UNIT-I

ANALOG MODULATION

Amplitude Modulation – AM, DSBSC, SSBSC, VSB – PSD, modulators and demodulators – Angle modulation – PM and FM – PSD, modulators and demodulators – Superheterodyne receivers

AMPLITUDE MODULATION

1.(a)Explain Communication system with the basic diagram.

Or

(b)With neat diagram,explain the components of communication system.

Communication system

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

In our day to day life we come across many ways of communication. For communication to take place, three essential things must be present i.e.

Sender or transmitter:

It sends information. For example TV transmitting station or radio transmitting stations are 'senders', since they transmit information

Receiver:

It receives information. For example all TV sets ad radios are receivers. They get information from transmitter.

Communication channel:

This is the path through which the signal propagates from transmitter to receiver.

Now let us consider different examples of communication that we use in our day to day life.

- Telephone
- Mobile phones(cellular phones)

Basic communication systems

We know that for any communication to take place, three things are essential. They are,

1. Transmitter
2. Receiver

3. Channel or transmission median

The above three things are present in any communication system. Whenever two people are talking with each other, it becomes a communication system. In that case voice signal is exchanged and the transmission medium is air. Fig.1.1 shows the block diagram of basic communication system.

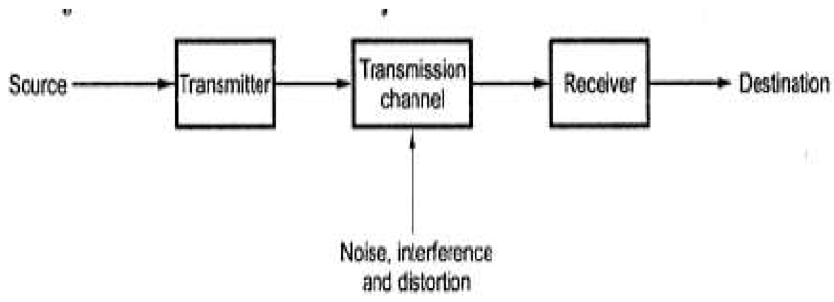


Fig. 1.1 Basic communication system

As shown in fig 1.1 the source generators the message to be transmitted. The transmitter or talker sends the message over the transmission channel. The transmission channel can be the median such as electric conductors, air. Or light (in case of optical fibre). The receiver or listener receives the message the message from transmission channel. It is then given to the destination by the receiver. During the transmission over the channel, the message is distorted and it becomes noisy.

The transmitter is required to make the signal suitable for conduction over the channel. And the receiver is required to convert the signal from transmission channel and make suitable for destination.

Modulation

Modulation is the process of placing the message signal over some carrier to make it suitable for transmission over long distance.

The carrier signal is basically of higher frequency than of message. And it has the ability to travel over long distance.

Demodulation:

The modulated carrier signal is transmitted by the transmitter. The receiver receives this carrier. It separates the message signal from the modulated carrier. This process is exactly opposite that of modulation at the transmitter.

Demodulation is the process of separating message signal from the modulated carrier signal.

Communication channel

The connection between transmitter and receiver is established through communication channel. The common problems associated with the channel are:

1. **Additive noise interference**: This noise is generated due to internal solid state devices and resistors etc. Used to implement the communication system.
2. **Signal attenuation**; It occurs due to internal resistance of the channel and fading of the signal.
3. **Amplitude and phase distortion**: The signal is distorted in amplitude and phase because of nonlinear characteristics of the channel.
4. **Multipath distortion**: This distortion occurs mostly in wireless communication channels. Signals coming from different paths tend to interfere with each other.

There are two main resources available with the communication channels. These two resources are

1. **Channel bandwidth** :

This is the maximum possible range of Frequencies that can be used for transmission. For example the bandwidth offered by wire line channels is less compared to fibre optic channel.

2. **Power in the transmitted signals :**

This is the power that can be put in the signal being transmitted. The effect of noise can be minimized by increasing the power. But this cannot be increased to very high value because of the equipment and other constraints.

2.(a)Why digital communication is so popular?write the advantages and disadvantages of digital communication..

Or

(b)Explain why digital communication is popular.enumerate its advantages and disadvantages.

There are few reasons due to which people are preferring digital communication over analogue communication.

1. Due to advancement in VLSI technology, it is possible to manufacture very high speed embedded circuits. Such circuit are used in digital communication.
2. High speed computers and powerful software design tools are available. They make the development of digital communication systems feasible.
3. Internet is spread almost in every city and towns. The compatibility of digital communication system with internet has open new area of applications.

Advantages and disadvantages of digital communication

Advantages:

1. Because of the advances in digital IC technologies and high speed computers digital communication system are simpler and cheaper compared to analogue systems.
2. Using data encryption, only permitted receivers can be allowed to detect the transmitted data. This is very useful in military application.
3. Wide dynamic range is possible since the data is converted to the digital form.
4. Using multiplexing, the speech, video and other data can be merged and transmitted over common channel.
5. Since the transmission is digital and channel encoding is used the noise does not accumulate from repeater to repeater in long distance communication.
6. Since the transmitted signal is digital a large amount of noise interference can be tolerated.
7. Since channel coding is used the errors can be detected and corrected in the receivers.
8. Digital communication is adaptive to other advanced branches of data processing such as digital signal processing, image processing, and data compression.

Disadvantages

Even though digital communications offer many advantages as given above, it has some drawbacks also.

But the advantages of digital communication outweigh disadvantages. They are as follows

1. Because of the analogue to digital conversion the data rate become high. Hence more transmission band with this required for digital communication.
2. Digital communication means synchronization in case of synchronous modulation.

Need of modulation or advantages of modulation:

The advantages of modulation are:

1. Easy of radiation
2. Adjustment of bandwidth
3. Reduction of height of antenna
4. Avoid mixing of signals
5. Increase the range of communication
6. Multiplexing, and
7. Improves quality of reception.

Principles of amplitude modulation

3.(a).Derive the expression for amplitude modulation.

Or

(b).Derive the expression that the bandwidth is twice the modulating frequency.

Definition:

Amplitude modulation is the process by which amplitude of the carrier signal is varied in accordance with the instantaneous value of the modulating signal, but frequency and phase remains constant.

Mathematical Expression of AM
AM Envelope

In amplitude modulation, the amplitude of a carrier signal is varied according to variations in the amplitude of modulating signal.

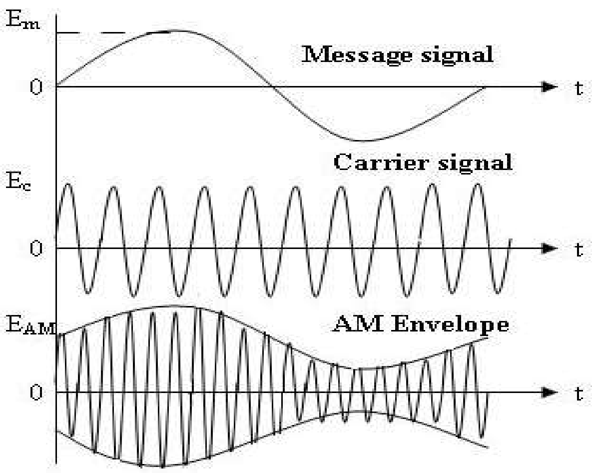


Fig. 1.1 AM waveform

Fig.1.1shows the modulating signal, high frequency carrier and amplitude modulated signal.It is observed from fig.1.1 that the carrier frequency remains same, but its amplitude varies according to amplitude variations of the modulating signal.

Let the instantaneous modulating signal be $e_m(t)$

$$e_m(t) = E_m \sin \omega_m t \qquad \text{..............................(1.1)}$$

and the instantaneous carrier signal be $e_c(t)$

$$e_c(t) = E_c \sin \omega_c t \qquad \ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots(1.2)$$

where Em is maximum amplitude of modulating signal.

Ec is maximum amplitude of carrier signal

ω_m is the frequency of modulating signal

ω_c is the frequency of carrier signal

The mathematical expression for the complete Amplitude modulated wave is

$$E_{AM} = E_C + e_m(t)$$

$$= E_c + E_m \sin \omega_m t$$

The instantaneous value of the amplitude modulated wave is

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

$$e_{AM} = (E_C + Em \sin \omega_m t) \sin \omega_c t \qquad \ldots\ldots\ldots\ldots\ldots\ldots\ldots(1.3)$$

Frequency Spectrum and Bandwidth

Consider the expression for AM wave

$$e_{AM} = (E_C + Em \sin \omega_m t) \sin \omega_c t \qquad \ldots\ldots\ldots\ldots\ldots\ldots\ldots(1.4)$$

$$\text{We have } m = \frac{E_m}{E_c} \qquad \ldots\ldots\ldots(1.5)$$

$$e_{AM} = (E_C + mE_C \sin \omega_m t) \sin \omega_c t$$

$$= E_C (1 + m \sin \omega_m t) \sin \omega_c t$$

$$= E_C \sin \omega_c t + mE_C \sin \omega_m t \sin \omega_c t$$

$$e_{AM} = E_C \sin \omega_c t + \frac{mE_c}{2} \operatorname{Cos}(\omega_c - \omega_m) t - \frac{mE_c}{2} \operatorname{Cos}(\omega_c + \omega_m) t$$

$$[\sin A \sin B = \frac{1}{2}(\cos(A-B) - \cos(A+B))]$$

$$e_{AM} = E_C \sin 2\pi f_c t + \frac{mE_c}{2} \cos 2\pi (f_c - f_m)t - \frac{mE_c}{2} \cos 2\pi (f_c + f_m)t$$

$$\boxed{e_{AM} = E_C \sin 2\pi f_c t + \frac{mE_c}{2} \cos 2\pi f_{LSB}{}^t + \frac{mE_c}{2} \cos 2\pi f_{USB}{}^t}$$

…………………..(1.6)

ie. $E_C 2\pi f_c t$ represents unmodulated carrier

second term represents lower side band

third term represents upper side band.

The above equations contain full carrier and both the side bands. Hence it is called double side band full carrier system. Above equation indicates AM envelope is a complex wave made up of a dc voltage, the carrier frequency and the sum ($f_c + f_m$) and difference (f_c - f_m) frequencies. The sum and difference frequencies are displaced from the carrier frequency by an amount equal to the modulating signal frequency. Therefore an AM signal spectrum contains frequency components spaced f_m Hz on either side of the carrier.

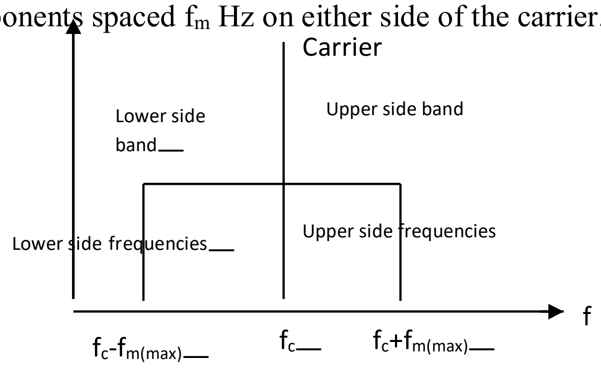


Fig.1.2 Frequency spectrum of AM

The AM spectrum extends from f_c-$f_{m(max)}$ to f_c+$f_{m(max)}$,

where

f_c is the carrier frequency

and $f_{m(max)}$ is the highest modulating signal frequency.

The band of frequencies between f_c-$f_{m(max)}$ and f_c is called lower side band frequency (LSB) and the band of frequencies between f_c and f_c-$f_{m(max)}$is called upper side band frequency (USB)

The Bandwidth of the signal

$$BW = f_{USB} - f_{LSB}$$

$$BW = f_c+f_{m(max)} - f_c-f_{m(max)}$$

$$BW = 2f_{m(max)} \quad \text{......................(1.7)}$$

Coefficient of Modulation and Percentage Modulation

The ratio of maximum amplitude of modulating signal to maximum amplitude of carrier signal is called **Coefficient of Modulation or Modulation Index.**

ie. $$\boxed{m = \frac{E_m}{E_C}} \quad \text{.....................(1.8)}$$

Value of E_m should be less than value of E_C to avoid any distortion in the modulated signal.Hence maximum value of modulation index will be equal to 1 when $E_m = E_c$.

Minimum value will be zero.

When modulation index is expressed in percentage it is also called **Percentage Modulation.**

AM POWER DISTRIBUTION:

4.(a).Write the expression for AM power distribution.

Or

(b).with mathematical expression,explain AM power distribution.

AM signal has three components

Unmodulated carrier

Lower side band and

Upper side band.

The total power of AM wave is sum of carrier power P_C and powers in the two side bands P_{USB} and P_{LSB} (ie)

$$P_{Total} = P_C + P_{USB} + P_{LSB} \qquad \text{......................(1.16)}$$

$$P_{Total} = \frac{E^2{}_{carr}}{R} + \frac{E^2{}_{LSB}}{R} + \frac{E^2{}_{USB}}{R} \qquad \text{...................(1.17)}$$

Here all the three voltages are rms values and R is characteristic impedance of antenna in which the power is dissipated.

The carrier power

$$P_C = \frac{E^2{}_{carr}}{R} = \frac{\left(E_C/\sqrt{2}\right)^2}{R} \qquad \text{...................(1.11)}$$

$$P_C = \frac{E_C{}^2}{2R} \qquad \text{...................(1.18)}$$

The power of USB and LSB are same.

$$P_{LSB} = P_{USB} = P_C = \frac{E_{SB}{}^2}{R} \qquad \text{...................(1.19)}$$

E_{SB} is rms voltage of side bands peak amplitude of both side bands is $\frac{mE_C}{2}$

Hence $E_{SB} = \frac{mE_C}{2}$

$$P_{LSB} = P_{USB} = \left(\frac{mE_C/2}{\sqrt{2}}\right)^2 \cdot \frac{1}{R} \qquad \text{...................(1.20)}$$

$$P_{LSB} = P_{USB} = \frac{m^2 E_C{}^2}{8R} \qquad \text{...................(1.21)}$$

Total power

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

$$= \frac{E_C^{\,2}}{2R}\left[1+\frac{m^2}{4}+\frac{m^2}{4}\right]$$

$$= \frac{E_C^{\,2}}{2R}\left(1+\frac{m^2}{2}\right)$$

$$P_{Total} = P_C\left(1+\frac{m^2}{2}\right)$$

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

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

$$\boxed{m = \sqrt{2\left(\frac{P_{total}}{P_C}-1\right)}}$$

Transmission Efficiency:

The transmission efficiency of an AM wave is the ***"ratio of the transmitted power which contains the information (i.e., the total sideband power) to the total transmitted power"***

$$\%\,\eta = \frac{m_a^2}{2+m_a^2} \times 100$$

If $m_a = 1$ then $\%\,\eta = 1/3 \times 100 = 33.3\%$

Only 33.3% of energy is used and remaining power is wasted by the carrier information along with the sidebands.

The maximum transmission efficiency of the AM is 33.3%. This means, that only one third of the total power is carried by the sidebands and the rest two third is a waste and is transmitted only for a low cost reception system.

Advantages, disadvantages and applications of AM

Advantages:

i) AM has the advantage of being usable with very simple modulators and demodulators.

ii) AM is a relatively inexpensive.

iii) AM wave can travel a long distance.

iv) It covers larger area than FM

Disadvantages:

i) Poor performance in the presence of noise.

ii) Inefficient use of transmitter power.

iii) Wastage in bandwidth.

Applications:

i) Low quality form of modulation that is used for commercial broadcasting of both audio and video signals.

ii) Two – way mobile radio communications such as citizens band radio.

iii)Aircraft communication in the VHF frequency range.

AM modulators and Demodulators

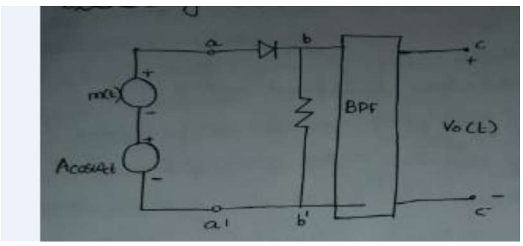
5.(a)Explain the methods to generate AM signal.

Or

(b)Explain in detail about Switching Modulator.

AM Modulators

Switching modulator: In switching modulator the diode conducts when the combined message and carrier signal is positive.since the carrier is stronger than message signal the switching of diode is regulated by a carrier only.The switching action can be approximated by a pulse train.



6.(a).Explain in detail about AM Demodulators

Or

(b).How AM signals are demodulated?.Explain in detail.

Demodulation of AM Wave

Demodulation or detection is the process by which the modulating voltage is recovered from the modulated signal. It is the reverse process of modulation. The devices used for demodulation or detection is called demodulators or detectors. For amplitude modulation, detectors or demodulators are classified as

- Square law detectors
- Envelope detectors.

AM signal with large carrier are detected by using the envelope detector. The envelope detector used in the circuit extracts the envelope of the AM wave is the baseband or modulating signal. But a low-level amplitude modulated signal can only be detected by using square-law detectors in which a device operating in the non-linear region is used to detect the modulating signal.

Square law detectors

In utilizes the non-linear region of voltage current dynamic characteristic of a diode. This dynamic characteristic is high non-linear, particularly in the low voltage region. Fig 1.12 shows the circuit of a square law detector. It may be observed that the circuit is very similar to the square in a square law modulator. The only difference lies in the filter circuit. In a square law modulator the filter used is a band pass filter, whereas in a square law detector, a low pass filter is used.

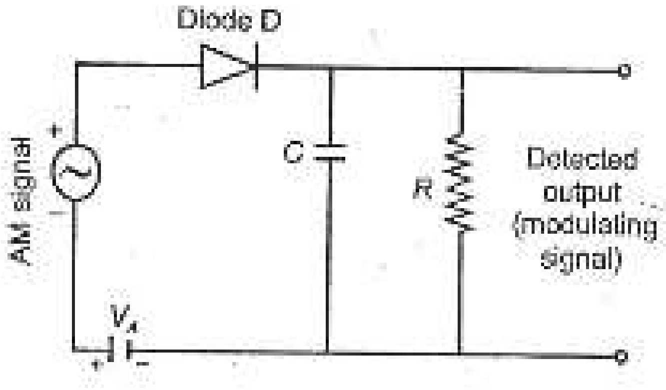


Fig 1.12 square law detector

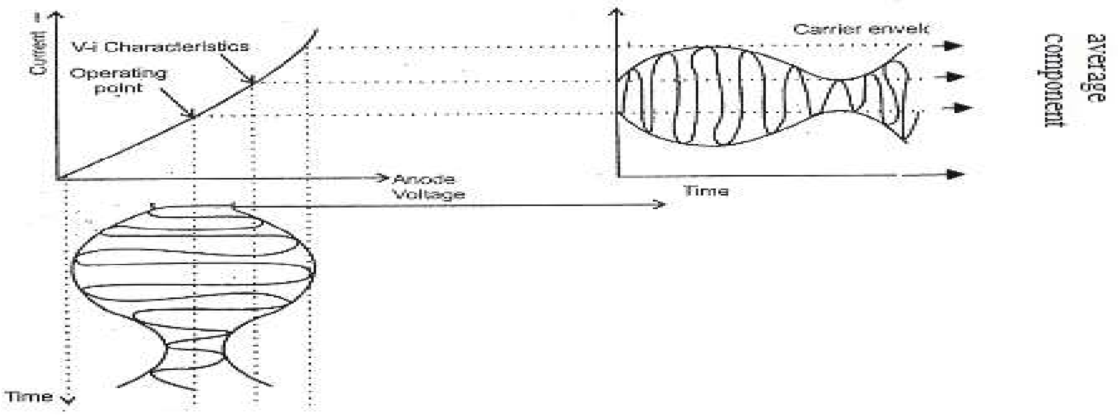


Fig 1.13 Wave shape of applied voltage and resulting currents in square law diode modulation

In the circuit the d.c. supply votage V_A is used to get a fixed operating point in the non-linear portion of the diode V-I characteristic. Since the operation is limited to the non-linear region of the diode characteristics, the lower half portion of the modulated wave form is compressed. This produces envelope applied distortion. Due to this, the average value of the diode-current is no longer constant, rather it varies with time as shown in fig 1.13.

This distorted output diode current is expressed by the non-linear V-I relationship (i.e. square law) as

$$i = a_1 V + a_2 V^2 \qquad (1.51)$$

V- modulating signal

We know that AM is expressed as

$$V = V_A + A_c(1 + m_a \sin \omega_m t)\sin \omega_c t \qquad (1.52)$$

Substitute of value of V in eqn.(1.52)

$$i = a_1(V_A + A_c(1 + m_a \sin \omega_m t)\sin \omega_c t) + a_2 \left[V_A + A_c(1 + m_a \sin \omega_m t)\sin \omega_c t\right]$$

$$= a_1 V_A + a_1 A_c \sin\omega_c t + a_1 A_c m_a \sin\omega_m t \sin\omega_c t + a_2 V_A^2 + a_2 A_c^2 (1 + m_a \sin\omega_m t)^2$$

$$\sin^2\omega_c t + 2a_2 V_A A_c (1 + m_a \sin\omega_m t)\sin\omega_c t$$

$$= a_1 V_A + a_1 A_c \sin\omega_c t + a_1 A_c m_a \sin\omega_m t \sin\omega_c t$$

$$+ a_2 V_A^2 + a_2 A_c^2 \left(1 + 2m_a \sin\omega_m t + m_a^2 \sin^2\omega_m t\right)\left(\frac{1 - \cos 2\omega_c t}{2}\right)$$

$$+ 2a_2 V_A A_c \sin\omega_c t + 2a_2 V_A A_c m_a \sin\omega_m t \sin\omega_c t$$

$$= a_1 V_A + a_1 A_c \sin\omega_c t + a_1 A_c m_a \sin\omega_m t \sin\omega_c t$$

$$+ a_2 V_A^2 + \frac{a_2 A_c^2}{2} - \frac{a_2 A_c^2 \cos 2\omega_c t}{2} + \frac{a_2 A_c^2}{2} 2m_a \sin\omega_m t - \frac{a_2 A_c^2}{2} 2m_a$$

$$\sin\omega_m t \cos 2\omega_c t + \frac{a_2 A_c^2}{2} m_a^2 \sin^2\omega_m t - \frac{a_2 A_c^2 m_a^2}{2} \sin^2\omega_m t \cos 2\omega_c t$$

$$+ 2a_2 V_A A_c \sin\omega_c t + 2a_2 V_A A_c m_a \sin\omega_m t \sin\omega_c t$$

The carrier (R.F) terms are bypassed through the capacitor and the circuit is tuned to modulating frequency ω_m. Thus the output contains ω_m and dc terms.

$$\therefore\ i = a_1 V_A + a_2 V_A^2 + \frac{a_2 A_c^2}{2} + a_2 A_c^2 m_a \sin\omega_m t + \frac{a_2 A_c^2 m_c^2}{4} \qquad (1.53)$$

Thus the modulating signal is recovered from modulated signal.

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

Diode Detector or Envelope Detector

The most commonly used AM detector is simple diode detector as the shown in Fig. 1.14. The AM signal at fixed IF is applied to the transformer primary. The signal at secondary is half wave rectified by diode D. This diode is the detector diode. The resistance R is load resistance to rectifier and C is the filter capacitor. In the positive half cycle of AM signal diode conducts and currents flows through R, whereas in negative half cycle, the diode is reverse biased and no current flows. Therefore only positive half of the AM wave appears across resistance R as shown in figure 1.15(B). The capacitor across capital R provides low impedance at the carrier frequency and much higher impedance at the modulating frequency. Therefore capacitor reconstructs the original modulating signal as sown in figure 1.15(C) and high frequency carrier is removed.

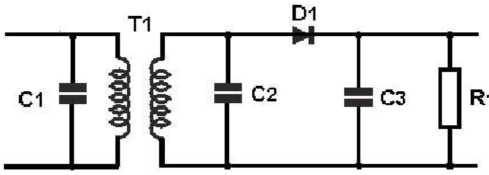


Fig1.14 Diode detector

Negative peak clipping in diode detector:

This is the distortion that occurs in the output of diode detector because of unequal *ac* and *dc* load impedances of the diode. The modulation index is defined as E_m / E_c.

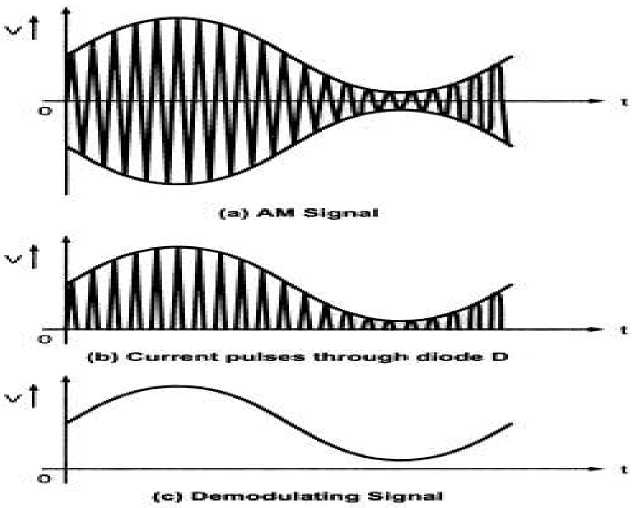


Fig. 1.15 Diode detector waveforms

Therefore it can also be defined as I_m / I_c with

$$I_m = E_m / z_m \qquad \text{and} \qquad I_c = E_c / R_c$$

Here Z_m is audio diode load impedance and R_c is the dc diode resistance. The audio load resistance of the diode is smaller than the dc resistance. Hence the AF current I_m is larger, in proportion to dc current. This makes the modulation index in the demodulated wave relatively higher than that of modulated wave applied at the detector input. This introduces the distortion due to over modulation in the detector signal for modulation index near 100%. This is illustrated in Fig.

In the figure observe that the negative peak of the detected signal takes place because of over modulation effect taking place in detector.

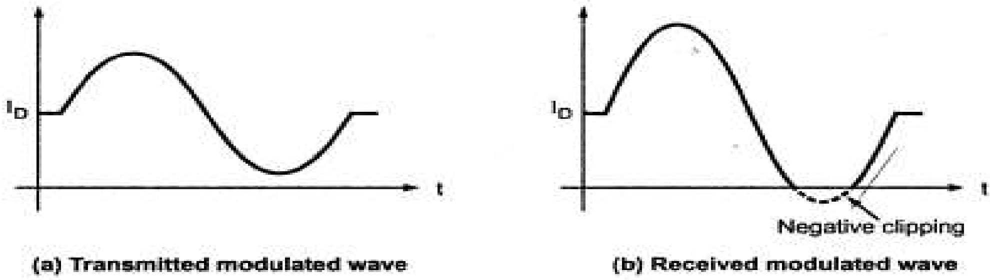


Fig.1.16 Negative peak clipping

Diagonal Clipping in Diode Detector

As the modulating frequency is increased, the diode ac load impedance, Z_m does not remain purely resistive. It does have reactive component also. At high modulation depths, the current changes so fast that the time constant of the load does not follow the changes. Hence the current decays slowly as shown in fig.1.17. The output voltage follows the discharge law of RC circuit. This introduces distortion in the detected signal and it is called diagonal peak clipping.

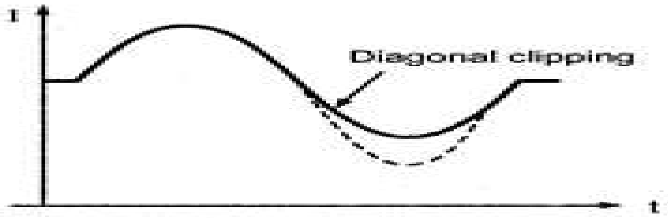


Fig.1.17 Diagonal peak clipping

DOUBLE SIDEBAND – SUPPRESSED CARRIER [DSB – SC]

7.(a).Derive the expression for DSBSC and its power calculation.

Or

b).Explain DSBSC and how power is saved in DSBSC with neat expression

In DSB – SC, the transmitted wave consists of only upper and lower sidebands. Transmitted power is saved here through the suppression of the carrier wave because it does not contain any useful information, but the channel bandwidth required is the same as before. Transmission

Bandwidth: The transmission bandwidth of DSB-SC is twice th frequency of the message signal.
BW=2fm

Expression for DSBSC

Let the modulating signal

$$V_m(t) = V_m \sin \omega_m t$$

The Carrier signal

$$V_c\ (t) = V_c \sin \omega_c t$$

$$V(t)_{DSB\text{-}SC} = V_c\ (t)\ \ V_m\ (t)$$

$$= V_c \sin \omega_c t\ V_m \sin \omega_m t$$

$$= V_c\ V_m\ \sin \omega_c t \sin \omega_m t$$

$$= V_m V_c\ /2(\ \mathrm{Cos}\ (\omega_c - \omega_m)\ t - \mathrm{Cos}\ (\omega_c + \omega_m)t)$$

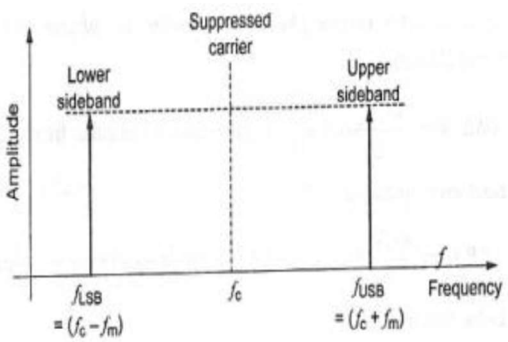


Fig 1.19 Frequency spectrum of DSB-SC-AM

In the above figure,it contains only two side bands.

Power Calculation: The total power transmitted in AM is

$$P_t = P_{carrier} + P_{LSB} + P_{USB} \quad (1.59)$$

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

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

$$= \frac{V_c^2}{2R}\left[1 + \frac{m_a^2}{2}\right]$$

$$P_t = P_c\left[1 + \frac{m_a^2}{2}\right] \quad (1.60)$$

Where

$$P_c = \frac{V_c^2}{2R}$$

If the carrier is suppressed, the the total power transmitted in DSB – SC – AM is,

$$P'_t = P_{LSB} + P_{USB} \quad (1.61)$$

$$= \frac{m_a^2 V_c^2}{8R} + \frac{m_a^2 V_c^2}{8R} = \frac{m_a^2}{2}\left[\frac{V_c^2}{2R}\right]$$

$$\boxed{P'_t = \frac{m_a^2}{2} P_c} \quad (1.62)$$

$$\text{Power savings} = \frac{P_t - P'_t}{P_t} \quad (1.63)$$

Substitute eqn.(1.60) and (1.62) in eqn.(1.63)

$$= \frac{P_c\left[1+\frac{m_a^2}{2}\right]-\frac{m_a^2}{2}P_c}{P_c\left[1+\frac{m_a^2}{2}\right]}$$

$$= \frac{P_c+P_c\frac{m_a^2}{2}-\frac{m_a^2}{2}P_c}{P_c\left[1+\frac{m_a^2}{2}\right]}$$

$$= \frac{1}{\left[1+\frac{m_a^2}{2}\right]} \qquad (1.64)$$

$$\% \text{ Power saving} = \frac{1}{\left[1+\frac{m_a^2}{2}\right]}\times 100$$

$$= \frac{2}{2+m_a^2}\times 100$$

If modulation index $m_a = 1$(100% modulation) then, the power saving is

$$= \frac{2}{3}\times 100$$

$$\boxed{P'_t = 66.67\%}$$

In DSB-SC 66.67% power is saved due to the suppression of the carrier wave.

Generation of DSBSC

8.(a)Explain how DSBSC signals are generated and Detected.

Or

(b)Briefly explain how DSBSC signals are modulated and demodulated

There are two ways of generating DSB-SC-AM

1.Balanced modulator

2.Ring Modulator

Balanced modulator

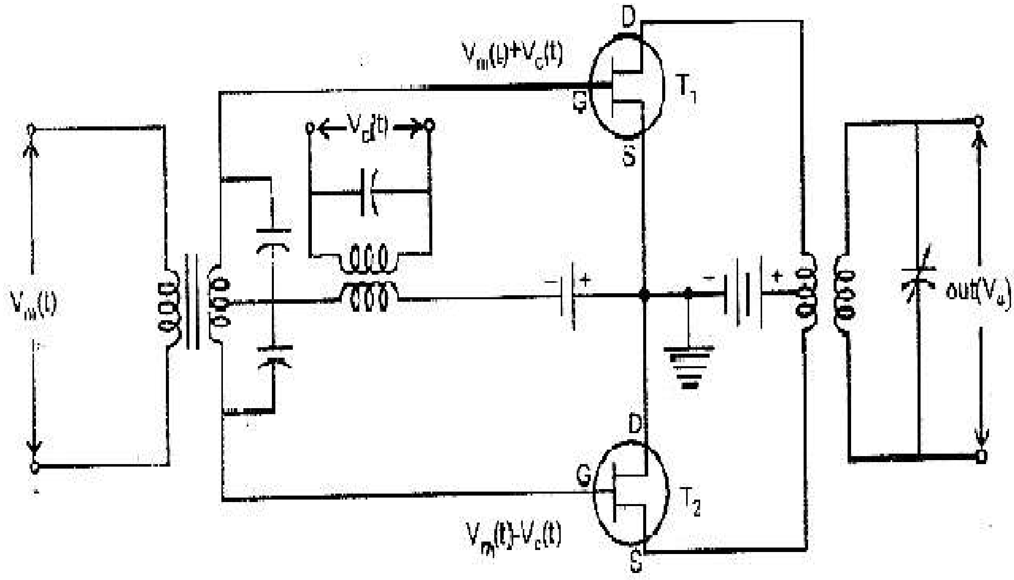


Fig1.11 Balanced modulator

Balanced modulator

A FET balanced modulator circuit used for AM generation is shown in Fig. 1.11 . In this modulator, two non-linear devices are connected in the balanced mode, so as to supply the currier wave i.e., it is assumed that the two FETs are identical and the circuit is symmetrical. Since the operation is confined in non-linear region of its transfer is equal and opposite in phase $V_c = - V'_c$

The input voltage to FET T_1 is given by

Similarly the input voltage to FET T_2 is given by

$$V_{GS} = C(t) + V_m(t)$$
$$= A_c \sin \omega_c t + A_m \sin \omega_m t$$

$$V'_{GS} = -C(t) + V_m(t)$$
$$= -A_c \sin \omega_c t + A_m \sin \omega_m t$$

By using the non-linearity relationship the drain current can be written as per square law equation.

$$i_d = a_1 V_{GS} + a_2 V^2_{GS}$$

$$i'_d = a_1 V'_{GS} + a_2 V'^2_{GS}$$

Substitute V_{GS} and V'_{GS} in i_d and i'_d

The output AM voltage V_0 is given by

$$V_0 = k(i_d - i'_d)$$

This is because i_d and i'_d flows in the opposite direction. K is a constant depending on impedance or other circuit parameters.

Substitute i_d and i'_d in V_0 and

$$m_a = \frac{2a2Am}{Ac}$$

Therefore,

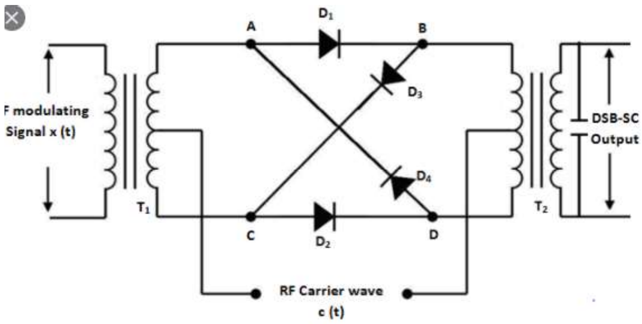
$$V_0 = 2ka_1A_c(1+m_a \sin\omega_m t)\sin\omega_c t$$

This same circuit can be used to generate DSB-SC-AM.The main difference between AM with carrier generation and DSB-SC-Amis that feeding points of the carrier and modulating signals are interchanged.

Advantages of balanced modulator over other non-linear modulator:

In simple non-linear circuit the undesirable harmonics are eliminated by a bandpass filter. In balanced modulator the undesirable harmonics are automatically balanced out, so the filter is not required.

Ring Modulator:



In ring modulator,on the positive half cycle of carrier diodes D1 and D2 are forward biased.At this time D3 and D4 are reverse biased and act like open circuits

On the negative half cycle of carrier the polarity of the carrier reverses,diodes D1 and D2 are reverse biased and diodes D3 and D4 conduct.The ring modulator is alson as double balanced modulator

Demodulation of DSBSC

Synchronous or coherent detector

Principle:

The synchronous or coherent detector uses locally generated carrier for detection. This technique needs the carrier which is in phase coherence with that of transmitter.

Block diagram:

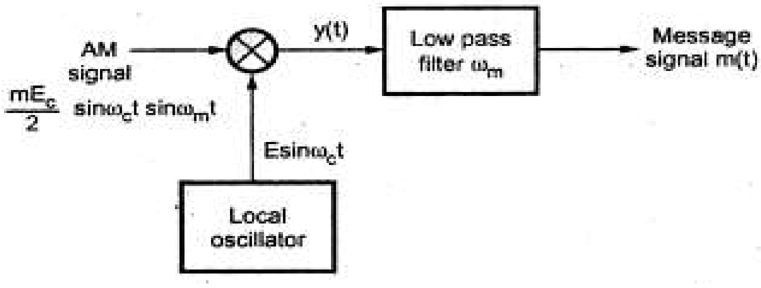


Fig 1.18 shows the block diagram of the synchronous detector.

Operation:

- The input signal can be DSB-SC or SSB-SC.
- It is multiplied by locally generated carrier of E sin
- The product signal is then passed through low pass filter of bandwidth

Mathematical analysis:

Let the input signal be DSB-SC signal,

$$m_1(t) = \frac{mE_c}{2} \sin\omega_c t \ \sin\omega_m t$$

When the signal is multiplied by local carrier,

$$\begin{aligned}
y(t) &= m_1(t) \cdot E\sin\omega_c t \\
&= \frac{mE_c}{2} \sin\omega_c t \ \sin\omega_m t \cdot E\sin\omega_c t \\
&= \frac{mEE_c}{2} \sin^2\omega_c t \sin\omega_m t \\
&= \frac{mEE_c}{2} \frac{1-\cos 2\omega_c t}{2} \sin\omega_m t \\
&= \frac{mEE_c}{4} \sin\omega_m t - \frac{mEE_c}{4} \cos 2\omega_c t
\end{aligned}$$

In above equation, the second term has frequency of $2\omega_c$. The low pass filter has the bandwidth of ω_m. Hence second term is not passed by low pass filter. Hence only first term appears at the output of low pass filter, i.e.

$$m(t) = \frac{mEE_c}{4} \sin \omega_m t$$

Thus the modulating signal is obtained at the output. Here note that even if modulation index is small, original signal is recovered.

SINGLE SIDEBAND SUPPRESSED CARRIER (SSB – SC – AM)

9(a)Write the mathematical expression for SSBSC

Or

(b).Why SSBSC is preferred over DSBSC and write its mathematical expression.

In a DSB signal, the basic information is transmitted twice, once in each sideband.
The sidebands are the sum and difference of the carrier and modulating signals, the information must be contained in both of them.
So either one sideband is enough for transmitting as well as recovering the useful message.
One sideband may be suppressed. The remaining sideband is called a single sideband suppressed carrier (SSBSC or SSB) signal.

Transmission Bandwidth

SSB requires half of the bandwidth of the DSB SC and use considerably less transmitted power.

BW = $\mathbf{f_m}$

The bandwidth of SSB –SC signal is f_m same as the bandwidth of the base band signal.
In order to suppress one of the sidebands, the input signal fed to the modulator 1 is 90° out of phase with that of the signal fed to the modulator '2'.

Expression for SSB – SC

Let

$$V_1(t) = V_m \sin(\omega_m t + 90)\, V_c \sin(\omega_c t + 90) \qquad (1.77)$$

$$V_1(t) = V_m \cos\omega_m t\, V_c \cos\omega_c t \qquad (1.78)$$

$$V_2(t) = V_m \sin\omega_m t\, V_c \sin\omega_c t \qquad (1.79)$$

Therefore,

$$V_{SSB}(t) = V_1(t) + V_2(t)$$

$$= V_m V_c \left[\sin\omega_m t \sin\omega_c t + \cos\omega_m t \cos\omega_c t\right] \quad (1.80)$$

We know that

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

Hence the equation (1.80) becomes,

$$V_{SSB}(t) = \frac{V_m V_C}{2} \cos(\omega_C - \omega_m)\, t \qquad (1.81)$$

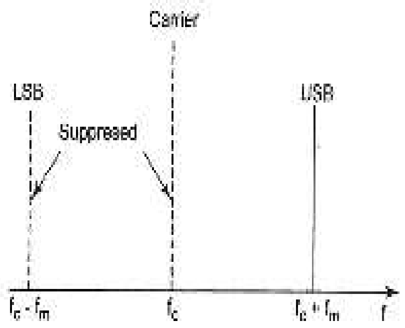


Fig1.26 frequency spectrum of SSB-SC-AM

Power calculation

10.(a)With mathematical expression explain how power is saved in SSBSC.

Or

(b)Write the mathematical expression for power saving in SSBSC

Total power saved in SSB-SC-AM is calculated as follows:
Power in SSB-SC-AM is

$P_t'' = \text{USB (OR) LSB} = \frac{1}{4} m_a^2 P_c$ (1.82)

Power savings with respect to AM with carrier,

$$\text{Power saving} = \frac{P_t - P_t''}{P_t} \qquad (1.83)$$

P_t = total power transmitted.

$$= \frac{\left[1+\frac{m_a^2}{2}\right]P_c - \left[\frac{m_a^2}{4}P_c\right]}{\left[1+\frac{m_a^2}{2}\right]P_c}$$

$$= \frac{\left[1+\frac{m_a^2}{2}-\frac{m_a^2}{4}\right]}{\left[1+\frac{m_a^2}{2}\right]}$$

$$= \frac{1+\frac{m_a^2}{4}}{1+\frac{m_a^2}{2}}$$

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

$$\text{Power saving} = \frac{4+m_a^2}{4+2m_a^2} \qquad (1.84)$$

If $m_a = 1$ then % power saving $5/6$ = 83.33%. In addition to carrier, one of the sidebands is also suppressed the power savings is 83.3% over AM with carrier.

Generation of SSB

11.(a) How SSB signals are generated? Explain

Or

(b)With neat block diagram ,explain the methods that generate SSB signal.

Or

(c)Explain how SSB signals are generated usingfilter method and phase shift method.

SSB – SC – AM waves can by generated in two ways.

1. Frequency discrimination (or) Filler method.
2. Phase discrimination method.

Phase discrimination method itself can be divided into two types.

1. Phase shift method.

2. Modified phase shift (or) weaver's method.

Suppression of Unwanted Sideband

In the previous subsection we studied the techniques to suppress the carr
ier. To generate single sideband suppressed carrier (SSBSC), we have to suppress the carrier as well as one of the sidebands. In this section let us consider the techniques to suppress one of the sidebands. These techniques are (i) filter method, (ii) phase shift method and (iii) The 'third' method.

Filter Method to Produce SSB

Fig. 1.28shows the block diagram of filter method to suppress one sideband. As shown in the block diagram, the balanced modulator produces DSB output. This DSB signal contains both the sideband. The filter must have a flat passband and extremely high attenuation outside the passband. In order to have this type of response the Q of the tuned circuits must be very high. The required value of Q factor increases as the difference between modulating frequency and carrier frequency increases. Carrier frequency is usually same as the transmitter frequency increases. Carrier frequency is usually same as the transmitter frequency. For higher transmitting frequencies them required value of Q is so high that there is no practical way of achieving it. In such situation, initial modulation is carried out at a low frequency carrier say 100 kHz by the balanced modulator. Then the filter suppresses one of the sidebands. The frequency of the SSB signal generated at output of filter is very low as compared to the transmitter frequency. The frequency is boosted up to the transmitter frequency by the balanced mixer and crystal oscillator. This process of frequency booting is also called as up conversion. The SSB signal having frequency equal to the transmitter frequency is then amplified by the linear amplifiers.

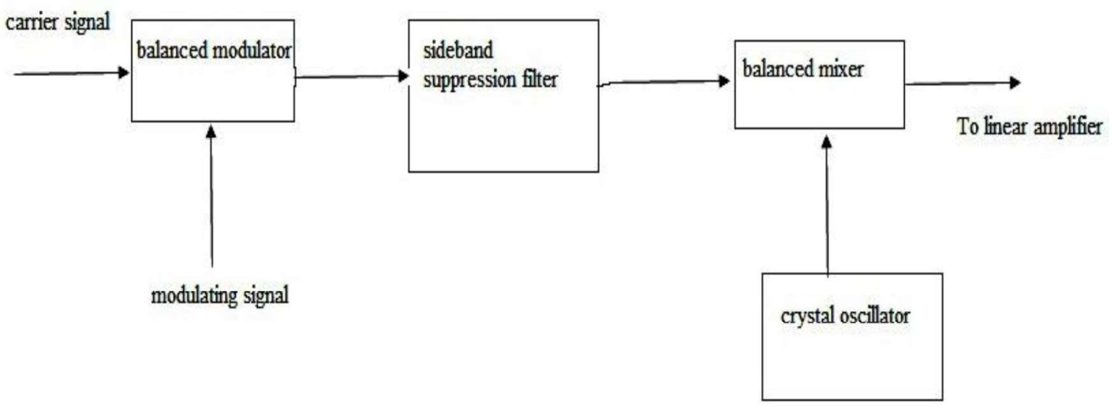


Fig.1.28 Filter method to suppress sideband

Phase Shift Method to Generate SSB

Fig.1.29 shows the block diagram of phase shift method to generate SSB. The carrier signal is shifted by 90° and applied to the balanced modulator M_1. The modulating signal is also directly applied to the balanced modulator M_2. The modulating signal is phase shifted by $90°$ and applied to balanced modulator M_2. Both the modulators produce an output consisting of only sidebands. The upper balanced modulator (M_1) generates upper sideband and lower sideband, but upper sideband is shifted by $+90°$ whereas lower sideband is shifted by -90.The output of balanced modulators are added by the summing amplifier. Since upper sidebands of both the modulators are phase shifted by $+90^0$, they are in phase and add to produce double amplitude signal. But lower sideband of the balanced modulator s are ($+90°$, $-90°$) $180°$ out of phase and hence cancel each other.

Thus the output of summing amplifier contains only upper sideband SSB signal. The carrier is already suppressed by balanced modulators.

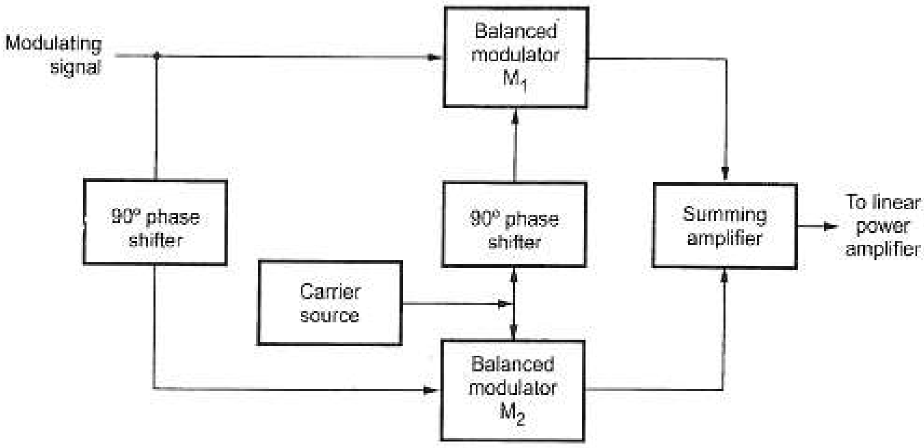


Fig.1.29 phase shift method to generate SSB

Let us see mathematically , how the sidebands add and cancel each other because of phase shifts. Input to the balanced modulator M_1 are $\sin \omega_m t$ and $\sin(\omega_c t + 90°)$. Hence output of M_1 will be,

$$\text{Output of } M_1 = \cos[(\omega_c t + 90°) - \omega_m t] - \cos[(\omega_c t + 90°) + \omega_m t]$$

$$= \cos(\omega_c t - \omega_m t + 90°) - \cos(\omega_c t + \omega_m t + 90°) \quad (1.85)$$

In the above equation observe that first term represents LSB with $+90°$ phase shift and second term represents USB with $+90°$ phase shift. Now inputs to the balanced modulator M_2 are, $\sin(\omega_m t + 90°)$ and $\sin \omega_c t$. Hence output of M_2 will be,

$$\text{Output of } M_2 = \cos[\omega_c t - (\omega_m t + 90°)] - \cos[\omega_c t + (\omega_m t + 90°)]$$

$$= \cos(\omega_c t - \omega_m t - 90°) - \cos(\omega_c t + \omega_m t + 90°) \quad (1.86)$$

In the above equation observe that first term represents LSB with $-90°$ phase shift and second term represents USB with $+90°$ phase shift. When signal of equation (1.85) and equation (1.86) add in the summing amplifier, the lower sidebands (first term) cancel each other since they are out of phase. The second terms add since they have same phase shift of $+90°$ (i.e., in phase). Thus SSB is generated at the output of summing amplifier.

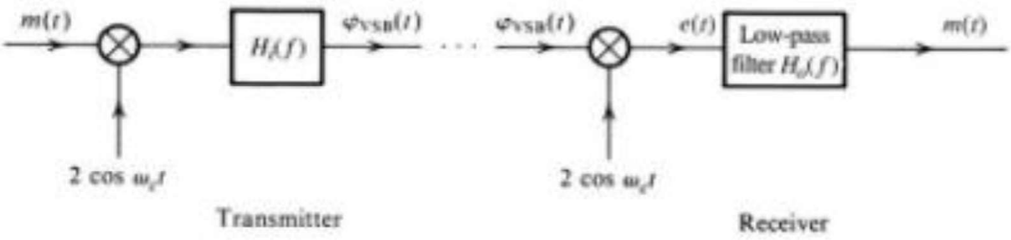
Vestigial Sideband

12.(a).Explain in detail about VSB.

Or

b)Write short notes on Vestigial Side band

In case of SSB modulation, when a sideband is passed through the filters, the band pass filter may not work perfectly in practice. As a result of which, some of the information may get lost. Hence to avoid this loss, a technique is chosen, which is a compromise between DSBSC and SSB, called as Vestigial Sideband (VSB) technique. The word vestige which means "a part" from which the name is derived. Both of the sidebands are not required for the transmission, as it is a waste. But a single band if transmitted, leads to loss of information. Hence, this technique has evolved. Vestigial Sideband Modulation or VSB Modulation is the process where a part of the signal called as vestige is modulated, along with one sideband.



VSB modulator and demodulator.

$$\varphi_{VSB}(f) = [M(f+f_c) + M(f-f_c)]H_i(f)$$

$$e(t) = 2\varphi_{VSB}(f)\cos\omega_c t \Leftrightarrow [\varphi_{VSB}(f+f_c) + \varphi_{VSB}(f-f_c)]$$

$$M(f) = M(f)[H_i(f+f_c) + H_i(f-f_c)]H_o(f)$$

$$H_o(f) = \frac{1}{H_i(f+f_c) + H_i(f-f_c)}$$

Advantages

- Highly efficient.
- Reduction in bandwidth.
- Filter design is easy as high accuracy is not needed.

- The transmission of low-frequency components is possible, without difficulties.
- Has good phase characteristics.

Disadvantages

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

Applications

The most prominent and standard application of VSB is for the transmission of television signals. Also, this is most convenient and efficient technique when bandwidth usage is considered.

COMPARISON OF AMPLITUDE MODULATION SYSTEMS

13.(a)Compare amplitude modulation systems.

Or

(b).Compare AM,DSB,SSB,VSB

Description	AM with carrier	DSB – SC – AM	SSB – SC - AM	VSB - AM
Band width	$2f_m$	$2f_m$	f_m	$f_m < BW < 2f_m$
Power Saving for Sinusoidal	33.33%	66.66%	83.3%	75%
Power Saving for non - Sinusoidal	33.33%	50%	75%	75%
Generation methods	Easier to generate	Not difficult	More difficult to generate	Difficult. But easier to generate than

				SSB-SC
Detection methods	Simple & Inexpensive	Difficult	More difficult	Difficult
Signal to noise	$\left[\frac{S}{N}\right]_O = \frac{1}{3}\left[\frac{S}{N}\right]_I$	$\left[\frac{S}{N}\right]_O = \left[\frac{S}{N}\right]_I$	$\left[\frac{S}{N}\right]_O = \left[\frac{S}{N}\right]_I$	$\left[\frac{S}{N}\right]_O = \left[\frac{S}{N}\right]_I$
Sidebands	Two sidebands	Two sidebands	One sideband	One of the sideband is partially suppressed and a vestige of the other sideband is transmitted to compensate for that suppression
Application	AM broadcast application	Short distance point to point communication	Long range high frequency communication.	Televison Transmission

Superheterodyne Receiver

14.(a)Explain in detail about superheterodyne receiver.

Or

(b)With neat block diagram,explain superheterodyne receiver and its performance parameters.

In a broadcasting system whether it is based on amplitude modulation or frequency modulation, the receiver not only have the task of demodulating the modulated signal, but it is also required to perform some other system functions.

Carrier frequency tuning, the purpose of which is to select the desired signal (i.e.) desired radio or TV station)

Filtering, which ir required to separate the desire signal from other modulation signals that may be picked up along the way.
Amplification, which is intended to compensate for the loss of signal power incurred in the course of transmission.
The superheterodyne receiver is a special type of receiver that fulfils all the three functions. Fig. 1.33 shows the block diagram of a superheterodyne receiver.

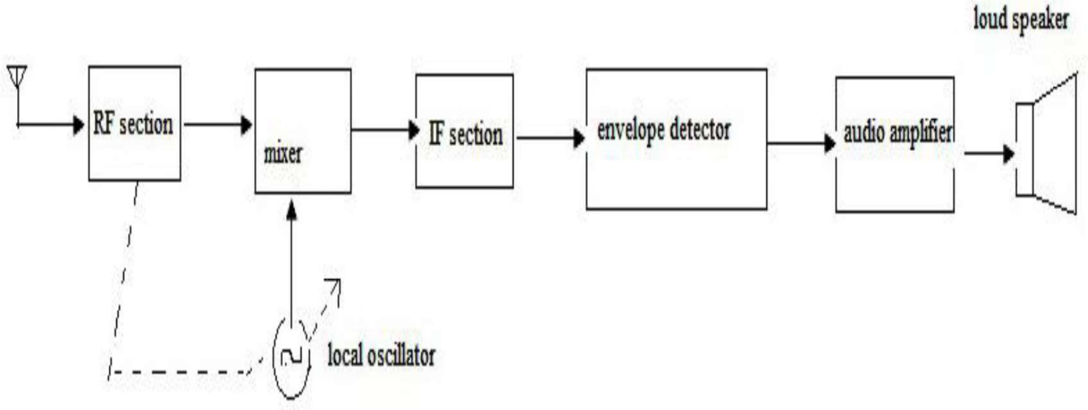


Fig.1.33 Block diagram of a superheterodyne receiver

R.F Section

The incoming amplitude modulated wave is picked up by the receiving antenna and is fed to the RF section. The RF section consists of a preselector and an RF Amplifier.
The preselector is a bandpass filter with an adjustable centre frequency that is tuned to the desired carrier frequency of the incoming signal. The main use of the preselected is to provide sufficient band limiting to prevent undesired ratio in frequency signal or image signal. The effectiveness of suppressing unwanted image signals increases as the number of selective stages in the RF section increases and as the ratio of intermediate to signal frequency increases. R.F amplifiers are used for better selectivity.

Frequency Changer

The combination of mixer and local oscillator provides a heterodyning function whereby the incoming signal is converted to a predetermined fixed intermediate frequency, usually lower than the incoming carrier frequency. This frequency translation is achieved without disturbing the relation of the sidebands to the carrier. The result of heterodyning is to produce an intermediate frequency carrier defined by $f_{IF} = f_{lo} - f_{RF}$
Where f_{LO} is the frequency of the local oscillator and f_{RF} is the carrier frequency of the incoming RF signal. Since the output of the frequency his neither the original input frequency not the final baseband frequency, it is called as intermediate frequency. Sometimes the frequency changer circuits are referred to as the first detector, in which case the demodulator is called as second detector.

IF Section

The IF section consists of one or more stages of turned amplification with a bandwidth corresponding to that required for the particular type of modulation that the receiver intended to handle. The IF section provides most of the amplification

Detector or Demodulator
purpose of which is to recover the baseband or message signal.
If coherent detection is used, then a coherent signal source must be provided in the receiver.
Audio Amplifiers
The final stage of the superheterodyne receiver consists of one or more audio amplifiers which is used for the power amplification of the recovered message signal.

Performance Parameters of Receivers

The performance of a Radio receiver is measured on the basic of its selectivity, sensitivity, fidelity and image frequency rejection selectivity.

Selectivity

The selectivity is the ability of the receiver to select a signal of a desired frequency while rejecting all others. The selectivity of the receiver is obtained partially by RF amplifier and mainly by IF amplifiers. The selectivity shows the attenuation that the receiver offers to signals at frequencies near to the one to which it is tuned. Fig.1.35 shows the typical selectivity curve of the receiver. The selectivity depends upon tuned LC circuits used in RF and IF stages, f_r is the resonating (tuned) frequency and Q is quality factor of these LC Circuits, As shown in Fig.1.35 bandwidth should be narrow for better selectivity. Hence Q of the coil should be high.

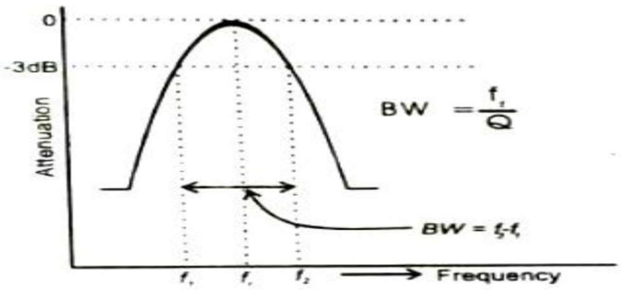


Fig.1.35 selectivity curve

Sensitivity

The ability of the receiver to pick up weak signals and simplify them is called sensitivity. It is often defined in terms of the voltage that must be applied to the receiver input terminals to give the standard output power, measured at the output terminals.

As the gain of the receiver is increased, sensitivity is also increased. The sensitivity is expressed in microvolts or decibels. Fig.1.36 shows the typical sensitivity curve of a receiver. As shown in the Fig.1.36, the sensitivity is decreased (i.e., voltage is increased) at high frequencies.

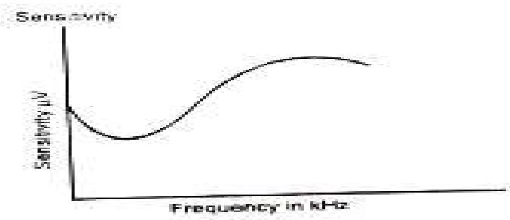


Fig1.36 sensitivity curve

Fidelity

Fidelity is a measure of the ability of a communication system to produce at the output of the receiver, an exact replica of the original source information. This may also be defined as the degree to which the system accurately reproduces at the output, the essential characteristics of signals that are impressed upon the input.

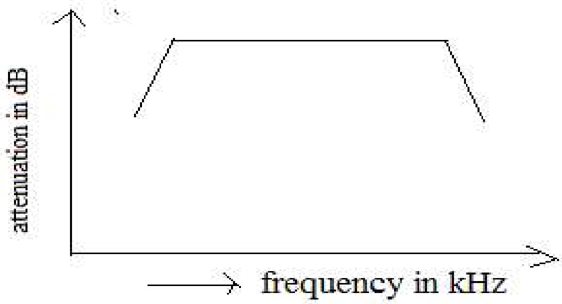


Fig.1.37 Fidelity curve

Fig.1.37 shows the typical fidelity curve of the receiver. The fidelity curve shown in the Fig.1.37basically represents frequency response of the receiver. A good fidelity requires wide band of frequencies to be amplified. Hence for good fidelity, more bandwidth of RF and IF stages are required.

Signal to noise Ratio

Signal to noise Ratio may be defined as the ratio of signal power to noise power at the receiver output. A good receiver should have high signal to noise ratio (SNR) which indicates negligible noise present at the output.

Image Frequency Rejection

We know that local oscillator frequency is made higher than the signal frequency such that $f_0 - f_s = f_i$. Here f_i is IF. That is $f_0 = f_s + f_i$. The IF stage passes only f_i. If the frequency $f_i = f_s + 2f_i$ appears at the input of the mixer, then the mixer will produce different frequency equal to f_i.

This is equal to IF. The frequency f_{si} is called image frequency and is defined as the signal frequency plus twice the IF. The image frequency is converted in the IF stage and it is also amplified by IF amplifiers. This is the effect of two stations being received simultaneously. The image frequency rejection is done by tuned circuit in the RF stage. It depends upon the selectivity of the RF stage. The image rejection should be done before the RF stage

Angle Modulation

15.(a)Explain in detail about angle modulation.

Or

(b)Briefly explain the relationship between phase and frequency modulation.

We know that amplitude, frequency or phase of the carrier can be varied by the modulating signal. Amplitude is varied in AM. When frequency or phase of the carrier is varied by the modulating signal, then it is called angle modulation. There are two types of angle modulation.

1. Frequency Modulation: When frequency of the carrier varies as per amplitude variations of modulating signal, then it is called Frequency Modulation (FM). Amplitude of the modulated carrier remains constant.
2. Phase Modulation: When phase of the carrier varies as per amplitude variations of modulating signal, then it is called Phase Modulation (PM). Amplitude of the modulated carrier remains constant.

The angle modulated wave is mathematically expressed as,

$e(t) = E_c \sin[\omega_c t + \theta(t)]$ or $s(t) = A_c \cos(\theta_i(t))$ (2.1)

Here e(t) is angle modulated wave

E_c is peak amplitude of the carrier

ω_c carrier frequency

$\theta(t)$ instantaneous phase deviation.

The phase deviation takes place in FM as well as PM. Hence phase is direct function of modulating signal. i.e.,

$\theta(t) \infty e_m(t)$

Here $e_m(t)$ is the modulating signal.

FM equation:

$e(t) = = E_c \sin\left[\omega_c t + \frac{k_1 E_m}{\omega_m} \sin \omega_m t\right]$ (2.2)

This is an equation for frequency modulated wave. Now let us derive an equation for phase modulated wave. Putting for θ(t) from equation (2.1.5) in equation (2.1.9) we get,

$e(t) = E_c \sin[\omega_c t + k\, e_m(t)]$

PM equation: $e(t) = E_c \sin[\omega_c t + k\, E_m \cos \omega_m t]$ (2.3)

This is an equation for phase modulated wave.

FM and PM Waveforms:

Fig.2.1 shows the waveforms of FM and PM.

In this figure following observations can be noted:

i. For FM signal, the maximum frequency deviation takes place when modulating signal is at positive and negative peaks.
ii. For PM signal the maximum frequency deviation takes place near zero crossing of the modulating signal.
iii. Both FM and PM waveforms are identical expect the phase shift.
iv. Form modulated waveform it is difficult to know, whether the modulation is FM or PM.

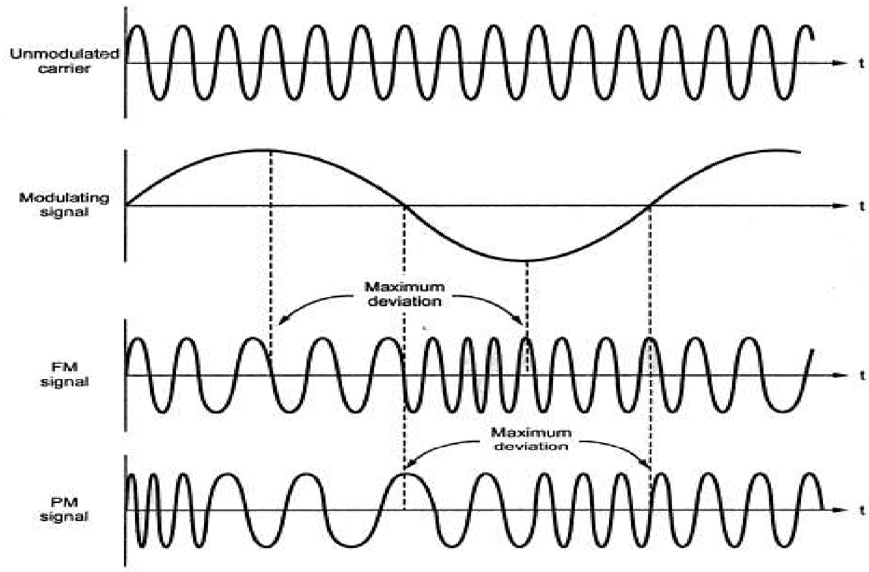


Fig.2.1 waveforms of FM and PM.

Phase deviation, modulation index and Frequency deviation.

The FM signal, in general is expressed as,

$e_{FM}(t) = E_c \sin[\omega_{ct} + m \sin\omega_m t]$ (2.4)

And the PM signal, in general is expressed as,

$e_{PM}(t) = E_c \sin[\omega_{ct} + m \cos\omega_m (t)]$ (2.5)

In both the above equations, the term 'm' is called modulation index. Note that the term m $\sin\omega_m t$ in equation (2.4) and m $\cos\omega_m (t)$ in equation (2.5) indicates instantaneous phase deviation θ(t). Hence 'm' also indicates maximum phase deviation. In other words, modulation index can also be defined as maximum phase deviation.

Modulation index for PM:

Comparing equation (2.5) and equation (2.3), we find that

Modulation index in PM : $m = k E_m$ rad (2.6)

Thus modulation index of PM signal is directly proportional to peak modulating voltage. And it's unit is radians.

Modulation index for FM:

Comparing equation (2.4) and equation (2.2) we find that, m = $\frac{K_1 E_m}{\omega_m}$ (2.7)

Thus modulation index of FM is directly proportional to peak modulating voltage, but inversely proportional to modulating signal frequency.

Since ω_m $=2\pi f_m$ above equation becomes,

Here $\frac{K_1 E_m}{2\pi f_m}$ is called frequency deviation. It is denoted by and its unit is Hz, i.e.,

Modulation index in FM :

$$m = \frac{\delta}{f_m} = \frac{\text{Maximum frequency deviation}}{\text{Modulating frequency}} \quad (2.8)$$

Thus modulation index of FM is unit less ratio. From above equation (2.1.14), note that the modulation index is differently defined for FM and PM signals.

Percentage modulation :

For angle modulation, the percentage modulation is given as the ratio of actual frequency deviation to maximum allowable frequency deviation. i. e.,

$$\% \text{ Modulation in} = \frac{\text{Actual frequency deviation}}{\text{Maximum allowable frequency deviation}} \quad (2.9)$$

Deviation Ratio (DR) :

The deviation ratio is the ratio of maximum frequency deviation to maximum modulating signal frequency. i.e.,

$$\text{Deviation ratio (DR)} = \frac{\text{Maximum frequency deviation}}{f_{m(max)}} \quad (2.10)$$

Thus the deviation ratio is basically the modulation index corresponding to maximum

FM and PM Modulators

We know that in FM, the frequency of the carrier is varied according to amplitude changes in the modulating signal. The carrier frequency is generated by LC oscillators. The carrier frequency can be changed by varying either the inductance or capacitance of the tank circuit. The devices like FET, BJT and varactor diodes have the property that their reactance can be varied by varying the voltage across them. Such devices can be used with LC tank circuits to vary the overall reactance. This reactance can be inductive or capacitive. The change in reactance changes the frequency of the oscillator.

There are two types of FM modulators:

i. Direct FM:

In this type of angle modulation, the frequency of the carrier is varied directly by the modulating signal. This means, an instantaneous frequency deviation is directly proportional to amplitude of the modulating signal.

ii. Indirect PM:

In this type of angle modulation, FM is obtained by phase modulation of the carrier. Instantaneous phase of the carrier is directly proportional to amplitude of the modulating signal.

Direct FM

Direct FM can be obtained by using FET and varactor diode.

16.(a)Explain in detail about FM modulators

Or

(b)Explain in detail about FM modulators using Direct and indirect method.

Frequency Modulation using varactor Diode or reactance modulation method or Direct method or frequency multiplier method.

All the diodes exhibit small junction capacitance in the reverse biased condition. The varactor diodes are specially designed to optimize this characteristic. The junction capacitance of the varactor diode changes as the reverse bias across it is varied. The variations in capacitance of this diode are wide and linear. The varactor diodes provide the junction capacitance in the range of 1 to 200 pF. Fig 2.9 shows how varactor diode can be used to generate FM. L_1 and C_1 form the tank circuit of the carrier oscillator. The capacitance of the varactor diode depends upon the fixed bias set by R_1 and R2 and the AF modulating signal. Either R1 or Rs is made variable so that the center carrier frequency can be adjusted over a narrow range. The Radio Frequency Choke (RFC) has high reactance at the carrier frequency to prevent the carrier signal from getting into the modulating signal circuits. At positive going modulating signal adds to the reverse bias applied to the varactor diode D, which decreases its capacitance and increases the carrier frequency. A negative going modulating signal subtracts from bias, increasing the capacitance, which decreases the carrier frequency.

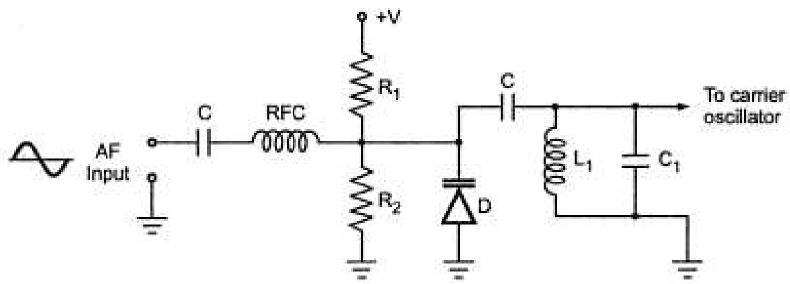


Fig 2.9 Varactor diode for FM generation

The frequency of the LC oscillator changes due to temperature effects. Hence crystals are used in FM generators to provide frequency stability.

Indirect method (Armstrong method)

In this method of FM generation, the modulating signal is integrated and it is phase modulated by crystal oscillator to get narrow band FM signal which later passed on to a frequency multiplier to get a wideband FM signal. The block diagram of an Armstrong method is shown in Fig

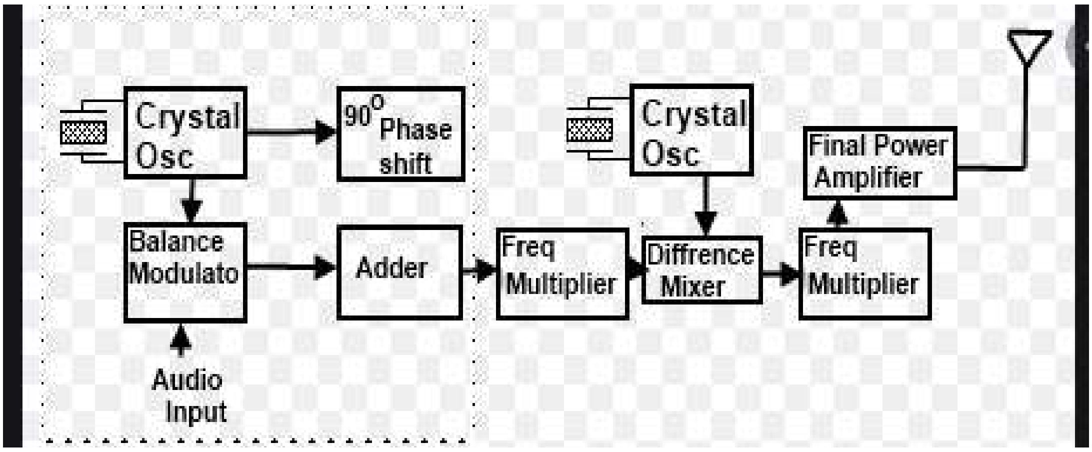


Fig. 2.11 Block diagram of Armstrong method for generating wideband FM signal

The crystal oscillator provides frequency stability. The value β is kept small to produce narrow band FM signal. The frequency multiplier consists of a memory less nonlinear device followed by a bandpass filter.

The balanced modulator is an amplitude modulator used to form an envelope of double sidebands and to suppress the carrier signal.This requires two input signals,the carrier signal and modulating message signal.The output of the modulator is connected to the adder circuit,here the 90^0 phase delayed carriers signal will be added back to replace the suppressed carrier and thus FM signal is generated

Advantages:

i) FM is generated from PM indirectly.
ii) Modulation takes place at low carrier frequency.

FM Demodulators

17.(a)Explain the various FM Demodulators.

Or

(b)Briefly explain the various types of FM demodulators.

The FM receivers also super heterodyne receivers. But they have different types of demodulators or detectors. FM receivers have amplitude limiters which are absent in AM receivers. The AGC system of FM receivers is different than that of AM receivers. RF amplifiers, mixers, local oscillators IF amplifiers, audio amplifiers etc. all are present in FM receivers . the detection of FM is totally different compared to AM. The FM detector should be able to produce the signal whose amplitude is proportional to the deviation in the frequency of signal. Thus the job FM detector is almost similar to frequency to voltage convertor. Here we will discuss these types of FM detectors. Slope detectors, phase discriminator and ratio detector.

Round – Travis detector or balanced slope detector (frequency Discriminator)

Fig. 2.13 shows the circuit of balanced slope detector. It consist of two identical circuit connected back to back . The FM signal is applied to the turned LC circuit. Two turned LC circuits are connected in series . The inductance of this secondary turned LC circuit is coupled with inductance of the primary (or input side) LC circuit . Thus it forms a turned transformer. In fig. 2.13, the upper turned circuit is shown as T1 and lower turned circuit is shown as T2. The input side LC circuit is turned to *fc,* carrier frequency. T1 tuned to $fc\text{-}\delta f$, which represents the minimum frequency of FM signal. The input FM signal is coupled to *T1* and *T2* 180° out of phase. The secondary side tuned circuits (*T1* and *T2)* are connected to diodes *D1* and *D2* with RC loads. The total output V_{out} is equal to difference between V_{01} andV_{02}, since they subtract (See Fig. 2.13) Fig. 2.14 shows the characteristic of the balanced slope detector. It shows V_{out} with respect to input frequency.

When the input frequency is equal to f_c both T1 and T2 produce the same voltage. Hence V_{01} and V_{02} are identical and they substract each other. Therefor V_{out} is zero. This is shown in Fig. 2.14. when the input frequency is $f_c + \delta f$,the upper circuit t1 produces maximum voltage since it is tuned to this frequency (i.e. f_c). Whereas lower circuit T2 is tuned to $f_c - \delta f$, which is quite away from $f_c + \delta f$. Hence T2 produces minimum voltage. Hence the output V_{01} is maximum where V_{02} is minimum. Therefore $V_{out} = V_{01} - V_{02}$ is maximum positive for f_c+δf.

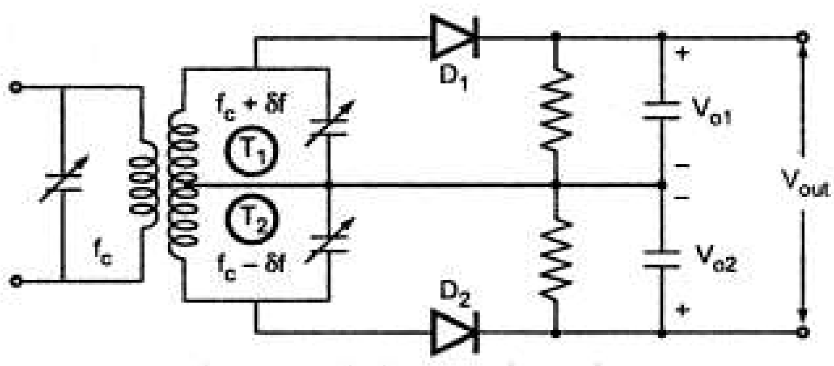


Fig.2.13 Balanced Slope detector

When input frequency is $f_c\text{-}\delta f$, the lower circuit T2 produces maximum signal. Hence rectified output V_{o2} is maximum and V_{o1} is minimum. Therefore output $V_{out} = V_{o1} - V_{o2}$ is maximum negative for $f_c - \delta f$. This is shown in Fig. 2.14.

For the other frequencies of input, the output (V_{out}) is produced according to the characteristic shown in Fig.2.14. For example if input frequency tries to increase above f_c then V_{01} will be greater than V_{o2} and net output V_{out} will be positive. It is desirable that the characteristic shown in Fig.2.14 should be linear between $f_c - \delta f$ and $f_c + \delta f$, then only proper detection will take place. The linearity of the characteristic depends upon alignment of tuning circuit and coupling characteristics of the tuned coils.

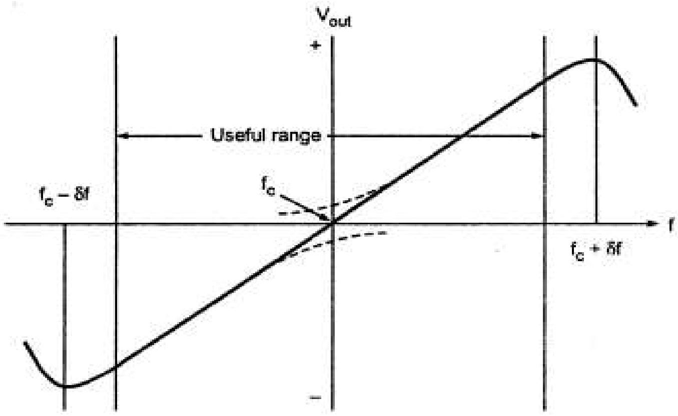


Fig.2.14 Characteristic of balanced slope detector, or 'S' curve

Foster-Seeley Discriminator (Phase Discriminator)

The phase shift between the primary and secondary voltages of the tuned transformer is a function of frequency. It can be shown that the secondary voltage lags primary voltage by 90^0 at the carrier center frequency. The carrier frequency (f_c) is the resonance (or tuned) frequency of the transformer. Foster-Seeley discriminator utilizes this principle for FM detection. Fig 2.16 (a) shows the circuit diagram of basic Foster-Seeley discriminator. In the figure observe that capacitor C_3 passes all the frequencies of FM. Thus the voltage V_1 is generated across RFC. RFC offers high impedance to frequencies of FM. The voltage V_1 thus appears across (RFC) center tap of secondary and ground also. The voltage of secondary is V_2 and equally divided across upper half and lower half of the secondary coil.

Fig 2.16 (b) shows the generator equivalent circuit of Foster-Seeley discriminator. In this figure observe that the voltage across diode D_1 is $V_{D1} = V_1 + 0.5V_2$ and that across D_2 is $V_{D2} = V_1 - 0.5V_2$.

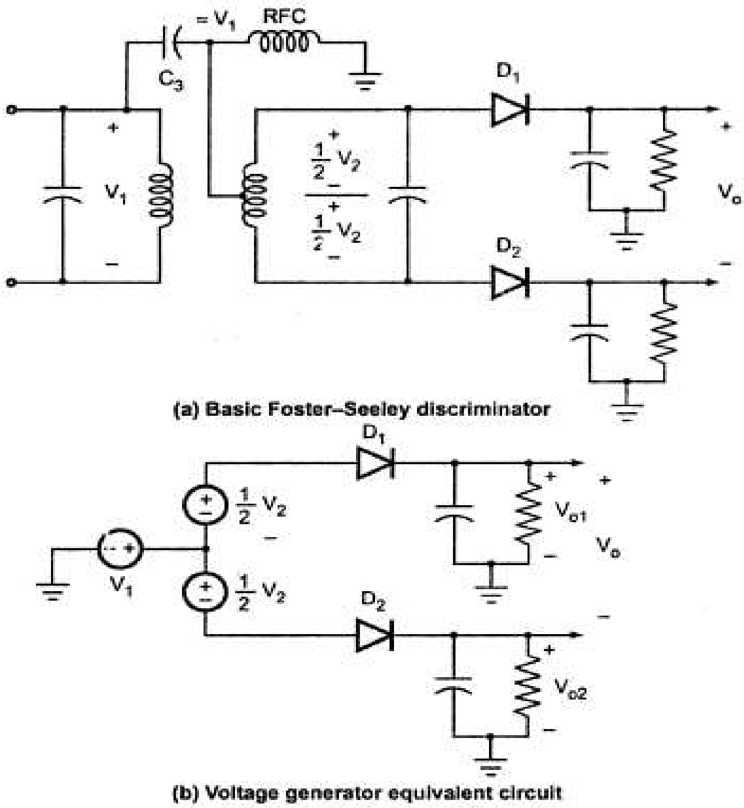


Fig.2.16 **Foster-Seeley Discriminator**

The output of upper rectifier is V_{01} and lower rectifier is V_{02}. The net output $V_0= V_{01}-V_{02}$. Since $V_{01}\approx |V_{D1}|$ and $V_{02} \approx |V_{D2}|$ output $V0 \approx |V_{01}| - |V_{D1}|$.Thus the net output depends upon the difference between magnitudes of $V_{D1}$and V_{D2} .

At the centre frequency both $V_{D1}$and V_{D2} will be equal, since V_2 will have 90^0 phase shift with V_1. Fig 2.17(a) shows how $V_{D1}$and V_{D2} are generated from V_1 and V_2. It shows that $|V_{D1}| = |V_{D2}|$. Hence the net output of the discriminator will be zero. Now consider the situation when input frequency increases above f_c. Hence the phase shift between V_1 and V_2 reduces. Therefore $|V_{D1}|$ is greater than $|V_{D2}|$. This is shown by vector addition in Fig.2.17 (b). Hence the net output $V_0= V_{01}-V_{02}$ will be positive. Thus the increase in frequency increases output voltage. Now consider the situation when frequency reduces below f_c. This makes $|V_{D1}|$ less than $|V_{D2}|$

This is shown is Fig.2.17. (c). Hence the output $V_0= V_{01}-V_{02}$will be negative. Thus the Foster-Seeley discriminator produces output depending upon the phase shift, The linearity of the output depends upon the linearity between frequency and induced phase shift. The characteristic of the Foster-Seeley discriminator (i.e. S-curve) is similar to that shown in Fig.2.14with more linearity in the operation.

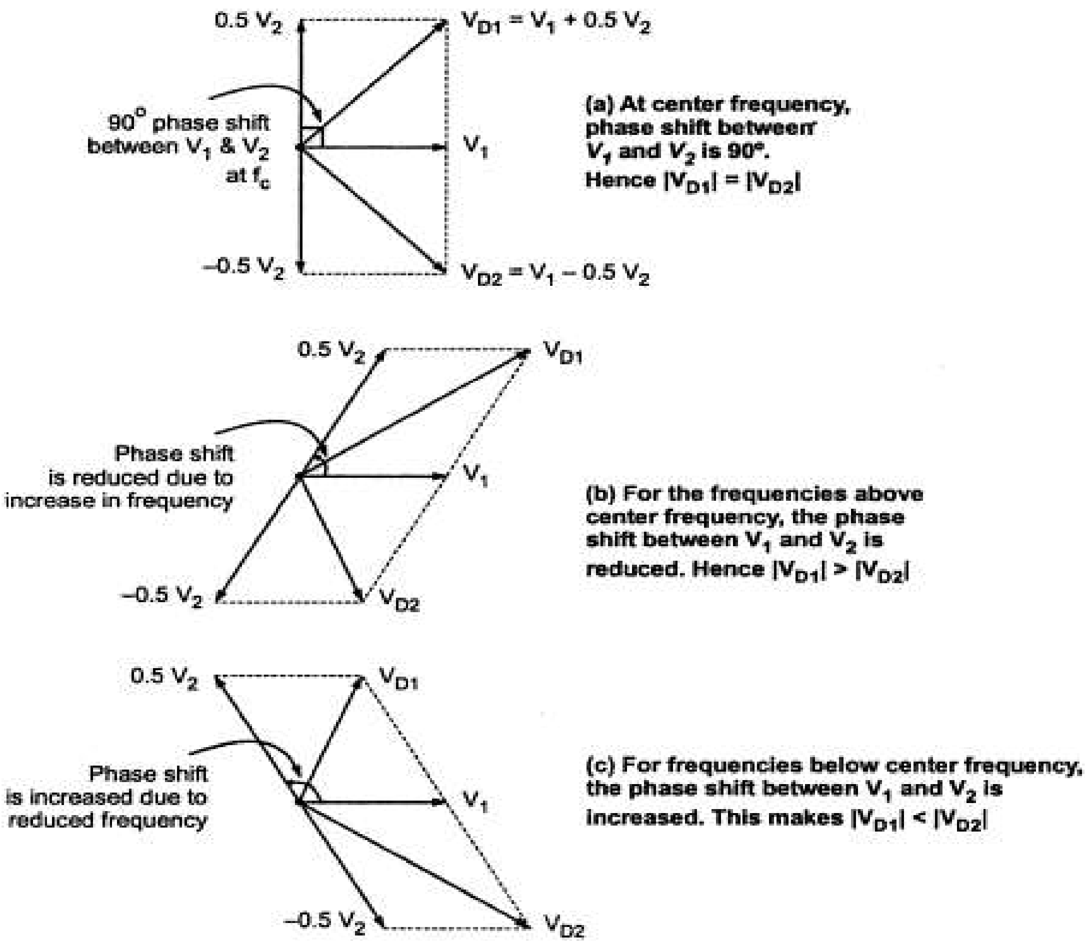


Fig.2.17 Phase shift between V_1 and V_2.

Ratio Detector:

Ratio detector can be obtained by slight modifications in the Fooster-Seeley discriminator. Fig.2.118 shows the circuit diagram of ratio detector .As shown in diagram the diode D_2 is reversed, and output is taken from different points.

The polarity of voltage in the lower capacitor is reversed, since connections of diode D_2 are reversed. Hence the voltages V_{01} and V_{02} across two capacitors add. We know that when V_{01} increases, V_{02} decreases and vice-versa as we have seen in Foster-Seeley circuit. Since V'_0 is sum of V_{01} and V_{02}, it remains constant.

From the circuit of Fig.2.17 we can write two equations for the output voltage V_0. The first equation will be

$$V_0 = \frac{1}{2} V'_0 - V_{02}$$

and $$V_0 = -\frac{1}{2} V'_0 + V_{01}$$

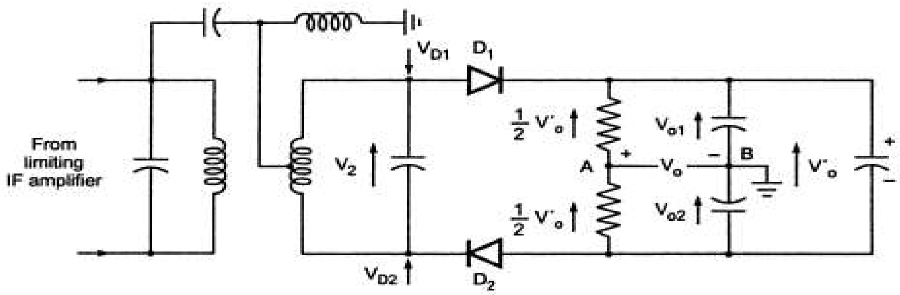


Fig. 2.18 Ratio detector circuit

Adding the above two equations,

$2V_0 = V_{01} - V_{02}$

Therefore $$V_0 = \frac{1}{2}(V_{01} - V_{02})$$

Since $V_{01} \approx |V_{D1}|$ and $V_{02} \approx |V_{D2}|$, above equation will be,

$$V_0 = \frac{1}{2}(|V_{D1}| - |V_{D2}|)$$

Here V_{D1} and V_{D2} are obtained as discussed earlier in Foster – Seeley circuit. The above equation shows that the output of ratio detector is half compared to that of Foster – Seeley circuit. We have seen earlier that as frequency increases above f_c, $|V_{D1}| > |V_{D2}|$, hence output V_0 is positive. Similarly if frequency decreases below f_c, $|V_{D1}| < |V_{D2}|$, hence output V_0 is negative.

Advantages:

1) As compared to Foster – Seeley circuit, this circuit does not respond to amplitude variations.

2) The output is bipolar (i.e., positive as well as negative).

Disadvantages:

1) Ratio detector does not tolerate variation in signal strength over performed period.

2) It requires an ACC signal

Advantage, Disadvantage and Application of FM

Advantages of FM:

i. Improved noise immunity

ii. Low power is required to transmit the signal

iii. Covers a large area with the same amount of transmitted power.

iv. Transmitted power remains constant

v. All the transmitted power is useful

vi. Adjacent channel interference is avoided due to guard bands.

Disadvantages of FM:

i. Very large bandwidth is required

ii. FM transmission and reception equipments are complex

iii. Compare to AM the area covered by FM is less.

Application of FM:

i. Radio broadcasting

ii. Sound broadcasting in TV

iii. Satellite communication

iv. Police wireless

v. Point to point communication

vi. Ambulances

vii. Taxicabs

Comparison of FM demodulator:

18.(a)Compare the various FM demodulators

Or

(b)Write the comparison of various FM demodulators.

S.No.	Parameter	Balanced slope FM detector	Ratio FM detector	Phase FM discriminator
1	Linearity of output characteristics	Poor	Good	Very good
2	Output characteristics depends on	Primary and Secondary frequency relation	Primary and Secondary phase relation	Primary and Secondary phase relation
3	Amplitude limiting	Not provided inherently	Provided inherently	Not provided inherently
4	Timing procedure	Circuit as three tuned	Not critical	Not critical

		circuit at frequencies		
5	Applications	Not used in practice	Narrowband FM receiver , TV receiver tuned section	Commercial FM radio receiver , Satellite receiver.

Comparison between AM and FM

19.(a)Compare AM and FM

Or

(b)Write the comparison of AM and FM

S.No.	Amplitude modulation	Frequency modulation
1	The modulation index is directly proportional to modulating voltage AM and inversely proportional to frequency	The modulation index is proportional to amplitude as well as phase
2	There are three components in AM. They are Carrier USB, LSB.	There are many frequency components in FM signal.
3	Power depends on the sideband	Total power remains constant
4	The bandwidth required is less compared to FM signal and is equal to $2f_m$. B.W = $2f_m$	Theoretically bandwidth of FM signal is infinite.
5	AM has poor fidelity due to narrow bandwidth	Since the bandwidth is large, fidelity is better
6	Adjacent channel interference is present	Adjacent channel interference is absent
7	Noise interference is more	Noise interference is less