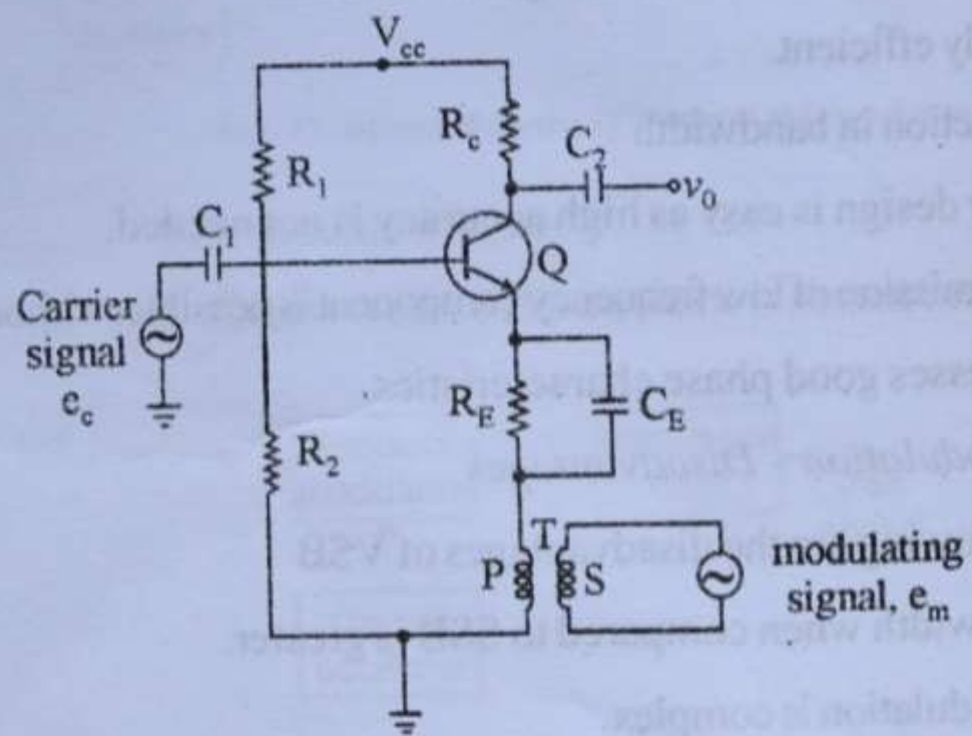



*Fig. 19. Circuit diagram of low level AM modulator*

Figure shows the circuit diagram of low level AM modulator. It as two imputs (i) carrier signal and (ii) modulating signal. The modulator perates in class A mode. In the absence of modulating signal, it is linear lass A amplifier and amplifies carrier signal ($e_c$)

$$A_v = A\,[1 + m \sin(\omega_m t)]$$

where, $A_v$ is voltage gain with modulation, $A$ is voltage gain without modulation. Since $\sin(\omega_m t)$ varies from $-1$ to $1$,

$$A_v = A\,(1 \pm m)$$

Above equation shows that voltage varies according to modulation index ($m$). Fig. shows the wave forms of this modulator.

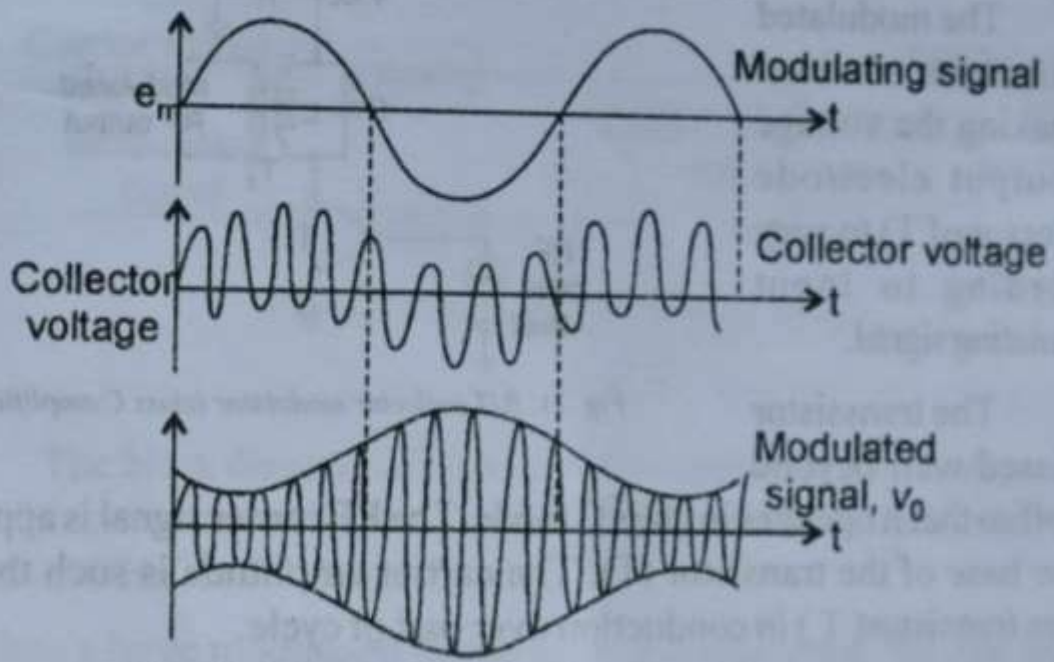



*Fig. 20. Waveforms of low level modulator*

The collector voltage varies according to modulating signal. The capacitor $C_2$ removes dc level in collector voltage and we get AM signal at the output.

*Advantages*

1. Modulator operates at low voltage level. Hence less power consumption in the modulator.

2. The circuit is very simple, since it is basically class A amplifier.

*Disadvantages*

The modulator operates in class A, hence its power efficiency is very low. Class B linear RF power amplifiers are usually employed for this purpose.

**Medium - power AM modulator (Collector modulator)**

Figure shows the circuit diagram of medium power AM modulator (callector modulator). It has two inputs (i) modulating signal and (ii) carrier (RF) signal.

The modulated output can be obtained by making the voltage on output electrode (collector of T) to vary according to input modulating signal.

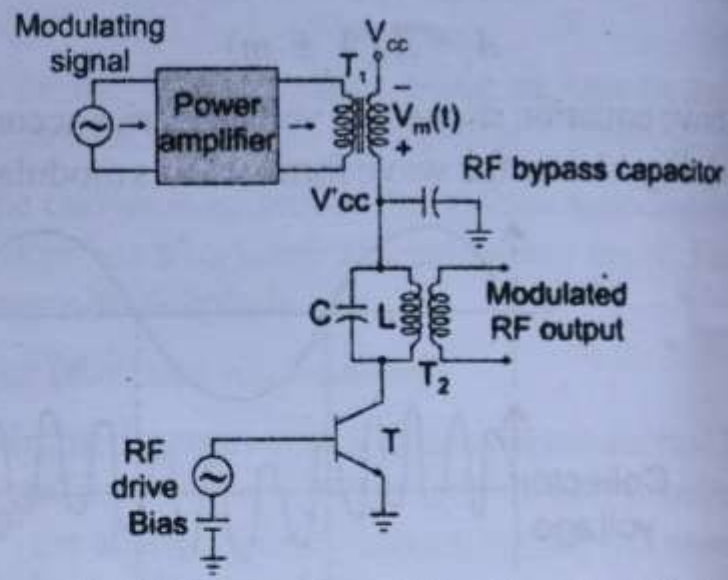



*Fig. 21. BJT collector modulator (class C amplifier)*

The transsistor is biased well beyond cut-off so that it operates in class C mode. The RF carrier signal is applied to the base of the transistor (T). The carrier amplitude is such that it drives transistor( T ) in conduction over part of cycle.

The modulating signal is passed through the power amplifier and applied to the collector through a low frequency transformer ($T_1$). This voltage is shown as $v_m(t)$.

The modulating voltage output at ($T_1$) is in series with the supply voltage $V_{CC}$ of transistor (T). Hence the collector voltage of (T) becomes $V'_{CC} = V_{CC} + v_m(t)$. Thus the output of the amplitude of carrier sine wave varies in accordance with the modulated signal.

*Working*

The tuned L-C circuit associated with transformer ($T_2$) in the collector of (T) receives the A.M. signal. The modulating signal can vary the net supply voltage of the collector of transistor (T) as $V_{cc}$.. ie, the net supply voltage of transistor changes according to the slow variation in $v_m(t)$. Hence RF carrier signal amplitude is also varied according to variation in $v_m(t)$. Thus AM signal is produced across the LC circuit at the collector. The AM signal is derived from the secondary of transformer($T_2$).

*Advantages*

The collector modulator produces the best type of AM modulation, but it requires a high power modulator circuit.

**High power AM modulation**

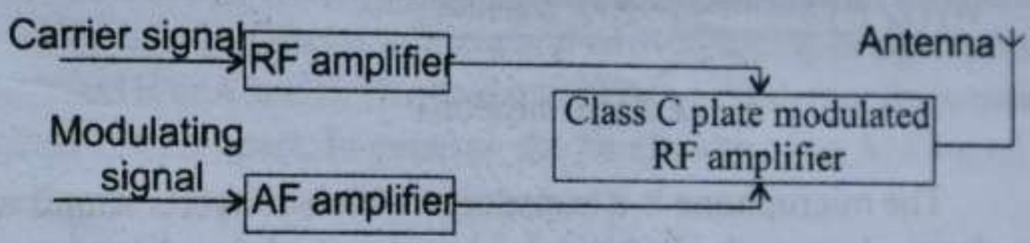



*Fig. 22. High level modulation system*

The block diagram of high level modulation system is shown in fig. In this case the carrier and modulating signals are amplified to the desired level before the modulation.

(1) When a large modulated power is to be transmitted, the low level modulation system is quite unsuitable because of low efficiency.

(2) After modulation which is usually carried out by using a plate modulated class C amplifier, the output is fed to the antenna without further amplification. Such a system is termed as high level modulation.

(3) power conversion efficiency is higher.

(4) This system requires less critical adjustments.

**Transmission of Radio waves: AM transmitter**

Audio signals (such as music, speech etc) are transmitted over large distances by converting them into corresponding electrical signals.

These signals are used to modulate high frequency carrier waves and the modulated waves are used to energise the antenna. The antenna radiates the radio waves into space. This is what happens in a radio transmitter.

The block diagram of a radio transmitter is shown in the figure. The important units of the transmitter are microphone, audio frequency amplifier, high frequency oscillator to generate carrier waves, an RF amplifier, a modulator and the antenna to radiate the radio waves.

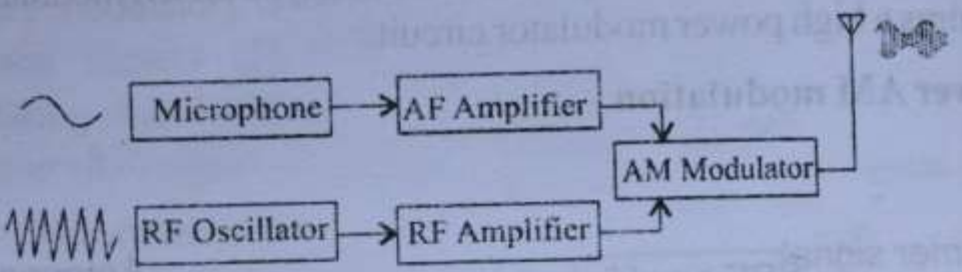



*Fig. 23. AM Transmitter*

## Broadcast AM transmitter

The purpose of most transmitter is radio communication of information over a distance. A broadcast transmitter refers to an installation used for broadcasting,. including radio transmitter or television transmitter equipment, the antenna, and often the localion of the broadcasting station. Radio broadcasting is transmission by radio waves intended to reach a wide audience.

A transmitter ( or radio transmitter) is an electronic device which produces radio waves with the help of an antenna. A transmitter generates a radio frequency alternating current applied to the antenna, which, in turn, radiates radio waves.

In an amplitude modulation (AM) transmitter, the information is added to the radio signal by varying its amplitude. In a frequency modulation (FM) transmitter, it is added by varying the radio signal's frequency slightly.

Transmitters that transmit AM signals are known as AM transmitters. These transmitters are used in medium wave (MW) and short wave (SW) frequency bands for AM broad cast. The MW band has frequencies between 550 kHz and 1650 kHz, and the SW band has frequencies ranging from 3MHz to 30 MHz. Two types of AM transmitters that are used based on their transmitting powers are:

(i) High level and (ii) Low level.

High level transmitters use high level modulation, and low level transmitters use low level modulation. The choice between the two modulation schemes depends on the transmitting power of the AM transmitter. In broadcast transmitters, where the transmitting power may be of the order of kilowatts, high level modulation is employed. In low power transmitters where only a few watts of transmitting power are required, low level modulation is used.

**High level AM transmitter**

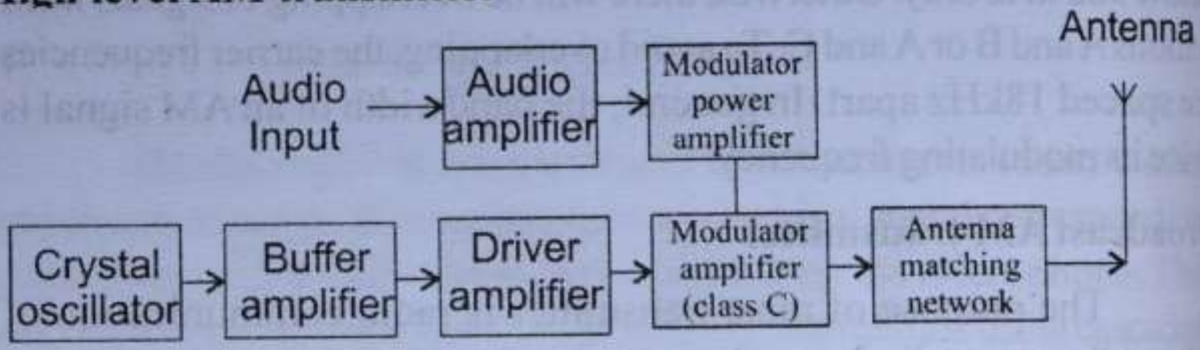



*Fig. 24. High level AM transmitter block diagram*

Figure shows the block diagram of AM transmitter. Crytstal oscillator generates carrier frequency. Buffer amplifier and driver amplifier amplify the power level of the carrier to required value. This amplified carrier signal is fed to class C modulating amplifier.

The modulating audio signal is amplified by audio amplifier. This signal is further amplified by audio power amplifier at a level suitable for modulation. The class C modulator amplifier modulates, the carrier input according to modulating audio signal. This AM modulated output is given to antenna through antenna matching network. The antenna matching network is a turned LC circuit in collector circuit of modulator amplifier. In this AM transmitter, the modulator amplifier operates at high power levels and delivers power directly to the antenna. The antenna transmits high power e.m. wave into the space surrounding it.

## Low level AM transmitter

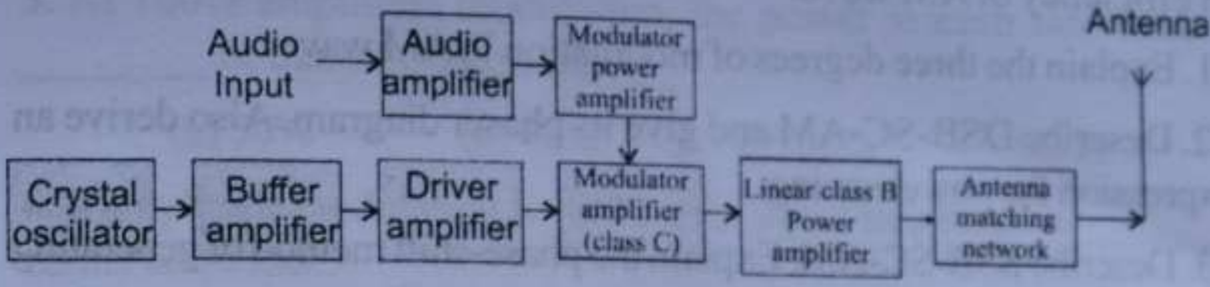



*Fig. 25. Low level AM transmitter block diagram*

The block diagram of low level AM transmitter is shown in figure. The linear class B power amplifier is used after class C modulator amplifier. The AM signal, amplified by linear class B power amplifier is fed to the antenna. The modulator amplifier performs modulation at relatively low power level. Hence this circuit is called low level modulated AM transmitter. The modulated AM signal is amplified by class B power amplifier to avoid distortion in the output. The antenna converts the AM modulated signal into electromagnetic wave and transmits the e.m. wave into the space surrounding it.

# UNIT II

## AMPLITUDE MODULATION –RECEPTION

Comparison of AM system – Quadrature amplitude modulation – principles of AM detection – AM receivers – receiver parameters – Tuned radio frequency (TRF) receiver or straight receiver – principles of superhetrodyne –double frequency conversion AM receiver

### Comparison of AM Systems

In standard AM Systems, the sidebands are transmitted in full, accompained by the carrier. Accordingly, demodulation is accomplished simply by using an envelope detector. On the other hand, in suppressed carrier systems, the receiver is more complex because additional circuitary must be provided for the purpose of carrier recovery. Hence in commercial AM radio broadcast systems which involves one transmitter and numerous receivers, standard AM is used in preference to DSB - SC or SSB modulation.

Suppressed carrier modulation systems have an advantage over standard AM systems in that they require much less power to transmit the same amount of information, which makes the transmitter for such a system less expensive than those required for standard AM. Suppressed - carrier systems are well suitated for point to point communication involving one transmitter and one receiver.

Single sideband modulation requires the minimum transmitter power and minimum transmission bandwidth possible for conveying a message signal from one point to another. The vestigial sideband requires a transmission bandwidth that is intermediate between that required for SSB and DSB-SC modulation. DSB-SC modulation, SSB modulation, VSB modulation are the examples of linear modulation.

Following Table lists the comparison of various AM systems.

| Sr. No. | Parameter | AM with carrier | DSB -SC | SSB -SC | VSB |
|---|---|---|---|---|---|
| 1 | Method | Carrier and both sidebands | Only sidebands | Only one sideband | One side-band and part of other sideband. |
| 2 | Bandwidth | $2f_m$ | $2f_m$ | $f_m$ | $f_m < BW < 2f_m$ |
| 3 | Generation | Easy | Easy | Complex | Complex |
| 4 | Transmission efficiency | 33.3% | 100 % | 100 % | $33.3\% < \eta < 100\%$ |
| 5 | Selective fading | Heavy distortion in received signal | More distortion compared to SSB-SC | Least distortion | Received singal is distorted. |

Fig 26. Table: *Comparison of various AM systems*

## Quadrature Amplitude Modulation (QAM)

The DSB -SC modulation signal occupies twice the bandwidth of the signal. This is very wasteful of the avilable frequencey spectrum.

The quadrature amplitude modulation (QAM) is similar to DSB - SC modulation. The only difference is that QAM sends *two message signals* over the same Spectrum. Since each side band of DSB - SC modulation carries independent signals with suppressed carrier, there is no waste of the available frequency spectrum of DSB -SC modulation. QAM is both an analog and a digital modulation technique.

Quadrature Amplitude modulation QAM is a signal in which two carriers with a phase shift of 90 degrees between them are modulated. The resultant output consists of both amplitude and phase variations. These are then phase shifted carrier- amplitude modulated with the two data streams known as I or In - phase and the Q or quadrature ($90^0$ phase shift) data streams. These are generated in the base band (message signal ) processing area.

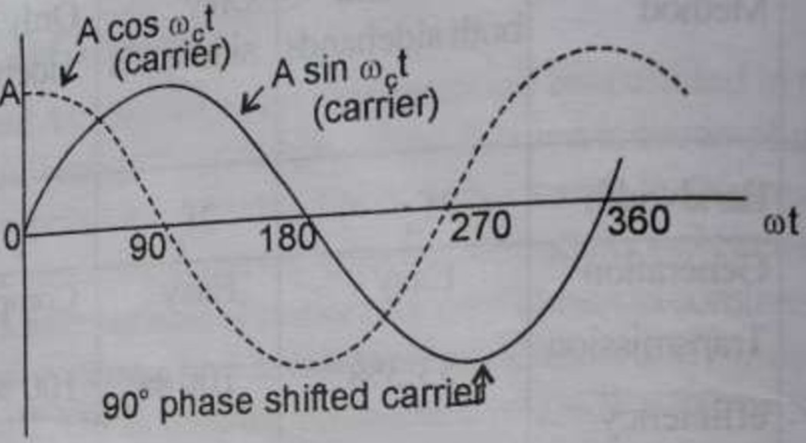



*Fig. 27. QAM carriers having phase difference of* 90°

Thus in QAM, the two carrier waves are out of phase by 90°; that is why this scheme is known as QAM and the carriers are quadrature carriers.

The two resultant signals are *summed* and then *processed* as required in the RF signal chain, typically converting them in frequency to the required final frequency and *amplifying* them as required.

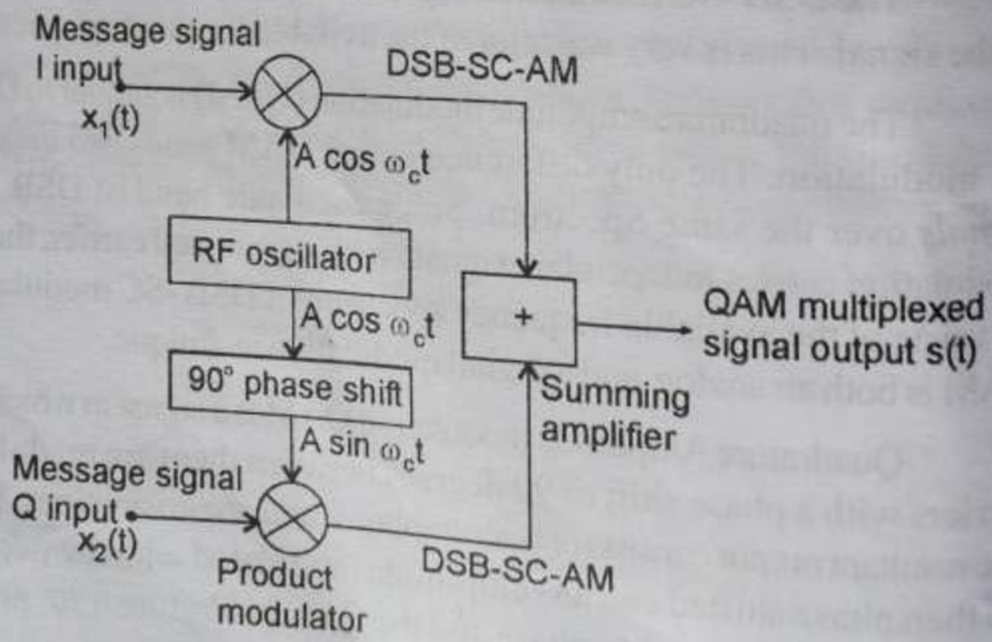



*Fig. 28. QAM modulator block - diagram*

But, if the locally generated carriers at the receivers have some phase error, then, it will cause a serious problem.

*Advantanges*

QAM increases the efficiency of transmission for radio communication systems by utilising both amplitude and phase variations.

*Dis advantages*

1. QAM is susceptible to noise because the states are closer together so that a lower level of noise is needed to move the signal to a different decision point.

2. QAM contains an amplitude component. Only linear amplifiers can be used in QAM. unfortunately linear amplifiers are less efficient and consume more power and this makes them less attractive for mobile applications.

*Uses and application of QAM*

1. Digital radio communication

2. Digital cable channels

3. Dial - up modems (A hardware device that allows a computer to send and receive data over a telephone line or a cable satellite connection).

4. Wi -Fi systems. (Wireless networking technology that uses radio waves to provide wireless high-speed internet connections).

**AM Receivers**

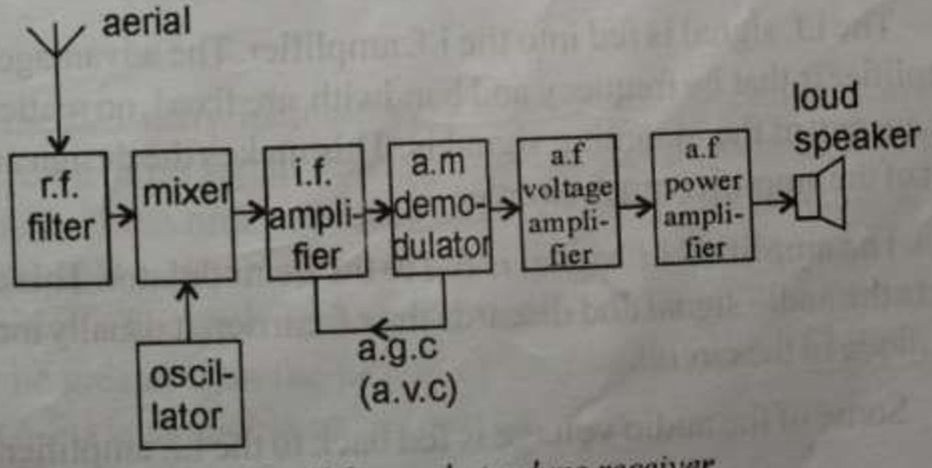



*Fig. 31. AM superhetrodyne receiver*

The block diagram of AM Reciver is shown in figure. The broadcast AM signal from transmitting station is converted into electromagnetic (e.m) waves by the broadcasting antenna. The e.m. waves are propagated in space in all directions. At the receiving centre, the e.m. waves induce weak alternating current R.F. signal ( of the order of microvolt) in the receiving antenna. The required RF signal is filtered out, amplified and the AF signal is demodulated. The demodulated AF signal is amplified and fed to the speaker. Thus the message is transmitted from transmitter to receiver.

The various stages of the AM receiver is shown in the figure. The filters, mixers, frequency changers, AM modulation and amplifiers are there in the receiver module.

There are signals from thousands of radio transmitters on many different frequencies inducing signal voltages in the aerial. The r.f. filter selects the desired station from the many. It is adjustable so that the selection frequency can be altered. This is called tuning.

The selected frequency is applied to the mixer.

The output of a (local) oscillator is also applied to the mixer.

The mixer and oscillator form a frequency change circuit.

The output from the mixer is the intermediate frequency (i.f)

The i.f. is a fixed frequency of about 455 kHz.

No matter what the frequency of the selected radio station is, the i.f.is always 455 kHz.

The i.f. signal is fed into the i.f. amplifier. The advantage of the i.f. amplifier is that its frequecy and bandwith are fixed, no matter what the frequency of the incoming signal is. This makes the design and operation of the amplifier much simpler.

The amplified i.f. signal is fed to the demodulator. This circuit recovers the audio signal and discards the r.f. carrier. It usually incorporates a diode in the circuit.

Some of the audio voltage is fed back to the i.f. amplifier as an *automatic gain control* voltage.

This ensures that when tuning from a weak station to a strong one, the loudness from the loudspeaker stays the same.

The audio signal voltage is increased in amplitude by a voltage amplifier.

The power level is increased sufficiently to drive the loud speaker by the power amplifier.

## Demodulation (AM detection) : Envelope detector

Separating the A.F. signal from the modulated carrier wave reaching the receiver is known as the demodulation or detection. The circuit employed for this purpose is known as a demodulator or detector. Demodulation can be done by using a circuit known as envelope detector. It operates by detecting the envelope of the incoming signal which it does to remove any radio frequency components of the signal at the output.

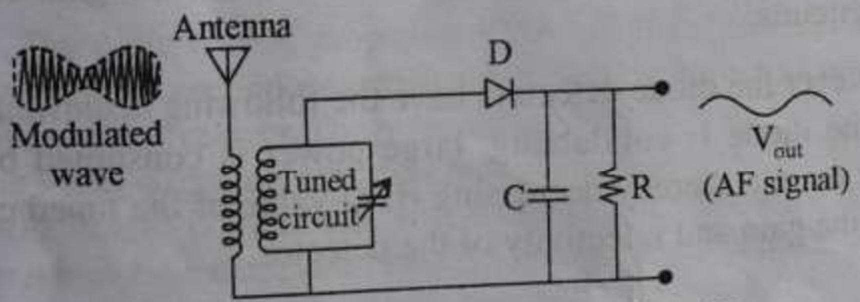



*Fig. 32 A.M. detection - Envelope detector*

The modulated wave incident on the antenna of receiver is applied to a detector. The signal voltage induced in the antenna of a radio receiver is of the order of mocro-volt. The circuit is made up of a crystal diode D with a capacitor C and resistance R forming a parallel combination. The time constant of the circuit $\tau = CR$ must be large. It must be greater than the period of the carrier wave. The detected signal (A.F.) is available as the output.

*Working* During each carrier cycle, the diode turns *on* for a small time and charges the capacitor to the maximum voltage of the wave each time. In the time interval between the peak voltage and the next peak voltage of the carrier wave, we get only a *slight* discharge between the cycles. Hence the carrier wave is absent in the output. The output looks like the upper envelope of the output but with a small variation. The small variation (ripples) can be removed by adding a low pass filter circuit at the output part of the demodulator. Thus the audio signal is seperated from the carrier wave and demodulation is effected.

*Advantages of diode detectors*

1. The diode detectors have the ability to handle input signals to well above 1 volt. i.e; they can handle relatively high power signals.
2. They can be operated as linear or non-linear detectors.
3. They have no distortion and hence have good linearity.
4. It provides a usable dc voltage for the automatic gain control (AGC) circuits.

However the diode detectors have the following *disadvantages*. When the diode is conducting, large power is consumed by the detector circuit, thereby decreasing (i) Q-value of the tuned circuit and (ii) the gain and selectivity of the detector.

**Tuned Radio Frequency (TRF) Receiver (Straight Receiver)**

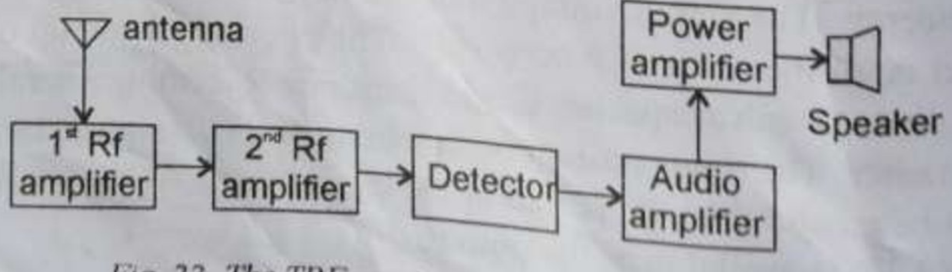



*Fig. 33. The TRF receiver*

Until shortly before World War II, most radio receivers were of TRF type whose block diagram is shown in figure.

Two or three RF amplifiers all tuning together (ganged) were employed to select and amplify the incoming frequency (radiowaves) and simultaneously to reject all others. The signal is amplified to a suitable level. Then the signal was demodulated ( detected). The detected, AF signal was amplified by audio amplifier, then by power amplifier. The amplified audio signal was fed to loud speaker. Such receivers were simple to design and align at broadcast frequencies (535 to 1640 kHz). They presented difficulties at higher frequencies.

TRF receiver suffered from a variation in bandwidth over the tuning range. Also it was unable to achieve sufficient selectivity at high frequencies. This was due to the use of single - tuned circuits. It was not possible to use double tuned RF amplifiers in this receiver.

There are problems of instabitity, insufficient adjacent - frequency rejection and bandwidth variation. These problems can be solved by the use of superheterodyne receiver.

## Superheterodye receiver

The radio waves, propagated through the space, reaches the receiving antenna and induces small e.m.f ( $\mu$ V) in it. This small voltage is fed to the radio receiver. The radio waves voltage is *amplified* and the message signal is separated from the radio frequency signal by the process of demodulation. The detected signal is amplified and fed to the loud speaker which reproduces the orginal sound waves. The block diagram of the *superheterodye radio receiver* is shown in figure.

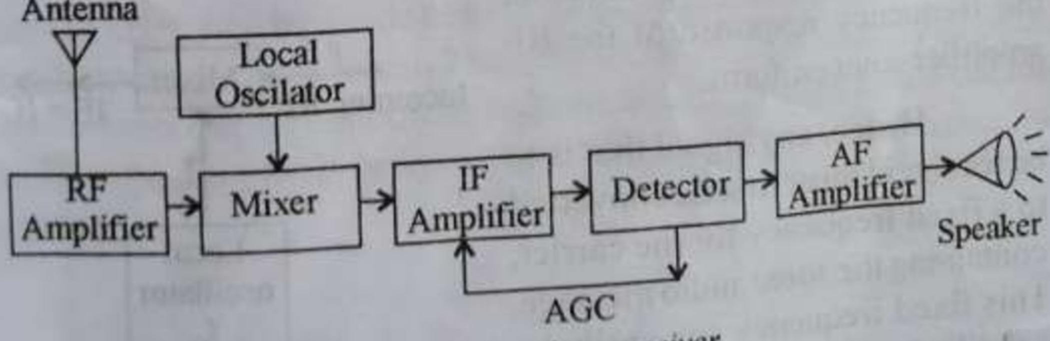



*Fig. 36. Superhetrodyne receiver*

The important units of the radio receiver are the antenna, RF amplifier, a local oscillator, mixer stage, IF amplifier, detector, AF amplifier and loud speaker.

(i) RF stage: The radio waves from various transmitting stations are intercepted by the receiving antenna. The antenna changes the radio signals (electromagnetic waves) into electric voltage signals.

The antenna is connected to the primary of the input transformer (fig.). The secondary coil of the transformer and a variable capacitor, connected parallel to each other form a parallel resonant circuit. The frequency of this circuit is $f = \frac{1}{2\pi\sqrt{LC}}$ .

By adjusting the capacitor, the circuit is *tuned* to select one radio station (of frequency $f$) at a time. This part of the receiver is known as the selector circuit. The weak signal picked up here is amplified by RF amplifier to a desired voltage level. The selector circuit and the RF amplifier form the RF stage of the receiver. The amplified radio sinal is fed to the next stage, known as mixer stage.

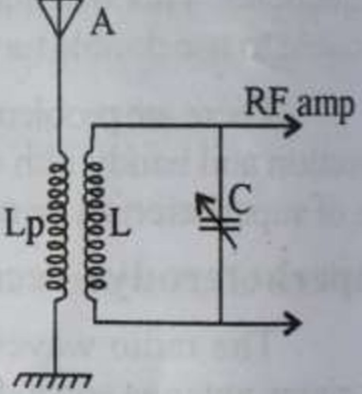



*Fig. 37*

(ii) *Mixer stage and 'Intermediate frequency'* The signal output from the RF amplifier has to be further amplified using more amplifiers, before detection. The amplification is not uniform for all the signals selected from various stations (of various carrier frequencies). This is because, the frequency response of the RF amplifier is not uniform.

Hence, any signal that is to be receiveid must first be converted to a fixed frequency for the carrier, containing the *same* audio message. This fixed frequency can easily be amplified using tuned amplifiers. The

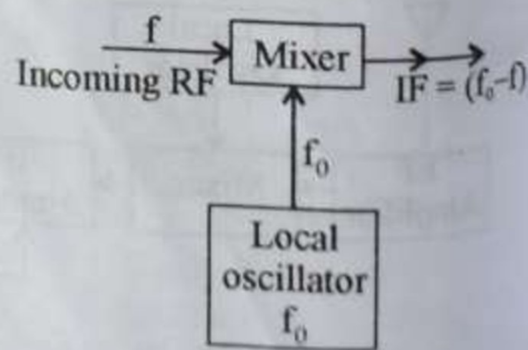



*Fig. 38*

fixed carrier frequency is called the Intermediate Frequency (IF) and it has a value of 455 kHz for AM radio waves. It is 10.7 MHz for FM waves. For television broadcast, IF = 44 MHz.

The IF is produced by the mixer stage. The mixer has two inputs and one output (fig. ). The output of the RF stage is connected as one input. The other input is from a local oscillator, which produces oscillations of radio frequency. The two frequencies mix together and the difference frequency (beat frequency) is drawn as the output of the mixer stage. The output is the intermediate frequency (IF). If $f$ is the frequency of the selected station and $f_o$ is the local oscillator frequency,

$$IF = (f_o - f)$$

$f_o$ is always kept greater than $f$ by an amount. I.F = 455 kHz.

To secure this condition, the tuning capacitor of the R.F stage and that of the local oscillator are ganged. As a result, by turning a single knob, the two units are simulateously tuned and IF is produced. Mixing the frequencies $f_o$ and $f$ to obtain IF = 455 kHz is known as superheterodyning. The term superterodyne refers to creating a beat frequency that is lower than the original signal. Superheterodying process raduces the signal frequency prior to processing.

The signal at this intermediate frequency contains the same modulation as the original carrier and it has to be amplified further by the IF amplifier stage.

(iii) *IF amplifier stage* This stage contains two or more I.F amplifiers, which are transformer coupled (fig. ). IF amplifiers are tuned amplifiers (for 455 kHz), which provide most of the gain necessary and the bandwidth requirement of the receiver. The selectivity and the sensitivity of the superhet receiver are uniform throughout its tuning range. This is because the IF amplifier gain is *independent* of frequency to which the receiver is tuned.

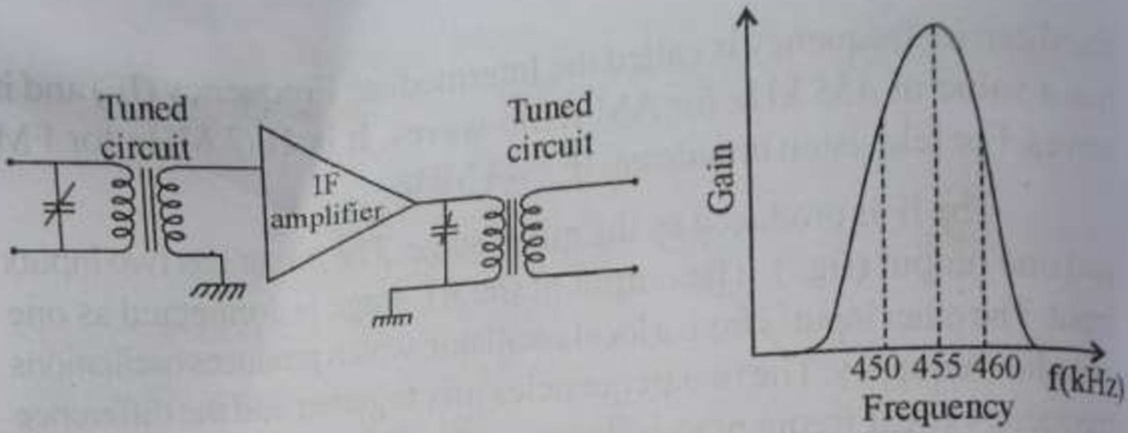



*Fig. 39. IF amplifier stage*

Having secured sufficient gain and the selectivity, the output from the IF stage is fed to the detector stage.

(iv) *Detector* In this stage demodulation occurs. The diode detector is most commonly used for AM demodulation since the distortion is less. Here audio signal is separated from the IF carrier. The audio signal is, however, weak and it is fed to the AF stage.

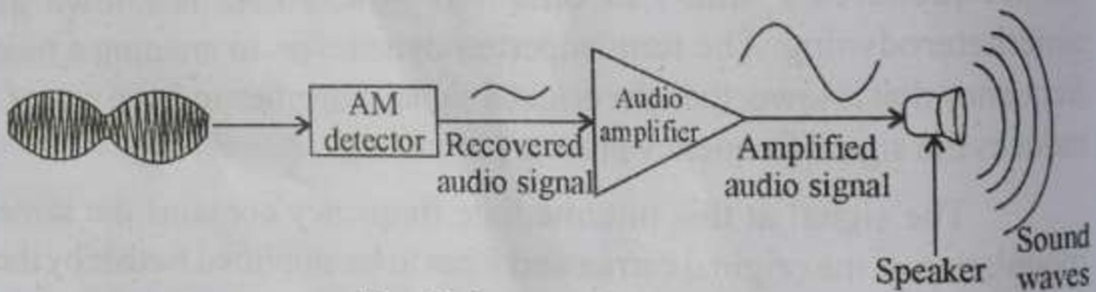



*Fig. 40. Detector stage*

*(v) AF stage* The AF stage consists of AF voltage amplifier and power amplifier, coupled to the loud speaker or display device (fig.). The audio signal is voltage-strengthened using the AF voltage amplifier in order to drive the power amplifier. The signal is power-amplified until it is strong enough to drive the loud speaker. There must be proper impedance matching between power amplifier and speaker. The loud speaker, as a transducer, converts the audio electrical signal into sound waves, corresponding to the original sound at the transmitting station. The IC LM 386 audio amplifier is an example of a low-power amplifier that is capable of providing several hundred milliwatts to an 8Ω speaker.

(vi) *Automatic Gain Control* (AGC) In a radio receiver, the incoming signal strength may vary due to different reasons such as fading (variation of intensity of radio signal). Accordingly the volume of the output sound may suddenly change (the output not uniform). The signal strength may also vary appreciably when the receiver is tuned across the band (different stations). The purpose of AGC is to maintain a relatively constant output from the receiver.

The AGC stage develops a control voltage, bassed on the strength of the signal reaching the detector. This control voltage is fed back to the IF stage and this in turn, adjusts the gain of the first IF amplifier. Thus, due to suitable feedback, AGC keeps the output from the speaker constant.

***Characteristics of a receiver (Receiver Parameters)***

The characteristics of a good radio receiver are -

(i) selectivity (ii) sensitivity, and (iii) fidelity

A good radio receiver responds only to the station of which it is turned at the same time it rejects signals from other stations. This property is known as the *selectivity* of the receiver i.e., selectivity is the ability of the receiver to differentiate one station signal from the other. In other words, selectivity is the ability to reject unwanted frequencies. The *selectivity* depends upon tuned LC circuits used in RF and IF stages. The bandwidth should be narrow for better selectivity. Hence Q of the coil should be high.

Again, a good receiver picks up even weak signal of the desired radio station and is capable of producing the signal output of sufficient strength. This property is known as the *sensitivity* of the receiver. The sensitivity is mainly determined by the gain of the IF amplifier. It is defined in terms of the voltage that must be applied to the receiver input terminals to give a standard output power, measured at the output terminals. Sensitivity can be improved by reducing noise level and bandwidth of the receiver. Sensitivity is expressed in microvolt or in decibels.

*Fidelity* of a receiver is its ability to reproduce the *full range* of AF signal, (in a concert), present in the modulated wave without any distortion ie, It is the ability of a receiver to produce the exact replica of the transmitted signal. A good fidelity requires wide band of frequency to be amplified. Hence for good fidelity, more bandwidth of RF and IF stages are required.

*Advantages of having intermediate frequency (IF) in a superhetrodyne receiver*

1. The IF amplifier is tuned to amplify only one frequency namely 455 kHz. The carrier wave of frequencies of different stations are all brought to this single frequency 455 kHz, which still contains the signal. It is easier to amplify this single frequency using a tuned amplifier rather than amplifying a wide range of frequencies (of different stations whose waves are intercepted by the aerial).

2. The IF is less than the radio frequency (SW or MW). Hence the amplification will be more stable in the amplifier.

3. Power loss in the tuned circuits, because of the use of IF transformer (IFT), is low at the IF.

4. The selectivity of the radio receiver is improved because of the use of IF.

Hence the superhetrodyne receiver is more efficient and cheaper than a straight receiver.

*Tracking in super hetrodyne receiver*

**Communication receiver: Double frequency conversion A.M. receiver**

The block diagram of a communicatioin receiver is given in figure.

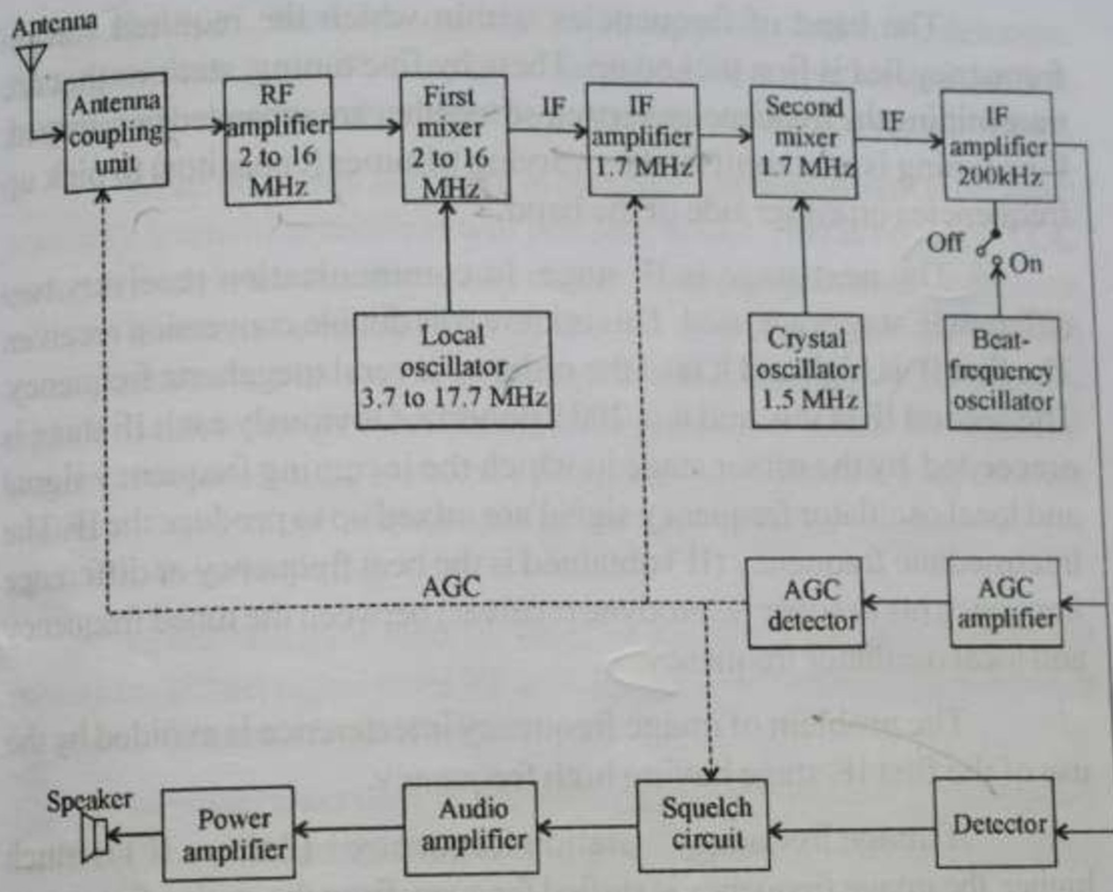



*Fig. 42. Communication receiver block diagram*

The main function of a communication receiver is to receive the signals used for communication between stations rather than the entertainment (home radio or television). The communication receiver is meant for low and high frequency reception much better than the home radio. The communication receiver finds application in the receiver system used by the police, ambulance, coast radio stations to receive signals from ships and high frequency impedance bridges (microwave parameters measurement) etc.

The electromagnetic signals in space intercept the antenna, which converts the e.m. signals into the electrical signals. The antenna coupling unit consists of a tuning circuit (*L* and *C* in parallel) and it helps us to pick up RF signal from a required station. Separate tuning circuits are used for tuning frequencies of different ranges. All these ranges will cover a band of frequencies.

The band of frequencies within which the required statio frequency lies is first picked up. Then, by fine tuning, stations that ar transmitting the frequencies very close together are separated or resolve Fine tuning is accomplished by varying trimmer (capacitor) to pick u frequencies on either side of the band.

The next stage is IF stage. In communication receivers tw different IF stages are used. This is known as double conversion receive The first IF is high and it is of the order of several megahertz frequenc The second IF is low and it is 200 kilohertz. Obviously each IF stage preceeded by the mixer stage in which the incoming frequency sign and local oscillator frequency signal are mixed up to produce the IF. Th intermediate frequency (IF) obtained is the beat frequency or differen frequency (as in super heterodyne receiver) between the tuned frequen and local oscillator frequency.

The problem of image frequency interference is avoided by th use of the first IF. stage having high frequency.

If image frequency = station frequency + (2 times IF) is muc higher, the image frequency is shifted far away from the station frequenc

The second IF stage of low frequency enables sharp selectivi and hence good rejection of adjacent channel frequencies. Each IF sign is amplified with the help of IF amplifiers.

The output of the second IF. stage is fed to the detector, whic demodulates and separates the AF. signal from the carrier. The AF. sign is amplified using audio voltage amplifier. Then it is power amplified an fed to the output display unit or a loud speaker, which faithfully reproduce the original signal or message.

As the incoming singnal strength increases, the received outpu voltage in the output stage also increases linearly if no automatic gai control (AGC) is used. The AGC enables constant output power leve even if suddenly changing incoming signal strength exceeds a definit limit. The limit is set by the AGC circuit. The AGC does not get its require bias voltage till a pre-determined input signal has been reached. It begin to function after the signal reaches the predetermined value. For this reason, the AGC, section consists of AGC amplifier and AGC detector. The signal from the AGC detector is fed to the RF amplifier and the first stage IF. The detector offers negative feed back to the above two stages.

When no carrier is present at the input, (no transmission between stations), a sensitive receiver will produce noise. This is because AGC does not work in the absence of any carrier. The noise produces unpleasant effects. This can be minimized by the use of a circuit called *squelch circuit* (also called muting or quieting). In the absence of AGC detector signal, squelch circuit controls the audio amplifier of the receiver and thereby unwanted noise is cut-off. Thus, the squelch enables the recivers output to remain cut-off, whenever the carrier is not present i.e., the station is not transmitting.

Communication receiver requires additional local oscillators and mixers to convert signal from RF to IF before conversion to base band. It increases the cost of receiver.

In modern version, main microprocessor control functions are built into the communication receiver by large scale integration (LSI) process.

***Questions***

1. Compare the various AM systems.
2. Explain Quadrature amplitude modulation. Describe the working of QAM with block diagrams.
3. What is demodulation? Describe the construction and /principle of AM detector.
4. Describe various parts of AM receiver using a block diagram.
5. Explain the working of various units of tuned radio frequency (TRF receiver with a block diagram.
6. What is meant by superheterodyning? Give the block diagram of superheterodyne radio receiver and explain the function and advantage of each unit in it.
7. Explain the pı cess of tracking. What is image frequency rejection

a receiver set? Explain why local oscilator frequency is kept greater than station frequency in a superheterodyne set?

8. Explain the parameters of radio receiver set.

9. Describe the principle and working of double frequency conversion AM receiver.

***Objective type Qestions***

1. Demodulation is done in __________
   (a) receiving antenna (b) transmitter
   (c) radio receiver (d) transmitting antenna.

2. A superheterodyne receiver with IF 450 kHz is tuned to a signal at 1200 kHz. The image frequency is
   (a) 2100 kHz (b) 900 kHz (c) 750 kHz (d) 1650 kHz

3. Sensitivity of a receiver depends upon the receiver's over all
   (a) bandwidth (b) selectivity
   (c) noise response (d) gain

4. A mixer in a superheterodyne receiver has a signal input 50 MHz frequency and a local oscillator freqency of 59 MHz. The IF is
   (a) 9 MHz (b) 50 MHz (c) 59 MHz (d) 109 MHz

5. A high Q tuned circuit will permit a radio receiver to have high
   (a) fidelity (b) frequency range
   (c) sensitivity (d) selectivity

6. In TV transmission, pitcture signal is _________ modulated
   (a) frequency (b) phase
   (c) amplitude (d) none of the above

7. In a radio receiver, noise is generally developed at
   (a) IF stage (b) receiving antenna
   (c) audio stage (d) RF stage.

8. The signal voltage induced in the aerial of a radio receiver is of the order of

(a) mV (b) μ V (c) V (d) none of the above

9. Superheterodyne principle refers to

(a) using a large number of amplifier stages

(b) using a push - pull circuit

(c) obtaining lower fixed intermediate frequency

(d) none of the above

10. If a radio receiver amplifies all the signal frequencies equally well, it is said to have high ______

(a) sensitivity (b) selectivity (c) distortion (d) fidelity

11. Most of the amplification in a superheterodyne receiver occurs at __________ stage

(a) IF (b) RF amplifier (c) audio amplifier (d) detector

12. The letters AVC stands for ______

(a) audio voltage control (b) abrupt voltage control

(c) automatic volume control (d) automatic voltage control

13. The superheterodyne principle provides selectivity at ______ stage

(a) RF (b) IF (c) audio (d) before RF

14. In superhet receiver, the input at mixer stage is ________

(a) IF and RF (b) RF and AF

(c) IF and AF (d) RF and local oscillator signal

15. In a superhet receiver, the difference frequency is chosen as the IF rather than the sum frequency because

(a) difference frequency is closer to oscillator frequency

(b) lower frequencies are easier to amplify

(c) only the difference frequency can be modulated

(d) None of the above

16. The diode detector in an AM radio receiver is usually found ______

(a) before the first RF stage

(b) after the first RF stage

(c) after serveral stages of amplification

(d) none of the above

17. In a TRF receiver, the RF and detection stages are tuned to ______

(a) radio frequency (b) IF

(c) audio frequency (d) none of the above

18. The intermediate frequency in a standard AM receiver is

(a) 455Hz (b) 455kHz (c) 4.55 MHz

(d) none of the above

19. The function of an AM detector circuit is to

(a) rectify the input signal

(b) discard the carrier

(c) provide audio signal

(d) all of the above

20. Which of the following should be used in order to prevent overloading on the last IF amplifier in a receiver?

(a) variable selectivity (b) variable sensitivity

(c) double conversion (d) squelch

21. When the station tuned by a receiver is not transmitting, the receiver's output can remain cut-off. This is due to the action of

(a) AGC (b) squelch (c) LSI (d) discriminator

*Answers*

| | | | | | |
|---|---|---|---|---|---|
| 1. (c) | 2. (a) | 3. (d) | 4. (a) | 5. (d) | 6. (c) |
| 7. (d) | 8. (b) | 9. (c) | 10. (d) | 11. (a) | 12. (c) |
| 13. (b) | 14. (d) | 15. (b) | 16. (c) | 17. (a) | 18. (b) |
| 19. (d) | 20. (b) | 21. (b) | | | |

# UNIT III-ANGLE MODULATION – TRANSMISSION

Introduction – Frequency modulation – Phase modulation – Phase deviation and modulation index – Multitone modulation – Transmission band width of FM – conversion of PM to FM or frequency modulator – conversion of FM to PM / phase modulators – commercial broadcast FM – phasor representation of an FM and PM – average power of an AM/FM wave – generation of FM – direct method of FM generation – reactance tube modulator – indirect method of FM wave generation – FM transmitters – indirect method – Comparison of AM and FM.

## Angle modulation

It is the process by which the frequency or the phase of the carrier signal is changed according to the instantaneous amplitude of the modulating (message) signal. The amplitude of the carrier remains constant in this process.

The angle modulation can be classified into two types: (1) Frequency modulation (2) Phase modulation.

## Frequency Modulation

Frequency modulation is the process by which the frequency of a high frequency carrier wave is changed in proportion to the instantaneous amplitude of the modulation (A.F.) signal. In this process, the amplitude of the carrier wave is not disturbed and it remains constant.

The frequency modulated wave is at its highest and lowest frequency when the A.F. signal is at its maximum positive and negative points. When the amplitude of the AF signal is larger, the change of FM signal frequency from the centre frequency is larger. When the amplitude of AF is zero, the FM signal is at its centre frequency.

The information about the frequencies of the A.F. signal (modulating signal) is indicated by *the rate* at which the frequency swings are produced in the F.M. signal.

The degree of modulation in FM is indicated by a quantity, called the modulation index. It is defined as the ratio of the frequency deviation to the audio signal frequency.

i.e., modulation index $M = \frac{f_d}{f_m}$

Thus, in the case of frequency modulated wave

1. the amount of frequency deviation indicates the amplitude of the audio signal and

2. the *rate* of frequency deviation indicates the frequency of the audio signal.

The total power of an FM wave is not changed due to modulation. When side band power increases, the carrier power decreases correspondingly. This is different from the case of AM where the total power increases with the depth of modulation.

*Advantages of frequency modulation*

1. The FM reception is free from noise and interference. Noise is a form of amplitude variation and FM receiver will reject such signals. For this reason FM finds application in intercommunication between moving vehicles.

2. The operating range is large since the efficiency is high.

3. High fidelity reception. This means faithful reproduction of the message.

4. FM stations operate over a broadcast band of high frequencies from 88 to 108 mega hertz. The stations are spaced every 0.2 mega hertz from the centre frequency of the carrier wave. So, a larger number of broadcast stations can be accommodated.

5. The adjacent channel selectivity is greater.

6. There is no chance of the image frequency interference in reception with F.M., since the intermediate frequency (I.F) is very high (= 10.7 mega hertz)

***Expression for frequency modulated voltage***

In frequency modulation, the frequency of the carrier wave is altered according to the amplitude of the modulating A.F. signal voltage. Let $\omega_C$ be angular frequency of the carrier wave and $V_C$ be the its amplitude. Then, the instantaneous voltage of the carrier wave is given by

$$\upsilon_C = V_C \sin(\omega_C t + \theta) \qquad \text{where } \theta \text{ is the intial phase.}$$

The phase angle of the carrier is $\phi = \omega_C t + \theta$

$$\therefore \quad \frac{d\phi}{dt} = \omega_C$$

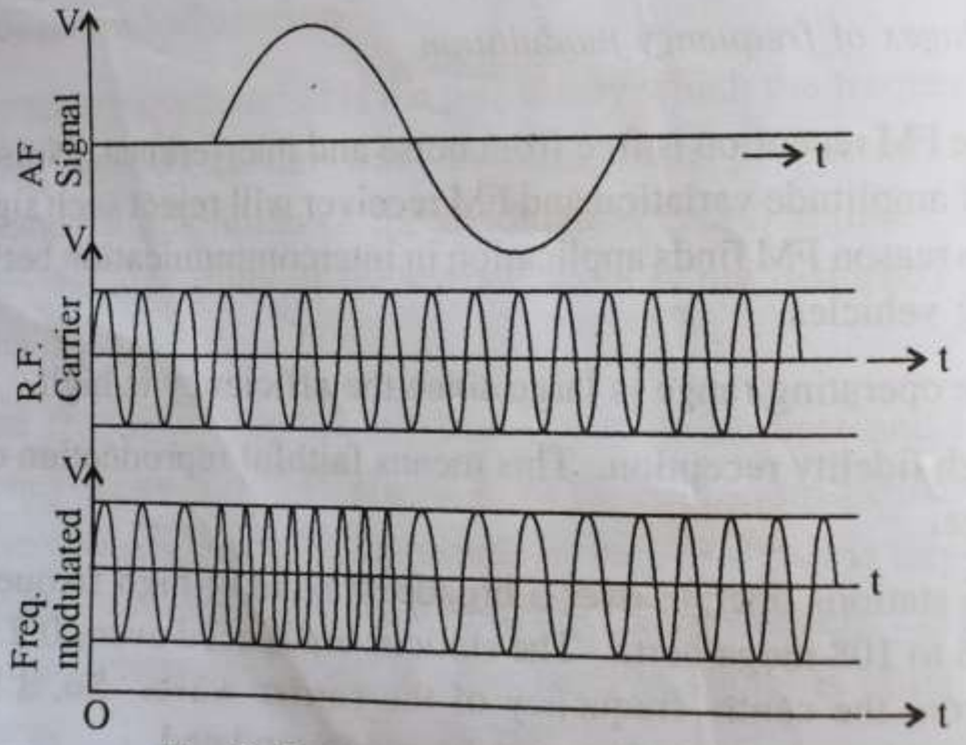



*Fig. 43 Frequency modulated voltage*

Let the modulating signal voltage at any instant be

$$\upsilon_m = V_m \cos \omega t$$

where $\omega_m$ is the angular frequency of the modulating voltage and $V_m$ is its amplitude. When frequency modulation is effected, the carrier no longer remains constant in frequency, but it varies with time according to the instantaneous value of the modulating voltage. The frequency of the carrier voltage after frequency modulation is given by

$$\omega = \omega_C + k.\upsilon_m \quad \text{at any instant}$$

$$\omega = \omega_C + k[V_m \cos \omega_m t] \qquad \text{... ... (1)}$$

Here $k$ denotes the constant of proportionality. To obtain phase angle $\phi$ of the modulated carrier, we integrate the above expression

$$\phi = \int \omega \, dt = \int \{\omega_C + k V_m \cos \omega_m t ]\} \, dt$$

$$\phi = \omega_C t + k V_m \frac{\sin \omega_m t}{\omega_m} + \phi_1$$

where $\phi_i$ is the constant of integration. It represents a constant phase angle, which may be neglected in the analysis of FM wave.

$$\therefore \quad \phi = \omega_c t + \frac{k.V_m}{\omega_m} \sin \omega_m t$$

With this value of $\phi$, we can write the instantaneous voltage of the F.M wave as

$$\upsilon = V_C \sin \phi$$

$$\upsilon = V_C \sin\left[\omega_C t + \frac{k.V_m}{\omega_m} \sin \omega_m t)\right] \qquad \text{... ... (2)}$$

**FM spectrum**

The equation representing FM wave is

$$v(t) = \sin[\omega_C t + \left(\frac{kV_m}{\omega_m}\right) \sin \omega_m t]$$

This is a sine of sine function.

Let $\omega_C t = A$ and $\frac{kV_m}{\omega_m} \sin \omega_m t = B$

$$\therefore\ v(t) = V_c \sin(A + B)$$
$$= V_c [\sin A \cos B + \cos A \sin B]$$

$$v(t) = V_C \left[\sin \omega_c t \cos\left(\frac{kV_m}{\omega_m} \sin \omega_m t\right)\right.$$
$$\left. + \cos \omega_C t \sin\left(\frac{kV_m}{\omega_m} \sin \omega_m t\right)\right]$$

The simplification involves the special function: Bessel functions of the first kind. The result is available in Table form or graphical form.

The FM spectrum has several components. The amplitude of the carrier and that of sidebands are known from the corresponding Bessel functions, $J_n(m_f)$, where $n$ is the order and $m_f$ is the argument of the Bessel function.

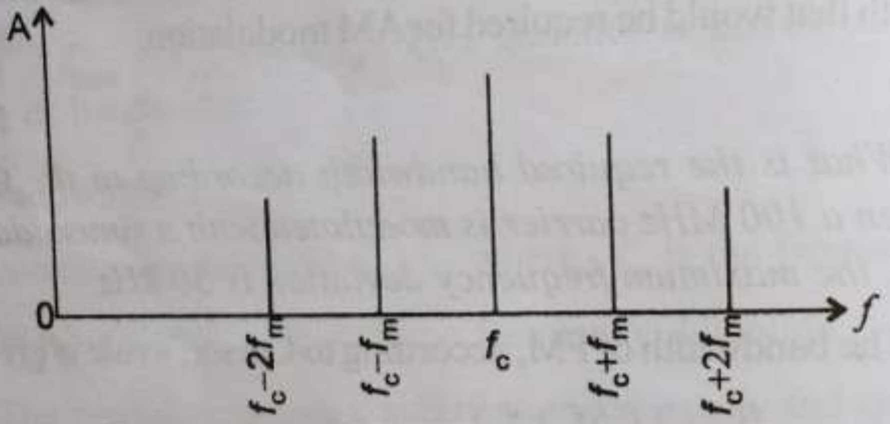



*Fig. 44. Frequency modulation - bandwidth spectrum, sidebands*

*Discussion*

1. FM has infinite number of side bands and the carrier. They are separated from carrier by $\omega_m$, $2\omega_m$, $3\omega_m$ etc.

2. The modulation index will determine how many sideband components have significant amplitude.

3. The sideband distribution in the spectrum is symmetrioal about the carrier frequency.

4. If $M < 1$, The FM carrier has only one pair of sidebands. This is equivalent to narrow band FM.

5. The amplitude of FM remains unchanged. Hence the power of FM is the same as that of unmodulated carrier.

6. The total power of FM signal depends upon the power of the unmodulated carrier unlike the AM where the power depends on the modulation index.

7. The modulation index signifies the strength and quality of the transmitted signal when $M$ is large, the audio signal will be stronger and clearer, the bandwidth is larger and the number of effective sideband pairs increases. Modulation index is normally limited to a value between 1 and 5.

$$M = \frac{k \cdot V_m}{2\pi f_m} \quad \text{or} \quad M = \frac{k\, V_m}{\omega_m}$$

## 2. Phase Modulation

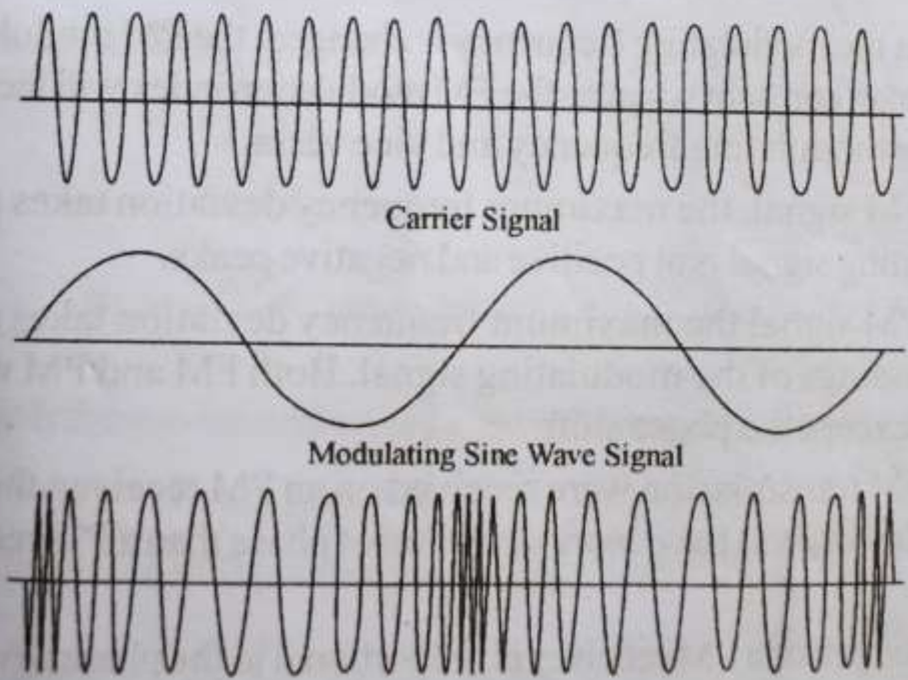



*Fig. 45 A. Phase modulated Signal*

It is the process by which the phase of the carrier signal is change according to the instantaneous amplitude of the message signal.

Let the modulating signal be $v_m(t) = V_m \cos \omega_m t$.

and the carrier signal be $v_c(t) = V_c \sin(\omega_c t + \theta)$

The phase angle of the carrier is changed according to the amplitude of the message signal.

$$\theta \propto V_m \cos \omega_m t \text{...... (instantaneous amplitude)}$$

$$\theta = KV_m \cos \omega_m t$$

After phase modulation, the instantaneous voltage will be

$$v_{pm}(t) = V_C \sin(\omega_C t + \theta)$$

$$= V_C \sin(\omega_C t + KV_m \cos t\, \omega_m t$$

$$v_{pm}(t) = V_C \sin(\omega_C t + m_p \cos \omega_m t)$$

where $m_p$ is the modulation index for $PM$. $m_p = KV_m$

This represents phase modulated signal. In phase modulation, the

modulation index $\propto$ amplitude of modulating voltage.

In frequency modulation

$$\text{the modulation index} \propto \frac{1}{\text{modulating frequency.}}$$

When the modulating frequency is changed, the PM modulation index will remian constant whereas the FM modulation index will increase for decreasing modulating frequency and vice versa.

For FM signal, the maximum frequency deviation takes place when modulating signal is at positive and negative peaks.

For PM signal the maximum frequency deviation takes place near zero crossings of the modulating signal. Both FM and PM waves are indential except the phase shift.

If a FM transmission were received on an PM receiver, the bass frequencies (lower) will have *more deviation* of phase than a PM receiver would give.

The output of a PM receiver is proportional to the phase deviation (or the modulation index).

So, the output signal would appear bass booted. If PM is received by an FM system, the output signal ...

... lacking in bass.

***Conversion of PM to FM***

Frequency modulated wave (FM) can be obtained from PM. This is achieved by integrating the modulating signal before applying to the phase modulator, as shown in the figure.

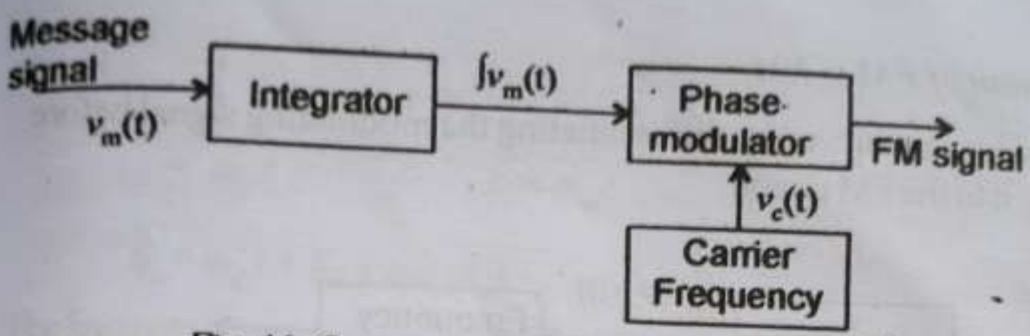



*Fig. 46. Conversion of PM to FM*

Let the modulating signal be

$$v_m(t) = V_m \cos \omega_m t$$

After integration, we have

$$e_m(t) = \int V_m \cos \omega_m t$$

$$e_m(t) = \frac{V_m}{\omega_m} \sin \omega_m t \qquad \text{.... .... ... (1)}$$

In phase modulation,

$$\theta \propto e_m(t)$$

$$\theta = K e_m(t)$$

$$\theta = \frac{K V_m}{\omega_m} \sin \omega_m t \qquad \text{.... ... ... (2)}$$

As regards the modulated voltage, its instantaneous value is given by

$$v(t) = V_C \sin(\omega_c t + \theta)$$

Putting the value of θ from equation (1)

$$v(t) = V_C \sin\left(\omega_C t + \frac{K V_m}{\omega_m} \sin \omega_m t\right)$$

But $\frac{K V_m}{\omega_m} = \frac{2\pi\Delta f}{\omega_m} = \frac{2\pi\Delta f}{2\pi\Delta f_m} = \frac{f}{f_m} = m$, which

is the modulation index in FM namely $m_f$.

$\therefore v(t) = V_C \sin[\omega_C t + m_f \sin \omega_m t]$

This is the expresion for FM. Thus PM is converted to FM.

***Conversion of FM to PM***

This is achieved by differentiating the modulating signal before applying it to the FM circuit.

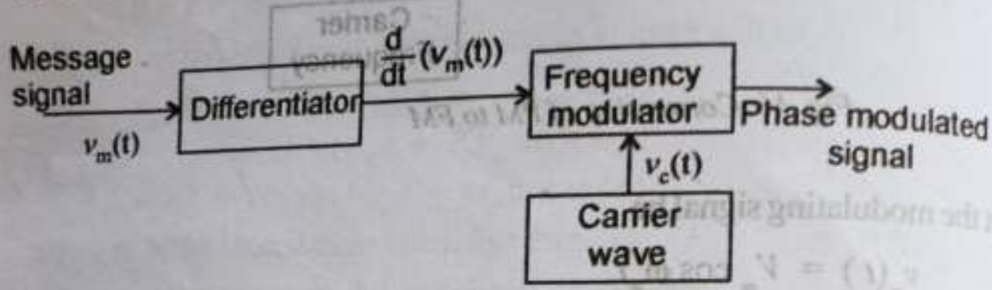



*Fig. 47. Conversion of FM to PM*

We start with the modulating signal $v_m(t)$ and differentiate it.

Let the modulating signal be $v_m(t) = V_m \cos \omega_m t$

$$\frac{dv_m(t)}{dt} = -V_m\omega_m \sin \omega_m t \quad \text{....... (1)}$$

In frequency modulation,

Angular frequency deviation $\propto$ rate of change modulating signal voltage.

$$\propto \frac{dv_m(t)}{dt} \text{ (instantaneous values)}$$

$$= \frac{K.dv_m(t)}{dt} \text{ where } K \text{ is proportionality constant.}$$

$\therefore$ The instantaneous angular frequency of FM is

$$\omega_i = \omega_C + \text{angular frequency deviation}$$

$$\omega_i = \omega_C + \frac{K.dV_m(t)}{dt} \quad \text{............... (2)}$$

Substituting equation (1) in eqn (2),

$$\omega_i = \omega_C - K\omega_m V_m \sin \omega_m t$$

We know that the instantaneous phase angle of the frequency modulated signal is

$$\phi_i = \int \omega\, dt$$

$$= \int (\omega_c - K\omega_m V_m \sin \omega_m t)\, dt$$

$$= \omega_C t + \frac{K\omega_m V_m}{\omega_m} \cos \omega_m t$$

$$\phi_i = \omega_C t + K_m v \cos \omega_m t$$

The instantaneous *voltage* after modulation is given by

$$v(t) = V_C \sin \phi_i$$

$$v(t) = V_C \sin [\omega_C + KV_m \cos \omega_m t]$$

$$v_{pm}(t) = V_C \sin [\omega_C t + M \cos \omega_m t]$$

where $KV_m = M$, the modulation index in PM.

This is the expression for phase modulated wave.

In the conversion, no new frequecies are generated since the process of integration and differentiation are linear.

**F.M. Transmitter**

The transistor reactance modulator shown in the diagram can be used as an FM transmitter. The circuit is basically a CE amplifier with a radio frequency choke (RFC) acting as the load. The RFC prevents the voltage fluctuation at the collector affecting the power supply. The resistance $R_1$ provides the feed back bias to the transistor amplifier.

The carrier voltage from an R.F oscillator is applied to the base of the transistor through the capacitor $C_2$. The audio signal input from microphone is fed to the base of transistor through another R.F. choke.

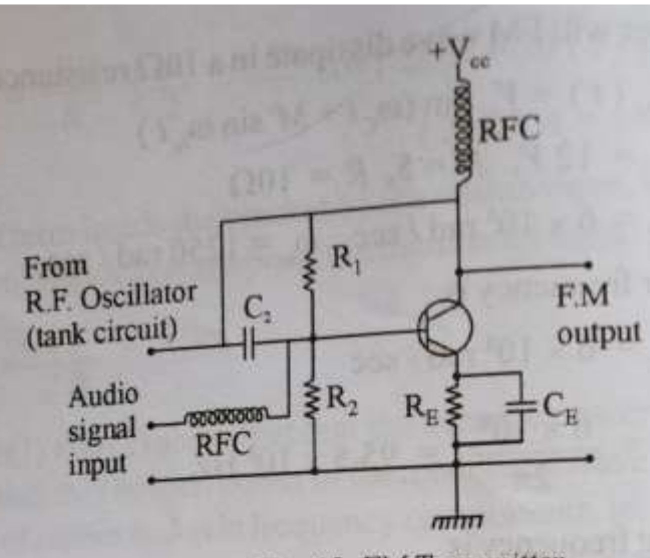



*Fig. 45. FM Transmitter*

For the given circuit parameters, there is a particular gain of the amplifier. The carrier wave is produced by a R.F. oscillator whose frequency is determined by its tank circuit parameters and the capacitance presented by the transistor circuit which depends on $C_2$, $h_{ie}$, $h_{fe}$ and $R_2$ values.

The reactance ($1/C\omega$) offered by the circuit to the tank circuit of the oscillator (carrier) varies according to the audio signal input voltage. Hence the frequency of the carrier wave is changed according to the strength of the audio signal at each instant. This is amplified by the transistor. The amplified frequency modulated output is made available at the collector of the transistor, with respect to the ground. The F.M. wave can be transmitted by coupling the output to an antenna.

**Generation of FM**

**A frequency modulation generator should produce a variable output frequency. The variation should be proportional to the instantaneous amplitude of the modulating voltage. This is the basic principle of FM generation. There are various methods of providing voltage variable rasistance, which can be connected to the tank circuit of an oscillator that produces FM.**

***Direct method of FM generation***

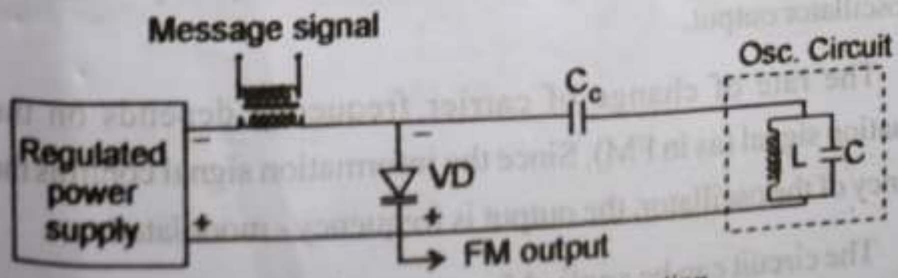



*Fig. 48. Basic circuit for FM generation*

Figure shows the basic circuit for FM generation. The varactor diode (VD) is connected across the resonant circuit of an LC oscillator through a coupling capacitor $C_c$ of fairly large value. Reactance of the capacitor $X_c = 1/C\omega$. As $C_c$ is large, $X_c$ will be small for the operating frequencies ($\omega$). This coupling capacitor also isolates the diode from the oscillator.

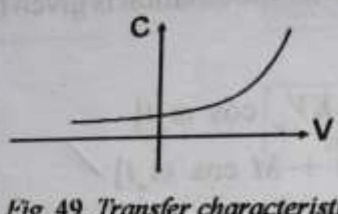



*Fig. 49. Transfer characteristics of varactor diode*

Varactor is a self variable capacitor due to variation of depletion region width ($d$) at the p-n junction. The dc bias to the varactor diode is regulated in such a way that oscillator frequency is not affected by varactor supply fluctuations.

The modulating signal is applied in series with this regulated supply. At any instant the effective bias (voltage) to the varactor equals the algebraic sum of the bias voltage plus the instantaneous values of the modulating signal. As a result, the capacitance of varactor changes with the amplitude of the modulating signal. This results in frequency modulation of the oscillator output.

The rate of change of carrier frequency depends on the information signal (as in FM). Since the information signal controls the frequency of the oscillator, the output is frequency - modulated.

The circuit can be applied for automatic frequency control and for remote tuning.



### Reactance tube modulator

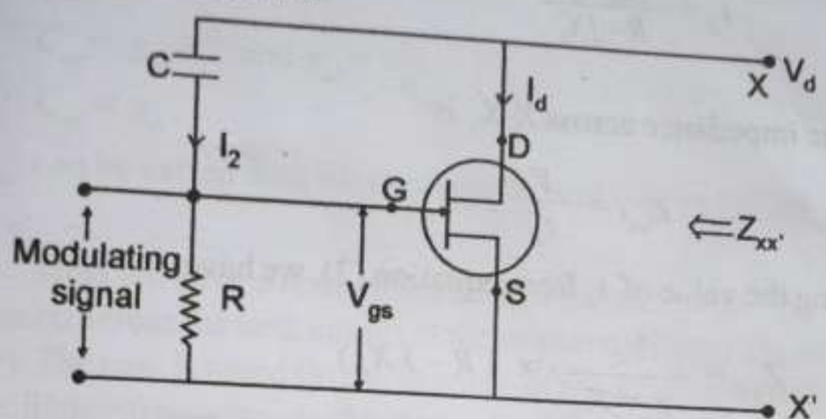



*Fig. 50. Reactance tube modulator*

Figure shows the reactance tube modulator with FET. The Hartley oscillator to produce carrier frequency (not shown in diagram) is connected across XX' of the reactance tube circuit. Our expectation is that the device should behave like 'reactance'. In the circuit the reactance appearing between X and X' corresponds to the reactance between the drain D and the source S of the field effect transistor (FET). The value of the terminal reactance is proportional to the transconductance of the FET, which in turn, can be made to depend on the gate bias and its variations (due to the modulating signal) at the third terminal, the Gate.

In order to make the impedance across the terminals X and X' to be pure reactance (without resistance part), the reactance modulator should satisfy the following two requirements.

1. The biasing current $i_2$ must be negligible, when compared to $i_d$. That is, the impedance of the bias network must be large enough.

2. The gate - to - drain impedance must be greater than the gate-to-source impedance and $X_C >> R$ (ie. $R$ is negligibly small for the gate voltage to be 90° out of phase with the drain current).

Thus we have a *voltage variable reactance* with FET circuit. It is placed across the tank circuit of the master oscillator (as described earlier). The tank is tuned (in the absence of modulating signal), so that the oscillator frequency = the desired carrier frequency. When the modulating signal is on, the reactance is changed with the modulating voltage. It increases as the modulating voltage increases positively and going the other way, when the modulating signal goes negative. The larger the departure of the modulating signal from zero, the larger will be the reactance variation and hence the frequency variation. Thus, the variable reactance modulator generates FM.

High power FM generation is possible to use direct method for generation of FM. This is an advantage of this method.

*Disadvantage of Direct method of FM generation*

1. To produce carrier frequency crystal oscillators meant for high frequency stability could not be used. Their frequency could not be varied as requied in FM.

2. In the direct method of FM generation we have to use the LC oscillator. The LC oscillator frequency is not stable.

3. Hence, it is not possible to use such oscillator for communication or broadcast purpose.

4. Distorted FM signal is generated due to harmonics of modulating signal.

**Indirect method of FM generation**

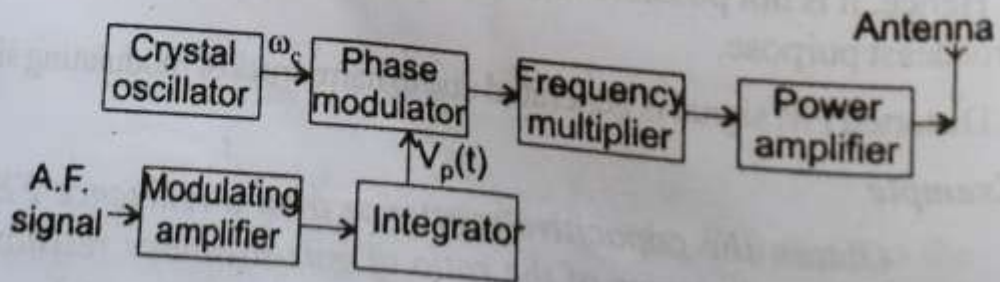



*Fig. 51. Indirect method of FM generation*

The block diagram of indirect method of FM generation is shown above. It comprises of a crystal oscillator, whose output is fed to a phase modulator. The audio signal is integrated (to prevent over modulation) and also applied to the phase modulator.

The resultant wave at the phase modulator output is passed through several stages of frequency multipliers to obtain the desired frequency deviation and also to increase the centre frequency of FM. The signal is then amplified by the power amplifier stage to the required power level.

*Circuit action*

Let the input audio signal be $e(t)\ V_m \sin \omega_m t$. The input to the phase modulator is given by the integrator.

$$v_p(t) = \int V_m \sin \omega_m t \, dt .$$

$$\text{ie. } v_p(t) = -V_m \cos \omega_m t / \omega_m$$

The phase shift produced at the modulator output is given by

$$\theta \propto v_p(t)$$

$\theta = K . v_p(t)$ where $K$ is proportionality constant

$$\theta = -(K v_m / \omega_m) \cos \omega_m t$$

$\therefore$ The instantaneous angular frequency of the wave is

$$\omega_i = \frac{d\theta}{dt}$$

$$= \left[\left(\frac{KV_m}{\omega_m}\right)\omega_m \sin \omega_m t\right]$$

$$\omega_i = KV_m \sin \omega_m t$$

The angular frequency deviation of the $FM = KV_m$. (maximum)

Thus the action of the circuit results in frequency modulation with deviation, proportional to the peak amplitude of the modulating signal.

**Phasor Repressntation of PM**

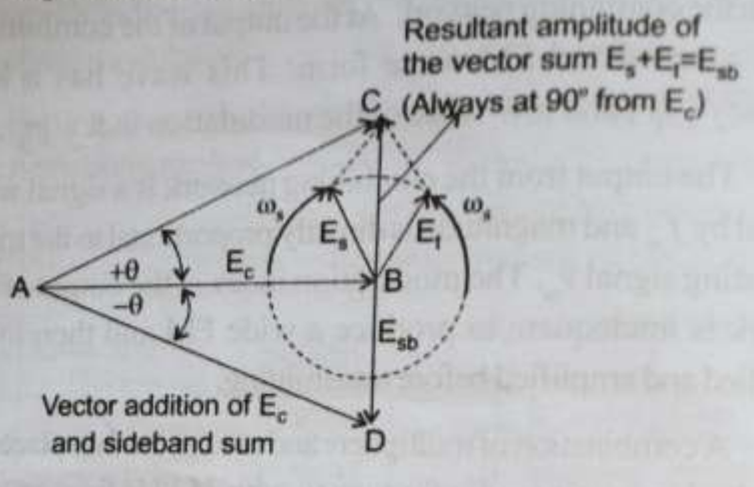



*Fig. 53. Phasor representation of PM*

The PM transmitter phasor diagram is shown in figure. The carrier is labelled as $E_C$ . The lower side band vector $E_S$ rotates in clock-wise direction and the upper side side band rotates in anti-clock-wise direction. The resultant of side bands $E_{Sb}$, varies from $+E_{Sb}$ to $-E_{Sb}$ through 0. The resusltant of side bands is always normal to the carrier vector $E_C$.

As the two side band vectors counter rotate, their resultant $E_{sb}$ will always be 90° from $E_C$, but will change amplitude and polarity from $+E_{Sb}$ to $-E_{Sb}$. The vector addition of $E_{Sb}$ and $E_C$ will form the hypotenuse of the triangle that changes through $\pm\theta$ as the sideband amplitude $E_{sb}$ changes from $+E_{Sb}$ to $-E_{Sb}$. The hypotenuse represents the output voltages ($E_O$) of the adder of the modulator. The angle changes from $+\theta$ through 0, to $-\theta$, the length of the hypotenuse ($E_O$) changes, and since this is the output of the adder, an undesirable amount of amplitude modulation appears at the adder output.

**Phasor representation of FM**

The instantaneous voltage of the $FM$ wave is,

$$v(t) = V_C \sin\left[\omega_C t + \frac{KV_m}{\omega_m} \sin \omega_m t\right]$$

A FM narrow band signal may be represented as shown in figure.

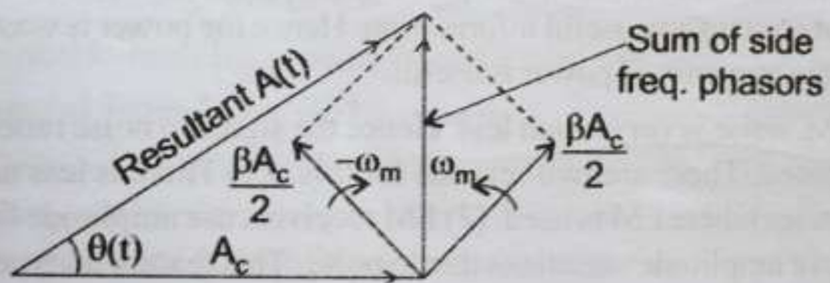



*Fig. 53(a). Phasor diagram for NBFM with tone modulation*

Since the suppressed carrier voltage $v_C$' is 90° out of phase with carrier voltage $v_C$, the upper and lower side bands combine to poroduce a component $v_m$ which is always perperdicular to $v_C$. We have taken the carrier phasor as reference. Comparing the phasor diagram for the Narrow band FM (NBFM) with that of AM signal, we make the following observation.

In the case of AM, the resultant of the side-band phasors is collinear with carrier phasor where as, it is perpendicular to the carrier in NBFM. It is this quadrature relationship between the two of NBFM that produces angle variations resultring corresponding changes in instantaneous frequency $f_i(t)$.

## Comparison of AM and FM

1. In AM system there are three components: The carrier, upper side band and lower side band frequencies. Hence the band width is finite.

But FM system has infinite side bands in addition to a single carrier. Each side band is separated by $f_m$. So the band width is infinite.

2. In AM, the amplitude of modulated wave depends on the modulation index ( $m$ ). But in FM, the amplitude of modulated wave is independent of modulation index ( $M$ ).

3. In AM, increased modulation index increases the side band power and hence it increases the total transmitted power. In FM, the total transmitted power always remains constant but an increase in the modulation index ( $M$ ) increases the bandwidth of the system.

4. In AM most of the transmitted power is used by the carrier. The carrier does not contain any useful information. Hence the power is wasted. In FM all the transmitted power is useful.

5. In F.M. noise is very much less. Hence the signal to noise ratio (S/N) is increased. There are two reasons for this : (1) There is less noise at frequencies where FM is used. (2) FM receivers use amplitude limiters to remove amplitude variations due to noise. This feature does not exist in AM.

6. Due to frequency allocations by CCIR (Internation Radio Consultative Committee) there are guard bands between FM stations. So there is less ajaceent channel interference than in AM.

7. FM system operates in UHF and VHF range of frequencies, where space wave is used for propagation, so that the radius of reception is limited slightly more than the line of sight. It is thus possible to operate several independent FM transmitters on the same frequency with less interterence than that is possible with AM.

## Transmission bandwidth in FM

FM has very large number of side bands. But higher order side bands are not significant. The significant side bands produced in FM may be considered to be an integer, approximately equal to the value of $m_f$ (ie, $n = m_f$ if $m_f >> 1$), where $m_f$ is the arguement of the Bessel function of a given order $n$, which represents the FM spectrum $m_f$ is inversely proportional to the modulating frequency.

The sidebands (of FM spectrum) are at equal distances from the carrier frequency $f_C$ have equal amplitudes. That is the sideband distribution is symmetrical about the carrier frequency. The bandwidth is an integral multiple of the modulating signal frequency.

From the theory of frequency modulation studied earlier,

the FM bandwidth $= 2(\Delta f)$ hetz

$= 2(\Delta\omega)$ radian

An appropriate rule for transmission band width of FM generated by a single tone modulating signal is

$$\text{bandwidth} = 2(\Delta\omega + \omega_m)$$

$$= 2\,\Delta\omega\,(1 + \omega_m / \Delta\omega) \text{ radian}$$

This empirical formula is known as Carson's rule.

## Commercial broadcast FM

Commercial broadcasting (also called private broadcasting) is the broadcasting of television programs and radio programming by privately owned corporate media, as opposed to state sponsorship.

Commercial Radio is a radio station that generates revenue through advertising, also called radio commercials, radio spots or advertising jingles. The advertisements are released by companies which find the commercial radio to be a good medium to take their message to their customers.

Generally commercial radio stations broadcast popular music to build their listener base. They also provide useful information like local traffic alerts, sports. AM is mostly talk radio and FM is mostly music. The AM and FM radio function and operate in the same ways. The difference between the two is the way in which the carrier waves are modulated. AM radio has a lower bandwidth and can accommodate more stations. FM radio has better sound quality which makes it more desirable for those who want to transmit clear and clean sounding audio. FM is preferred by those who want to start their own low power broadcast.

India got its first commercial radio in 1967 when Vividh Bharati started airing commercials. This followed the recommendations of the committee on Broadcasting and Information Media, which is better known as Chanda Committee. Today, Vividh Bharati runs commercials from serveral broadcast stations across India. It also has an all-night service to entertain listeners. Commercial radio got a big boost in India following the introduction of FM radio. All India Radio (AIR) charges Rs. 3,000 per hour, but the private companies advertisers charges Rs. 250 - 300 for a 10 second commercial.

## Multitone modulation

Modulation with more than one message signal is called multitone modulation. ie, The message signal contains more than one frequency component then the modulated signal is called as multitone modulated signal. Let us consider the message signal as

$$v_m(t) = V_{m1}\cos\omega_1 t + V_{m2}\cos\omega_2 t \qquad \text{......... (1)}$$

Let the carrier signal be

$$v_C(t) = V_C \sin(\omega_C t + \theta) = V_c \sin\phi.$$

Let $(\omega_C t + \theta) = \phi$

$$\therefore \frac{d\phi}{dt} = \omega_C, \qquad \text{the angular frequency.}$$

During frequency modulation the frequency of carrier is changed in accordance with the amplitude of modulating signal. Hence the modulated signal is

$$\omega_i = \omega_C + KV_m(t)$$

$$\omega_i = \omega_C + K[V_{m1}\cos\omega_1 t + V_{m2}\cos\omega_2 t]$$

$$\omega_i = \omega_C + KV_{m1}\cos\omega_1 t + K\,V_{m2}\cos\omega_2 t \qquad \text{.......... (2)}$$

The frequency deviation will be maximum when

$$\cos\omega_1 t = \pm 1; \quad \cos\omega_2 t = \pm 1 .$$

The frequency deviation is proportional to the amplitude of modulating signal. Hence equ (2) can be written as

$$\omega_i = \omega_c + KV_{m1} \pm KV_{m2} \qquad \omega_d = 2\pi\Delta f = KV_m$$

$$\omega_i = \omega_C + 2\pi\Delta f_1 \cos\omega_1 t + 2\pi\Delta f_2 \cos\omega_2 t$$

The instantaneous phase is given by

$$\phi_i = \int \omega_i \, dt$$

$$i = \int(\omega_C + 2\pi\Delta f_1 \cos\omega_1 t + 2\pi\Delta f_2 \cos\omega_2 t)\, dt$$

$$= \omega_c t + \frac{2\pi\Delta f_1}{\omega_1} \sin\omega_1 t + \frac{2\pi\Delta f_2}{\omega_2} . \sin\omega_2 t$$

$$\phi_i = \omega_C t + \frac{\Delta f_1}{f_1}, \sin \omega_1 t + \frac{\Delta f_2}{f_2} . \sin \omega_2 t \text{ ....... (3) } \omega = 2\pi f$$

After frequency modulation, the instantaneous amplitude of the modulated signal is

$e_{fm}(t) = V_C \sin \theta_i$ . Hence using equ (3)

$$e_{fm}(t) = V_C \sin [\omega_C t + \frac{\Delta f_1}{f_1}, \sin \omega_1 t + \frac{\Delta f_2}{f_2} . \sin \omega_2 t ]$$

$$e_{fm}(t) = V_C \sin [\omega_C t + m_{f1} \sin \omega_1 t + m_{f2} \sin \omega_2 t] \quad \text{....... (4)}$$

where modulation index of $FM$ system is

$$m = \frac{\omega_d}{\omega_m} = \frac{2\pi\Delta f}{2\pi f_m} = \frac{\Delta f}{f_m}$$

If $\alpha_1 = m_{f1} \sin \omega_1 t$ ; $\alpha_2 = m_{f2} \sin \omega_2 t$ ,

then, equation (4) canbe written as

$$e_{fm}(t) = V_C \sin [\omega_C t + (\alpha_1 + \alpha_2)]$$
$$= V_C [\sin \omega_C t . \cos (\alpha_1 + \alpha_2) + \cos \omega_C t . \sin (\alpha_1 + \alpha_2)]$$
$$e_{fm}(t) = V_C \{ \sin \omega_C t . [\cos \alpha_1 . \cos \alpha_2 - \sin \alpha_1 \cos \alpha_2]\} +$$
$$\cos \omega_C t [\sin \alpha_1 . \cos \alpha_2 + \cos \alpha_1 \sin \alpha_2]\} \quad \text{..... (5)}$$

In order to simplify the above equation the Bessel function can be used. Hence the resultant equation is given as

$$e_{fm}(t) = V_C \sum_{m=-\infty}^{\infty} J_n(\alpha_1) . J_m(\alpha_2) (\cos \omega_C t + n \omega_1 t + n\omega_2 t) \quad \text{...... (6)}$$

Equation (6) has four frequency terms

(i) A carrier compenent with an amplitude $[J_0(\omega_1) . J_0(\omega_2) . E_C]$

(ii) A set of side bands corresponding to first tone $\omega_1$. The side bands have amplitude $J_n(\alpha_1) . J_0(\alpha_2)$ and frequencies $(\omega_1 + n\omega_2)$.

where $n = 1,2,3.....$

The multitone transmission uses two or more coordinated passband (QAM or QAM-like) signals to carry a single bit stream over a communication channel. The passband signals are idependently demodulated in the receiver and then re-multiplexed into the original bit stream. The bandwidth of each of the 'tones' is sufficienctly narrow, then no ISI (inter symbol interference) occurs on any subchannel.

***Questions***

1. Define frequency modulation. Derive an expression for (i) instantaneous voltage and (ii) modulation index in FM.

2. Discuss the nature of frequency spectrum of FM.

3. Explain phase modulation and derive expressions for

   (i) instantaneous voltage of PM output and

   (ii) modulation index of PM.

4. Discuss phase deviation and modulation index in PM

5. Write about transmission bandwidth requirements in FM.

6. Explain, giving block diagrams, the conversion of

   (i) FM to PM and (ii) PM to FM

7. Explain, with the help of vector diagrams, the phasor representation of (i) FM and (ii) PM.

8. List out the methods of generation of FM. Discuss in detail, any one method.

9. Describe direct method of FM generation.

10. Draw the circuit diagram of reactance tube modulator and explain its action.

11. Describe, giving block diagram, indirect methord of FM generation.

12. Draw the circuit diagram of an FM transmitter and explain its action.

13. Give the block diagram of indirect method of FM/ PM transmitter. Explain the function of various units in it.

14. Compare AM and FM.

15. Write the similarity and difference of FM and PM.

16. Explain commercial broadcast FM.

17. Define multitone modulation. Give the theory of multitone modulation.

***Objective type questions***

1. In FM the modulation index is

(a) $\frac{f_d}{f_m}$ (b) $\frac{f_m}{f_d}$ (c) $\frac{f_C - f_m}{f_m}$ (d) $\frac{f_m}{f_c - f_m}$

2. In FM when the modulation index is increased, the total power is

(a) increased (b) decreased

(c) first increases and then decreases (d) ramains constant

3. FM broadcast range is

(a) 88 KHz to 108 kHz (b) 88 MHz to 108 MHz

(c) 455 kHz to 515 kHz (d) 455 MHz to 505 MHz

4. In FM the intermediate frequency is

(a) 455 KHz (b) 455 MHz (c) 10.7 kHz (d) 10.7 MHz

5. In FM the frequency of carrier wave is altered according to ______ of the modulating signal.

(a) phase (b) frequency (c) amplitude (d) none

6. FM spectrum has

(a) two sidebands (b) three sidebands

(c) no sidebands (d) infinite number of side bands.

7. The special function used in the analysis of amplitudes of sidebands in FM spectrum is

(a) Lorentz function (b) Bessel function

(c) Legendre function (d) Carson function

8. The number of significant sidebands in FM is determined by

(a) modulation index (b) frequency deviation

(c) modulating signal (d) centre frequency

9. The power of frequency modulated wave is _______ the unmodulated wave.

(a) higher than (b) lower than

(c) same as (d) none

10. The total power after modulation depends on modulation idex in

(a) AM (b) FM (c) PM (d) all the above

11. FM band width is

(a) proporational to integral multiple of modulating frequency

(b) independent of modulating frequency

(c) sine of frequency deviation (d) none

12. In FM the frequency deviation is

(a) proportional to modulating frequency

(b) independent of modulating frequency

(c) depends on centre frequency

(d) none

13. In phase modulation the modulation index is

(a) $Kv_m$ (b) $V_m/V_c$ (c) $V_c/V_m$ (d) $\cos\phi$

14. In PM modulation index is

(a) directly proportional to the modulating voltage

(b) directly proportional to the modulating frequency

(c) inversely proportional to the modulating frequency.

(d) all the above.

15. The output of PM receiver in proportional to

(a) centre frequency (b) the phase deviation

(c) carrier amplitude (d) none

16. Varactor diode is

(a) variable resistor diode (b) variable inductance diode

(c) variable capacitance diode (d) variable transsconductance

17. The expression ($g_m$RC) has the dimension of

(a) capacitance (b) resistance

(c) inductance (d) none

18. In FET version of reactance tube FM modulator the gate-to-drain voltage must be ______ with the drain current.

(a) inphase (b) $90^o$ out of phase (c) $180^0$ out of phase

(d) none.

19. In an FM system the modulation index is doubled by halving the modulating signal frequency. The maximuam deviation

(a) remains the same (b) is doubled

(c) halved (d) decrased four times

20. Carson's rule is for

(a) AM bandwidth (b) FM bandwidth

(c) PM bandwidth (d) all bandwidths

*Answers*

1. (a) 2. (d) 3. (b) 4. (d) 5. (c) 6. (d) 7. (b) 8. (a) 9. (c)
10. (a) 11. (a) 12. (b) 13. (a) 14. (a) 15. (b) 16. (c) 17. (a)
18.(b) 19. (a) 20. (b).

Unit IV

FM RECEPTION

FM detectors – Balanced slope detector – Foster seely discriminator – ratio detector – F'M super heterodyne receiver – FM noise suppression – threshold extension by FMFB technique.

## F.M. detectors

Separating the A.F signal (message) from the modulated carrier wave reaching the receiver is known as demodulation or detection. The circuit employed for this purpose is known as detector. The function of FM demodulation (or frequency-to-amplitude changer) is to change the frequency deviation (in the FM wave) of the incoming carrier into an AF (audio frequency) amplitude modulation. This must be identical to the one that caused the frequency deviation.

It is now necessary to think of a circuit, which produces an output where amplitude depends on the frequency deviation of the input voltage.

### FM Slope Detector Demodulator

The FM slope detector uses the attenuation slope of a tuned circuit to convert frequency variations into amplitude variations that can then use a diode detector to give the required audio.

FM slope detection is a concept that can be used to recover the modulation from an FM signal, but it is not widely used.

There are far more efficient methods that can be used for FM detection / demodulation. However as a concept it is useful to understand.

*FM slope detection basics*

The very simplest form of FM demodulation is known as slope detection or demodulation. It consists of a tuned circuit that is tuned to a frequency slightly offset from the carrier of the signal.

As the frequency of the signals varies up and down in frequency according to its modulation, so the signal moves up and down the slope of the tuned circuit. This causes the amplitude of the signal to vary in line with the frequency variations. In fact at this point the signal has both frequency and amplitude variations.

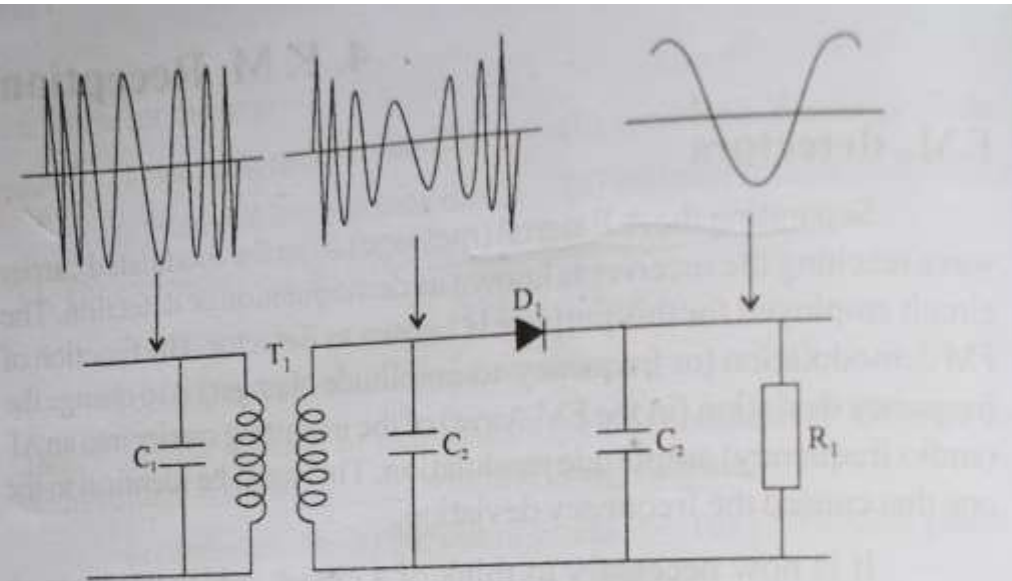


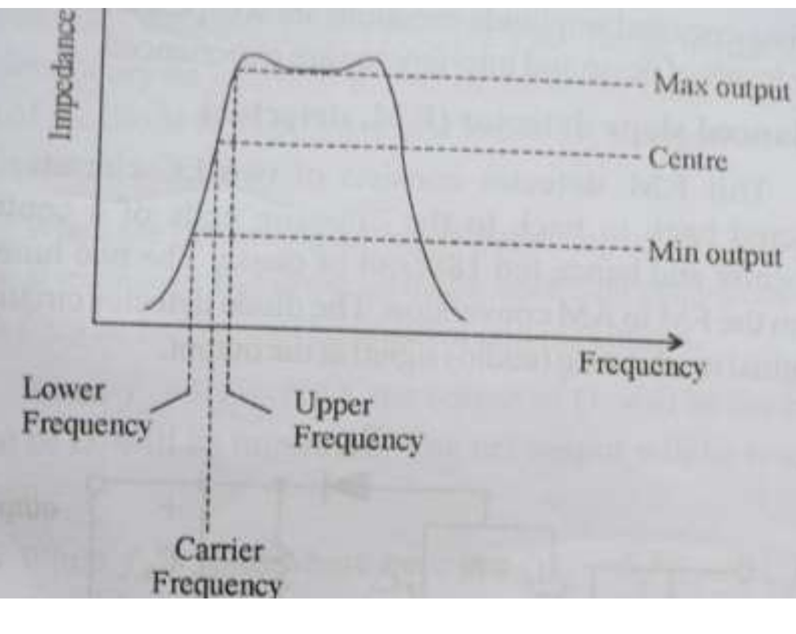

*Fig. 55 A. FM slope detection concept*

*FM slope detection concept*

It can be seen from the diagram that changes in the slope of the filter, reflect into the linearity of the demodulation process. The linearity is dependent not only on the filter slope as it falls away, but also the tuning of the receiver - it is necessary to tune the receiver off frequency and to a point where the filter characteristic is relatively linear.

The final stage in the process is to demodulate the amplitude modulation and this can be achieved using a simple diode circuit. One of the most obvious disadvantages of this simple approach is the fact that both amplitude and frequency variations in the incoming signal appear at the output. However the amplitude variations can be removed by placing a limiter before the detector.

A variety of FM slope detector circuits may be used, but the one below shows one possible circuit with the applicable waveforms. The input signal is a frequency modulated signal. It is applied to the tuned transformer (T1, C1, C2 combination) which is offset from the centre carrier frequency. This converts the incoming signal from just FM to one that has amplitude modulation superimposed upon the signal.



This amplitude signal is applied to a simple diode detector circuit, D1. Here the diode provides the rectification, while C3 removes any unwanted high frequency components, and R1 provides a load.

*FM slope detection advantages & disadvantages*

FM slope detection is not widely used, and yet it has some limited applications. Knowing the advantages and disadvantages enables the technique to be used where applicable.

*Advantages*

Simple - can be used to provide FM demodulation when only an AM detector is present.

Enables FM to be detected without any additional circuitry.

*Disadvantages*

Not linear as the output is dependent upon the curve of a filter.

Not particularly effective as it relies on centring the signal part of the way down the filter curve where signal strengths are less.

Both frequency and amplitude variations are accepted and therefore much higher levels of noise and interference are experienced.

**1. Balanced slope detector (F.M. detector)**

This F.M. detector consists of two LC circuits. They are connected back to back to the opposite ends of a centre-tapped transformer and hence fed $180^0$ out of phase. The two tuned circuits perform the FM to AM conversion. The diode detector circuits recover the original modulating (audio) signal at the output.

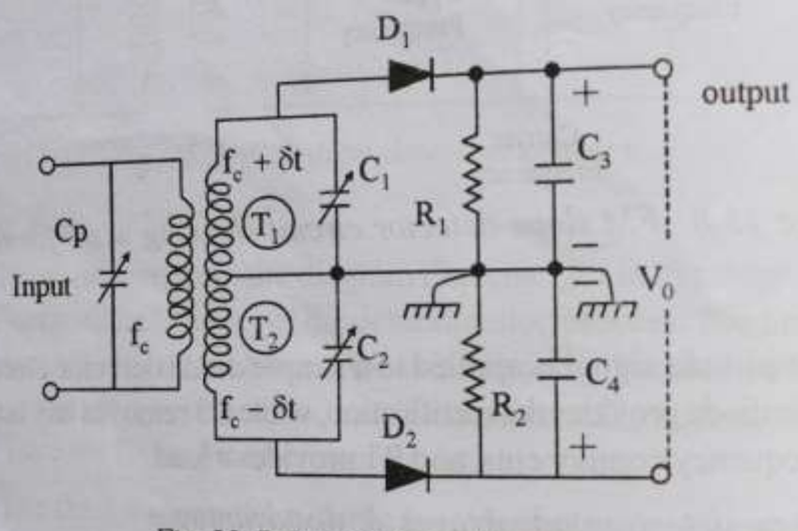



*Fig. 55. Balanced slope detector*

Let the primary winding be tuned to a frequency $= f_c$, the intermediate frequency (I.F) of the carrier wave. The top secondary is tuned (by adjusting the capacitance $C_1$) above the I.F by an amount $\Delta f$. The boltom circuit is similarly tuned below the IF by the same amount. Each tuned circuit is connected to a diode detector with an RC load. The output is taken from across the series combination of the loads so that the output is the sum of the two individual outputs. The receiver demodulates the received signal and recovers the original modulating signal.

*Working*

When the input frequency is $f_c$, the voltage across the upper half of the secondary coil will be less than the maximum voltage. This is because the maximum voltage occurs at resonance that will take place only at its resonance frequency ($f_c + \Delta f$). The voltage across lower half of the secondary be identical to that of the upper half. Therefore, the output of the diode $D_1$ is positive and that of $D_2$ is negative. Hence the detector output is zero.

When the instantaneous input frequency is $f_m = f_c + \Delta f$, circuit I will be at resonance. But in circuit II, the input is far away from resonance ($f_c - \Delta f$). The output will be positive and maximum.

When $f_m = (f_c - \Delta f)$, the output of $D_2$ will be the maximum and that of $D_1$ will be minimum. The net output will be negative and maximum.

When $f_m$ is somewhere between ($f_c - \Delta f$) and ($f_c + \Delta f$), the output will the between the above two extremes,. It will be positive or negative, depending on which side of $f_c$, the input frequency lies.

The positive and negative halves of the modulating signal are avilable across the output terminals. The operating range of the overall response curve is a straight line. Hence the operation is linear.

When the input frequency goes outside the range, the tuned circuit's response will make the output fall. Thus the output and input frequency characteristics of the detector will be the required S-shaped curve.

*Disadvantages*

1. The transformer has to be adjusted for theree different frequencies. [$f_c$, ($f_c + \Delta f$) and ($f_c - \Delta f$) ].
2. Alignment is tricklier.
3. Amplitude limiting is not provided in this modal of FM demodulation. Hence, this detector responds to the amplitude variations of the input and it will not provide the true modulating signal. Limiter circuit is used to remove any amplitude present in the received signal.
4. Distortion due to the lowpass RC filter in the circuit may arise.

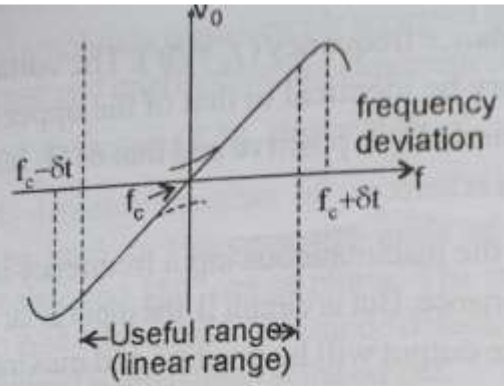



*Fig. 56. Frequency demodulation characteristics.*

## 2. Foster-Seely discriminator : centre-tuned discriminator

*What is a discriminator*

The FM signal which arrives at the antenna of the receiving station is likely to have variation of amplitude due to noise voltages, fading, reflections and absorptions. These extraneous variations in the amplitude must first be smoothed out. Otherwise the reception would be distorted and noisy.

In the next stage the detector output voltage must be made zero for the unmodulated carrier wave and it should rise and fall in exact agreement with the deviation of frequency of the modulated wave. The device used for this purpose is known as discriminator.

We see here Foster Seely discriminator circuit that gives fairly the required S - shaped frequency response circuit.

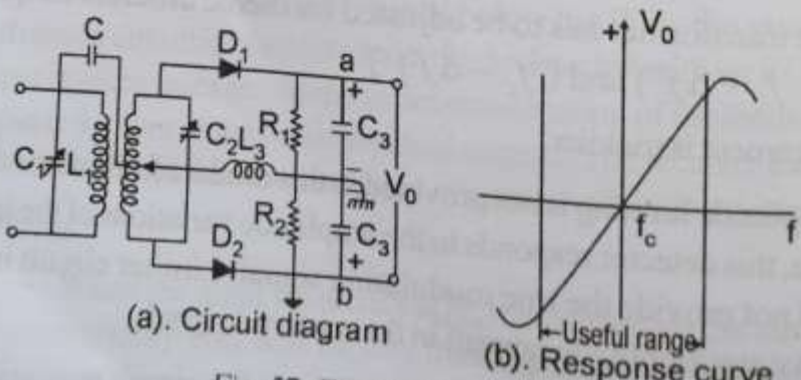



*Fig. 57. Foster - Seely discriminator*

In this circuit both primary and secondary windings are both tuned to the centre frequency of the incoming signal. This is desirable because it greatly simplifies alignment and also gives better linearity than slope detection mothod. In this circuit the same diode and RC load arrangement is used as in balanced slope detector.

The primary and secondary voltages are exactly $90^0$ out of phase. When the input frequency is greater than the centre frequency $f_c$, the voltages are less than $90^0$ out of phase. When the input frequency is below $f_c$, the primary and secondary voltages become more than $90^0$ out of phase.

Although the individual component voltages will be the same at the diode inputs at all frequencies, the vector sum will differ with $\phi$, the phase diffrence between the primary and secondary windings. The result will be: the individual output voltages will be equal only at the centre frequency, $f_c$. At all other frequencies, the output of one diode will be greater than that of the other. Accordingly, the overall ouput will be +ve or –ve according to the input frequency.

The magnitude of the ouput will depend on the deviation of the input frequency from $f_C$, as we need. Thus the resulting voltage at the output corresponds to the modulation on the incoming signal.

*Advantages*

1. Much easier alignment because it has only two tuned circuit and both are tuned to the same frequency $f_C$.

2. The working of the circuit is based primarily on primary -secondary phase relationship and so linearity in the output is better.

*Disadvantage*

This discriminator needs a separate amplitude limiting circuit like a Ratio detector which is another phase discriminator.

### 3. Ratio detector

When compared with Foster -Seely circuit, three important changes have been made to achieve amplitude limiting. The resulting circuit is the Ratio Detector.

(i) One of the diodes has been reversed;

(ii) A large capacitor (electrolytic) is placed across the (previous) output terminals.

and (iii) The output now is taken in between resistor and capacitor net works.

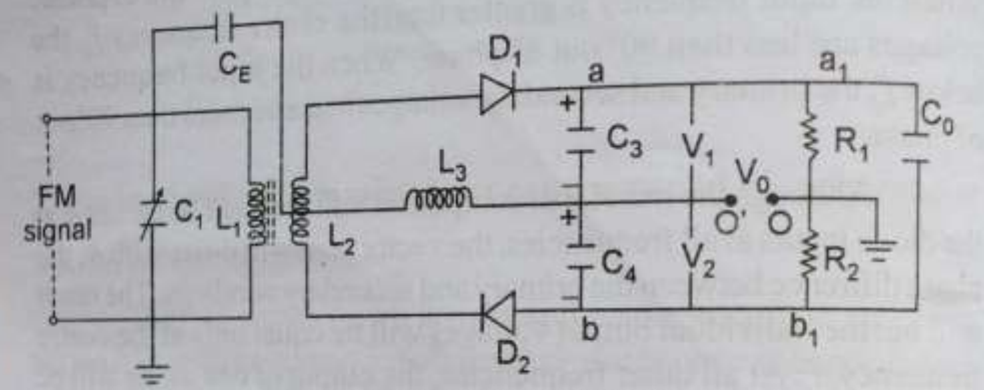



*Fig. 58. Ratio Detector Circuit*

*Operation*

The detector is tuned with the use of inductance $L_1$ and capacitance $C_1$ to receive FM wave of frequency $f_m$. As the diode $D_2$ is reverse-connected, the polarity at $a$ and $b$ are opposite (reversed). Let $V_1$ and $V_2$ be the voltage drop across $D_1$ and $D_2$ respectively.

The voltage across $a_1$ and $b_1$ is the sum of the voltages rather than the difference.

This sum voltage is kept constant due to the large capacitor $C_o$ connected between $a_1$ and $b_1$.

Once $C_0$ is connected, the output has to be taken across O and O'.

The capacitor $C_0$ gets charged to the sum voltage ( $V_1 + V_2$). which is constant. Normally $R_1$ and $R_2$ are kept equal i.e. $R_1 = R_2 = R$. The voltage at the junction of $R_1$ and $R_2$, measured with respect to the bottom point is

$$\frac{(V_1+V_2)}{2}$$

This voltage is also fixed due to large $C_0$ and large time constant.

*Case* (i) When the signal frequency $f_m$ equals $f_C$ ($f_m = f_C$), the input to both the diodes will be equal. As the capacitance $C_1$ is always kept equal to $C_2$, the diode voltages $V_1$ and $V_2$ are equal. The potential difference across the output terminals = 0.

*Case (ii)* $f_m > f_C$

When the signal frequency $f_m$ is greater than the centre frequency $f_C$, the input for diode $D_1$ exceeds the input for $D_2$.

$$V_0 = -\Delta V$$

*case (iii)* $f_m < f_C$. For signal frequencies below $f_C$, the voltage across $D_1$ is reduced to ( $V_1 - \Delta V$) and the voltage through $D_2$ is increased to ($V_2 + \Delta V$).

$$V_0 = \Delta V$$

Thus, it is seen that the output voltage follows the deviation of frequency of the signal. Thus this circuit translates FM signal into original modulating signal.

Let us now see, how amplitude limiting is achieved with Ratio detector i.e. how the detector responds to amplitude changes at the input.

(i) If the input voltage $V_{12}$ is constant, the large capacitor $C_o$ will charge upto the potential existing between a and b. This will be a d. c. voltage and no current will flow through $C_0$. Now $\omega = 0$. Hence $C\omega$ = 0 and $1/C\omega$ = infinity reactance. The total inpedance for the two diodes is the sum of $R_1$ and $R_2$. The output will be not changing. ie; it remains the same.

(ii) Let us say that there is an increase in input due to noise. In order that the output follows the input, the capacitor $C_0$ must get charged to that level. For this purpose, the capacitor $C_0$ draws charging current from the input resonant circuit, thereby loading the circuit to a great extent. As a result, the Q -factor (magtnification factor) of the circuit is lowered, thereby reducing the input signal level.

(iii) Let there be a decrease in the input signal level. The capacitor must discharge to the input level. Thus there is a discharging current through $R_1$ and $R_2$. The current drawn from the input circuit is reduced. This reduces loading upon the input resonant circuit, causing the Q-factor to increase.As a reasult, the input signal level is increased.

Thus, the ratio detector output remains free from amplitude fluctuations of the signal input and it converts frequency changes into amplitude changes.



## FM superheterodyne receiver

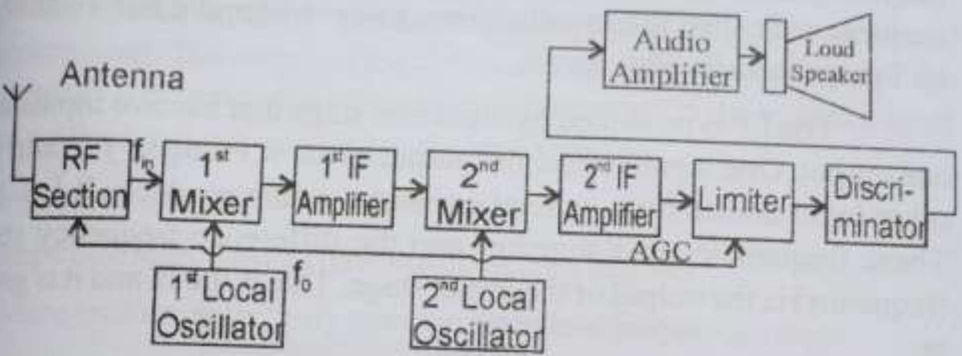



*Fig. 59. Double Frequency Conversion FM Superheterodyne Receiver*

A receiver, designed to receive frequency - modulated radio waves is known as FM receiver. The principle of superheterodyning i.e; the amplified incoming RF carrier wave is mixed with the output of a local oscillator and obtaining thereby the intermediate frequency (I.F = 10.7 MHz) is used in the FM receiver. The noise-free feature of FM is attained, when the carrier amplitudes are limited to a particular value. So a limiter and a discriminatior are necessarily used in the FM receiver. The block diagram shows various units, used in the receiver.

*RF section*

This is similar to that used in AM receivers. Many modulated signals intercept the antenna. The desired station modulated insgnal is selected by tuning and then amplified to the required level.Unlike AM, the R.F sections in FM receivers are disigned to operate at VHF and UHF frequency ranges...

... FM receivers are disigned to operate at VHF and UHF frequency ranges, employing space wave propagation, the FM signals are not prone to selective or general fading. Moreover the FM receiver circuits do not require an AGC bias.

*Frquency changer*

It is the combination of mixer and local osillator. The frequency response of RF amplifier in the previous stage. ie the amplification is not uniform for all the signals, selected from varous station carrier frequencies. Hence any signal received must first be converted in to a fixed frequency

for the carrier containing the same audio messge. The fixed carrier frequency can be easily amplified by using tuned amplifiers. The fixed frequency is called intermediate frequency (I.F) and it has a value of 10.7 megahertz for FM. waves.

The I.F is produced by the mixer stage that has two inputs and one output. One input is from the output of the R.F stage ( $f$ ) and the other input comes from a local oscillator which produces R.F. ( $f_0$). These frequencies mix together and the difference frequency (beat frequency) is the output of the mixer stage. This is the IF and it is given by

$$I.F = f_0 - f$$

The processof obtaining I.F by mixing is known as superheterodyning.

*I.F section*

It consists of one or more stages of tuned amplifiers, which have sufficient bandwidth to accomodate the F.M. signals. The I.F amplifier amplifies the intermediate frequency present at the output of the mixer stage.

The IF section provides most of the gain and selectivity of the receiver.

*Limiter*

The purpose of the limiter is to provide contant amplitude I.F signal with the same frequency deviation as produced by the modulating signal at the output of the discriminator (demodulator). The discriminator resolves the modulatinge signal. Due to the action of the limiter circuit, the amplitude variations in FM due to external and internal noise (stray voltages) do not reach the receiver output.

*De - emphasis circuit*

This unit is employed to restore relative magnitudes of different components of A.F signals as in the original modulating singnal. This accounts for better quality of sound note heard with FM receivers. De-emphasis attenuates higher frequencies to bring them back to the original amplitude as these were boosted or emphasized before transmission.

*Detector*

The FM detector is usually a Foster-Seely discriminator or Ratio detector to recover the original modulating signal from the modulated carrier signal. The output of the detector is passed to an audio amplifier to amplify the detected baseband signal (message) to the required level. The audio signal is finally fed to the loud speaker.

*Advantages of superheterodyne receiver*

1. Improved selectivity in terms of adjacent channels.
2. More uniform selectivity over the complete frequency range.
3. Improved receiver stability.
4. Higher gain per stage because I.F amplifiers are operated at a lower frequency.
5. Uniform bandwidth because of fixed I.F.

**F.M Noise suppression**

Pre-emphasis and De-emphasis circuits are used for suppression of the unwanted noise. The noise ( $N$ ) has greater effect on higher modulating frequencies ( $F$ ) than on the lower ones. (ie, $N \propto F^2$) The effect of noise on higher frequencies can be reduced by artificially boosting them at the transmitter and correspondingly *attennating* them at the receiver. This boosting of high modulating frequencies at the transmitter is called pre-emphasis and attenuating them at the receiver is called *de-emphasis*. Normally *pre-emphasis* and de-emphasis are carried out according to standard curves. Figure shows the 75 μs (value of time constance RC) emphasis curves.

A 75 μs pre-emphasis corresponds to a frequency response curve which is 3 dB up at a frequency whose time constant RC is 75 μs ie; with RC = 75 μ sec.

$$f = \frac{1}{2\pi\, RC} = \frac{1}{2\pi \times 75 \times 10^{-6}} = 2120 \text{ Hz}$$

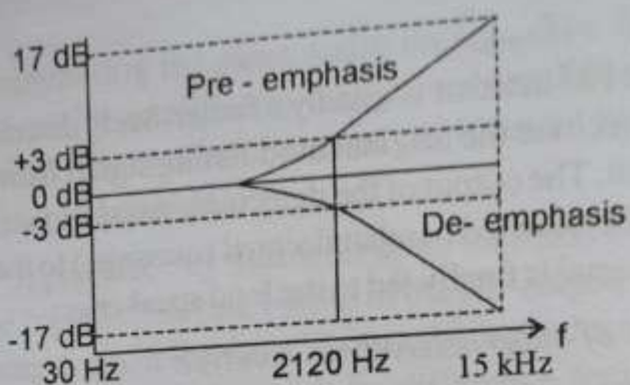
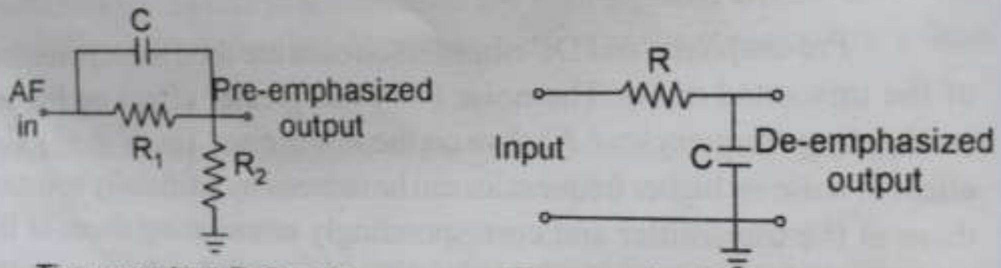



*Fig. 60. 75 μs emphasis curves*

In figure we observe that the emphasis curves have 3 dB frequency of 2120 Hz. The pre-emphasis circuit is shown below. The values of $R_1C$ are ajusted to get required $R_1C$ pre-emphasis. Similarly de-emphasis circuit is shown below. Its RC time coustant can be adjusted to get required de-emphasis curve.




Time constant $R_1C = 75\mu s$

(a). Pre-emphasis circuit

Time constant $RC = 75\ \mu s$

(b). De-emphasis circuit

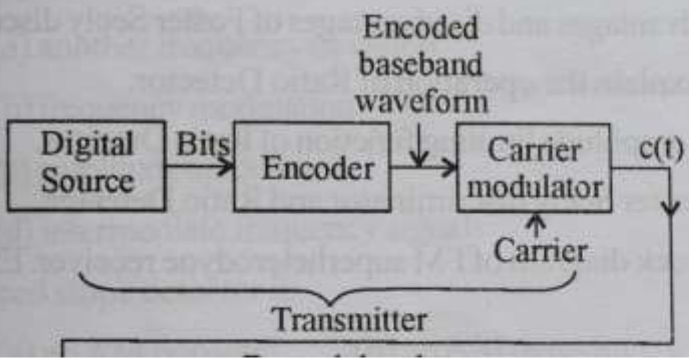
*Fig. 61. Pre-emphasis and De-emphasis circuits*

## Threshold extension by frequency modulation feed back (FMFB)

The criteria for evaluating the performance of an FM demodulator are based on the capability of the device to provide a linear relationship between the output and input signal-to-noise ratios. The useful operating range of all FM demodulators is limited by the fact that this linear relationship becomes nonlinear below a certain value of input SNR. This input SNR value is called the point of theshold for the demodulator.

FM threshold is defined as a carrier-to-noise ratio at which demodulated signal-to-noise (S/N) ratio falls 1 dB below the linear relationship. This is the effect produced in an FM receiver when noise limits the desired information signal. The concept of lowering the threshold or as it is commonly called, *theshold extension*, usually means to extend the region relative to a conventional FM demodulator.

The theshold reduction or extension can be achieved in FM demodulator with negative feedback. Figure shows the block diagram of a FMFB demodulator.



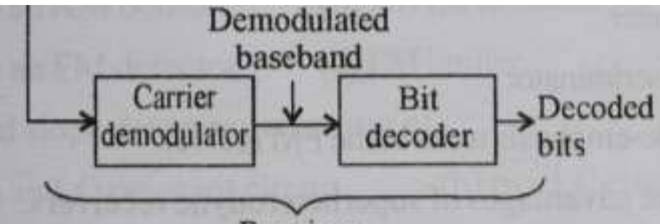



*Fig. 62. Digital communication block diagram*

In the above figure the local oscillator is replaced by voltage controlled oscillator (VCO). The instantaneous output frequency of VCO is controlled by demodulated signal. Therefore output frequency of VCO changes as per low frequency variations of demodulated signal. In other words VCO frequency does not depends upon high frequency variations of narrowband noise. Thus FMFB demodulator acts as a tracking filter. It tracks only the slowly varying frequency of wideband FM waves. Therefore it responds only to narrowband of noise centered around instantaneous carrier frequency. This reduces the threshould of FMFB receiver. The threshold reduction of about 5 - 7 dB is possible.

*Questions*

1. Explain the principle of FM detection. Draw and explain the characteristics of frequency domodulation.

2. Draw the circuit diagram of balanced slope FM detector and explain its working. What are its disadvantages?

3. Describe the function of a discriminator.

4. Draw and explain the operation of Foster Seely discriminator. Discuss its frequency response curve.

5. Write the advantages and disadvantages of Foster Seely discriminator.

6. Draw and explain the operation of Ratio Detector.

7. Explain the amplitude limiting function of Ratio Detector.

8. Compare Foster Seely discriminator and Ratio Detector.

9. Draw the block diagram of FM superheterodyne receiver. Explain the function of

   (i) limiter

   (ii) discriminator

   (iii) de-emphasis units in the FM reciver.

10. List out the advantages of superheterodyne receivers.

11. Define FM demodulator theshold and explain in brief. Draw the block diagram of FMFB demodulator and explain how threshold reduction is obtained.

12. What is Pre-emphasis and De-emphasis? Why it is required?

***Objective type questions***

1. In an FM detector the output amplitude should vary as

(a) amplitude of the input voltage

(b) phase of the output voltage

(c) frequency of the input voltage

(d) frequency deviation of the input voltage

2. FM demodulators change the frequency deviations of the incoming carrier into

(a) another frequency deviaton

(b) frequency modulation

(c) amplitude modulation

(d) intermediate frequency signals

3. Balanced slope detector is

(a) an AM booster (b) an AM detector

(c) an FM detector (d) FM limiter

4. Balanced slope detector makes use of

(a) an LC resonant circuit (b) two LC circuits

(c) two RC networks (d) none

5. Foster Seely discriminatior is

(a) AM detector (b) FM detector

(c) Phase stabliser (d) none

6. Noise supression in FM is possible with the use of

(a) discriminator (b) frequency change

(c) superheterodyning (d) all the above

7. Frequency response graph of an FM detector need to be

(a) bell shaped (b) Inverted bell shaped

(c) S-shaped (d) any shape

8. Which one is amplitude-limiting circuit:

(a) Balanced modulator (b) Foster Seely detector

(c) Ratio detector (d) all the above

9. The Ratio detector converts

(a) input frequency changes into output amplitude changes.

(b) input amplitude changes into output frequency changes.

(c) low impedance at the input to high impedance at the output

(d) none

10. The I.F for FM transmission and reception is

(a) 10.7 kHz (b) 10.7 MHz (c) 455 kHz (d) 455 MHz

11. I.F contains

(a) baseband signal

(b) no base band signal

(c) local oscillator frequency only

(d) all the above

12. Noises in Ratio receivers are

(a) high frequency sound

(b) low frequency sound waves

(c) sudden bursting of speaker coils

(d) spurious voltages

13. In FM super heterodyne receiver, noise is prevented from reaching the final output due to the action of

(a) superheteradyning (b) I.F amplifier

(c) limiter (d) de-emphasiser

14. Better quality of musical note heard with FM receivers is due to the action of

(a) attenuator (b) limiter

(c) de-emphasizer (d) all the above.

15. The additional units used in FM superhet receivers, when compaired to AM receivers are

(a) limiter and de-emphasize units

(b) R.F and I.F amplifiers

(c) Notch filter (d) all the above

*Answers*

| | | | | | | |
|---|---|---|---|---|---|---|
| 1.(d) | 2. (c) | 3. (c) | 4. (b) | 5. (b) | 6. (a) | 7.(c) |
| 8. (c) | 9. (a) | 10. (a) | 11. (a) | 12. (d) | 13. (c) | |
| 14. (c) | 15. (a) | | | | | |

## UNIT V

## DIGITAL MODULATION

In digital communication systems, source information (data) is first converted into sequences of digital symbols (data bits 1, 0,....) which are then encoded into wave forms for ( R.F) carrier modulation and transmitted over the link (medium). At the receiver the demodulated wave forms are decoded back into the digital sequence of bits from which the source information is then obtained. The advantage of digital communication over analog mathods is that the bits can be transmitted error - free with lesser carrier power than in analog system.

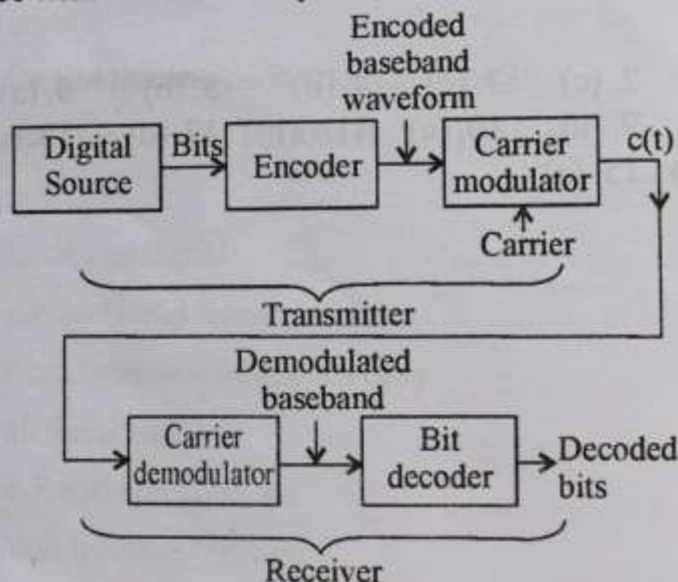



*Fig. 62. Digital communication block diagram*

We know that modulation is the process of encoding source data on to a carrier signal with frequency $f_C$. Data transmision may be obtained by modulating the carrier signal. To transmit *analog data* (message signal) over *digital carrier* signal, the following two digital schemes can be used:

(i) Pulse code modulation (PCM) and

(ii) Delta modulation (D.M)

At the outset we shall have a review of some basic concepts and for terminologies that will be useful for further reading.

### 3. Frequency shift keying (FSK)

In FSK the Frequency of the carrier signal is varied to represent the information data. The frequency of the modulated signal is constant for the duration of one signal element but changes for the next signal elements of the data element change. ( 1> 0 or 0 >1)

### 4. Binary FSK (BFSK)

Consider two carrier frequencies $f_1$, and $f_2$. We use the first carrier if the data element is 0. We use the second carrier if the data element is 1. Practically the carrier frequencies are very high and the difference between them is very small. With this scheme, the '1; is called the mark frequency and '0' is called the space frequency.

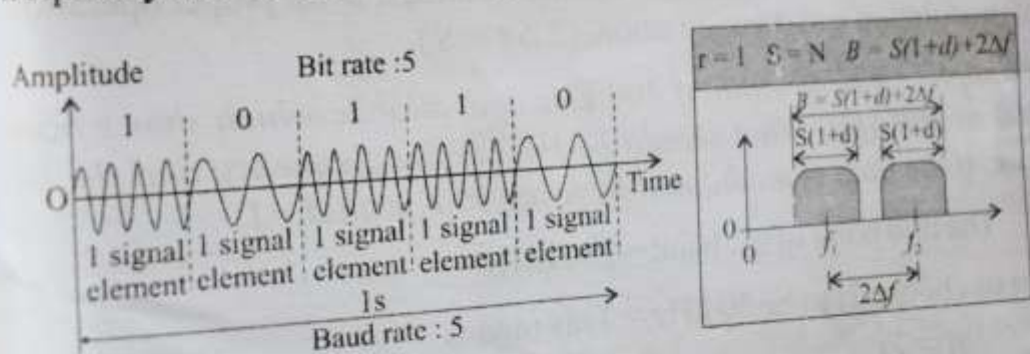



*Fig. 78. Binary frequency shift keying*

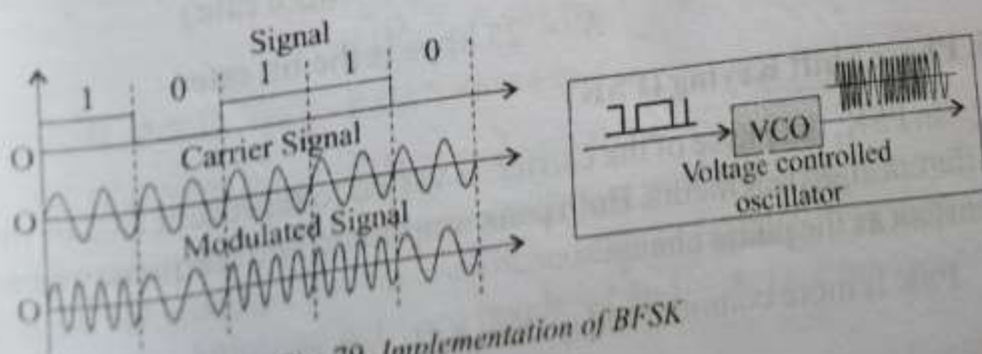



*Fig. 79. Implementation of BFSK*

As seen in figure the middle of one bandwidth is $f_1$ and the middle of the other is $f_2$. Both $f_1$ and $f_2$ are $\Delta f$ apart from the mid point between the two bands. The difference between the two frequencies is $2\Delta f$.

*Bandwith for BFSK*

Figure above also shows the bandwidth. The carrier signals are only simple sine waves but the modulation creates a non-periodic composite signal with continuous frequencies. We can think of, FSK as two ASK signals, each with its own carrier frequencies ( $f_1$ or $f_2$). If the difference between the two frequencies is $f$, then the required bandwidth is

$$B = (1 + d) \times S + 2\Delta f.$$

*What should be the minimum value of $\Delta f$?*

In figure we have chosen a value greater than $(1 + d)\,S$. It can be shown that the minimum value should be atleast $S$ for proper operation of modulation and demodulation. ($2\,\Delta f = S$).

*Example: A bandwidth of 100 kHz is available, which shows from 200 to 300 kHz. What should be the carrier frequency and the bit rate, if the data is modulated by using FSK with d =1.*

The mid point of the band = 250 kHz.

Let us choose 2f to be 50 kHz = This means

$$B = (1 + d) \times S + 2\,\Delta\, S$$

$$100 = 2 \times S + 50 \qquad \therefore 2S = 100 - 50 = 50$$

$$\therefore S = 25 \text{ baud (symbol rate)}$$

$$N = 25 \text{ kbps is the bit rate.}$$

**5. Phase Shift Keying (PSK)**

In PSK, the phase of the carrier is varied to represent two or more different signal elements. Both peak amplitude and frequency remain constant as the phase changes.

PSK is more common today than ASK or FSK

## 6. Binary phase shift keying (BPSK)

In BPSK, we have only two signal elements, one with a phase of $0^0$ and the other with a phase of $180^0$. Figure below gives a view of BPSK.

*Implementation*

The implementation is as simple as ASK. This is because the signal element with a phase of $180^0$ can be seen as the complement of the signal element with phase $0^0$. Thgis gives us a clue on how to implement BPSK. Here in BPSK, the phase of the carrier is switched between two values, according to the two possible messages $m_1$, and $m_2$. The two phases are separated by $\phi$ radian. PSK can be implemented using the system shown below.

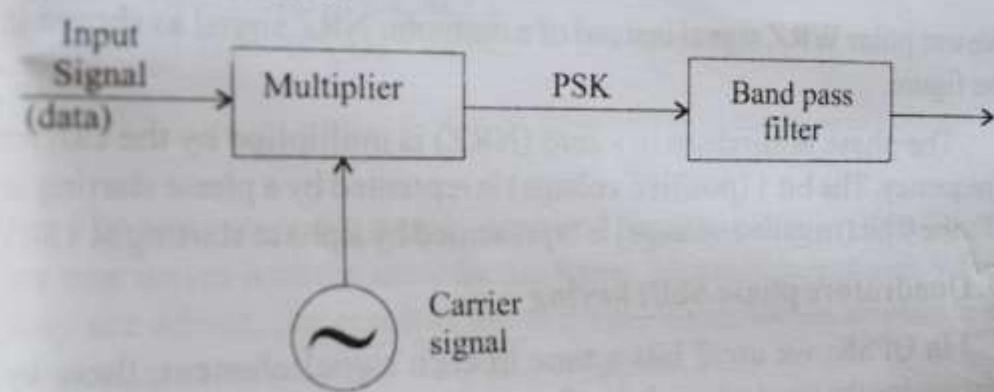



*Fig. 80(a). PSK System*

The input data is translated into NRZ (non return -to-zero) code, having no d.c. component. The resulting pulses have amplitudes $+A$ or $-A$ representing binary 1s and 0s. The output of the multiplier will be PSK signal. It can be represented as

$$V(t) = A\cos(\omega_c t + \phi)$$

If $\phi = 0$, $\quad V(t) = A\cos\omega_c t$.

If $\phi = \pi$ $\quad V(t) = -A\cos\omega_c t$.

The main difference between ASK and PSK wave forms is that in the ASK scheme, the carrier is switched ON and OFF. In PSK scheme, the carrier is , suitched between levels $+A$ and $-A$. ($+A$ for 1 and $-A$ for 0)

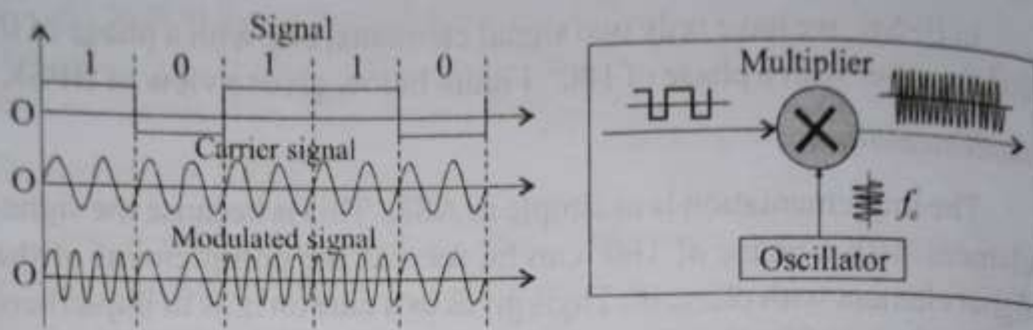



*Fig. 80(b). Implementation of BASK*

We use polar WRZ signal instead of a nonpolar NRZ signal as shown in the figure.

The phase non-return to - zero (NRZ) is multiplied by the carrier frequency. The bit 1 (positive voltage) is reprented by a phase starting at $0^0$; the 0 bit (negative voltage) is represented by a phase starting at $180^0$.

**7. Quadrature phase Shift keying**

In QPSK, we use 2 bits a time in each signal element, there by decreasing the baud rate and also the required bandwidth. The scheme is called quadrature PSK because it uses two separate BPSK modulations. One is in phase, the other in quadrature (out of phase). The two systems operate independently. QPSK allows the signal to carry twice as much information as ordinary PSK using the same bandwidth.

The incoming bits are first passed through a serial-to-paralled conversion arrangement that sends one bit to one modulator and the next bit to the other modulator.

If the duration of each bit in the incoming signal is $T$, the duration of each bit sent to the corresponding BPSK signal is $2T$. That is, the bit to each BPSK signal has one half the frequency of the original signal.

Figure below shows this idea.

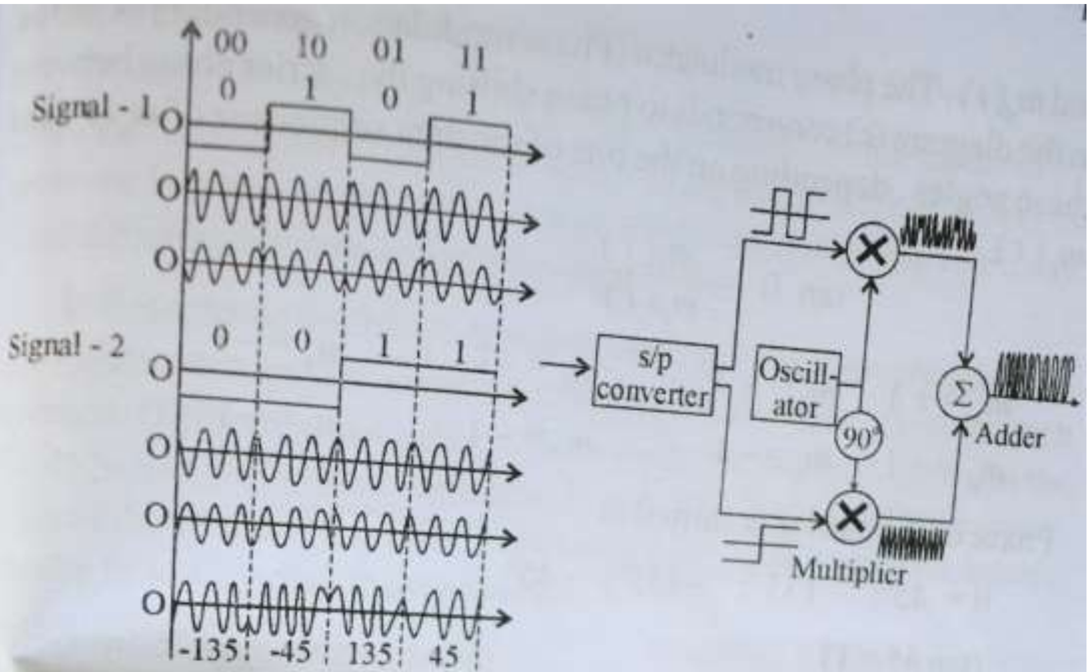



*Fig. 81. QPSK and its implementation*

The two composite signals, produced by each multiplier (refer figure) are sine waves with the same frequency ($\omega_c$) but different phases. When they are added, the result is another sine wave, whose amplitude is proportional to $\sqrt{m_C^2(t)+m_S^2(t)}$ and phase, given by

$$\tan\theta = \frac{m_S(t)}{m_C(t)}$$

where $m_C$ is the carrier bit ($\pm 1$ for every t) and ms is the signal bit.

Since $m_C = \pm 1$ and $m_S = \pm 1$ at every $t$,

$\sqrt{m_C^2 + m_S^2} = 1$ for every $t$.

ie, the QPSK carrier has constant envelope. The QPSK carriers can be formed either by combining two separate binary phase shift keying quadrature carriers or by phase shifting a carrier, according to the ratio.

$$\frac{m_S(t)}{\quad}$$

and $m_S(t)$. The phase modulator (Phase modulation generator) as shown in the diagram (c) coresponds to phase shifting the carrier phase between these angles, depending on the bits of the data sequences of $m_S(t)$ and $m_C(t)$

$$\tan\theta = \frac{m_S(t)}{m_C(t)}$$

$m_S = +1$ $\quad m_S = +1$ $\quad m_S = -1$ $\quad m_S = -1$

$m_C = +1$ $\quad m_C = -1$ $\quad m_C = -1$ $\quad m_C = +1$

Phase of the carrier is shifted at

$\theta = 45^0;\quad 135^0;\quad -135^0;\ -45^0$

$(\tan 45 = 1)$

There are four kinds of signal elements in the output signal. We can send 2 bits of information per signal element ( ie; $r = 2$).

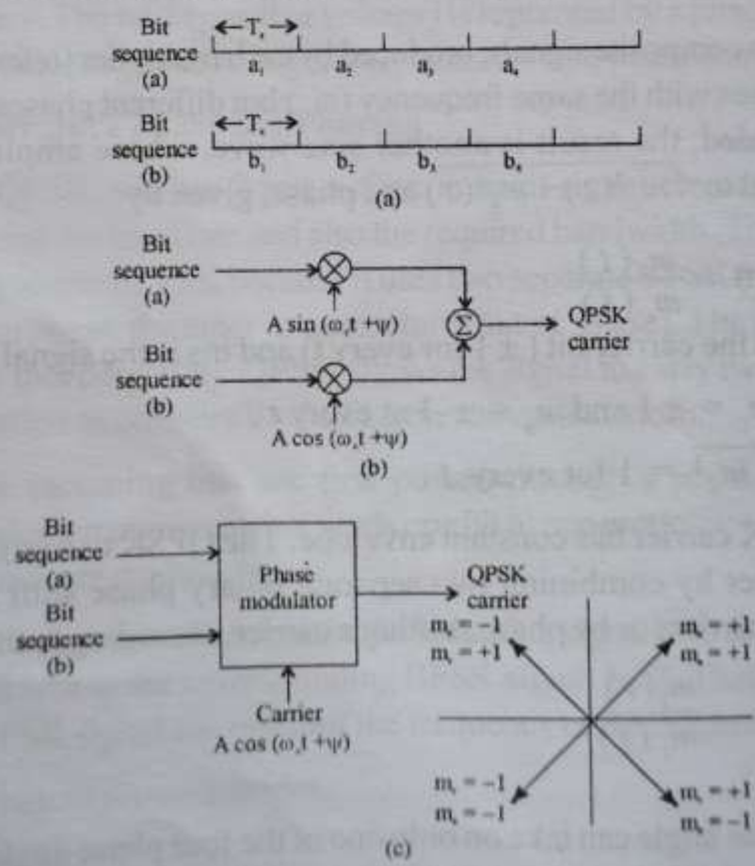



Fig. 82. *Four kinds of signal elements*

## Differential Phase Shift Keying (DPSK)

In phase shift keying, the information resides on the phase of the carrier. It is difficult to get the absolute phase at the receiver unless the receiver is given some *indication* about the proper phase reference. This difficulty is overcome in differential PSK.

In this scheme the information is encoded in terms of the phase change ($\delta\phi$) between adjacent signals rather than on absolute phase for each symbol. Differential encoding of a message sequence is shown in the Table below. An orbitrary reference binary digit is chosen for the initial digit of the encoded sequence. For example, '1' is taken as the reference for the following digit in the sequence.

*Differential encoding example*

| | | | | | | | | | | |
|---|---|---|---|---|---|---|---|---|---|---|
| Message sequence | | 1 | 0 | 0 | 1 | 1 | 1 | 0 | 0 | 0 |
| Encoded sequence | 1 | 1 | 0 | 1 | 1 | 1 | 1 | 0 | 1 | 0 |
| Reference digit<br>Transmitted phase | 0 | 0 | $\pi$ | 0 | 0 | 0 | 0 | $\pi$ | 0 | $\pi$ |

(alike, result 1; unlike, result 0.)

Consider the message sequence. A '1' is encoded as no change of state. A '0' is encoded as a transition (change) from the state of reference digit to the opposite state in the encoded *message sequence*. In the example shown the first digit in the *message sequence* is a '1' and so no change of state takes place in the *encoded sequence* and a '1' appears as the next digit in the encoded sequence. This serves as the reference for the next digit to be encoded. The next digit appearing in the message sequence is '0'; the corresponding encoded digit is opposite of the reference digit and it is '0', and so on.

The message sequence then phase-shift keys a carrier with a phase 0 or $\pi$ as shown in the last row of Table-above.

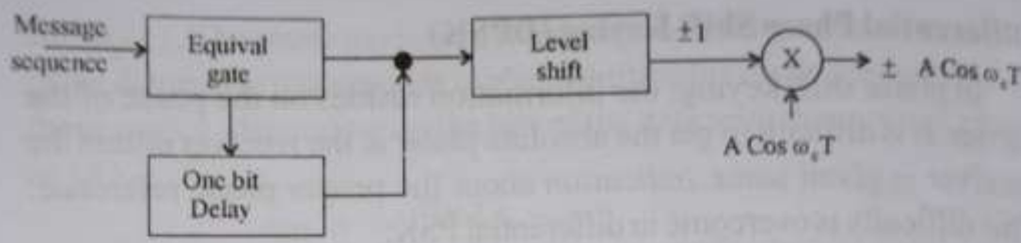



*Fig. 83. Block diagram of a DPSK modulator*

The block diagram shown here illustrates the generation of differential PSK. The equivalence gate in the diagram is the negation of Exclusive - OR gate (XOR) a logic circuit that performs as listed in the Table below.

| Input 1 (message) | input 2 (reference) | output logic |
|---|---|---|
| 0 | 0 | 1 |
| 0 | 1 | 0 |
| 1 | 0 | 0 |
| 1 | 1 | 0 |

Back to the block diagram, a simple level shifter at the output of the logic circuit makes the encoded message to become bipolar.The differential PSk signal is produced by multiplication with the carrier signal.

***Performance comparison of digital modulation schemes***

1. If bandwidth is of primary importance, FSK is not uset since bandwidth utilisation is poor, when compared to ASK and PSK.
2. If power requirement is most important, differential phase shift keyinhg is most desirable. ASK is least desirable.

| Scheme | Input $S_1(t)$: $S_2(t)$ | Band width | Signal to Noise ratio | Equipment complexity | comments |
|---|---|---|---|---|---|
| FSK | $S_1(t)=A\cos(\omega_c-\omega_d)t$ $S_2(t)=A\cos(\omega_c-\omega_d)t$ | >2B | 10.6 | Major | seldom used |
| PSK | $S_1(t)=A\cos\omega_c t$ | = 2B | 8.45 | Major | used for high speed data communication |
| Differential PSK | same as above | = 2B | 9.3 | Moderate | Most commonly used |
| ASK | Same as above | 2B | 14.45 | Moderate | Rarely used |

## M' ary PSK system

Instead of just varing the phase, frequency or amplitude of the RF signal, modern modulation techniques allow both envelope (amplitude) and phase (or frequency) of the RF carrier to vary. Because the envelope and phase provide two degrees of freedom such modulation technique make base band data into four or more possible RF carrier signals. Such modulation techniques are known as $M'$ ary modulation. In $M'$ ary modulation scheme two or more bits are grouped together to form symbols and one of possible signals $S_1(t)$ $S_2(t)$ ..... $S_m(t)$ is transmitted during each symbol period $T_S$. Normally, the number of possible signals is $M=2n$, where $n$ is an interger.

Sampling, quantizing, and coding are the initial stages in PSK and other schemes of modulation. In binary system, pulses with 2 possible levels are used. (0,1). But in $M'$ ary system, the pulses are allowed to take one of $m$ possible levels instead of 2 ( $m$>2). Each level corresponds to a definite input symbol (not like 011 in binary).

Let there be $N$ quantization levels. Let us assume that there are 4 voltages representing an input sample. Let the corresponding codes be 0, 1, 2, 3. Then, how many bits would be necessary or (sufficient) to describe the sampled levels ?

To make things easier, let us take up the binary system in which there are $T_n$ quantized levels and let $N=8$ levels. $N$ represens a maximum of

8, we need 3 bits to represent the sampled pulses as follows:

$$\left.\begin{array}{l}000 \text{ for } 0\\001 \text{ for } 1\\010 \text{ for } 2\\011 \text{ for } 3\end{array}\right\}\quad\left.\begin{array}{l}100 \text{ for } 4\\101 \text{ for } 5\\110 \text{ for } 6\\111 \text{ for } 7\end{array}\right\}\quad \ldots\ \ldots\ldots\ldots\ldots\ 8 \text{ levels}$$

$2^3 = 8$ number of quantized levels.

where $3 \rightarrow$ number of bits are sufficient to describe sampled level and $2 \rightarrow$ bin ary system.

In the M ary system, we may write

$$N = 4^x$$

where $N$ is the number of quantized levels and $x$ is the number of symbols needed to represent a sample. (Codes given, symbols are founded out)

$$4^x = N$$

In general,

$$m^x = N$$

where $m$ is the number of codes that correspond to the input levels. In our discussion in the previous paragraph,. $m = 4$

If, for example, $N = 32$,

we have $4^x = 32$

$$x \log_{10} 4 = \log_{10} 32$$

$$\therefore x = \frac{\log_{10} 32}{\log_{10} 4} = \frac{1.505}{0.602}$$

$$= 2.5$$

We need 2.5 symbols to represent a sample. When compared with the binary system,

$$2^x = 32$$

$$x = \frac{\log_{10} 32}{\log_{10} 2} = \frac{1.505}{0.3010} = 5$$

ie, we may need 5 symbols to represent the sample. We find that the number of symbols is less in M' ary system and it needs lesser bandwidth than the binary system.

As the number of signals or number of M increases the error also increases. It is seen that higher - order modulation exhibit - higher error rates. Increasing the data rate will increase the SNR. But increaseing bit rate will cause more noise.

The main disadvantage with M ' ary is that the transmitter needs more power because noise is more affective due to larger number of

## Correlative coding and Duo binary signaling

We have seen that ISI as an undesirable phenomenon that produces a degradation in system performance. But by adding ISI into the transmitted signal in a controlled manner, it is possible to achieve a bit rate of $2B_0$ bits per second in a channel of bandwidth $B_0$ Hz. Such a scheme is *correlative coding* or partial response signaling scheme. One such example is Duo binary signaling. Duo means transmission capacity of system is doubled.

In telecommunication, bipolar encoding is a type of return-to-zero (RZ) line code, where two non zero values are used so that the three values are +, −, and zero. Such a signal is called duo binary signal. Standard bipolar encoding are designed to be DC - balaced, spending equal amounts of time in the + and − states.

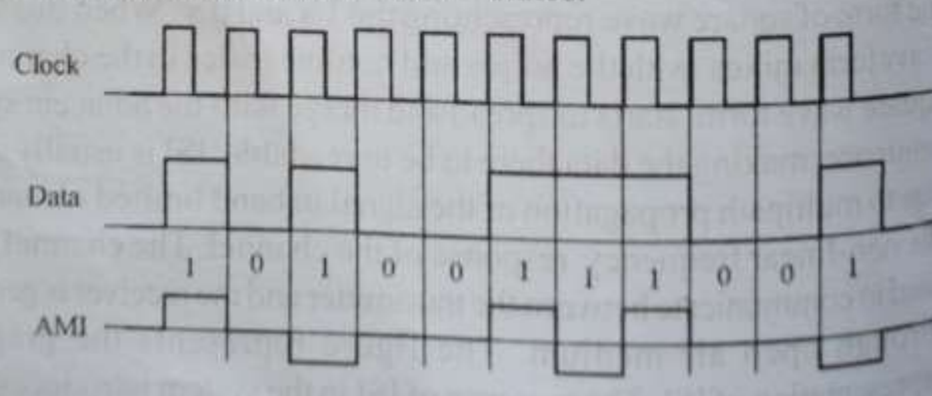



*Fig. 85. An example of bipolar encoding, known as AMI (Alternate mark inversion)*

The reason why Bipolar encoding is classified as a return to zero (RZ) is because when a bipolar encoded channel is idle, the line is held at a constant 'zero' level; and when it is transmitting bits, the line is either in a $+V$ or $-V$ state corresponding to the binary bit being transmitted. Thus the line always is returned to zero level to denote optionally a separation of bits or to denote idleness of the line.

**Duobinary encoding**

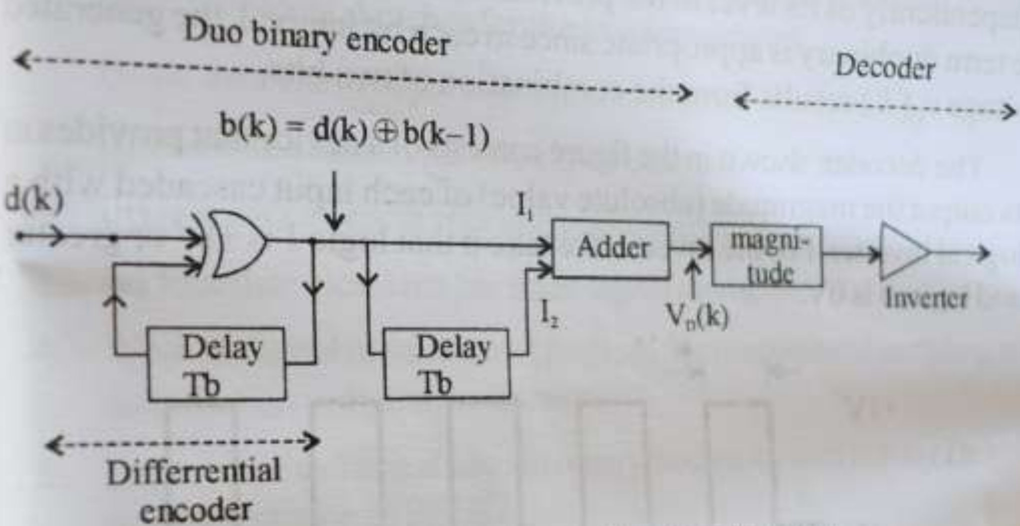



*Fig. 86. Duo binary encoder / decoder system*

One method for duobinary encoding and decoding is shown in figure. The signal $d(k)$ is the data bit stream with bit duration $T_b$. It makes excussions between logic 1 and 0, and we take the corresponding voltage levels to be + 1 V and − 1V. The signal $b(k)$ as the output of the differential encoder also makes excursion between +1 V and −1 V. The wave form $v_D(k)$ is therefore

$$v_D(k) = b(k) + b(k-1)$$

which can take on the values $v_D(k = +2V,\ 0\,V$ and $-2\,V)$. The values of $v_D(K)$ in any interval $k$ depends on both $b(k)$ and $b(k-1)$. Hence there is a correlation between the values of $v_D(k)$ in any two sucessive intervals. For this reason the coding of the above figure is refered to as *correlative coding*.

The wave form $d(k)$ has the appearance of square wave of period $2T_b$ and frequency $(1/2T_b)$ is shown in fig. (a). If $d(k)$ is the input to the luobinary encoder, $b(k)$ appears as in fig (b). The form $v_D(k)$ which is o be transmitted appears as in fig. (c).

The correlation can be made apparent in another way. When the ransition is made from one interval to the next, it is not possible for $v_D(k)$ to change from + 2V to – 2V or vice-versa. In short, in any nterval, $v_D(k)$ cannot always assume any of the possible levels ndependently of its level in the previous intervals. Finally, we note that he term duobinary is appropriate since in each bit interval, the generated voltage $v_D(k)$ results from the combination of two bits.

The decoder, shown in the figure consists of a device that provides at its output the magnitude (absolute value) of each input cascaded with a logical inverter. For the inverter we take it that logic 1 is +IV or greater and logic 0 is 0V.

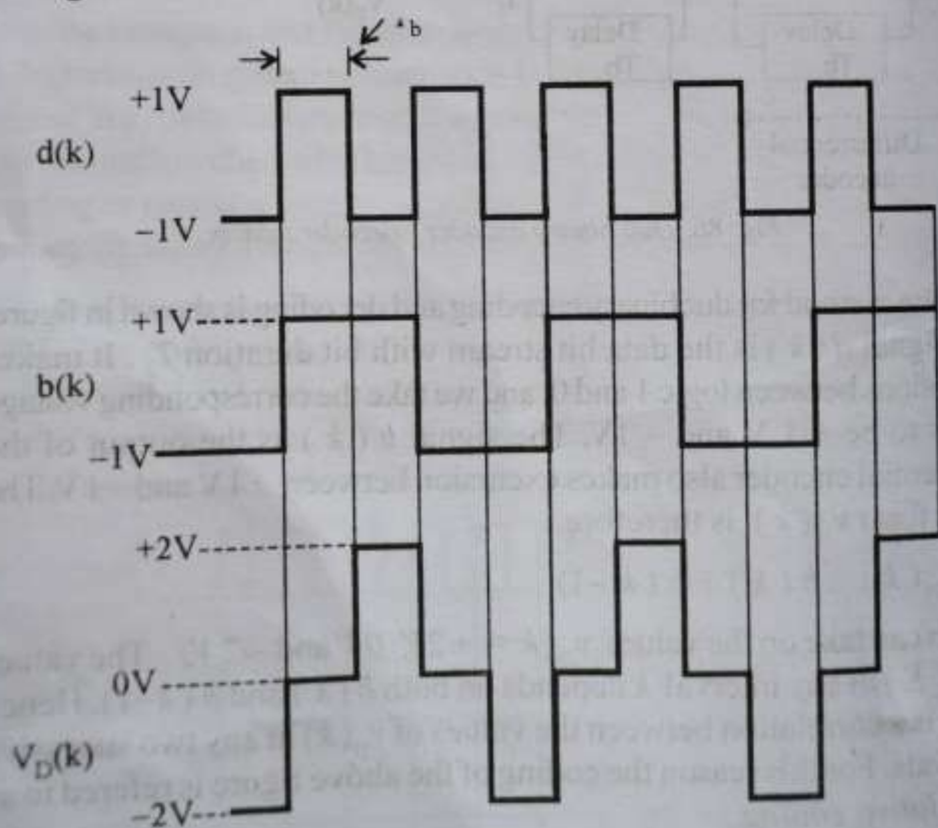



*Fig. 87. Waveforms of d(k), b(k) and $v_D(k)$*

## Questions

1. List out digital modulation techniques. Sketch the modulated wave form in each case.
2. Draw a sketch of a sine wave in time domain and frequency domain and explain its features.
3. Explain the difference between signal element and data element. Give graphical sketches for the following cases:

   i) one data element per one signal element.

   ii) one data element per two signal element.

   iii) two data elements per one signal elements.

   iv) four data elements per three signal elements.
4. What is digital modulation? Explain the processes of sampling and quantization in digital modulation.
5. Explain the technique used in binary frequency shift keying. Describe implementation of BFSK.
6. Explain binary phase shift keying. Describe its implementation.
7. Explain quadrature phase shift keying.
8. Explain differential encoding in the differential phase shift keying technique (DPSK) .
9. Compare the performances of different digital modulation schemes.
10. Explain M' ary PSK system. Write its specific advantage and disadvantage.
11. Explain correlative coding and Duobinary encoding.

***Objective type questions***

1. Most commonly used digital modulation scheme is
   (a) ASK (b) BFSK (c) BPSK (d) DPSK
2. The first step in pulse code modulation is
   (a) A/D conversion (b) D/A conversion
   (c) quantization (d) sampling
3. It a signal has L levels, each level during sampling needs
   (a) L bits (b) 2L bits
   (c) $\log_2 L$ level (d) $\log_{10} L$ level
4. If the number of bits in the code is to be $n$, then the signal must be divided into _______ number of equal levels.
   (a) $n$ (b) $n/2$ (c) $2^n$ (d) $2n$
5. The process of assessing binary values to each sample is known as
   (a) quantization (b) sampling (c) encoding (d) decoding
6. BFSK can be implemented using
   (a) voltage controlled oscillator (b) low pass filter
   (c) band pass filter (d) differential amplifier
7. Compared to binary system, the number of symbols necessary to represent a sample in M' ary system is
   (a) greater (b) lesser (c) rational (d) irrational
8. It needs more power to trans mit with M'ary PSK due to
   (a) samll number of voltage levels
   (b) large number of voltage levels
   (c) high antnna rsistance
   (d) none of the above
9. The possible phase angles of quadrature PSK modulated signal are
   (a) $0, \pi$ (b) $45^0, 135^0$
   (c) $45^0, 135^0, -45^0, -135^0$ (d) $90^0, -180^0, 270^0, -360^0$
10. To transmit analogue message signal over digital carrier, the digital modulation scheme suitable is
    (a) pulse code modulation (b) phase shift modulation
    (c) single sideband transmission (d) any one of the above

11. The information is encoded as phase changes between adjacent symbols. The digital modulation scheme is
    (a) PSK (b) BFSK (c) DPSK (d) ASK
12. In low speed digital modulation schemes, which one is used
    (a) FSK (b) PSK (c) DPSK (d) anyone
13. Phase schift keying is
    (a) linear modulation (b) non-linear modulation
    (c) complex modulation (d) none
14. In quadrature PSK scheme, how many PSK systems are used:
    a) one b) two c) three d) four
15. The advantage of digital modultion is
    (a) the bits can be transmitted error free
    (b) the bits can be transmitted with lesser carrier power than analog system.
    (c) both (a) and (b) (d) neither (a) nor (b)
16. The method inwhich small amount of controlled ISI is introduced into the data stream rather than trying to eliminate it completely is called
    (a) correlative coding (b) Duobinary coding
    (c) partial response signalling (d) all the above.
17. In duobinary signalling method, for M-ary transmission, the number of output obtained is
    (a) 2 M (b) 2M+1 (c) 2 M−1 (d) $M^2$
18. The interference caused by the adjacent pulses in digital transmission is called
    (a) Inter symbol interference (ISI) (b) White noise
    (c) Image frequency interference (d) Transit time noise

*Answers:*

1. (d) 2. (d) 3. (c) 4. (c) 5. (a) 6. (a) 7. (a)
8. (b) 9. (c) 10. (a) 11. (c) 12. (a) 13. (b) 14. (b)
15. (c) 16. (d) 17 (c) 18. (a)