UNIT-I
AMPLITUDE MODULATION

1.1Communication 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)

1.1.1Basic 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.

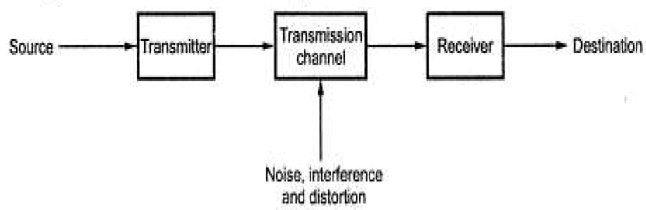


Fig. 1.1 Basic communication system

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. We will come back to the discussion of modulation later on. Now we will first see what is demodulation.

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.

1.1.2Communication channel

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

a) **Additive noise interference**:

This noise is generated due to internal solid state devices and resistors etc. Used to implement the communication system.

b) **Signal attenuation**; It occurs due to internal resistance of the channel and fading of the signal.

c) **Amplitude and phase distortion**: The signal is distorted in amplitude and phase because of nonlinear characteristics of the channel.

d) **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

i) **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 .

ii) **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.

1.1.3Analogue versus Digital Communication

Presently most of the communication is digital. For example cellular communication, satellite communication radar and sonar signal, facsimile ,data transmission over internet etc. all use digital communication. Practically, after 20 years analogue communication will be totally replaced by digital communication.

Why digital communication is so popular?

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.

1.1.4Advantages 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.

1.1.5Need of modulation or advantages of modulation:

The advantages of modulation are:

i) Easy of radiation

ii) Adjustment of bandwidth
iii) Reduction of height of antenna
iv) Avoid mixing of signals
v) Increase the range of communication
vi) Multiplexing, and
vii) Improves quality of reception.

1.2Principles of amplitude modulation

Introduction:

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.

Amplitude modulation is a relatively inexpensive, low quality form of modulation that is used for commercial broadcasting of both audio and video signals.

1.2.1Mathematical representation of an AM wave:

Let the modulating signal

$v_m(t) = v_m sin\omega_m t$ (1.1)

Carrier signal $v_c(t) = v_c sin\omega_c t$ (1.2)

Where,

v_{c-} Amplitude of the carrier signal (volts)

v_{m-} Amplitude of the modulating signal (volts)

The amplitude of the carrier signal is changed after modulation.

$v_{AM} = v_c + v_m(t)$ (1.3)

Substitute equation (1.1) in equation (1.3)

$= v_c + v_m\ sin\omega_c t$

$v_{AM} = v_c[1 + m_a\ sin\omega_m t]$ (1.4)

Hence AM wave is given by

$v_{AM}\ (t) = v_{AM}\ sin\omega_c t$ (1.5)

Substitute equation (1.4) in equation(1.5)

$v_{AM} = v_c[1 + m_a\ sin\omega_m t]sin\omega_c t$ (1.6a)

$v_{AM} = v_c[1 + m_a\ sin(2\pi f_m t)]sin(2\pi f_c t)$ (1.6b)

Where, m_a = Modulation index

This expression represents the time domain representation of an AM signal.

AM Envelope:

- AM is simply called as double side band full carrier (DSB-FC) is probably the most commonly used.
- AM DSB-FC is sometimes called conventional AM or simply AM.

- The shape of the modulated wave (AM) is called AM envelope which contains all the frequencies and is used to transfer the information through the system.
- An increase in the modulating signal amplitude causes the amplitude of the carrier to increase.
- Figure 1.2, shows a single – frequency sine wave modulating a higher frequency carrier signal.
- Without modulating signal, the AM output waveform is simply the carrier signal.
- The repetition rate of the envelope is equal to the frequency of the modulating signal.
- The shape of the envelope is identical to the shape of the modulating signal.

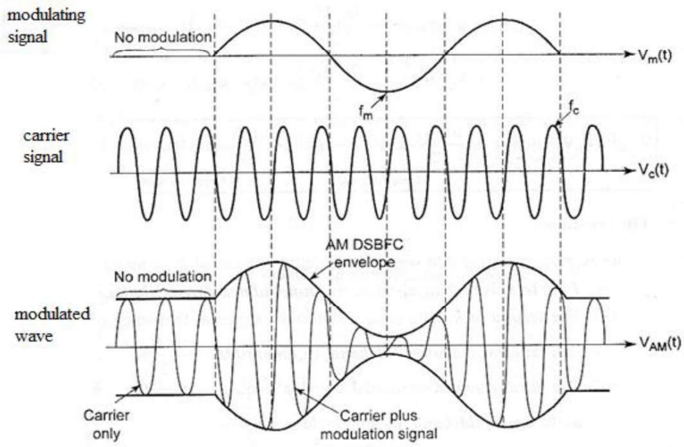


Fig.1.2 AM generation

AM Frequency Spectrum and Bandwidth:

An AM modulator is a nonlinear device. Therefore, nonlinear mixing occurs, and the output 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.

- **Frequency spectrum and bandwidth:**

The AM wave is given by:

$$v_{AM}(t) = v_c[1 + m_a \sin\omega_m t]\sin\omega_c t$$

$$= v_c \sin\omega_c t + m_a v_c \sin\omega_m t \sin\omega_c t$$

We know that,

$\mathrm{Sin}\omega_m t \sin\omega_c t = \frac{\cos(\omega_c-\omega_m)t-\cos(\omega_c+\omega_m)t}{2}$

$V_{AM}(t) = v_c \sin\omega_c t + \frac{m_a v_c}{2}[\cos(\omega_c - \omega_m)t - \cos(\omega_c + \omega_m)t]$

$= v_c \sin\omega_c t + \frac{m_a v_c}{2}[\cos(\omega_c - \omega_m)t - \cos(\omega_c + \omega_m)t]$

$V_{AM}(t) = v_c \sin\omega_c t + \frac{m_a v_c}{2}\cos(\omega_c - \omega_m)t - \frac{m_a v_c}{2}\cos(\omega_c + \omega_m)t$ (1.7)

Observation:

The observation for the AM wave shows that it consists of three terms:

i) **First term** is nothing else but the **unmodulated carrier signal.**

ii) The **second term** represents a sinusoidal signal at frequency ***(f_c - f_m).*** It is called as the lower sideband. Its amplitude is $\frac{m_a V_c}{2}$.

iii) The third term is sinusoidal signal at frequency (f_c+f_m). It is called as the upper sideband. Its amplitude is $\frac{m_a V_c}{2}$.

iv) The (-) sign associated with the USB represents a phase shift of 180°

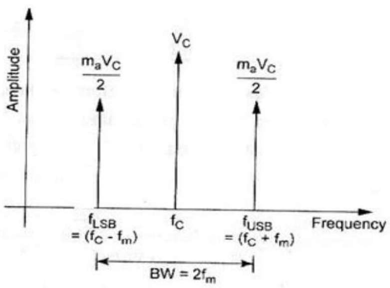


Fig.1.3 Frequency domain representation of AM wave

Sidebands:

Whenever a carrier is modulated by an information signal, new signals at different frequencies are generated as part of the process. These new frequencies are called side frequencies or sidebands.

The sidebands are occurs in the frequency spectrum directly above and below the carrier frequency.

Assuming a carrier frequency of f_c and a modulating frequency of f_m, the upper sideband f_{USB} and lower sideband f_{LSB} are computed as follows:

$$f_{USB} = f_c + f_m$$
$$f_{LSB} = f_c - f_m \qquad (1.8)$$

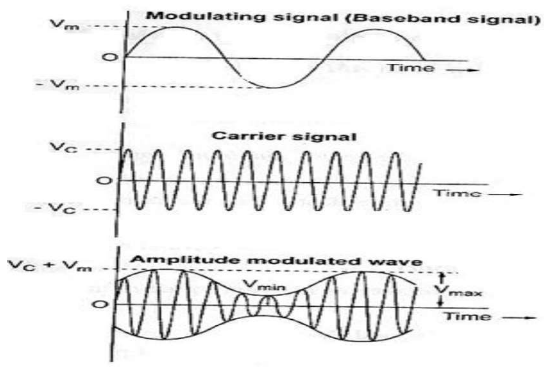


Fig.1.4 AM waveform for sinusoidal modulating signal

Bandwidth of AM:

The Bandwidth of AM signal is given by the subtraction of the higher and the lowest frequency component in the frequency spectrum.

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

$= (f_c + f_m) - (f_c - f_m)$

$B = 2 \times f_m$ (1.9)

Where,

B – Bandwidth in hertz

f_m – Highest modulation frequency in hertz.

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

Phasor representation of an AM with carrier:

The amplitude variation in an AM system can be explained with the help of a Phasor diagram.

The Phasor for the upper sideband rotates anticlockwise at an angular frequency of ω_m.

Similarly, the Phasor for the lower sideband rotates clockwise at the same angular frequency ω_m.

The upper side frequency rotates faster than the carrier ($\omega_m > \omega_c$), and the lowerside frequency rotates slower ($\omega_m < \omega_c$).

The resulting amplitude of the modulated wave at any instant is the vector sum of the two sideband Phasors.

Vc is carrier wave Phasor taken as reference Phasor and the resulting Phasor is V_{AM} (t).

The Phasors for the carrier and the upper and lower side frequencies combine, sometimes in phase and sometimes out of phase.

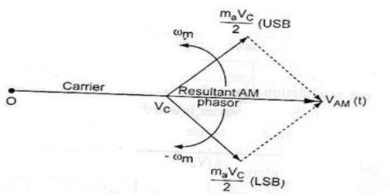


Fig.1.5 Phasor representation of AM with carrier

1.2.2Modulation Index and Percent Modulation

Coefficient of modulation or modulation index:

Modulation index is a term used to describe the amount of amplitude change present in an AM waveform.

The extent of amplitude variation in AM about an unmodulated carrier amplitude is measured in terms of a factor called modulation index.

In AM wave, the modulation index (m_a) is defined as the ***ratio of maximum amplitude of modulating signal to maximum amplitude of carrier signal.***

$m_a = \frac{v_m}{v_c}$ (1.10)

Value of Vm must be less than value of Vc (Vm < Vc) to avoid distortion in the modulated signal. Hence the maximum value of m_a is 1 (Vc = Vm).
The modulation index should be a number between 0 and 1

Percent modulation:

When modulator index is expressed in percentage, it is called percent modulation.
Percent modulation gives the percentage change in the amplitude of the output wave when the carrier is acted on by a modulating signal.

For example: $m_a = 0.5$ corresponds to 50% modulation.

$$\%modulation = \frac{V_m}{V_c} \times 100 \qquad (1.11)$$

Calculation of modulation index from AM waveform:

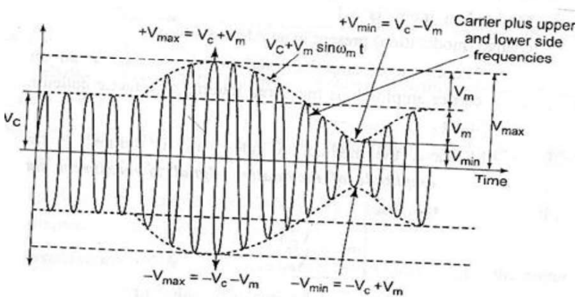


Fig. 1.6 AM wave for calculation of m_a

From the above figure,

$V_{max} = V_C + V_m$ (a)

$V_{min} = V_C - V_m$ (b)

From equations (a) – (b), we get

$2 V_m = V_{max} - V_{min}$

$$V_m = \frac{Vmax - Vmin}{2} \qquad (1.12)$$

From equation (a), $V_C = V_{max} - V_m$ (1.13)

Sub. Equation (12) in (13)

$$V_C = V_{max} - \left[\frac{Vmax - Vmin}{2}\right]$$

$$V_C = \frac{Vmax + Vmin}{2} \qquad (1.14)$$

Modulation index $m_a = \frac{V_m}{V_c}$ (1.15)

So substituting the values of V_C and V_m from the equations (1.12) and (1.14) in equation (1.15), we get

$$m_a = \frac{\frac{1}{2}(V_{max} - V_{min})}{\frac{1}{2}(V_{max} + V_{min})}$$

$$m_a = \frac{V_{max} - V_{min}}{V_{max} + V_{min}} \qquad (1.16)$$

Equation (1.16) is expressed as in percentage is called the percentage modulation.

$$\%\ m_a = \frac{V_{max} - V_{min}}{V_{max} + V_{min}} \times 100 \qquad (1.17)$$

The peak change in the amplitude of the output wave V_m is the sum of the voltages from the upper and lower side frequencies.

$V_m = V_{USB} - V_{LSB}$ (1.18)

And $V_{USB} = V_{LSB}$ then

$V_m = 2\ V_{USB}$

$$V_{USB} = \frac{Vm}{2} \qquad (1.19)$$

$$V_{USB} = V_{LSB} = \frac{Vm}{2} \qquad (1.20)$$

Sub equation (1.12) in (1.20)

$$= \frac{\frac{1}{2}\left[V_{max} - V_{min}\right]}{2}$$

$$V_{USB} = V_{LSB} = \frac{1}{4}(V_{max} + V_{min}) \qquad (1.21)$$

For 100% modulation $V_m = V_C$, the maximum amplitude of the envelope $V_{max} = 2V_c$ and the minimum amplitude of the envelope $V_{min} = 0$

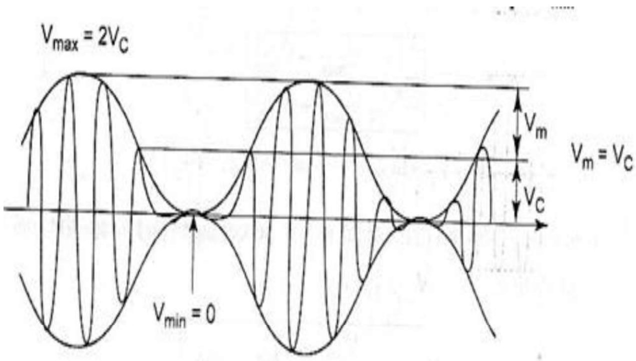


Fig.1.7 100% modulated wave

Modulation index for multiple modulating frequencies:

When two or more modulating signals are modulated by a single carrier. Then the modulation index is given by,

$m_a = \sqrt{m_1^2 + m_2^2 + \ldots}$ (1.21)

Where,

m_a = total resultant modulation index

m_1, m_2, ... = modulation indices due to individual modulating components.

1.2.3Degree of modulation

The modulating signals preserved in the envelope of amplitude modulated sig**n**al only if Vm < Vc, then ma <1.

Vm = Maximum amplitude of modulating signal.

Vc = Maximum amplitude of carrier signal.

In AM, three types of degree of modulation are available. It depends upon the amplitude of the modulating signal relative to carrier amplitude.

i) Under modulation.
ii) Critical modulation
iii) Over modulation

Under modulation:

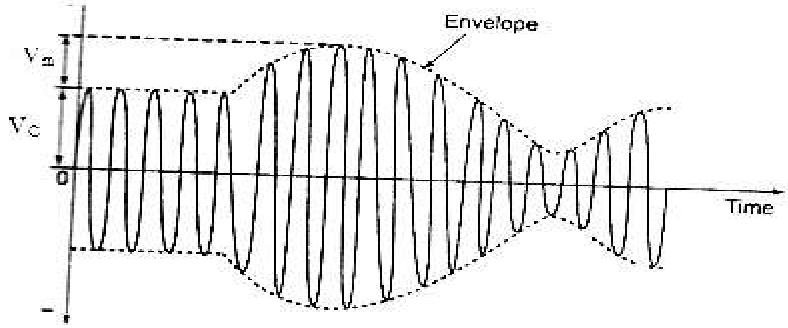


Fig 1.8 AM wave for percentage modulation less than 100%

Here the envelope of amplitude modulated signals does not reach the zero amplitude axis. Hence the messages signal in fully preserved in the envelope of the AM wave.
An envelope detector can recover the message signal without any distortion.

$\boxed{\text{AM wave with } m_a < 1 \text{ i.e., } V_m < V_C}$

Critical modulation:

$\boxed{m_a = 1 \text{ when } V_m = V_C}$

Here the envelope of the modulated signal just reaches the zero amplitude axis. The message signal remains preserved.
An envelope detector can recover the message signal without any distortion.
AM wave with m_a =i.e., 100% modulation $V_m = V_c$

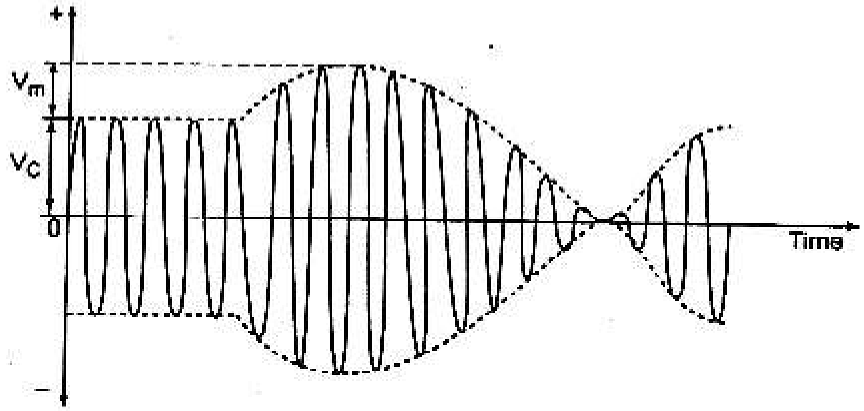


Fig 1.9 AM wave with 100% modulation

Over modulation:
$m_a > 1$ when $V_m > V_c$
Here both positive and negative extensions of the modulating signals are cancelled or clipped out.
The amplitude of a base band signal exceeds carrier amplitude
The envelope and message signal are not same. This is called as envelope distortion. Due to this envelope detector provides distorted message signals.
An over modulated signal can be recovered using a costly and as well as complex technique***; synchronous detection.***

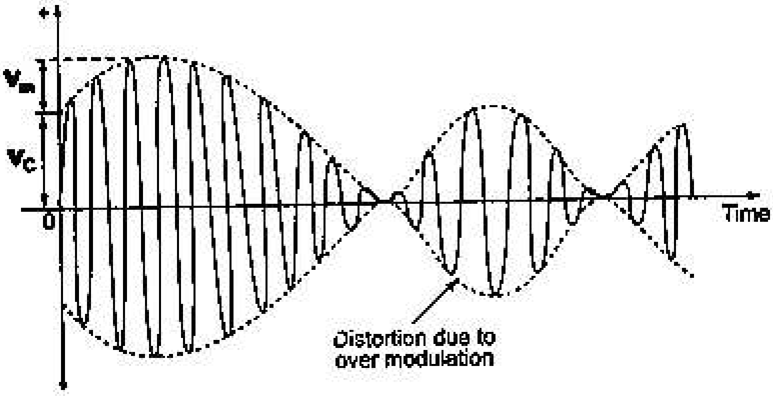


Fig 1.10 over modulation $m_a > 1$

1.2.4AM POWER DISTRIBUTION:

An AM wave consists of carrier and two sidebands.

The carrier component of the modulated wave has the same amplitude as the unmodulated carrier. The modulated wave contains extra power in the two sideband components.

The amplitude of the sidebands depends on the modulation index 'ma'. Therefore the total power in the modulated wave will depend on the modulation index also.

The total power in the modulated wave will be

Pt = [carrier power] + [power in LSB] + [power in USB]

$$P_t = P_c + P_{LSB} + P_{USB} \qquad (1.22)$$

$$= \frac{V_c^2}{R} + \frac{V_{LSB}^2}{R} + \frac{V_{USB}^2}{R} \text{ (rms)}$$

Where all three voltages are in rms values, and R is the resistance (ex. Antenna resistance), in which the power is dissipated.

Carrier Power (Pc) :

The average power dissipated in a load by an unmodulated carrier is equal to the rms carrier voltage squared divided by the load resistance.

$$P_C = \frac{V_{carr}^2}{R} \qquad (1.23)$$

$$P_C = \frac{(0.707 V_C)^2}{R}$$

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

$$\boxed{P_C = \frac{V_C^2}{2R}} \qquad (1.24)$$

Where,

Pc __ Carrrier power (watts)
Vc __ Peak carrier voltage (volts)
R ___ Load resistance (ohms)

- **Power in the sidebands:**

The upper and lower sideband powers are expressed mathematically as

$$P_{USB} = P_{LSB} = \frac{(0.707V_{SUB})^2}{R} \quad (1.25)$$

The peak voltage of the upper and lower side band frequencies

$$V_{SUB} = \frac{maVc}{2} \quad (1.26)$$

Substitute equation (1.26) in (1.25)

$$P_{USB} = P_{LSB} = \frac{\left(\frac{m_a V_C/2}{\sqrt{2}}\right)^2}{R}$$

$$\boxed{P_{USB} = P_{LSB} = \frac{m_a^2 V_C^2}{8R}} \quad (1.27)$$

Where,
P_{USB} __ Upper side band power (Watts),
P_{LSB} ___ Lower sideband power(Watts).
Equation (1.27), may be written as

$$P_{USB} = P_{LSB} = \frac{m_a^2}{4} \times \frac{V_C^2}{2R} \quad (1.28)$$

We know $P_C = \frac{V_C^2}{2R}$

$$\boxed{P_{USB} = P_{LSB} = \frac{m_a^2}{4} P_C} \quad (1.29)$$

- **Total power in AM Wave:**

Substituting equations (1.24) & (1.29) and into equation (1.22) , we get the total Power in AM wave,

$$P_t = P_C + P_{USB} + P_{LSB}$$

$$= \frac{V_C^2}{2R} + \frac{m_a^2}{4}\frac{V_C^2}{2R} + \frac{m_a^2}{4}\frac{V_C^2}{2R} \quad = P_C + \frac{m_a^2}{4}P_C + \frac{m_a^2}{4}$$

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

$$\boxed{P_t = P_C\left[1 + \frac{m_a^2}{2}\right]} \qquad (1.30)$$

$$\frac{p_t}{p_c} = 1 + \frac{m_a^2}{2}$$

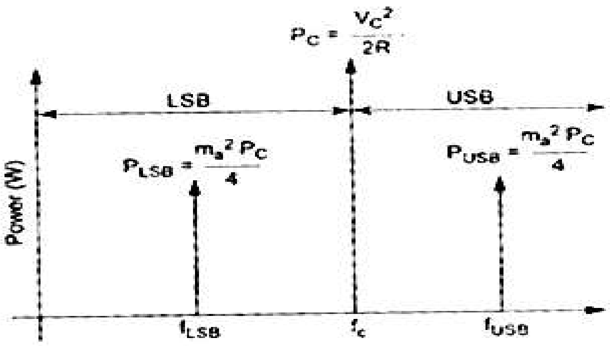


Fig 1.11 power spectrum for an AM wave

Equation (1.30) relates the total power in the amplitude modulated wave to the unmodulated carrier power with increases in the value of 'm_a', the total power also increases.

Power spectrum for an AM wave single frequency modulating signal:

f $m_a = 1$ for 100% modulation

$$\frac{Pt}{Pc} = 1.5$$

$$P_t = 1.5Pc \qquad (1.31)$$

$$\frac{Pt}{Pc} = 1 + \frac{m_a^2}{2} \qquad (1.32)$$

Modulation index in terms of P_t and P_c

Equation (1.32) becomes

$$\frac{m_a^2}{2} = \frac{Pt}{Pc} - 1$$

$$m_a = \sqrt{2\left(\frac{Pt}{Pc} - 1\right)} \quad (1.33)$$

Current equations:

From equation (1.30), we get

$$Pt = Pc\left[1+\frac{m_a^2}{2}\right]$$

We know that

$$P_t = I_t^2 \times R \text{ and } P_C = I_C^2 \times R$$

Where,

P_t – Total transmit power (watts)
P_c – Carrier power (watts)
I_t – Total transmit current (ampere)
I_C – Carrier current (ampere)
R – Antenna resistance (ohms)

$$\frac{P_t}{P_C} = \frac{I_t^2 R}{I_C^2 R} = \left(\frac{I_t}{I_C}\right)^2 = 1+\frac{m_a^2}{2}$$

$$\frac{I_t}{I_C} = \sqrt{1+\frac{m_a^2}{2}}$$

$$\boxed{I_t = I_C\sqrt{1+\frac{m_a^2}{2}}} \quad \textbf{(1.34)}$$

Modulation index in terms of current:

We know that

$$\frac{I_t^2}{I_C^2} = \left[1+\frac{m_a^2}{2}\right]$$

$$\frac{m_a^2}{2} = \left[\left(\frac{I_t}{I_C}\right)^2 - 1\right]$$

$$\boxed{m_a = \sqrt{2\left[\left(\frac{I_t}{I_C}\right)^2 - 1\right]}} \quad \textbf{(1.35)}$$

1.2.5Transmission Efficiency:

The amount of useful message power present in AM wave is expressed by a term called transmission efficiency

AM wave expression contains carrier, USB & LSB, but carrier does not contain any information. So we consider two sideband only.

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*"**

$$\%\,\eta = \frac{\text{Power in sideband}}{\text{Total power}} \times 100$$

$$= \frac{P_{LSB} + P_{USB}}{P_t} \times 100$$

$$= \frac{\dfrac{m_a^2 V_C^2}{8R} + \dfrac{m_a^2 V_C^2}{8R}}{\dfrac{V_C^2}{2R}\left[1 + \dfrac{m_a^2}{2}\right]} \times 100 \qquad \because\ P_C = \frac{V_C^2}{2R}$$

$$= \frac{\dfrac{m_a^2 V_C^2}{2 \times 4R} + \dfrac{m_a^2 V_C^2}{2 \times 4R}}{\dfrac{V_C^2}{2R}\left[1 + \dfrac{m_a^2}{2}\right]} \times 100$$

$$= \frac{\dfrac{V_C^2}{2R}\left[\dfrac{m_a^2}{4} + \dfrac{m_a^2}{4}\right]}{\dfrac{V_C^2}{2R}\left[1 + \dfrac{m_a^2}{2}\right]} \times 100$$

$$= \frac{\dfrac{m_a^2}{2} P_C}{P_C\left(1 + \dfrac{m_a^2}{2}\right)} \times 100$$

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

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

$$\% \eta = \frac{m_a^2}{2 + m_a^2} \times 100 \qquad (1.36)$$

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.

1.2.6Advantages, 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.

1.3Types of AM modulation:

Introduction

- The generating circuits for AM wave are called as amplitude modulator circuits.
- The power in an amplitude modulated signal increases with modulation. The extra power goes into the sidebands. At maximum modulation, the total power is 50% greater than the power in the unmodulated carrier.
- The methods for generation of AM waves are broadly divided into two parts. They are:

(a) Square law or non linear modulator circuits and
(b) Linear modulator circuits.

- Based on the power level at which modulation is carried out and may be termed as,

(a) Low level modulation and
(b) High level modulation.

In general square law modulators are low level modulators while linear modulations are high level modulators.
The location in a transmitter where modulation occurs determines whether the circuit is a low or a high level transmitter.

Low level modulation:

- The generation of AM wave takes place at a low power level.
- With low- level modulation, the modulation takes place prior to the output element of the final stage of the transmitter.
- The linear amplifiers are required in order to avoid any waveform distortion.

Advantages:
Here, less modulating signal power is required to achieve a high percentage of modulation.
Disadvantages:
For high power applications, when all the amplifiers that follow the modulator stage must be linear amplifiers, which is extremely inefficient.

High level modulation:

- In this method, the generation of AM wave takes place at high power levels.
- The carrier and the modulating signal both are amplified first to an adequate power level and the modulation takes place in the last RF amplifier stage of the transmitter.

Advantage:
Highly efficient class C amplifiers are used in high level modulation hence the efficiency of high level modulation is higher than that of low level modulation.

1.4AM MODULATOR CIRCUITS
1.4.1Low level AM modulator
Emitter modulator:
The carrier is applied to the base and the modulating signal to the emitter. This configuration is called emitter modulation.

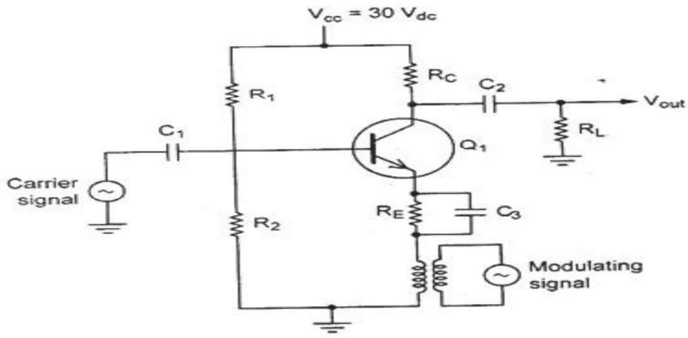


Fig.1.12 Single transistor emitter modulator

- Figure shows a circuit which is basically a small signal class A amplifier. It is called as an emitter modulator circuit and uses as amplitude modulator.
- The circuit has two inputs namely the RF carrier and the modulating signal.
- When the modulating signal is absent, only the carrier is applied, the circuit works only as a class A amplifier and we get amplified carrier at the output.
- When the modulating signal is applied, the amplifier operates as non- linear device and multiplication of the carrier and modulating signal will take place.
- The gain of the amplifier is dependent on the modulating signal.
- The modulating signal varies the gain of the amplifier at a sinusoidal rate equal to the frequency of the modulating signal. Depending upon the gain variations, carrier is amplified.
- The depth of modulation (m_a) is proportional to the amplitude of modulating signal.
- The ***voltage gain*** of emitter modulator is given as

$$A_V = A_Q \left[1 + m_a \sin\left(2\pi f_m t\right)\right] \qquad (1.37)$$

Where, A_v = Voltage gain with modulation
A_Q = Voltage gain without modulation

- This equation (1.37) shows that the gain A varies sinusoid ally with the modulating signal.
- $\sin(2\pi f_m t)$ goes from maximum value of +1 to a minimum value of -1. Thus equation (1.37) reduces to

$$A_V = A_Q [1 \pm m_a] \qquad (1.38)$$

- At 100% modulation, $m_a = 1$, and equation (1.38) reduces to

$$A_{v(max)} = 2A_Q$$
$$A_{V(min)} = 0 \qquad (1.39)$$

Operation

The modulating signal i applied through isolation transformer T_1 to the emitter of Q_1 and the Carrier is applied directly to the base.

The modulating signal drives the circuit into both saturation and cut off, thus producing the ***non-linear amplification*** necessary for modulation to occur.

The collector waveform includes the carrier and upper and lower side frequencies as well as a component at the modulating signal frequency.

The unwanted modulating signal from the AM waveform is removed by the coupling capacitor C_2, thus producing a symmetrical AM envelope at V_{out} (across R_L).

In emitter modulation, the amplitude of the output signal depends on the amplitude of the input carrier and the voltage gain of the amplifier.

Disadvantages:

i) Emitter modulation circuit is a amplifier operates in ***Class A*** mode, which is ***extremely inefficient.***

ii)Emitter modulators are Also incapable of producing high power output waveforms

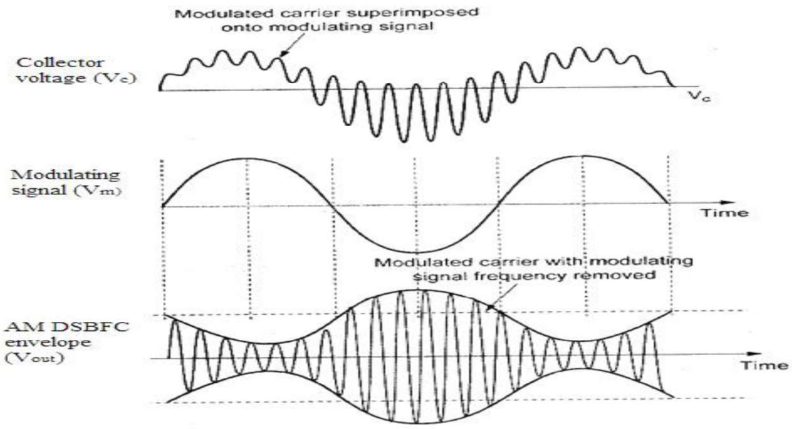


Fig.1.14 Output waveform

1.4.2High Level AM Modulator [Collector Modulator]

This circuit is called as **collector modulator** because the modulating signal is applied to the collector. This stage acts as an ***output*** stage of the ***transmitter.*** It is a high level modulator and operates as ***class C amplifier.***Transistors T1 forms RF amplfoer in **"Class C"** mode for higher efficiency. The carrier signal is applied at the base of T1 V_{cc} is the collector supply used for biasing. Transistor T2 forms a "***class B"*** amplifier used to amplify the ***base band*** (modulating) signal.

After amplification the modulating signal appears across the modulating transformer.

This modulating signal exists in series with the collector supply V_{cc}.The capacitor "C" offers very low path for the carrier signal and as such the carrier is prevented from flowing through modulating transformer $T_{rl.}$

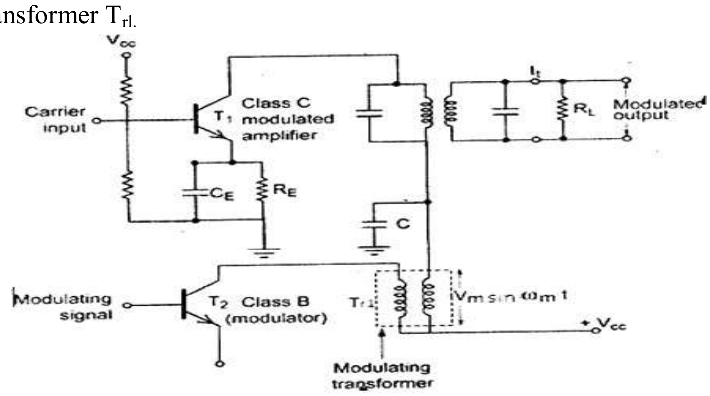


Fig.1.5 A collector modulation circuit

Principle of Operation:

The carrier signal is amplified by the class C modulated amplifier and its amplitude remains constant equal to V_{cc} as no voltage exists across the modulating transformer T_{rl} , in the absence of modulating voltage.

When modulating voltage $V_m \, Sin\omega_m t$ appears across the modulating transformer $T_{rl.}$, its voltage will added with the supply voltage $V_{cc.}$ The net effect is a ***slow variation*** in supply voltage V_{cc} .

This slow variation in supply voltage changes the amplitude of the carrier voltage at the output of the modulated class C amplifier.

The envelope of the output voltage is identical with the modulating voltage, and thus an AM Wave is generated.

The slowly varying supply voltage with respect to modulating voltage is given by

$$V_c = V_{cc} + V_m \sin \omega_m t \tag{1.40}$$

$$= V_{cc} \left[1 + \frac{V_m}{V_{cc}} \sin \omega_m t \right]$$

$$= V_{cc} \left[1 + m_a \sin \omega_m t \right] \tag{1.41}$$

The Modulated output voltage Vo (t) is given by

$$V_0(t) = V_{cc} \left[1 + m_a \sin \omega_m t \right] \sin \omega_c t \tag{1.42}$$

Power and efficiency calculation:

The modulated power delivered to the output load depends on the ***input supplied*** by the supply voltage and the ***power dissipation*** in the collector circuit.

Out of the total power input in the collector circuit, only a part of it reaches the output load, the remaining power is lost in the collector circuit.

This loss is due to the collector resistance, and other dissipating components.

Let, $P_m =$ Input power in the collector circuit.

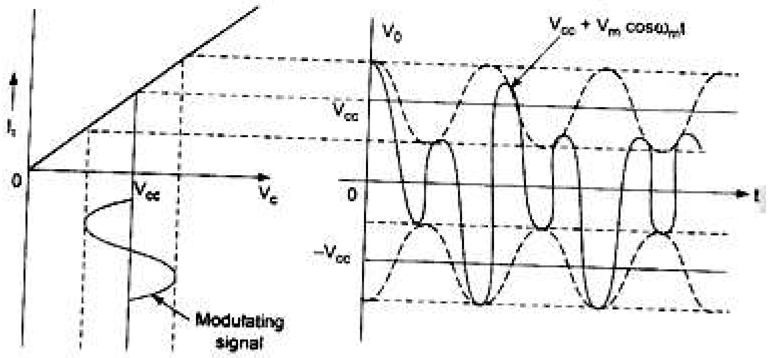
$P_{out} =$ Output power delivered to a load R_L and

P_d = Power dissipation in the collector circuits.

$$P_{in} = P_{out} + P_d \tag{1.43}$$

$$P_d = P_{in} - P_{out}$$

$$= P_{in}\left[1 - \frac{P_{out}}{P_{in}}\right]$$



Collector Efficiency:

The ratio of output power and input power in the collector circuit is defined as collector circuit efficiency denoted by η_c.

$$\eta_c = \frac{P_{out}}{P_{in}} \qquad (1.44)$$

Hence

$$P_d = P_{in}\,(1 - \eta_c) \qquad (1.45)$$

Here,

$$P_{in} = P_{cc}\left[1 + \frac{m_a^2}{2}\right] \qquad (1.46)$$

Where,

$$P_{cc} = V_{cc} \times I_c$$

P_{cc} is the power supplied by V_{cc}.

The total input power is composed of two parts: P_{cc} and $\frac{m_a^2}{2} P_{cc}$

The second part represents the sideband power introduced by the modulating amplifier in the Collector circuit.

Substitute equation (1.46) in equation (1.45) .

$$\boxed{P_d = P_{cc}\left[1 + \frac{m_a^2}{2}\right](1 - \eta_c)} \qquad (1.47)$$

This dissipation in the Collector circuit increases with an ***increase in modulation index***. This dissipation generates heat.

$$\boxed{P_{out} = \eta P_{in} = \eta \times P_{cc}\left[1 + \frac{m_a^2}{2}\right]} \quad (1.48)$$

The modulated output is obtained at the collector. The collector modulator has higher collector efficiency and higher power ***output per transistor.***

Advantages of Collector Modulator:

- Better linearity
- High efficiency and
- Higher output power per transistor
- It produces a more symmetrical envelope.

1.5Comparison of Low Level and High Level Modulation

Sl. No.	Parameter	High level modulation	Low level modulation
1.	Power level	Modulation takes place at high power level.	Modulation takes place at a low power level.
2.	Types of amplifiers	High efficient class C amplifiers are used.	Linear amplifiers (A, AB or B) are used after modulation.
3.	Efficiency	Very high.	Lower than high level modulators.
4.	Devices used	Vacuum tubes, Transistors FET	Transistors, FET, OP – AMPs, Diodes
5.	Design of AF power amplifier	Complex due to very high power involved.	Easy as it is to be done at low power.
6.	Applications	High power broadcast transmitters.	Used for wireless intercom, remote control, walkie-talkie, sometimes used in TV transmitters.
7.	Modulation	Modulation takes place prior to the output element of the final stage of the transmitter.	Modulation takes place in the last RF amplifier stage of the transmitter.

1.6Non – Linear Modulation

A simple diode or transistor or FET can be used as a non-linear modulator by restricting the operation over the non-linear region of its characteristics. In this case the input to the diode is kept so small, the diode operates in a non-linear region of its V-I characteristics. This method is useful only for small signal amplitude modulation.

1.6.1Square law modulator

A square law modulator requires to add up the carrier and modulating signal to obtain AM with carrier. Thus a square law modulator has three features as shown in Fig. 1.7.

1. Summer – To add carrier and modulating signal
2. A non – linear (active) element.
3. Bandpass filter for extracting desired modulating products.

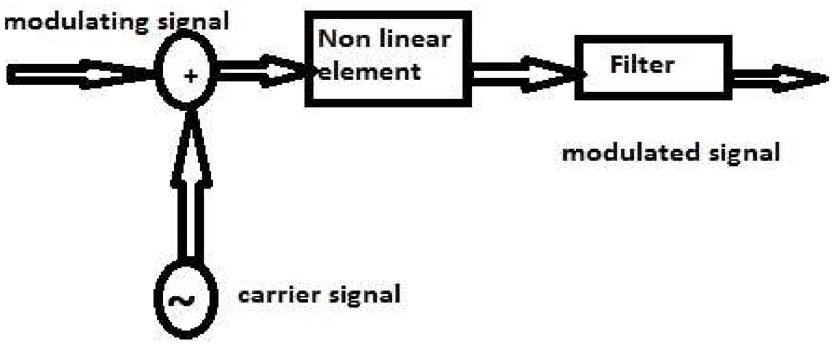


Fig1.7 Square law modulator

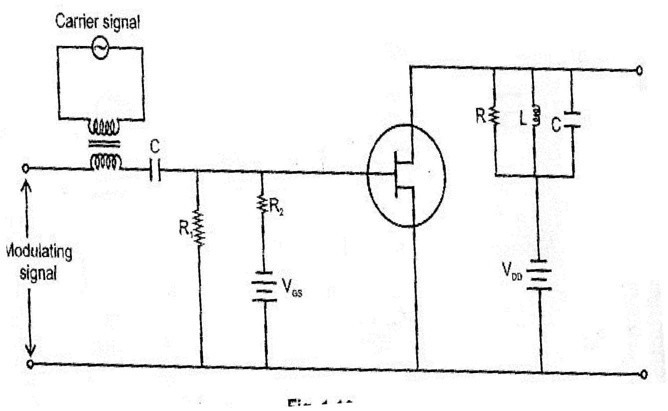


Fig.1.8 circuit diagram of square law detector using FET

The Fig.1.8 shows the circuit diagram of square law detector using FET. The FET is biased in a non-linear region of its transfer characteristics, to obtain the desired output, and the output tank circuit RLC is tuned to the carrier frequency to select the desired modulating components.

Analysis

When a non – linear element such as diode or FET is suitably biased and operated in a restricted portion of its non-linear transfer characteristics, the resulting current i_o will be given by the square law equation

$$i_o = a_1 V_1 + a_2 V_1^2 + \ldots\ldots\ldots \quad (1.49)$$

where V_1 = input voltage applied to the FET

$V_1 = A_m \sin\omega_m t + A_c \sin\omega_c t$

Substituting above Eq.(1.49)

$$\begin{aligned} i_o(t) &= a_1 (A_m \sin\omega_m t + A_c \sin\omega_c t) + a_2 (A_m \sin\omega_m t + A_c \sin\omega_c t)^2 \ldots \\ &= a_1 A_m \sin\omega_m t + a_1 A_c \sin\omega_c t + a_2 A_m^2 \sin^2\omega_m t + a_2 A_c^2 \sin^2\omega_c t \\ &\quad + 2A_m A_c a_2 \sin\omega_m t \sin\omega_c t \ldots\ldots \end{aligned}$$

Neglecting second and higher order terms, we get

$$\begin{aligned} &= a_1 A_m \sin\omega_m t + a_1 A_c \sin\omega_c t + 2a_2 A_m A_c \sin\omega_c t \sin\omega_m t \\ &= a_1 A_m \sin\omega_m t + a_1 A_c \sin\omega_c t + \frac{2a_2 A_m A_c}{2} (\cos(\omega_c - \omega_m)t - \cos(\omega_c + \omega_m)t) \\ &= a_1 A_m \sin\omega_m t + a_1 A_c \sin\omega_c t + a_2 A_m A_c \cos(\omega_c - \omega_m)t - a_2 A_m A_c \cos(\omega_c + \omega_m)t \end{aligned}$$

when the bandpass filter is tuned to the carrier frequency it allows only ω_c, $(\omega_c - \omega_m)$ and $(\omega_c + \omega_m)$ terms and it eliminates all other terms. Hence we obtain.

$$i(t) = \underbrace{a_1 A_c \sin\omega_c t}_{\text{Carrier}} + \underbrace{a_2 A_m A_c \cos(\omega_c - \omega_m)t}_{\text{LSB}} - \underbrace{a_2 A_m A_c \cos(\omega_c + \omega_m)t}_{\text{USB}} \quad (1.50)$$

1.6.2 Square law diode modulator

Square law diode modulation circuit makes use of non linear current voltage characteristics of diode. This method is suited at low voltage levels because of the fact that current – voltage characteristic of a diode is highly non-linear particularly in the low voltage region as shown in Fig.1.9.

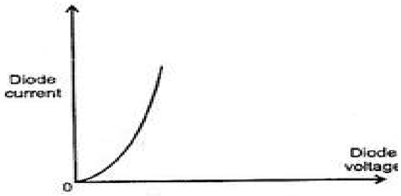


Fig.1.9 Current – voltage Characteristics of a diode

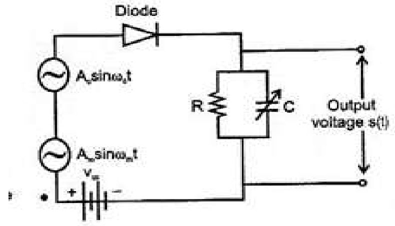


Fig.1.10 Square law diode modulator

It may be observed from the Fig.1.10 that the carrier and modulating signals are applied across the diode. A d.c. battery V_{cc} is connected across the diode to get a fixed operating point on the V_I characteristics of diode. The working of this circuit may be explained by considering the fact when two different frequencies are passed through a non linear device, the process of amplitude modulation takes place. Hence when carrier and modulating frequencies are applied at the input of diode, the different frequency terms appear at the output of diode. these different frequency terms are applied across a tuned circuit which is tuned to the carrier frequency and has a narrow bandwidth just to pass two sidebands along with the carrier and reject other frequencies. Hence at the output of tuned circuit, carrier and two sidebands are obtained i.e., Amplitude modulated (AM wave is produced).

1.6.3Balanced modulator

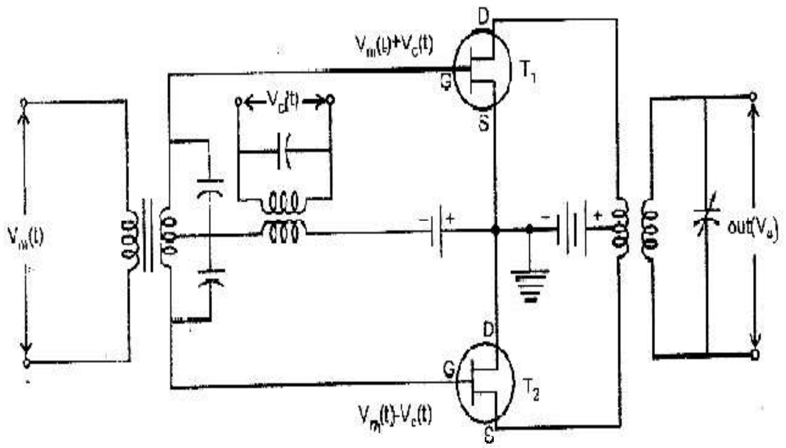


Fig1.11 Balanced modulator

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

Si ... given by

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

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

Therefore,

$V_0 = 2ka_1A_c(1+m_a \text{ Sin } \omega_m t)\text{Sin } \omega_c t$

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.

1.7Demodulation 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
and
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.

1.7.1Square 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.

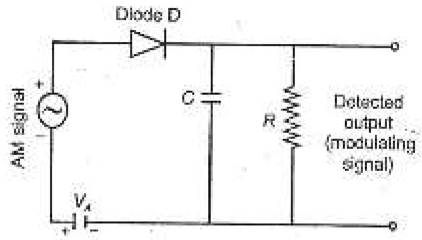


Fig 1.12 square law detector

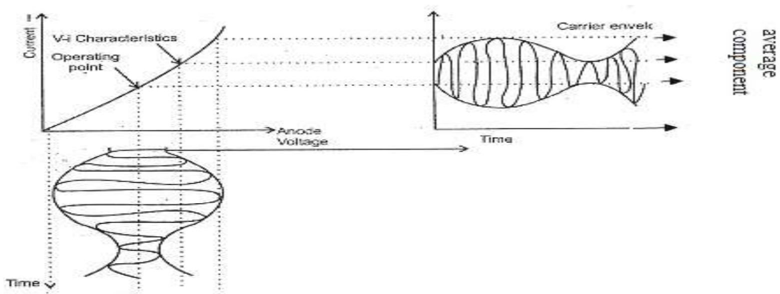


Fig 1.13 Wave shape of applied voltage and resulting currents in square law diode modulation

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.

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

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.

1.7.2Diode 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.

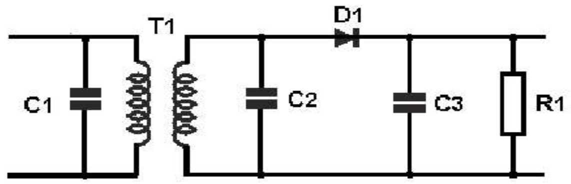


Fig1.14 Diode detector

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.

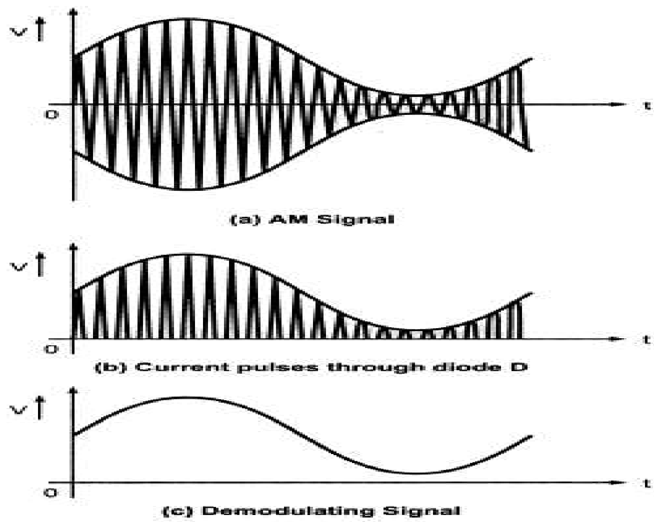


Fig. 1.15 Diode detector waveforms

Therefore it can also be defined as I_m / I_c with

$I_m = \frac{Em}{Zm}$ and $I_c = \frac{Ec}{Rc}$

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

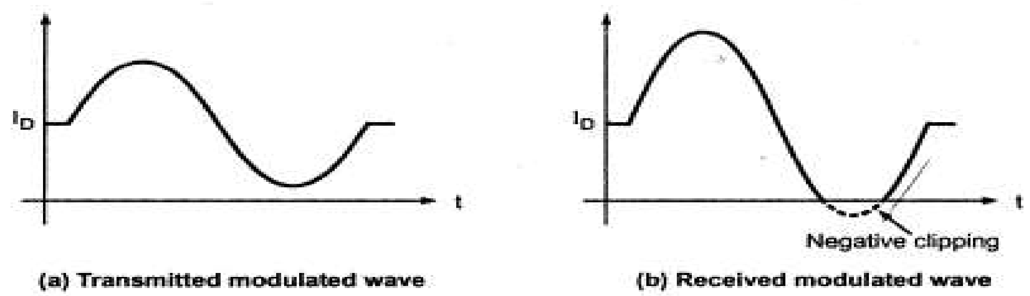


Fig.1.16 Negative peak clipping

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.

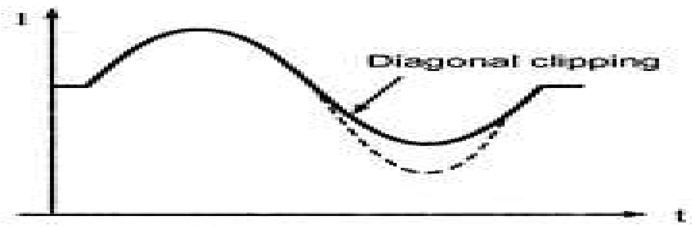


Fig.1.17 Diagonal peak clipping

1.7.3Synchronous 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:

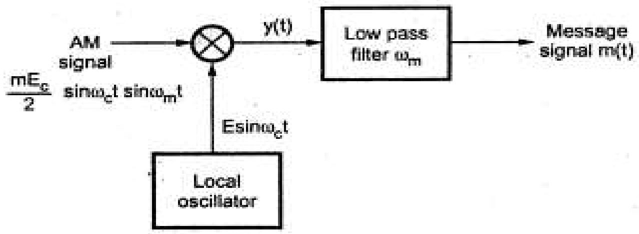


Fig 1.18 shows the block diagram of the synchronous detector.

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.

1.8DOUBLE SIDEBAND – SUPPRESSED CARRIER [DSB – SC]

Introduction

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=2f_m

1.8.1Expression for DSB –SC:

Let the modulating signal,

$$V_m(t) = V_m \sin\omega_m t \quad (1.54)$$

The carrier signal

$$V_c(t) = V_c \sin\omega_c t \quad (1.55)$$

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

$$V(t)_{DSB-SC} = V_m \sin\omega_m t \,.\, V_c \sin\omega_c t$$

$$= V_m V_c \sin\omega_m t \sin\omega_c t \quad (1.56)$$

$$\boxed{V(t)_{DSB-SC} = \frac{V_m V_C}{2}\left[\cos(\omega_C - \omega_m)t - \cos(\omega_C + \omega_m)t\right]} \quad (1.57)$$

We know that

$$V(t)_{AM} = V_c \sin\omega_c t + \frac{m_a V_c}{2}\left[\cos(\omega_C - \omega_m)t - \cos(\omega_C + \omega_m)t\right]. \quad (1.58)$$

Comparing equation (1.57) and (1.58) the carrier terms $V_c \sin\omega_c t$ is missing and only upper and lower sidebands are present.

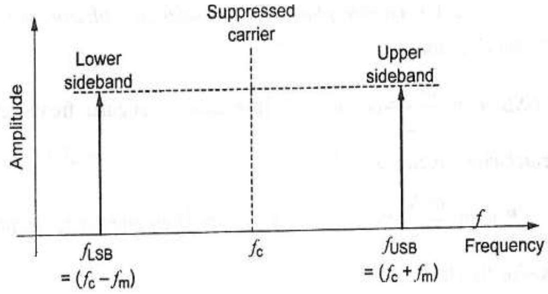


Fig 1.19 Frequency spectrum of DSB-SC-AM

In this fig, the carrier term f_c is suppressed. It contains only two sideband terms having the frequency of $(f_c - f_m)$ and $(f_c + f_m)$.

The DSB - SC modulated signal V(t) $_{DSB-SC}$ under goes a phase reversal whenever the message signal Vm(t) across zero.

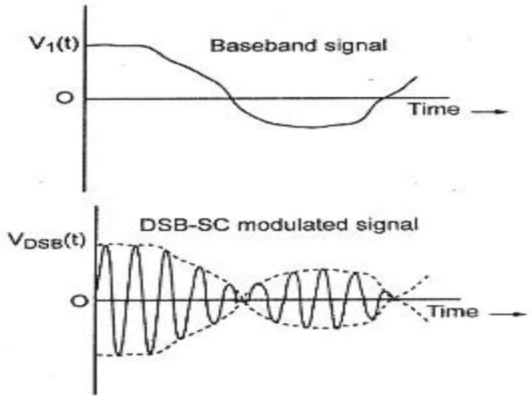


Fig.1.20 Graphical representation of DSB-SC-AM

Phasor Diagram of DSB – SC – AM

Assume that the coordinate system rotates in anticlockwise direction at an angular frequency of Let us assume the carrier phasor is the reference and oriented in horizontal direction.

The USB term $\frac{maVc}{2}\cos(\omega_c + \omega_m)t$ rotates an angular frequency of ω_m in anticlockwise direction.

The LSB term $\frac{maVc}{2}\cos(\omega_c - \omega_m)t$ rotates at an angular frequency of ω_m in clockwise direction.

The resultant amplitude of the modulated wave at any point is the vector sum of the two sidebands.

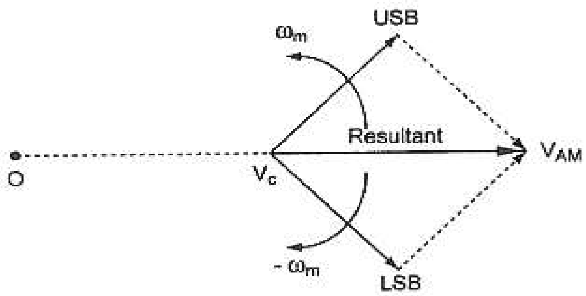


Fig1.21 phasor diagram of DSB-SC-AM

1.8.2Power Calculation

The total 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)

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

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

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

% 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, 66.7% of power is saved due to the suppression of the carrier wave.

1.9Generation of DSB – SC – AM

There are two ways of generating **DSB – SC – AM** such as ,

(i) Balanced modulator,

(ii) Ring modulator.

1.9.1Balanced Modulator

The circuit that is very commonly used for **DSB – SC generation**.

In balanced modulator, two non-linear devices are connected in the balanced mode, so as to suppress the carrier wave.It is assumed that the two transistors are identical and the circuit is symmetrical. Since the operation is confined in non-linear region of its transfer characteristics.

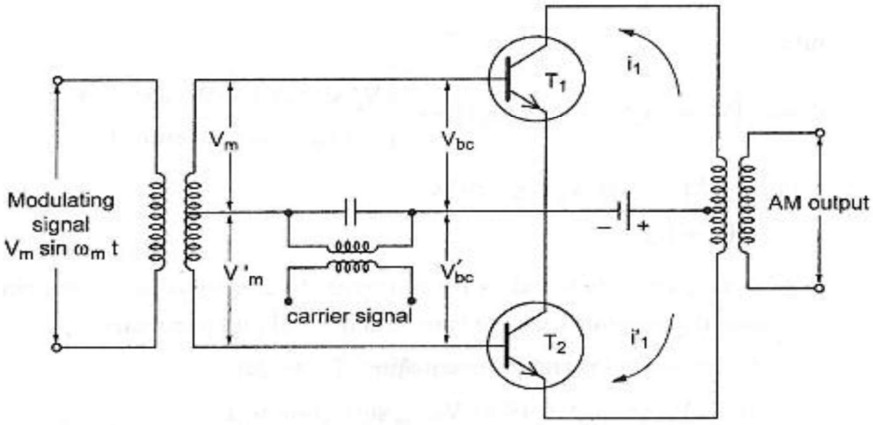


Fig1.22 Balanced modulator

The modulating voltage across the two windings of a centre-tap transformer is equal,and opposite in phase,

i.e., $V_m = V'_m$

The input voltage to transistor T_1 is given by

$$V_{bc} = V_c + V_m$$
$$= V_c \sin\omega_c t + V_m \sin\omega_m t \quad (1.65)$$

Since both V_c and V_m are in phase.

Similarly, the input voltage to transistor T_2 is given by

$$V'_{bc} = V'_m + V_c$$
$$= -V_m \sin\omega_m t + V_c \sin\omega_c t \quad (1.66)$$

By the non-linearity relationship the collector current can be written as

$$i_1 = a_1 V_{bc} + a_2 V_{bc}^2 \quad (1.67)$$

$$i'_1 = a_1 V'_{bc} + a_2 V'^2_{bc} \quad (1.68)$$

Substituting the values of V_{bc} and V'_{bc} from equation (1.65) and (1.66), we get

$$i_1 = a_1 \left[V_c \sin\omega_c t + V_m \sin\omega_m t\right] + a_2 \left[V_c \sin\omega_c t + V_m \sin\omega_m t\right]^2$$
$$= a_1 \left[V_c \sin\omega_c t + V_m \sin\omega_m t\right] + a_2 \begin{bmatrix} V_c^2 \sin^2\omega_c t + V_m^2 \sin^2\omega_m t \\ + 2V_m V_c \sin\omega_m t \sin\omega_c t \end{bmatrix} \quad (1.69)$$

Similarly,

$$i_1' = a_1\left[V_c \sin\omega_c t - V_m \sin\omega_m t\right] + a_2\left[\begin{array}{l} V_c^2 \sin^2\omega_c t + V_m^2 \sin^2\omega_m t \\ -\,2V_m V_c \sin\omega_m t \sin\omega_c t \end{array}\right] \quad (1.70)$$

The output AM voltage V_0 is given by

$$V_o = K(i_1 - i'_1) \quad (1.71)$$

This is because currents i_1 and i'_1 flow in opposite directions in a tuned circuit.
K is a constant depending on impedance and other circuit parameters.
Substituting equation (1.69) and (1.70) in equation (1.71), we get

$$V_o = 2K a_1 V_m \sin\omega_m t + 4K a_2 V_c V_m \sin\omega_c t \sin\omega_m t \quad (1.72)$$

The other terms are balanced out.
Balanced modulator equation (1.72) shows the carrier has been cancelled out, leaving only the sidebands and modulating signal.
Equation (1.72) can be written as

$$V_o = 2K V_m a_1\left[1 + \frac{2a_2 V_m}{a_1}\sin\omega_m t\right]\sin\omega_m t$$

$$V_o = 2Ka_1 V_m \left[1 + m_a \sin\omega_c t\right]\sin\omega_m t \quad (1.73)$$

Where $m_a = \dfrac{2a_2 V_m}{a_1}$ is the modulation index

Transistors are amplifying devices. The circuit can be fabricated using other amplifying devices like FET or electron tubes. The diode circuit is cheaper.
Whereas amplifying devices provide power gain.

1.9.2Ring Modulator (or) Diode Balanced Modulator
Introduction:
The one of the most popular method of generating a **DSB – SC** wave is ring modulator.
The circuit employs diodes as non-linear devices and the carrier signal is connected between centre taps of the input and output transformers.
There is no need for a band pass filter at the output. The four diodes are controlled by a carrier $V_c(t)$ of frequency f_c
The carrier signal acts as a switching signal to alternate the polarity of the modulating signals at the carrier frequency. For better understanding of the operation, assume that the modulating input is zero. Only carrier signal is present.

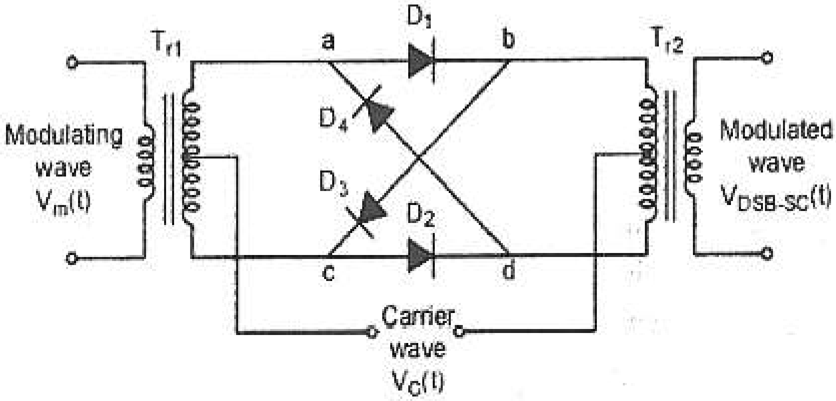


Fig. 1.23 Ring modulator

Positive Half Cycle of Carrier:

Diodes D_1 and D_2 are formward biased. AT this time D_3 and D_4 are reverse biased and act like open circuits. The current divides equally in the upper and lower portions of the primary winding of T_{12} .

The current in the upper part of the winding produces a magnetic field that is equal and opposite to the magnetic field produced by the current in the lower half of the secondary.

Therefore, these magnetic fields cancel each other out and no output is induced in the secondary. Thus the carrier is effectively suppressed.

Negative Half Cycle of Carrier:

When the polarity of the carrier reverses. Diodes D_1 and D_2 are reverse biased and diodes D_3 and D_4 conduct. Again the current flows in the secondary winding of T_{r1} and the primary winding of $T_{r2.}$

The equal and opposite magnetic fields produced in T_{r2} cancel each other out and thus result in zero carrier output. The carrier is effectively balanced out.

Principle of Operation:

When both the carrier and the modulating signals are present, during positive half cycle of the carrier, diodes D_1 and D_2 conduct, while diodes D_3 and D_4 does not conduct.

During negative half cycle of the carrier voltage diodes D_3 and D_4 conduct and D_1 and D_2 does not conduct.

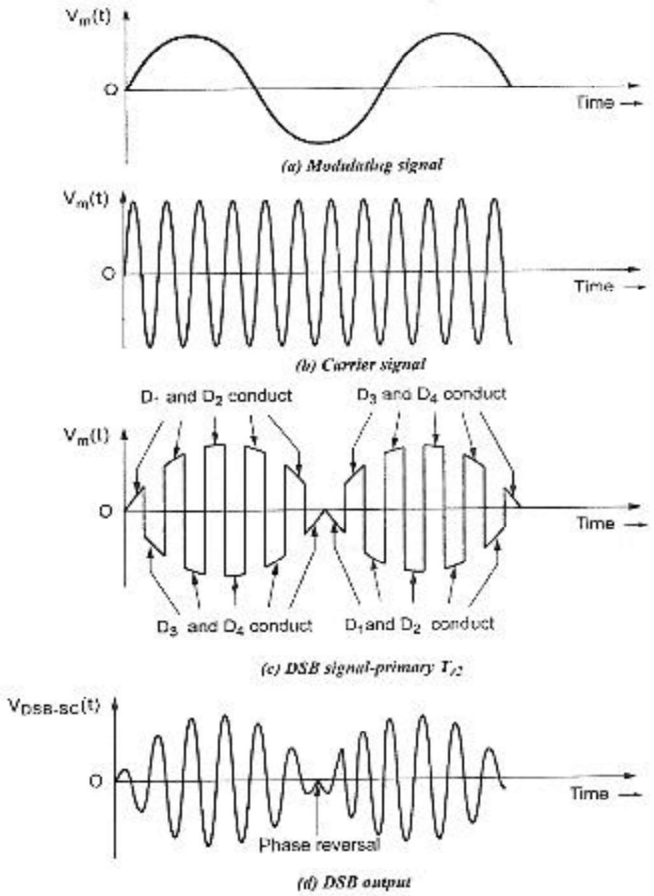


Fig.1.24 Graphical representation of DSB-SC signals

Phase Reversal

When polarity of the modulating signal changes, the result is a 180° phase reversal.

At the time, during the positive half cycle of the carrier, diodes D_3 and D_4 are in forward bias and the negative half cycle of the carrier, diodes D_1 and D_2 are in reverse bias.

Consider the modulating signal $V_m(t)$ and carrier signal $V_c(t)$, such that,

$$V_m(t) = V_m \sin \omega_m t \qquad (1.74)$$

$$V_c(t) = V_c \sin \omega_c t \qquad (1.75)$$

Output voltage $V_o(t) = V_m(t) \cdot V_c(t)$

$$V_o(t) = \frac{V_m V_c}{2}\left[\cos(\omega_c - \omega_m)t - \cos(\omega_c - \omega_m)t\right] \qquad (1.76)$$

The equation (1.76) shows that the output is free from the carrier and other higher order terms, and it contains upper and lower sidebands only.
The ring modulator circuit is also known as double balanced modulator because comparing to balanced modulator here two more diodes are used.

Advantages and Disadvantages of DSB – SC
Advantages:
1. DSB –SC is more efficient in transmitted power as compared to DSBFC .
2. DSB –SC has better signal to noise ratio as compared to single side band (SSB) transmission.

Disadvantage:
Even though the carrier is suppressed the bandwidth of DSBSC remains same as DSBFC.

1.10SINGLE SIDEBAND SUPPRESSED CARRIER (SSB – SC – AM)
Introduction
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.

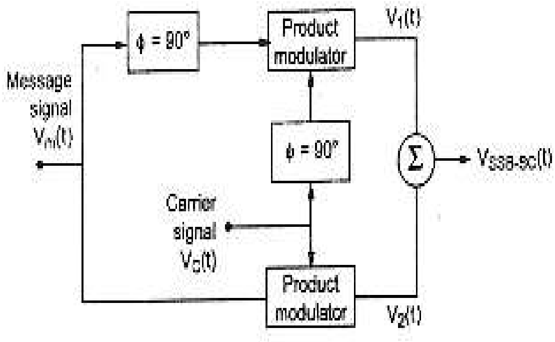


Fig.1.25 Block diagram of DSB-SC-AM

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'.

1.10.1Expression for DSB – SC

Let

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

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

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

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,

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

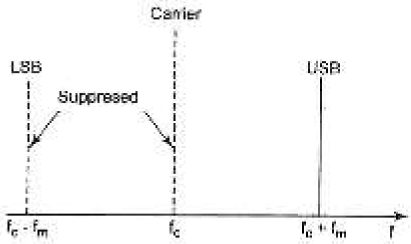


Fig1.26 frequency spectrum of SSB-SC-AM

Phasor diagram of SSB-SC-AM

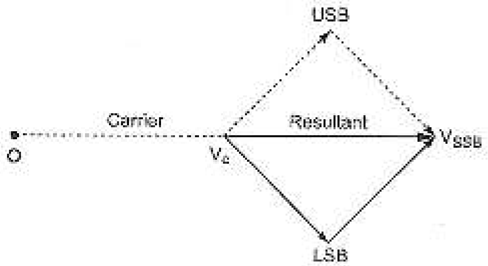


Fig1.27 Phasor diagram of SSB-SC-AM

1.10.2Power calculation

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,

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

P_t = total power transmitted.

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

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

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

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

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

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.

1.11Generation of SSB

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.

1.11.1Filter 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.

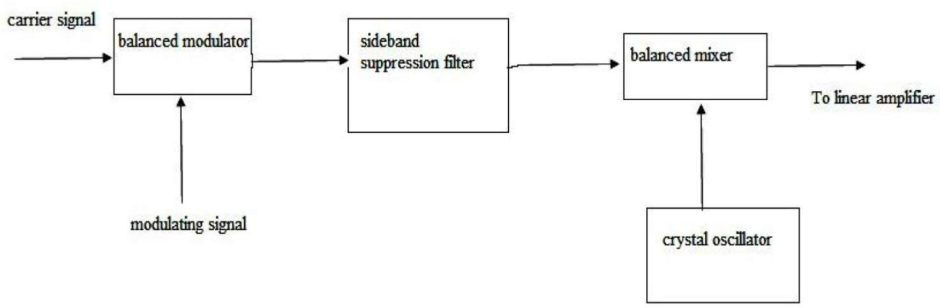


Fig.1.28 Filter method to suppress sideband

1.11.2Phase 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 ($+90^{\circ}$, -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.

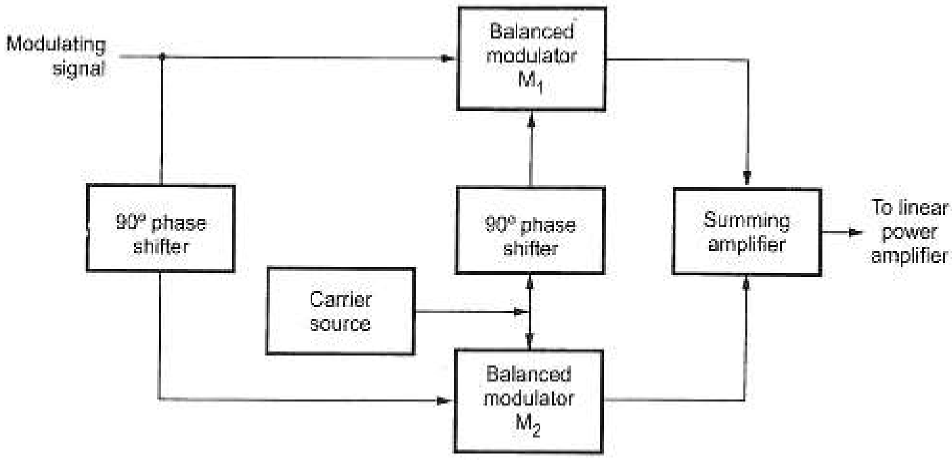


Fig.1.29 phase shift method to generate SSB

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 $\sin(\omega_c t + 90^{\circ})$. Hence output of M_1 will

be,

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

$$= \cos(\omega_c t - \omega_m t + 90^{\circ}) - \cos(\omega_c t + \omega_m t + 90^{\circ}) \quad (1.85)$$

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

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

$$= \cos(\omega_c t - \omega_m t - 90^{\circ}) - \cos(\omega_c t + \omega_m t + 90^{\circ}) \quad (1.86)$$

In the above equation observe that first term represents LSB with -90° phase shift and second term represents USB with $+90^{\circ}$ 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 $+90^{\circ}$ (i.e., in phase). Thus SSB is generated at the output of summing amplifier.

1.11.3Weaver's Method or Third Method of SSB Generation

Principle: It uses two carriers: One is audio subcarrier at frequency f_o and other is RF carrier at frequency f_c .

Block diagram and operation

Fig. 1.30 shows the block diagram of weaver's method.

The input to the modulator is s(t) = cos(2∏f_mt). It is the modulating signal of frequency f_m.

The audio subcarrier generates the carrier of frequency f_0. The input to upper multiplier is cos(2∏f_0t) and lower multiplier is sin(2∏f_0t).

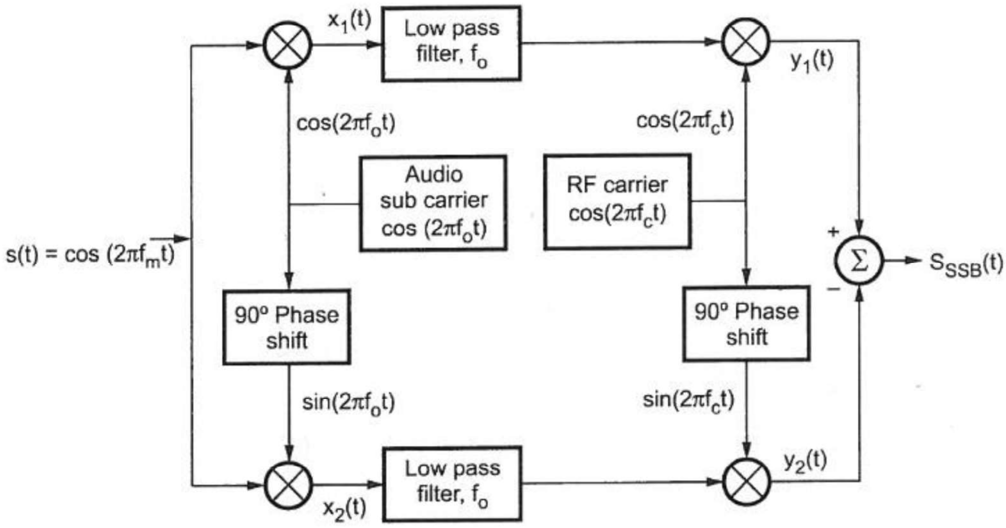


Fig.1.30 Block diagram of weaver's method

The signals $x_1(t)$ and $x_2(t)$ will be,

$$x_1(t) = cos\left(2\pi f_m t\right) \cdot cos\left(2\pi f_o t\right)$$

$$x_1(t) = \frac{1}{2}\left\{cos\ 2\pi(f_o + f_m)\ t + cos\ 2\pi(f_o - f_m)\ t\right\} \tag{1.87}$$

And

$$x_2(t) = cos\left(2\pi f_m t\right) \cdot sin\left(2\pi f_o t\right)$$

$$x_2(t) = \frac{1}{2}\left\{sin\ 2\pi(f_o + f_m)t + sin\ 2\pi(f_o - f_m)t\right\} \tag{1.88}$$

The low pass filter have a bandwidth of f_0. Hence it passes only difference frequency terms in equation (1.87) and (1.88). Thus the outputs of low pass filters.

$$\text{Upper lowpass filter} \quad : \quad \frac{1}{2} \cos 2\pi(f_o - f_m)\, t$$

$$\text{Lower lowpass filter} \quad : \quad \frac{1}{2} \sin 2\pi(f_o - f_m)\, t$$

i) The RF carrier generates the carrier of frequency f_c. . The carriers of $\cos(2\pi f_c t)$ and $\sin(2\Pi f_c t)$ are given to upper and lower multipliers respectively.

ii) The SSB signal at output is obtained by subtraction of $y_1(t)$ and $y_2(t)$. Thus ,

$$S_{SSB}(t) = y_1(t) - y_2(t)$$
$$= \frac{1}{2} \cos 2\pi(f_c + f_o - f_m)\, t$$

Above signal contains only lower sideband at $f_c + f_0 - f_m$.

1.12COMPARISON OF AMPLITUDE MODULATION SYSTEMS

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	$\left[\frac{S}{N}\right]_0 = \frac{1}{3}\left[\frac{S}{N}\right]_I$	$\left[\frac{S}{N}\right]_0 = \left[\frac{S}{N}\right]_I$	$\left[\frac{S}{N}\right]_0 = \left[\frac{S}{N}\right]_I$	$\left[\frac{S}{N}\right]_0 = \left[\frac{S}{N}\right]_I$

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

1.13Superheterodyne Receiver

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.

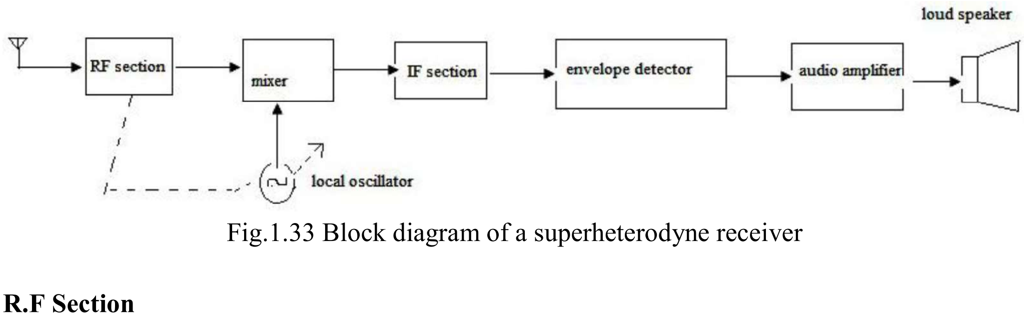


Fig.1.33 Block diagram of a superheterodyne receiver

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.

1.13.1FM Superheterodyne Receiver

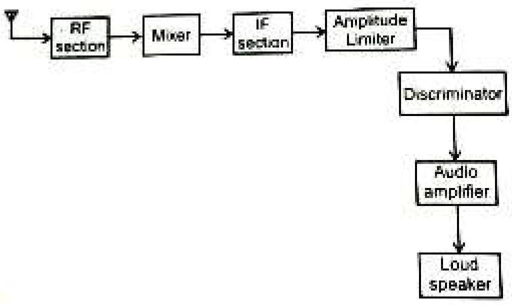


Fig.1.34 FM superhedrodyne receive

Amplitude Limiter

The basic difference between AM and FM superheterodyne receiver lies in the use of an FM demodulator such as limiter frequency discriminator. In FM system the message signal is transmitted by the instantaneous value of carrier signal & its amplitude remain constant. Therefore any variation of the carrier's amplitude at the receiver input must result from noise or interference. An amplitude limiter following the IF section is used to remove amplitude variations by clipping the modulated wave is rounded by a bandpass filter that suppresses harmonics of the carrier frequency. Thus the filter output is again sinusoidal, with an amplitude that in practically independent of the carrier amplitude of the receiver input. The Fig 1.34 shows basic block diagram of FM superheterodyne receiver.

1.13.2 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.

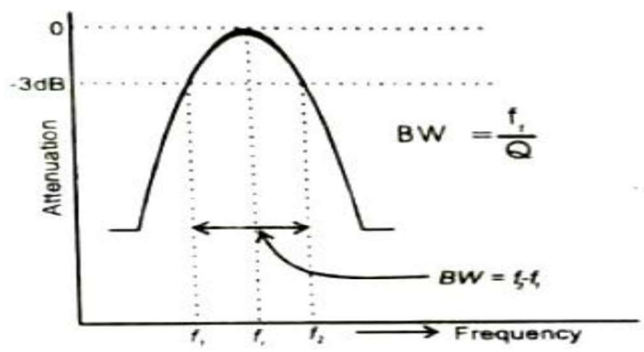


Fig.1.35 selectivity curve

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.

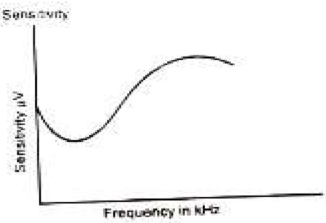


Fig1.36 sensitivity curve

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.

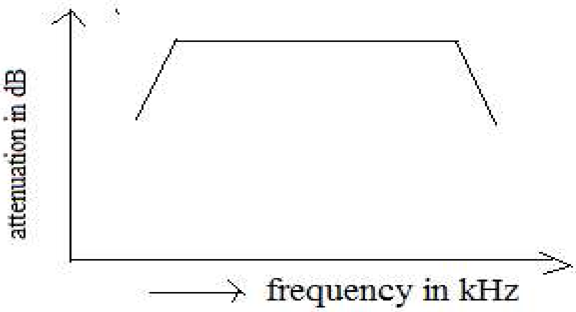


Fig.1.37 Fidelity curve

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.

1.14Hilbert Transform

If every frequency components of a signal f(t) is shifted by (-π/2) the resultant signal f_h (t) is the Hilbert transform of f(t).

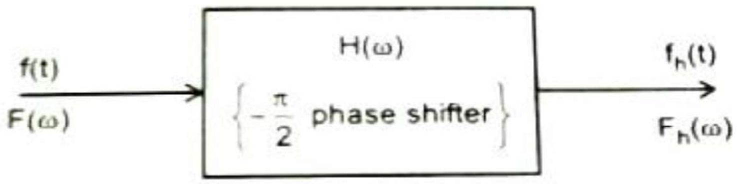


Fig1.38 phase shifting system

A signal f(t) is passed through a phase shift system H(ω) and the output f_h (t) shown in fig. 1.38.The characteristics of the of the system specified as follows:

i) The magnitude frequency components present in f(t) remains unchanged when it is passed through the system that is H(ω) = 1 and

ii) The phase of the positive frequency components if shifted by -π/2 . Since the phase spectrum θ(ω) has an odd symmetry, the phase of the negative frequency components is shifted by π/2. H(ω) and θ(ω) are plotted in Fig 1.39 by continuous and dotted lines respectively.

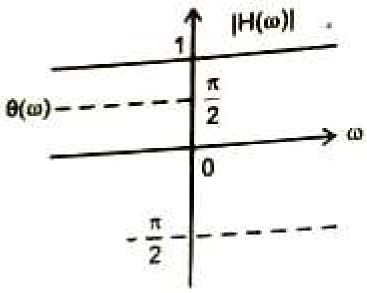


Fig1.39 Transfer function -π/2 phase shifter

The transfer is given as

$$H(\omega) = |H(\omega)|\, e^{j\theta\omega} = 1 e^{jH(\omega)} \quad (1.89)$$

It is evident from fig. That

$$\theta(\omega) = \begin{cases} \dfrac{\pi}{2} & \omega < 0 \text{ (i.e., negative frequencies)} \\ -\dfrac{\pi}{2} & \omega > 0 \text{ (i.e., positive frequencies)} \end{cases}$$

Therefore Eq 1.89 may be written as

$$H(\omega) = \begin{cases} e^{j\pi/2} & \omega < 0 \\ e^{-j\pi/2} & \omega > 0 \end{cases}$$

It is known that

$$e^{j\pi/2} = \cos\frac{\pi}{2} + j\sin\frac{\pi}{2} = j$$

$$e^{-j\pi/2} = \cos\frac{\pi}{2} - j\sin\frac{\pi}{2} = -j$$

Hence H(ω) can becomes

$$\frac{H(\omega)}{j} = \begin{cases} 1 & \omega < 0 \\ -1 & \omega > 0 \end{cases}$$
$$= -\operatorname{sgn}(\omega)$$
$$H(\omega) = -j\operatorname{sgn}(\omega) \qquad (1.90)$$

The response $F_h(\omega)$ of the system is related to signal $F(\omega)$ as

$F_h(\omega)=F(\omega)H(\omega)$ (1.91)

Where

$$f(t) \xrightarrow{FT} F(\omega) \text{ and } f_h(t) \xrightarrow{FT} F_h(\omega)$$

Substituting Eq 1.90 in Eq 1.91 we get

$$F_h(\omega) = F(\omega)(-jsgn(\omega)) \qquad (1.91(\ i))$$

Taking inverse Fourier transform of both sides of Eq 1.91(i)

$$f_h(t) = F^{-1}[-jF(\omega)\operatorname{sgn}(\omega)]$$

The time domain of sgn (ω) *is given* by

$$\frac{j}{\pi t} \xleftrightarrow{FT} \operatorname{sgn}(\omega) \quad \& \quad F(t) \xleftrightarrow{FT} F(\omega)$$

Using time convolution function

$$f_h(t) = -jF(t) * \frac{j}{\pi t} \qquad j^2 = -1$$
$$f_h(t) = \frac{1}{\pi}\left[f(t) * \frac{1}{t}\right]$$
$$= \frac{1}{\pi}\int_{-\infty}^{\infty} \frac{f(\tau)}{t-\tau} d\tau$$

Which is the Hilbert transform of f(t) .

$$f(t) = -\frac{1}{\pi}\int_{-\infty}^{\infty}\frac{f_h(\tau)}{t-\tau}d\tau$$

The inverse Hibert transform is given by

Using the inverse Hilbert transform, the orginal signal f(t) can be reconstructed from $f_h(t)$.

The function f(t) and $f_h(t)$ are considered to constitute a Hibert transform pair.

A Hilbert transform is referred to an ideal device in which the amplitude of all frequency components in the signal are un affected by transmission through it. However it produces a phase shift of +90 degres phase shift for all negative frequencies and -90 degrees for all positive frequencies of the input signals.

1.14.1Properties of Hilbert transform

Hilbeert transform has the following properties

1. A signal f(t) and its Hilbert transform have the same magnitude spectrum.
2. A signal f(t) and its Hilbert transform $s_h(t)$ have the same energy density spectrum.
3. If the Hilbert transform of $f_h(t)$ is –f(t), then $f_h(t)$ is Hilbert transform of s(t) that is if

$$H[f(t)] = f_H(t)$$
$$H[f_H(t)] = -f(t')$$

Where H denotes the Hilbert transform.

4. A signal f(t) and its Hilbert transform $f_h(t)$ are mutually orthogonal over the time interval (-∞,∞) that is

$$\int_{-\infty}^{\infty} f(t)f_h(t)dt = 0$$

5. A signal f(t) and its Hilbert transform $f_h(t)$ have the same auto correlation function.

Some useful Hilbert transform.

1. $\cos\omega_c t \xrightarrow{H} \sin\omega_c t$
2. $\sin\omega_c t \xrightarrow{H} \cos\omega_c t$
3. $\sin(\omega_c t + \theta) \xrightarrow{H} \cos\left(\omega_c t + \theta - \frac{\pi}{2}\right)$
4. Let $m(t)$ be a low pass signal with cutoff frequency W_1 and $c(t)$ a high pass signal with lower cut off frequency $\omega_2 > W_1$. Then

$$m(t)c(t) \xrightarrow{H} m(t)\overset{n}{c}(t)$$

1.14.2Application of Hilbert transform pair

Generation of SSB signal
Design of minimum phase type filters
Representation of band pass signals.

1.15 Pre Envelope and Complex Envelope

Pre envelope or analytic signal

The concept of free envelope also known as the analytic function is useful in driving the general expression of the SSB –SC signal. The pre envelope of real valued signal f(t) is defined as

$$\mathbf{f_p\,(t) = f(t) + jf_h(t)} \qquad (1.92)$$

Where $f_h(t)$ is a Hilbert transform of f(t). Obviously, the pre envelope $F_p(t)$ is a complex valued signal. The real part of $f_p(t)$ is F(t) and the imaginary part is its Hilbert transform $f_h(t)$. The complex conjugate of the pre envelope denoted by $f^*_p(t)$ is given by

$$\mathbf{f^*_p(t) = f(t) - jf_h(t)} \qquad (1.93)$$

Complex envelope

The new quality based on the analytic signal, called the complex envelope is defined as

$$\mathbf{f(t) = f(t)e^{j2\pi fct}}$$

The part f(t) is called the complex envelope of the signal f(t).
Taking Fourier transform

$$F(\omega)\begin{cases} 2F[f+f_c] & \text{for } f>0 \\ F(0) & \text{for } f=0 \\ 0 & \text{for } f<0 \end{cases}$$

The complex envelope is just the low pass signal, part of the analytic signal. The analytic low pass signal has been multiplied by the complex exponential at the carrier signal.

1.16 Costas PLL detection scheme for DSB-SC AM (Costas receiver)

This system is used for synchronous detection of DSB-SC signal. The schematic representation of Costas receiver is shown in figure 1.40.

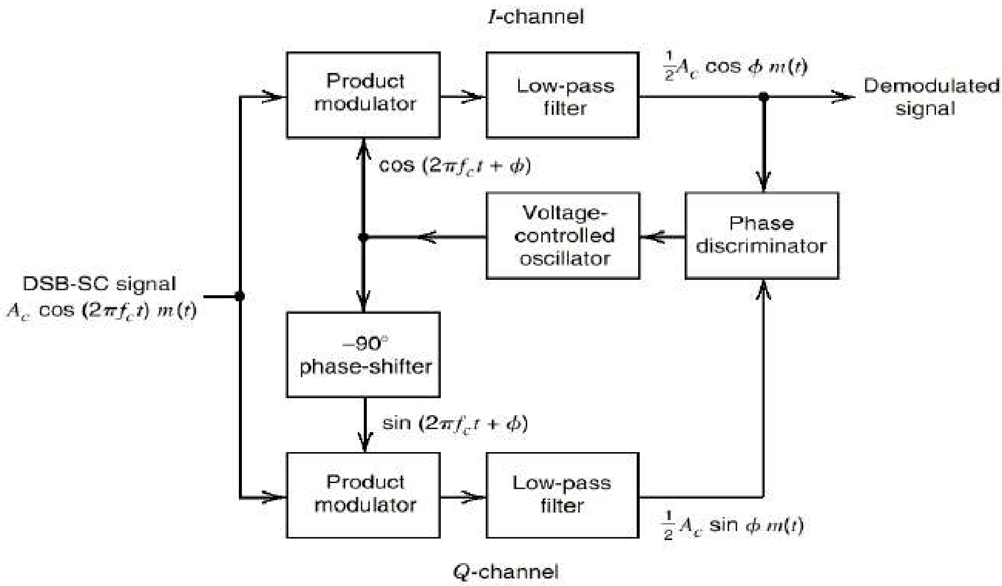


Figure 1.40 Costa's receiver

The system has two synchronous detectors, one detector is fed with a locally generated carrier which is inphase with the transmitted carrier. This detector is known as **Inphase coherent detector or I-channel.**

The other synchronous detector employs a local carrier which is in phase quadrature with the transmitted carrier, and is known as **Quadrature phase coherent detector or Q-channel.**

Operation principle

Consider that the local carrier signal is synchronized with the transmitted carrier. The output of the I-channel is the desired modulating signal, but the output of the Q-channel is zero due to quadrature null effect. Now consider that the local oscillator frequency drifts slightly, I-channel output is almost unchanged but the Q-channel is not zero rather some signal appears at its output.

The output of the Q-channel,

- is proportional to Φ.
- Will have a polarity same as the I-channel for the other direction of phase shift in local oscillator, whereas the polarity will be opposite to I-channel for the direction of phase shift.

The phase discriminator provides a DC control signal which may use to correct local oscillator phase error. The local oscillator is a VCO.

Limitations

- The Costas receiver ceases phase control where there is no modulation.
- The re-establishment is so rapid that distortion is not perceptible in voice communication.