UNIT 1 BASICS OF ELECTRONIC COMMUNICATION AND NOISE THEORY

Review of time and frequency domain description of signals - Communication system: point to point and broad cast - Basic model of a communication system: transmitter, receiver and channel - Fundamental limitations: Technological, Physical; Noise, bandwidth (signal and channel) and information capacity - Need for modulation and types - classification of communication based on modulation and channel - Base band and Pass band transmission - Electromagnetic spectrum allocation for various communication systems. Noise in Communication systems: Types and sources of noise; Atmospheric Noise, Thermal Noise, Shot noise, Partition noise, Flicker noise, Transit time noise - noise factor, noise factor for cascaded amplifier (Friss formula) -Noise figure - Equivalent noise temperature and bandwidth - Signal to Noise Ratio.

UNIT-1

1.1 Time/Frequency Domain Representation of Signals

Electrical signals have both time and frequency domain representations. In the time domain, voltage or current is expressed as a function of time as illustrated in Figure 1. Most people are relatively comfortable with time domain representations of signals. Signals measured on an oscilloscope are displayed in the time domain and digital information is often conveyed by a voltage as a function of time. Time domain visualization provides information such as shape of the signal and variation in voltage with respect to time. But it did not provide complete information regarding frequency content of a signal.

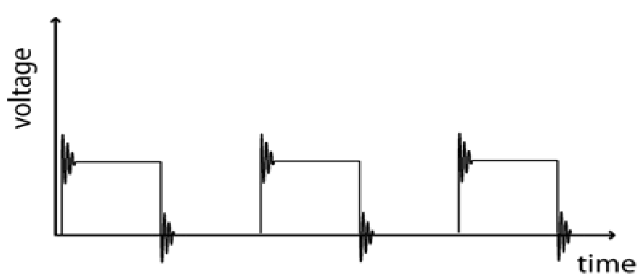


Figure 1. Time domain representation of an electrical signal.

Signals can also be represented by a magnitude and phase as a function of frequency. Signals that repeat periodically in time are represented by a discrete power spectrum as illustrated in Figure 2. Signals that are non periodic are represented by a continuous energy spectrum as illustrated in Figure 3.

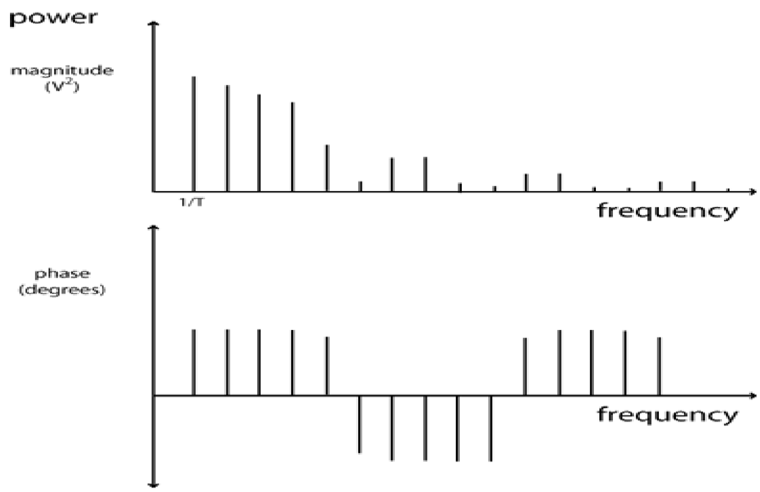


Figure 2. Power spectrum of a periodic signal.

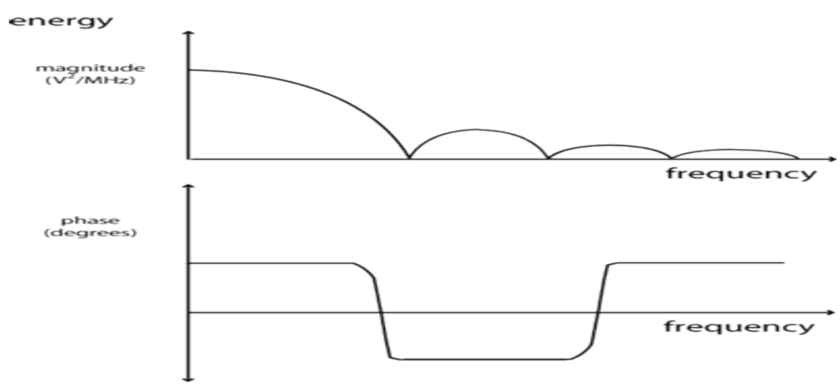


Figure 3. Energy spectrum of a time-limited (transient) signal.

Frequency domain representations are particularly useful when analyzing linear systems. EMC and signal integrity engineers must be able to work with signals represented in both the time and frequency domains. Signal sources and interference are often defined in the time domain. However, system behavior and signal transformations are more convenient and intuitive when working in the frequency domain.

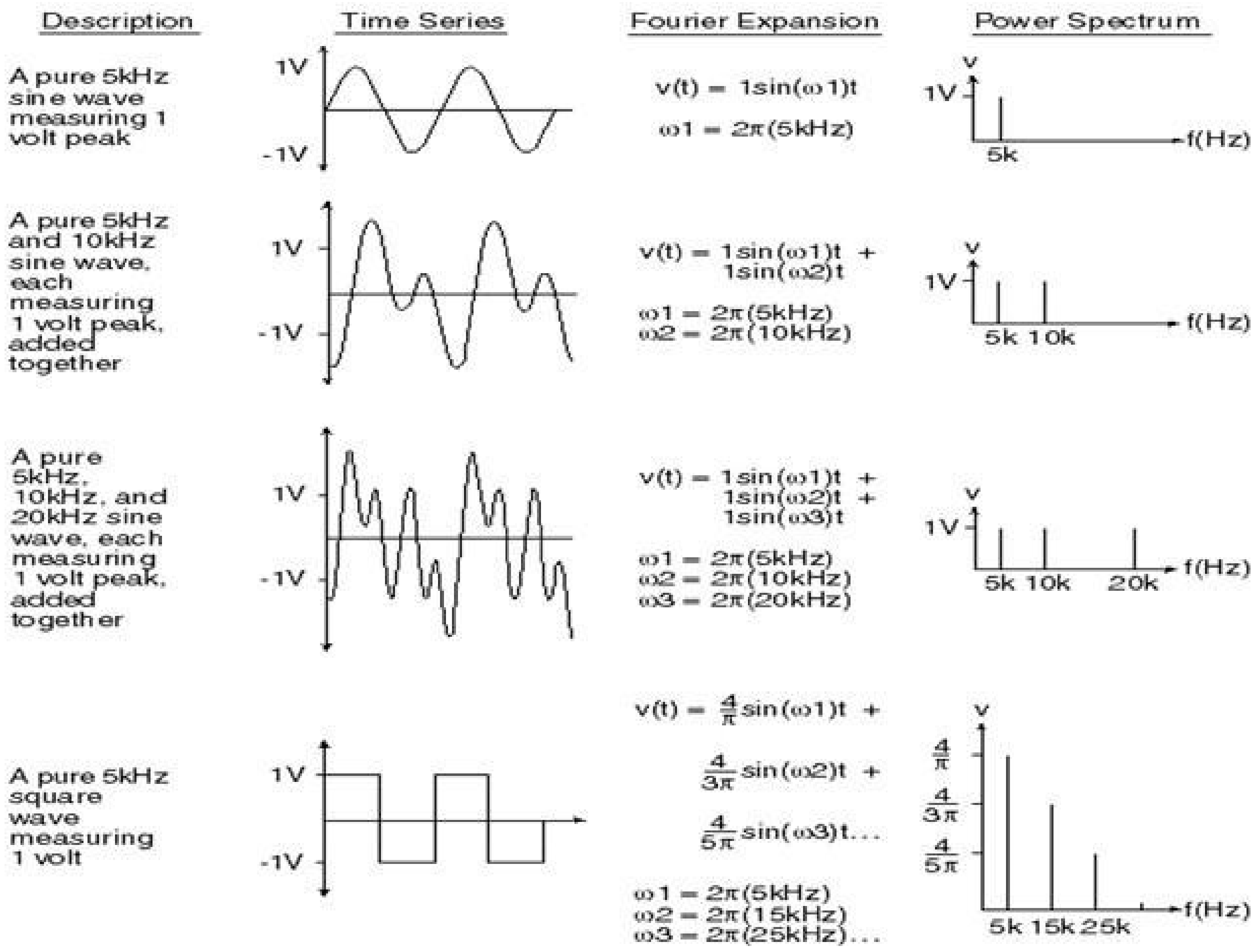


Figure 5. Periodic signals in the time and frequency domain.

Example 1: Frequency Domain Representation of a Pulse Train

Determine the frequency domain representation for the pulse train shown in Figure 6.

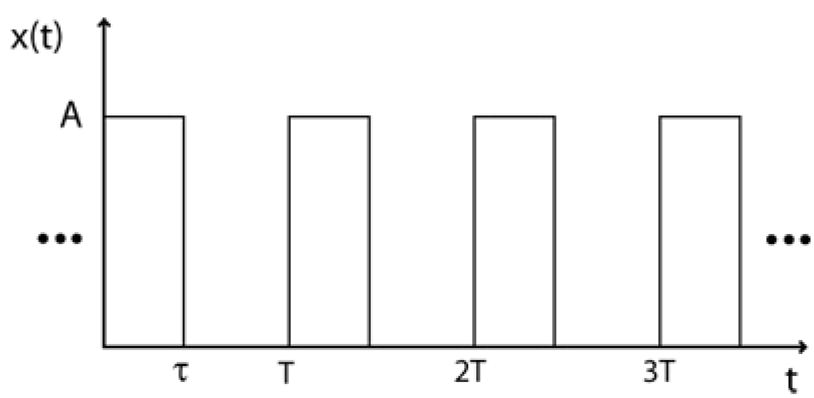


Figure 6: A pulse train.

In the time domain this signal is described by the following formula:

$$x(t)=\begin{cases} 1\ v & nT<t<nT+\tau \\ 0 & otherwise \end{cases} \qquad n=\pm 1,\pm 2,\pm 3,\cdots \qquad (1.1)$$

The coefficients of the Fourier series are then calculated using Equation (7b) as,

$$\begin{aligned} c_n &= \frac{1}{T}\int_0^T x(t)\, e^{-jn\omega_o t}dt \\ &= \frac{1}{T}\int_0^\tau (A)\, e^{-jn2\pi t/T}dt \\ &= \frac{A}{T}\int_0^\tau e^{-jn2\pi t/T}\, dt \\ &= \frac{A\tau}{T}\left[\frac{\sin\left(n\pi\tau/T\right)}{\left(n\pi\tau/T\right)}\right]e^{-j\left(n\pi\tau/T\right)} \end{aligned} \qquad (1.2)$$

Note that as $\tau \rightarrow 0$, our time domain signal looks like an impulse train and the amplitudes of all the harmonics approach the same value. As $\tau \rightarrow T/2$, the signal becomes a square wave and the magnitude of the harmonics becomes,

$$c_n = \frac{A}{2}\left|\frac{\sin\left(\frac{n\pi}{2}\right)}{\left(\frac{n\pi}{2}\right)}\right|\left|e^{-j\left(\frac{n\pi}{2}\right)}\right| = \begin{cases} \frac{A}{n\pi} & n = \pm 1, \pm 3, \pm 5 \cdots \\ 0 & n = \pm 2, \pm 4, \pm 6 \cdots \end{cases} \quad .(1.3)$$

In this case, the amplitude of the even harmonics is zero and the odd harmonics decrease linearly with frequency (n).

1.1.1 Fourier Transform

Transient signals (i.e. signals that start and end at specific times) can also be represented in the frequency domain using the Fourier transform. The Fourier transform representation of a transient signal, x(t), is given by,

$$X(\omega) = \int_{-\infty}^{\infty} x(t)\, e^{-j\omega t} dt \qquad (1.4)$$

The inverse Fourier transform can be used to convert the frequency domain representation of a signal back to the time domain,

$$x(t) = \frac{1}{2\pi}\int_{-\infty}^{\infty} X(\omega) e^{j\omega t} d\omega \quad .(1.5)$$

Some transient time domain signals and their Fourier transforms are illustrated in Figure.7.

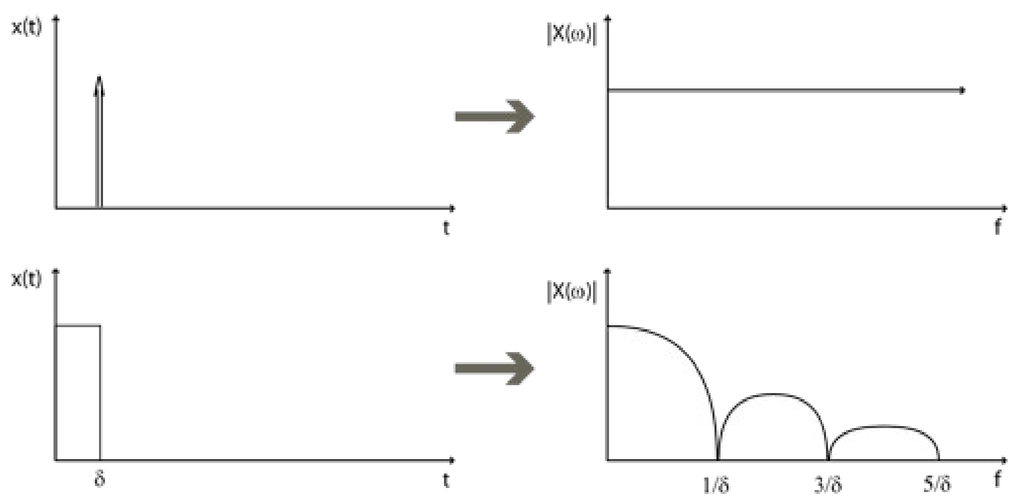


Figure 7. Transient signals in the time and frequency domain.

Note that transient signals have zero average power (when averaged over all time), but they have finite energy. The total energy in a transient time domain signal is given by,

$$E = \int_{-\infty}^{\infty} x^2(t)\, dt \qquad (1.6)$$

This must equal the total energy in the frequency domain representation of the signal,

$$E = \int_{-\infty}^{\infty} \left| X(\omega) \right|^2 d\omega \qquad (1.7)$$

Example 2: Spectrum of a pure sinusoidal signal

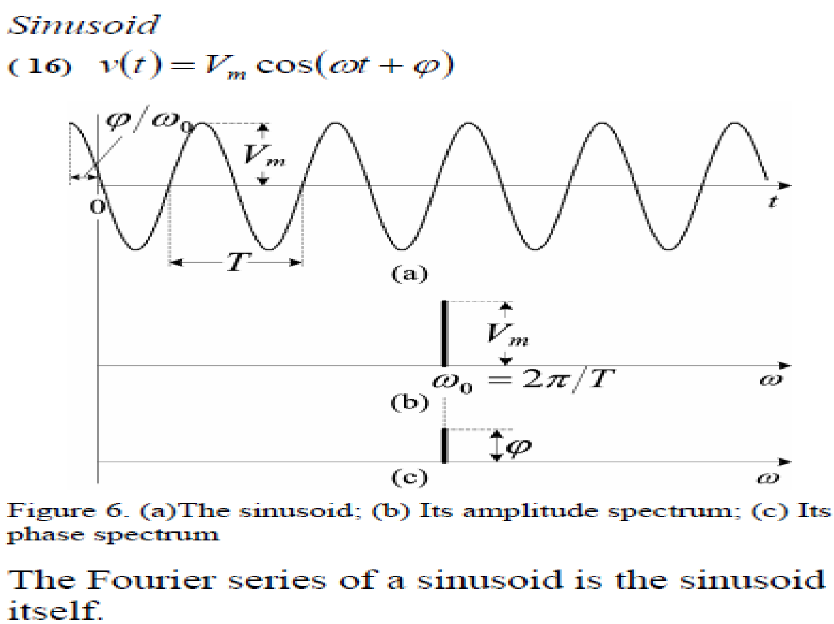


Figure 8: spectrum of a sinewave

Note:

Frequency domain representation of a pure sine wave will consist of only one frequency component at w_0.

All other signals (except pure sine wave) will consists of more than one frequency component in the spectrum

Example 3: Spectrum of rectangular and square wave forms

Train of Rectangular Pulses

(17) $v(t) = \begin{cases} V_m \text{ for } nT \le t < nT + t_p, \; n\text{ - integer;} \\ 0 \text{ otherwise} \end{cases}$

T – period;

V_m – height of the pulses;

t_p – pulse duration;

$d = \dfrac{t_p}{T}$ - duty cycle (usually in %)

Fourier series:

(18) $$v(t) = V_m \frac{t_p}{T} + \sum_{n=1}^{\infty} \frac{2V_m \sin\left(\frac{n\pi t_p}{T}\right)}{n\pi} \cos\left(\frac{n2\pi}{T}t\right)$$

If $d = 1/N$ where N is integer, then every N^{th} harmonic is 0.

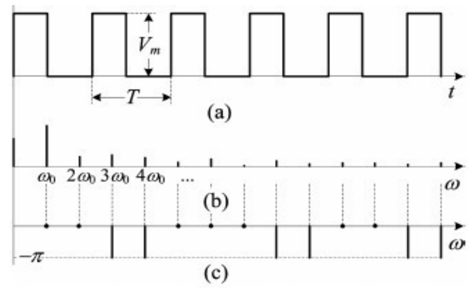


Figure 7. (a) The pulse train; (b) Its amplitude spectrum. (c) Its phase spectrum. The phase is either 0 or $-\pi$ depending on the sign of the sin function in (18).

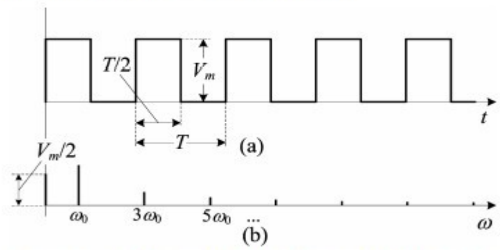


Figure 8. (a) Pulses with 50% duty cycle; (b) Its amplitude spectrum. The spectrum has only odd harmonics.

Figure:9. Time and frequency domain representation a pulse train

Note:

- Spectrum of rectangular pulse train shows that it consists of fundamental frequency component w_0 and harmonic frequencies at $2w_0$,$3w_0$,$4w_0$, ...
- Spectrum of square wave(50% duty cycle) shows that it consists of fundamental frequency component w_0 and harmonic frequencies at $3w_0$,$5w_0$,$7w_0$, ...
- Time domain representation do not give this information about harmonics

Example:4 Spectrum of a non sinusoidal signal

Frequency domain representation of the below non-sinusoidal signal shows that it consist of two sine waves at frequencies f and 3f.

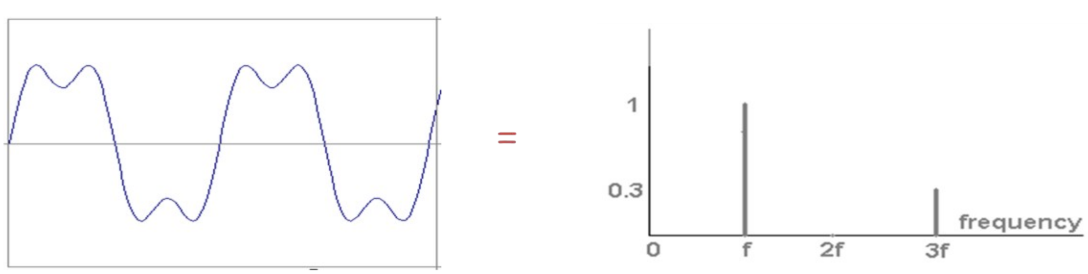


Figure 10: Spectrum of a non-sinusoidal signal

1.2 Introduction to Electronic Communication Systems

Communication is the process of establishing connection or link between two points for information exchange or Communication is simply the basic process of exchanging information.

The electronics equipments which are used for communication purpose, are called communication equipments. Different communication equipments when assembled together form a **communication system**. Typical example of communication system are line telephony and line telegraphy, radio telephony and radio telegraphy, radio broadcasting, point-to-point communication and mobile communication, computer communication, radar communication, television broadcasting, radio telemetry, radio aids to navigation, radio aids to aircraft landing etc.Fig.1 shows the block diagram of a general communication system, in which the different functional elements are represented by blocks.

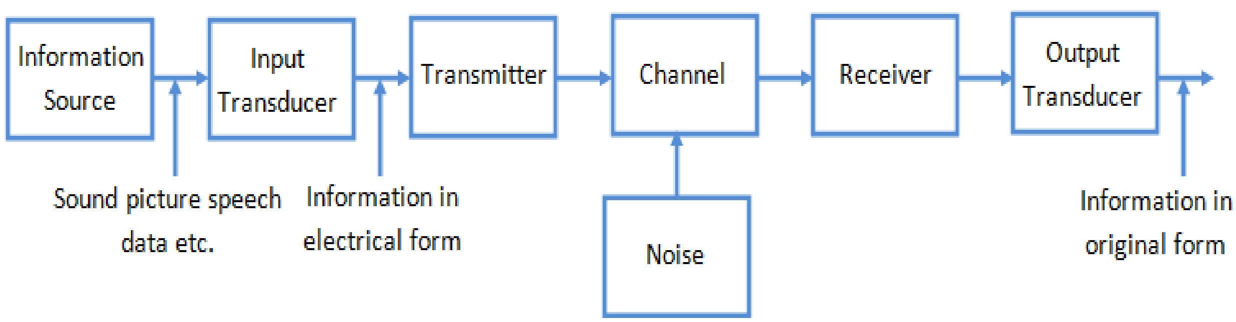


Fig 11: Block Diagram of Communication System

The essential components of a communication system are information source, input transducer, transmitter, communication channel, receiver and destination.Now, we shall discuss the functioning of these blocks.

(i) Information Source

As we know, a communication system serves to communicate a message or information. This information originates in the information source.In general, there can be various messages in the form of words, group of words, code, symbols, sound signal etc. However, out of these messages, only the desired message is selected and communicated.

Therefore, we can say that the function of information source is to produce required message which has to be transmitted.

(ii) Input Transducer

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

For example, in case of radio-broadcasting, a microphone converts the information or massage which is in the form of sound waves into corresponding electrical signal.

(iii) Transmitter

The function of the transmitter is to process the electrical signal from different aspects. It does modulation and amplification of the signal to be transmitted.

In the modulation process, some parameter of the carrier wave (such as amplitude, frequency or phase) is varied in accordance with the modulating signal . This modulated signal is then transmitted by the transmitter. The modulating signal is nothing but the baseband signal or information signal while the carrier is a high frequency sinusoidal signal. In the process of modulation the carrier wave actually carries the information signal from the transmitter to receiver .

For example in radio broadcasting the electrical signal obtained from sound signal, is processed to restrict its range of audio frequencies (upto 5 kHz in amplitude modulation radio broadcast) and is often amplified, modulated and then given to antenna for radiation in to space. In wire telephony, no real processing is needed. However, in long-distance radio communication, signal amplification is necessary before modulation.

Modulation is the main function of the transmitter. In modulation, the message signal is superimposed upon the high-frequency carrier signal. All these processings of the message signal are done just to ease the transmission of the signal through the channel.

(iv) Channel and the Noise

The term channel means the medium through which the message travels from the transmitter to the receiver. In other words, we can say that the function of the channel is to provide a physical connection between the transmitter and the receiver. There are two types of channels, namely point-to-point channels and broadcast channels.

Example of point-to-point channels is wire lines, microwave links and optical fibres. Wire-lines operate by guided electromagnetic waves and they are used for local telephone transmission. In case of microwave links, the transmitted signal is radiated as an electromagnetic wave in free space. Microwave links are used in long distance telephone transmission.

An optical fibre is a low-loss, well-controlled, guided optical medium. Optical fibres are used in optical communications. Although these three channels operate differently, they all provide a physical medium for the transmission of signals from one point to another point. Therefore, for these channels, the term point-to-point is used.

On the other hand, the broadcast channel provides a capability where several receiving stations can be reached simultaneously from a single transmitter. An example of a broadcast channel is a satellite in geostationary orbit, which covers about one third of the earth‟s surface.

Noise is an unwanted signal which tends to interfere with the required signal During the process of transmission and reception the signal gets distorted due to noise introduced in the system.. Noise signal is always random in character. Noise may interfere with signal at any point in a communication system. However, the noise has its greatest effect on the signal in the channel.

(v) Receiver

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

(vi) Destination

Destination is the final stage which is used to convert an electrical message signal into its original form.For example in radio broadcasting, the destination is a loudspeaker which works as a transducer i.e. converts the electrical signal in the form of original sound signal.

Points to Remember

Basic operations in the transmitter

• Modulation

• Amplification

Basic operations in the receiver

• Amplification

• Filtering

• Demodulation

Effects of the channel on the transmitted signal

• Attenuation: decreasing the signal strength;

• Distortion of the signal waveform(change in waveshape): caused by channel characteristics (linearity, frequency response, etc.)

• Noise: contamination of random natural signals added to the transmitted signal

• Interference: contaminations of extraneous signal of human sources – machinery, power lines, digital switching circuits, etc.

1.3 Types of Communication

1.3.1 Broadcast vs. Point-to-point

Broadcast: A method of sending a signal where multiple parties may hear a single sender. Radio stations are a good example of everyday life "Broadcast Network". In this example, you can see a single station is broadcasting a message to multiple locations that may or may not be able to hear it, and if they are able to hear it, may choose to listen or not.

Point-to-point: A method of communication where one "point" (person or entity) speaks to another entity.

1.3.2 Classification Based on Direction of Communication

Based on whether the system communicates only in one direction or otherwise, the communication systems are classified as under:

1. Simplex System
2. Half duplex System
3. Full duplex System

Simplex System

In these systems, the information is communicated in only one direction. For example, the radio or TV broadcasting system can only transmit, they cannot receive. Another example of simplex communication is the information transmitted by the telemetry system of a satellite to earth. The telemetry system transmits information about the physical status of the satellite such as its position or temperature.

Half duplex System

These systems are bidirectional, i.e. they can transmit as well as receive but not simultaneously. At a time, these systems can either transmit or receive, for example, a transceiver or walky talky set. The direction of communication alternates. The radio communications such as those in military, fire fighting, citizen band (CB) and amateur radio are half duplex system.

Full duplex System

These are truly bidirectional systems as they allow the communication to take place in both the directions simultaneously. These systems can transmit as well as receive simultaneously. For example, the telephone systems.

However, the bulk of electronic communications is two-way. The best example of full duplex communication system is telephone system. Fig. 2 shows this classification .

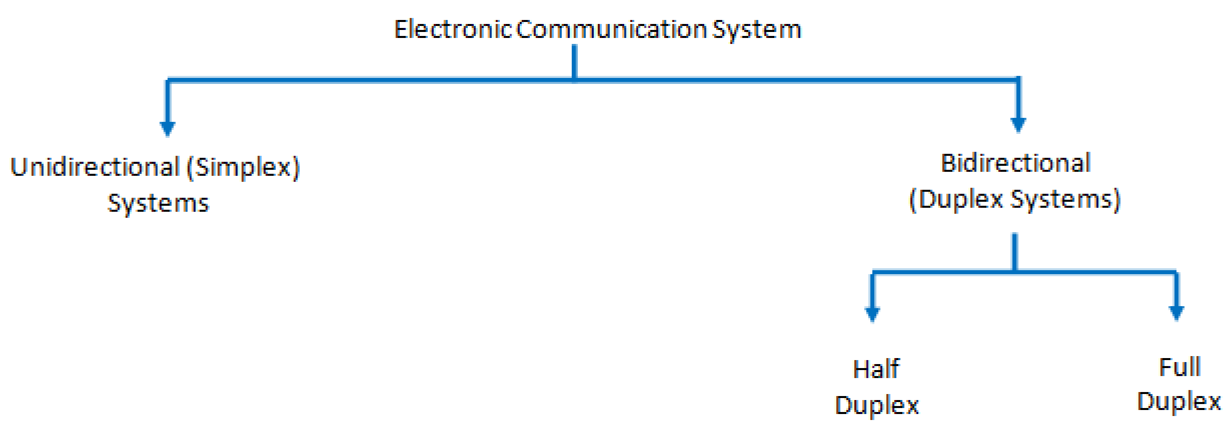


Fig.12. Simplex and Duplex communication

1.4 Fundamental Limitations in Communications

A. Limitations Due to Technological Problems

- Hardware availability
- Economic factors
- International and national regulating norms

B. Fundamental Physical Limitations

Available transmission bandwidth, level of noise generated in the electronic system, atmospheric conditions and maximum capacity of the channel used for communication puts a limit of the rate of information exchange. The impact of each of these factors are given below.

a) Transmission Bandwidth(B)

Limits the spectrum of the transmitted signal, i.e. the maximum speed of variation of the transmitted signal. The time required for transmission of a given amount of information is inversely proportional to the transmission bandwidth B.

b) Noise

Noise is generated in all conductors and in electronic devices as well. Noises generated in electronic systems(Thermal, shot, flicker, popcorn, avalanche noise) degrades the signal quality or fidelity in analog communication systems and produces errors in digital communications. Noise generation limits the weakest transmitted signal. Its impact is significant in long-distance communications when the signal attenuation is large. A measure of noise level is Signal-to-noise ratio (S/N) given by

S/N=(Power of Signal)/(Power of Noise)

***c)* Channel capacity (C)**

Hartley-Shannon law or channel capacity theorem states that the rate of information transmission cannot exceed the channel capacity C.

C=B log(1+S/N)

1.5 Modulation and Demodulation

In the modulation process, some parameter of the carrier wave (such as amplitude, frequency or phase) is varied in accordance with the modulating signal . This modulated signal is then transmitted by the transmitter. The modulating signal is nothing but the baseband signal or information signal while the carrier is a high frequency sinusoidal signal. In the process of modulation the carrier wave actually carries the information signal from the transmitter to receiver .

The receiver demodulates the received modulated signal and gets the original information signal back. Thus, demodulation is exactly opposite to modulation .

Terms to remember:

Modulating signal – *represents the message.*

Carrier wave – *High frequency signal on which message is imposed through modulation process. Usually the modulating signal is much slower than the carrier wave.*

Modulation – *altering one or more of the parameters (amplitude, frequency, phase, pulse width) of the carrier in correspondence with the modulating signal.*

1.6 Need for modulation

You may be ask, when the baseband signal can be transmitted directly why to use the modulation ? The answer is that the baseband transmission has many limitations which can be overcome using modulation . It is explained below.

1.6.1 Advantages of Modulation

The process of modulation provides the following benefits:

- **Reduction in the height of antenna**
- **Avoids mixing of signals**
- **Increases the range of communication**
- **Multiplexing is possible**
- **Improves quality of the signal**

1. Reduction in the height of antenna

For the transmission of radio signals, the antenna height must be multiple of λ/4 ,where λ is the wavelength .

$$\lambda = c / f$$

where c : is the velocity of light

f: is the frequency of the signal to be transmitted

The minimum antenna height required to transmit a baseband signal of f = 10 kHz is calculated as follows :

$$Minimum\ antenna\ height = \frac{\lambda}{4} = \frac{c}{4f} = \frac{3 \times 10^8}{4 \times 10 \times 10^3} = 7500\ meters\ i.e. 7.5\ km$$

The antenna of this height is practically impossible to install .Now, let us consider a modulated signal at carrier frequency of fc = 1 MHz . The minimum antenna height is given by,

$$Minimum\ antenna\ height = \frac{\lambda}{4} = \frac{c}{4f} = \frac{3 \times 10^8}{4 \times 10 \times 10^6} = 75\ meters$$

This antenna can be easily installed practically. **Thus, modulation reduces the height of the antenna.**

2. Avoids mixing of signals

If the baseband sound signals are transmitted without using the modulation by more than one transmitter, then all the signals will be in the same frequency range i.e. 0 to 20 kHz . Therefore, all the signals get mixed together and a receiver cannot separate them from each other. Hence, if each baseband sound signal is used to modulate a different carrier then they will occupy different slots in the frequency domain (different channels). Thus, modulation avoids mixing of signals.

Example : FM stations broadcasting at different carrier frequencies.

3. Increase the Range of Communication

The frequency of baseband signal is low, and the low frequency signals cannot travel long distance when they are transmitted. They get heavily attenuated. The attenuation reduces with increase in frequency of the transmitted signal, and they travel longer distance. The modulation process increases the frequency of the signal to be transmitted. Therefore, it increases the range of communication.

4. Multiplexing is possible

Multiplexing is a process in which two or more signals can be transmitted over the same communication channel simultaneously. This is possible only with modulation. The multiplexing allows the same channel to be used by many signals.

Hence, many TV channels can broadcast simultaneously without getting mixed with each other as they use different carrier frequencies. It is referred to as frequency division multiplexing.

5. Improves Quality of Reception

With frequency modulation (FM) and the digital communication techniques such as PCM, the effect of noise is reduced to a great extent. This improves quality of reception.

1.7 Analog and Digital Communications

In analog communication systems, the message signals are transmitted in analog form itself. AM, FM and PM are common analog modulation schemes which uses sinusoidal carrier signal. In pulse modulation systems such as PAM, PWM and PPM, the carrier signal is a pulse train but the message signal is in analog form. Therefore PAM, PWM and PPM are also called as analog modulation schemes. They are generally not used for wireless communications.

In digital communication systems, the analog information is converted to digital binary data (ones and zeros) using Analog to digital convertor ICs. Then the binary data is modulated with a sinusoidal carrier and transmitted. Amplitude shift keying (ASK), Frequency shift keying (FSK) and Phase shift keying(PSK) are some digital modulation schemes.

1.7.1 Types of modulation

There are various types of modulation techniques used for transmitting information. If the carrier is sinusoidal, then its amplitude, frequency or phase is changed in accordance with the modulating signal to obtain AM, FM or PM respectively. These are continuous wave modulation systems.

Analog modulation can be pulsed modulation as well. Here the carrier is in the form of rectangular pulse. The amplitude, width or position of the carrier pulses is varied in accordance with the instantaneous value of modulating signal to obtain the PAM, PWM or PPM outputs.

Some commonly used analog and digital modulation techniques are outlined below in figure 13, 14 and15. AM, FM, PM, PAM, PWM and PPM are analog modulation schemes. ASK, FSK and PSK are digital modulation schemes.

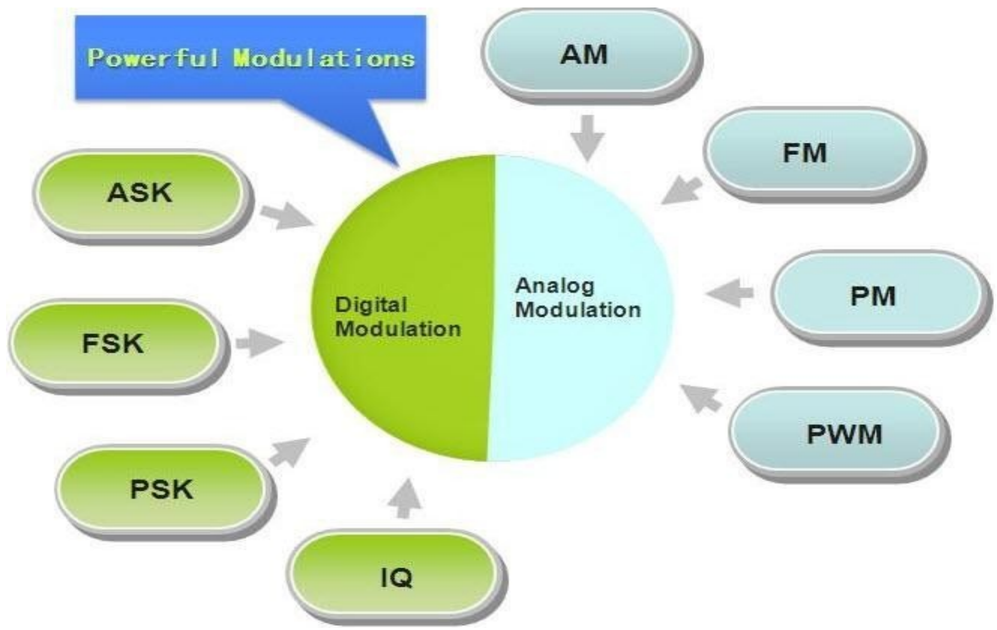


Figure 13: Types of modulation

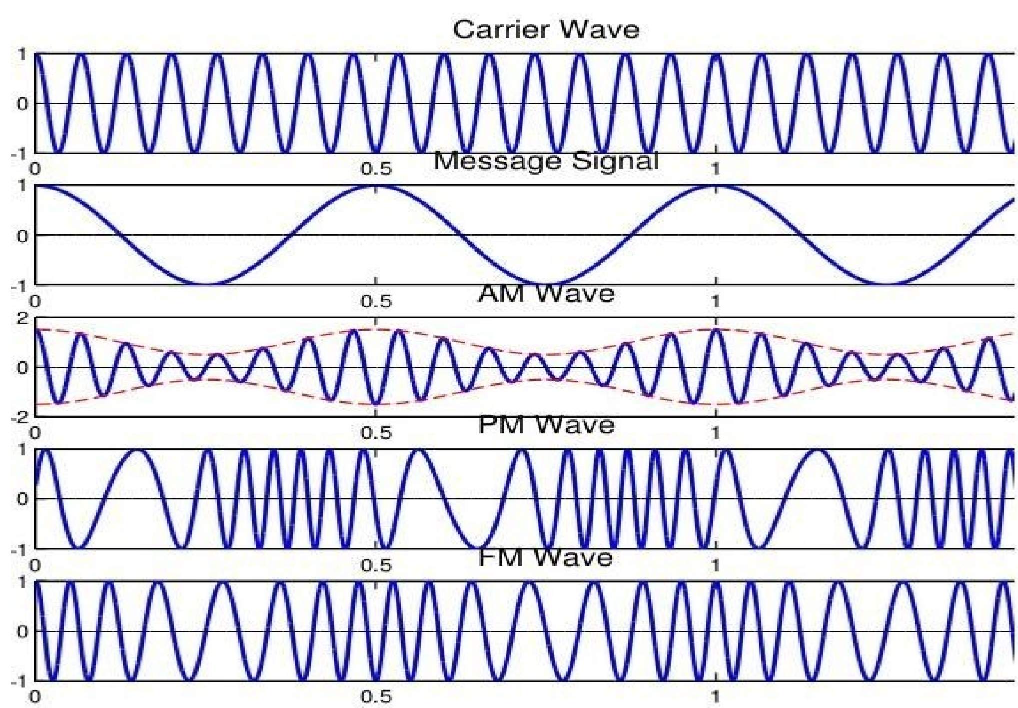


Figure 14.Examples of the basic continuous modulation schemes(AM,PM,FM).

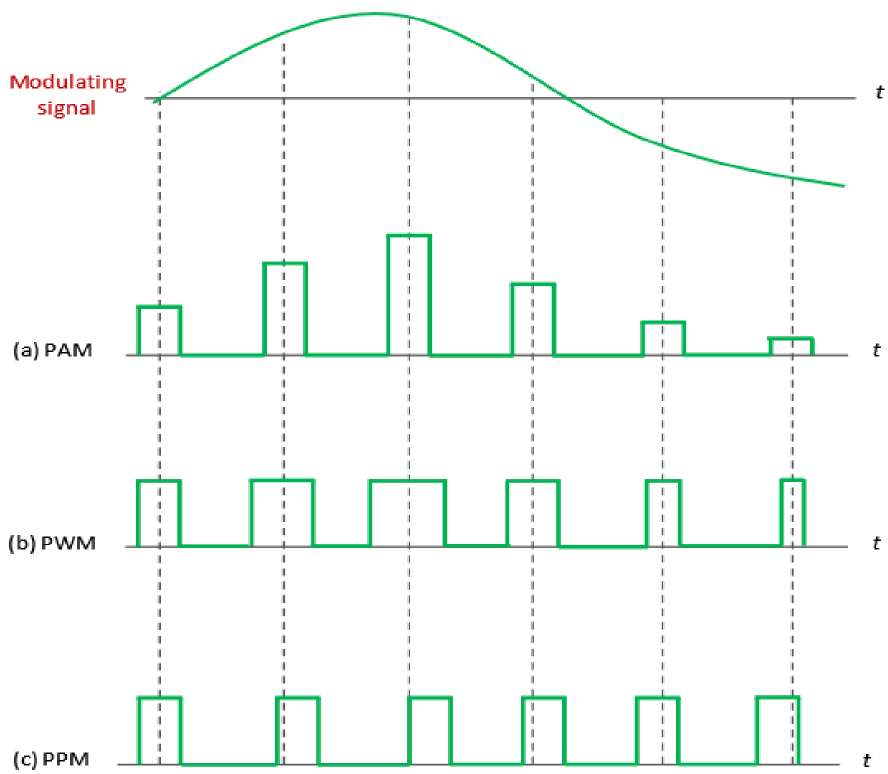


Figure.15. Examples of the basic pulse modulations

Advantages of analog communication

1. Transmitters and receivers are simple
2. Low bandwidth requirement
3. FDM (Frequency division multiplexing) can be

used Drawbacks of analog communication

1. Noise affects the signal quality
2. It is not possible to separate noise and signal
3. Repeaters cannot be used between transmitter and receiver
4. Coding is not possible
5. It is not suitable for the transmission of secret

information Applications

1. Radio broadcasting (AM and FM)
2. TV broadcasting(AM for video and FM for audio)

Advantages of Digital Communication

1. Due to the digital nature of the transmitted signal, the interference of additive noise does not introduce many errors. Hence, digital communication has a better noise immunity.
2. Due to the channel coding techniques used in digital communication, it is possible to detect and correct the errors introduced during the data transmission .
3. Repeaters can be used between transmitter and receiver to regenerate the digital signal. This improves the noise immunity further.
4. Due to the digital nature of the signal, it is possible to use the advanced data processing techniques such as digital signal processing, image processing, data compression etc.
5. TDM (Time Division Multiplexing) technique can be used to transmit many voice channels over a single common transmission channel .
6. Digital communication is useful in military applications where only a few permitted receivers can receive the transmitted signal.
7. Digital communication is becoming simpler and cheaper as compared to the analog communication due to the invention of low cost and high speed computers and integrated circuits (ICs).

Drawbacks of Digital Communication

1. The bit rates of digital systems are high. Therefore, they require a larger channel bandwidth as compared to analog system.
2. Digital modulation needs synchronization.

Applications of Digital Communications

1. Long distance communication between earth and space ships .
2. Satellite communication
3. Military communication
4. Telephone systems
5. Data and computer communications
6. DTH, etc.,

1.8 Classification Based on the Technique of Transmission

Based on the technique used for the signal transmission, we can categorise the electronic communication system as under :

1. Baseband transmission system
2. Communication systems using modulation

1.8.1 Baseband Transmission

In baseband transmission systems, the baseband signals (original information signals) are directly transmitted.

Example of these type of systems are telephone networks where the sound signal converted into the electrical signal is placed directly on the telephone lines for transmission. Another example of baseband transmission is computer data transmission over the coaxial cables in the computer networks.

Thus, the baseband transmission is the transmission of the original information signal as it is.

Limitation of Baseband Transmission

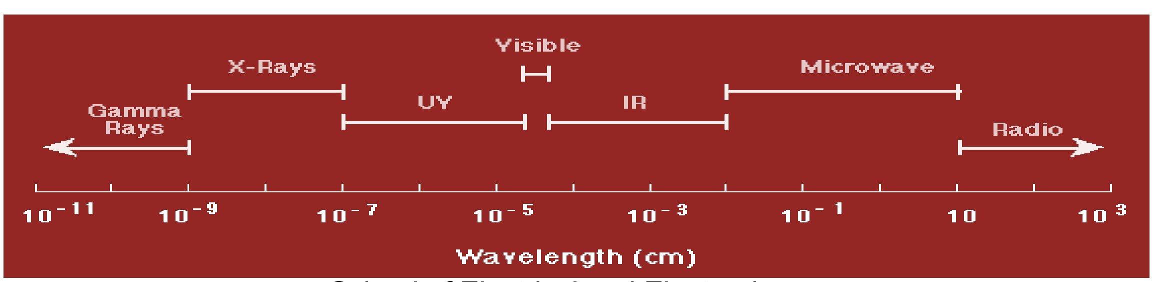
The baseband transmission cannot be used with certain mediums e.g., it cannot be used on the radio transmission where the medium is free space. This is because the voice signal cannot travel long distance in air. It gets suppressed after a short distance Therefore, for the radio communication of baseband signal, a technique called modulation is used.

1.9 Electromagnetic spectrum

The electromagnetic spectrum consists of a group of radiations that all travel at the speed of light. The varieties of electromagnetic radiation form a continuum known as the Electromagnetic Spectrum. Its broad categories are called

- Radio waves
- microwaves
- infrared light
- visible light
- ultraviolet light
- X-rays
- gamma rays

The types of electromagnetic radiation are listed below in decreasing order of wavelength, and hence increasing order of frequency.



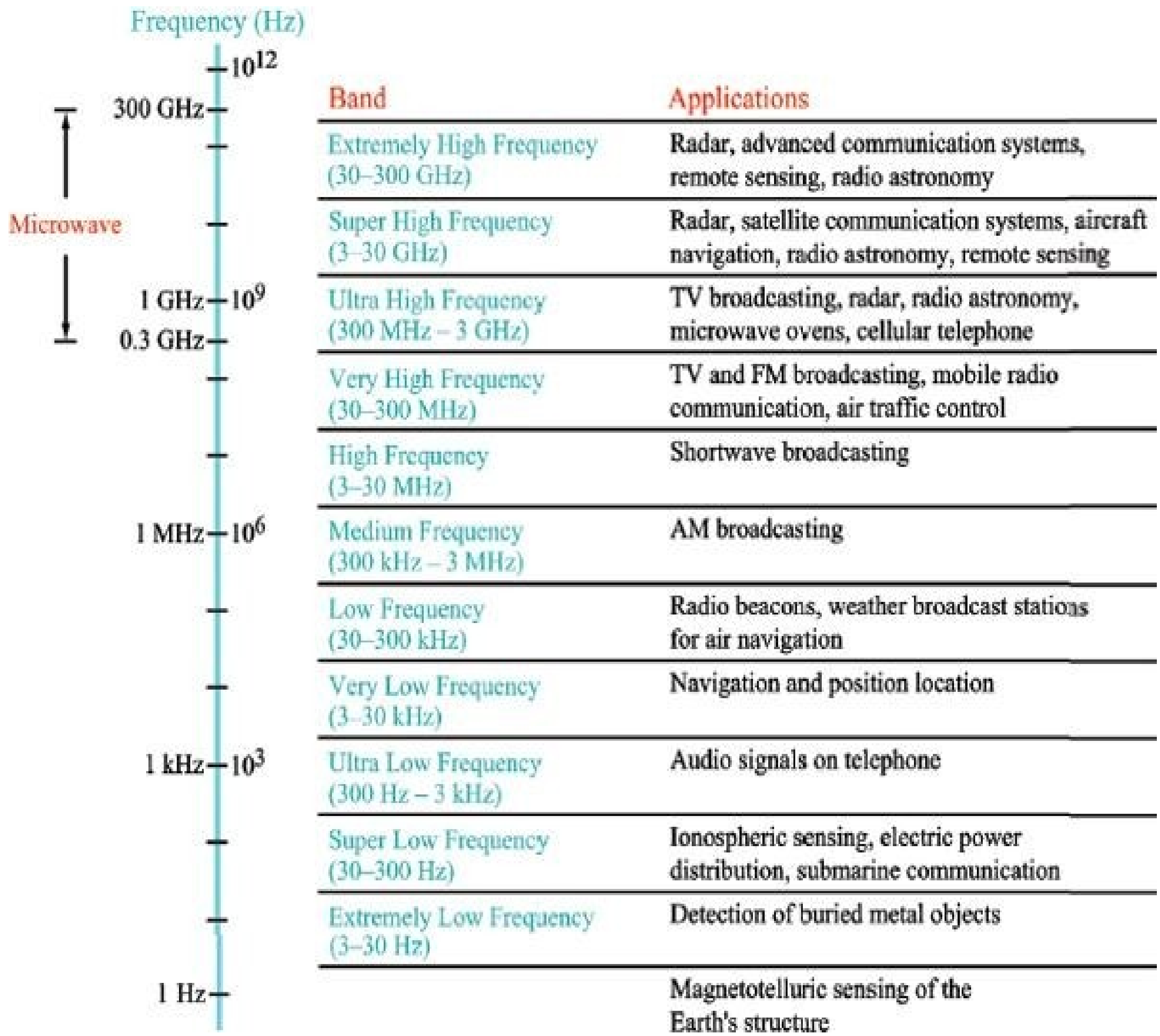


Figure: Classification of radio waves and their applications

a. Very Low Frequency (VLF)

The VLF radio wave has a very long wavelength between 10km and 100km, and it propagates on the ground surface, past small mountains.

b. Low Frequency (LF)

The LF radio wave has a wavelength between 1km and 10km, and it propagates very far. The LF had been in use for radiotelegraphy till around 1930; however, it has not gradually been in use for the purpose as it required a large-scale antenna and transmitting device and the high frequency communication has been largely developed. The LF is partially utilized for the sound broadcasting in Europe, Africa, and some other regions, while in Japan, it is used for Loran C stations for radio navigation, navigation beacons for vessels and aircrafts, and standard frequency and time signal stations providing information on the standard frequency and time signal.

c. Medium Frequency (MF)

The MF radio wave has a wavelength between 100m and 1000m, and it propagates by reflecting on the E layer of the ionosphere formed at the altitude of about 100km. Because of MF dario wave's characteristics ensuring stable propagation in a long distance it is suitable for sound broadcasting. While transmission of MF radio wave requires a large-scale transmitter and antenna, only a simple type of receiver is necessary for its reception.

d. High Frequency (HF)

The HF radio wave has a wavelength between 10m and 100m, and it can travel to the opposite side of the planet by repeating reflecting on the F layer of the ionosphere formed at the altitude of about 200-400km the ground surface. As it enables a long-distance communication, it is utilized for ocean vessel communication, aeronautical communication, international broadcasting, and amateur radio communications.

e. Very High Frequency (VHF)

The VHF radio wave has a wavelength between 1m and 10m, and it propagates straightforwardly not reflecting the ionosphere, while it reaches behind mountains or buildings to an extent. As it can carry more information than the high frequency does, it is utilized for the VHF TV broadcasting, FM broadcasting, or mobile communications.

f. Ultra High Frequency (UHF)

The UHF radio wave has a wavelength between 10cm and 1m, and it has stronger straightforwardness than very short waves does, while it reaches behind small mountains or buildings. It is used for mobile communications as it is suitable for the transmission of a large amount of information with small antennas, transmitters and receivers. It is also used for UHF TV broadcasting.

g. Super High Frequency (SHF)

The SHF radio wave has a wavelength between 1cm and 10cm. Since it propagates straightforwardly, it is suitable for transmission into a specific direction. As it is suitable for transmission of a fairly large amount of information, it is used for fixed links between telephone exchanges, satellite communications, and satellite broadcasting.

Furthermore, it is also used for radars.

h. Extremely High Frequency (EHF)

The EHF radio wave has a very short wavelength between 1mm and 10mm. It has a strong straightforwardness similarly to the light, and it is attenuated by rain or mist in a bad weather, resulting in difficulties in propagating in a long distance. Therefore, it is used for short range radio communications such as the simple radio communication for image transmission or fixed wireless access systems. Furthermore, the development of new types of systems such as vehicle collision prevention radars, radio LANS, etc utilizing this frequency band has been in progress.

i. Sub-millimeter Wave

The sub-millimeter wave has a wavelength between 0.1mm and 1mm, and it has similar characteristics to the light. Currently it is not used for radio communications because its transmission requires large-scale facilities and it is largely absorbed by steam. The sub-millimeter wave is used for scientific studies such as radio astronomy.

NOISE:

Noise can be defined as an unwanted signal that interferes with the communication or measurement of another signal. A noise itself is a signal that conveys information regarding the source of the noise. For example, the noise from a car engine conveys information regarding the state of the engine. The sources of noise are many, and vary from audio frequency acoustic noise emanating from moving, vibrating or colliding sources such as revolving machines, moving vehicles, computer fans, keyboard clicks, wind, rain, etc. to radio-frequency electromagnetic noise that can interfere with the transmission and reception of voice, image and data over the radio-frequency spectrum. Signal distortion is the term often used to describe a systematic undesirable change in a signal and refers to changes in a signal due to the non–ideal characteristics of the transmission channel, reverberations, echo and missing samples. Noise and distortion are the main limiting factors in communication and measurement systems. Therefore the modelling and removal of the effects of noise and distortion have been at the core of the theory and practice of communications and signal processing. Noise reduction and distortion removal are important problems in applications such as cellular mobile communication, speech recognition, image processing, medical signal processing, radar, sonar, and in any application where the signals cannot be isolated from noise and distortion.

Internal Noise:

- It is due to random movement of electrons in electronic circuits
- Major sources are resistors, diodes, transistors etc.
- Thermal noise or Johnson noise and shot noise are

examples External Noise:

- Man- made and natural resource
- Sources over which we have no contro
- Examples are Motors, generators, atmospheric sources.

- Thermal Noise: This noise is generated due to thermal motion (Brownian motion) of electrons inside resistor. This noise is zero at absolute zero degree Kelvin and generated when temperature rises, also called thermal noise. Also called Johnson noise who invented it. Thermal noise also referred as „White noise" since it has uniform spectral density across the EM Spectrum.

  - PSD of thermal noise $Sn\ (f)$ is $Sn\ (f) = kT/2$ (where k is Boltzman's constant and T is temperature) kT is denoted by N0 Then $Sn\ (f) = N0\ /2$
- Work of Johnson and Nyquist gave the expression for noise power $Pn = \overline{vn}^2 = 4kTBR\ volt$

k = Boltzman constant

T = Absolute temp.

(Kelvin) B = Bandwidth

(Hz)

R = Resistance (ohms)

Where $\overline{vn}$ is mean noise voltage.

- SHOT NOISE It is electronic noise that occurs when there are finite number of particles that carry energy such as electrons or photons. Due to analogy of lead shots called shot noise. It has uniform spectral density like thermal noise.

Noise figure is always > 1

- Noise temperature Equivalent noise temperature is not the physical temperature of amplifier, but a theoretical construct, that is an equivalent temperature that produces that amount of noise power $Te = T(F - 1)$

- Noise figure of cascaded stages

  FRIIS formula for calculating total noise factor of several cascaded amplifiers
  $F = F1 + F2 - 1\ /G1 + F3 – 1/\ G1G2 + - - - - -$
  $F1, F2, - -$ & $G1, G2, etc\ are$ Noise figure and gains of different stages in cascade. Note that noise figure is mainly dominated by first two stages.

- Acoustic noise: emanates from moving, vibrating, or colliding sources and is the most familiar type of noise present in various degrees in everyday environments. Acoustic noise is generated by such sources as moving cars, air-conditioners, computer fans, traffic, people talking in the background, wind, rain, etc.
- Electromagnetic noise: present at all frequencies and in particular at the radio frequencies. All electric devices, such as radio and television transmitters and receivers, generate electromagnetic noise.
- Electrostatic noise: generated by the presence of a voltage with or without current flow. Fluorescent lighting is one of the more common sources of electrostatic noise.
- Channel distortions, echo, and fading: due to non-ideal characteristics of communication channels. Radio channels, such as those at microwave frequencies used by cellular mobile phone operators, are particularly sensitive to the propagation characteristics of the channel environment.
- Processing noise: the noise that results from the digital/analog processing of signals, e.g. quantisation noise in digital coding of speech or image signals, or lost data packets in digital data communication systems.

Depending on its frequency or time characteristics, a noise process can be classified into one of several categories as follows: (a) Narrowband noise: a noise process with a narrow bandwidth such as a 50/60 Hz „hum' from the electricity supply. (b) White noise: purely random noise that has a flat power spectrum. White noise theoretically contains all frequencies in equal intensity. (c) Band-limited white noise: a noise with a flat spectrum and a limited bandwidth that usually covers the limited spectrum of the device or the signal of interest. (d) Coloured noise: non-white noise or any wideband noise whose spectrum has a non-flat shape; examples are pink noise, brown noise and autoregressive noise. (e) Impulsive noise: consists of short-duration pulses of random amplitude and random duration. (f) Transient noise pulses: consists of relatively long duration noise pulses.

Transient Noise Pulses

Transient noise pulses often consist of a relatively short sharp initial pulse followed by decaying low-frequency oscillations The initial pulse is usually due to some external or internal impulsive interference, whereas the oscillations are often due to the resonance of the communication channel excited by the initial pulse, and may be considered as the response of the channel to the initial pulse. In a telecommunication system, a noise pulse originates at some point in time and space, and then propagates through the channel to the receiver. The noise pulse is shaped by the channel characteristics, and may be considered as the channel pulse response. Thus we should be able to characterize the transient noise pulses with a similar degree of consistency as in characterizing the channels through which the pulses propagate.The initial high-amplitude pulse response of the playback system to the physical discontinuity on the record medium, followed by; (b) decaying oscillations that cause additive distortion. The initial pulse is relatively short and has a duration on the order of 1–5 ms, whereas the oscillatory tail has a longer duration and may last up to 50 ms or more.

Thermal Noise

Thermal noise, also referred to as Johnson noise (after its discoverer J. B. Johnson), is generated by the random movements of thermally energised particles. The concept of thermal noise has its roots in thermodynamics and is associated with the temperature-dependent random movements of free particles such as gas molecules in a container or electrons in a conductor. Although these random particle movements average to zero, the fluctuations about the average constitute the thermal noise. For example, the random movements and collisions of gas molecules in a confined space produce random fluctuations about the average pressure. As the temperature increases, the kinetic energy of the molecules and the thermal noise increase. Similarly, an electrical conductor contains a very large number of free electrons, together with ions that vibrate randomly about their equilibrium positions and resist the movement of the electrons. The free movement of electrons constitutes random spontaneous currents, or thermal noise, that average to zero since in the absent of a voltage electrons move in all different directions. As the temperature of a conductor, provided by its surroundings, increases, the electrons move to higher-energy states and the random current flow increases.

Shot Noise

The term shot noise arose from the analysis of random variations in the emission of electrons from the cathode of a vacuum tube. Discrete electron particles in a current flow arrive at random times, and therefore there will be fluctuations about the average particle flow. The fluctuations in the rate of particle flow constitutes the shot noise. Other instances of shot noise are the flow of photons in a laser beam, the flow and recombination of electrons and holes in semiconductors, and the flow of photoelectrons emitted in photodiodes. The concept of randomness of the rate of emission or arrival of particles implies that shot noise can be modelled by a Poisson distribution. When the average number of arrivals during the observing time is large, the fluctuations will approach a Gaussian distribution. Note that whereas thermal noise is due to “unforced” random movement of particles, shot noise happens in a forced directional flow of particles.

Electromagnetic Noise

Virtually every electrical device that generates, consumes or transmits power is a potential source of electromagnetic noise and interference for other systems. In general, the higher the voltage or the current level, and the closer the proximity of electrical circuits/devices, the greater will be the induced noise. The common sources of electromagnetic noise are transformers, radio and television transmitters, mobile phones, microwave transmitters, ac power lines, motors and motor starters, generators, relays, oscillators, fluorescent lamps, and electrical storms. Channel Distortions 39 Electrical noise from these sources can be categorized into two basic types: electrostatic and magnetic. These two types of noise are fundamentally different, and thus require different noise-shielding measures. Unfortunately, most of the common noise sources listed above produce combinations of the two noise types, which can complicate the noise reduction problem. Electrostatic fields are generated by the presence of voltage, with or without current flow. Fluorescent lighting is one of the more common sources of electrostatic noise. Magnetic fields are created either by the flow of electric current or by the presence of permanent magnetism. Motors and transformers are examples of the former, and the Earth's magnetic field is an instance of the latter. In order for noise voltage to be developed in a conductor, magnetic lines of flux must be cut by the conductor. Electric generators function on this basic principle. In the presence of an alternating field, such as that surrounding a 50/60 Hz power line, voltage will be induced into any stationary conductor as the magnetic field expands and collapses. Similarly, a conductor moving through the Earth's magnetic field has a noise voltage generated in it as it cuts the lines of flux.

16. NOISE FIGURE AND NOISE FACTOR

* Noise factor (F) and noise figure NF are *figures of merit* used to indicate how much the signal to noise ratio deteriorates as a signal passes through a circuit.

Noise factor can be defined as the ratio of the available SNR power supplied to the input terminal of a receiver or amplifier to the available SNR power at the output or load resistor. It can be denoted by

$$F = \frac{\text{Available SNR power at input}}{\text{Available SNR power at output}} = \frac{(SNR)_i}{(SNR)_o}$$

$$F = \frac{(P_{Si} / P_{ni})}{(P_{SO} / P_{nO})} = \frac{P_{Si}}{P_{ni}} \times \frac{P_{nO}}{P_{SO}} \quad (35)$$

Where

P_{si} = Signal power at input; P_{ni} = Noise power at input

P_{so} = Signal power at output P_{no} = Noise power at output

For comparing the performance of radio receivers or amplifiers working at different impedance levels the use of equivalent noise resistance is very difficult. In that case noise factor is used to calculate the noise level present in its output.

Available power from a source is the maximum average power that a source can deliver and is obtained under matched conditions.

The available input power may not actually be delivered to the input of the amplifier because of mismatch, but the available output power depends on the actual input power and the mismatch is taken into account.

We know that the available signal power gain $G_a = \frac{P_{SO}}{P_{Si}}$

Hence the noise factor $F = \frac{P_{Si}}{P_{ni}} \times \frac{P_{nO}}{P_{SO}} = \frac{P_{nO}}{G_a P_{ni}}$ (36)

or $P_{nO} = FG_aP_{ni} = FG_aKTB$

Noise figure in db $NF = 10 \log_{10} F$

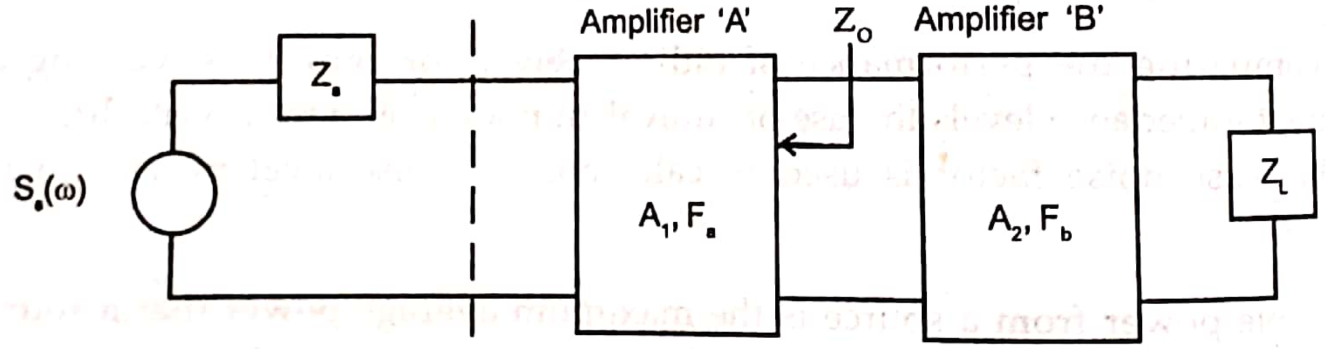
Noise figure is unity for noiseless ideal receiver which introduces no noise of its own.

17. NOISE FIGURE OF CASCADED STAGES OR FRII'S FORMULA

* When the amplifier consists of more than one stage, its noise figure can be computed in terms of the noise figures of individual stages. It is intuitively obtained that the noise generated in earlier stages is amplified by later stages, and hence the noise figures of the first stage is much more significant in determining the overall noise figures of the amplifier.

Consider a two stage amplifier as shown in figure 12. Two amplifiers have power gains A_1 and A_2 respectively in order to determine the overall noise figure (F_{ab}) of the cascaded amplifier.

We will first determine (S_{no}) (i.e.) the available total noise power at the output terminals. This consists of two components, S_1 the total noise power density available at the output due to first stage noise. The component S_1 obviously A_2 times the total noise power available at the output of the second stage, but the noise power available at the input of the second stage is the total noise power available at the output of the first stage.



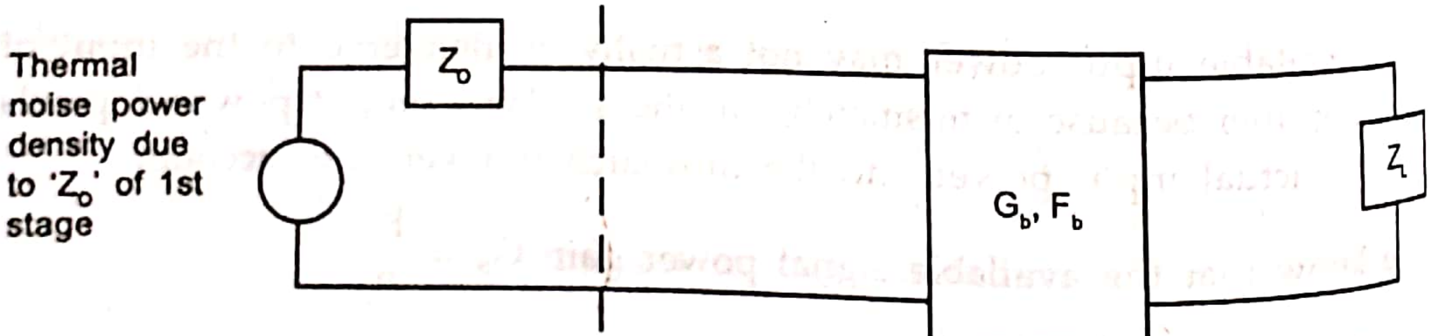


Figure 12 : Noise figure of cascaded stages

The available noise power spectral density from any two terminal networkcan be written as

$$S_{ni} = \frac{KT}{2} = P_{ni}$$

we know the noise factor

$$F = \frac{P_{nO}}{GP_{ni}} = \frac{2P_{nO}}{GKT}$$

$$P_{no} = \frac{FGKT}{2} = S_1$$

$$S_1 = \frac{F_a A_1 KT}{2}$$

where G = gain of the amplifier = A_1.

Let S_2 is the noise power available at the output due to the second stage alone. The source impedance for the second stage is the impedance seen at the output terminals of the first stage i.e., Z_O.

Let the noise figure of amplifier B is F_b. Here we assume source impedance Z_O is thermal impedance thus its power density is KT/2. The available noise power density at the output due to amplifier B alone is S_2 and is given by

$$S_2 = \frac{F_b A_2 KT}{2} - \frac{A_2 KT}{2} \tag{38}$$

Where, $A_2 \frac{KT}{2}$ = noise power density due to second amplifier.

$\frac{F_b A_2 KT}{2}$ = total noise power density at the output of second stage.

$$S_2 = \frac{(F_b - 1) A_2 KT}{2} \tag{39}$$

and $S_2{}^1$ = noise power due to amplifier 1 which is amplified by amplifier 2

$$= \frac{F_a KT A_1 A_2}{2}$$

Hence, the total noise power at the output can be written as

$$(S_{no})_{av} = S_{21} + S_2$$

$$= \frac{KT}{2}\left[(F_b - 1) A_2 + F_a A_1 A_2\right] \tag{40}$$

$$= \frac{F_{ab} A_{ab} KT}{2} \quad (41)$$

Where A_{ab} is the available gain of the cascaded amplifier. It can be easily seen from the definition of the available gain that the gain of cascaded stages is equal to the product of the gain of individual stages.

$$A_{ab} = A_1 A_2$$

where
$$F_{ab} = F_a + \frac{F_b - 1}{A_1}$$

In general for multistage amplifiers,

$$F = F_a + \frac{F_b - 1}{A_1} + \frac{F_C - 1}{A_1 A_2} + \cdots \quad (44)$$

The equation 44 is known as *Friss formula*.

It is evident that the first stage is the most significant in determining the noise figure of an amplifier. Hence in low noise amplifier the primary consideration in design of the first stage is to obtain a low noise figure even at the cost of gain.

18. NOISE FIGURE OF AN AMPLIFIER

* When a signal is being processed through any system, additional noise is being added. The ratio of signal power to noise power is a good indication of the purity of the signal.

  When a signal is amplified, additional noise generated in the amplifier is being added to the original noise in the signal. This causes deterioration in the S/N of output compared to that output of noiseless amplifier.

* In any amplifier the noise generated in the source is amplified and delivered to load. Additional noise is generated inside the amplifier and also delivered to the load. Thus the noise at the output is contributed by the source as well as the amplifier.

* *Noise figure* of amplifier is defined as the ratio of the total noise density in the load to the noise power density delivered to the load due to the source.

19. NOISE TEMPERATURE

Noise temperature of a system is defined as the temperature at which a noisy resistor, has to be maintained such that by connecting this resistor to the input of the noiseless system, it produces the same noise power at the output of the system, as is produced by all the sources of noise in the actual system.

We know the available noise power is directly proportional to temperature and is independent of resistor value. This provides a convenient way of expressing available noise power from different noise sources. The power specified in terms of temperature is called as ***noise temperature.***

$$T_n = \frac{P_n}{KB} \quad \text{where } P_n = \text{maximum noise power}$$

The noise temperature is referef at the input of a two part network which accounts the internal noise produced by the network and thereafter the network is considered to be noise free. Let us consider a two port network shown in figure 15.

The input noise of the network is n_{si} and its power density $\frac{KT}{2}$. If the available power gain of the network is G_a then available power gain at the output of the network is.

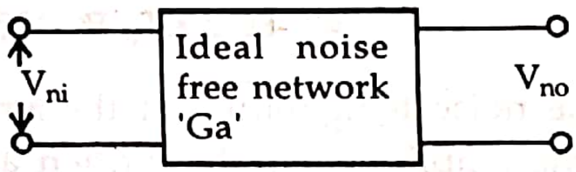


Figure 15 : Network for messaging noise temperature

$$S_{no} = G_a\left(\frac{KT}{2}\right) \tag{1}$$

However, the practical two port network introduces its own noise hence the available power density at the output of the noisy network in higher than the noise free network. This increases in noise power density may be accounted interms effective noise temperature (T_e) at the input then the network may be considered as noise free. Thus the total output power density is given as

$$S_{no} = G_a\frac{KT}{2} + G_a\frac{KT_e}{2} = (T+T_e)\frac{KG_a}{2} \tag{2}$$

where T = temperature of the source

T_e = temperature of network generated noise we know the figure of merit (from eqn. 36)

$$f = \frac{P_{no}}{G_a P_{ni}} = \frac{S_{no}}{G_a S_{ni}} = \frac{S_{no}}{G_a \frac{KT}{2}} \tag{3}$$

now compare the equation (2) and (3)

$$\frac{S_{no}}{K\frac{G_a}{2}} = F.T = (T + T_e)$$

or $$F = \frac{T+T_e}{T} = 1+\frac{T_e}{T} \quad \text{or} \quad (f-1) = \frac{T_e}{T}$$

The input noise power of amplifier or network is given by

$$P_{IN} = KT_e B = K(F-1)\,T.B \tag{4}$$

The output noise power $P_O = FG_a P_{IN} = FG_a KT_e B$ (5)

The noise generated with in the amplifier = $P_O - G_a P_{IN}$

$$= FG_a KT_e B - KT_e BG_a = KT_e BG_a (F-1) \tag{6}$$

If the noise temperature of the amplifier is T_a then noise power generated by the amplifier may be written as

Noise power generated in the amplifier = KT_aA_pB (7)

Equating the equations (52) and (53) we get

$$KT_aA_pB = KTBA_p(F-1)$$

$$\therefore \quad \frac{T_a}{T} = (F-1) \tag{8}$$

According to Frii's formula $F = F_1 + \frac{F_2-1}{G_1} + \frac{F_3-1}{G_1G_2} + \ldots\ldots$

Subtracting by '1' on both sides, we get,

$$F-1 = (F_1-1)+\frac{F_2-1}{G_1}+\frac{F_3-1}{G_1G_2}+\ldots\ldots \tag{9}$$

Since $\frac{T_a}{T} = F-1$ and $\frac{T_{a1}}{T} = F_1 - 1$ and so on.

Hence the equation (55) can be rewritten as,

$$\frac{T_e}{T} = (F - 1) = \frac{T_{a1}}{T} + \frac{T_{a2}}{TG_1} + \frac{T_{a3}}{TG_1G_2} + \ldots\ldots$$

Therefore effective noise temperature of cascaded system is given by

$$\boxed{T_e = T_{a1} + \frac{T_{a2}}{G_1} + \frac{T_{a3}}{G_1G_2} + \ldots\ldots} \qquad (10)$$

20. EFFECTIVE OR EQUIVALENT NOISE RESISTANCE

The equivalent noise resistance is the input resistance that will produce the same amount of noise at the output as the actual amplifier. Thus the actual amplifier may be replaced by a noiseless amplifier with an equivalent noise resistance at its input. This resistance will produce the same amount of source at the output as does the actual amplifier.

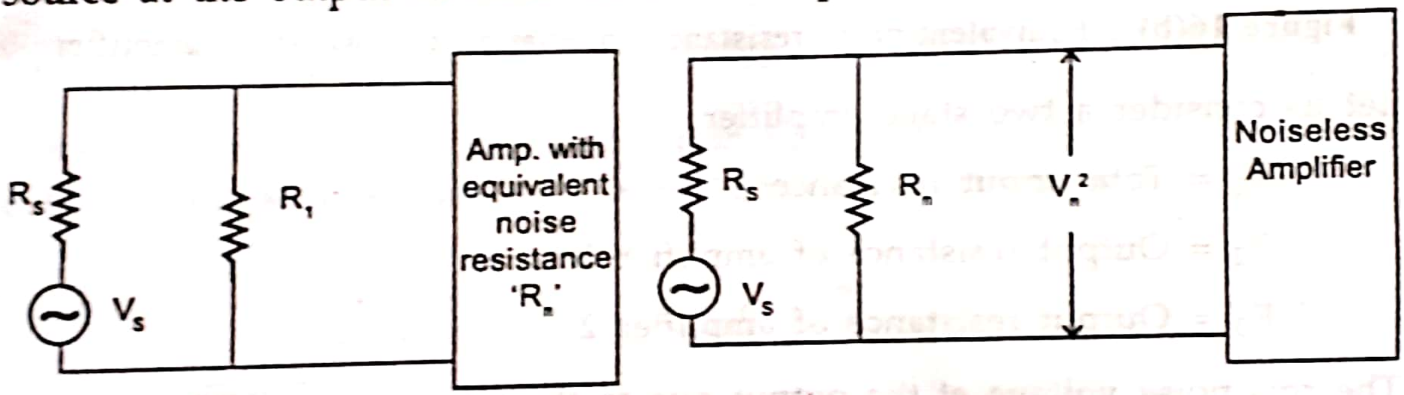


Figure 16(a) : Equivalent noise resistance

It is often convenient to represent the noise which originates in device by means of a fictitious resistance R_n assumed to generate the noise at room temperature, the actual device then being assumed to be noiseless refer figure 16.

An amplifier may have a specified noise resistance R_n and an actual input resistance R_1, hence the equivalent mean square noise voltage at the input is given by

$$V_n^2 = 4(R_i + R_n)\, KT_O\, B$$

The main advantage of using R_n is that it enables the amplifier noise contribution to be compared directly to the input circuit contribution. It is necessary to understand the concept of fictitious R_n which does not affect the real input resistance.

22. NOISE FIGURE FROM EQUIVALENT NOISE RESISTANCE

In order to correlate noise figure 'F' and equivalent noise resistance R_{eq}' it is thus convenient to define R_{eq} which is a noise resistance that does not incorporate input resistance R_1 and which is given by

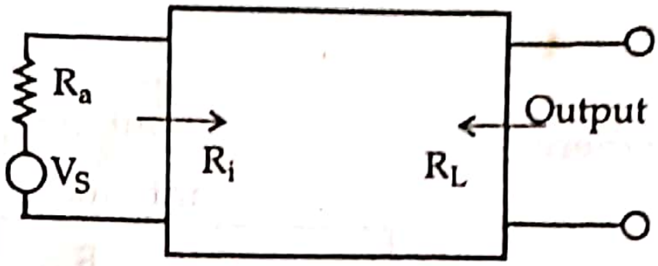


Figure 18 : Equivalent noise resistance

$$R_{eq}' = R - R_i' \tag{1}$$

The total equivalent noise resistance for this receiver will be

$$R = R_{eq}' + \frac{R_a R_i}{R_a + R_i} \tag{2}$$

Hence the noise voltage at the input of receiver will be

$$V_{ni} = \sqrt{4KTBR} \tag{3}$$

We know that $P_{no} = \dfrac{V_{no}^{\ 2}}{R_L} = \dfrac{(AV_{ni})^2}{R_L} = \dfrac{A^2 4KTBR}{R_L}$ (4)

and $P_{ni} = \dfrac{4KTB A^2 R_a}{(R_i + R_a)}$ (5)

We know the general expression noise figure

$$F = \frac{R_L P_{no}(R_a + R_i)}{4KTB A^2 R_a R_i} = \frac{R_L(R_a + R_i)}{4KTBA^2 R_a R_i} \cdot \frac{A^2 4KTBR}{R_L}$$

$$= R\left(\frac{R_a + R_i}{R_a R_i}\right) = \left(R'_{eq} + \frac{R_a R_i}{R_a + R_i}\right)\left(\frac{R_a + R_i}{R_a R_i}\right) \tag{6}$$

$$F = 1 + R_{eq}'\left(\frac{R_a + R_i}{R_a R_i}\right) \tag{7}$$

This is the required expression for the noise figure interms of equivalent noise resistance. From the equation (6), it is observed that noise figure is minimum for given value of R_a if the ratio $\left(\dfrac{R_a + R_i}{R_i}\right)$ is minimum. So that R_i

24. NOISE EQUIVALENT BANDWIDTH

Equivalent noise bandwidth of the ideal bandpass system which produces the same noise power as actual system.

When a source of white noise with zero mean and power spectral density $N_0/2$ is connected to the input of ideal low pass filter with bandwidth 'B'

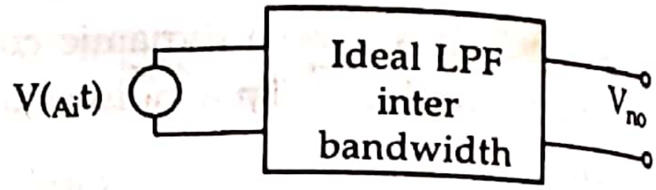


Figure 20(a)

The average output noise power = N_0B.

We know the 3dB bandwidth or cutoff frequency of low pass filter $B = \frac{1}{2\pi RC}$.

Thus the average output noise power of the filter is proportional to the bandwidth.

Generation of the above statement for all kinds of low pass filter with transfer function H(f) is obtained as follows:

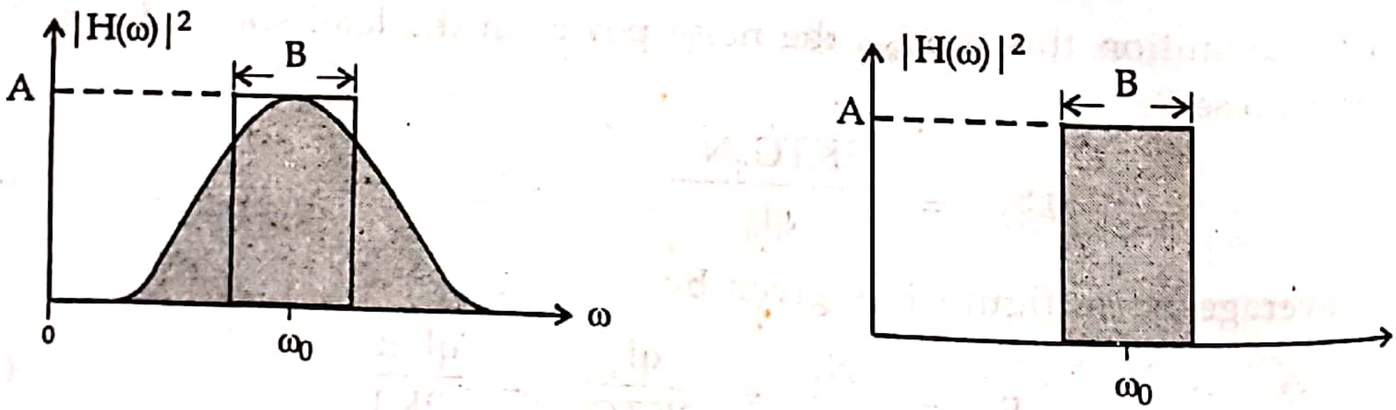


Figure 20(b) & (c) : Frequency response of actual ideal LPF

The noise power at the output of the system is

$$P_o = v_{no}^2 = \frac{1}{2\pi}\int_{-\infty}^{\infty} S_{ni}(\omega)|H(\omega)|^2\, d\omega = \frac{1}{\pi}\int_{0}^{\infty} S_{ni}(\omega)|H(\omega)|^2\, d\omega \quad (1)$$

Assume $S_{ni}(\omega) = K$.

then $$P_o = \frac{K}{\pi}\int_{0}^{\infty} |H(\omega)|^2\, d\omega \quad (2)$$

$$R(\tau) = \frac{1}{2\pi}\int_{-\infty}^{\infty} F(\omega)\,F^*(\omega)\; e^{j\omega\tau}\,.d\omega = \frac{1}{2\pi}\int_{-\infty}^{\infty} F(\omega)^2\; e^{j\omega\tau}\,.d\omega$$

$$= F^{-1}\{|F(\omega)^2| = F^{-1}[\Psi(\omega)]$$

equivalently $F[R(\tau)] = \Psi(\omega)$

Hence the autocorrelation and the energy density spectrum are F.T. pair.

13. INTRODUCTION TO NOISE

- *Noise* may be defined as an unwanted from of electrical signal which interfere with proper (and easy) reception and reproduction of desired signals. Noise may be picked or added up by a signal during its transmission from a transmitter to a receiver, which is commonly termed as *External noise* it also known as additive noise, because it directly added with the original signal.

- Noise may also be produced within a receiving equipment while it is receiving a signal, which is termed as *Internal noise*. For example, in radio receiver, noise may produce hiss sound in loud speaker output while in, pulse communication systems, noise may produce unwanted pulses which could lead to serious error in the detected signal. The classification of noise is shown in figure 3 broadly it can be devided into two types such as correlated noise, and uncorrelated noise.

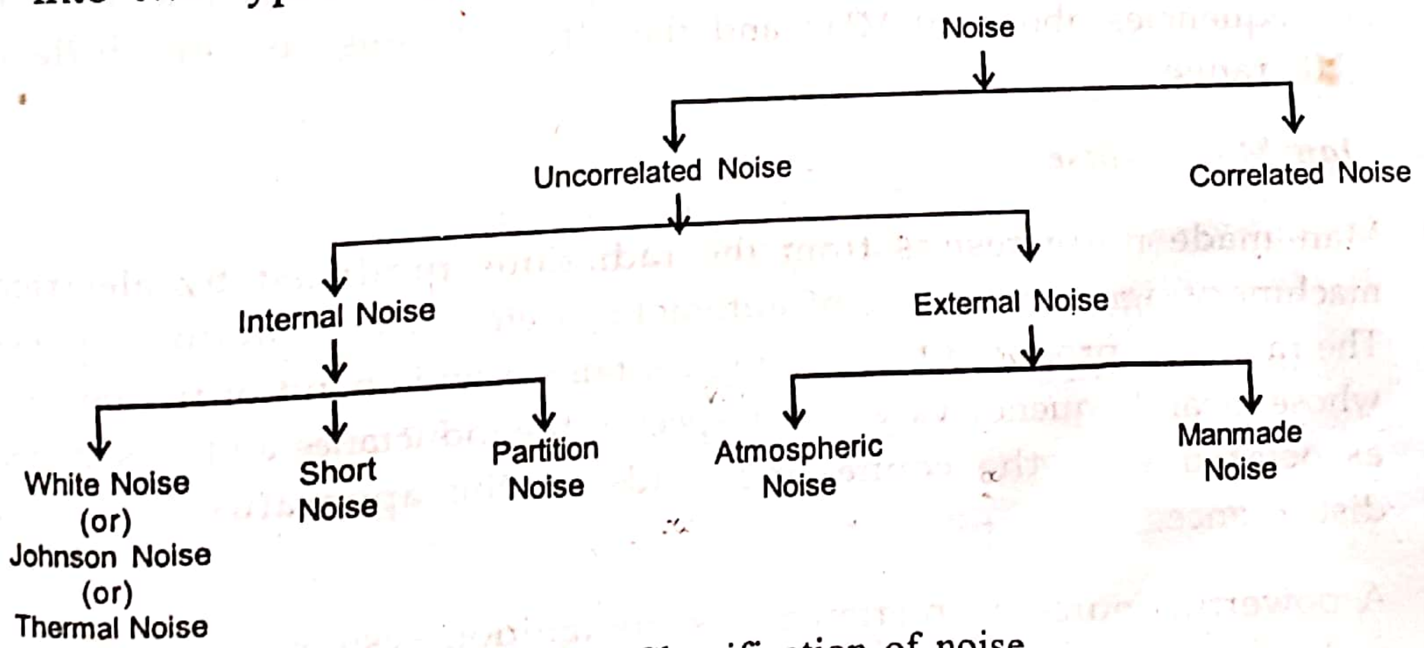


Figure 3 : Classification of noise

- Correlated noise is a noise that is correlated to the signal and cannot be present in a circuit unless there is an input signal. Correlated noise is produced by non linear amplification and includes harmonic, intermodulation distortion, non linear distortion is also produced when signals pass through non linear devices such as diodes.

- Un correlated noise is present regardless of whether there is a signal present or not, it can be divided into two types such as i) External noise ii) Internal noise.

14. EXTERNAL NOISE

The external noise may be classified as

a) Atmospheric noise and b) Man made noise

a) Atmospheric Noise

- This noise is produced due to several electrical disturbances occurring in the atmosphere and lightning discharges in thunderstorms. These occur in the form of impulses and they spread over the entire radio frequency spectrum. These disturbances are also called as *static noise* and the actual amount of noise added depends on the bandwidth of a receiver. Thus these noises would be avoided if the bandwidth of the receiver is limited.

- The field strength of this noise signal is approximately inversely proportional to the frequency. Thus atmospheric noise become less severe at frequencies above 30 MHz and the effect of noise is very little in VHF range.

b) Man Made Noise

- Man made noise results from the radiations produced by electrical machinery, ignition system of automobiles etc., which produce sparks. The radiation produced by sparking contains a wide band of frequencies, whose mean frequency vary with respect to the inductance and capacitance associated with the connecting leads of the apparatus providing disturbance.

- A powerful source of interference is the ignition system of automobiles and aircrafts. This could be avoided by shielding the ignition leads of engines and spark plugs.

* Interference due to sparking in DC motors can be reduced by fitting a capacitor or choke-capacitor combination of a sufficiently low HF resistance across the motor terminals that damp these radiations to a negligible level.

* Interference may also be caused by insulators used on power transmission lines, when such insulators are dirty or coated with salts, leakage currents will flow through it. These are unsteady currents and probably travel by small arcs between deposited particles.

* RF components in these currents are radiated by the line conductors acting as aerials causing disturbances in communication services. Similarly interference may reach a receiver in the following ways.

  i) Direct radiations from noise source to receiver.

  ii) Due to RF current in the main lines and may reach either directly or through inductive or capacitive coupling to receiver aerials.

  iii) Interference may be caused by stray coupling between the two supply systems.

* These interferences can be reduced by placing receiver aerials in a high, open position and connecting them to receivers through shielded cables. Large ships contain a heavy number of electrical machines and it becomes necessary to use radio transmission while signals are received at other frequency and it becomes necessary to shield the receiving room completely.

15. INTERNAL NOISE

* This noise is created by various components used in processing the received signal and is completely internal to the system. This can be reduced by proper designing of the system. These would be classified into three types namely

  (a) Thermal Noise (b) Partition noise and (c) Shot noise.

i) *Thermal Noise*

* The noise generated in a resistance or any resistive component is called as *thermal noise.* This noise is due to the rapid and random motion of the molecules and electrons. The free electrons in a conductor are in continuous motion with a velocity that depends on the temperature of the conductor. The motion of each free electron contributes a minute current, but over a long time interval, the sum of total current is zero.

* At any particular instant there may be a current either in one direction or in the other direction. This flow of charge creates a random voltage at the terminals of the conductor and the effect is called as Thermal ***agitation noise.***

* Noise power generated by a resistor is proportional to the absolute temperature and bandwidth over which noise is to be measure. Thus

$$P_n \propto TB \quad \text{or} \quad P_n = KTB \qquad (1)$$

Where K = Constant of proportionality or Boltzmann's constant = 1.38 x 10^{-23} J/K.

* It may be mentioned that all formulae referring to random noise are applicable only to rms value of such noise, but not to its instantaneous value which is quite unpredictable. The equivalent circuit of a resistor as a noise generator is shown in figure 4, from which noise voltage may be calculated as follows:

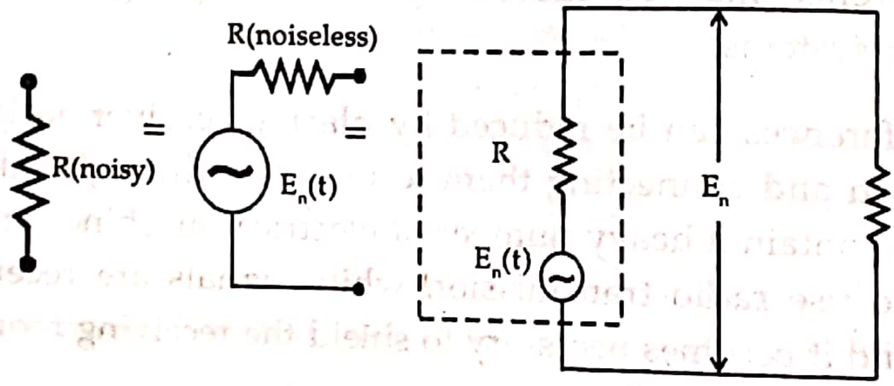


Figure 4 : Thevenin's Equivalent circuit for noise source

* Consider R_L is noiseless resistor, receiving the maximum noise power generated by 'R' and as per the condition of maximum power transfer theorem R_L = R.

We know that the noise power

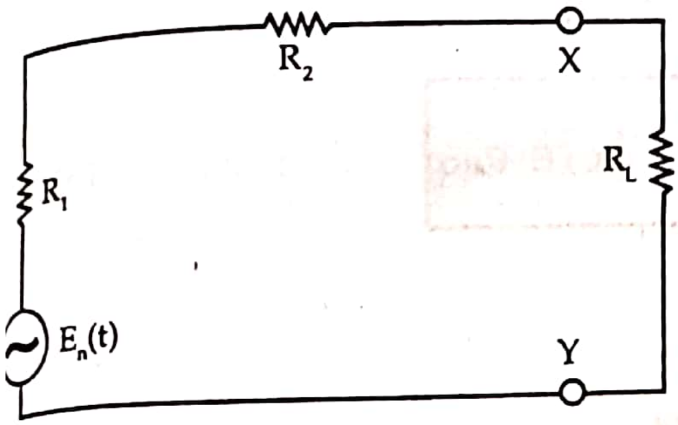
$$P_n = \frac{E_{rms}^2}{(R_L + R)} = \frac{E_{rms}^2}{2R} = \frac{(E_n/\sqrt{2})^2}{2R} = \frac{E_n^2}{4R}$$

$$E_n^2 = 4RP_n = 4RKTB$$

$$E_n(t) = \sqrt{4KTBR} = \text{Equivalent noise voltage} \qquad (2)$$

Where E_n = Peak or maximum value of noise voltage.

t is quite independent of the frequency at which it is measured.



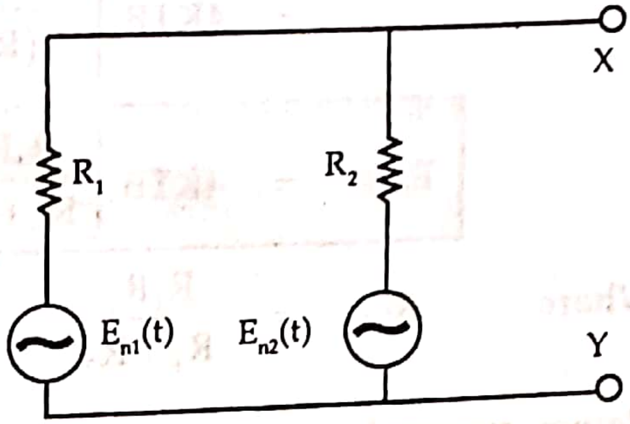


Figure 5 : Noise resistors in series **Figure 6 :** Noise resistors in parallel

- If two resistors R_1 & R_2 are connected in series as shown in figure 5, then the total mean square noise voltage is (3)

$$E_n^2 = 4KTB(R_1 + R_2)$$

This is for the resistors at the same temperature.

If the temperature are T_1 & T_2 then (4)

$$E_n^2 = 4KTB(T_1R_1 + T_2R_2)$$

- If two resistors are at the same temperature T_1 and are connected in parallel, as shown in figure 6 then the random noise voltage act independently of each other.
- The total mean square value across common terminals is the sum of the mean square voltages developed across it by each generator when other is short circuited.
- The output noise voltage at x – y is then given by using potential division rule,

$$E_n(t) = 4KTBR_1\left[\frac{R_2}{R_1+R_2}\right]^2 + 4KTBR_2\left[\frac{R_1}{R_1+R_2}\right]^2$$

$$= 4KTB\left[\frac{R_1R_2^2}{(R_1+R_2)^2} + \frac{R_2R_1^2}{(R_1+R_2)^2}\right]$$

$$= 4KTB\left[\frac{R_1R_2(R_1+R_2)}{(R_1+R_2)^2}\right]$$

$$\boxed{E_n(t) = 4KTB\left[\frac{R_1R_2}{R_1+R_2}\right] = 4KTB\ Req} \quad (5)$$

Where $\quad Req = \dfrac{R_1R_2}{R_1+R_2}$

Power spectral density of thermal noise

The power spectral density of the noise current due to the free electrons is given by

$$S_i(\omega) = \left[\frac{2KTG\alpha^2}{\alpha^2+\omega^2}\right] = \frac{2KTG}{1+\left(\dfrac{\omega}{\alpha}\right)^2} \quad (6)$$

Where K is the Boltzmann's constant

α is the average number of collision / second.

T is the ambient temperature in degrees kelvin

G is the conductance of the conducting medium

A plot of this power density spectrum is shown in the figure 7. The spectrum may be considered as flat for $\omega/\propto < 0.1$, thus if we assume $\omega/\propto = 0.01$ then the power density spectrum may be considered as constant.

Then $S_i(\omega) = 2\ KTG \quad (7)$

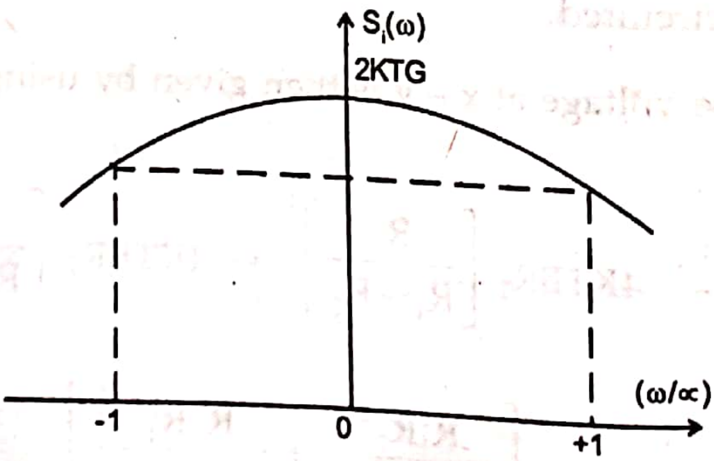


Figure 7 : Power density spectrum of thermal agitation noise

* The contribution of thermal noise in any circuit is limited only by the bandwidth of the circuit. The thermal noise has a power density spectrum which is constant at all frequencies, hence it is also called as ***white noise***, it means that frequencies corresponding to all colours of the light spectrum are present in equal amount in the thermal noise. It is also called ***Johnson noise*** after. J.B. Johnson who invested the phenomenon of thermal noise.

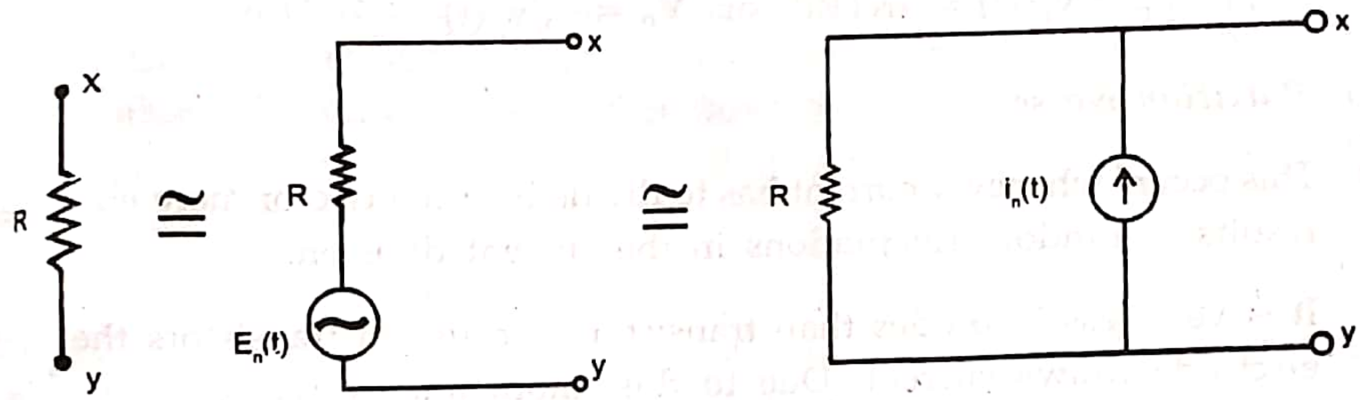


Figure 8(a) : Resistor R (b) Thevenin's voltage source (c) Norton's noise current source

In general all the resistors act as a source of thermal noise and may be represented in terms of Thevenin's voltage source or Norton's current source as shown in figure 8.

It is clear that the noise voltage generated by a resistor is given as

$$E_n(t) = i_n(t).R \tag{9}$$

The power density spectrum $S_i(\omega)$ of $i_n(t)$ is a function of the square of $i_n(t)$. Similarly the power density spectrum $S_v(\omega)$ of $v_n(t)$ is a function of the square of $v_n(t)$.

$$\frac{S_v(\omega)}{S_i(\omega)} = \left(\frac{v_n(t)}{i_n(t)}\right)^2 = R^2$$

$$S_v(\omega) = R^2 S_i(\omega) = R^2\,(2KTG)$$

$$S_v(\omega) = 2KTR \qquad \left[G = \frac{1}{R}\right] \tag{10}$$

Power carried by the frequency components with a frequency bandwidth of B is Hz is,

$$P_n = 2S_v(\omega)\, B = 4KTRB = v_n^2 \tag{11}$$

We know the average power of the signal is the same as its mean square value of voltage or current

i.e., $p_n = v_n^2(t) = 4KTRB$ or $V_n = \sqrt{v_n^2(t)} = 4KTRB$

ii) Partition Noise

* This occurs whenever current has to divide between two or more electrodes results in random fluctuations in the current division.

  It is very less in diodes than transistors because in transistors the third electrode draws current. Due to this, more microwave receivers often use diode circuits. Recently gallium arsenide FET which draws zero gate current, has been developed for low noise microwave amplification. The spectrum for partition noise is flat.

  The mean squared value of partition noise in transistor is

$$i_{ne}^2 = 2I_e\left(1 - \frac{|\alpha|^2}{\alpha_0}\right) \tag{12}$$

Where I_e = average collector current

α = current amplification factor

α_0 = current amplification factor at low frequencies.

iii) Shot Noise

* Shot noise is present in both vacuum tube and semiconductor devices. In vacuum tubes shot noise occurs due to random emission of electrons from the cathode. In semiconductor devices, this effect arises due to the random diffusion of minority carriers as well as random generation and recombination of electron hole pairs.

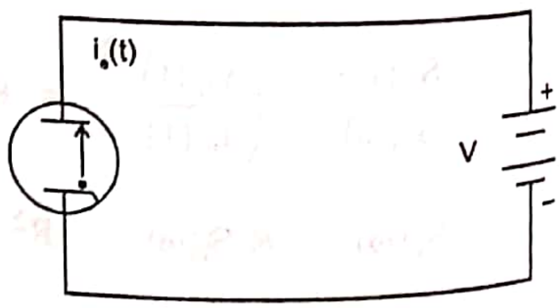


Figure 9 : Shot noise source

* Consider the electron emission from the hot cathode of a parallel plate diode. At a given temperature the average number of electrons emitted per second is constant. The process of electron emission is however random. This means that if we divide the time axis into a large number of small intervals of T seconds each, the number of electrons emitted during each of these intervals is not constant but in random. However the average rate of emission of electrons is constant, provided that it is averaged over a sufficiently long interval of time, it helps the current formed by emitted electrons is not constant but fluctuates about a mean value. The diode current thus fluctuates about a certain mean value. We may consider the total current i(t) as compared if a constant current I_O and a noise current $i_n(t)$ which has a zero mean value.

$$i(t) = I_O + i_n(t) \qquad (13)$$

* The nature of fluctuations can be understood better by considering the process of the induction of current in the plate of the diode due to the emission of an electron. Assume 'that a single electron is, emitted from the cathode. This electron acquires velocity as it moves towards the plate and induces a current $i_C(t)$.

* If the plate and the cathode of a diode are separated by 'd' units, then the emitted electron experiences a force of magnitude of 'qV/d' in the direction of the plate, where 'q' is the charge of an electron and V is the applied voltage.

* The electron will acquire an acceleration of qV/m_d units. where 'm' is the mass of an electron. The initial velocity of the emitted electron is usually much smaller than the final velocity acquired by the electron at the time it strikes the plate, hence the velocity will be assumed to be zero. The velocity v(t) at any time 't' will be given by,

$$v(t) = \frac{qV}{md}t \qquad (14)$$

The kinetic energy (K.E) acquired by an electron at any instant 't' is -1/2 $mv^2(t)$. Substitute the value of v(t) from equation (14) then,

$$K.E = \frac{q^2V^2}{2md^2}t^2 \qquad (15)$$

If the motion of this electron induces a charge 'q' on the plate, then the amount of work 'W' that must be done to induce the charge on the plate with potential 'V is given by $W = V.Q$

Equating this work with the kinetic energy of the electron $Q = \frac{q^2Vt^2}{2md^2}$

The current $i_e(t) = \frac{dQ}{dt} = \frac{q^2V}{md^2}t = \frac{q}{d}\, v(t)$ (16)

Note that the induced current is proportional to the velocity of the electron.

The time required for the electron to reach the plate from cathode is known as the transist time 'τ_a' and can be found from equation (14)

We know that $v(t) = \frac{d}{t}$

and $d = \frac{1}{2}\frac{qv}{md}\tau_a^2$ (17)

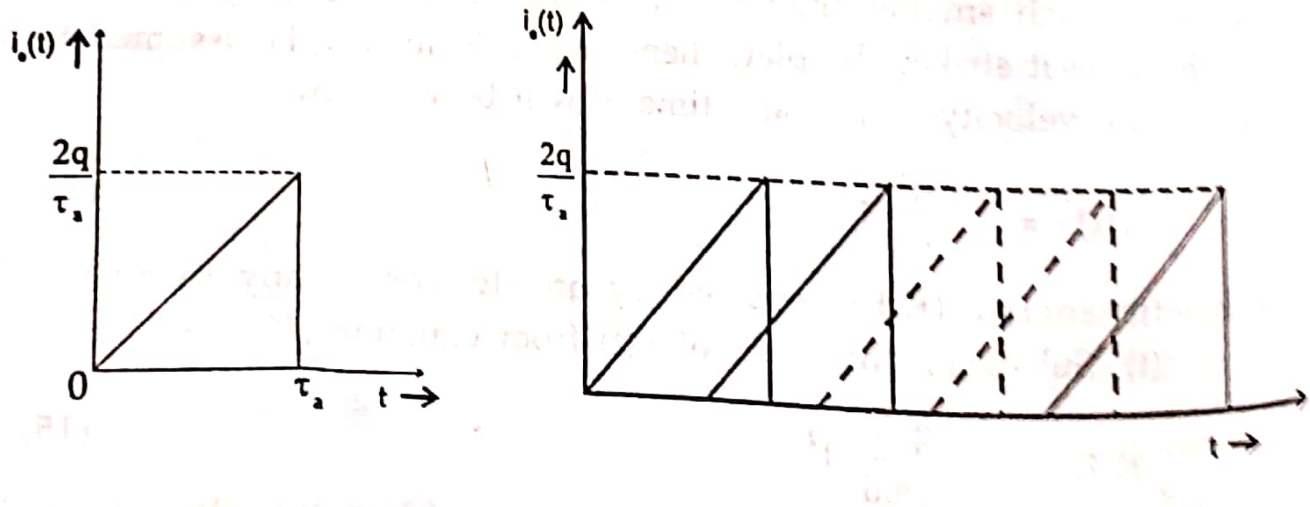
where d = distance between anode and cathode.

$$\tau_a = \left(\sqrt{\frac{2m}{qv}}\right) d$$

substituting τ_a in the equation (16), we get

$$d = \begin{cases} \frac{2q}{\tau_a^2} & \text{for } (0 < t < \tau_a) \\ 0 \end{cases} \quad (18)$$

Obviously the induced current goes to zero as soon as the electron reaches the plate at $t = I_a$. The current pulse induced by a single electron is shown in figure 8.



(a) (b)

Figure 9

Each emitted electron induces such a pulse. Thus the total plate current is composed of a large number of triangle pulses distributed randomly as shown in figure 9(a).

Power Density Spectrum of Shot noise

The shot noise current consists of two components, a constant current component I_O and the time varying component $i_n(t)$. The component $i_n(t)$ as it is random, cannot be specified as a function of time. However $i_n(t)$ represents a stationary random signal and can be specified by its power density spectrum. Since there are $\bar{n}$ pulses per second, it is reasonable to expect that the power density spectrum of $i_n(t)$ will be $\bar{n}$ times the energy density spectrum of $i_e(t)$. Thus if

$$i_e(t) \longleftrightarrow I_e(\omega)$$

The $S_i(\omega)$ the power density spectrum of $i_n(t)$ is given by

$$S_i(\omega) = \bar{n}\,|I_e(\omega)|^2 \tag{19}$$

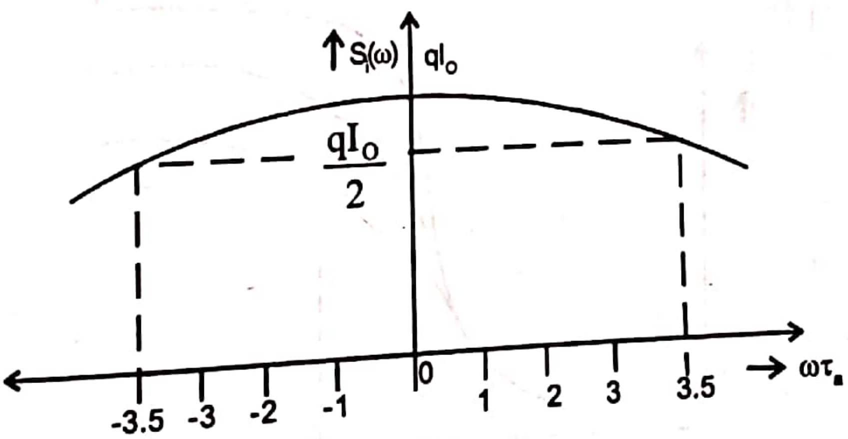


Figure 10 : Power density spectrum for shot noise current

$I_e(\omega)$ is the Fourier transform of $i_e(t)$ and can be found as follows

$$i_e(t) = \frac{2q}{\tau_a^2}[tu(t) - \tau_a\, u\,(t - \tau_a) - (t - \tau_a)\, u(t - \tau_a)\,] \tag{20}$$

Taking Laplace transform on both sides of the above equation (20) and substituting $j\omega$ for 'S' we get

$$i_e(t) \leftrightarrow I_e(\omega) = \frac{2q}{-\omega^2\tau_a^2}\left[1 - e^{-j\omega\tau_a} - j\omega\tau_a e^{-j\omega\tau_a}\right] \tag{21}$$

Substituting this in equation (19) we get,

$$S_i(\omega) = \bar{n}\,|I_O(\omega)|^2 \frac{4I_O q}{(\omega\tau_a)^4}\left[(\omega\tau_a)^2 + 2(1-\cos\omega\tau_a - \omega\tau_a \sin\omega\tau_a)\right]$$

The average power density spectrum $S_i(\omega)$ can be plotted as a function 'ω', and is shown in figure 10.

iv) Noise in Diode

* The operating characteristics of a diode may be divided into two regions. The ***temperature limited region*** and the ***space charge limited region***. In the ***temperature limited region***, the diode current is limited by the temperature of the cathode. The electric field in this region is high enough to attract every electron emitted towards the plate. The increase in electric field cannot increase the average current I_O. The average current can only be increased by increasing the temperature of the cathode and thereby increasing the rate of electron emission. The VI characteristics of a typical diode is shown in figure 11.

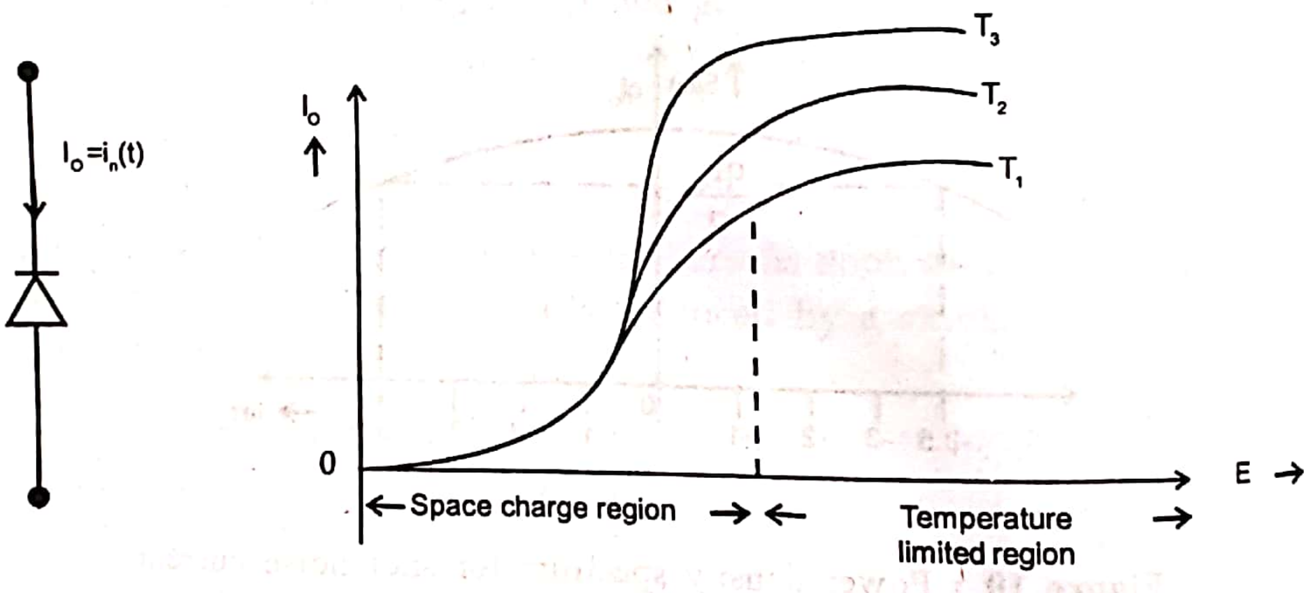


Figure 11 : VI characteristics of diode

* At lower voltages, all the electrons emitted by the cathode does not reach the anode, some electrons remain inside the region between anode and cathode hence this region is called as ***space charge region***. The diode operation in this region is known as ***space charge limited region***. In this region the diode current I_O is, increased by increasing the anode voltage.

* The shot noise derived in the previous article is for the temperature limited region of the diode. Now let us discuss the noise due to space charge region.

The sum of all such pulses constitute the diode current i(t) the area under each pulse is 'q' units, hence the average value of the plate current is given by

$$I_O = \bar{n}q \tag{23}$$

when $\bar{n}$ is the average number of electrons emitted per second.

Noise in Space Charge Limited Region

* When sufficient high voltage is present at the anode then all electrons are attracted and the plate current is limited only by number of electrons emitted per second and thus, it is limited by the cathode temperature. There is no space charge region.

* At lower voltage an electron cloud (space charge) is present between the anode - cathode space. The space charge depresses the potential just outside the cathode and repels some of the emitted electrons.

* Although the electrons are emitted independently from the cathode, there is a strong interaction among the electrons once they are emitted. Hence the arrival of an electron at the plate depends on the previously emitted electrons. These space charge has a smoothing effect upon the random fluctuation.

* If the emission rate increases, the extra electrons increases the space charge which repels more emitted electrons back to the cathode. On the other hand, if the emission rate decreases the space charge is reduced and more electrons reach the plate. Hence the space charge has a tendency to maintain constant current, hence the current fluctuations are smoothened out. The power density spectrum of the noise current $i_n(t)$ for a space charge limited diode is given by

$$S_i(\omega) = \alpha q I_O \tag{24}$$

where $$\alpha = 3\left(1 - \frac{\pi}{4}\right)\frac{2KT_C g_d}{qI_O} \tag{25}$$

then $S_i(\omega) = 1.288\ KT_C g_d$ (26)

T_C is the cathode temperature, K is the Boltzmann constant (K = 1.38×10^{-23} Jules/K) and g_d is the dynamic conductance of the diode.

$$g_d = \frac{\partial I_O}{\partial V} \tag{27}$$

In the space charge region, V and I_O are related by $I_O = BV^{3/2}$ (using child's law).

Hence $$g_d = \frac{\partial I_O}{\partial V} = \frac{3}{2}\frac{E}{I_O} \tag{28}$$

The value of smoothing constant α varies from 0.01 to 1.

Then, $S_i(\omega)$ can be expressed as

$S_i(\omega) = qI_O$ (for temperature limited operation)

$= \alpha I_O$ (for space charge limited operation) (29)

A diode in general can be represented by a noiseless diode in parallel with a noise current source $i_n(t)$ whose power density spectrum is given by the above equation.

e) Shot Noise in Multi electrode devices

The shot noise current for multielectrode devices such as tubes or transistors are as follows :

We know that the power density spectrum of the noise current $i_n(t)$ for a space charge region of the device is given by

$$S_i(\omega) = \alpha q I_O$$

Where $$\alpha = 3\left(1-\frac{\pi}{4}\right)\frac{2KT_C g_d}{qI_O}$$

= smoothing constant (it varies from 0.01 to 1)

Hence $S_i(\omega) = 1.288\ KT_C g_d$

g_d = Dynamic conductance of the transistor or device

$$g_d = \frac{\partial I}{\partial V}$$

Where $V = \sigma(V_g + V_p / \mu)$

V_g = gate or base voltage

V_p = anode or drain or cathode voltage

μ = amplification factor

$$= \frac{\delta I}{\delta\left[\sigma\left(V_g + V_p/\mu\right)\right]} = \frac{\delta I.\delta V_g}{\delta V_g \delta\left[\sigma\left(V_g + V_p/\mu\right)\right]} = \frac{g_m}{\sigma}$$

$$S_i(\omega) = 1.288\ KT_C g_d = \frac{1.288 KT_C g_m}{\sigma}$$

$$= 2KT\left(\frac{0.644\,T_C}{\sigma T}\right)g_m \quad (30)$$

$S_i(\omega)$ is the power spectrum of noise voltage Vn(t). We know the power density is the function of the square of the signal, then the ratio of the power density spectra of $i_n(t)$ and $V_n(t)$ will be g_m^2.

i.e. $$\frac{V_n^2(t)}{i_n^2(t)} = \frac{S_v(\omega)}{S_i(\omega)} = \frac{1}{g_m^2} \quad \text{(or)} \quad S_V(\omega) = \frac{S_i(\omega)}{g_m^2}$$

$$\therefore \quad S_V(\omega) = \frac{1}{g_m^2}\,\frac{2KT(0.644)T_C g_m}{\sigma T} = 0.644\frac{2KT_C}{\sigma g_m} \quad (31)$$

The power density spectrum over the bandwidth of '2B' is given by

$$S_V(\omega) = \frac{0.644\ 2T_C}{\sigma g_m}.2B \quad (32)$$

$$S_V(\omega) = \frac{0.644\ KT_C\,B.4}{\sigma g_m}.V_n^2(t) \quad (33)$$

The shot noise voltage is given by

$$V_n(t) = \sqrt{\left(\frac{0.644}{\sigma}\right)4KT_C\frac{B}{g_m}} \quad (34)$$

f) Low frequency Noise or Flicker Noise

- At low frequencies (below few KHz), a component of noise appears, the spectral density of which increases as the frequency decreases. This known as *Flicker noise.*
- In vacuum tube it arises from slow changes in the oxide structures of oxide coated cathodes and from the migration of impurity ions through the oxide.
- In semiconductors Flicker noise arises from the fluctuations in the carrier densities which in turn give rise to fluctuations in the conductivity of the material.
- It plays an important role in limiting the sensitivity of microwave diode mixers used for dopplar radar systems. This is because the input frequency in the mixer are in the microwave frequency range, the dopplar frequency output is in the lower range where flicker noise is significant.

Burst Noise

- It is the another type of low frequency noise occurs in transistor the name arising because the noise appears as a series of burst at two or more levels. The spectral density increases as frequency decreases.

Avalanche Noise

- The reverse bias characteristics of a diode shows a region where the reverse current increases rapidly with a slight increase in the magnitude of the reverse bias voltage. It is because, holes and electrons in the depletion region gain sufficient energy from the reverse bias to ionize atoms by collision. This ionizing process means that additional holes and electrons are produced which in turn contribute to the ionization process.

The collisions that result in the avalanching occurs at random, with the result that large noise spikes are present in the avalanche current. In zener diode avalanche noise is less, hence it is neglected.

1. INTRODUCTION

Today, communication has entered our daily lives in so many different forms, that it is very difficult to lead a life without the various appliances, gadgets and tools that are born out of the world of communication. Communication, is just not a part of our lives, but is slowly, steadily growing to engulf our entire life.

2. WHAT IS COMMUNICATION?

Communication is the process of conveying or transferring messages from one point to another. Generally it can be classified into two types:

i) Communication within line of sight.

ii) Communication beyond the line of sight between point to point

If the two points are beyond the line of sight, then the branch of communication, engineering comes into picture, and it is known as Telecommunication engineering.

3. COMMUNICATION PROCESS AND ITS COMPONENTS

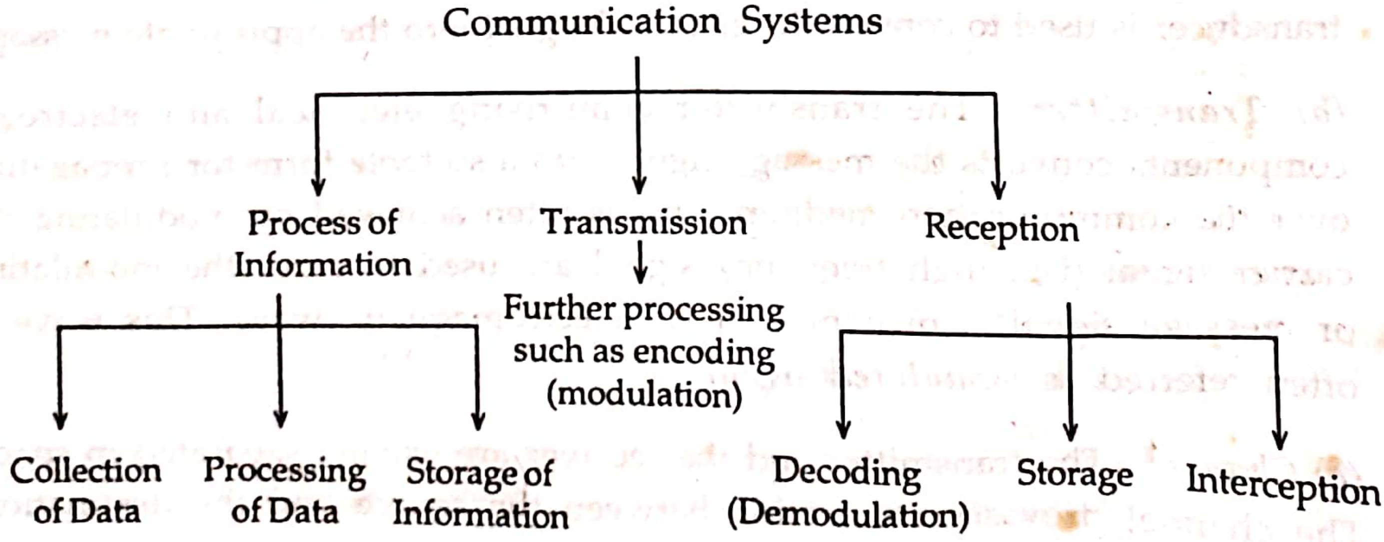


Figure 1(a) : Various Stages of Communication systems

In communication engineering, the physical message such as sound, words, pictures etc., are converted into equivalent electrical values, called signals. This electrical signal is conveyed to a distant place, through a communication media, and at the receiving end, this electrical signal is reconverted back into the original message through some media.

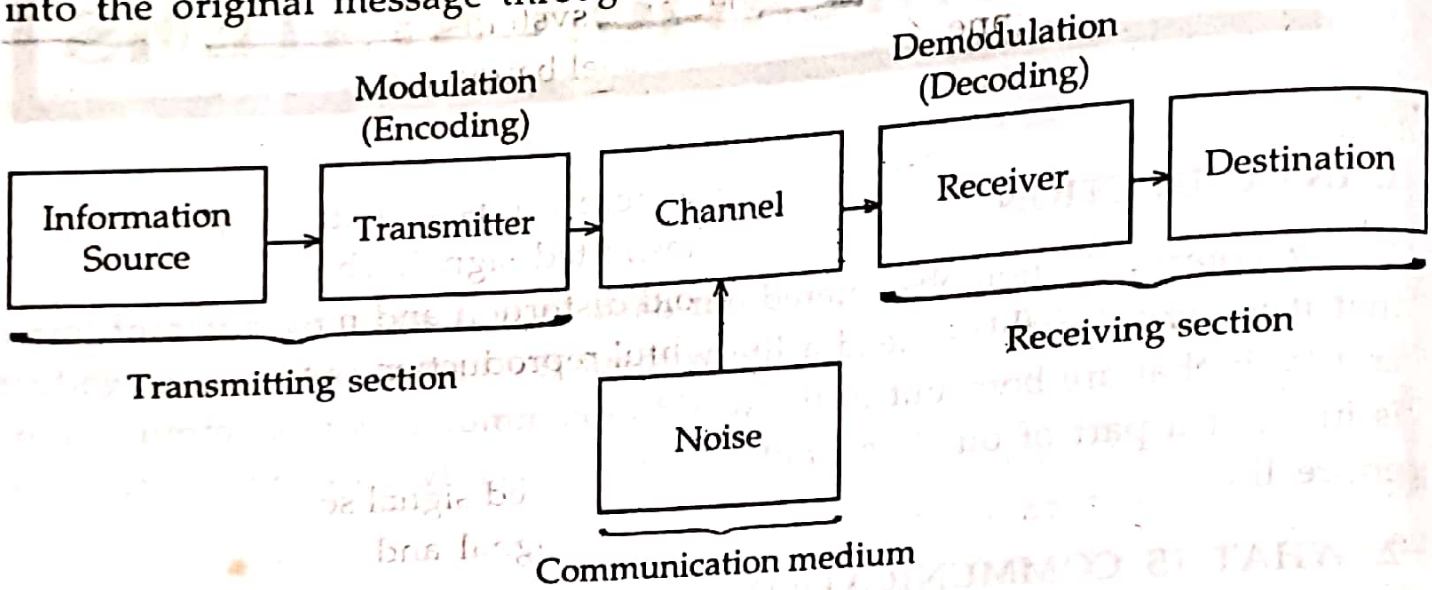


Figure 1(b) : Block Diagram of Communication System

The various stages and the block diagram of a communication system are shown in figure 1(a) and 1(b). The various components of a communication system, with reference to the figure 1(b) are summarised below:

(a) Information source : The message produced by the information source is not electrical in nature, but it may be a voice signal, a picture signal etc. So, an input transducer is required to convert the original physical message into a time varying electrical signal. These signals are called ***base band signals (or) message signals or modulating signals.*** At the destination, another transducer is used to convert the electrical signal into the appropriate message.

(b) Transmitter : The transmitter comprising electrical and electronic components converts the message signal into a suitable form for propagating over the communication medium. This is often achieved by modulating the carrier signal (i.e., high frequency signal are used to carry the modulating or message signal) which may be an electromagnetic wave. This wave is often referred as ***modulated signal.***

(c) Channel : The transmitter and the receiver are usually separated in space. The channel provides connection between the source and the destination.

The channel can be of many forms like coaxial cable, microwave links, radiowave links (or) an optical fibre.

Regardless of its type, the channel degrades the transmitted signal in a number of ways which produces the signal distortion. This occurs in a channel due to, imperfect response of the channel and systems, undesirable electrical interference in the channel, insufficient channel bandwidth and contamination of signals due to noise.

(d) Receiver : The main function of the receiver is to extract the message signal from the degraded version of transmitted signal. The transmitter and the receiver are carefully designed to avoid distortion and minimise the effect of the noise from the receiver. So that faithful reproduction of message emitted by the source is possible.

The receiver has the task of operating on the received signal so as to reconstruct the recognizable form of the original message signal and to deliver it to the user destination.

ATION

Modulation is the process by which some characteristics of a high frequency carrier signal is varied in accordance with the instantaneous value of the another signal called modulating or message signal.

Signal containing information or intelligence to be transmitted is known as modulating or message signal, it is also known as ***Base band signal***. The term Base band designates the band of frequencies representing the signal supplied by the source of information. Usually the frequency of carrier is greater than the modulating signal. The signal resulting from the process of modulation is called modulated signal.

6. NEED FOR MODULATION

The various aspects as to why we should modulate the message signal are summarised below :

(a) For Easy Transmission : If the communication medium is the free space, then antennas are needed to radiate and receive the signal. The antenna radiates effectively when its height is in the order of wave length of the signal to be transmitted.

For example : For a frequency of 1KHz, the height of the antenna required for effective radiation would be half the wavelength,

$$\text{i.e., Antenna height} = \frac{\lambda}{2} = \frac{C}{2f} = \frac{3 \times 10^8}{2 \times 1 \times 10^3} = 150 \text{ Km.}$$

where λ = wave length of the signal to be transmitted

C = velocity of light

f = frequency of the signal to be transmitted

But it is highly impractical to construct and install such an antenna. However, the height of the antenna can be reduced by modulation technique and it achieves effective radiation. The process of modulation provides frequency shifting (or) frequency translation i.e., audio frequency (A.F.) signals are translated into Radio Frequency (R.F.) signals. These R.F. signals act as carrier signal and the A.F. signals act as message signal. Hence, the height of the antenna required is very much reduced. Here, the 1 KHz base band signal is translated into a high frequency signal of 1 MHz.

$$\text{Hence the antenna height} = \frac{\lambda}{2} = \frac{C}{2f} = \frac{3 \times 10^8}{2 \times 1 \times 10^6} = 150 \text{ m.}$$

This height of the antenna is practically achievable.

b) Narrow Banding : Let us assume that the Base Band signal in a broadcast system is radiated directly with the frequency range extending from 50 Hz to 10 KHz. The ratio of the highest to the lowest wavelength is 200. If an antenna is designed for 50Hz, it will be too large for 10 KHz, and vice-versa. Hence we require a wide band antenna which can operate for band edge ratio of 200 Hz which is practically impossible.

However, if the audio signal is modulated (or) translated to radio frequency range of 1 MHz (10^6 Hz) then the ratio of lowest to highest frequency will be $(10^6 + 50) / (10^6 + 10^4) \cong 1/1.01$ which is approximately unity, and the same antenna will be suitable for the entire band extending from $(10^6 + 50)$ Hz to $(10^6 + 10^4)$. Thus modulation converts a wide band signal to a narrow band. This is called narrow banding.

(c) Multiplexing : If more than one signal uses a single channel then modulation may be used to translate different signals to different spectral location, thus enabling the receiver to select the desired signal. Application of multiplexing includes data telemetry, FM stereo phonic, broadcasting and long distance telephones.

(d) To Overcome equipment limitations : Occasionally, in signal processing applications the frequency of the signal to be processed and frequency range of processing apparatus does not match. If the equipment is elaborate and complex, it is operated in some fixed frequency range, thus translating the frequency range of the processing signal corresponding to this fixed range of the equipment. The modulation can be used to accomplish this frequency translation.

(e) *Modulation for Frequency assignment* : Modulation allows several radio and TV stations for broadcasting simultaneously at different carrier frequency and allows different receivers to be tuned to select different stations.

(f) *Modulation to reduce noise and interferences* : The effect of noise and interference, cannot be completely eliminated in a communication system, however it is possible to minimise the effect, by using certain types of modulation schemes. These schemes generally require a transmission band width larger than the band width of message signal.

REVIEW QUESTIONS

PART - A

1. Draw the block diagram of a communication system.
2. Why do we go for modulation?
3. What are the classifications of communication system
4. Define modulation.
5. Calculate the antenna dimension for receiving 30KHz signal.
6. Name some transmission media.
7. In which system the signals are essentially baseband?
8. Is modulation must for all communication system? Explain Why?
9. What do you mean by narrow banding?
10. Classify the different types of modulation.

PART - B

1. With a neat block diagram explain the principle of operation of a basic communication system.
2. Briefly describe the need for modulation.

user destination.

4. TYPES OF COMMUNICATION SYSTEM

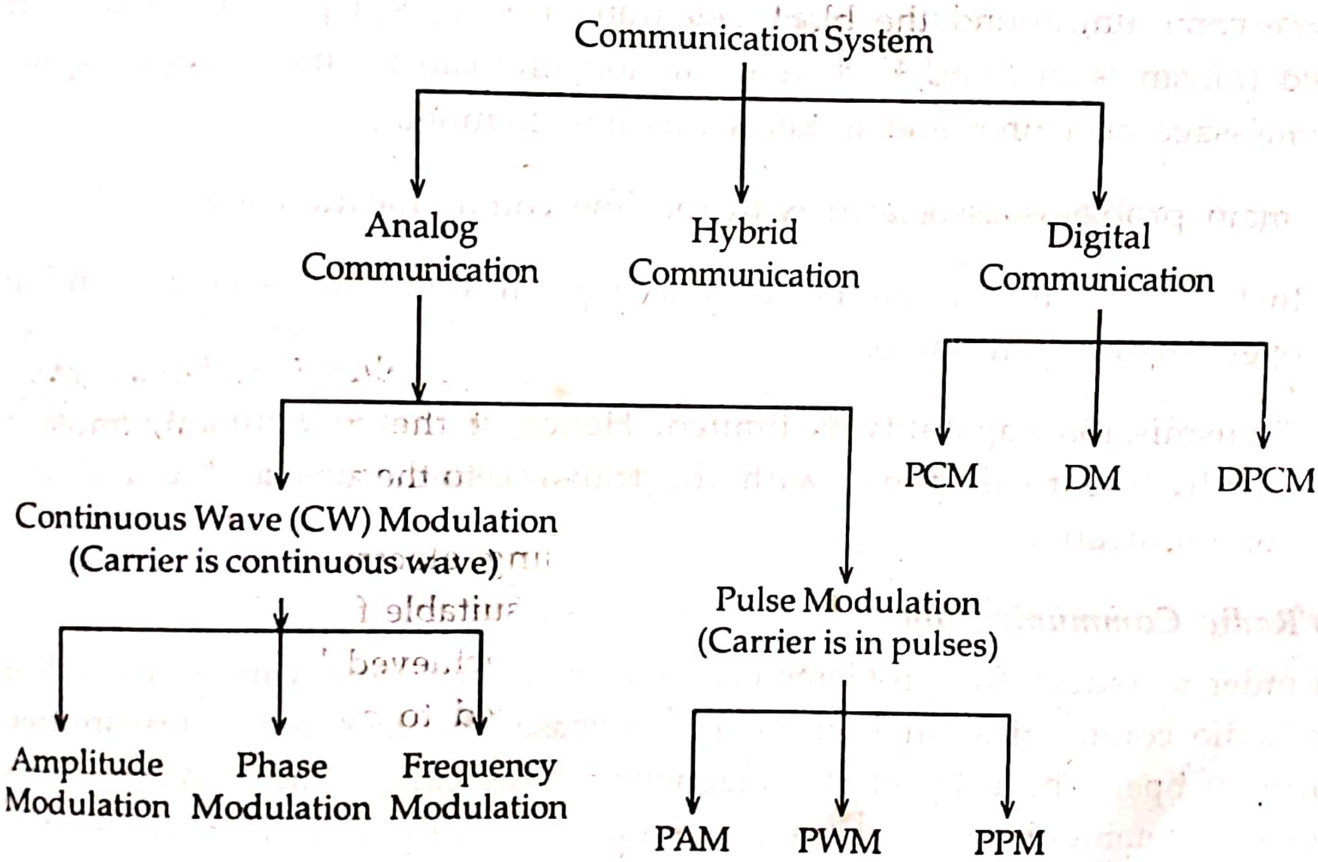


Figure 2 : Overview of Communication Systems

An overview of the types of communication system is shown in Fig. 2. Based on the type of modulation scheme used and the nature of output of the information source, we can divide the communication system into 3 categories.

(a) *Analog communication system* : It is designed to transmit analog information using analog modulation schemes such as Amplitude modulation and Angle modulation.

(b) *Digital communication system* : It is designed to transmit digital information using digital modulation schemes such as PCM, DM, DPCM etc.

(c) *Hybrid communication system* : It is designed to use digital modulation schemes for transmitting sampled and quantised value of analog signals.

Communication engineering is further divided into two categories depending on the transmission media (or) channel used, such as

i) Line Communication   ii) Radio Communication

i) Line Communication

In line communication the medium of transmission is a pair of conductors called transmission lines. Each transmission line can normally convey only one message at a time and is known as line channel.

The main problems associated with the line communication are

* Installation and maintenance of a transmission line is costly and it overcrowds open space.
* Transmission capability is limited. Hence, in many applications it is worthwhile to dispense with the transmission lines and use Radio communication.

ii) Radio Communication

In order to reduce the problems occuring in line communication, a wireless or Radio communication is used. In this case, the message is transmitted through open space by electro-magnetic waves called Radio Waves. This mode of communication is known as Radio communication. Here the signals from various sources are transmitted through the open space. This causes interference among various signals, and as such no useful message is received by the receiver.

The problem of interference is solved by translating the message signal to different R.F. spectra. This is achieved by transmitter using the process called *modulation*. Each radio frequency spectrum behaves as a separate frequency channel and thus avoids interference. The receiver selects the desired radio frequency and produces the useful message by another process called demodulation.

Table 1 : Frequency Ranges in Communication Systems

Type of Signal	Frequency Range	Applications
VLF	3 to 30 KHz	Long distance, point - point communication
LF	30 to 300 KHz	Radio Navigation
MF	300 KHz to 3 MHz	Broadcasting, Marine applications
HF	3 to 30 MHz	Radio Telephony
VHF	30 to 300 MHz	FM broadcasting, TV, mobile radio, Radio navigation
UHF	300 to 3000 MHz	FM broadcasting, TV, mobile radio, Radio navigation
EHF	3 to 30 GHz (Microwaves)	Multichannel telephony links, Radar, Satellite communication.

The radio frequency spectrum is classified into seven service bands and its uses are listed in the Table 1.

The main objectives of communication system designer is to transmit messages as quickly as possible with least probability of error. This is possible by

* Reducing the time of each message.
* Simultaneous transmission of several messages over a single channel. This is known as **multiplexing**.

An error in the received message is due to the various types of distortion that occurs in the system and channel. Hence, a compromise has to be made, to reduce the probability of error and hence we obtain the optimum system.

Communication system covers the study of all the aspects of message transmission with particular emphasis on the following :

i) Accuracy

ii) Band width

iii) Cost

iv) Complexition

v) Speed of transmission

vi) Noise

AMPLITUDE MODULATION AND DEMODULATION. STD-AM (DSB-FC)

Mathematical representation - waveform, frequency spectrum, bandwidth, power relations and Modulation index - Multi tone modulation - Limitations and Modifications in STD-AM: DSB-SC, SSB-SC and VSB AM Generation (Modulators): DSB-FC; square law modulator, Collector and base modulator circuits - DSBSC; Balanced modulator circuit using BJT/FET - SSB: Phase shift method and Filter method - VSB; Filter method Application and Comparison of various AM schemes - AM transmitter: Low and high level Modulation. AM Detection (Demodulators) - Envelope detector, Significance of RC time constant - Square law detector - Costa's PLL detector

AMPLITUDE MODULATION AND DEMODULATION

In communication systems, we often need to design and analyse systems in which many independent message can be transmitted simultaneously through the same channel. It is possible with a technique called *frequency multiplexing*, in which each message is translated in frequency to occupy a different range of spectrum. This involves an auxiliary signal called *carrier* which determines the amount of frequency translation. It requires the amplitude, frequency or phase of the carrier be instantaneously varied as according to the instantaneous value of the message signal. The resulting signal then is called a modulated signal. When the amplitude of the carrier is changed as according to the instantaneous value of the message/baseband signal, it results in *Amplitude Modulation*. The systems implanting such modulation are called as Amplitude modulation systems

Frequency Translation

Frequency translation involves translating the signal from one region in frequency to another region. A signal band-limited in frequency lying in the frequencies from f_1 to f_2, after frequency translation can be translated to a new range of frequencies from f_1' to f_2'. The information in the original message signal at baseband frequencies can be recovered back even from the frequency-translated signal. There are so many benefits which are satisfied by the frequency translation techniques:

1. Frequency Multiplexing: In a case when there are more than one sources which produce band-limited signals that lie in the same frequency band. Such signals if transmitted as such simultaneously through a channel, they will interfere with each other and cannot be recovered back at the intended receiver. But if each signal is translated in frequency such that they encompass different ranges of frequencies, not interfering with other signal spectrums, then each signal can be separated back at the receiver with the use of proper band-pass filters. The output of filters then can be suitably processed to get back the original message signal.

2. Practicability of antenna: In a wireless medium, antennas are used to radiate and to receive the signals. The antenna operates effectively, only when the dimension of the antenna is of the order of magnitude of the wavelength of the signal concerned. At baseband low frequencies, wavelength is large and so is the dimension of antenna required is

impracticable. By frequency translation, the signal can be shifted in frequency to higher range of frequencies. Hence the corresponding wavelength is small to the extend that the dimension of antenna required is quite small and practical.

3. Narrow banding: For a band-limited signal, an antenna dimension suitable for use at one end of the frequency range may fall too short or too large for use at another end of the frequency range. This happens when the ratio of the highest to lowest frequency contained in the signal is large (wideband signal). This ratio can be reduced to close around one by translating the signal to a higher frequency range, the resulting signal being called as a narrow-banded signal. Narrowband signal works effectively well with the same antenna dimension for both the higher end frequency as well as lower end frequency of the band-limited signal.

4. Common Processing: In order to process different signals occupying different spectral ranges but similar in general character, it may always be necessary to adjust the frequency range of operation of the apparatus. But this may be avoided, if by keeping the frequency range of operation of the apparatus constant, every time the signal of interest is translated down to the operation frequency range of the apparatus.

Amplitude Modulation Types:

1. Double-sideband with carrier (DSB-FC) or AM
2. Double-sideband suppressed carrier (DSB-SC)
3. Single-sideband suppressed carrier (SSB-SC)
4. Vestigial sideband (VSB)

Consider a sinusoidal carrier signal C (t) is defined as

$$C(t) = A_c \cos(2\ f_c t +)\ t$$

For our convenience, assume the phase angle of the carrier signal is zero. An amplitude-modulated (AM) wave S(t) can be described as function of time is given by

$$S(t) = A_c\,[1+k_a m(t)] \cos 2\ f_c t$$

Where ka = Amplitude sensitivity of the modulator.

The amplitude modulated (AM) signal consists of both modulated carrier signal and un modulated carrier signal. There are two requirements to maintain the envelope of AM signal is same as the shape of base band signal. The amplitude of the $k_a m(t)$ is always less than unity i.e., $|k_a m(t)|<1$ for all 't'. The carrier signal frequency f_c is far greater than the highest frequency component *W* of the message signal m (t) i.e., $f_c >> W$. Assume the message signal m (t) is band limited to the interval $-W < f < W$

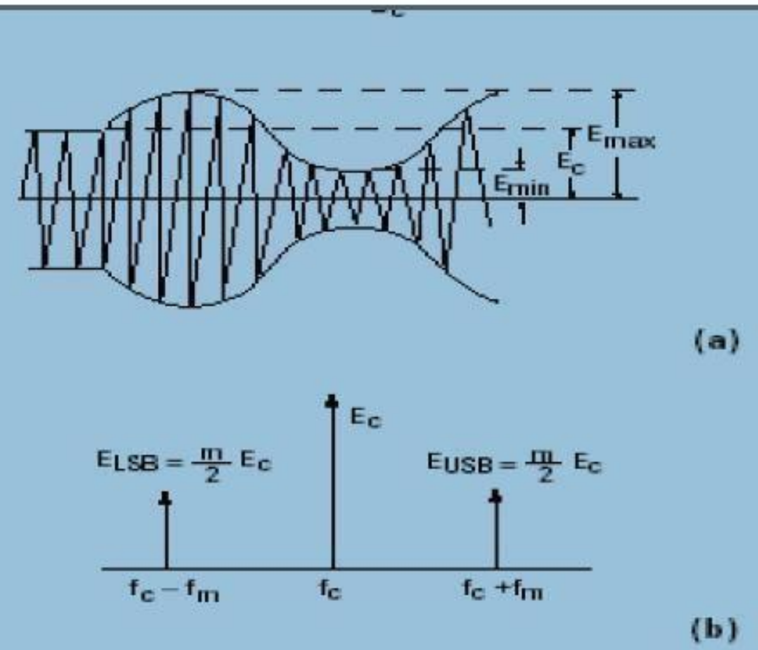


Figure 2(a)(b). Calculation of degree of amplitude modulation from time domain and frequency domain displays

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$$S(f) = A_c/2\,[\delta(f-f_c)+\delta(f+f_c)]+k_aA_c/2[M(f-f_c)+M(f+f_c)]$$

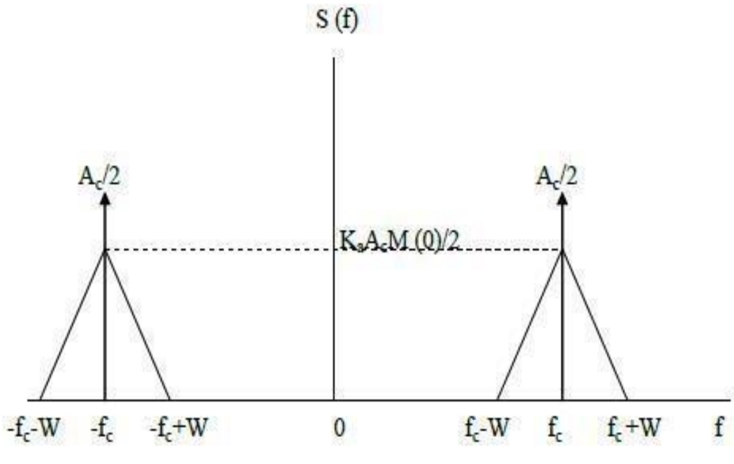


Fig : Spectrum of AM signal

The AM spectrum consists of two impulse functions which are located at fc and -fc and weighted by Ac/2, two USBs, band of frequencies from fc to fc +W and band of frequencies from -fc-W to –fc, and two LSBs, band of frequencies from fc-W to fc and -fc to -fc+W. The difference between highest frequency component and lowest frequency component is known as transmission bandwidth. i.e.,

$$BT = 2W$$

The envelope of AM signal is Ac [1+kam (t)].

Single-tone modulation:

In single-tone modulation modulating signal consists of only one frequency component where as in multi-tone modulation modulating signal consists of more than one frequency component.

$$S (t) = Ac[1+kam(t)]\cos 2 fct .... (i)$$

Let m (t) = Amcos 2 fmt

Substitute m (t) in equation (i)

$$S (t) = Ac [1+ka Amcos 2 fmt] \cos 2 fct$$

Replace the term ka Am by which is known as modulation index or modulation factor.

Modulation index is defined as the ratio of amplitude of message signal to the amplitude of carrier signal. i.e.,

= Am/Ac

(In some books modulation index is designated as ***"m"***)

Which can also be expressed in terms of Amax and Amin:

= (Amax-Amin)/ (Amax+Amin)

Where Amax = maximum amplitude of the modulated carrier signal

Amin = minimum amplitude of the modulated carrier signal

S (t) = Ac cos (2 fct)+Ac /2[cos2 (fc+fm)t]+ Ac /2[cos2 (fc-fm)t]

Fourier transform of S (t) is

S (f) =Ac/2[(f-fc) + (f+fc)] +Ac /4[(f-fc-fm) + (f+fc+fm)]

+ Ac /4[(f- fc+fm) + (f+fc-fm)]

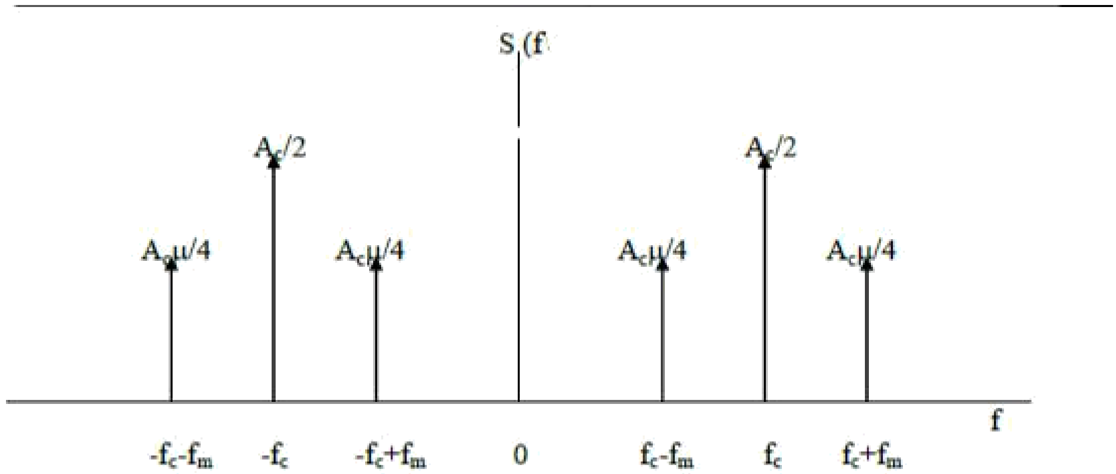


Fig. Spectrum of Single tone AM signal

POWER CALCULATIONS OF SINGLE-TONE AM SIGNAL:

The standard time domain equation for single-tone AM signal is given

S (t) = Accos (2 fct) +Ac /2[cos2 (fc+fm) t] + Ac /2[cos2 (fc-fm) t]

Power of any signal is equal to the mean square value of the signal

Carrier power Pc = $Ac^2/2$

Upper Side Band power PUSB = $Ac^2\ ^2/8$

Lower Side Band power P LSB = $Ac^2\ ^2/8$

Total power PT = Pc + PLSB + PUSB

Total power PT = $Ac^2/2 + Ac^2\ ^2/8 + Ac^2\ ^2/8$

PT = Pc [1+ $^2/2$]

Multi-tone modulation:

In multi-tone modulation modulating signal consists of more than one frequency component where as in single-tone modulation modulating signal consists of only one frequency component.

S (t) = Ac [1+kam (t)] cos 2 fct ....(i) Let m (t) = Am1cos 2 fm1t + Am2cos 2 fm2t

Substitute m (t) in equation (i)

S (t) = Ac [1+ka Am1cos 2 fm1t+ka Am2cos 2 fm2t] cos 2 fct

Replace the term ka Am1 by 1 and Am2 by 2

S (t) = Accos (2 fct) + Ac 1/2[cos2 (fc+fm1) t]+ Ac 1/2[cos2 (fc-fm1) t] + Ac 2/2[cos2 (fc+fm2) t] + Ac 2/2[cos2 (fc-fm2) t]

Fourier transform of S (t) is

$S(f)=A_c/2[\delta(f-f_c)+\delta(f+f_c)]+A_c\mu_1/4[\delta(f-f_c-f_{m1})+\delta(f+f_c+f_{m1})]+$

$A_c\mu_1/4[\delta(f-f_c+f_{m1})+\delta(f+f_c-f_{m1})]+A_c\mu_2/4[\delta(f-f_c-f_{m2})+\delta(f+f_c+f_{m2})]+$
$A_c\mu_2/4[\delta(f-f_c+f_{m2})+\delta(f+f_c-f_{m2})]$

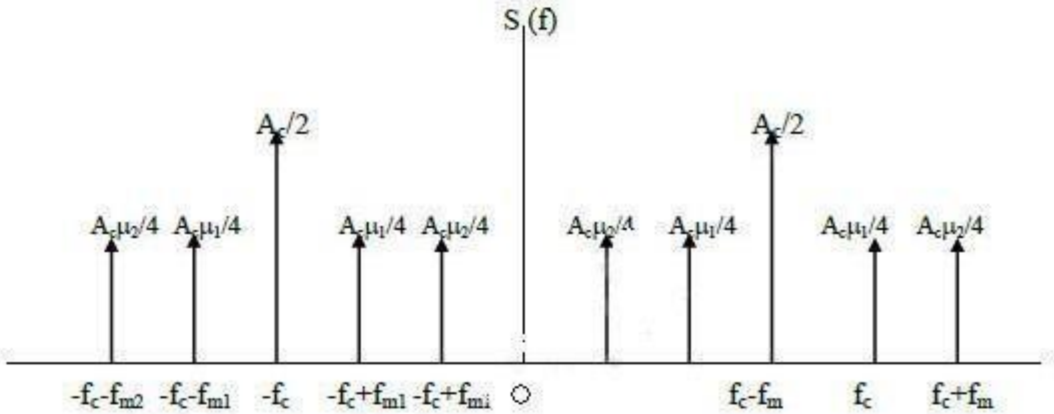


Fig. Spectrum of Multi tone AM signal

Power of Multi-tone AM signal is given by

$P_T = P_c\,[1+\mu_1^2/2+\mu_2^2 2+ \ldots\ldots\ldots +\mu_n^2 2]$

$P_T = P_c\,[1+\mu_t/2]$

Where $\mu_t=\sqrt{\mu_1^2+\mu_2^2+\ldots\ldots\ldots+\mu_n^2}$

Transmission efficiency :-

Transmission efficiency is defined as the ratio of total side band power to the total transmitted power.

i.e., =PSB/PT or 2/ (2+ 2)

Advantages of Amplitude modulation:-

Generation and detection of AM signals are very easy. It is very cheap to build, due to this reason it I most commonly used in AM radio broad casting.

Disadvantages of Amplitude of modulation:-

Amplitude modulation is wasteful of power

Amplitude modulation is wasteful of band width

Application of Amplitude modulation: -

AM Radio Broadcasting

Generation of AM waves

There are two methods to generate AM waves

1. Square-law modulator

2. Switching modulator

Square-law modulator: -

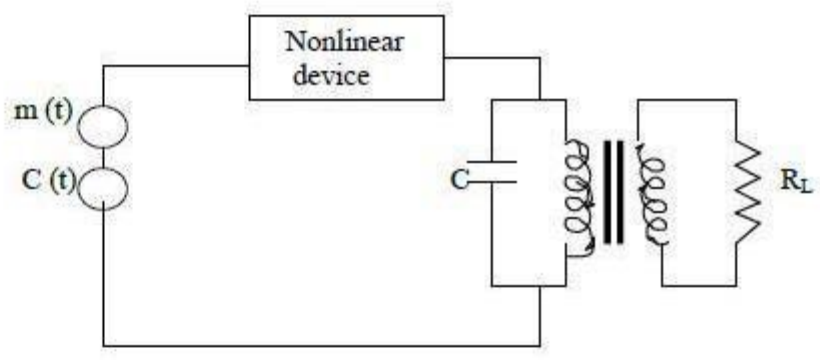


Fig. Square-law Modulator.

A Square-law modulator requires three features: a means of summing the carrier and modulating waves, a nonlinear element, and a band pass filter for extracting the desired modulation products. Semi-conductor diodes and transistors are the most common nonlinear devices used for implementing square law modulators. The filtering requirement is usually satisfied by using a single or double tuned filters.

When a nonlinear element such as a diode is suitably biased and operated in a restricted portion of its characteristic curve, that is ,the signal applied to the diode is relatively weak, we find that transfer characteristic of diode-load resistor combination can be represented closely by a square law :

$V_0(t) = a_1 V_i(t) + a_2 V_i^2(t)$ .... (i)

Where a1, a2 are constants

Now, the input voltage Vi (t) is the sum of both carrier and message signals i.e., Vi (t) =Accos 2 fct+m (t)...................... (ii)

Substitute equation (ii) in equation (i) we get
V0 (t) =a1Ac [1+kam (t)] cos2 fct +a1m (t)+a2Ac2cos22 fct+a2m2(t) .... (iii)

Where ka =2a2/a1

Now design the tuned filter /Band pass filter with center frequency fc and pass band frequency width 2W.We can remove the unwanted terms by passing this output voltage V0(t) through the band pass filter and finally we will get required AM signal.

V0 (t) =a1Ac [1+2a2/a1 m (t)] cos2 fct

Assume the message signal m (t) is band limited to the interval –W<f< W

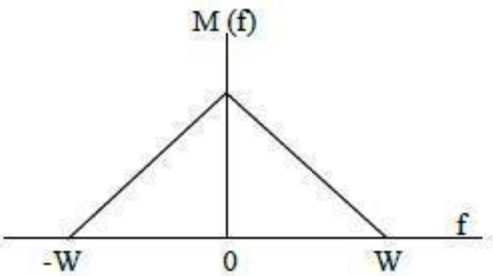


Fig .Spectrum of message signal

The Fourier transform of output voltage V_O (t) is given by

$$V_O (f) = a_1 A_C/2[\delta(f-f_c) + \delta(f+f_c)] + a_2 A_C [M (f-f_c) + M (f+f_c)]$$

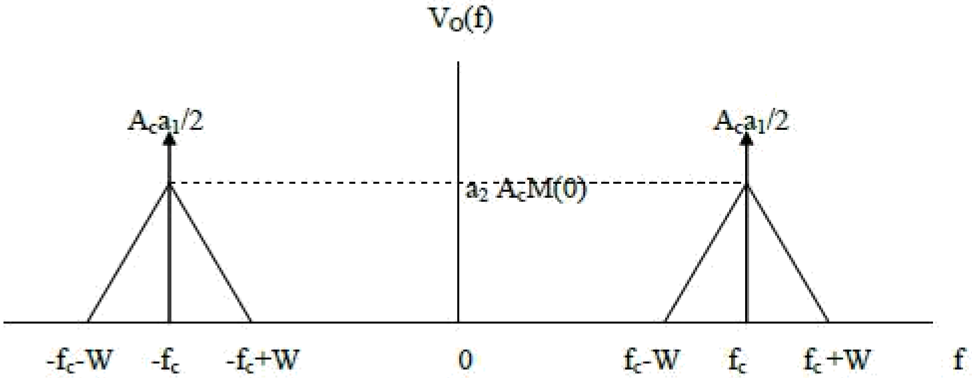


Fig: Spectrum of AM signal

The AM spectrum consists of two impulse functions which are located at fc & -fc and weighted by Aca1/2 & a2Ac/2, two USBs, band of frequencies from fc to fc +W and band of frequencies from -fc-W to –fc, and two LSBs, band of frequencies from fc-W to fc & -fc to -fc+W.

<u>*Szñtc* bing Alodulator: -</u>

C (I) =.@cos2u -t

$-f_c$-W $-f_c$ $-f_c$+W

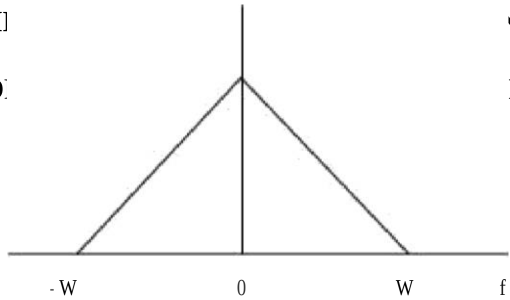
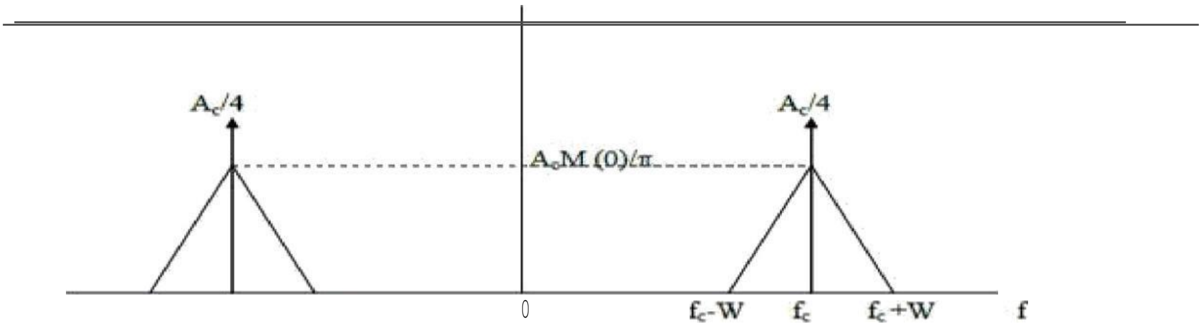


Fig: Spectrum of message signal

T

The Fourier transform of output voltage V_O (t) is given by

$$V_O(f) = A_C/4[\delta(f-f_c) + \delta(f+f_c)] + A_C/\pi[M(f-f_c) + M(f+f_c)]$$



F ig. Spectrum of AhI sign at

Assume that carrier wave C (t) applied to the diode is large in amplitude, so that it swings right across the characteristic curve of the diode .we assume that the diode acts as an ideal switch, that is, it presents zero impedance when it is forward-biased and infinite impedance when it is reverse-biased. We may thus approximate the transfer characteristic of the diode-load resistor combination by a piecewise-linear characteristic.

The input voltage applied Vi (t) applied to the diode is the sum of both carrier and message signals.

Vi (t) =Accos 2 fct+m (t)..................... (i)

During the positive half cycle of the carrier signal i.e. if C (t)>0, the diode is forward biased, and then the diode acts as a closed switch. Now the output voltage Vo (t) is same as the input voltage Vi (t) . During the negative half cycle of the carrier signal i.e. if C(t) <0, the diode is reverse biased, and then the diode acts as a open switch. Now the output voltage VO (t) is zero i.e. the output voltage varies periodically between the values input voltage Vi (t) and zero at a rate equal to the carrier frequency fc.

i.e., Vo (t) = [Accos 2 fct+m (t)] gP(t).............(ii)

Where gp(t) is the periodic pulse train with duty cycle one-half and period Tc=1/fc and which is given by

gP(t)= ½+2/ ∑ [(-1)n-1/(2n-1)]cos [2 fct(2n-1)]................(iii)

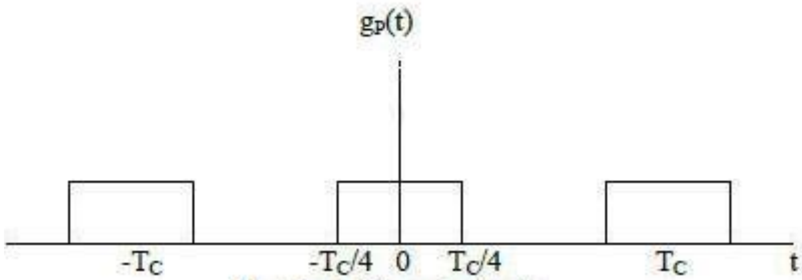


Fig. Periodic pulse train

V0 (t) =Ac/2[1+kam (t)] cos2 fct +m (t)/2+2AC/ cos 22 fct……….(iii) Where ka = 4/ AC

Now design the tuned filter /Band pass filter with center frequency fc and pass band frequency width 2W.We can remove the unwanted terms by passing this output voltage V0(t) through the band pass filter and finally we will get required AM signal.

V0 (t) =Ac/2[1+kam (t)] cos2 fct

Assume the message signal m(t) is band limited to the interval –W<f< W

The AM spectrum consists of two impulse functions which are located at fc & -fc and weighted by Aca1/2 & a2Ac/2, two USBs, band of frequencies from fc to fc +W and band of frequencies from -fc-W to –fc, and two LSBs, band of frequencies from fc-W to fc & -fc to -fc+W.

Demodulation of AM waves:

There are two methods to demodulate AM signals. They are:

Square-law detector

Envelope detector

Square-law detector:-

A Square-law modulator requires nonlinear element and a low pass filter for extracting the desired message signal. Semi-conductor diodes and transistors are the most common nonlinear devices used for implementing square law modulators. The filtering requirement is usually satisfied by using a single or double tuned filters. When a nonlinear element such as a diode is suitably biased and operated in a restricted portion of its characteristic curve, that is ,the signal applied to the diode is relatively weak, we find that transfer characteristic of diode-load resistor combination can be represented closely by a square law :

V0 (t) = a1Vi (t) + a2 Vi2 (t) .... (i)

Where a1, a2 are constants

Now, the input voltage Vi (t) is the sum of both carrier and message signals i.e., Vi (t) = Ac [1+kam (t)] cos2 fct ....(ii)

Substitute equation (ii) in equation (i) we get

V0 (t) = a1Ac [1+kam (t)] cos2 fct + 1/2 a2Ac2 [1+2 kam (t) + ka2m2 (t)] [cos4 fct] ....(iii)

Now design the low pass filter with cutoff frequency f is equal to the required message signal bandwidth. We can remove the unwanted terms by passing this output voltage V0 (t) through the low pass filter and finally we will get required message signal.

V0 (t) = Ac2 a2 m (t)

The Fourier transform of output voltage VO (t) is given by VO (f) = Ac2 a2 M (f)

Envelope detector is used to detect high level modulated levels, whereas square-law detector is used to detect low level modulated signals (i.e., below 1v). It is also based on the switching action or switching characteristics of a diode. It consists of a diode and a resistor-capacitor filter. The operation of the envelope detector is as follows. On a positive half cycle of the input signal, the diode is forward biased and the capacitor C charges up rapidly to the peak value of the input signal. When the input signal falls below this value, the diode becomes reverse biased and the capacitor C discharges slowly through the load resistor Rl . The discharging process continues until the next positive half cycle. When the input signal becomes greater than the voltage across the capacitor, the diode conducts again and the process is repeated.

The charging time constant RsC is very small when compared to the

carrier period 1/fc i.e., RsC << 1/fc

Where Rs = internal resistance of the voltage source. C = capacitor ,fc = carrier frequency

i.e., the capacitor C charges rapidly to the peak value of the signal. The discharging time constant RlC is very large when compared to the charging time constant i.e.,

1/fc << RlC << 1/W

Where Rl = load resistance value

W = message signal bandwidth

i.e., the capacitor discharges slowly through the load resistor.

Advantages:

It is very simple to design

It is inexpensive

Efficiency is very high when compared to Square Law detector

Disadvantage:

Due to large time constant, some distortion occurs which is known as diagonal clipping i.e., selection of time constant is somewhat difficult

Application:

It is most commonly used in almost all commercial AM Radio receivers.

Types of Amplitude modulation:-

There are three types of amplitude modulation. They are:

- Double Sideband-Suppressed Carrier(DSB-SC) modulation
- Single Sideband(SSB) modulation
- Vestigial Sideband(SSB) modulation

DOUBLE SIDEBAND-SUPPRESSED CARRIER (DSBSC)

MODULATION

Double sideband-suppressed (DSB-SC) modulation, in which the transmitted wave consists of only the upper and lower sidebands. Transmitted power is saved through the suppression of the carrier wave, but the channel bandwidth requirement is same as in AM (i.e. twice the bandwidth of the message signal). Basically, double sideband-suppressed (DSB-SC) modulation consists of the product of both the message signal m (t) and the carrier signal c(t),as follows:

S (t) =c (t) m (t)

S (t) =Ac cos (2 fct) m (t)

The modulated signal s (t) undergoes a phase reversal whenever the message signal m (t) crosses zero. The envelope of a DSB-SC modulated signal is different from the message signal.

The transmission bandwidth required by DSB-SC modulation is the same as that for amplitude modulation which is twice the bandwidth of the message

signal, 2W. Assume that the message signal is band-limited to the interval –W ≤f≤ W

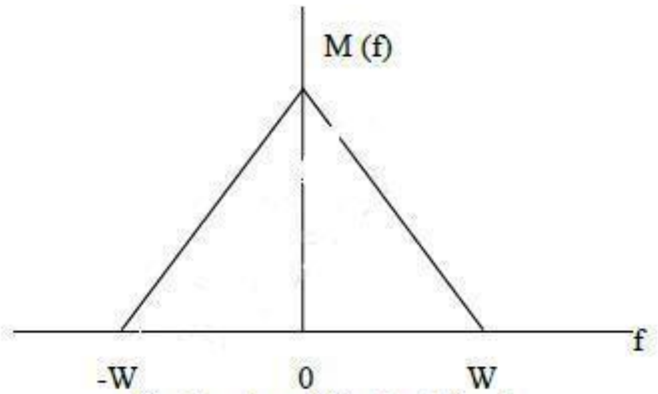


Fig. Spectrum of Baseband signal

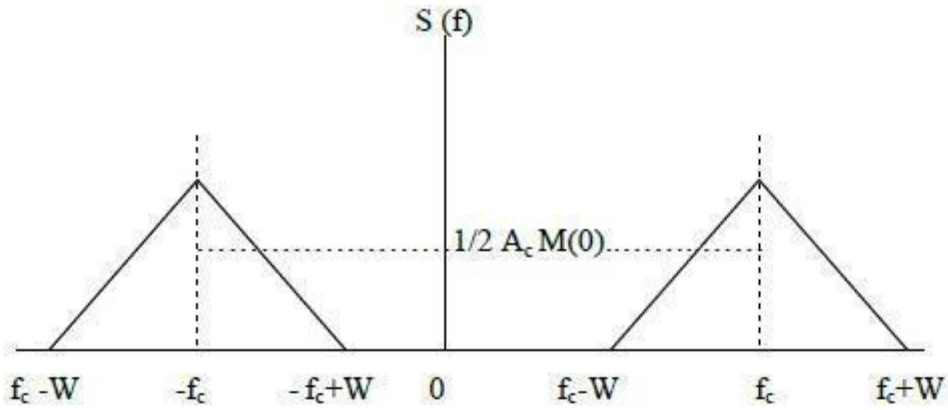


Fig. Spectrum of DSBSC wave

Single-tone modulation:-

In single-tone modulation modulating signal consists of only one frequency component where as in multi-tone modulation modulating signal consists of more than one frequency components. The standard time domain equation for the DSB-SC modulation is given by

S (t) =Ac cos (2 fct) m (t) ....(1)

Assume m (t) =Amcos (2 fmt)...........................(2)

Substitute equation (2) in equation (1) we will get S (t) =Ac Am cos (2 fct) cos (2 fmt)

S (t) = Ac Am/2[cos 2π (fc-fm) t + cos 2π (fc+fm) t].....................(3)

The Fourier transform of s (t) is

S (f) =Ac Am/4[δ (f-fc-fm) + δ (f+fc+fm)] + Ac Am/4[δ (f-fc+fm) + δ(f+fc+fm)]

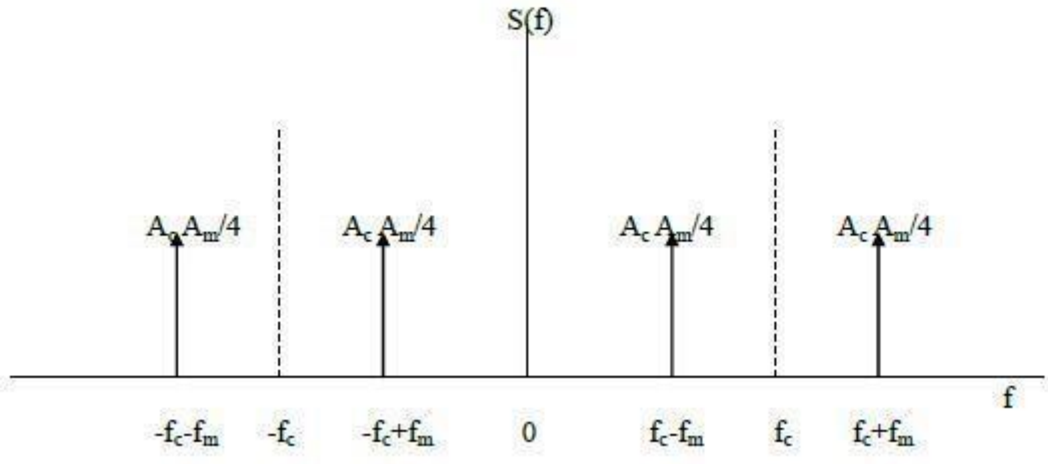


Fig. Spectrum of Single tone DSBSC wave

Power calculations of DSB-SC waves:-

Total power PT = PLSB+PUSB

Total power PT =Ac2Am2/8+Ac2Am2/8

Total power PT =Ac2Am2/4

Generation of DSB-SC waves:-

There are two methods to generate DSB-SC waves. They are: Balanced modulator,Ring modulator.

BALANCED MODULATOR

One possible scheme for generating a DSBSC wave is to use two AM modulators arranged in a balanced configuration so as to suppress the carrier wave, as shown in above fig. Assume that two AM modulators are identical, except for the sign reversal of the modulating signal applied to the input of one of the modulators. Thus the outputs of the two AM modulators can be expressed as follows:

S1 (t) = Ac [1+kam (t)] cos 2 fct and

S2 (t) = Ac [1- kam (t)] cos 2 fct Subtracting S2 (t) from S1 (t), we obtain

S (t) = S1 (t) – S2 (t)

S (t) = 2Ac kam (t) cos 2 fct

Balanced Modulator:-

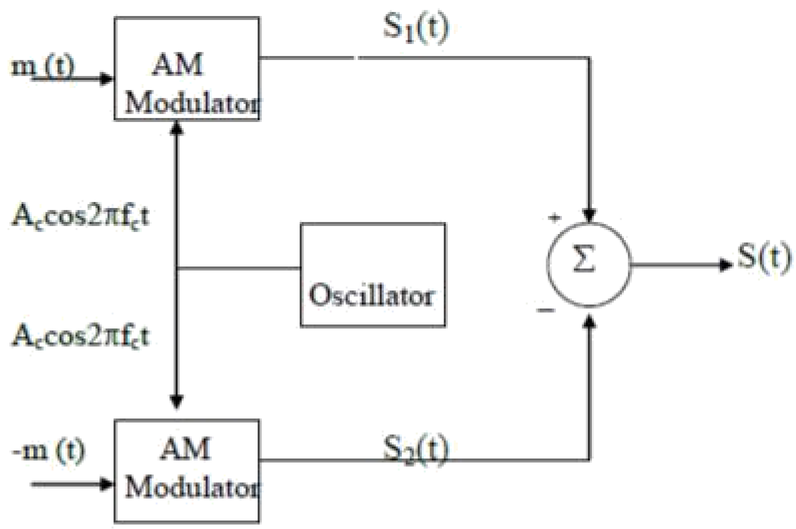


Fig. Balanced Modulator

Hence, except for the scaling factor 2ka the balanced modulator output is equalto product of the modulating signal and the carrier signal. The Fourier transform of s (t) is S (f) =kaAc [M (f-fc) + M (f+fc)] Assume that the message signal is band-limited to the interval –W≤f≤ W.

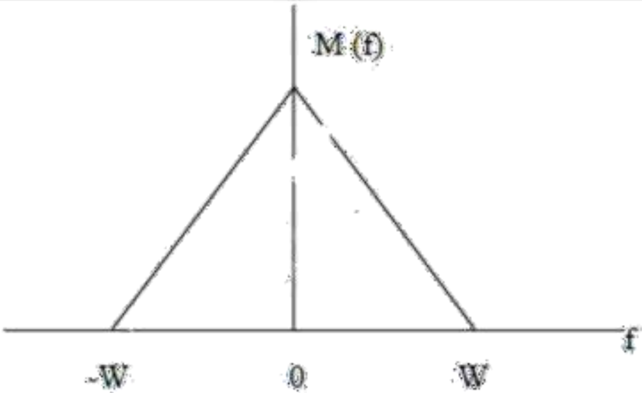
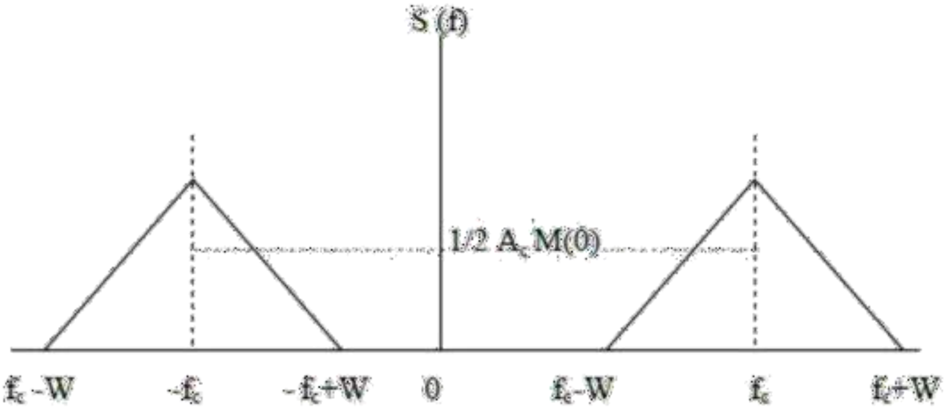
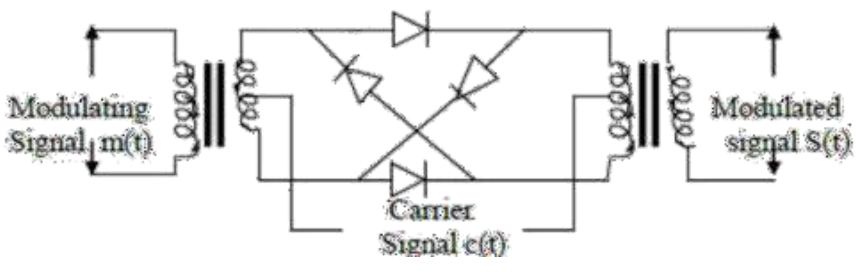


Fig. Spectrum of Baseband signal



Ring modulator:-



One of the most useful product modulator, well suited for generating a DSBSC wave, is the ring modulator shown in above figure. The four diodes form ring in which they all point in the same way-hence the name. The diodes are controlled by a square-wave carrier c (t) of frequency fc, which applied longitudinally by means of to center-tapped transformers. If the transformers are perfectly balanced and the diodes are identical, there is no leakage of the modulation frequency into the modulator output. On one half-cycle of the carrier, the outer diodes are switched to their forward resistance rf and the inner diodes are switched to their backward resistance rb .on other half-cycle of the carrier wave, the diodes operate in the opposite condition. The square wave carrier c (t) can be represented by a Fourier series as follows:

c (t)=4/π Σ (-1)n-1/(2n-1) cos [2πfct(2n-1)] n=1

When the carrier supply is positive, the outer diodes are switched ON and the inner diodes are switched OFF, so that the modulator multiplies the message signal by +1 When the carrier supply is positive, the outer diodes are switched ON and the inner diodes are switched OFF, so that the modulator multiplies the message signal by +1.when the carrier supply is negative, the outer diodes are switched OFF and the inner diodes are switched ON, so that the modulator multiplies the message signal by -1. Now, the Ring modulator output is the product of both message signal m (t) and carrier signal c (t).

S (t) =c (t) m (t)

S (t) =4/π Σ (-1) n-1/ (2n-1) cos [2πfct (2n-1)] m (t) n=1

For n=1

S (t) =4/π cos (2πfct) m (t)

There is no output from the modulator at the carrier frequency i.e the modulator output consists of modulation products. The ring modulator is sometimes referred to as a double-balanced modulator, because it is balanced with respect to both the message signal and the square wave carrier signal. The Fourier transform of s (t) is

S (f) =2/π [M (f-fc) + M (f+fc)]

Assume that the message signal is band-limited to the interval –W≤f≤ W

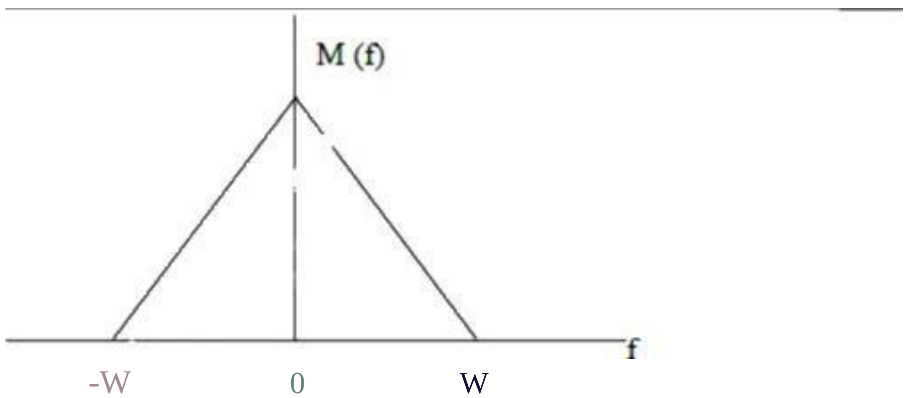
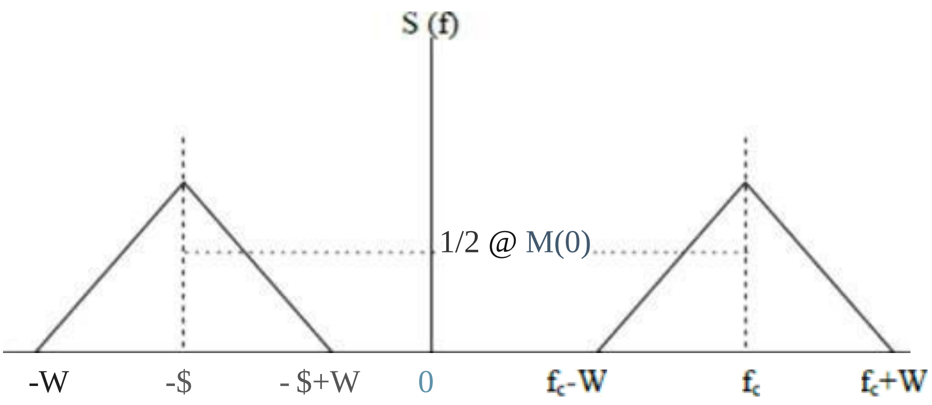


Fig. Specm n of 8astbaed signal



Coherent Detection of DSB-SC’ ¥¥'aves--

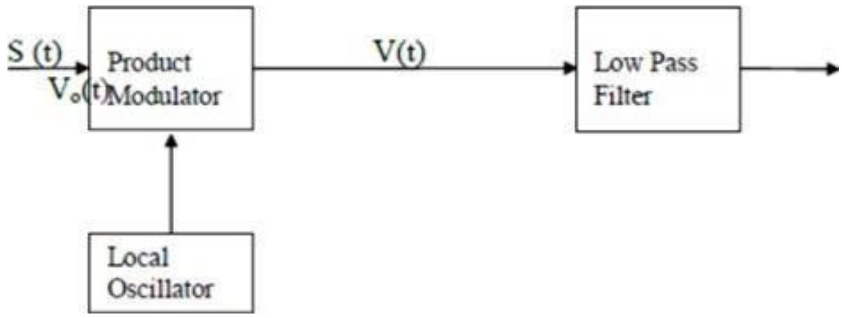


Fig. Coherent detection of DSBSC wave form

The base band signal m (t) can be recovered from a DSB-SC wave s (t) by multiplying s(t) with a locally generated sinusoidal signal and then low pass filtering the product. It is assumed that local oscillator signal is coherent or synchronized, in both frequency and phase ,with the carrier signal c(t) used in the product modulator to generate s(t).this method of demodulation is know as coherent detection or synchronous demodulation. The product modulator produces the product of both input signal and local oscillator and the output of the product modulator v (t) is given by

v (t) =Áccos (2πfct+Ø) s (t)

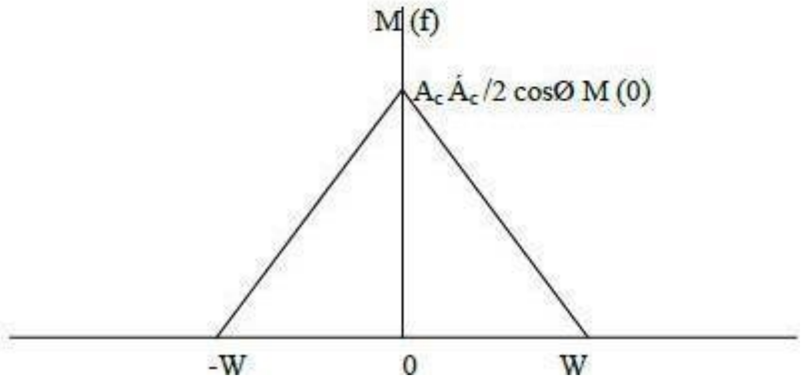
v(t) =Áccos (2πfct+Ø) Accos2πfct m (t)

v (t) =Ac Ác/2 cos(2πfct+Ø) m(t)+ Ac Ác/2 cosØ m(t)

The high frequency can be eliminated by passing this output voltage to the Low Pass Filter. Now the Output Voltage at the Low pass Filter is given by v0 (t) = Ac Ác/2 cosØ m(t)

The Fourier transform of v_o (t) is

$$V_O(f) = A_c \acute{A}_c/2 \cos Ø\, M(f)$$



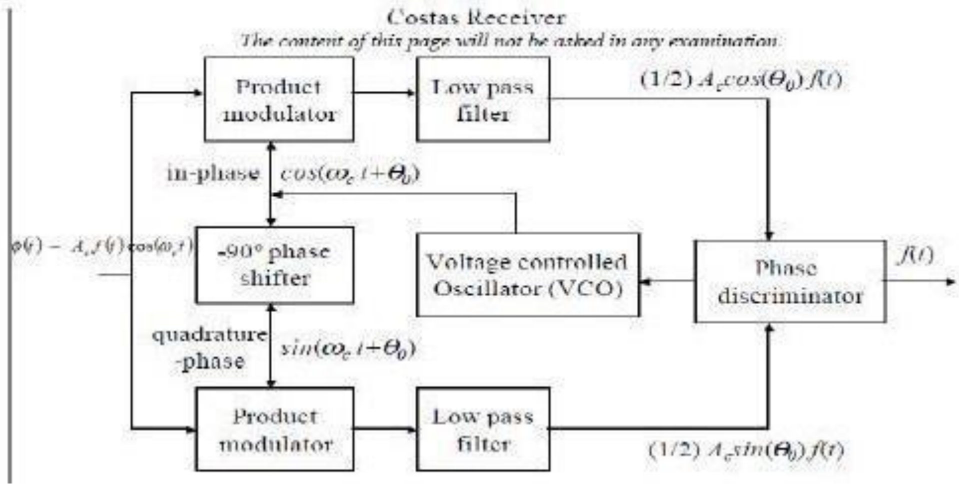
The demodulated signal is proportional to the message signal m (t) when the phase error is constant. The Amplitude of this Demodulated signal is maximum when Ø=0, and it is minimum (zero) when Ø=±π/2 the zero demodulated signal, which occurs for Ø=±π/2 represents quadrature null effect of the coherent detector.

Conventional AM DSB communication systems have two inherent disadvantages.

First, with conventional AM, carrier power constitutes two thirds or more of the total transmitted power .This is a major drawback because the carrier contains no information.

Conventional AM systems utilize twice as much bandwidth as needed with SSB systems. With SSB transmission, the information contained in the USB is identical the information contained in the LSB.Therefore, transmitting both sidebands is redundant. Consequently, Conventional AM is both power and bandwidth inefficient, which are the two predominant considerations when designing modern electronic communication systems.

COSTA'S Loop



SSB MODULATION

Generation of SSB waves:

- Filter method
- Phase shift method
- Third method (Weaver's method)

Demodulation of SSB waves:

Coherent detection: it assumes perfect synchronization between the local carrier and that used in the transmitter both in frequency and phase.

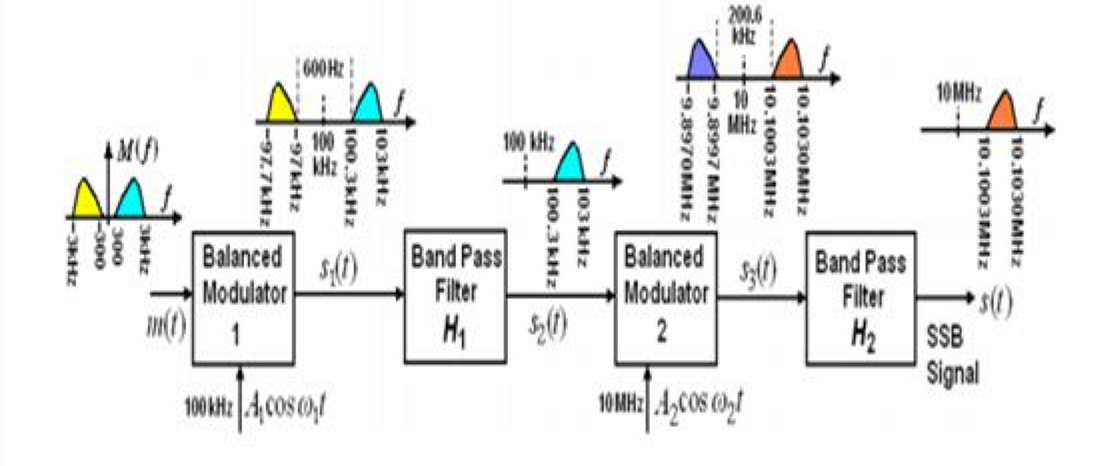
Effects of frequency and phase errors in synchronous detection-DSB-SC, SSB-SC:

Any error in the frequency or the phase of the local oscillator signal in the receiver, with respect to the carrier wave, gives rise to distortion in the demodulated signal. The type of distortion caused by frequency error in the demodulation process is unique to SSB modulation systems. In order to reduce the effect of frequency error distortion in telephone systems, we have to limit the frequency error to 2-5 Hz. The error in the phase of the local oscillator signal results in phase distortion, where each frequency component of the message signal undergoes a constant phase shift at the demodulator output. This phase distortion is usually not serious with voice communications because the human ear is relatively insensitive to phase distortion; the presence of phase distortion gives rise to a Donald Duck voice effect.

Frequency Discrimination / Filter Method : An SSB modulator based on frequency discrimination consists of a balanced modulator and a filter which is designed to pass the desired sideband and suppress the undesired one. This method is also known as filter method of generation of SSB signal. The most severe requirement of this method of SSB generation arises from the desired sideband by twice the lowest frequency component of the modulating signal. A typical arrangement for generating SSB signal by frequency discrimination method is shown in Figure. For a satisfactory performance of the system the following two requirements have to be satisfied.

1. The pass band of the filter should be same as that of the desired sideband. 2. The transition band

of the filter should not exceed twice the minimum frequency component present in the baseband signal. This kind of frequency discrimination is possible by using highly selective filter, which can be realized using high Q (on the range of 1000 to 2000) crystal resonator. There is another problem associated with the generation of SSB by frequency discrimination method when the SSB wave occupies a frequency band which is much larger than the baseband signal. For example, consider the translation of voice signals (approximately 300 to 3400 Hz) to high frequency range of radio spectrum. In such cases it is difficult to design a filter to pass the desired band and reject the other using the simple arrangement. To overcome this difficulty, multistage modulation and filtering scheme may be used to ease the filtering requirements. This is shown in Figure, where two stage modulation has been used. In this arrangement the SSB wave at the first filter output is used as the modulating wave for the second balanced modulator, which produces DSB-SC wave with a spectrum that is symmetrically spaced around the second carrier. The frequency separation between the two sidebands of the DSB-SC wave is effectively twice the first carrier frequency. This enables the easy removal of the unwanted sidebands. The following example will make this more clear.

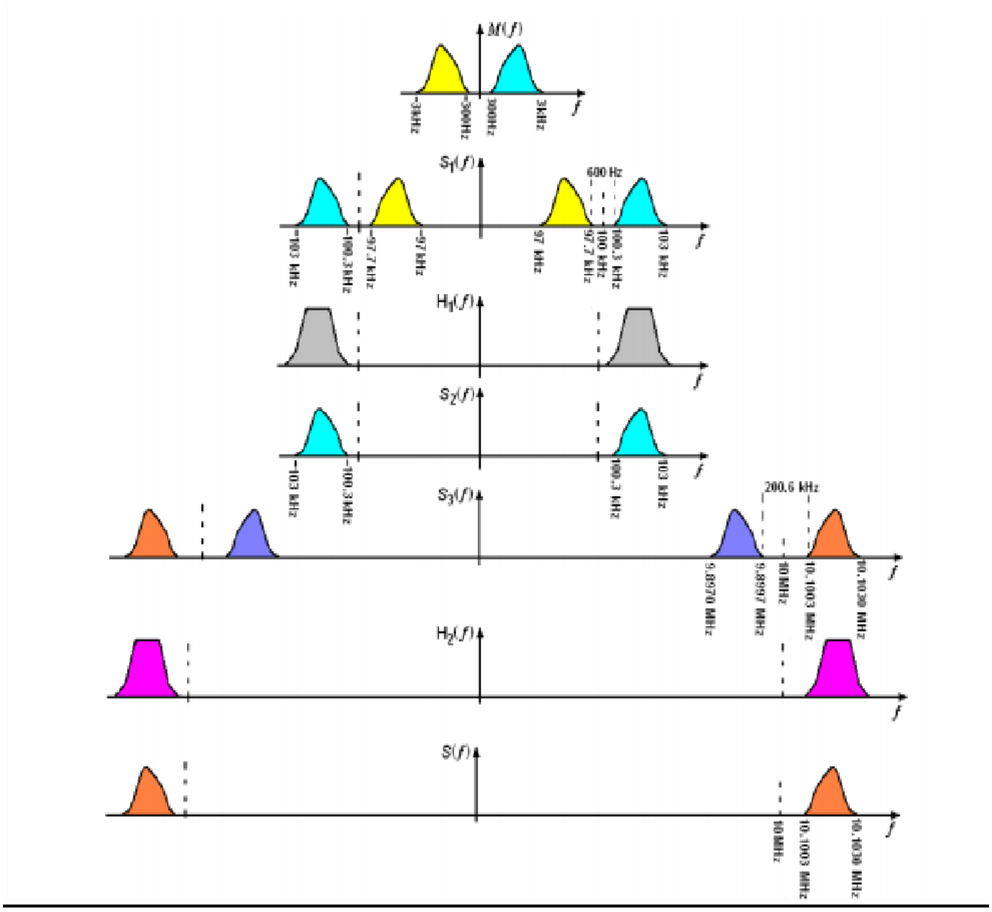


Block diagram of Frequency discrimination SSB Modulation

Consider that we desire to generate an SSB signal with a carrier of 10 MHz and the baseband signal consists of voice signal occupying frequency band 300 Hz to 3kHz. Suppose we use a two stage modulation scheme in which we select first carrier frequency f1=100 kHz and the second one as f2 = 10 MHz.

Stage 1: In the first stage, a balanced modulator with carrier frequency f1=100 kHz is used for generation of DSB-SC wave. The spectrum of the DSB-SC wave s1(t) appearing at the output of first balanced modulator has a lower sideband occupying the frequency band 97 to 99.7 kHz and the upper sideband occupying 100.3 to 103 kHz. By assuming that only USB occupying 100.3 to 103 kHz is selected resulting in the SSB wave.

In order to achieve this, the first band pass fillter (transfer function H1) must have a lower transition band from 99.7 to 100.3 kHz, so as to suppress the LSB. The transition band is 600 Hz as shown in Figure. This requirement which is 6% of the center frequency can met easily.



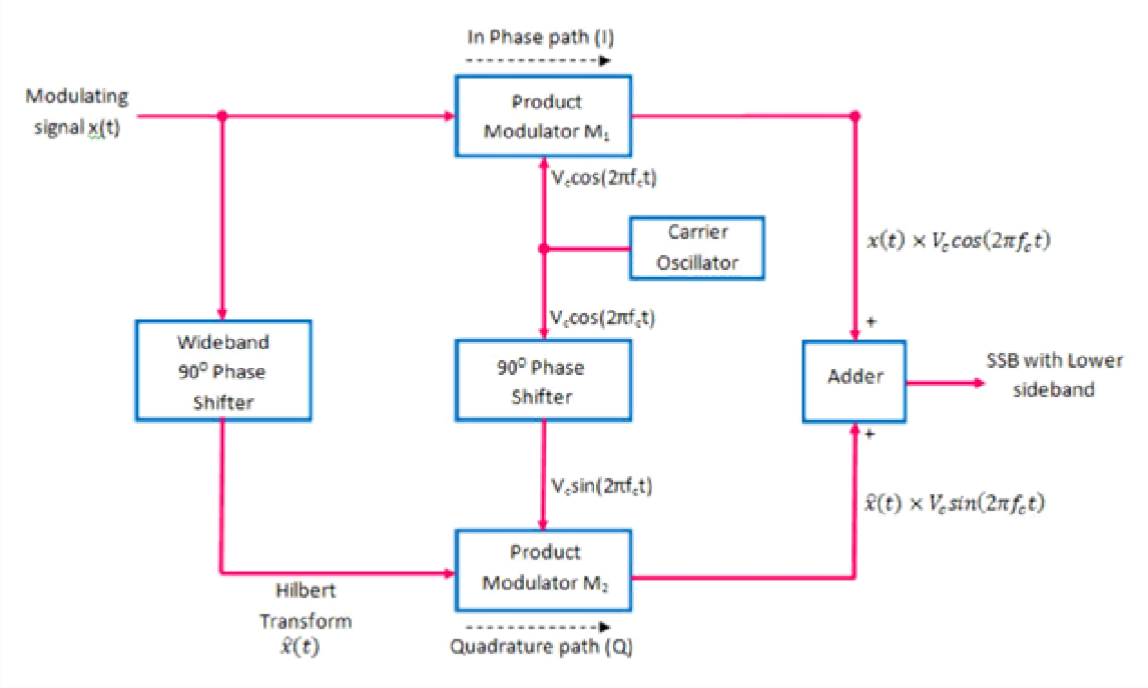
SSB generation using two stage filtering method

Stage 2: In the second stage, a balanced modulator with carrier frequency f2=10 MHz is used for generation of DSB-SC wave. The spectrum of the DSB-SC wave s2(t) appearing at the output of second balanced modulator has a lower sideband occupying the frequency band 9.897 to 9.8997 MHz and the upper sideband occupying 10.1003 to10.1030 MHz. Here again assuming that only USB occupying 10.1003 to10.1030 MHz is selected by filtering using second BPF (transfer function H2) resulting in the desired SSB wave. The separation between the lowest frequency USB and highest frequency LSB is 200.6 kHz in this case. Thus the transition band of second filter is to be adjusted is 200.6 kHz which is approximately 2% of center frequency can be easily designed. Thus by multistage modulation and filtering scheme, a SSB wave with desired carrier wave cane easily designed. This would not have been possible in the single stage scheme. For example in single stage modulation scheme, one requires the transition band of the filter to be adjusted within 600 Hz at a center frequency of 10 MHz, a percentage frequency change of 0.006% of the carrier frequency. Filters such sharp selectivity are extremely difficult to design. Hence multistage modulation and filtering scheme is much useful to generate SSB wave.

Phase Shift Method for the SSB Generation

Figure shows the block diagram for the phase shift method of SSB generation.

This system is used for the suppression of lower sideband. This system uses two balanced modulators M1 and M2 and two 90o phase shifting networks as shown in figure.



Phase shift method for generating SSB signal

Working Operation

The message signal x(t) is applied to the product modulator M1 and through a 90o phase shifter to the product modulator M2 .Hence, we get the Hilbert transform at the output of the wideband 90o phase shifter . The output of carrier oscillator is applied as it is to modulator M1 whereas it s passed through a 90o phase shifter and applied to the modulator M2 .

$$Output\ of\ M_1 = x(t) \times V_c cos(2\pi f_c t)$$

$$and\ \ Output\ of\ M_2 = \hat{x}(t) \times V_c sin(2\pi f_c t)$$

The outputs of M1 and M2 are applied to an adder .

Generation of VSB Modulated wave:

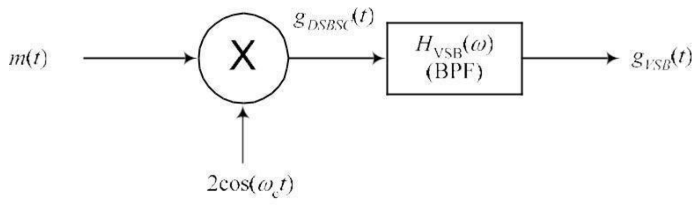
To generate a VSB modulated wave, we pass a DSBSC modulated wave through a sidebandshaping filter.

Comparison of amplitude modulation techniques:

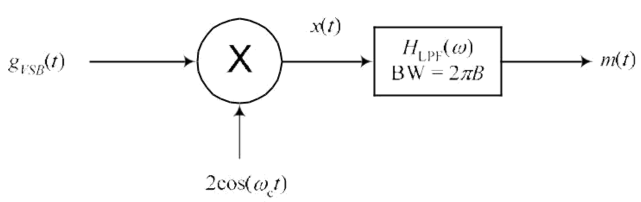
In commercial AM radio broadcast systems standard AM is used in preference to DSBSC or SSB modulation. Suppressed carrier modulation systems require the minimum transmitter power and minimum transmission bandwidth. Suppressed carrier systems are well suited for point –to-point communications. SSB is the preferred method of modulation for long-distance transmission of voice signals over metallic circuits, because it permits longer spacing between the repeaters. VSB modulation requires a transmission bandwidth that is intermediate between that required for SSB or DSBSC. VSB modulation technique is used in TV transmission DSBSC, SSB, and VSB are examples of linear modulation. In Commercial TV broadcasting, the VSB occupies a width of about 1.25MHz, or about one-quarter of a full sideband.

Vestigial Side Band Modulation

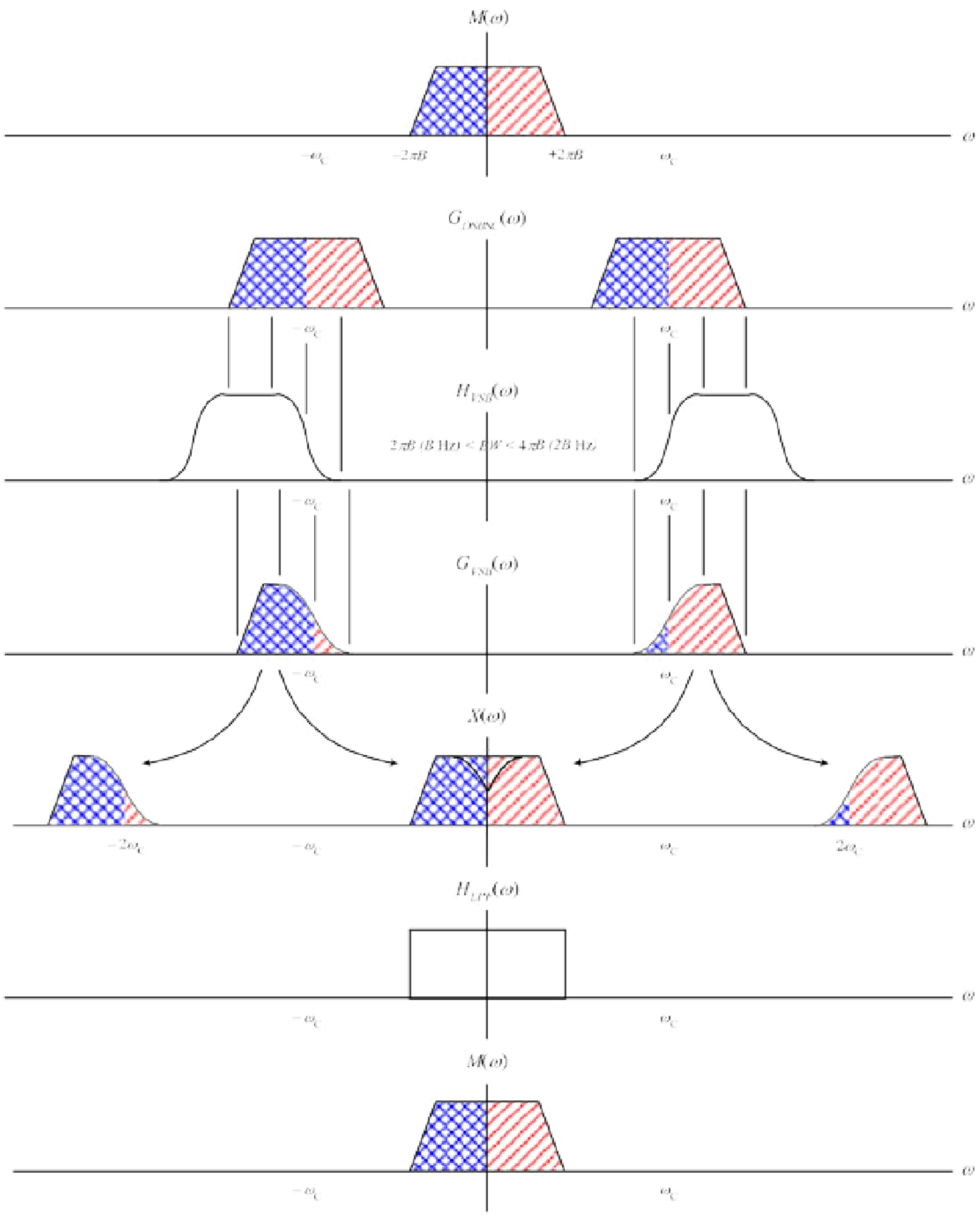
As mentioned last lecture, the two methods for generating SSB modulated signals suffer some problems. The selective–filtering method requires that the two side bands of the DSBSC modulated signal which will be filtered are separated by a guard band that allows the bandpass filters that are used to have non–zero transition band (so it allows for real filters). An ideal Hilbert transform for the phase–shifting method is impossible to build, so only an approximation of that can be used. Therefore, the SSB modulation method is hard, if not impossible, build. A compromise between the DSBSC modulation and the SSB modulation is known as Vestigial Side Band (VSB) modulation. This type of modulation is generated using a similar system as that of the selective–filtering system for SSB modulation. The following block diagram shows the VSB modulation and demodulation.



VSB Modulator (transmitter)



VSB Demodulator (receiver)


M(ω)
−2πB
+2πB
ω
H_VSB(ω)
2πB (B Hz) < BW < 4πB (2B Hz)
G_VSB(ω)
X(ω)
H_LPF(ω)
M(ω)

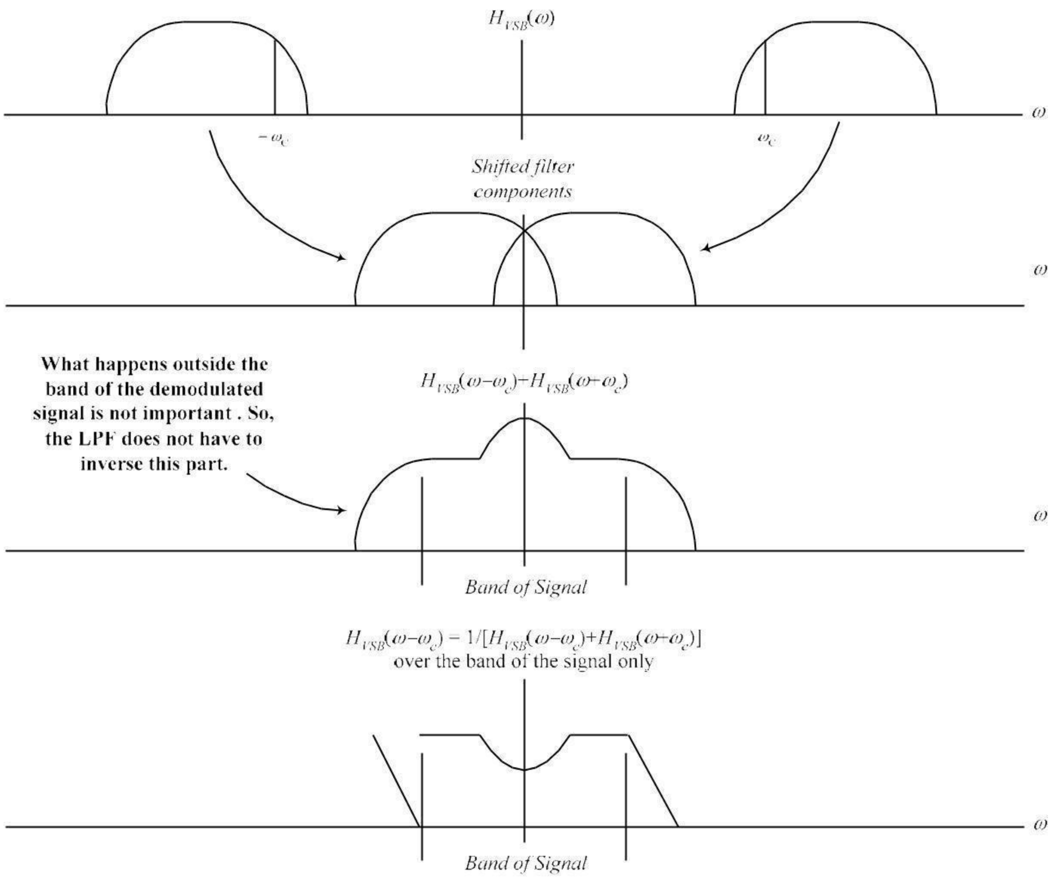
The above example for generating VSB modulated signals assumes that the VSB filter

(HVSB( )) that the transition band of the VSB filter is symmetric in a way that adding the part that remains in the filtered signal from the undesired side band to the missing part of the desired side band during the process of demodulation produces an undusted signal at baseband. In fact, this condition is not necessary if the LPF in the demodulator can take care of any distortion that happens when adding the different components of the bandpass components at baseband. To illustrate this, consider a baseband message signal m(t) that has the FT shown in the following figure. The DSBSC modulated signal from that assuming that the carrier is 2cos(Ct) (the 2 in the carrier is placed there for convenience) is

$$g_{DSBSC}(t)\ m(t)\cos(_{C}t)$$

In frequency–domain, this gives Passing this signal into the VSB filter shown in the modulator block diagram. Note that the VSB filter is not an ideal filter with flat transfer function, so it has to appear in the equation defining the VSB signal. Now, let us demodulate this VSB signal using the demodulator shown above but use a non–ideal filter $H_{LPF}(\ )$ (the carrier here is also multiplied by 2 just for convenience). Passing this through the non–ideal LPF in the demodulator gives an output signal.

For this communication system to not distort the transmitted signal, the output signal must be equal to the input signal. This gives us the following relationship between the LPF at the demodulator and the VSB fitler at the modulator. So, this filter must be a LPF that has a transfer function around 0 frequency that is related to the VSB filter as given above. To illustrate this relationship, consider the following VSB BPF example.


$H_{VSB}(\omega)$
ω
$-\omega_c$
ω_c
Shifted filter components
ω
What happens outside the band of the demodulated signal is not important . So, the LPF does not have to inverse this part.
$H_{VSB}(\omega-\omega_c)+H_{VSB}(\omega+\omega_c)$
ω
Band of Signal
$H_{VSB}(\omega-\omega_c) = 1/[H_{VSB}(\omega-\omega_c)+H_{VSB}(\omega+\omega_c)]$
over the band of the signal only
ω
Band of Signal

AM Transmitter :

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

- High Level
- Low Level

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

High-Level and Low-Level Transmitters

Below figure's show the block diagram of high-level and low-level transmitters. The basic difference between the two transmitters is the power amplification of the carrier and modulating signals.

Figure (a) shows the block diagram of high-level AM transmitter.

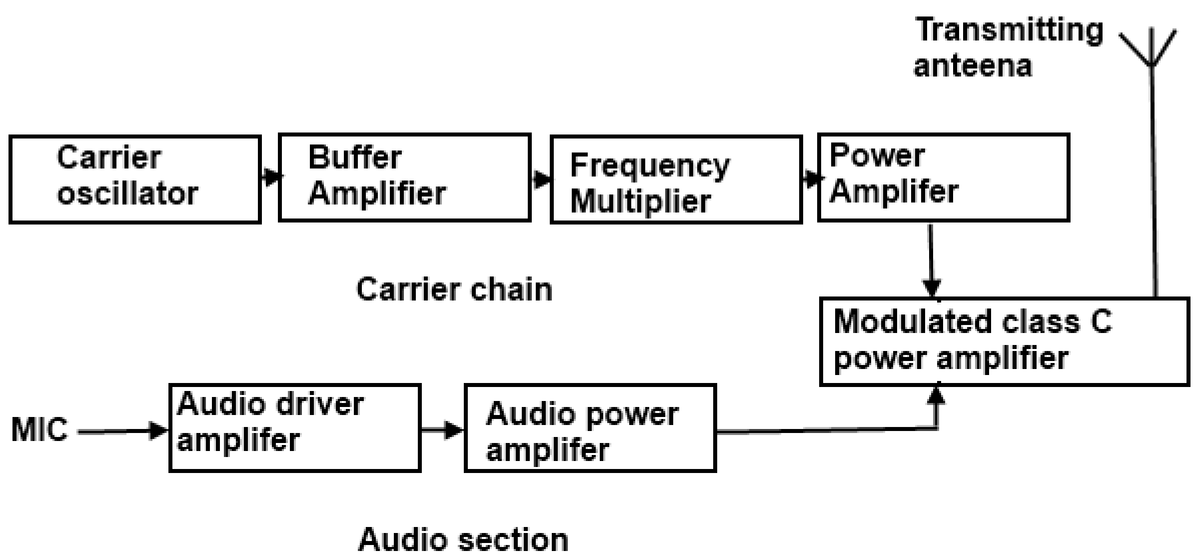


Figure (a) Block diagram of high level AM transmitter

In high-level transmission, the powers of the carrier and modulating signals are amplified before applying them to the modulator stage, as shown in figure (a). In low-level modulation, the powers of the two input signals of the modulator stage are not amplified. The required transmitting power is obtained from the last stage of the transmitter, the class C power amplifier.

The various sections of the figure (a) are:

- Carrier oscillator
- Buffer amplifier
- Frequency multiplier
- Power amplifier
- Audio chain
- Modulated class C power amplifier

Carrier oscillator

The carrier oscillator generates the carrier signal, which lies in the RF range. The frequency of the carrier is always very high. Because it is very difficult to generate high frequencies with good frequency stability, the carrier oscillator generates a sub multiple with the required carrier frequency. This sub multiple frequency is multiplied by the frequency multiplier stage to get the required carrier frequency. Further, a crystal oscillator can be used in this stage to generate a low frequency carrier with the best frequency stability. The frequency multiplier stage then increases the frequency of the carrier to its required value.

Buffer Amplifier

The purpose of the buffer amplifier is two fold. It first matches the output impedance of the carrier oscillator with the input impedance of the frequency multiplier, the next stage of the carrier oscillator. It then isolates the carrier oscillator and frequency multiplier.

This is required so that the multiplier does not draw a large current from the carrier oscillator. If this occurs, the frequency of the carrier oscillator will not remain stable.

Frequency Multiplier

The sub-multiple frequency of the carrier signal, generated by the carrier oscillator , is now applied to the frequency multiplier through the buffer amplifier. This stage is also known as harmonic generator. The frequency multiplier generates higher harmonics of carrier oscillator frequency. The frequency multiplier is a tuned circuit that can be tuned to the requisite carrier frequency that is to be transmitted.

Power Amplifier

The power of the carrier signal is then amplified in the power amplifier stage. This is the basic requirement of a high-level transmitter. A class C power amplifier gives high power current pulses of the carrier signal at its output.

Audio Chain

The audio signal to be transmitted is obtained from the microphone, as shown in figure (a). The audio driver amplifier amplifies the voltage of this signal. This amplification is necessary to drive the audio power amplifier. Next, a class A or a class B power amplifier amplifies the power of the audio signal.

Modulated Class C Amplifier

This is the output stage of the transmitter. The modulating audio signal and the carrier signal, after power amplification, are applied to this modulating stage. The modulation takes place at this stage. The class C amplifier also amplifies the power of the AM signal to the reacquired transmitting power. This signal is finally passed to the antenna., which radiates the signal into space of transmission.

Figure shows the block diagram of a low-level AM transmitter.

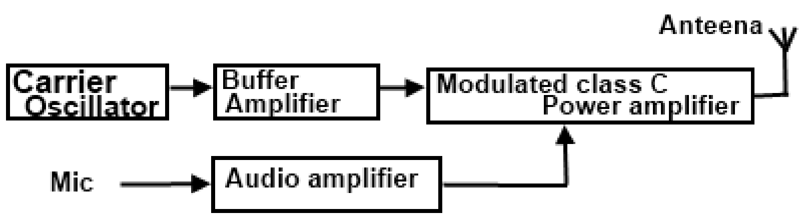


Figure (b) Block diagram of Low-level AM transmitter

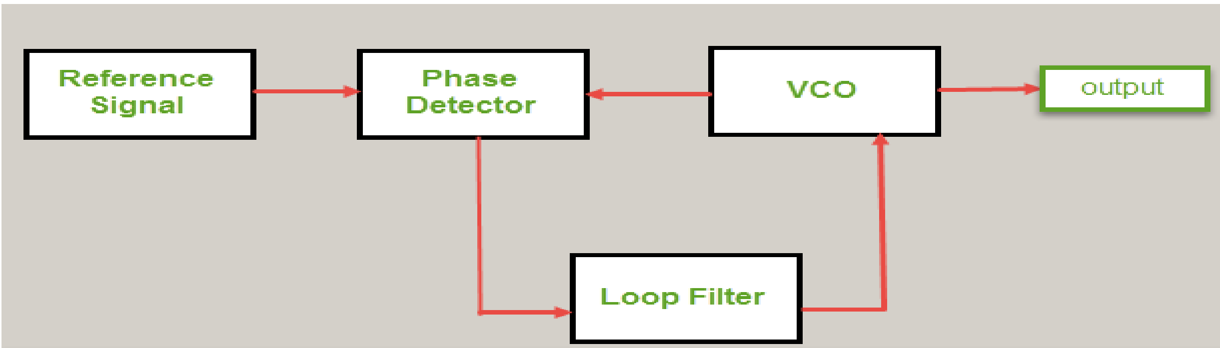
The low-level AM transmitter shown in the figure (b) is similar to a high-level transmitter, except that the powers of the carrier and audio signals are not amplified. These two signals are directly applied to the modulated class C power amplifier.

Modulation takes place at the stage, and the power of the modulated signal is amplified to the required transmitting power level. The transmitting antenna then transmits the signal.

COSTAS PLL DETECTOR:

A phase-locked loop consists of a phase detector and a voltage controlled oscillator. The output of the phase detector is the input of the voltage-controlled oscillator (VCO) and the output of the VCO is connected to one of the inputs of a phase detector which is shown below in the basic block diagram. When these two devices are feed to each other the loop forms.

Block Diagram And Working Principle Of PLL



BASIC DIAGRAM OF PHASE LOCKED LOOP

The phase-locked loop consists of a phase detector, a voltage controlled oscillator and, in between them, a low pass filter is fixed. The input signal 'Vi' with an input frequency 'Fi' is conceded by a phase detector. Basically the phase detector is a comparator that compares the input frequency fi through the feedback frequency fo. The output of the phase detector is (fi+fo) which is a DC voltage. The out of the phase detector, i.e., DC voltage is input to the low pass filter (LPF); it removes the high-frequency noise and produces a steady DC level, i.e., Fi-Fo. The Vf is also a dynamic characteristic of the PLL. The output of the low pass filter, i.e., DC level is passed on to the VCO. The input signal is directly proportional to the output frequency of the VCO (fo). The input and output frequencies are compared and adjusted through the feedback loop until the output frequency is equal to the input frequency. Hence, the PLL works like free running, capture, and phase lock.

When there is no input voltage applied, then it is said to be a free-running stage. As soon as the input frequency applied to the VOC changes and produces an output frequency for comparison, it is called a capture stage. The below figure shows the block diagram of the PLL.

Phase-Locked Loop Detector

The phase-locked loop detector compares the input frequency and the output frequency of the VCO to produces a DC voltage which is directly proportional to the phase distinction of the two frequencies. The analog and digital signals are used in the phase-locked loop. Most of the monolithic PLL integrated circuits use an analog phase detector and the majority of phase detectors are from the digital type. A double balanced mixture circuit is used commonly in analog phase detectors.

UNIT - 3

ANGLE (FM + PM) MODULATION AND DEMODULATION

Angle Modulation:

Angle modulation is the process of varying the total phase angle of a carrier wave in accordance with the instantaneous value of the modulating signal, keeping amplitude of the carrier constant.

Two types:

1. Frequency modulation
2. Phase modulation

Frequency modulation has an important advantage over amplitude modulation ie interference due to noise is reduced in freq. modulation.

However, this advantage of noise - immunity is at the cost of increased bandwidth and hence comparitively a less number of channels can be accomodate in a given frequency space.

The angle modulated wave is mathematically expressed as,

$$e(t) = E_c \sin(\omega_c t + \theta_0)$$

where,

$E_c \Rightarrow$ amplitude of carrier

$\theta_0 \Rightarrow$ phase angle

phase is direct function of modulating signal ie,

$$\theta_0 \propto e_m(t)$$

$$\boxed{\varphi = \omega_c t + \theta_0} \quad \text{——} \; ①.$$

$$e_c(t) = E_c \sin\varphi$$

To find carrier frequency, ω_c differentiate eqn ①

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

This derivative $\frac{d\varphi}{dt}$ is constant with time for an unmodulated carrier

But in general, this derivative may not be constant with time, rather it may vary in time. This time dependent angular velocity is called instantaneous angular velocity and is denoted by ω_i

$$\therefore \frac{d\varphi}{dt} = \omega_i$$

$$\varphi = \int \omega_i \, dt$$

The time dependent angular velocity ω_i of the phasor φ provides a time varying instantaneous frequency of the carrier wave. This implies that the frequency of the carrier wave changes from one cycle to another

Mathematical representation of FM single tone modulation.

Let the message signal,

$$e_m(t) = E_m \cos \omega_m t$$

Carrier signal,

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

Frequency modulated signal is given by,

$$e_{FM}(t) = E_c \sin \varphi_i$$

where,

$$\varphi_i = (\omega_c t + \theta)$$

After frequency modulation, the frequency of the carrier signal is changed in accordance with the instantaneous amplitude of message signal.

∴ The frequency of the carrier after modulation is given as,

$$\omega_i = \omega_c + k E_m \cos \omega_m t.$$

∴ Phase angle of the modulated signal,

$$\varphi_i = \int \omega_i \, dt.$$

$$= \int [\omega_c + k E_m \cos\omega_m t]\, dt$$

$$\varphi_i = \omega_c t + \frac{k E_m}{\omega_m} \sin\omega_m t$$

$$\therefore\ e_{FM}(t) = E_c \sin\varphi_i$$

$$= E_c \sin\left[\omega_c t + \frac{k E_m}{\omega_m} \sin\omega_m t\right]$$

$$e_{FM}(t) = E_c \sin[\omega_c t + m_f \sin\omega_m t]$$

where $m_f = \dfrac{k E_m}{\omega_m}$ = modulation index of FM

Modulation index:

Modulation index of FM can be defined as the ratio of maximum freq deviation to the modulating frequency

$$m_f = \frac{\omega_d}{\omega_m}.$$

The maximum and minimum value of cosine term is ± 1

$\therefore$ Maximum value of angular frequency.

$$\omega_{max} = \omega_c + k E_m$$

Minimum value of angular frequency.

$$\omega_{min} = \omega_c - k E_m$$

Freq. deviation

$$\omega_d = \omega_{max} - \omega_c = \omega_c - \omega_{min}$$

$$= kE_m.$$

$$\therefore m_f = \frac{kE_m}{\omega_m} = \frac{\delta}{\omega_m}$$

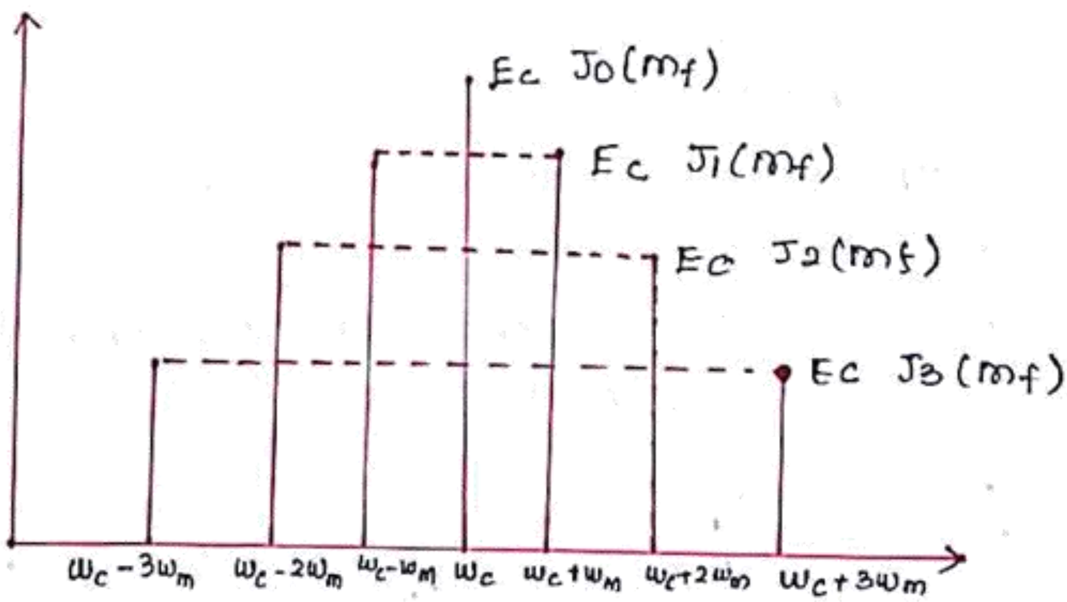
$$e_{FM}(t) = E_c \sin(\omega_c t + m_f \sin\omega_m t)$$

$$e_{FM}(t) = E_c \sin\omega_c t \cos(m_f \sin\omega_m t) + \cos\omega_c t \sin(m_f \sin\omega_m t)$$

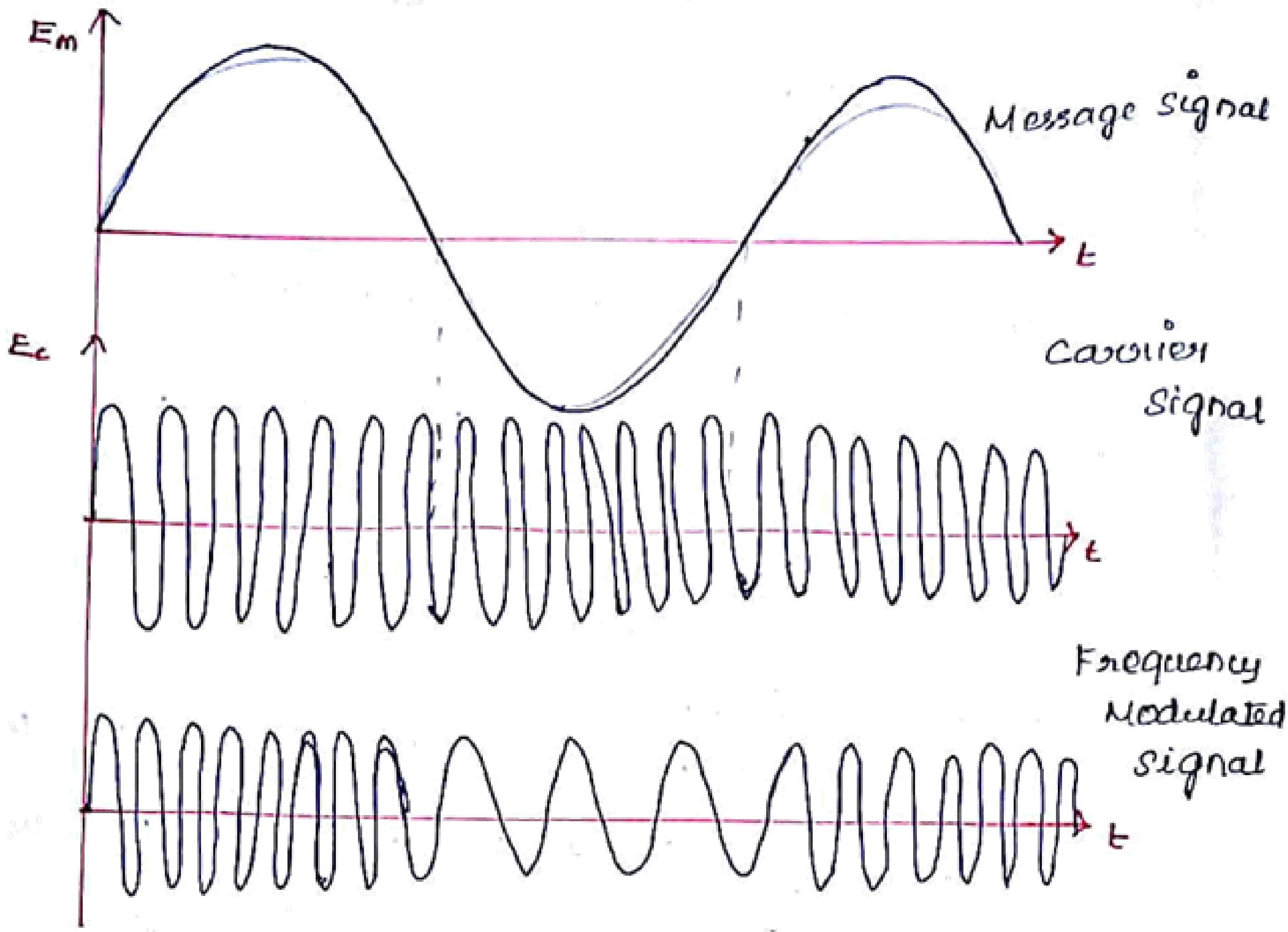
Using Bessel function, the above equation becomes,

$$= E_c [J_0(m_f) \sin\omega_c t + J_1(m_f) [\sin(\omega_c + \omega_m)t - \sin(\omega_c - \omega_m)t] + E_c [J_2(m_f)] [\sin(\omega_c + 2\omega_m)t - \sin(\omega_c - 2\omega_m t) + E_c [J_3(m_f)] [\sin(\omega_c + 3\omega_m)t - \sin(\omega_c - 3\omega_m)t]$$

Frequency spectrum of FM :



Graphical representation of FM Wave :



FM signal has the following components :

→ carrier term $\sin \omega_c t$ with magnitude $E_c J_0(m_f)$ i.e, magnitude of the carrier term is reduced by a factor $J_0(m_f)$.

Side band :

→ Theoretically infinite number of side bands are Produced and the amplitude of each sideband is decided by the corresponding Bessel function $J_n(m_f)$

Bandwidth :

→ the presence of infinite number of side bands takes the bandwidth of the FM signal infinite.

Power content in FM signal :

→ since the amplitude of FM remains unchanged the power of the FM signal is same as that of unmodulated carrier .

Mathematical Representation of FM [Multitone Modulation]:

→ Modulation done with more than one message signal is called multitone modulation.

Let us consider the message signal as

$$e_m = E_{m1} \cos \omega_1 t + E_{m2} \cos \omega_2 t$$

Carrier signal,

$$e_c = E_c \cos \omega_c t .$$

After Frequency Modulation, the frequency of the modulated signal is

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

$$\omega_i = \omega_c + K E_{m1} \cos \omega_1 t + K E_{m2} \cos \omega_2 t .$$

If $2\pi \Delta f_1 = K E_{m1}$, $2\pi \Delta f_2 = K E_{m2}$.

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

Phase angle $\theta_i = \int \omega_i \, dt$.

$$= \int [\omega_c + 2\pi \Delta f_1 \cos \omega_1 t + 2\pi \Delta f_2 \cdot \cos \omega_2 t] \, dt$$

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

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

Instantaneous amp.. of modulated signal,

$$e_{fm}(t) = E_c \sin \phi_i$$

$$= E_c \sin \left[\omega_c t + \frac{\Delta f_1}{f_1} \sin \omega_1 t + \frac{\Delta f_2}{f_2} \sin \omega_2 t\right]$$

$$= E_c \sin \left[\omega_c t + m_{f_1} \sin \omega_1 t + m_{f_2} \sin \omega_2 t\right].$$

if $\alpha_1 = m_{f_1} \sin \omega_1 t$; $\alpha_2 = m_{f_2} \sin \omega_2 t$.

$$e_{fm}(t) = E_c \sin \left[\omega_c t + (\alpha_1 + \alpha_2)\right]$$

$$\boxed{e_{fm}(t) = E_c \left[\sin \omega_c t . \cos (\alpha_1 + \alpha_2) + \cos \omega_c t . \sin (\alpha_1 + \alpha_2)\right]}$$

Mathematical representation of phase modulation :

→ phase modulation is defined as the process by which changing the phase of the carrier signal in accordance with the instantaneous amplitude of the message signal.

→ Amplitude and frequency remains constant.

Let the modulating signal be

$$e_m(t) = E_m \cos \omega_m t$$

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

After phase modulation,

$$e_{PM}(t) = E_c \sin (\omega_c t + \theta)$$

where θ is the phase angle of the carrier,

$$\boxed{\theta = K E_m \cos \omega_m t}.$$

$$e_{PM}(t) = E_c \sin[\omega_c t + K E_m \cos \omega_m t]$$

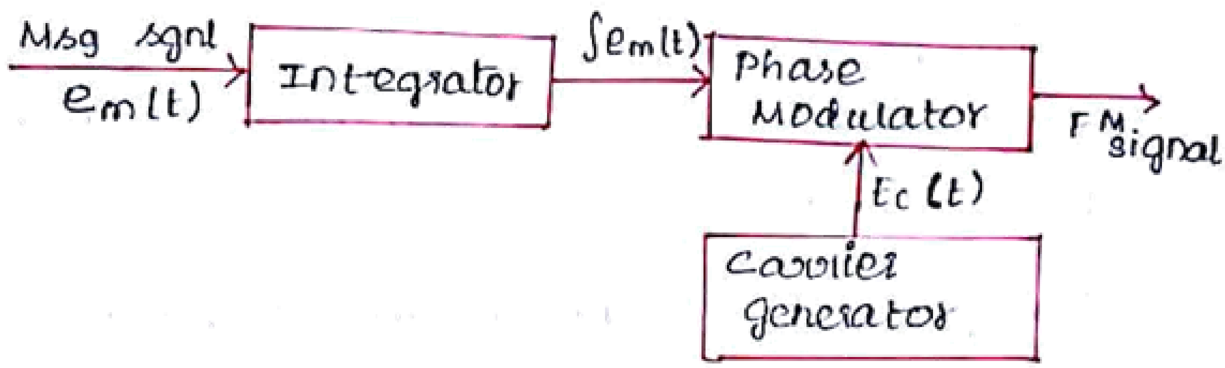
$$\boxed{e_{PM}(t) = E_c \sin[\omega_c t + m_p \cos \omega_m t]}$$

where,

m_p = modulation index of phase modulation.

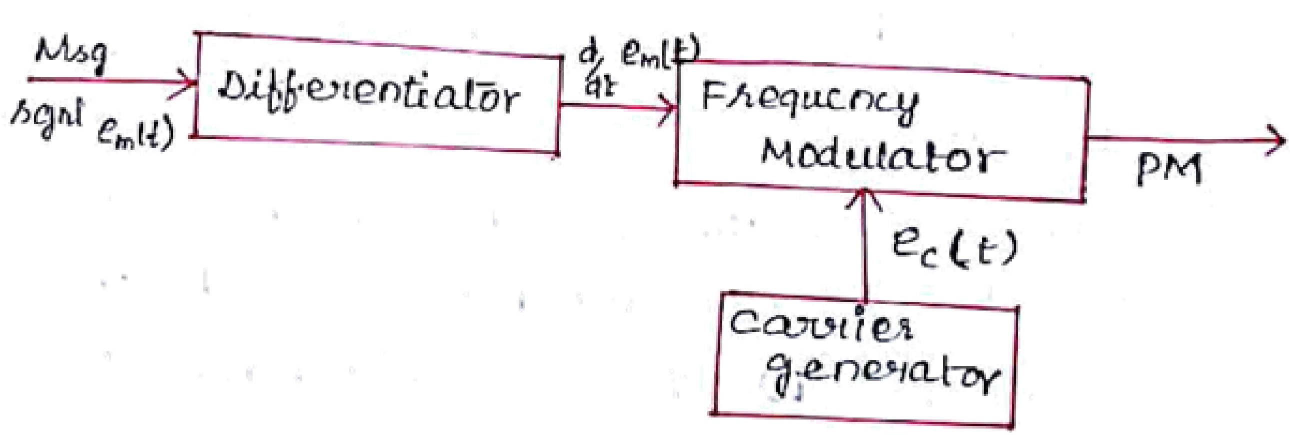
$m_p = K E_m$.

FM Generation using Phase Modulator :



→ We can get FM by using PM, provided that at first the modulating signal is integrated and there applied to the phase modulator.

PM Generation using Frequency Modulator :



→ We can generate a PM wave using frequency Modulator provided that msg sgnl is first differentiated and then applied to the frequency modulator.

Types of Frequency Modulation :

→ The Bandwidth of FM signal depends on the Modulation index.

→ If the deviation is high, the band width will be large.

→ If the deviation is small, the band width will be small.

→ Depending on the value of modulation, then we can divide FM into two categories,

1. Narrow Band FM
2. Wide band FM.

→ Narrow Band FM : When m_f is small, the bandwidth of FM is narrow.

→ Wide Band FM : When m_f is large, the FM signal has a wide bandwidth.

Bandwidth :

→ The Bandwidth of a narrow band FM is same as that of AM, which is twice the baseband.

→ The bandwidth of a wideband FM is too large ideally infinite.

Narrow Band FM :

Let the message signal be,

$$e_m(t) = E_c \cos W_m t$$

Carrier signal be,

$$e_c(t) = E_c \cos W_c t$$

After Frequency Modulation,

$$e_{FM}(t) = E_c \sin(W_c t + \theta)$$

$$w_i = W_c + K E_M \cos W_m t$$

Freq. deviation, $2\pi \Delta f = K E_M$

$$w_i = W_c + 2\pi \Delta f \cos W_m t$$

$$Q_i = \int w_i \, dt.$$

$$= \int (W_c + 2\pi \Delta f \cos W_m t) \, dt$$

$$Q_i = W_c t + \frac{2\pi \Delta f}{W_m} \sin W_m t$$

$$Q_i = W_c t + \frac{\Delta f}{f_m} \sin W_m t.$$

$$e_{FM}(t) = E_c \sin Q_i$$

$$= E_c \sin\left[W_c t + \frac{\Delta f}{f_m} \sin W_m t\right]$$

$$e_{FM}(t) = E_c \sin[W_c t + m_f \sin W_m t]$$

$$\boxed{\begin{aligned} e_{FM}(t) = {} & E_c \sin W_c t \cdot \cos(m_f \sin W_m t) + \\ & E_c \cos W_c t \cdot \sin(m_f \sin W_m t) \end{aligned}}$$

For narrow band FM, the modulation index m_f is small compared to one radian,

$$\therefore \cos(m_f \sin \omega_m t) = 1$$

$$\sin(m_f \cdot \sin \omega_m t) = m_f \sin \omega_m t .$$

$$\boxed{\therefore e_{FM}(t) = E_c \sin \omega_c t + E_c \cos \omega_c t \, (m_f \sin \omega_m t)}$$

The above eqn is similar to eqn. of AM.

If the modulation index is high, then it is called wideband FM.

Carson's rule :

Carson's rule gives approximate minimum bandwidth of angle modulated signal as,

$$\text{Bandwidth} = 2\left[\delta + f_m(max)\right] Hz .$$

Here $f_m(max)$ is the maximum modulating frequency.

Generation of FM :

Frequency Modulated signals can be Generated in two ways

1. Direct FM
2. Indirect FM

Direct FM :

→ In this Method, the frequency of the carrier is varied directly by the modulating signal, That is the instantaneous frequency deviation is directly proportional to amplitude of the modulating signal.

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

<u>Varactor diode Modulator</u> :

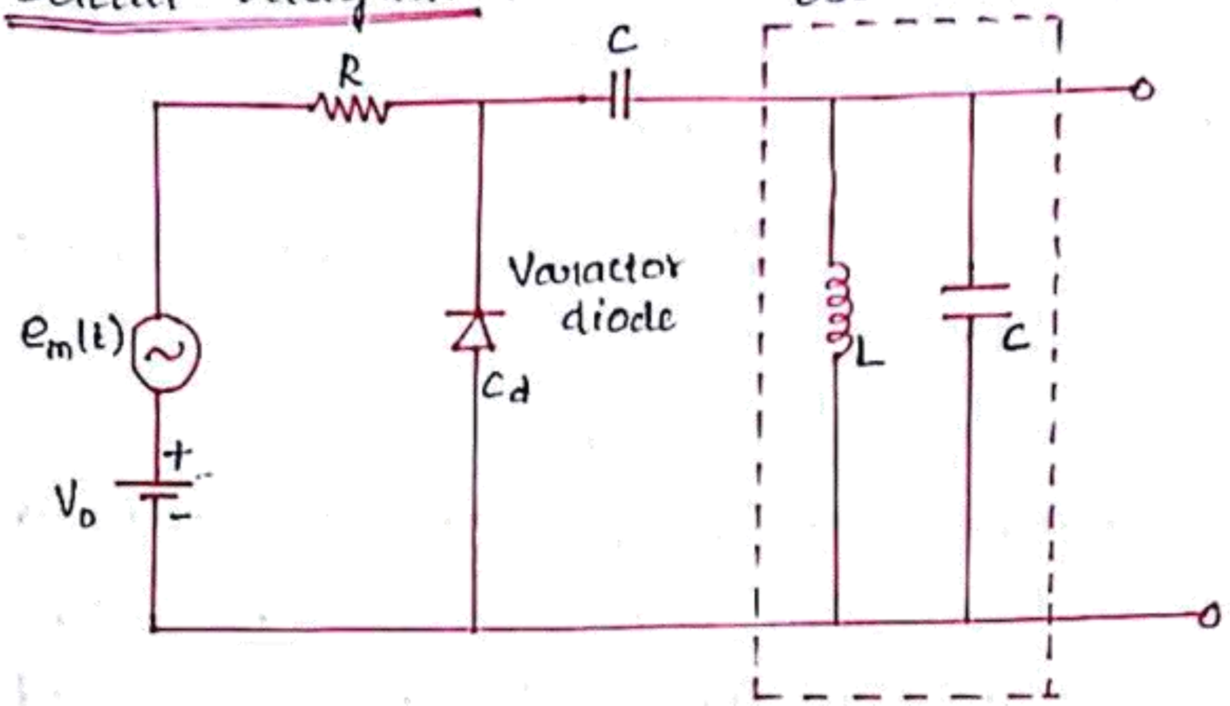
Varactor diode means variable capacitor, diode. The varactor diode is a specially fabricated PN Junction diode which is used as a variable capacitor in the reverse biased condition.

It is used as a variable reactance and is placed across the tank circuit,

When the tank circuit is tuned, the L or C changed with Modulating voltage.

The larger the departure of the modulating voltage from zero, the larger the reactance variation and hence larger the frequency Variation.

circuit diagram :



→ The varactor diode is connected across the resonant circuit of the oscillator through a coupling capacitor of relatively large value.

→ This coupling capacitors isolates the varactor diode from the oscillator as far as DC is concerned and provide an effective short circuit at the operating frequencies.

→ The Modulating signal is FED in series with the, dc supply and at any instant, the effective bias to the varactor diode equals the algebraic sum of the dc bias volt V and instantaneous value of the modulating sgl.

→ As a result, the capacitance changes with the amplitude of the modulating signal resulting in frequency Modulation of the oscillator output.

Advantages of Direct Method :

→ The direct method is used for high power FM generation in many applications.

Limitation :

→ In the direct method it is difficult to obtain a high order stability in carrier frequency.

→ This is because the carrier generation is directly affected by the modulating signal.

→ The modulating signal directly controls the tank circuit of the carrier generator and hence a stable oscillator like the crystal oscillator, cannot be used. (The crystal oscillator provides a stable but fixed freq).

→ Thus carrier generation cannot be of high stability which is an essential requirement.

Remedy :

→ A remedy of this problem is the indirect method of FM generation.

→ In this method, the carrier oscillator is not required to respond to the modulating signal directly, rather the carrier generation is isolated from other parts of the circuit

→ Hence stable crystal oscillations can be used for generating the carrier signal.

Indirect Method of FM Generation :

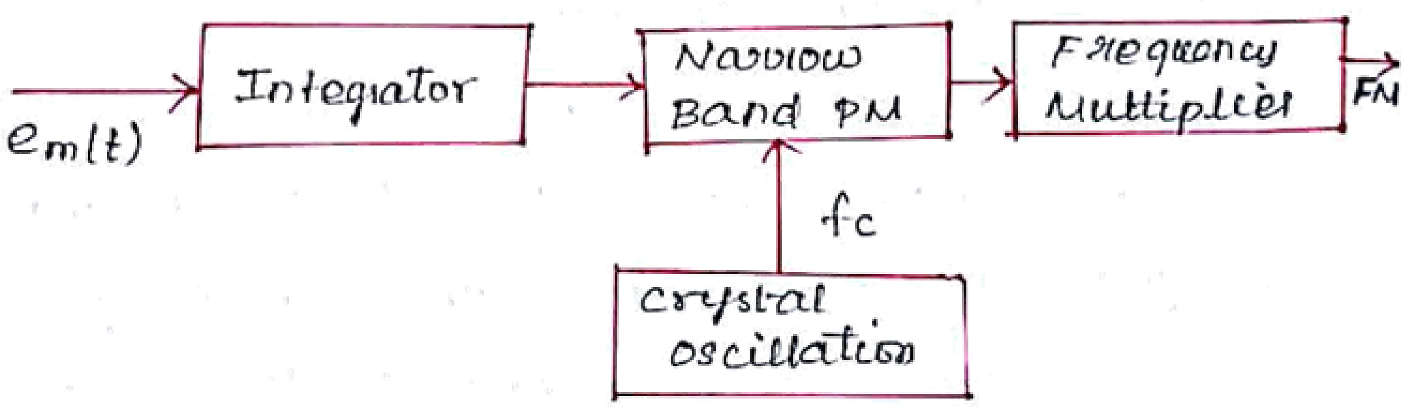
Armstrong Method :

→ The basic principle of this method is the generator a narrowband FM indirectly by using the phase modulation technique and then converting this NBFM to a wideband FM.

→ The distortion is low in NBFM as the modulation index is small.

→ The multiplier circuit, apart from multiplying the carrier frequency, also increases the frequency deviation and thus the NBFM is converted into WBFM.

Armstrong Method :



FM Transmitter :

The Frequency Modulated wave can be produced by two methods namely,

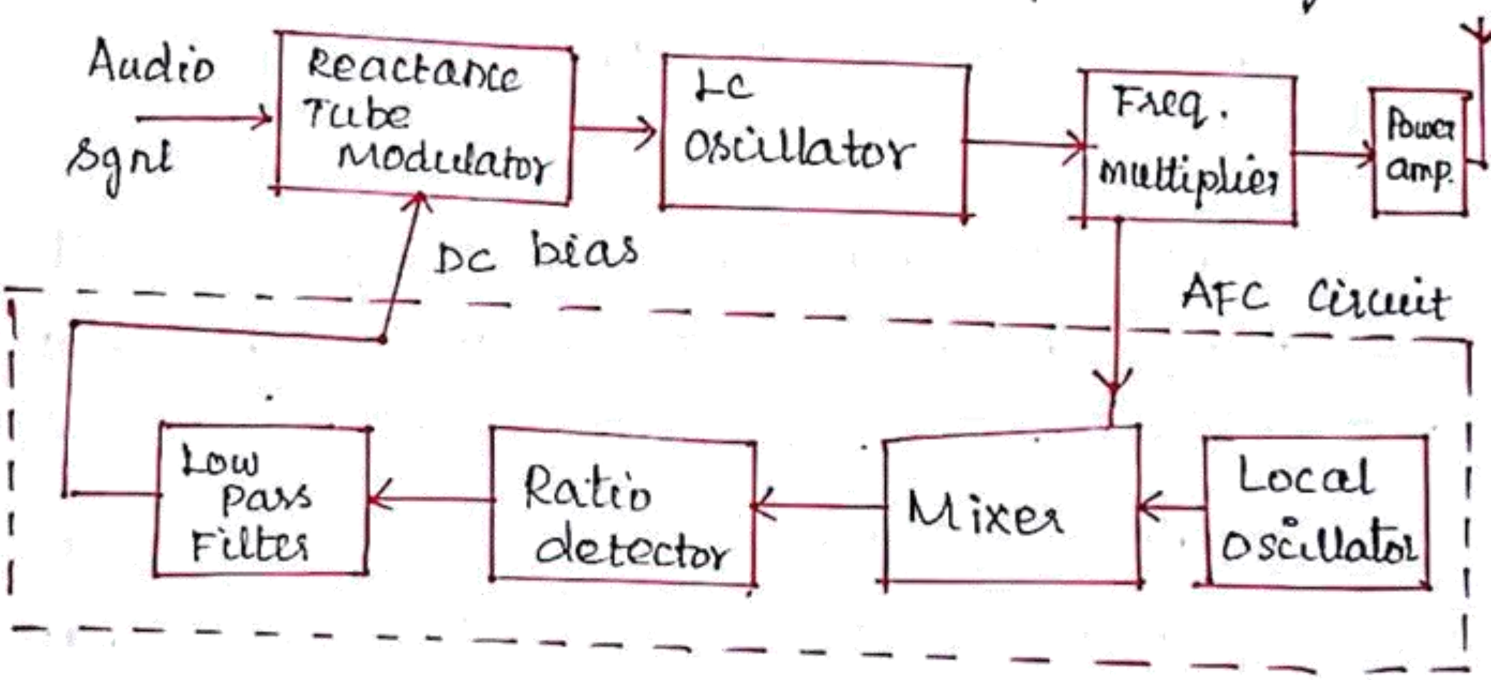
1. Directly Modulated FM Transmitter
2. Indirectly Modulated FM Transmitter

Directly Modulated FM Transmitter :

→ In this method, the modulating system directly produce FM waves by varying the master oscillator frequency such circuit employ LC circuits in Master oscillator circuits.

→ The oscillator frequency is likely to change with changes in circuit parameters.

→ It produces more frequency deviation and requires less number of freq. multiplier stages.



→ The transmitter employs a reactance tube Modulator to produce a frequency deviation in proportion to the signal amplitude.

→ The resulting FM is passed through a number of freq. multiplier stages.

→ The Modulated wave is then amplified to the required powers level by class 'C' Power amplifier stages and then transmitted through antenna.

→ A part of the output of the freq. multiplier stages is passed to AFC circuit.

AFC circuit :

→ The automatic Frequency correction is incorporated in FM transmitter to keep carrier frequency stable.

→ The discriminator reacts only to small change in the carrier frequency but not to freq. deviations in the carrier.

→ Suppose frequency of the carrier increases This higher frequency is fed to the mixer for which the other input frequency is from the stable crystal oscillator.

→ It is fed to discriminator which is tuned to the correct frequency difference which should exist between the LC oscillator and crystal oscillator,

→ The discriminator will develop a positive dc voltage.

→ This voltage is applied to the reactance modulator whose g_m is increased by the +ve voltage. This increased thereby decreasing the oscillator frequency.

→ The frequency increases in the carrier frequency is thus lowered and brought to the correct value.

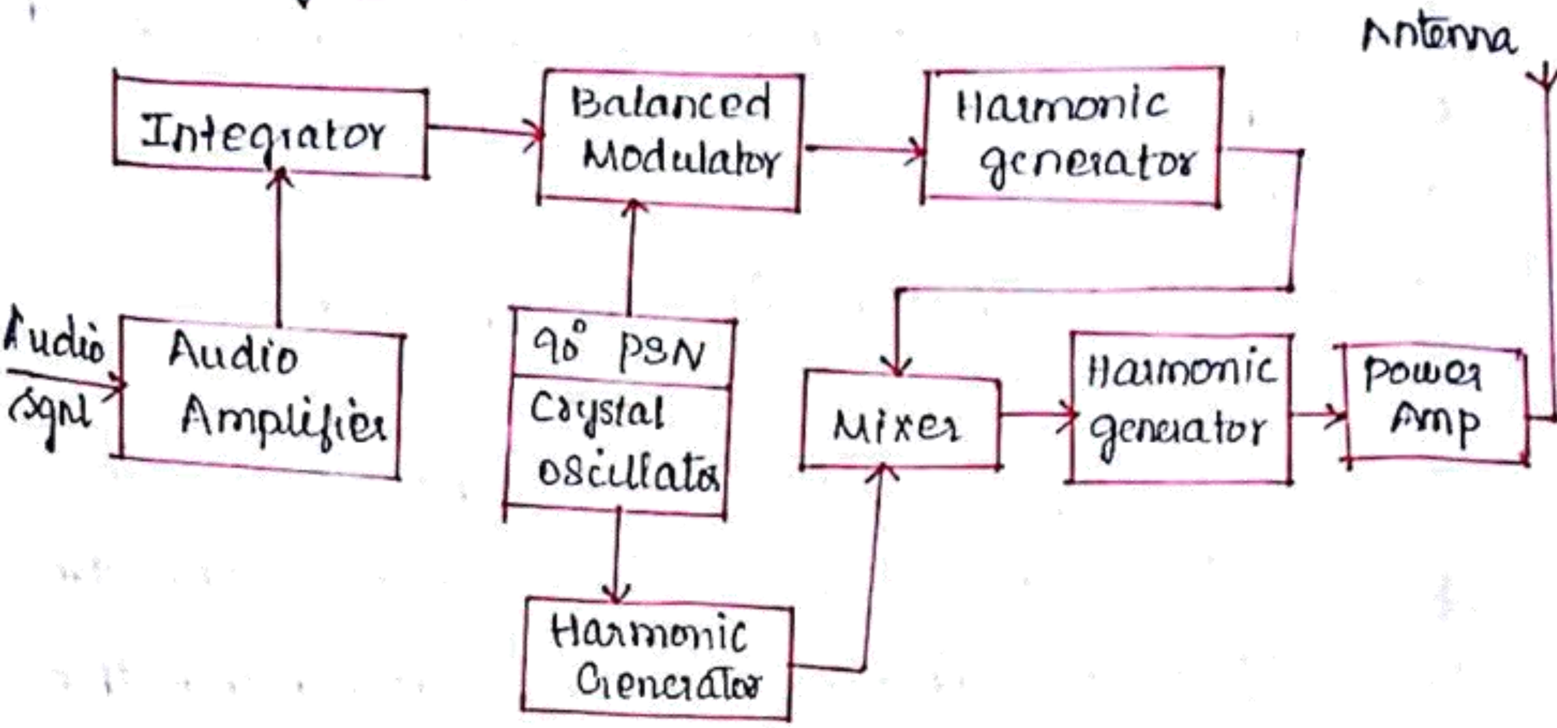
Indirectly Modulated FM Transmitter:

→ The transmitter equipment may contain a crystal oscillator which is phase modulator by the audio signals.

→ This tx gives a drift free frequency.

→ The phase modulator circuit produces smaller freq. deviation and requires more number of frequency multiplier stages.

Armstrong Method of FM Transmitter :



→ The Carrier frequency is generated with the help of crystal oscillator

→ This method utilizes a balanced Modulator with audio and carrier signal after 90° phase shift.

→ The balanced output gives DSB-SC AM.

→ The frequency of the sidebands is increased in a harmonic generator stages and fed to mixer stages the other input to this stage being the carrier signal after passing through another harmonic generator.

→ The different freq. components at the mixer output are the carrier & side band frequencies.

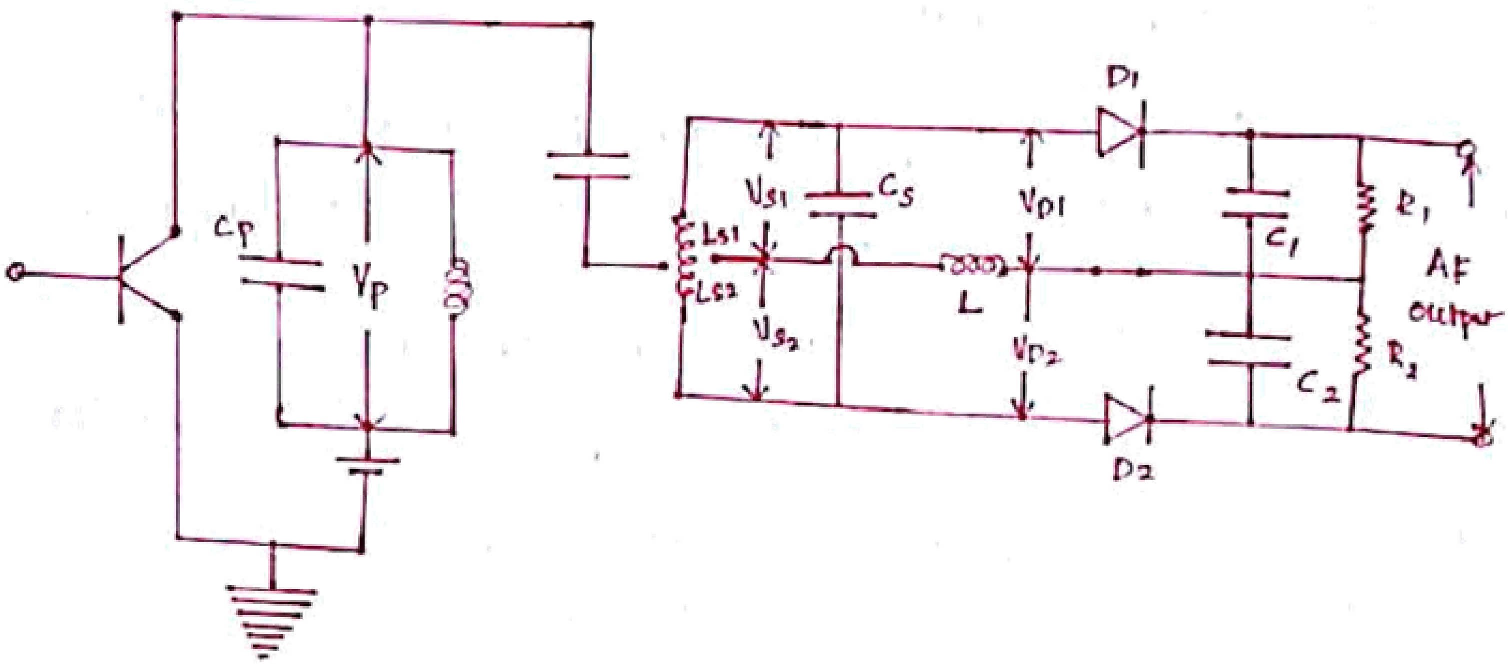
→ This output is again multiplied by a no. of freq. multiplier stages, raised to the required Power level and then it is transmitted.

Comparison with linear Modulation System :

Amplitude Modulation	Frequency Modulation
1. Amplitude of the Carrier is varied according to the amplitude of modulating signal.	Frequency of the carrier is varied according to amplitude of the modulating signal.
2. AM Broadcast operates in MF and HF range	FM Broadcasts operates in UHF and VHF range.
3. AM has poor fidelity due to narrow bandwidth	since the bandwidth is large, fidelity is better.
4. Noise interference is more	Noise interference is minimum.
5. Adjacent channel interference is present	Adjacent channel interference is avoided due to wide bandwidth
6. In AM only carrier and two sidebands are present	Infinite number of sidebands are present.

7.	Transmitted power varies according to the modulation index	Transmitted power remains constant irrespective of modulation index.
8.	Depth of modulation have limitation. It cannot be increased above 1.	Depth of modulation have no limitation. It can be increased by increasing freq. deviation.

Foster - Seeley Discriminator :



→ Foster seeley discriminator circuit consists of two identical diode detector circuits Ls1, D1, R_1, C_1 and Ls2, D2, C_2, R2.

→ The FM signal is inductively coupled to the secondary.

→ The input to diode D1 equals the vector sum of the primary voltage Vp, and the voltage

V_{S1} developing across the winding L_{S1}.

→ Similarly, the input to the diode D_2, is the vector sum of V_p and V_{S2} developing across L_{S2}.

→ The secondary voltages are 180° out of phase with primary voltage V_p.

→ Because of transformer action, there is an induced voltage V_s. Which causes a current. I_s in the secondary circuit.

→ The current I_s develops voltages V_{S1} and V_{S2} across the two halves of the secondary.

→ These voltages when measured with respect to tap point are 180° out of phase with one another.

Case (1) When $f_{in} = f_c$:

When the incoming signal has a frequency equal to f_c,

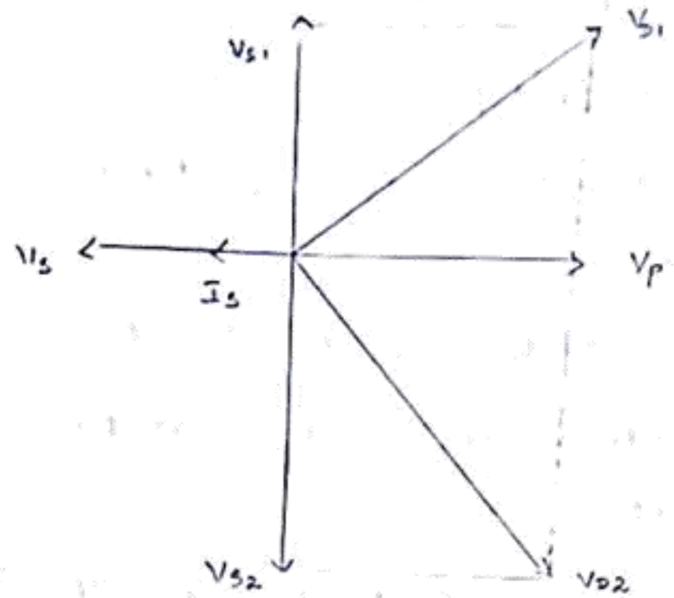
→ primary and secondary voltages are 90° out of phase.

→ V_s and I_s are in phase.

→ V_{S1} and V_{S2} lag by 90° but are 180° out of phase with respect to centre point.

→ The voltages V_{D1} and V_{D2} are equal. This makes rectified voltages across C_1R_1 and C_2R_2 equal.

→ The A.F output is zero.



Case (2) $f_{in} > f_c$:

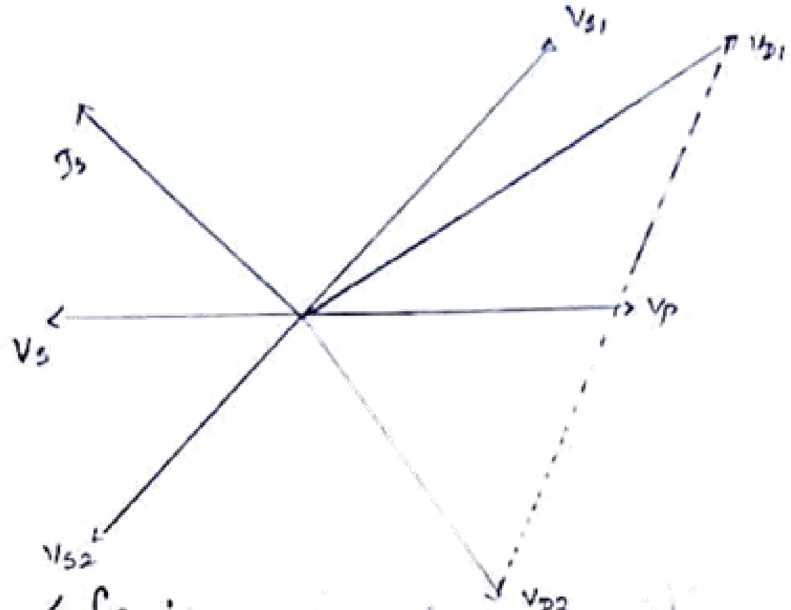
When incoming signal frequency is greater than fc, the secondary circuit acts as inductive.

→ The Primary and secondary voltages are less than 90° out of phase.

→ Is lags V_s by an angle less than 90°.

→ Input Voltage V_{D1} to diode D_1 becomes greater than V_{D2}.

→ The AF output becomes positive :



Case (3) $f_{in} < f_c$:

→ When incoming signal frequency is less than f_c, the secondary circuit behaves as capacitive

→ The primary and secondary voltages are more than 90° out of phase.

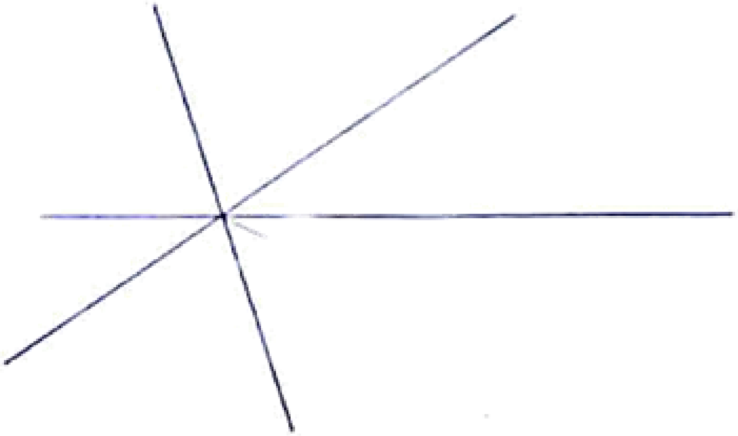
→ I_S leads V_S by angle that lies between 0° and 90°.

→ Vector sum of V_P and V_{S1} which gives the input to diode D_1 is smaller than the vector sum of V_P and V_{S2}.

→ This gives a larger voltage output to diode D_2 compared to D_1.

→ This results in greater DC voltage across R_2C_2 compared to R_1C_1.

→ The resultant output is negative.



→ Thus, the circuit is capable of converting frequency variations into amplitude variations.

Advantages:

The phase discriminator is much easier to align than the balanced slope detector.

→ only two tuned circuits are necessary and both are tuned to same frequency.

→ Linearity is better, because the device relies less upon the frequency response and more on the primary - secondary phase relations which is quite linear.

Disadvantages:

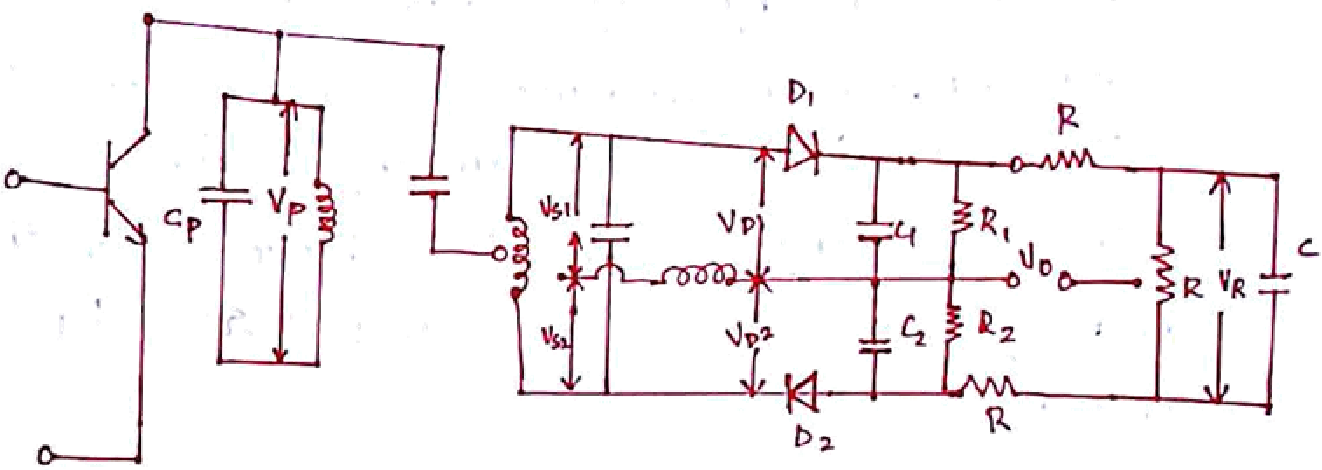
→ The effect of variation in the input signal amplitude is to vary the amplitude of the output correspondingly and the Foster seeley discriminator is sensitive to input signal amplitude variations.

→ To avoid this, circuit is always proceded by a limiter stage which makes the amplitude of FM signal free from amplitude variations.

→ It needs a seperate amplitude limiting circuit.

Ratio Detector :

Circuit diagram :



→ Ratio detector closely resembles the Foster - seeley discriminator in certain respects.

→ The input circuits of the ratio detectors and the foster - seeley discriminator are similar but the connections of the diode D_2 are reversed in ratio detector.

→ As a result, the rectified voltages of D_1 and D_2 becomes additive.

→ A large capacitor C_o is placed across resistor R_1 and R_2. This capacitor

→ The output voltage V_0 is taken across the terminal t_1, t_2.

$$V_0 = V_{t_1,t_2} = |V_{02}| - \left|\frac{V_R}{2}\right|$$

$$V_R = V_{01} + V_{02}$$

$$V_0 = V_{02} - \frac{V_{01} + V_{02}}{2}$$

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

<u>Amplitude limiting by ratio detector</u>:

→ An important feature of the ratio detector circuit is its limiting action.

→ This limiting action is achieved with the help of capacitor C connected across R_1 and R_2.

→ The time constant of this circuit is kept large consider a rapid increase in the incoming signal because of a noise impulse.

→ Inorder that the output voltage follows the input, the capacitor C_0 must get charged to that level.

→ To charge itself to the increased input level, the capacitor C_0 draws charging current from the input resonant circuit thereby loading this circuit to a greater extent.

→ As a result, the magnification factor Q of the circuit is lowered reducing the input signal level.

→ For a momentary decrease in the signal, the result is opposite.

→ In order that the output follows this decrease, the capacitor C must discharge to the input level.

→ Thus there is a discharging current through R_1 and R_2. The current drawn from the input circuit is therefore, reduced.

→ this reduces loading upon the input resonant circuit causing its magnification factor Q to increase.

→ As a result, the input signal level is increased. Thus, the ratio detector output remains free from rapid amplitude fluctuation in the signal input and converts frequency changes into amplitude changes.

<u>Advantages</u>:

→ The main advantages of using ratio detector is there is no need for separate amplitude limiting circuitry.

Threshold Effect:

→ When a noise is large as compared to the signal at the input of the detector, the detector output has a message signal completely mingled with the noise.

→ It means that if the input signal to noise ratio is below a certain level called threshold level, the noise dominates over the message signal.

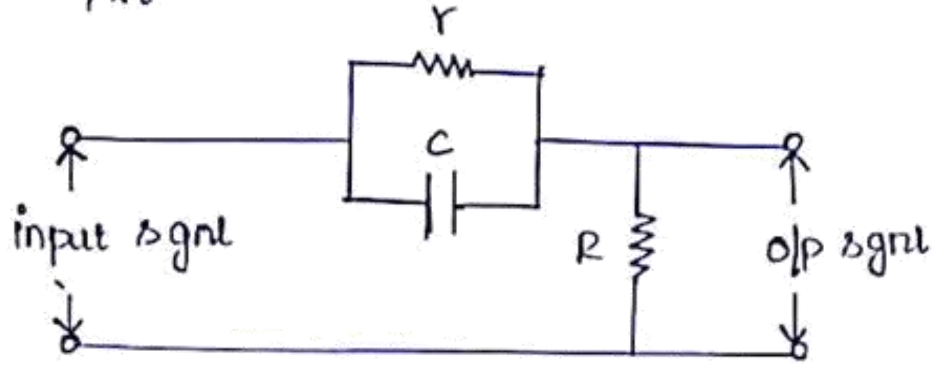
→ Threshold is defined as the value of input signal to noise ratio below which the output signal noise ratio deteriorates much more rapidly than the input signal to noise ratio.

→ Lower the threshold level (S_i/N_i), better is the system. The process of lowering the threshold level is known as threshold improvement.

→ Threshold improvement can be done by Pre-emphasis and deemphasis circuits.

→ The threshold effect can be avoided by improving S_o/N_o at the higher edge of the message band. This is done by a simple R-C network known as pre-emphasis circuit which boosts the signal amplitude of higher frequency in the message band before they modulate the Carrier.

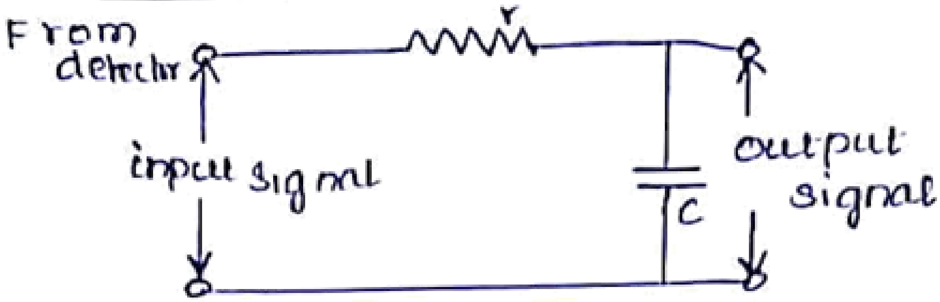
→ The boosted signal at transmitter increases the So/No ratio at the detector Output.



→ Thus So/No ratio becomes large enough to improve the threshold level over the entire message band.

→ However, the high frequency components of the message signal reproduced by the detector are at a raised amplitude level, and therefore amplitude distribution of the base band is disturbed.

→ An inverse action is needed at the discrimin-ator output to bring back the original level of high frequency components and restore the amplitude distribution of Message band. This done by another R-C network known as de-emphasis circuit.



→ The transfer functions $H_d(\omega)$ of the deemed and $H_p(\omega)$ of pre-emphasis circuits have inverse relationship, so that their product i constant for the entire message band.

$$H_d(\omega) = \frac{K}{H_p(\omega)}$$

$$H_d(\omega) H_p(\omega) = K$$

→ the pre-emphasis circuit at the transmitter a high pass network which behaves like a differentiator.

→ The de-emphasis circuit at the receiver is low pass network which behaves like a integrator.

The transfer function for pre-emphasis circuit is given by,

$$H_p(\omega) = \frac{R}{r \parallel \frac{1}{j\omega C}}$$

$$= \frac{R}{\frac{r/j\omega C}{r + \frac{1}{j\omega C}}}$$

$$= \frac{R}{r/j\omega c} \times \left(r + \frac{1}{j\omega c}\right)$$

$$= \frac{R}{r} j\omega c \left[r + \frac{1}{j\omega c}\right]$$

$$= \frac{R}{r}\left[1 + rj\omega c\right]$$

$$H_p(\omega) = \frac{R}{r}\left[1 + \frac{j\omega}{\omega_1}\right]$$

Where,

$$\omega_1 = \frac{1}{rc}$$

→ The frequency components between break frequencies ω_1 and ω_2 have been boosted. The rate of increase in amplitude is 6dB/octave.

→ For the FM broadcast purpose, the lower break frequency f_1 is about 2.1 KHz and higher break frequency f_2 is chosen to be much higher than the highest frequency term in the message band, so that f_2 lies outside the base band spectral range. For audio range F_2 may be taken as 30kHz.

→ The corresponding de-emphasis ckt has an invent characteristic. The transfer function $H_d(\omega)$ of the de-emphasis circuit is given by.

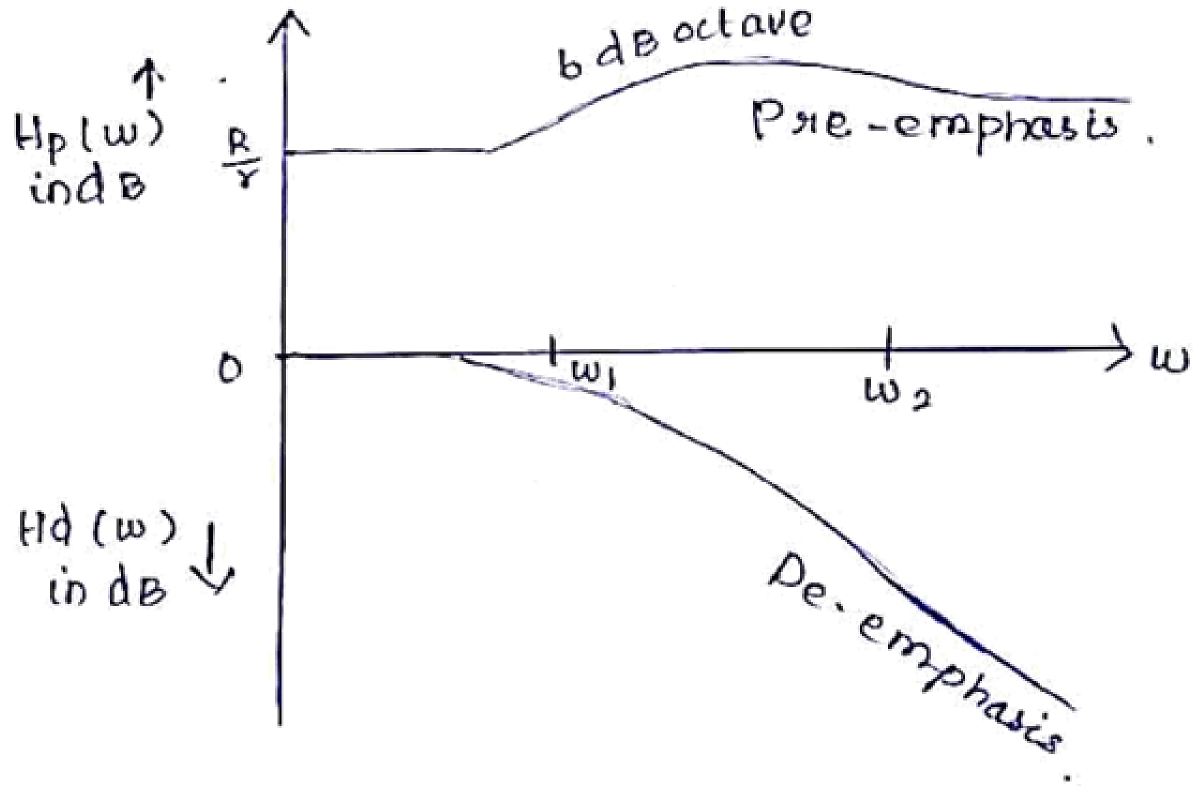
$$H_d(\omega) = \frac{1/j\omega c}{r + 1/j\omega c}$$

$$= \frac{1}{j\omega c} \times \frac{j\omega c}{1 + rj\omega c}$$

$$= \frac{1}{1 + rj\omega c}$$

$$H_d(\omega) = \frac{1}{1 + \frac{j\omega}{\omega_1}}$$

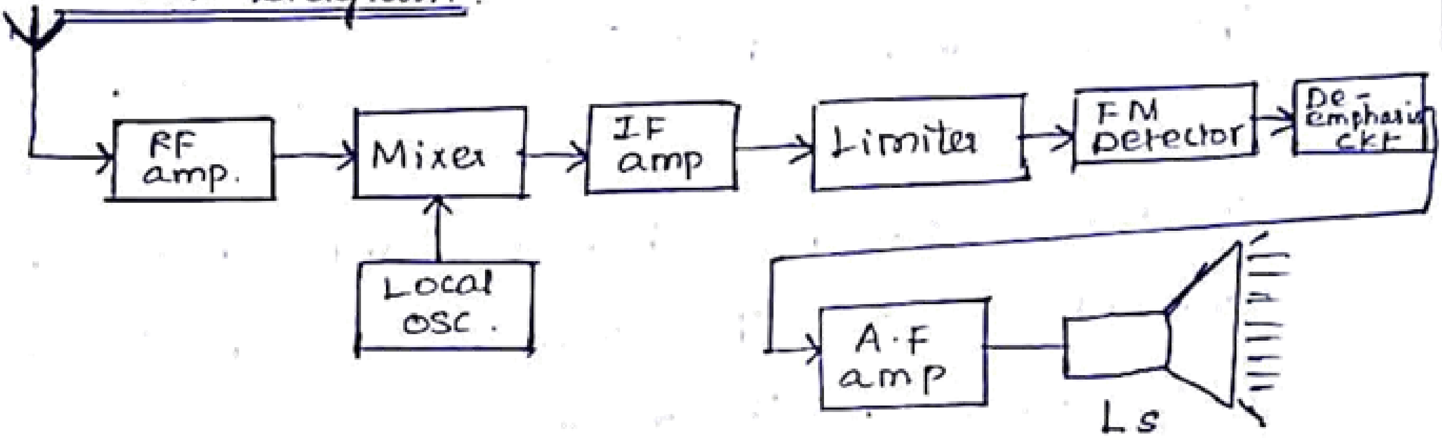
→ The amplitude of frequenquency components above ω_1 are reduced at a rate of -6dB octave and thus does the inverso action of Pre-emphasis circuit to restore the amplitude distributions of the message band.



FM Superheterodyne Receiver :

→ The Function of the FM receiver is to intercept the FM signal incoming from an FM transmitter, and then recover the original modulating signal.

Block Diagram :



RF Amplifier :

→ This stage amplifies the received radio signal. FM uses radio Frequency ranging from 40MHZ to 1GHZ for various application such as FM broad casting, television sound trans-mission, police radio, military systems etc. The bandwidth needed for an RF amplifier may also be large (150 KHz) The RF amplifier also reject the image signal as in AM receivers.

Frequency Mixer and Local Oscillators :

→ Separate active devices are used for the mixer and local oscillator as the frequency involved is in VHF and UHF range. The IF frequency in an FM receiver is much higher than AM. The FM broadcast receivers use 10.7 MHz as intermediate frequency.

IF amplifier :

→ This stage amplifies the intermediate frequency signals. It comprises of multistage double-tuned or stagger-tuned amplifiers to provide a high gain and a high overall bandwidth (150 KHz). This stage is responsible for sensitivity and selectivity of the reciever.

Limiter :

→ The limiter keeps the IF amplifier output voltage constant to a predetermined value and removes all amplitude fluctuations due to noise, or any other interferences.

This is essential because the FM detector need a constant amplitude FM voltage at its input a satisfactory operation. The limiter may be ignored if the ratio detector has been used. diodes and amplifying devices are used in making a limiter circuit.

FM detector:

→ It recovers the modulating Signal from the IF Signal. Various types of FM detector circuit have been discussed. De emphasis Circuit does the inverse job of the Pre-emphasis circuit. The high modulating frequency terms boosted by pre-emphasis are brought back to Original amplitude level by de emphasis circuit.

AF amplifier and Speaker:

→ This Stage amplifies the audio frequency modulating Signal recovered by the FM detector. The amplifier has wider bandwidth than that of AM reciever. The loud Speaker converts the electrical Signal into Sound Signal.

Characteristics of recievers

→ The overall performance of radio recievers is measured from the Characteristics or Parameter of the reciever.

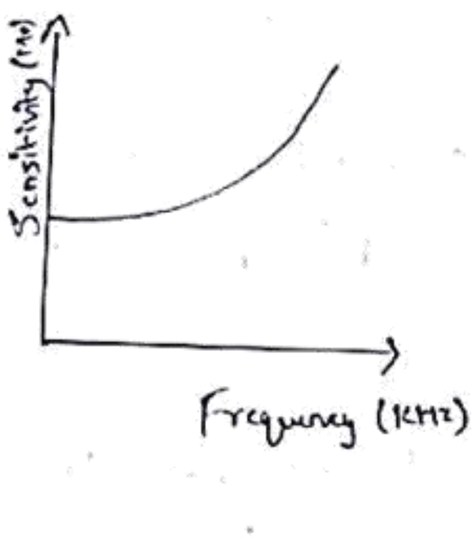
→ It helps us to evaluate the relative merits of Particular circuit design and also helps us to find out the usefulness of a reciever under Special Operating Conditions.

→ The Various Characteristics of the reciever are as follows.

Sensitivity:

→ Sensitivity of a reciever is defined as a measure of its ability to recieve weak Signals.

→ The Sensitivity of an reciever depends on the noise Power Present at the input to the reciever. Thus the Sensitivity of the reciever may be improved by reducing the noise level at the input it can be achieved by reducing the temperature or bandwidth of the reciever.



Selectivity:

→ Selectivity of a reciever is defined as its ability to Select the desired Signal among the various Signals present and rejecting all the other Unwanted Signals.

→ For Example, in AM broadcast, each station is allocated a 10KHz bandwidth, So the recievers must select only within the range of frequency assigned for it.

→ If the recievers passband is more than 10KHz, then more than one channel also be recieved simultaneously results in cross talk and distortion occurs in the reciever.

→ Thus the reciever should have high Selectivity.

→ The Selection is done by resonant circuits, the selectivity is entirely dependent upon the frequency response of these resonant circuit used in it.

→ A resonant circuit with high quality factor Q is more selective than a circuit with low Q.

Stability:

→ It is the ability of a reciever to deliever, a Constant amount of output for a given period Of time, when the reciever is Supplied with a Signal of constant amplitude and frequency.

Fidelity:

Fidelity is a measure of the ability of a communication system to produce an exact replica of the original information

It can also be defined as the degree to which a system accurately reproduces at the output the essential characteristics of signals that are impressed upon the input.

Signal to Noise Ratio:

It can be defined as the ratio of signal power to noise power at the reciever output.

A good reciever should have high SNR which indicates negligible noise power at the output.

Comparison between AM and Angle Modulation

Angle modulation has following advantages over AM :

i) The amplitude of FM is constant. It is independent of depth of modulation. Hence transmitter power remains constant in FM whereas it varies in AM.

ii) Since amplitude of FM is constant, the noise interference is minimum in FM. Any noise superimposing an amplitude can be removed with the help of amplitude limits. Whereas it is difficult to remove amplitude variations due to noise in AM.

iii) The depth of modulation have limitation in AM. But in FM the depth of modulation can be increased to any value by increasing the deviation. This does not cause any distortion in FM signal.

iv) Since guard bands are provided in FM, there is less possibility of adjacent channel interference.

v) Since space waves are used for FM, the radius of propagation is limited to line of sight. Hence it is possible to operate several independent transmitters on same frequency with minimum interference.

vi) Since FM uses UHF and VHF ranges, the noise interference is minimum compared to AM which uses MF and HF ranges.

There are some disadvantages of FM compared to AM as follows :

i) The bandwidth requirement of FM is much higher than that of AM.

ii) The FM transmitting and receiving equipment is more complex and costly.

iii) Since FM uses UHF and VHF range of frequencies, its area of reception is limited only to line of sight. This is much lower than area covered by AM.

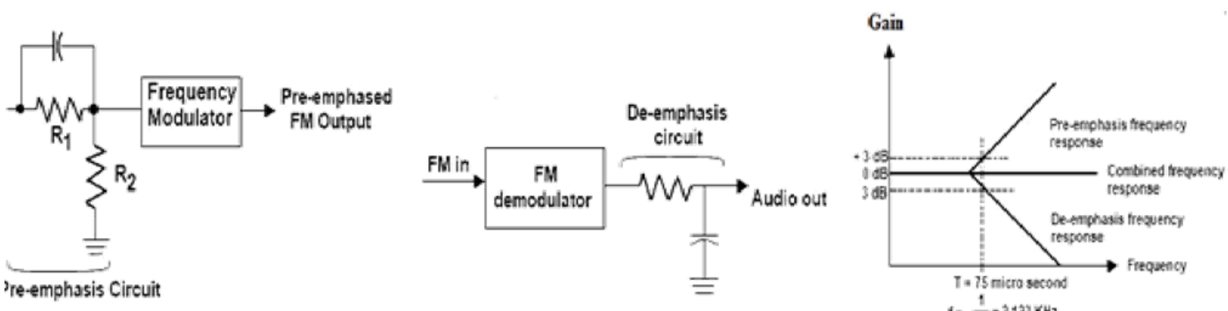
Table shows the comparison between FM and AM.

	Amplitude Modulation	Frequency Modulation
1.	Amplitude of the carrier is varied according to amplitude of modulating signal.	Frequency of the carrier is varied according to amplitude of the modulating signal.
2.	AM has poor fidelity due to narrow bandwidth.	Since the bandwidth is large, fidelity is better.
3.	Most of the power is in carrier hence less efficient.	All the transmitted power is useful.
4.	Noise interference is more.	Noise interference is minimum.
5.	Adjacent channel interference is present.	Adjacent channel interference is avoided due to wide bandwidth.

PRE-EMPHASIS AND DE-EMPHASIS

In *frequency modulation* the modulation or sideband frequencies often extend outward to as much as 15 kHz. This provides a much broader audio range than is usually found in conventional A.M. broadcasting. The higher audio frequencies produced by the program source are generally *weaker* than the lower audio frequencies, and thus produce a *smaller frequency deviation* in the carrier. In the normal transmitting process a certain amount of *interference* is encountered, such as atmospherics, which have little effect on the large amplitude low-frequency signals. However, with regard to the weak high-frequency signals, the *signal-to-noise ratio* is low. To correct this condition, the transmitter employes a *pre-emphasis* network.

This consists of a high-pass filter having a time constant of 75 micro seconds. The pre-emphasis at the FM transimtter provides for an audio gain of 1 at 50 Hz, to a gain of 7 at 15 kHz. From 50 to 500 Hz, the rise is very slight, but from there it increases rapidly. Thus *the net effect of pre-emphasis is to increase the signal-to-noise ratio at the higher audio frequencies.*



(a) Pre-emphasis Circuit (b) De-emphasis circuit Combined frequency response

Fig 3.9

If the pre-emphasis were not *corrected* at the receiver, the sound signal would have a *heavy treble effect.* To compensate for this, the FM receiver employs a *de-emphasis network,* a low-pass filter. As in the case of the high-pass filter, the low-pass filter in the de-emphasis network also has a time constant of 75 micro-seconds.

The de-emphasis network can be placed in any one of the several places in the receiver, but practice finds it located at the output of the FM detector circuit. It is an *RC* filter that shunts the higher audio frequencies to ground while affecting the lower audio frequencies to a lesser extent. *The overall effect of the pre-emphasis and de-emphasis networks is to provide a noise free signal at the output of the receiver that is a true replica of the sound signal at the studio.*

FM Stereo broadcast Transmission

Figure shows a simplified block diagram for a stereo FM transmitter. The L and R audio channels are combined in a matrix network to produce the L + R and L − R audio channels. The L − R

audio channel modulates a 38-kHz subcarrier and produces a 23-kHz to 53-kHz L − R stereo channel. Because there is a time delay introduced in the L − R signal path as it propagates through the balanced modulator, the L + R stereo channel must be artificially delayed somewhat to maintain phase integrity with the L − R stereo channel for demodulation purposes. Also for demodulation purposes, a 19-kHz pilot is transmitted to regenerate the carrier at the receiver.The composite baseband signal is fed to the FM transmitter, where it modulates the main carrier.

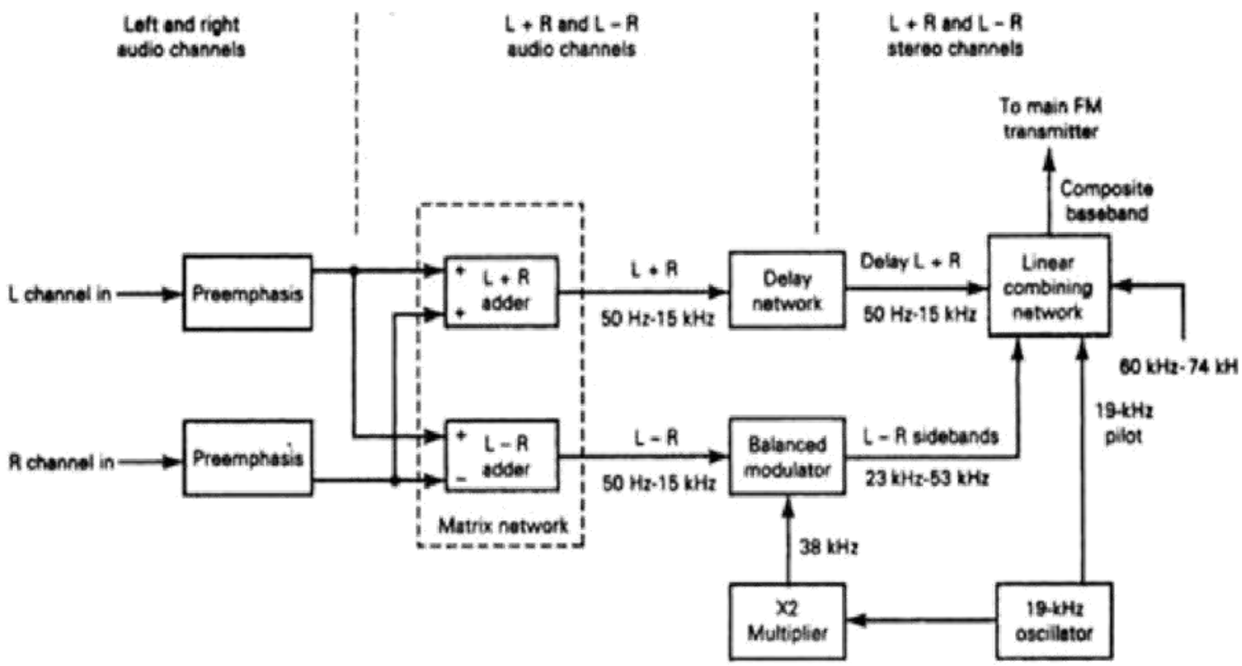


FIGURE Stereo FM transmitter using frequency-division multiplexing

The information contained in the L + R and L − R stereo channels is identical except for their phase. With this scheme, mono receivers can demodulate the total baseband spectrum, but only the 50-Hz to 15-kHz L − R audio channel is amplified and fed to all its speakers. Therefore, each speaker reproduces the total original sound spectrum. Stereophonic receivers must provide additional demodulation of the 23-kHz to 53-kHz L − R stereo channel, separate the left and right audio channels, and then feed them to their respective speakers.

FM STEREO BROADCAST RECEIVER:

The block diagram of stereo FM demodulation with optional SCA output is shown in the Fig.

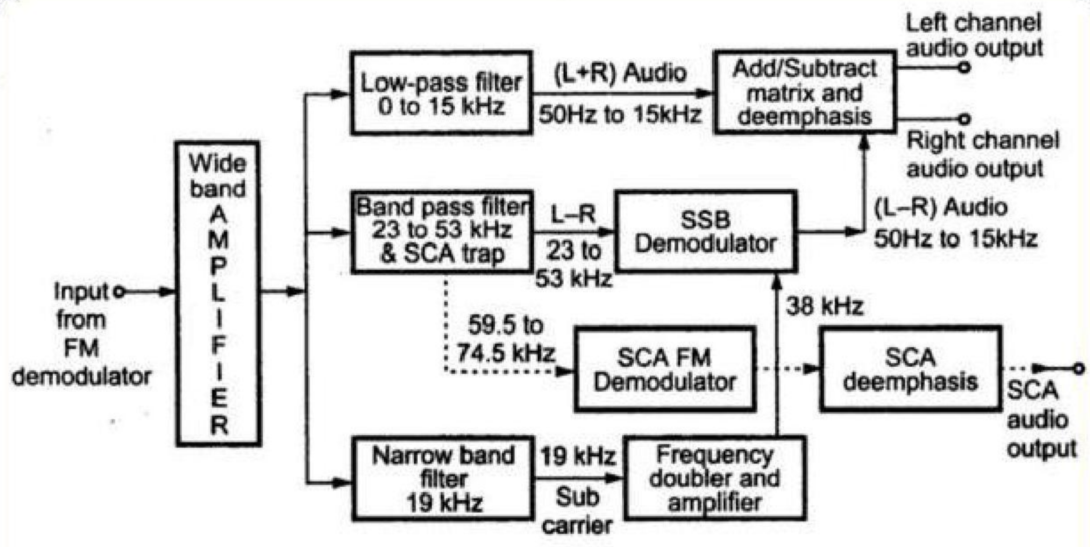


Fig FM Stereo broadcast receiver

Note: Subsidiary Communication Authorization [SCA] signal may also be transmitted along with the primary stereo signal. It is applied as input to adder. This SCA signal is not always transmitted, it is optional.

The low pass filter attenuates all frequencies above 15 kHz and thus the sum signal [L + R] is available at its output. In manaural receiver, this output will be further passed through the deemphasis network to the audio amplifier, for reproduction of stereo broadcast in monaural way.

The band pass filter has a passband extending from 23 kHz to 53 kHz, which corresponds to the difference signal [L – R], rejecting the (optional) SCA signal above 59.5kHz.

The difference signal [L – R] is fed to the SSB demodulator which also receives 38 kHz carrier necessary for demodulation. After demodulation, [L – R] audio signal in the frequency range 50 Hz to 15 kHz is available which is applied as the input to Add/Subtract Matrix and deemphasis block, also receiving [L + R] signal as its other input. The matrix produces the left channel from an adder and the right channel from a subtractor. After deemphasis, they are further amplified to yield ultimately stereo reproduction.

The 19 kHz subcarrier, transmitted at much lower level and serving as a pilot carrier, is separated from the composite signal, using a narrow band filter tuned to 19 kHz. The frequency doubler converts it to the wanted 38 kHz carrier for SSB demodulator.

If SCA signal is present, it is separated using band pass filter, then demodulated by FM detector, passed through deemphasis and produced finally as a separated audio output.

Armstrong method for FM generation

- The direct methods cannot be used for the broadcast applications. Thus the alternative method i.e. indirect method called as the Armstrong method of FM generation is used.
- In this method the FM is obtained through phase modulation. A crystal oscillator can be used hence the frequency stability is very high.
- The block diagram of the Armstrong method is shown below:

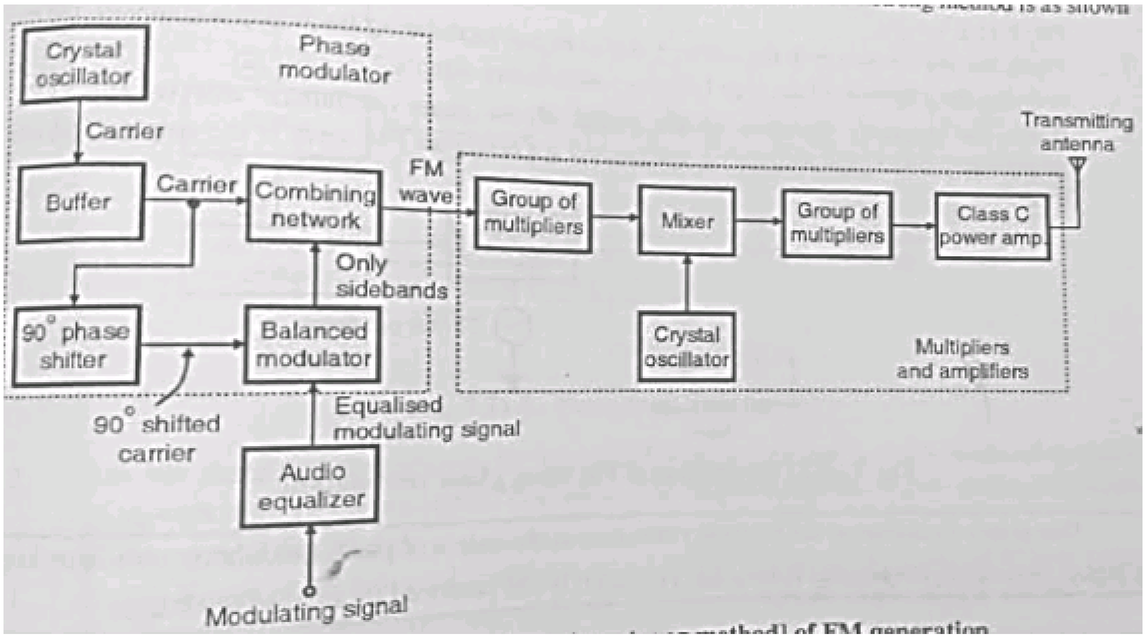


Fig: Indirect method [Armstrong method] of FM generation

Operation:

- The crystal oscillator generates the carrier at low frequency typically at 1MHz. This is applied to the combining network and a 90° phase shifter.
- The modulating signal is passed through an audio equalizer to boost the low modulating frequencies .The modulating signal is then applied to a balanced modulator.
- The balanced modulator produced two side bands such that their resultant is 90° phase shifted with respect to the unmodulated carrier.
- The unmodulated carrier and 90° phase shifted sidebands are added in the combining network.
- At the output of the combining network we get F_m wave. This wave has a low carrier frequency f_c and low value of the modulation index m_f

- The carrier frequency and the modulation index are then raised by passing the FM wave through the first group of multipliers. The carrier frequency is then raised by using a mixer and then the f_c and m_f,both are raised to required high values using the second group of multipliers.

- The FM signal with high f_c and high m_f is then passed through a class C power amplifier to raise the power level of the FM signal.

- The Armstrong method uses the phase modulation to generate frequency modulation. This method can be understood by dividing it into four parts as follows:

1.Generation of FM from phase modulator:

- The modulating signal is passed through a low pass RC filter.
- The filter output is then applied to a phase modulator along with carrier.
- Hence the extra deviation in the carrier f_c due to higher modulating frequency is compensated by reducing the amplitude of the high frequency modulating signals.

    - Hence the frequency deviation at the output of the phase modulator will be effectively proportional only to the modulating voltage and we obtain an FM wave at the output of phase modulator.

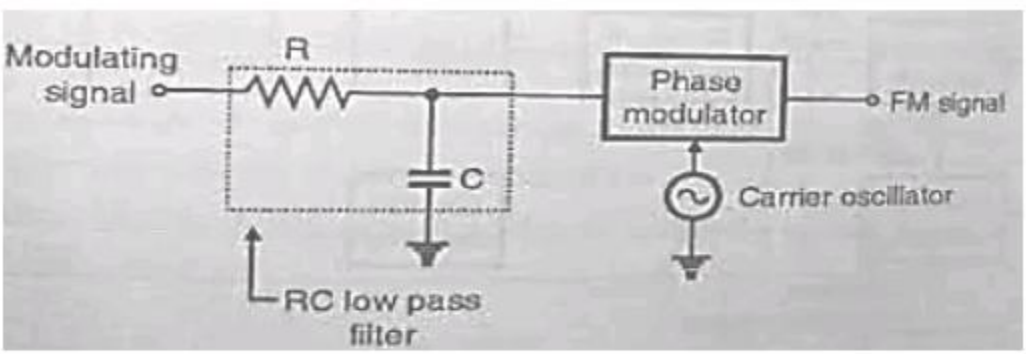


Fig: Generation of FM using phase modulation

2.Implementation of phase modulator:

- The crystal oscillator produces a stable unmodulated carrier which is applied to the "90° phase shifter" as well as the "combining network" through a buffer.

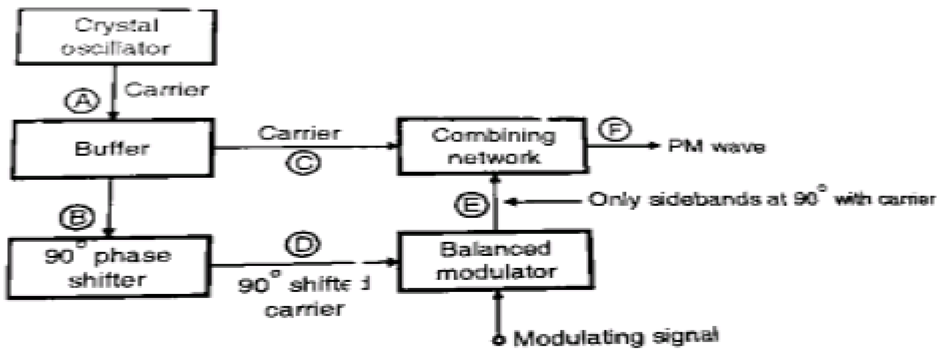


Fig: Phase modulator circuit

- The 90° phase shifter produces a 90° phase shifted carrier. It is then applied to the balanced modulator along with the modulation signal.
- At the output of the balanced modulator we get DSBSC signal i.e. AM signal without carrier. This signal consists of only two sidebands with their resultant in phase with their resultant in phase with the 90° phase shifted carrier.

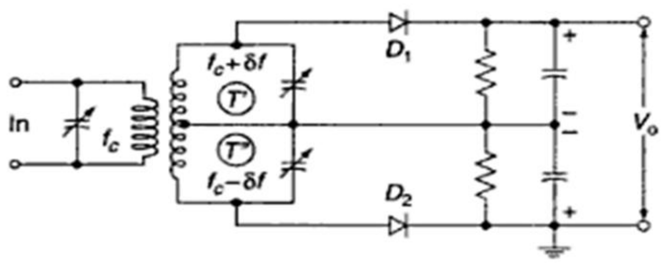
3.Combining parts 1 and 2 to obtain The FM:

- Combining the parts 1 and 2 we get the block diagram of the Armstrong method of FM generation

4.Use of frequency multipliers and amplifiers:

- The FM signal produced at the output of phase modulator has a low carrier frequency and low modulation index. They are increased to an adequately high value with the help of frequency multipliers and mixer. The power level is raised to the desired level by the amplifier.

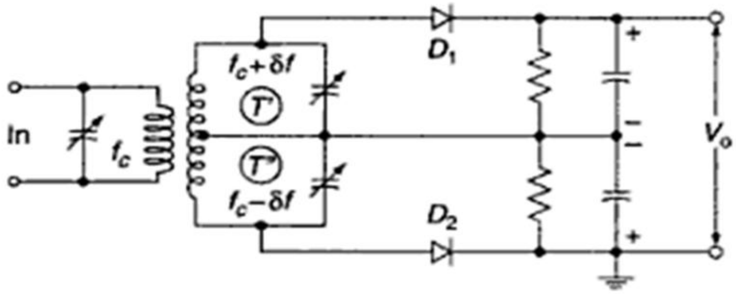
Balanced Slope Detector

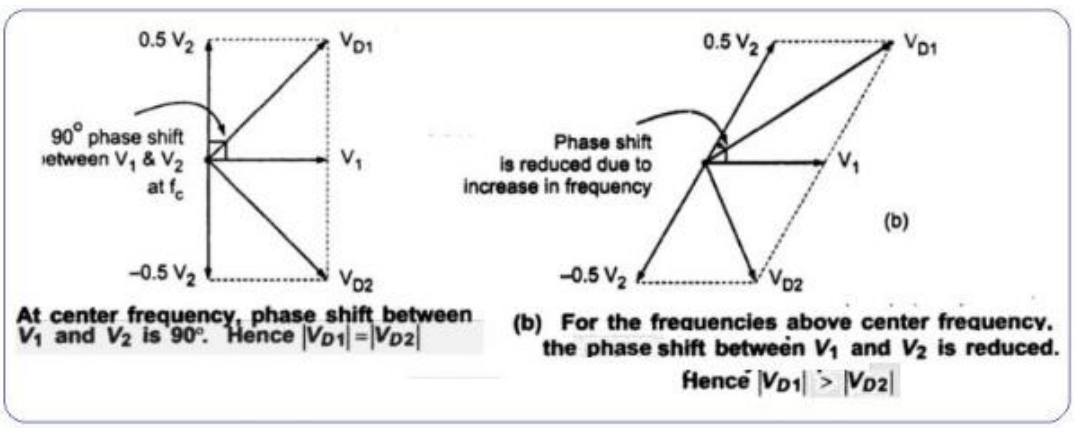


- Travis detector/Triple tuned discriminator/Amplitude discriminator
- Uses two slope detectors connected back to back to the opposite ends of a centre tapped transformer
- Hence fed 180 out of phase.

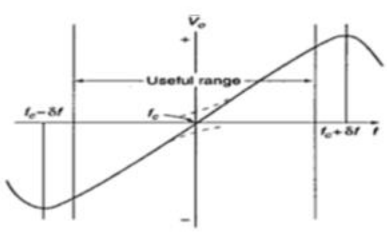
Primary tuned to fc (which is the IF)

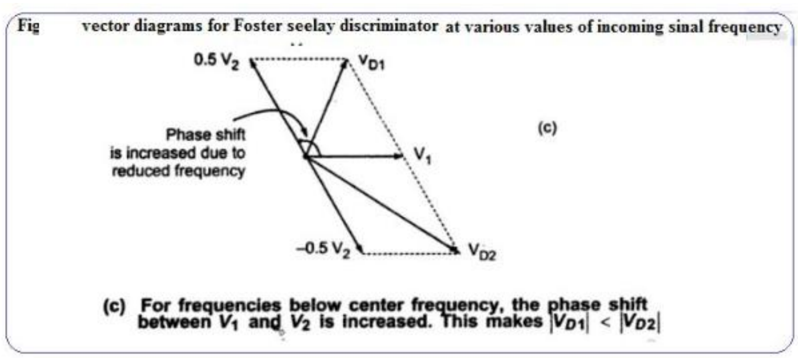
• Top secondary tuned above the IF by an amount δf ie fc + δf
• Bottom secondary tuned below the IF by an amount δf ie fc – δf
• Diode detector with RC load
• Output taken across the series combination of the two loads
• Working principle – explanation
– Input freq = fc; Voltage across T1 and T2 are same ; V0 = 0
– Input freq = fc + δf; Output of D1 > output of D2; V0 is positive and max.
– Input freq = fc – δf; Output of D2 will be large negative ; Output of D1 is small positive; V0 is negative and max.





S-shaped Frequency modulation characteristic





- More efficient than slope detector
  - Difficult to align (Three different tuning)
  - Better linearity – but not good enough
  - No amplitude limiting

Multiplexing and Analog Pulse Modulation

Multiplexing is the process in which multiple Data Streams, coming from different Sources, are combined and Transmitted over a Single Data Channel or Data Stream.

In Electronic Communications, the two basic forms of Multiplexing are Time Division Multiplexing (TDM) and Frequency Division Multiplexing (FDM).

In Time Division Multiplexing, Transmission Time on a Single Channel is divided into non-overlapped Time Slots. Data Streams from different Sources are divided into Units with same size and interleaved successively into the Time Slots.

In Frequency Division Multiplexing, Data Streams are carried simultaneously on the same Transmission medium by allocating to each of them a different Frequency Band within the Bandwidth of the Single Channel.

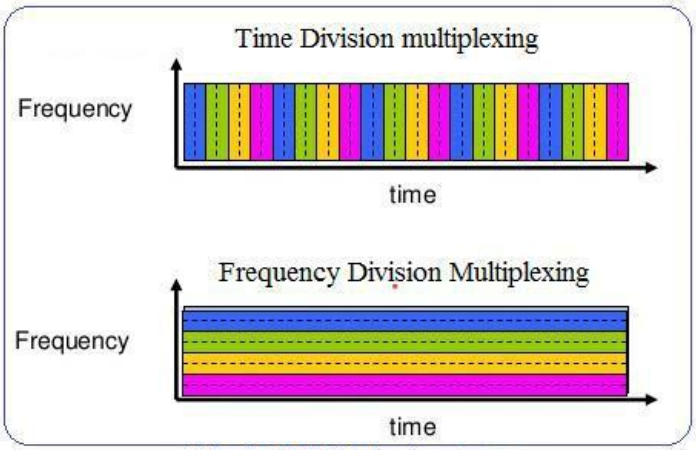


Fig 4.1 Multiplexing types

Concepts of Multiplexing

More efficient communication system can be obtained if a station transmits more than one "message" on the same carrier and on the same channel, or number of transmitters are transmitting simultaneously on the same channel. This process is known as "**multiplexing**" and has been used for many years in long distance telephony.

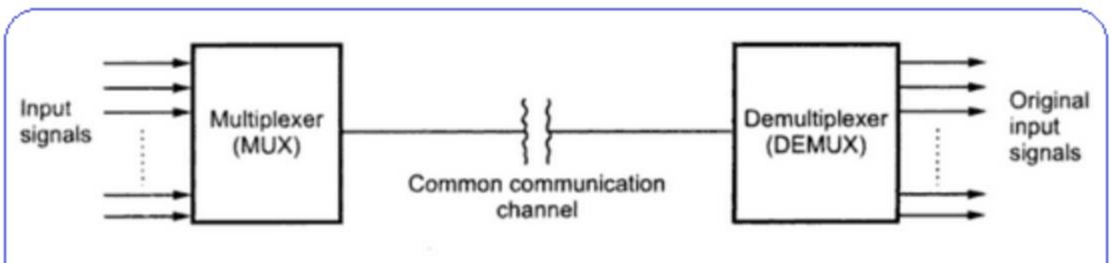


Fig. 4.2 Concept of multiplexing

The Fig. 4.2 illustrates the concept of multiplexing. The multiplexer combines all input signals into a single composite signal and transmits it through a common channel to the receiver. At the receiver end, the demultiplexer is used to separate original input signals from the composite signal.

Types of Multiplexing

There are two types of multiplexing :

- **Frequency Division Multiplexing (FDM) :** In FDM, many signals are transmitted simultaneously where each signal occupies a different frequency slot within a common bandwidth. Usually, FDM systems are used in analog communication.
- **Time Division Multiplexing (TDM) :** In TDM, the signals are not transmitted at a time; however, they are transmitted in differnt time slots. Usually, TDM systems are used in digital communication.

Freqeuncy Division Multiplexing (FDM)

Multiplexing requires that the signals be kept apart so that they do not interfere with each other, and thus they can be separated at the receiving end. This is accomplished by separating the signal either in frequency or time. The technique of separating the signals in frequency is referred to as frequency division multiplexing (FDM), whereas the technique of separating the signals in time is called time division multiplexing. In this section, we discuss frequency division multiplexing systems, referred hereafter as FDM.

Fig. 4.3 shows the block diagram of FDM system. As shown in the Fig. 4.3 input message signals, assumed to be of the low-pass type are passed through input low-pass filters. This filtering action removes high-frequency components that do not contribute significantly to signal representation but may disturb other message signals that share the common channel. The filtered message signals are then modulated with necessary carrier frequencies with the help of modulators. The most commonly method of modulation in FDM is single sideband modulation, which requires a

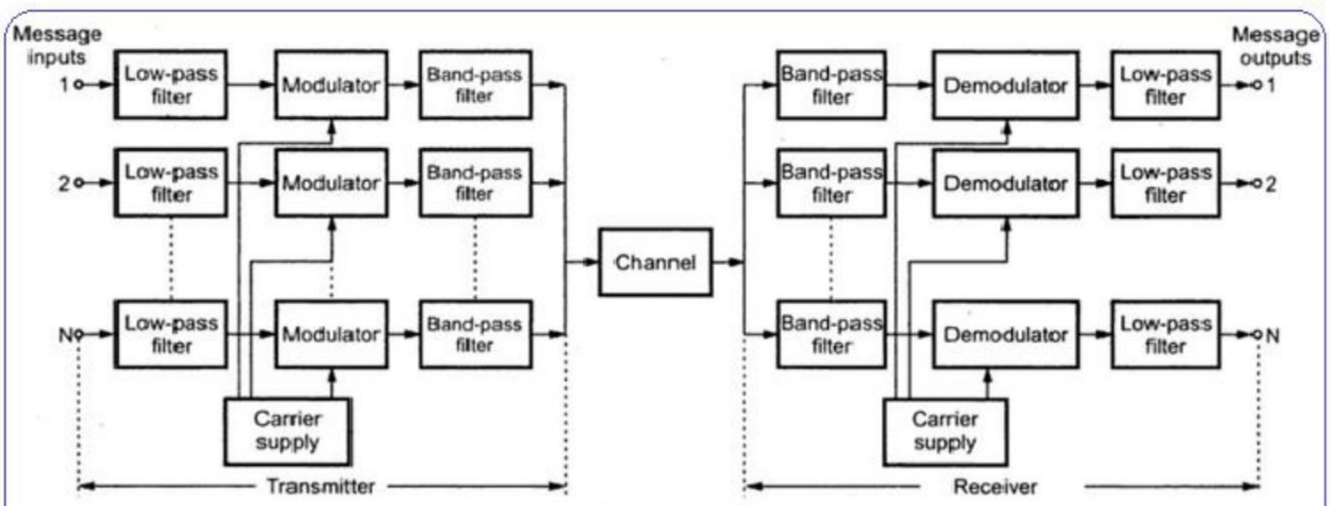


Fig 4.3 Block diagram of FDM System

bandwidth that is approximately equal to that of original message signal. The band pass filters following the modulators are used to restrict the band of each modulated wave to its prescribed range. The outputs of band-pass filters are combined in parallel which form the input to the common channel.

At the receiving end, bandpass filters connected to the common channel separate the message signals on the frequency occupancy basis. Finally, the original message signals are recovered by individual demodulators.

Guard Bands

The Fig. **4.4** shows the frequency spectrum of FDM signal. As shown in the Fig. **4.4**, the adjacent spectrums in the spectrum of an FDM signal do not touch each other. They are separated from each other by guard bands to avoid any interference between them. Wider the guard band, smaller the interference.

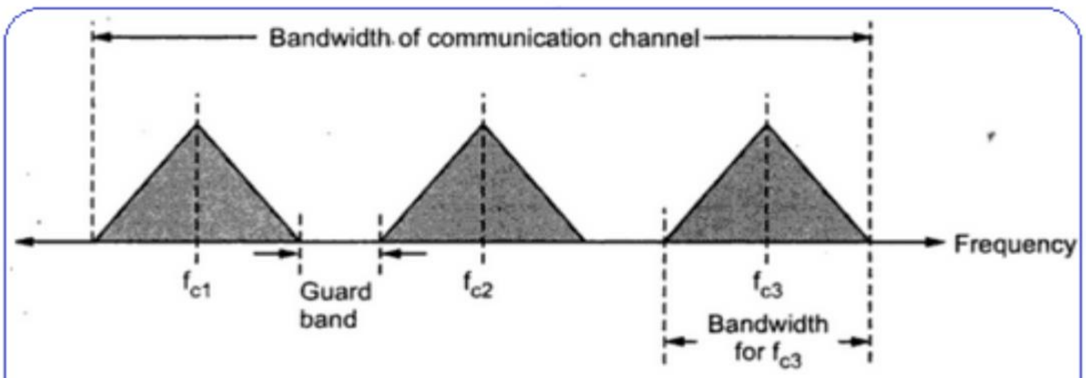


Fig. 4.4 Spectrum of FDM signal

Transmission Bandwidth

Let us determine the transmission bandwidth required to transmit 24 independent voice message inputs. Assume bandwidth of 4 kHz for each voice message input. We know that, in SSB modulation, bandwidth required is equal to the bandwidth of the message signal. Therefore, the transmission bandwidth required to transmit 24 voice message signals is given as

$$B = 24 \times 4 = 96 \text{ kHz}$$

This bandiwdth may increase if we consider the guard band.

Using FDM for Telephone Lines

Let us consider a system based on FDM providing 960 channels, each occupying 4 kHz. Here, modulation is done in several cascaded stages. The stages are divided into basic groups or pregroups, master (groups or simply) groups, and super groups

Each pregroup modulator provides three carrier oscillators at 12, 16, and 20 kHz. Here, USB (upper sideband) modulation is used. Thus, the first channel from baseband signal 0-4 kHz occupies the first channel slot from 12 kHz-16 kHz. The second channel goes in 16 kHz to 20 kHz slot and the third goes in the 20 kHz to 24 kHz slot. All these channels are added to form a pregroup and sent to one of the group modulator inputs. This is illustrated in Fig. 4.5 (a). A separate pregroup modulator is required for every three channel, so to accomodate 960 channels we require 960/3 = 320 pregroup modulators.

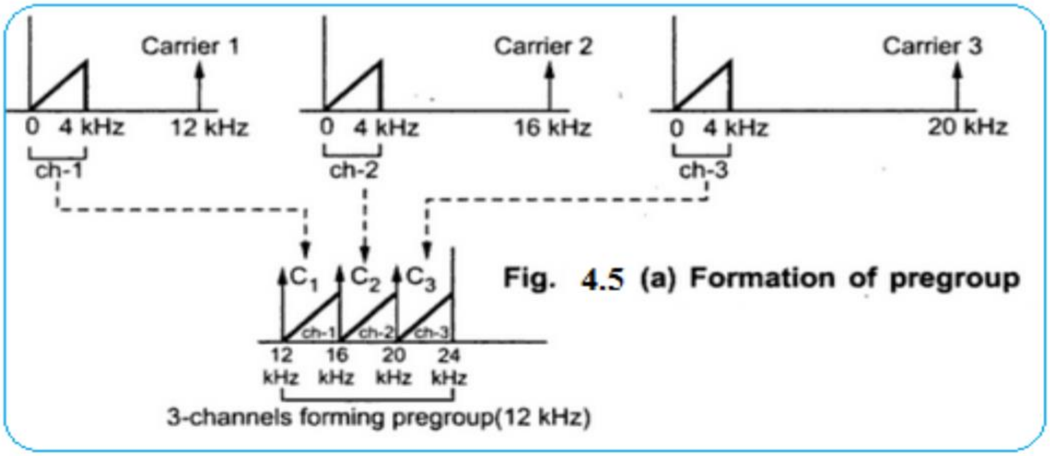


Fig. 4.5 (a) Formation of pregroup

The group modulator provides four carriers at 84 kHz, 96 kHz, 108 kHz, and 120 kHz. Here, four pregroup signals are moved by LSB modulation into four consecutive frequency slots at 60 kHz to 72 kHz, 72 kHz to 84 kHz, 84 kHz to 96 kHz and 96 kHz to 108 kHz. Because of LSB modulation, the order of frequencies are reversed. Again all four carrier are suppressed and output from four modulators are added to form a group. The added output is then sent to input of the one of the super group modulator. This is illustrated in Fig. 4.5 (b). As shown in the Fig. 4.5 (b), the output frequency range of a group is 60 kHz to 108 kHz.

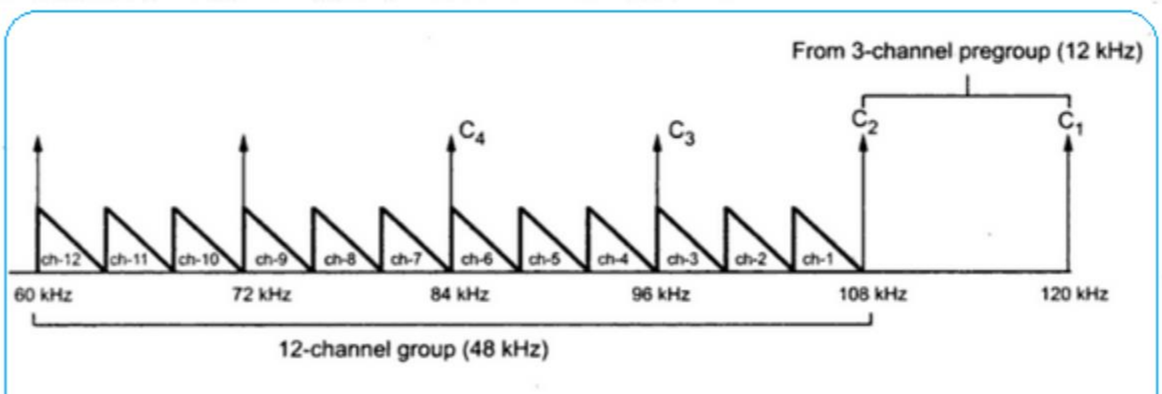


Fig. 4.5 (b) Formation of group

Each supergroup modulator provides carriers at 420, 468, 516, 564 and 612 kHz. Again LSB modulation takes place and it places the five group signals in the slots 312 to 360, 360 to 408, 408 to 456, 456 to 504 and 504 to 522 kHz. Again the order of frequencies within each slot is reversed by the LSB modulation and all carriers are suppressed. The five outputs are then added and sent to the input of master group modulator, each with frequencies from 312 to 552 kHz. This is illustrated in Fig. 4.5 (c).

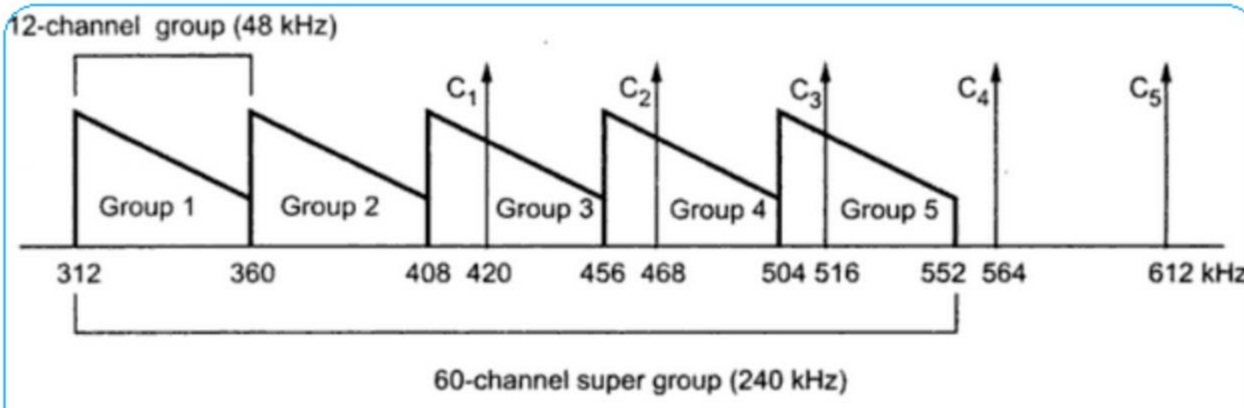


Fig. 4.5 (c) Formation of supper group

Each master group modulator provides 16 carrier, frequencies. These 16 carrier frequencies are again LSB modulated to given 16 supergroups. The output of these 16 supergroups are added to give master group. This is illustrated in Fig. 4.5 (d).

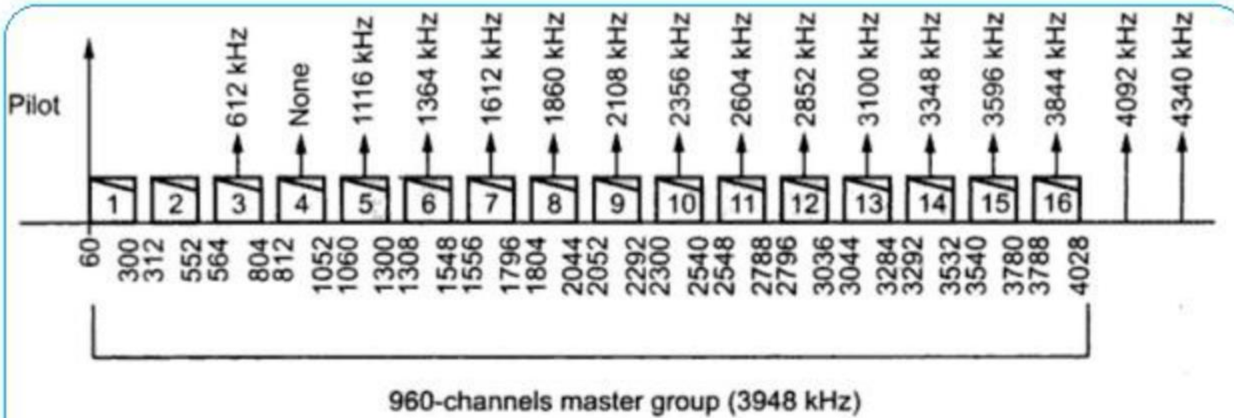


Fig. 4.5 (d) Formation of master group

As shown in the Fig. 4.5 (d), the first three slots are separated by 12 kHz and the remainder by 8 kHz. All carriers are suppressed, and a single pilot carrier is transmitted at 60 kHz to provide demodulation synchronization. Therefore, complete signal contains 960 FDM channels each 4 kHz wide in a frequency range from 60 to 4028 kHz.

Advantages of FDM

1. Number of signals can be transmitted simultaneously.
2. Do not require synchronization between transmitter and receiver.
3. Only a single channel gets affected due to slow narrow band fading.

Disadvantages of FDM

1. Requires larger bandwidth of communication channel.
2. Suffers from crosstalk problem due to imperfect bandpass filter.
3. Requires complex circuitry at transmitter and receiver.
4. More number of modulators and filters are required.
5. All FDM channels get affected due to wideband fading.

Applications of FDM

Some important applications of FDM are as follows :

1. In radio broadcasting using AM (Amplitude Modulation) and FM (Frequency Modulation).
2. In TV broadcasting.
3. In Telephone systems.
4. In first generation of celluar phones.

Time Division Multiplexing (TDM)

As the number of messages to be transmitted increases, the frequency division technique presents problems. The number of subcarriers needed increases, and stability problems can arise. Additional circuitry is required, both at transmitting and receiving ends to handle each added channel. The bandwidth requirements increase directly with the number of channels. These problems are eliminated to a great extent by using Time Division Multiplexing [TDM], together with pulse modulation.

In TDM, each intelligence signal to be transmitted [voice or telemetry data] is sampled sequentially and the resulting pulse code is used to modulate the carrier. The same carrier frequency is used to transmit different pulses sequentially, one after other, thus each intelligence, to be transmitted, has been allotted a given time slot. Since only one signal modulates the carrier at any time, no added equipment and no increase in bandwidth is needed when multiplexing. The number of sequential channels that can be handled is limited by the time span required by any one channel pulse and the interval between samples.

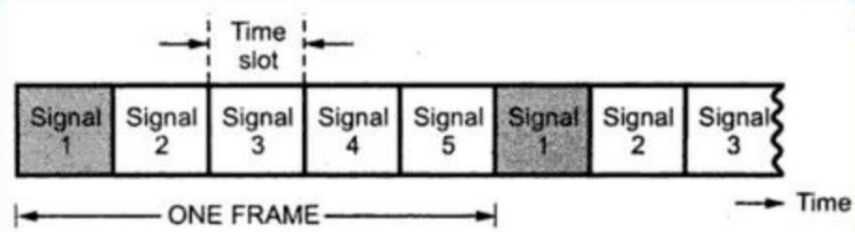


Fig. 4.6 Time frame

Thus, in TDM, each signal occupies the entire bandwidth of the channel. However, each signal is transmitted for only a short period of time, as shown in Fig. 4.6

Here five signals are time division multiplexed. Each signal is allowed to use the channel for a fixed interval of time, called **time slot**. The five signals use the channel sequentially one after other.

One transmission of each channel completes one cycle of operation, called a "**frame.**" Once all the signals have been transmitted, the cycle repeats again and again, at a high rate of speed. The Fig. 4.7 shows the block diagram of a TDM system which is used to multiplex the five signals.

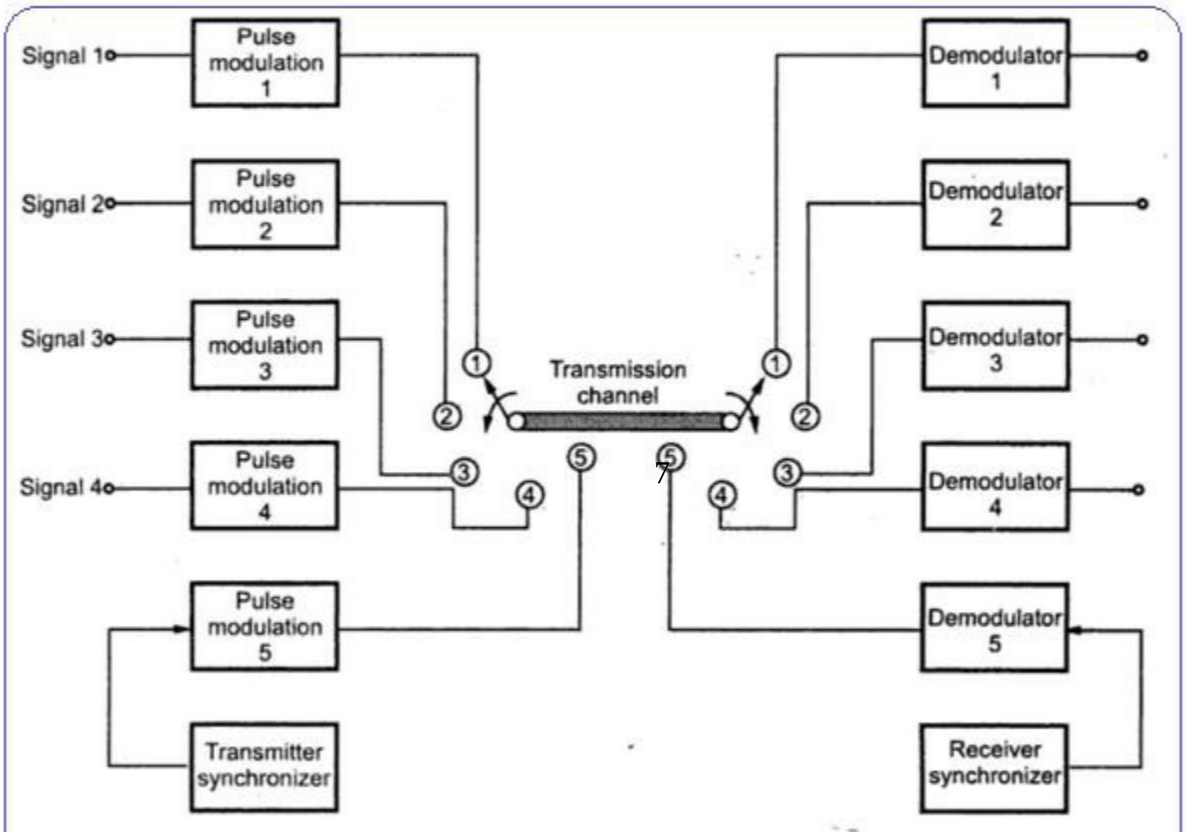


Fig. 4.7 Block diagram TDM system

A rotating switch called a **commutator** connects the output of each channel modulator to the communication channel input in turn. The commutator is realized with electronic switches since it has to rotate at high speed. The commutator remains at each contact for a fixed interval of time, which is the times slot allotted for each channel.

At the receiver, another switch, rotating in synchronism with the sending end commutator is used. This switch connects the pulses received to the appropriate demodulator circuits. For the proper operation of the system, absolute synchronism is very essential, between transmitter and receiver.

Crosstalk

The interference of the adjacent channels or overlapping of information between adjacent channels is called **crosstalk**. For faithful communication crosstalk must be avoided. In TDM crosstalk may occur due to insufficent transmission bandwidth to preserve the shape of the TDM pulses. In FDM the crosstalk may occur when frequency response of filter is not sharp enough. To eliminate or to reduce crosstalk a guardband (in case of FDM and guard interval in TDM) is provided between the adjacent channels.

Comparison of FDM and TDM

The FDM and TDM, being multiplexing techniques, accomplish the same goal, i.e. transmitting more than one message, on the same channel. Thus they are dual techniques.

Frequency Division Multiplexing requires modulators, filters, demodulators ; while Time Division Multiplexing require commutator at the transmitting end and a distributor, working in perfect synchronism with commutator at the receiving end. A perfect synchronism between transmitter and receiver is absolutely essential for proper operation of TDM system. Thus TDM synchronization is more demanding than that of FDM with suppressed-carrier modulation. For demodulating the SSB signal used in FDM; the carrier is locally generated in the receiver. This local carrier must be exactly identical to one used at the transmitting end with respect to frequency and phase.

There is no crosstalk in TDM if the pulses are completely isolated and non-overlapping since message separation is achieved by decommutation, rather by filtering as done in FDM. Actual pulse shapes, having decaying tails, do tend to overlap. However, the resulting crosstalk can be effectively reduced by providing guard-time between pulses, analogous to the guard bands in FDM. A practical TDM system has both guard times and guard bands, the former to suppress crosstalk, the latter to facilitate message reconstruction with practical filters. In PCM technique, pulse code modulated signals are converted into pulse amplitude modulated pulses, which are passed through low-pass filter to receive back the original modulating signal.

The difference between TDM and FDM can be summarized as shown in Table **4.1**

Advantages of TDM

1. Entire channel bandwidth can be utilized for each channel.
2. In TDM, intermodulation distortion is absent.
3. Crosstalk porblem is not severe in TDM.
4. Does not require very complex circuitry.

Disadvantages of TDM

1. Perfect synchronization between transmitter and receiver is required.
2. All TDM channels may get affected due to slow narrow fading.

Table 4.1 Comparison between TDM and FDM

S. N.	TDM	FDM
1.	It is a technique for transmitting several messages on one channel by dividing time domain slots. One slot for each message.	In this technique to transmit several messages on one channel, message signals are distributed in frequency spectrum such that they do not overlap.
2.	It requires commutator at the transmitting end and a distributor, working in perfect synchronization with commutator at the receiving end.	FDM requires modulator, filters and demodulators.
3.	Perfect synchronization between transmitter and receiver is required.	Synchronization between transmitter and receiver is not required.
4.	Crosstalk problem is not severe in TDM.	FDM suffers from crosstalk problem due to imperfect bandpass filter.
5.	It is usually preferred for digital signal transmission.	It is usually preferred for analog signal transmission.
6.	It does not require very complex circuitry.	It requires complex circuitry at transmitter and receiver.

PULSE ANALOG MODULATION TECHNIQUES

The pulse analog modulation techniques are of three types namely, PAM, PWM and PPM. This section describes each of them and also about the recovery of message from them.

Pulse Amplitude Modulation (PAM)

Pulse amplitude modulation is defined as the process of varying the amplitude of the pulse in proportion to the instantaneous variations of message signal.
Let the message signal be given by

$$v_m = V_m \sin\omega_m t \tag{4.1}$$

If $x(t)$ is a periodic signal with period T_0, then it should satisfy the defnition stated as $x(t) = x(t + T_0)$. The pulse train is a periodic signal with some fundamental period say T_0. Then the information present in each period of the pulse train is given by

$$p = V_p \qquad 0 \le t \le \Delta \tag{4.2}$$

$$= 0 \qquad \Delta \le t \le T_0 \tag{4.3}$$

where Δ is the width of the pulse and the leading edge of the pulse is assumed to be coinciding with the starting of the interval in each period.

The pulse amplitude modulated wave in the time domain is obtained by multiplying the message with the pulse train and is given by

$$p_a = p \times v_m \tag{4.4}$$

Substituting p in the above equation we get

$$p_a = V_p V_m \sin\omega_m t \qquad 0 \le t \le \Delta \tag{4.5}$$

$$= 0 \qquad \Delta \le t \le T_0 \tag{4.6}$$

Figure 4.8 shows the message, pulse train and PAM signal. The amplitude of the PAM signal follows the message signal contour and hence the name. It can be shown that the spectrum of PAM signal is a *sinc* function present at all frequencies (for derivation, please refer to the topic of Fourier series in any of the signals and systems textbook). Of course, its significant spectral amplitude values will be in the low frequency range and tapers off as we move towards the high frequency range. The message signal is a low frequency signal. Multiplication of the two for generating the PAM signal results in the convolution of their spectra in the frequency domain. Thus PAM signal still retains the message spectrum in the low frequency range after modulation. This is the difference between amplitude modulation of sine wave and pulse train. Therefore, PAM is not useful like AM for communication. Alternatively, PAM is found to be useful in understanding the *sampling process* to be described next.

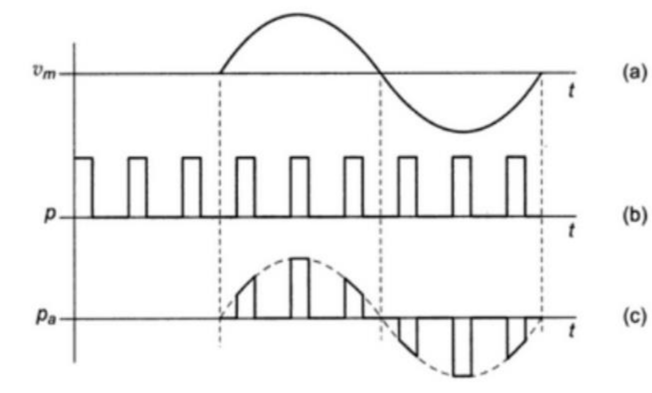


Fig. 4.8 *Generation of PAM signal: (a) Message, (b) Pulse train, and (c) PAM.*

Sampling Process Sampling is a signal processing operation that helps in sensing the continuous time signal values at discrete instants of time. The sampled sequence will have amplitudes equal to signal values at the sampling instants and undefined at all other times. This process can be conveniently performed using PAM described above. The sampling process can be treated as an electronic switching action as shown in Fig. 4.9 The continuous time signal to be sampled is applied to the input terminal. The pulse train is applied as the control signal of the switch. When the pulse occurs, the switch is in ON condition, that is, acts as short circuit between input and output terminals. The output value will therefore be equal to input. During the other intervals of the pulse train, the switch is in OFF condition, that is, acts as open circuit. The output is therefore undefined. The output of the switch will be essentially a PAM signal. Any active device like diode, transistor or FET can be used as a switch.

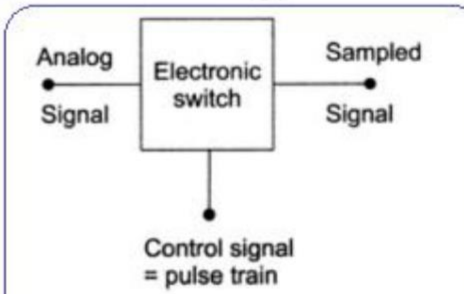


Fig. 4.9 *Illustration of sampling process.*

In the context of sampling process, there are other aspects that need to be considered with respect to the pulse train. The first and foremost is how often the signal needs to be sampled or sensed, so that when needed an approximate version of the continuous time signal can be reconstructed. This is based on the well known *sampling theorem* which states that *the sampling frequency (F_s) i.e., number of samples per second should be greater than or equal to twice the maximum frequency component (F_m) of the input signal.*

$$F_s \geq 2F_m \tag{4.7}$$

The minimum possible value of sampling frequency is termed as *Nyquist rate*. Thus the sampling theorem will decide the periodicity associated with the pulse train. The second important aspect is, the width of the pulse Δ should not infleunce the amplitude of the sampled value. Even though this point is not obvious in the time doman, it can be understood by observing the frequency domain behavior of the PAM process due to the convolution of sinc function of pulse train with the input signal spectrum. To minimize this effect, for all practical processing $\Delta \rightarrow 0$, so that the pulse train becomes on impulse train. The Fourier transform of an impulse train is also an impulse train in the frequency domain. Therefore convolution will not affect the shape of the sampled signal. It only leads to periodicity of the spectrum!

Example 1

A message signal made of multiple frequency components has a maximum frequency value of 4 kHz. Find out the minimum sampling frequency required according to the sampling theorem.

Solution

$$F_m = 4 \text{ kHz}$$
$$F_s \geq 2 \times F_m = 2 \times 4 \text{ kHz} = 8 \text{ kHz}$$

Example 2

A message signal has the following frequency components: a single tone sine wave of 500 Hz and sound of frequency components with lowest value of 750 Hz and highest value of 1800 Hz. What should be the minimum sampling frequency to sense the information present in this signal according to the sampling theorem?

Solution

$$F_m = 1800 \text{ Hz}$$
$$F_s \geq 2 \times F_m = 3600 \text{ Hz}$$

Classification of PAM based on Signal Polarity

The PAM signal can be classified according to signal polarity.

- Single polarity PAM
- Double polarity PAM

The Fig. **4.9.a** shows the single polarity PAM. Here, a fixed d.c. level is added to the modulating signal x(t), such that the modulated output i.e. PAM signal is always positive.

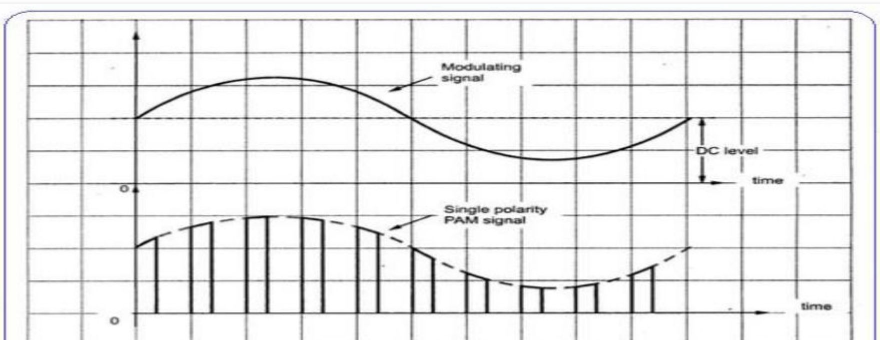


Fig. 4.9.a Single polarity PAM signal

A PAM Modulator Circuit (Practical circuit)

A PAM modulator circuit is shown in Fig. **4.10** This circuit is a simple emitter follower. In the absence of the clock signal, the output follows the input. The modulating signal is applied as the input signal. Another input to the base of the transistor is the clock signal. The frequency of the clock signal is made equal to the desired carrier pulse train frequency. The amplitude of the clock signal is so chosen that the high level is at ground (*O* V), and the low level is at some negative voltage which is sufficient to bring the transistor in the cut-off region. Thus, when the clock signal is high, the circuit behaves as an emitter follower, and the output follows the input modulating signal. When the clock signal is low, the transistor is cut-off and the output is zero. Thus the output waveform, shown in Fig. **4.10** is the desired pulse amplitude modulated signal.

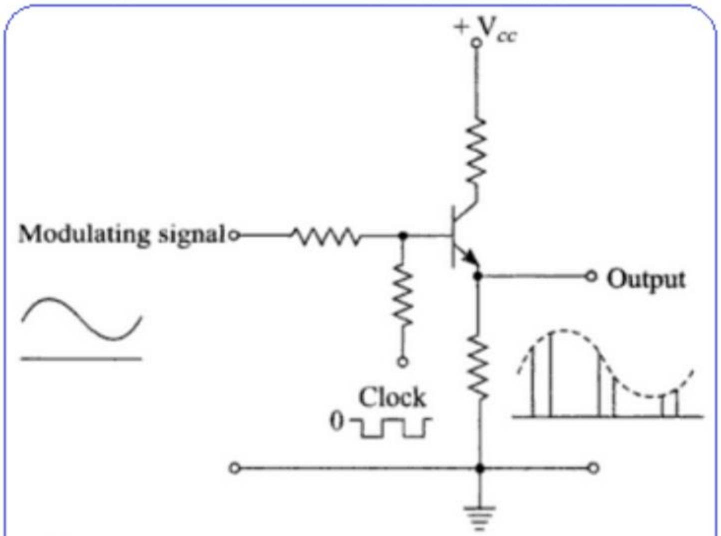


Fig. 4.10 *A PAM Modulator*

A PAM Demodulator Circuit

A PAM demodulator circuit is shown in Fig. **4.11** It is just an envelope detector followed by a low pass filter. The diode and R-C combination work as the envelope detector. This is followed

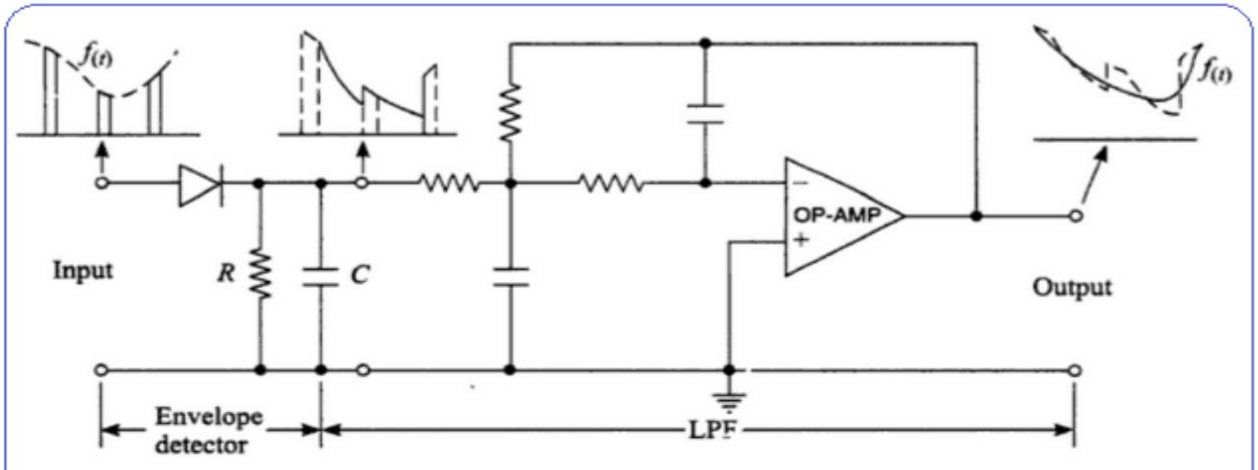


Fig. 4.11 *A PAM Demodulator*

by a second order OP-AMP low pass filter to have a good filtering characteristic. Thus, for the received pulse amplitude modulated signal as the input signal, the desired demodulated signal [i.e. the baseband signal $f(t)$] shown in Fig. **4.11** is the output.

Time Division Multiplexed PAM System

Normally, in a PAM system, the duration of the pulse (τ) is much less than the time period of pulses T_s, i.e., $\tau << T_s$, as shown in Fig. **4.12** (a). Thus, no information is being transmitted through the system for most of the time. The time space $T_s - \tau$ can be utilized to transmit information from

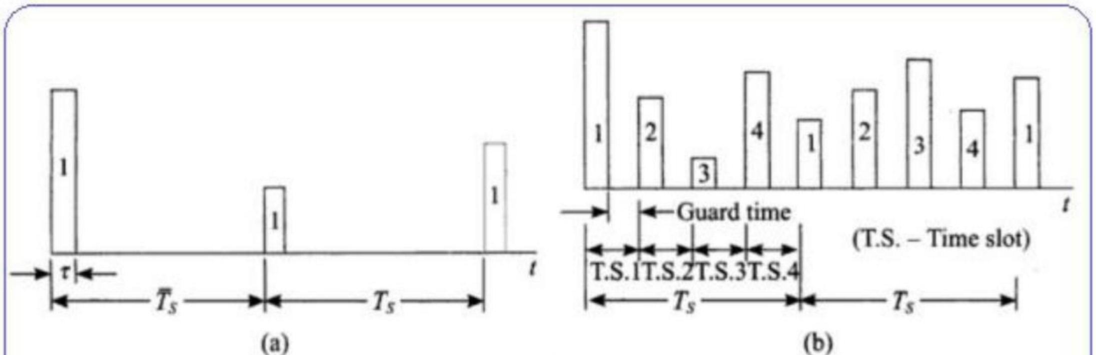


Fig. 4.12

other signals. In Fig. 4.12 (b), the signal numbers 2, 3 and 4 are transmitting information with the help of samples numbered 2, 3 and 4, respectively. This is alongwith the samples numbered 1 of the signal number 1. The time period T_s is equally divided between the four signals, thus allocating a time slot of $\frac{T_s}{4}$ to each signal. The duration of time slot is such that $\frac{T_s}{4} > \tau$. Thus, there is a guard time $\frac{T_s}{4} - \tau$ between all successive sampling pulses, ensuring that there is less cross-talk between signals. (More about cross-talk at a later stage in this section.) The arrangement by which the information from more than one signal is transmitted in this manner is known as time division multiplexing (TDM). A time division multiplexed PAM system is shown in Fig. 4.13 which transmits information from n signals. The switch 1 and switch 2, respectively known as commutator and decommutator, are synchronized electronic switches which rotate at the same speed of $2f_M$ rotations per second. The commutator samples and combines the samples, while the decommutator separates the samples belonging to individual signals.

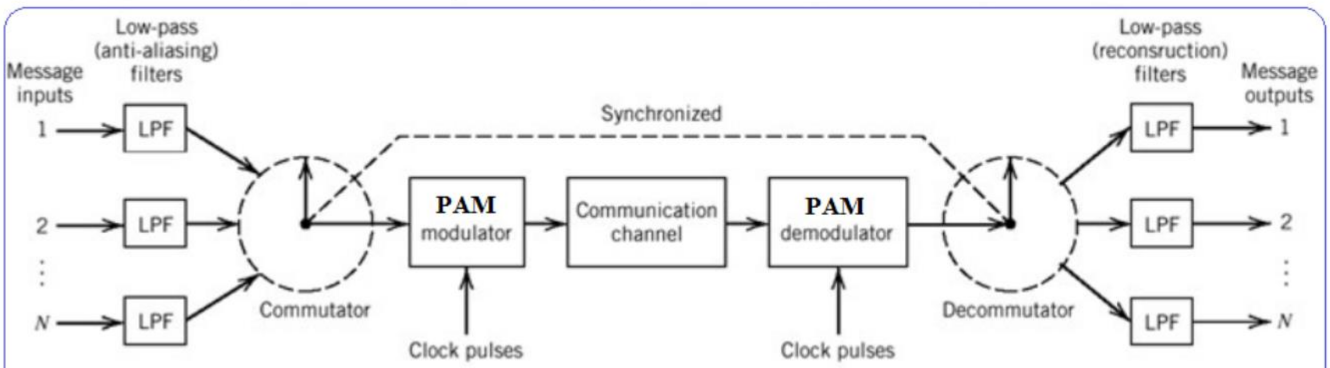


Fig 4.13 **PAM TDM System**

Synchronization is the most crucial operation in any TDM system. Thus, for example, if the commutator is at position 2, the decommutator must also be in position 2. To provide synchronization, a synchronizing pulse is transmitted in every frame (time interval between two successive samples of the same signal, i.e. T_s). Thus to multiplex n channels, $n + 1$ time slots are provided in a frame; n for channels and 1 for the synchronizing pulse. The synchronizing pulse is chosen in such a way that it is easily distinguishable.

Pulse Width Modulation

Pulse width modulation (PWM) is defined as the process of varying the width of the pulse in proportion to the instantaneous variations of message.

Let Δ be the width of the pulse in the unmodulated pulse train. In PWM

$$\Delta \propto v_m \quad \textbf{(4.8)}$$

Mathematically, the width of pulse in PWM signal is given by

$$\Delta_m = \Delta(1 + v_m). \quad \textbf{(4.9)}$$

When there is no message, i.e., $v_m = 0$, then the width of the pulse will be equal to the original width Δ. For positive values of message, the width will be proportionately increases by $(1 + v_m)$ factor. For negative values of message, the width decreases by $(1 - v_m)$ factor.

Figure 4.14 shows the generation of PWM signal. The amplitude of the pulse remains constant in this case. Thus PWM is more roubst to noise compared to PAM. This is the difference with respect to PAM signal. The mathematical treatement about the frequency domain aspect of PWM is an involved process. However, the resulting PWM will still have the spectrum in the baseband region itself. The illustration given in **Fig. 4.14** is made only using trailing edge of the pulse. We can also perform the same using either leading edge or both.

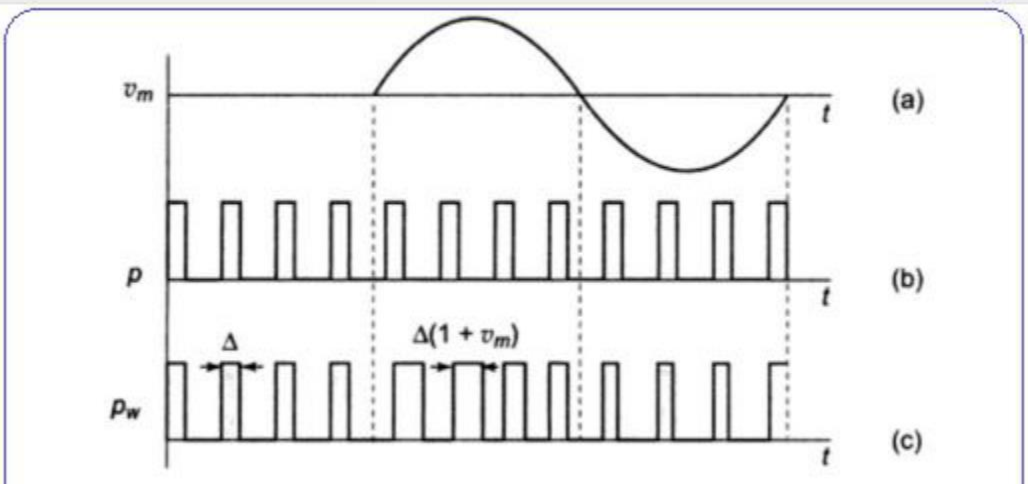


Fig. 4.14 *Generation of PWM signal. (a) Message, (b) pulse train and (c) PWM.*

Practical PWM Generator Circuits

Fig. 4.15 (a) shows pulse width modulator. It is basically a monostable multivibrator with a modulating input signal applied at the control voltage input. Internally, the control voltage is adjusted to the 2/3 V_{CC}. Externally applied modulating signal changes the control voltage, and hence the threshold voltage level. As a result, time period required to charge the capacitor up to threshold voltage level changes, giving pulse width modulated signal at the output.

Fig. **4.15.b** shows another monostable multivibrator circuit to generate pulse width modulation. The stable state for above circuit is T_1 OFF and T_2 ON. The positive going trigger pulse at B_1 switches T_1 ON. Due to this the voltage at C_1 falls as T_1 now begins to draw the collector current. As a result, voltage at B_2 also falls and T_2 is

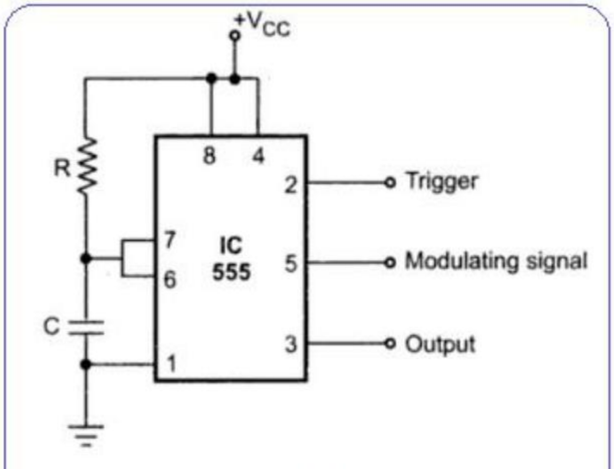


Fig. 4.15 (a)

switched OFF, C determined by the supply voltage and the RC time constant of the charging network, the base of the T_2 becomes sufficiently positive to switch T_2 ON. The transistor T_1 is simultaneously switched OFF by regenerative action and stays OFF until the arrival of the next trigger pulse. To make T_2 ON, the base of the T_2 must be slightly more positive than the voltage across resistor R_E. This voltage depends on the

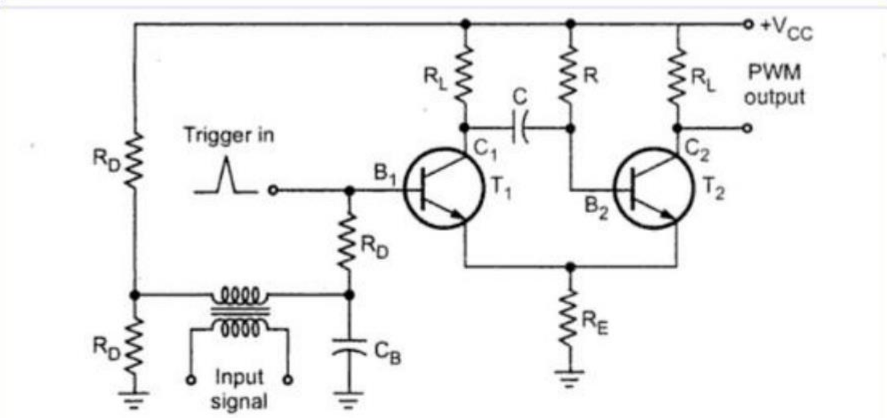


Fig. 4.15.b Monostable multivibrator generating pulsewidth modulation

emitter current I_E which is controlled by the signal voltage applied at the base of transistor T_1. Therefore, the changing voltage necessary to turn OFF transistor T_2 is decided by the signal voltage. If signal voltage is maximum, the voltage that capacitor should charge to turn ON T_2 is also maximum. Therefore, at maximum signal voltage, capacitor has to charge to maximum voltage requiring maximum time to charge. This gives us maximum pulse width at maximum input signal voltage. At minimum signal voltage, capacitor has to charge for minimum voltage and we get minimum pulse width at the output. With this discussion it can be noted that pulse width is controlled by the input signal voltage, and we get pulse width modulated waveform at the output.

Demodulation of PWM Signal

Fig. 4.16 (a) shows the block diagram of PWM detector. As shown in the Fig. 4.16 (a), the received PWM signal is applied to the schmitt trigger circuit. The schmitt trigger circuit removes the noise in the PWM waveform. The regenerated PWM is then applied to the ramp generator and the synchronization pulse generator. The ramp generator produces ramps for the duration of pulses such that height of ramps are proportional to the widths of PWM pulses. The maximum ramp voltage is retained till the next pulse. On the other hand synchronous pulse generator produces reference pulses with constant amplitude and pulse width. These pulses are delayed by specific amount of delay as shown in the Fig. 4.16 (b). The delayed reference pulses and the output of ramp generator is added with the help of adder. The output of adder is given to the level shifter. Here, negative offset waveform is clipped by rectifier.

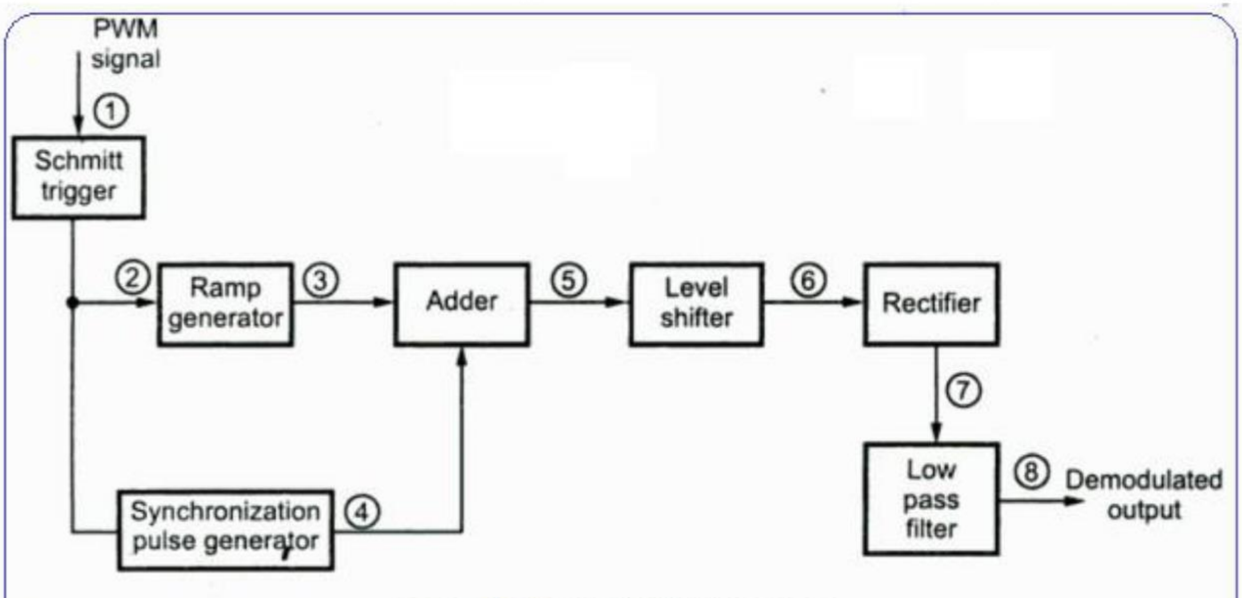


Fig. 4.16 (a) PWM detector

Finally, the output of rectifier is passed through low-pass filter to recover the modulating signal, as shown in the Fig. 4.16 (b) (8).

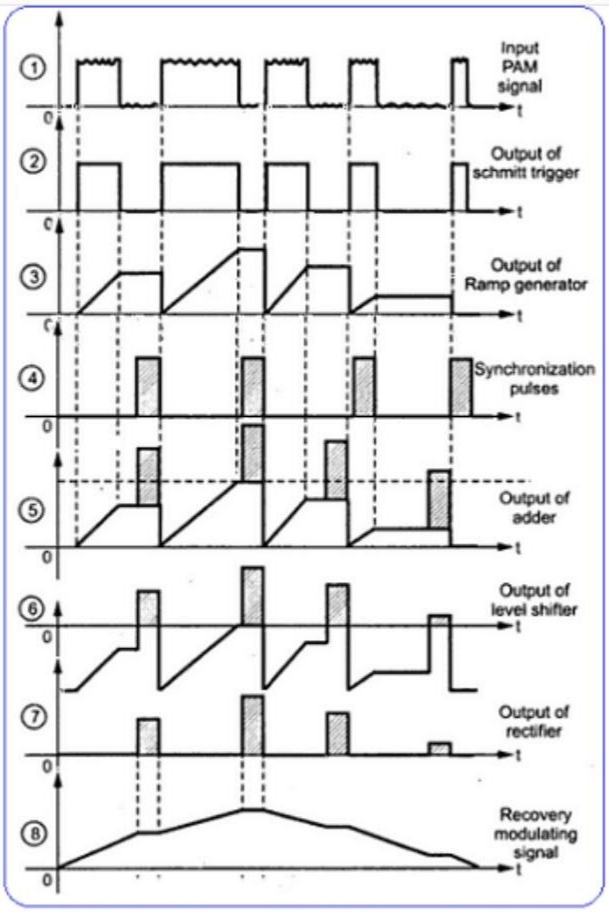
Advantages of PWM

1. Unlike, PAM, noise is less, since in PWM, amplitude is held constant.
2. Signal and noise separation is very easy, as shown in Fig. 4.16 (b) (2).
3. PWM communication does not require synchronization between transmitter and receiver.

Disadvantages of PWM

1. In PWM, pulses are varying in width and therefore their power contents are variable. This requires that the transmitter must be able to handle the power contents of the pulse having maximum pulse width.

2. Large bandwidth is required for the PWM communication as compared to PAM.



4.16 b Waveforms of PWM detection circuit

Note:

To have a better separation with respect to frequency, between highest frequency of baseband signal and lower sidebands of f_s (sampling frequency), a higher sampling frequency which is more than Nyquist rate is used; and pulse width deviation is kept small.

PAM Demodulator Circuit

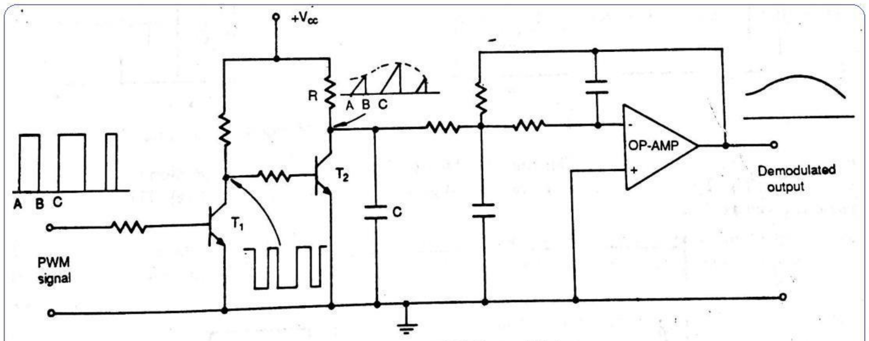


Fig. 4.16 C A PWM Demodulator

Fig 4.16 c shows the circuit diagram of the PWM demodulator. In this the transistor T_1 acting as a driver to T_2 and also provides the inverted version of the input to the base of T_2. When input is in on condition (between AB) the base drive to T_2 is inverted and hence it will be in cutoff. During this period the capacitor C charges through R which builds up an exponential (nearly ramp) voltage at the collector of T_2. During the period BC of the input T_1 is off and its collector voltage becomes high which drives the T_2 into saturation. This provides T_2 an instant discharge path and hence it completely discharges making the T_2 collector voltage as saw tooth. It remains in the same state till the period C and after that it starts charging and the process repeats. Thus PWM is converted in to varying height PAM signal. The op-amp circuit which follows the capacitor is second order low pass filter demodulates the PAM signal.

Pulse Position Modulation

Pulse position modulation (PPM) is defined as the process of varying the position of the pulse with respect to the instantaneous variations of the message signal.

Let t_p indicates the timing instant of the leading or trailing edge of the pulse in each period of the pulse train. In PPM

$$t_p \propto v_m \tag{4.10}$$

Mathematically, the position of the leading or trailing edge of the pulse (in each period) in PPM signal is given by

$$t_p = f(v_m) \tag{4.11}$$

When there is no message, then the position of the leading or trailing edge of the pulse will be equal to the original position and hence $t_p = 0$. For positive values of message, the position will be proportionately shifted right by $t_p = f(v_m)$. For negative values of message, the position will be proportionately shifted left by $-t_p = -f(v_m)$ factor. One way of generating PPM is to generate PWM and postprocess the same to get PPM.

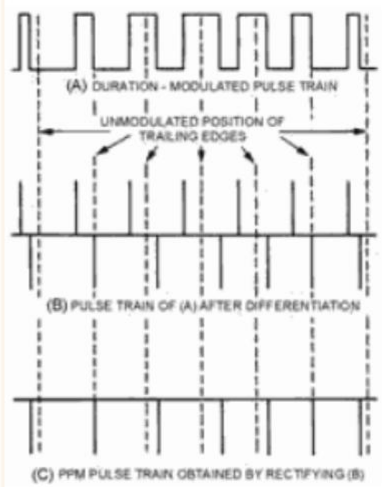


Figure 4.17.—Pulse-position modulation (PPM).

PPM can be generated in several ways, but we will discuss one of the simplest. Figure 4.17 shows three waveforms associated with developing PPM from PDM. The PDM pulse train is applied to a differentiating circuit. This providespositive- and negative-polarity pulses that correspond to the leading and trailing edges of the PDM pulses. Considering PDM and its generation, you can see that each pulse has a leading and trailing edge. In this case the position of the leading edge is fixed, whereas the trailing edge is not, as shown in view (A) of figure 4.17 The resultant pulses after the differentiation are shown in view (B). The negative pulses are position-modulated in accordance with the modulating waveform. Both the negative and positive pulse are then applied to a rectification circuit. This application eliminates the positive, non-modulated pulses and develops a PPM pulse train, as shown in view (C).

Generation of PPM Signal

The Fig. 4.18 shows the block diagram of PPM generator. It consists of PWM generator followed by the monostable multivibrator. Since, in PPM, output remains high for fix duration from the trailing edges of the PWM signal, the trailing edge of the PWM signal is used as a trigger input for the monostable multivibrator.

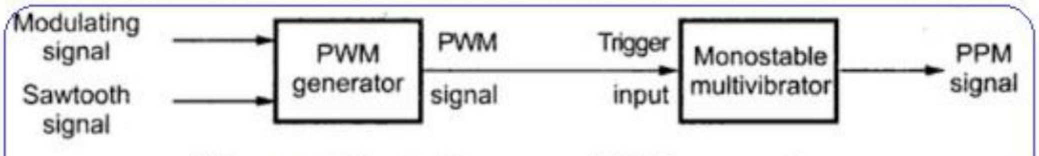


Fig. 4.18 Block diagram of PPM generator

Practical PPM Generator Circuit

Fig. 4.19 (a) shows the PPM generator. It consists of differentiator and a monostable multivibrator. The input to the differentiator is a PWM waveform. The differentiator generates positive and negative spikes corresponding to leading and trailing edges of the PWM waveform. Diode D_1 is used to bypass the positive spikes. The negative spikes are used to the trigger monostable multivibrator. The monostable multivibrator then generates the pulses of same width and amplitude with reference to trigger to give pulse position modulated waveform, as shown in the Fig. 4.19 (b).

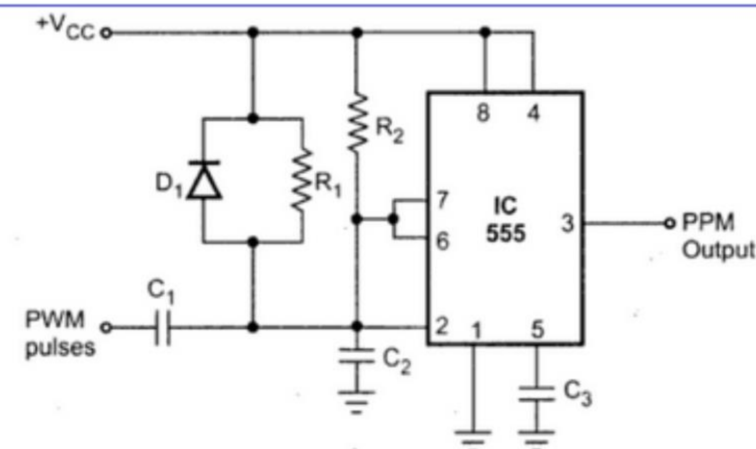


Fig. 4.19 (a) PPM generator

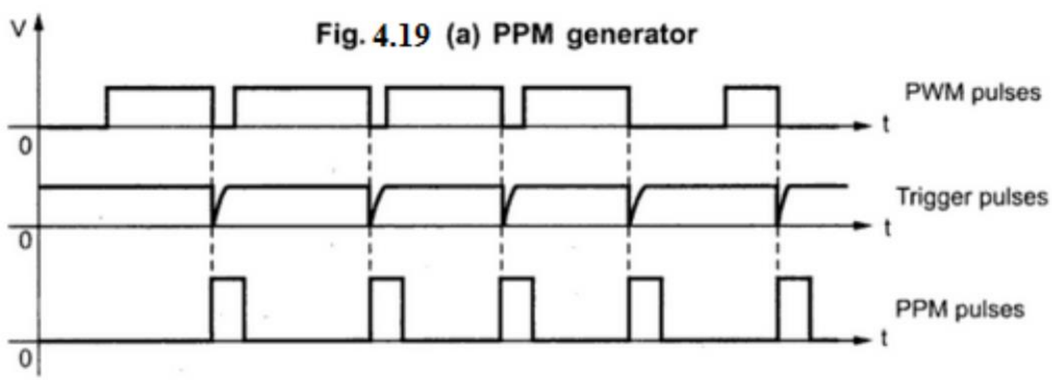


Fig. 4.19 (b) Waveforms of PPM generator

Demodulation of PPM Signal

In the case of pulse-position modulation, it is customary to convert the received pulses that vary in position to pulses that vary in length. One way to achieve this is illustrated in Fig. **4.20**

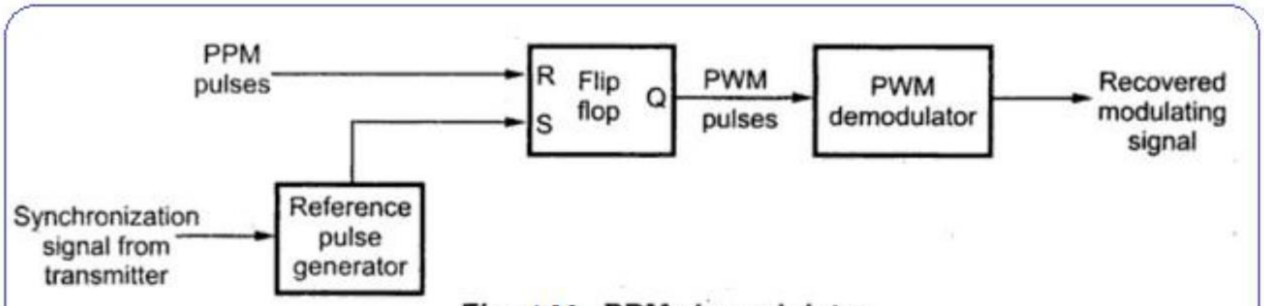


Fig. 4.20 PPM demodulator

As shown in the Fig. **4.20** flip-flop circuit is set or turned 'ON' (giving high output) when the reference pulse arrives. This reference pulse is generated by reference pulse generator of the receiver with the synchronization signal from the transmitter. The flip-flop circuit is reset or turned 'OFF' (giving low output) at the leading edge of the position modulated pulse. This repeats and we get PWM pulses at the output of the flip-flop.

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

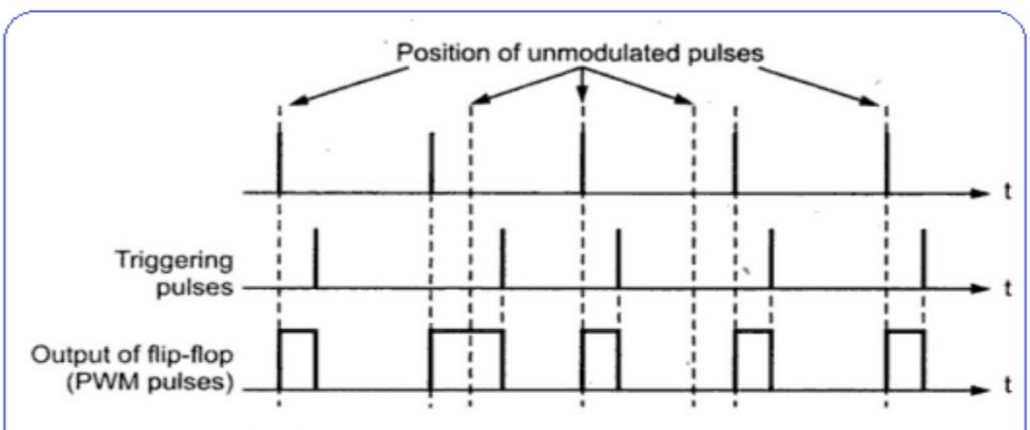


Fig. 4.21 Demodulation waveforms for PPM

2.4.3 Advantages of PPM

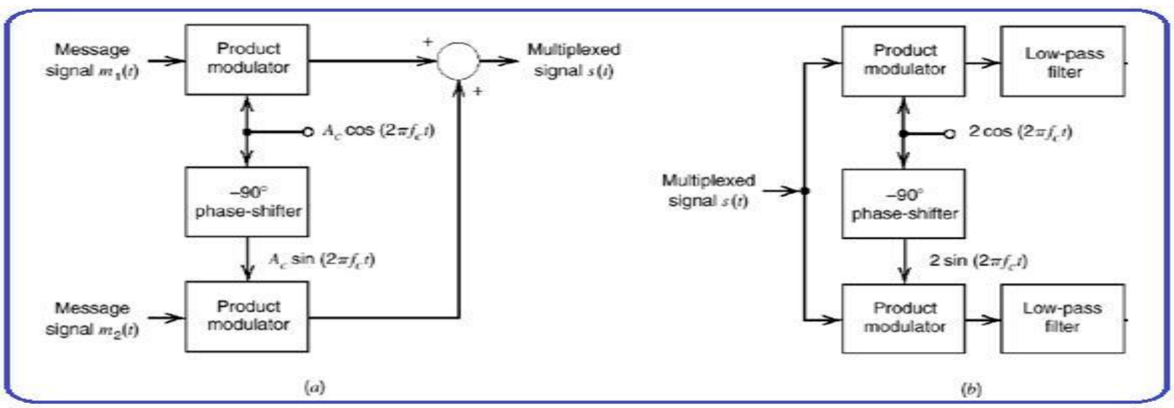
1. Like PWM, in PPM amplitude is held constant thus less noise interference.
2. Like PWM, signal and noise separation is very easy.
3. Due to constant pulse widths and amplitudes, transmission power for each pulse is same.

Quadrature Multiplexing

Quadrature multiplexing is an efficient way of transmitting two signals that are located within the same frequency band. If the input signals $m_i(t)$, $i = 1, 2$ have bandwidths B_i, then the transmission bandwidth of the QAM signal is given by $2B_{\max}$, $B_{\max} = \max(B_i)$. We are transmitting two signals in the same band and therefore we have saving in the transmission bandwidth. The quadrature multiplexing can be used for both DSB and SSB siganls. Non-coherent demodulation of QM is not possible since the envelope of the modulated signal is $A_c\sqrt{[m_1^2(t) + m_2^2(t)]}$. Quadrature multiplexing is very useful and has significant number of applications. One, in particular, is the use of QM for transmission of color signals in commercial television broadcasting.

Note: QAM is referred to as *frequency-domain multiplexing* rather than frequency-division multiplexing as the modu lated spectra of the signals in QM overlap rather than apart.

A Quadrature Multiplexing or Quadrature Amplitude Modulation (QAM) method enables two DSBSC modulated waves, resulting from two different message signals to occupy the same transmission band width and two message signals can be separated at the receiver. The transmitter and receiver for QCM are as shown below fig (a) and (b)



The transmitter involves the use of two separate product modulators that are supplied with two carrier waves of the same frequency but differing in phase by -90°. The multiplexed signal s(t) consists of the sum of the two product modulator outputs given by the equation 4.12.

$$s(t) = A_c m_1(t)\cos(2\pi f_c t) + A_c m_2(t)\sin(2\pi f_c t) \qquad \text{----------} (4\text{-}12)$$

where m1(t) and m2(t) are two different message signals applied to the product modulators. Thus, the multiplexed signal s(t) occupies a transmission band width of 2W, centered at the carrier frequency fc where W is the band width of message signal m1(t) or m2(t), whichever is larger.

At the receiver the multiplexed signal s(t) is applied simultaneously to two separate coherent detectors that are supplied with two local carriers of the same frequency but differing in phase by 90°. The output of the top detector is 1/2 [$A_c m_1(t)$] and that of the bottom detector is 1/2 [$A_c m_2(t)$]

For the QCM system to operate satisfactorily, it is important to maintain correct phase and frequency relationships between the local oscillators used in the transmitter and receiver parts of the system.

AM Receivers: TRF receivers -Super heterodyne receivers: choice of IF, double conversion technique, tracking, AGC- characteristics of receiver - noise in AM receiver. FM Receivers: FM stereo broadcast receivers - AFC - Noise in FM- Capture effect, FM threshold effect. Communication Receivers: Sensitivity, fidelity and selectivity - Squelch circuit – Beat frequency Oscillator. Overview of Telephony, Telegraphy, Television,CCTV and Cable television.

Introduction

In the previous chapters, we have seen how information is modulated and transmitted with the help of modulator and transmitter. The transmitter transmits the modulated carrier through the transmitting antenna to the air. The modulating signal thus propagated is collected by the receiving antenna. In this chapter, we are going to see the receiver section of the AM and FM wave communication.

Functions of Receiver

The primary requirement of any communication receiver is that it should have the ability to select the desired signal from among thousands of others present and to provide sufficient amplification to recover the modulating signal. To provide this primary requirement receiver has to carryout different functions as given below.

1. Collect the electromagnetic waves transmitted by the transmitter.
2. Select the desired signal and reject all others.
3. Amplify the selected modulated carrier signal.
4. Detect the modulating signal from the modulated RF signal.
5. Amplify the modulating signal to operate the loudspeaker.

CHARACTERISTICS OF A RADIO RECEIVER

There are three main characteristics by which the quality of a receiver can be judged. These are *selectivity*, *sensitivity* and *fidelity*. These are also known as the performance characteristics or the specifications of a receiver.

Selectivity Selectivity of a receiver is its ability to select a desired signal frequency without any objectionable interference from other neighbouring stations. It is a measure of the extent to which the receiver can reject all other neighbouring stations and accept only the desired station. A good, selective receiver will select the desired station and reject all other unwanted stations. The selectivity is generally expressed in the form of a curve shown in Fig. 5.1 In this curve, the strength of the input signal at the resonant frequency required to produce a given output is taken as the reference and the strength of the modulated carrier at neighbouring frequencies required to produce the same output is plotted on the vertical axis. The sharper the selectivity curve, the more selective a receiver is.

The sensitivity of a receiver is expressed in microvolts. The smaller the input in microvolts the greater is the sensitivity of a receiver. A high grade broadcast receiver will have a sensitivity of less than 10 μV. The sensitivity curve for a standard broadcast receiver is shown in Fig. 5.2

Fidelity Fidelity is the ability of a receiver to reproduce faithfully all the audio frequencies with which the carrier is modulated. This is generally expressed as a frequency response curve shown in Fig. 5.3

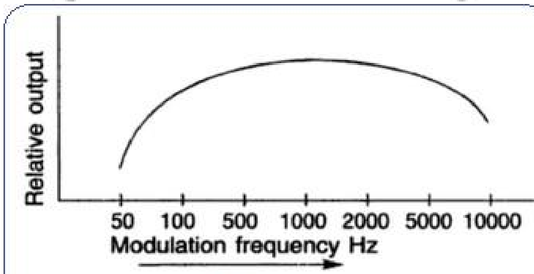


Fig. 5.3 Fidelity Curve

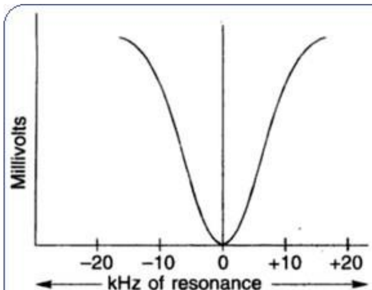


Fig. 5.1 Selectivity Curve

Sensitivity Sensitivity of a receiver is its ability to respond to weak signals. This is expressed as the minimum voltage or power that must be applied to the input of the receiver for getting a standard output of 0.5 W in the loudspeaker

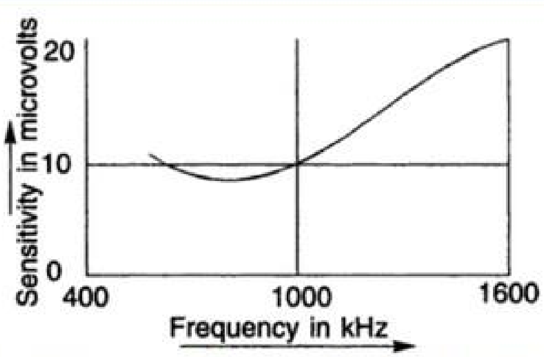


Fig 5.2 Sensitivity curve for a Standard Broadcast Receiver

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DEPARTMENT OF ELECTRONICS AND COMMUNICATION

Subject Name: Analog Communication UNIT V Subject Code:SEC1209

School of Electrical and Electronics

Other Important desired characteristics of any radio receiver are Stability, Noise Figure and Image rejection ratio.

Stability

Stability of a radio receiver is defined as the ability of the receiver to maintain the standard output produced, irrespective of the (unintentional) variations that is occurring in the input (due to noise or any other reasons)

Noise Figure

NOISE FIGURE

The noise figure (NF) frequently is used to measure the receiver's "goodness"; that is, its departure from the ideal. Therefore, it is a figure of merit. The noise figure is the noise factor converted to decibel notation:

$$NF = 10 \log (F_N)$$

where

NF is the noise figure in decibels (dB);
F_N is the noise factor;
log refers to the system of base-10 logarithms.

NOISE FACTOR F_N, in terms of the output and input signal-to-noise ratios:

$$F_N = \frac{S_{NI}}{S_{NO}}$$

where

S_{NI} is the input signal-to-noise ratio;
S_{NO} is the output signal-to-noise ratio.

Image Frequency and its Rejection

In standard broadcast receiver the local oscillator frequency is made higher than the signal frequency by an amount equal to intermediate frequency (IF). Therefore $f_o = f_s + f_i$. When f_o and f_s are mixed, the difference frequency, which is one of the by products, is equal to f_i only f_i is passed and amplified by the IF stage.

If a frequency f_{si} $(f_o + f_i)$, i.e. $f_{si} = f_s + 2\,f_i$, appears at the input of the mixer then it will produce the sum and difference frequencies regardless of the inputs. Therefore, the mixer output will be the difference frequency at the IF value. The terms f_{si} is called the **image** frequency and is defined as the signal frequency plus twice the intermediate frequency. Unfortunately, this image frequency signal is also amplified by the IF amplifiers resulting in interference. This has the effect of two stations being received simultaneously and is naturally undesirable.

The **image rejection ratio of** a single tuned circuit is **defined as** the ratio of the gain at the signal frequency to the gain at the image frequency. It is given by

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

where

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

and Q = loaded Q of tuned circuit

If the receiver has an RF stage, then there are two tuned circuits, both tuned to f_s ; the rejection of each will be calculated by the same formula, and the total rejection will be the product of two.

The image rejection depends on the selectivity of the RF amplifier and tuned circuits and must be achieved before the IF stage. Once the spurious frequency enters the first IF amplifier, it becomes impossible to remove it from the wanted signal.

RECEIVER TYPES

Of the various forms of receivers proposed at one time or another, only two have any real practical or commercial significance—the tuned radio-frequency (TRF) receiver and the superheterodyne receiver. Only the second of these is used to a large extent today, but it is convenient to explain the operation of the TRF receiver first since it is the simpler of the two. The best way of justifying the existence and overwhelming popularity of the superheterodyne receiver is by showing the shortcomings of the TRF type.

Tuned Radio-Frequency (TRF) Receiver

The TRF receiver block diagram is shown in Fig. 5.4. The TRF receiver is a simple "logical" receiver. A person with just a little knowledge of communications would probably expect all radio receivers to have this form. The virtues of this type, which is now not used except as a fixed-frequency receiver in special applications, are its simplicity and high sensitivity.

Two or perhaps three RF amplifiers, all tuning together, were employed to select and amplify the incoming frequency and simultaneously to reject all others. After the signal was amplified to a suitable level, it was demodulated (detected) and

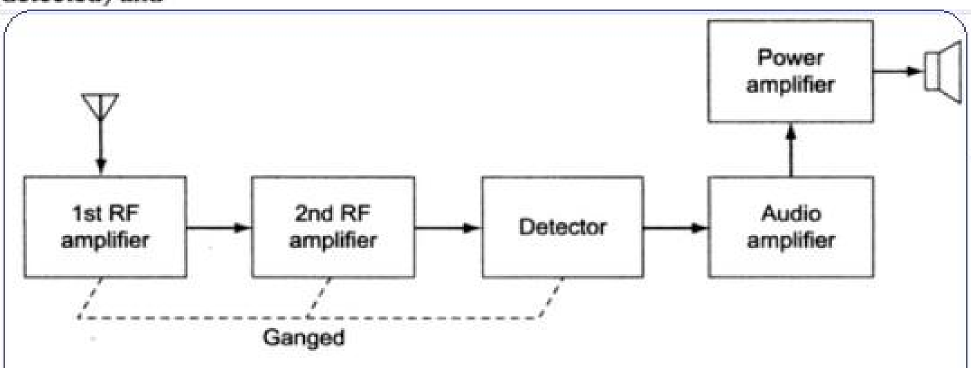


Fig. 5.4 *The TRF receiver*

fed to the loudspeaker after being passed through the appropriate audio amplifying stages. Such receivers were simple to design and align at broadcast frequencies (535 to 1640 kHz), but they presented difficulties at higher frequencies. This was mainly because of the instability associated with high gain being achieved at one frequency by a multistage amplifier. In addition the TRF receiver suffered from a variation in bandwidth over the tuning range. It was unable to achieve sufficient selectivity at high frequencies, partly as a result of the enforced use of single-tuned circuits.

Problems in TRF Receivers

1. Tracking of Tuned Circuit

In a receiver, tuned circuits are made variable so that they can be set to the frequency of the desired signal. In most of the receivers, the capacitors in the tuned circuits are made variable. These capacitors are 'ganged' between the stages so that they all can be changed simultaneously when the tuning knob is rotated. To have perfect tuning the capacitor values between the stages must be exactly same but this is not the case. The differences in the capacitors cause the resonant frequency of each tuned circuit to be slightly different, thereby increasing the pass band.

2. Instability

As high gain is achieved at one frequency by a multistage amplifier, there are more chances of positive feedback (of getting back the small part of output of the last stage at the input to the first with the correct polarity) through some stray path, resulting in oscillations. These oscillations are unavoidable at high frequencies.

3. Variable Bandwidth

TRF receivers suffer from a variation in bandwidth over the tuning range. Consider a medium wave receiver required to tune over 535 kHz to 1640 kHz and it provides the necessary bandwidth of 10 kHz at 535 kHz. Let us calculate Q of this circuit.

$$Q = \frac{f}{\text{Bandwidth}} = \frac{535\,\text{kHz}}{10\,\text{kHz}} = 53.5$$

Now consider the frequency at the other end of the broadcast band, i.e. 1640 kHz. At 1640 kHz, Q of the coil should be 164 (1640 kHz / 10 kHz). However, in practice due to various losses depending on frequency, we will not set so large increase in Q. Let us assume that at 1640 kHz frequency Q is increased to value 100 instead of 164. With this Q of the tuned circuit bandwidth can be calculated as follows

$$\text{Bandwidth} = \frac{f}{Q} = \frac{1640\,\text{kHz}}{100} = 16.4\text{ kHz}$$

We know, necessary bandwidth is 10 kHz. This increase in bandwidth of tuned circuit, pick up the adjacent stations along with station it is tuned for, providing insufficient adjacent frequency rejection. In other words we can say that in TRF receivers the bandwidth of the tuned circuit varies over the frequency range, resulting in poor selectivity of the receiver.

Because of the problems of tracking, instability and bandwidth variation, the TRF receivers have almost been replaced by superheterodyne receivers.

Superheterodyne Receivers

To solve basic problems of TRF receivers, in these receivers, first all the incoming RF frequencies are converted to a fix lower frequency called **intermediate frequency** (IF). Then this fix intermediate frequency is amplified and detected to reproduce the original information. Since the characteristics of the IF amplifier are independent of the frequency to which the receiver is tuned, the selectivity and sensitivity of superheterodyne receivers are fairly uniform throughout its tuning range.

Mixer circuit is used to produce the frequency translation of the incoming signal down to the IF. The incoming signals are mixed with the local oscillator frequency signal in such a way that a constant frequency difference is maintained between the local oscillator and the incoming signals. This is achieved by using ganged tuning capacitors.

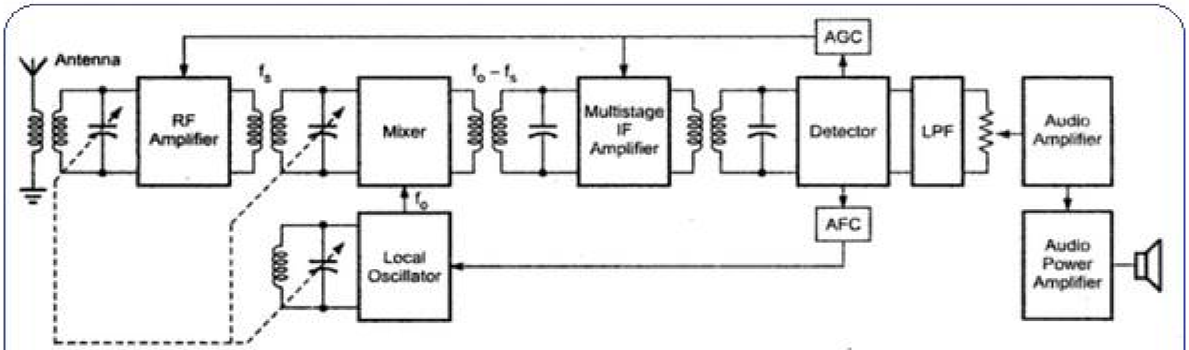


Fig. 5.5 Block diagram of superheterodyne receiver

Fig.5.5 shows the block diagram of superheterodyne receiver. As shown in the Fig.5.5 antenna picks up the weak radio signal and feeds it to the RF amplifier. The RF amplifier provides some initial gain and selectivity. The output of the RF amplifier is applied to the input of the mixer. The mixer also receives an input from local oscillator.

The output of the mixer circuit is difference frequency ($f_o - f_s$) commonly known as IF (Intermediate Frequency). The signal at this intermediate frequency contains the same modulation as the original carrier. This signal is amplified by one or more IF amplifier stages, and most of the receiver gain is obtained in these IF stages.

The highly amplified IF signal is applied to detector circuits to recover the original modulating information. Finally, the output of detector circuit is fed to audio and power amplifier which provides a sufficient gain to operate a speaker.

Another important circuit in the superheterodyne receiver are AGC and AFC circuit. AGC is used to maintain a constant output voltage level over a wide range of RF input signal levels.

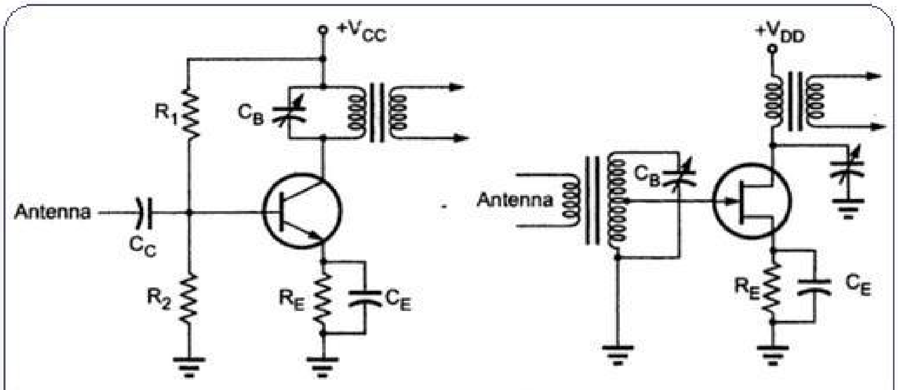
It derives the dc bias voltage from the output of detector which is proportional to the amplitude of the received signal. This dc bias voltage is feedback to the IF amplifiers, and sometimes to the RF amplifier, to control the gain of the receiver. As a result, it provides a constant output voltage level over a wide range of RF input signal levels. AFC circuit generates AFC signal which is used to adjust and stabilize the frequency of the local oscillator.

AM Receivers

In the above section we have seen the principle of superheterodyne receiver. In various forms of modulation same principle is used. In this section we are going to see the detail circuits of each block in the receiver.

RF Amplifier

As mentioned earlier, RF amplifier provides initial gain and selectivity. Fig. shows the RF amplifier circuits. It is a tuned circuit followed by an amplifier The RF amplifier is usually a simple class A circuit. A typical bipolar circuit is shown in Fig. 5.6 (a) , and a typical FET circuit is shown in Fig. 5.6 (b).



(a) Fig. 5.6 (b)

The values of resistors R_1 and R_2 in the bipolar circuit are adjusted such that the amplifier works as class A amplifier. The antenna is connected through coupling capacitor to the base of the transistor. This makes the circuit very broad band as the transistor will amplify virtually any signal picked up by the antenna. However the collector is tuned with a parallel resonant circuit to provide the initial selectivity for the mixer input.

The FET circuit shown in Fig. 5.6 (b) is more effective than the transistor circuit. Their high input impedance minimizes the loading on tuned circuits, thereby permitting the Q of the circuit to be higher and selectivity to be sharper.

The receiver having an RF amplifier stage has following advantages :

1. It provides greater gain, i.e. better sensitivity.
2. It improves image-frequency rejection.
3. It improves signal to noise ratio.
4. It improves rejection of adjacent unwanted signals, providing better selectivity.
5. It provides better coupling of the receiver to the antenna.
6. It prevents spurious frequencies from entering the mixer and heterodyning there to produce an interfering frequency equal to the IF from the desired signal.
7. It also prevents reradiation of the local oscillator through the antenna of the receiver.

Mixer or Frequency Converter

The frequency converter is a nonlinear resistance having two sets of input terminals and one set of output terminal. The two inputs to the frequency converter are the input signal along with any modulation and the input from a local oscillator (LO). The output contains several frequencies including the difference between the input frequencies. The difference frequency is called intermediate frequency and output circuit of the mixer is tuned for the intermediate frequency.

Separately Excited Mixer

Fig. 5.7 shows the separately excited mixer using FET. Here, one device acts as a mixer while the other supplies the necessary oscillations. The bipolar transistor T_2, forms the

Hartley oscillator circuit. It oscillates with local frequency (f_o). FET T_1, is a mixer, whose gate is fed with the output of local oscillator and its bias is adjusted such that it operates in a nonlinear portion of its characteristic. The local oscillator varies the gate bias of the FET to vary its transconductance in a nonlinear manner, resulting intermediate frequency (IF) at the output. The output is taken through double tuned transformer in the drain of the mixer and fed to the IF amplifier. The ganged tuning capacitor allows simultaneous tuning of mixer and local oscillator.

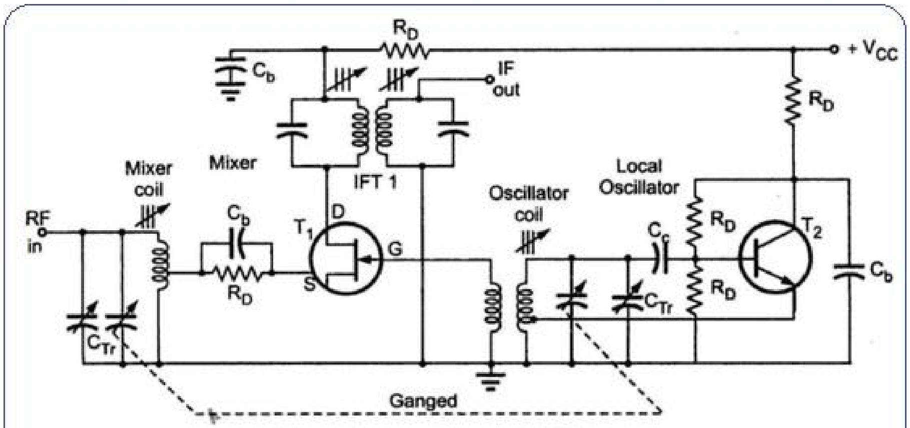


Fig. 5.7 Separately excited FET mixer

Self Excited Mixer

It is possible to combine the function of the mixer and local oscillator in one circuit. The circuit is commonly known as self excited mixer. Fig. 5.8 shows self excited bipolar transistor mixer. The circuit oscillates and the transconductance of the transistor is varied in a nonlinear manner at the local oscillator rate. This variable transconductance (g_m) is used by the transistor to amplify the incoming RF signal.

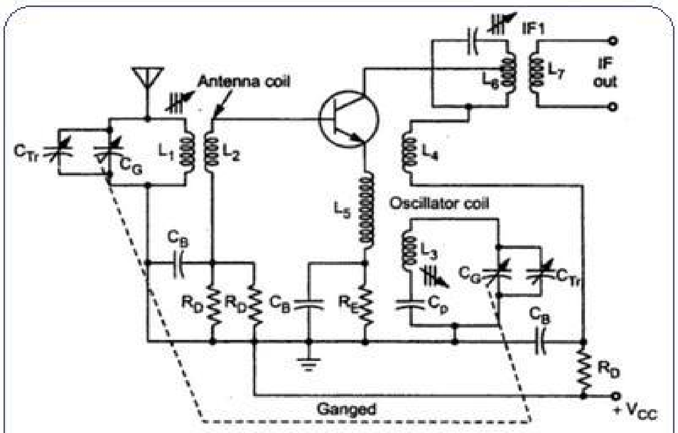


Fig. 5.8 Self excited mixer

Tracking

The superheterodyne receiver has number of tunable circuits which must all be tuned correctly if any given station is to be received. The ganged tuning is employed to do this work, which mechanically couples all tuning circuits so that only one tuning control or dial is required. Usually, there are three tuned circuits : Antenna or RF tuned circuit, mixer tuned circuit and local oscillator tuned circuit. All these circuits must be tuned to get proper RF input and to get IF frequency at the output of mixer. The process of tuning circuits to get the desired output is called **Tracking.** Any error that exists in the frequency difference will result in an incorrect frequency being fed to the IF amplifier. Such errors are known as **'Tracking Errors'** and these must be avoided.

To avoid tracking errors standard capacitors are not used, and ganged capacitors with identical sections are used. A different value of inductance and special extra capacitors called trimmers and padders are used to adjust the capacitance of the oscillator to the proper range. There are three common methods used for tracking. These are

- Padder tracking
- Trimmer tracking
- Three-point tracking

Padder tracking

Fig. 5.9 (a) shows connection of tuned circuit for padder tracking. In padder tracking the oscillator tunes below the frequency it should be in midband, so the IF created is higher than it should be, and positive error is created as shown in the Fig. 5.9 (b).

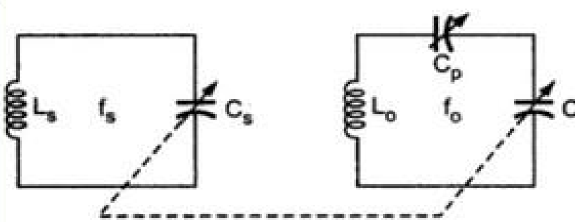


Fig. 5.9 (a) Padder tracking

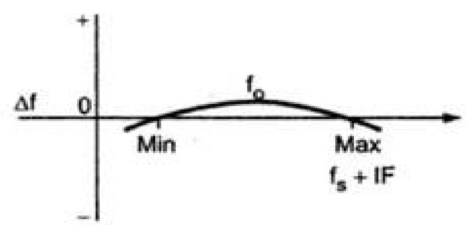


Fig. 5.9 (b) Tracking error in padder tracking

Trimmer tracking

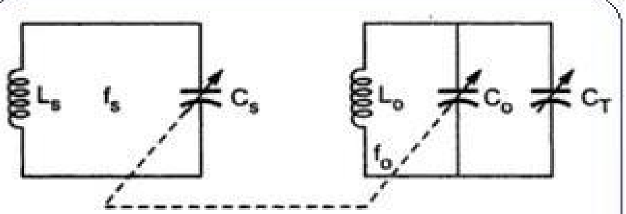


Fig. 5.10 (a) Trimmer tracking

Fig. 5.10 (a) shows connection of tuned circuit for trimmer tracking. In trimmer tracking, the oscillator tunes higher the frequency it should be in midband, so the IF created is less than it should be, and a negative error is created as shown in the Fig. 5.10 (b).

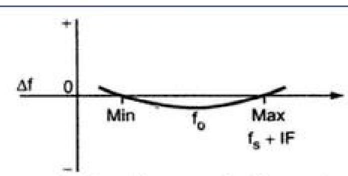


Fig. 5.10 (b) Tracking error in trimmer tracking

Three point tracking

The combination circuit called three point tracking can be adjusted to give zero error at three points across the band, at each end, and at the middle. Fig. 5.11 (a) and (b) show the connection of tuned circuit for three point tracking and its error, respectively.

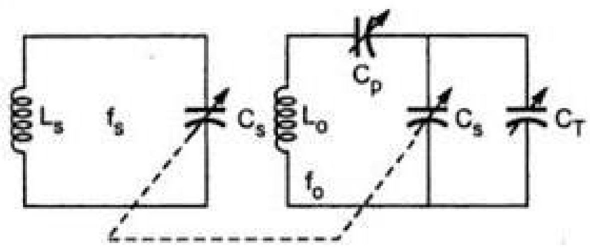


Fig. 5.11 (a) Three point tracking

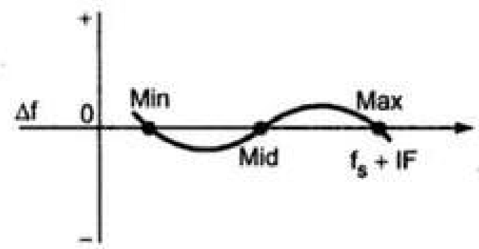


Fig. 5.11 (b) Tracking error in three point tracking

Local Oscillator

In shortwave broadcasting, the operating limit for receivers is 36 MHz. For such operating limit local oscillators such as Armstrong, Hartley, Colpitts, Clapp or ultra-audion are used. The Colpitts, clapp and ultra-audion oscillators are used at the top of the operating limit, whereas Hartley oscillator is used for frequencies below 120 MHz. All these oscillators are LC oscillators and each employs only one tuned circuit to defermine its frequency. When higher frequency stability of local oscillator is required, the circuits like AFC (Automatic Frequency Control) are used.

Relation between oscillator frequency and signal frequency :

We know that, for medium wave band, the signal frequency ranges from 540 kHz to 1650 kHz, and IF is usually 455 kHz. Therefore, the frequency range of local oscillator is 995 kHz (540 + 455) to 2105 kHz (1650 + 455), giving a ratio of maximum to minimum frequencies of 2.11 : 1 (2105 : 995). If the local oscillator had been designed to be below signal frequency, the range would have been 85 kHz (540 – 455) to 1195 kHz (1650 – 455), and the ratio would have been 14 : 1 (1195 : 85). The normal tunable capacitor has a capacitance ratio of approximately 10 : 1, giving a frequency ratio of 3.16 : 1. Hence the 2.11:1 ratio required for the local oscillator operating above signal frequency is well within range, whereas the frequency ratio of 14 :1 just cannot be obtained using practically available tunable capacitors. That is why the local oscillator frequency is always made higher than the signal frequency in receivers with variable frequency oscillators.

Ex. 5.1 : *Find the tuning range necessary for the oscillator capacitor in a MW superheterodyne receiver which tunes over the range of signals from 530 kHz to 1650 kHz and uses an IF of 455 kHz if the oscillator frequency is :*

1. Higher than the signal frequency

2. Lower than the signal frequency.

Sol. : Given $f_{s\,min} = 530$ kHz, $f_{s\,max} = 1650$ kHz and IF = 455 kHz

1. For $f_o > f_s$

$$f_{o\,min} = f_{s\,min} + IF = 530 + 455 = 985 \text{ kHz}$$

$$f_{o\,max} = f_{s\,max} + IF = 1650 + 455 = 2105 \text{ kHz}$$

We know that,

$$\frac{C_{o\,max}}{C_{o\,min}} = \left[\frac{f_{o\,max}}{f_{o\,min}}\right]^2 = \left[\frac{2105}{985}\right]^2 = 4.567$$

Therefore, tuning range for oscillator capacitor is 4.567 : 1.

2. For $f_o < f_s$

$$f_{o\,min} = f_{s\,min} - IF = 530 - 455 = 75$$

$$f_{o\,max} = f_{s\,max} - IF = 1650 - 455 = 1195$$

$$\left[\frac{C_{o\,max}}{C_{o\,min}}\right] = \left[\frac{1195}{75}\right]^2 = 253.87$$

Therefore, tuning range of oscillator capacitor is 253.87 : 1. **Which is impractical**

I F Amplifiers:

Intermediate-frequency Amplifiers The IF amplifier is a fixed-frequency amplifier, with the very important function of rejecting adjacent unwanted frequencies. It should have a frequency response with steep skirts. When the desire for a flat-topped response is added, the resulting recipe is for a double-tuned or stagger-tuned amplifier. Whereas FET and integrated circuit IF amplifiers generally are double-tuned at the input and at the output, bipolar transistor amplifiers often are single-tuned. A typical bipolar IF amplifier for a domestic receiver is shown in Fig. 5.12. It is seen to be a two-stage amplifier, with all IF transformers single tuned. This departure from a single-stage, double-tuned amplifier is for the sake of extra gain, and receiver sensitivity.

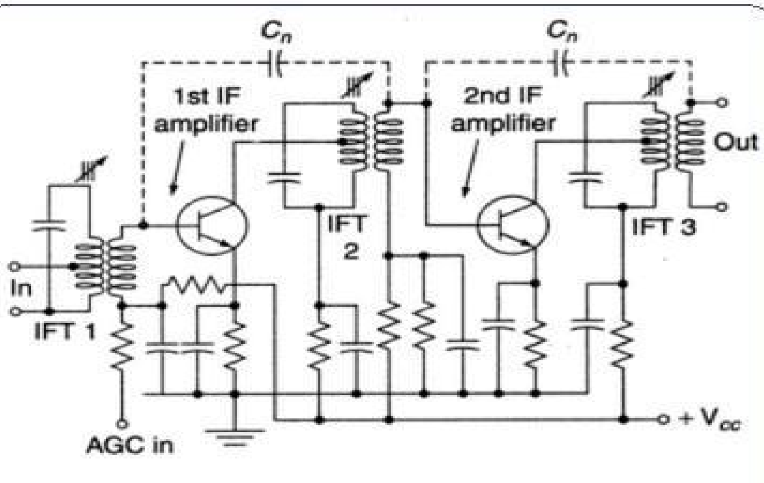


Fig. 5.12 *Two-stage IF amplifier*

Choice of Intermediate Frequency

Selection of the intermediate frequency depends on various factors. While choosing the intermediate frequency it is necessary to consider following factors.

1. The IF must not fall in the tuning range of the receiver, otherwise instability will occur and heterodyne whistles will be heard, making it impossible to tune to the frequency band immediately adjacent to the intermediate frequency.
2. Very high intermediate frequency will result in poor selectivity and poor adjacent channel rejection.
3. A high value of intermediate frequency increases tracking difficulties.
4. At low values of intermediate frequency, image frequency rejection is poor.
5. At very low values of intermediate frequency, selectivity is too sharp. Cutting off the sidebands.
6. At very low IF, the frequency stability of the local oscillator must be correspondingly high because any frequency drift is now a larger proportion of the low IF than of a high IF.

With the above considerations the standard broadcast AM receivers [tuning to 540 to 1650 kHz] use an IF within the 438 kHz to 465 kHz range. The 465 kHz IF is most commonly used.

Intermediate Frequencies Used

As a result of many years' experience, the previous requirements have been translated into specific frequencies, whose use is fairly well standardized throughout the world (but by no means compulsory). These are as follows:

1. Standard broadcast AM receivers [tuning to 540 to 1650 kHz, perhaps 6 to 18 MHz, and possibly even the European long-wave band (150 to 350 kHz)] use an IF within the 438- to 465-kHz range, with 455 kHz by far the most popular frequency.
2. AM, SSB and other receivers employed for shortwave or VHF reception have a first IF often in the range from about 1.6 to 2.3 MHz, or else above 30 MHz. (Such receivers have two or more different intermediate frequencies.)
3. FM receivers using the standard 88- to 108-MHz band have an IF which is almost always 10.7 MHz.
4. Television receivers in the VHF band (54 to 223 MHz) and in the UHF band (470 to 940 MHz) use an IF between 26 and 46 MHz, with approximately 36 and 46 MHz the two most popular values.
5. Microwave and radar receivers, operating on frequencies in the 1- to 10-GHz range, use intermediate frequencies depending on the application, with 30, 60 and 70 MHz among the most popular.

By and large, services covering a wide frequency range have IFs somewhat below the lowest receiving frequency, whereas other services, especially fixed-frequency microwave ones, may use intermediate frequencies as much as 40 times lower than the receiving frequency.

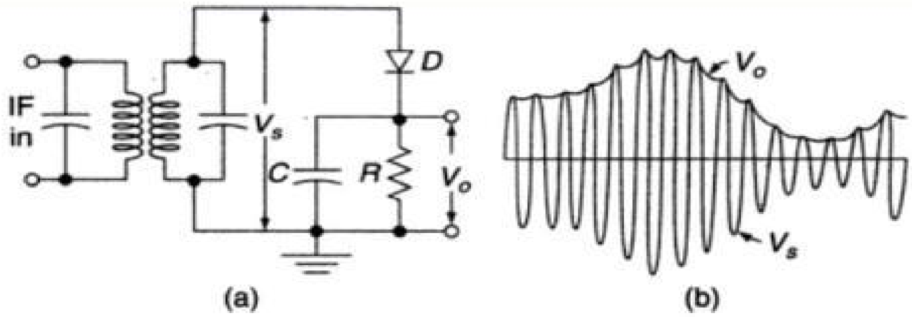


Fig. 5.13 *Simple diode detector. (a) Circuit diagram; (b) input and output voltages*

Detection and Automatic Gain Control (AGC)

Operation of Diode Detector The diode is by far the most common device used for AM demodulation (or detection), and its operation will now be considered in detail. On the circuit of Fig. 5.13*a*, C is a small capacitance and *R* is a large resistance. The parallel combination of *R* and *C* is the load resistance across which the rectified output voltage V_o is developed. At each positive peak of the RF cycle, *C* charges up to a potential almost equal to the peak signal voltage V_s. The difference is due to the diode drop since the forward resistance of the diode is small (but not zero). Between peaks a little of the charge in *C* decays through *R*, to be replenished at the next positive peak. The result is the voltage V_o, which reproduces the modulating voltage accurately, except for the small amount of RF ripple. Note that the time constant of *RC* combination must be slow enough to keep the RF ripple as small as possible, but sufficiently fast for the detector circuit to follow the fastest modulation variations.

This simple diode detector has the disadvantages that V_o, in addition to being proportional to the modulating voltage, also has a dc component, which represents the average envelope amplitude (i.e., carrier strength), and a small RF ripple. The unwanted components are removed in a practical detector, leaving only the intelligence and some second harmonic of the modulating signal.

Practical Diode Detector A number of additions have been made to the simple detector, and its practical version is shown in Fig. 5.14. The circuit operates in the following manner. The diode has been reversed, so that now the negative envelope is demodulated. This has no effect on detection, but it does ensure that a negative AGC voltage will be available, as will be shown. The resistor *R* of the basic circuit has been split into two

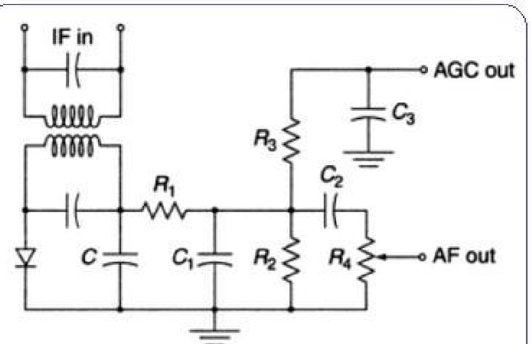


Fig. 5.14 *Practical diode detector*

parts (R_1 and R_2) to ensure that there is a series dc path to ground for the diode, but at the same time a low-pass filter has been added, in the form of $R_1 - C_1$. This has the function of removing any RF ripple that might still be present. Capacitor C_2 is a coupling capacitor, whose main function is to prevent the diode dc output from reaching the volume control R_4. Although it is not necessary to have the volume control immediately after the detector, that is a convenient place for it. The combination $R_3 - C_3$ is a low-pass filter designed to remove AF components, providing a dc voltage whose amplitude is proportional to the carrier strength, and which may be used for automatic gain control.

It can be seen from Fig. 5.14 that the dc diode load is equal to $R_1 + R_2$, whereas the audio load impedance Z_m is equal to R_1 in series with the parallel combination of R_2, R_3 and R_4, assuming that the capacitors have reactances which may be ignored. This will be true at medium frequencies, but at high and low audio frequencies Z_m may have a reactive component, causing a phase shift and distortion as well as an uneven frequency response.

Principles of Simple Automatic Gain Control Simple AGC is a system by means of which the overall gain of a radio receiver is varied automatically with the changing strength of the received signal, to keep the output substantially constant. A dc bias voltage, derived from the detector as shown and explained in connection with Fig. 5.14 is applied to a selected number of the RF, IF and mixer stages. The devices used in those stages are ones whose transconductance and hence gain depends on the applied bias voltage or current. It may be noted in passing that, for correct AGC operation, this relationship between applied bias and transconductance need not be strictly linear, as long as transconductance drops significantly with increased bias. The overall result on the receiver output is seen in Fig. 5.15

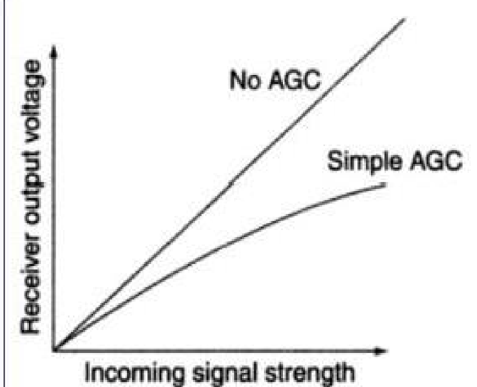


Fig. 5.15 *Simple AGC characteristics*

All modern receivers are furnished with AGC, which enables tuning to stations of varying signal strengths without appreciable change in the volume of the output signal. Thus AGC "irons out" input signal amplitude variations, and the gain control does not have to be readjusted every time the receiver is tuned from one station to another, except when the change in signal strengths is enormous. In addition, AGC helps to smooth out the rapid fading which may occur with long-distance shortwave reception and prevents overloading of the last IF amplifier which might otherwise have occurred.

FM. Receiver

The block diagram of typical F.M. receiver is shown in the Fig. 5.16
Let us briefly study different stages in F.M. receiver.

R.F.Amplifier Stage

Since F.M. signal has a larger bandwidth it is likely to encounter more noise. Hence to reduce the noise figure of the receiver, an RF amplifier stage is used. The RF amplifier stage matches the antenna to the receiver. For this purpose and to avoid neutralization, grounded-base or grounded-gate circuits are employed for this stage. Both these circuits have low input impedance, suitable for matching with antenna impedance,

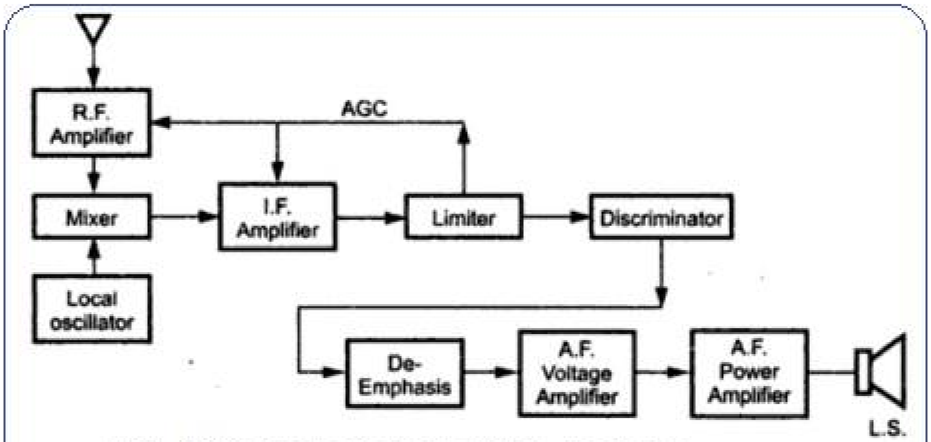


Fig. 5.16 Block diagram of F.M. receiver

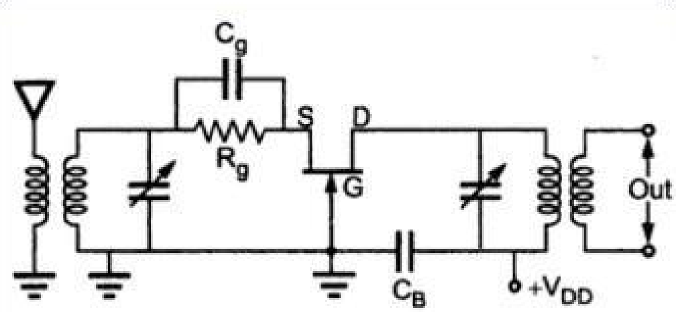


Fig.5.17 Grounded-gate FET RF amplifier

A typical circuit is shown in the Fig.5.17 Since the gate terminal is grounded, the input and output sides are isolated for RF purposes. There is no possibility of feedback and thereby instability in operation. Therefore the circuit does not require neutralization. The low input impedance of the FET amplifier can be easily matched to antenna through a single secondary tuned RF transformer. Both the input and output tank circuits are tuned to carrier frequency.

Mixer Stage

With the help of local oscillator, this stage down converts the incoming carrier frequency to I.F, which is 10.7 MHz for F.M. receiver. The local oscillator is usually the Clapp oscillator, suitable for VHF operation. Maintaining a constant difference between the carrier frequency and local oscillator frequency is not a problem in FM receiver unlike in an AM receiver. Compared to A.M. receiver, tuning range of incoming carrier frequencies for FM receiver is small, from 88 MHz to 108 MHz, i.e. about 1.25:1. Thus the tracking is comparatively easy in FM receiver.

Since FETs are less noisy than BJTs, RF amplifier stage and mixer stage uses FETs, with local oscillator constructed with BJT.

The mixer stage uses a tuned circuit as its load. The circuit is tuned to Intermediate Frequency of 10.7 MHz and hence selects the difference between incoming carrier frequency and locally generated oscillator frequency.

I.F. Amplifier Stage

In the I.F. amplifier stages, the most of the gain of receiver is developed. the intermediate frequency and bandwidth requirements are normally much larger than in AM receiver. The typical values, for an F.M. receiver operating in FM band from 88 MHz to 108 MHz are 10.7 MHz for I.F. and 200 kHz for bandwidth. Generally two I.F. amplifier stages are employed. However while designing these stages, shrinkage of bandwidth due to cascading must be taken in account.

The I.F. amplifier stage uses a tuned circuit as its load. The circuit is tuned to intermediate frequency.

Limiter Stage

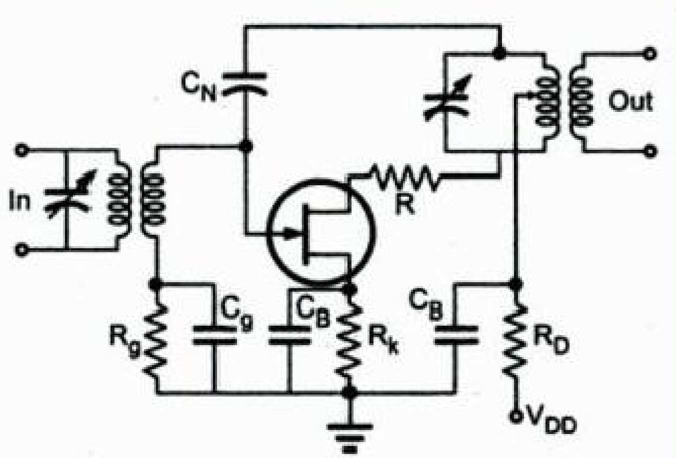


Fig. 5.18 Amplitude Limiter

Frequency modulation is developed to provide a communication system which is less noise-sensitive than AM system. The most of the noise accompanying the desired signal accompanies it as A.M. Thus, if the intelligence is contained in the frequency variation of signal; we can, at the receiver, remove all amplitude variations without loss of the information content of the desired signal. To remove the amplitude variations of the signal is precisely the function of the limiter. At the output of the limiter state, we get a constant amplitude signal, even though the amplitude of input signal may be varying. The limiter is thus basically a clipper circuit which clipps off the undesired amplitude variations of the input signal, as shown in the Fig. 5.18 . The input signal provides the bias for the FET circuit. Negative bias increases as input increases and hence it lowers the gain of the amplifier for high amplitude of the input signal and output voltage tends to remain constant.

Audio Amplifier

This stage consists of an *RC* coupled voltage amplifier followed by a push-pull power amplifier. The fidelity of the receiver is determined by the frequency response characteristic of the stage. The more the bandwidth of the stage, the better the fidelity.

The voltage amplifier is a class 'A' amplifier provides the necessary drive for the succeeding power amplifier, The gain control of the voltage amplifier serves as the volume control of the radio receiver. The output of the push-pull amplifier is transformer coupled to match the impedance of the output stage with that of the loudspeaker.

FM Stereo Broadcast receiver

The block diagram of stereo FM demodulation with optional SCA output is shown in the Fig. 5.19

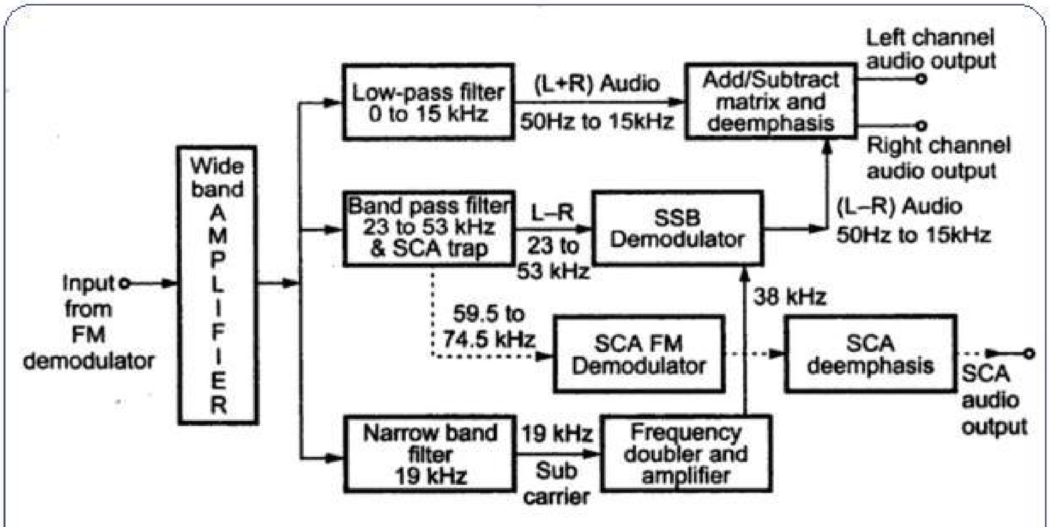


Fig 5.19 FM Stereo broadcast receiver

Note: Subsidiary Communication Authorization [SCA] signal may also be transmitted along with the primary stereo signal. It is applied as input to adder. This SCA signal is not always transmitted, it is optional.

The low pass filter attenuates all frequencies above 15 kHz and thus the sum signal [L + R] is available at its output. In manaural receiver, this output will be further passed through the deemphasis network to the audio amplifier, for reproduction of stereo broadcast in monaural way.

The band pass filter has a passband extending from 23 kHz to 53 kHz, which corresponds to the difference signal [L – R], rejecting the (optional) SCA signal above 59.5kHz.

The difference signal [L – R] is fed to the SSB demodulator which also receives 38 kHz carrier necessary for demodulation. After demodulation, [L – R] audio signal in the frequency range 50 Hz to 15 kHz is available which is applied as the input to Add/Subtract Matrix and deemphasis block, also receiving [L + R] signal as its other input. The matrix produces the left channel from an adder and the right channel from a subtractor. After deemphasis, they are further amplified to yield ultimately stereo reproduction.

The 19 kHz subcarrier, transmitted at much lower level and serving as a pilot carrier, is separated from the composite signal, using a narrow band filter tuned to 19 kHz. The frequency doubler converts it to the wanted 38 kHz carrier for SSB demodulator.

If SCA signal is present, it is separated using band pass filter, then demodulated by FM detector, passed through deemphasis and produced finally as a separated audio output.

Automatic Frequency Control (AFC)

Where the main oscillator is only an L-C oscillator then it is not capable of keeping the carrier frequency stable. The carrier frequency can be kept stable by using an automatic frequency control [AFC] circuit. The block diagram of a typical AFC circuit is shown in the Fig. 5.20 .

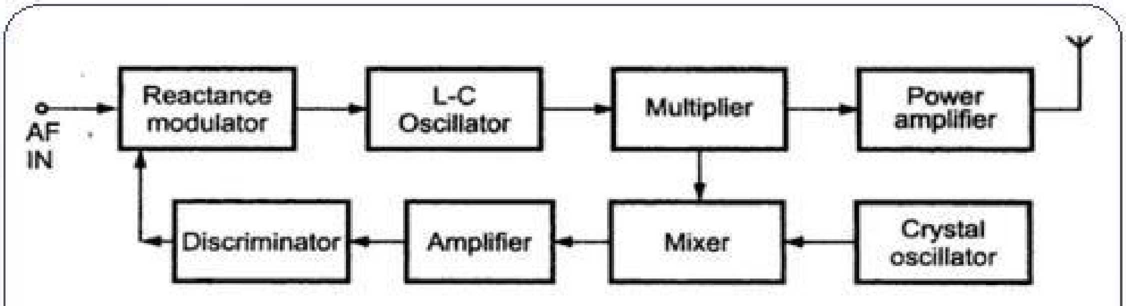


Fig. 5.20 Typical AFC circuit

The discriminator reacts only to small changes in the carrier frequency but not to the frequency deviations in the carrier (since it is too fast).

Suppose frequency of the carrier increases. This higher frequency is fed to the mixer for which the other input frequency is from the stable crystal oscillator. A some what higher frequency will be fed to the discriminator. Since the discriminator is tuned to the correct frequency difference which should exist between the LC oscillator and crystal oscillator , and its input frequency is now somewhat higher, the discriminator will develop a positive dc voltage. This voltage is applied to the reactance modulator whose transconductance is increased by the positive voltage developed by the discriminator. This increases the equivalent capacitance of the reactance modulator thereby decreasing the oscillator frequency. The frequency increase in the carrier frequency is thus lowered and brought to the correct value.

The correcting dc voltage developed by the discriminator may be fed to a varactor diode connected across the tank circuit of the oscillator and be used for AFC purposes.

AUTOMATIC FREQUENCY CONTROL (in receivers)

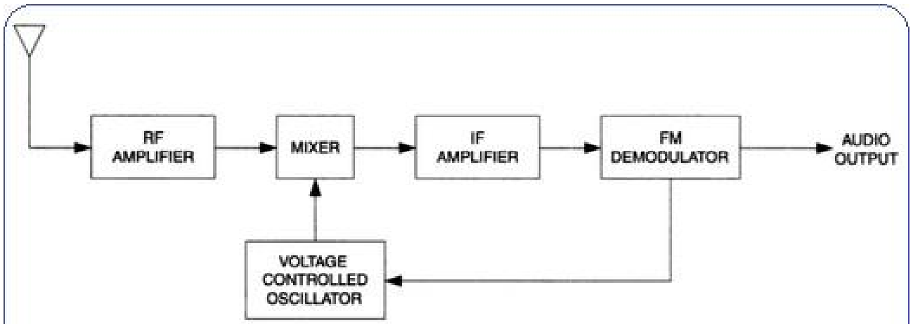


Fig. 5.21 Automatic frequency control circuit.

Automatic frequency control is used in some FM receivers to keep the tuning frequency stable. This type of circuit is confined largely to older, analog FM receivers, because modern digital frequency synthesizers prevent the kind of drift and outright frequency shifting common in analog receivers.

A typical AFC system is shown in Figure 5.21 The AFC output of the FM demodulator is 0 V when the input signal frequency is directly on the resonant frequency of the demodulator, but positive or negative voltage depending whether the frequency is above or below resonance. This voltage is fed back to a voltage controlled oscillator circuit, which feeds the mixer. In practice, a varactor (variable capacitance diode) is used to shunt the L-C or quartz crystal in the oscillator, and this varactor converts the oscillator into a VCO. which then takes care of the frequency correction or restoration

Noise in F.M.

In F.M. system, carrier frequency for a transmitter is fixed at the transmitting end, which may be crystal controlled. Noise voltages at the input to receiver can produce only amplitude and phase modulation of the carrier; noise cannot directly frequency modulate the carrier. In the FM receiver, by using limiter circuit, amplitude modulation can be removed. However phase modulation of the carrier produces indirectly frequency modulation, which is detected as noise at the output of the receiver. As shown in the Fig.5.22, the noise voltage can be represented by a phasor, E_n, having a randomly varying amplitude, and a randomly varying phase angle θ_n with respect to the carrier phasor E_c.

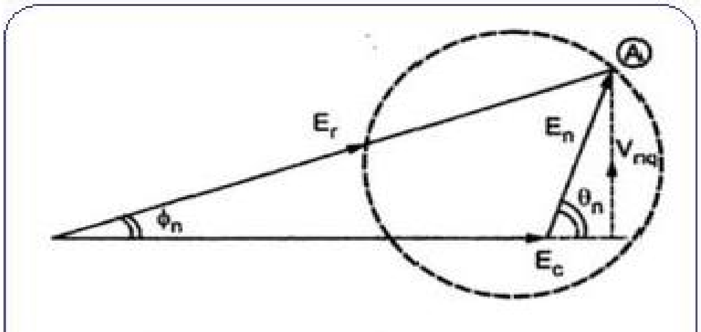


Fig. 5.22 Phasor diagram for carrier voltage very much greater than noise

The resultant of the phasor E_c and E_n is the phasor E_r. The tip of this phasor, point A traces at a random path, shown dotted in figure. Due to this, the phase angle between the phasors E_r and E_c, ϕ_n, varies at random. Normally the carrier amplitude E_c is much larger than the amplitude of the noise, E_n. Then the phase modulation of the carrier is given approximately by,

$$\phi_n = \tan^{-1}\left[\frac{V_{nq}}{E_c}\right], \frac{V_{nq}}{E_c}$$

where V_{nq} is the component of noise phasor, which is at right angles to the phasor E_c.

For a phase modulated wave, the phase modulation is expressed by

$$\theta(t) = \phi_c + \phi(t) \qquad \text{... (I)}$$

where ϕ_c is the phase angle of the carrier, and $\theta(t)$ is the change in phase angle with time.

Differentiating both sides of the above equation (I) :

$$\frac{d\theta(t)}{dt} = \frac{d\phi_c}{dt} + \frac{d}{dt}[\phi(t)] \quad \text{... (II)}$$

But we know that the instantaneous angular frequency, ω_i :, is related to the phase angle by

$$\omega_i = \frac{d\theta(t)}{dt}.$$

Hence, substituting in (II).

$$\omega_i = \omega_c + \omega_{eq}(t) \quad \text{... (III)}$$

where $\omega_{eq}(t)$ is the equivalent frequency (angular) deviation produced due to phase modulation.

Then

$$\omega_{eq}(t) = \frac{d}{dt}[\phi(t)]$$

$$\therefore \quad 2\pi f_{eq}(t) = \frac{d}{dt}[\phi(t)]$$

$$\therefore \quad f_{eq}(t) = \frac{1}{2\pi}\frac{d}{dt}[\phi(t)]$$

Here $f_{eq}(t)$ is equivalent frequency modulation due to the noise. The FM receiver will detect this frequency modulation, giving rise to the output noise.

Thus, the demodulated noise output voltage is proportional to the rate of change of the phase modulation or equivalent frequency deviation (modulation). Thus the noise voltage becomes proportional to frequency shift. Then the spectrum density, expressed in V/Hz, for the noise voltage is proportional to frequency.

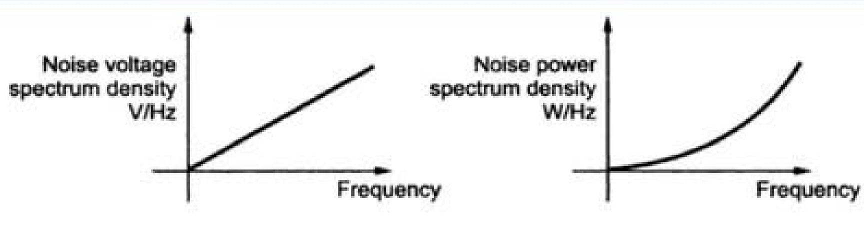


Fig. 5.23 (a) Voltage spectrum density curve (b) Power spectrum density curve

As a result of noise spectrum density increasing with frequency, noise is more at higher audio frequencies, or signal-to-noise ratio reduces at higher frequency. The remedy for this is the use of pre-emphasis and de-emphasis circuits.

Since the power is proportional to $(\text{voltage})^2$, noise power density spectrum will vary with $(\text{frequency})^2$; and the graph of noise power spectrum density V_s frequency will show a square-law or it will be parabolic in nature.

As a result of noise voltage spectrum density increasing with frequency, noise voltage increases as sideband frequency increases. At the carrier frequency, noise will be minimum. Assuming noise frequencies to be evenly distributed across the bandpass of the receiver, the noise output from the receiver changes uniformly with noise sideband frequency for F.M. In Amplitude modulation, since there is no frequency shift, the noise voltage will be constant. This is shown in the Fig. 5.24

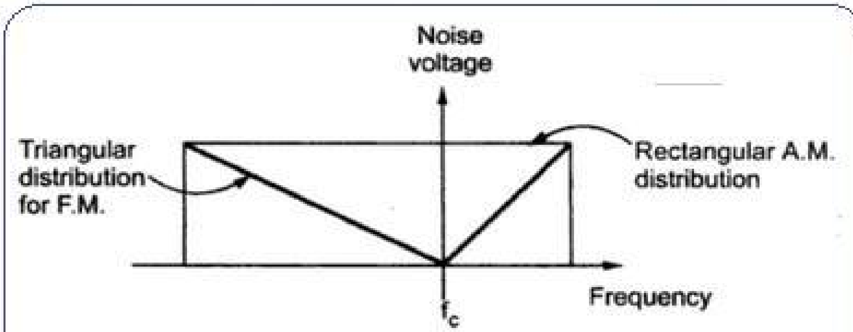


Fig. 5.24 Noise distribution

The triangular noise distribution for FM is called the "noise triangle". The corresponding distribution for AM is rectangle, since frequency is not varied in A.M. From above discussion it can be concluded that as modulation index decreases, effect of noises increases for constant frequency deviation. When modulation frequency is small, modulation index becomes large producing large number of sidebands. Hence the effect of noise gets reduced.

Capture Effect

A major advantage of FM is that interfering signals on the same frequency will be effectively rejected. As the limiter stage is used in FM receivers, a peculiar effect is observed when two or more FM signals occur simultaneously on the same frequency. If the signal strength of one is more than twice the amplitude of the other, the stronger signal will 'capture" the channel and will totally eliminate the weaker, interfering signal. This is shown as the capture effect in FM. However, when two AM signals occupy the same frequency, both signals will generally be heard regardless of their relative signal strengths. When the signal strengths of the AM signals are nearly the same, they will interfere with one another making both of them unintelligible. But in FM, the capture effect allows the stronger signal to dominate while totally suppressing the weaker signal.

However, when the strengths of the two FM signals are nearly the same the capture effect may cause the signals to dominate alternately. This is more experienced in mobile FM communication. As some time one signal may be stronger than the other, thus capturing the channel. At some other time, the other signal may have more strength, and thus capturing the channel. In any case, once the strong signal dominates; the weaker is not heard at all on the channel.

FM Threshold Effect

Definition

As the carrier to noise ratio is reduced, clicks are heard in the receiver output. As the carrier to noise ratio reduces further, crackling, or sputtering sound appears at the receiver output. Near the breaking point the theoretically calculated output signal to noise ratio becomes large, but its actual value is very small. This phenomenon is called threshold effect.

Definition of threshold

It is the minimum carrier to noise ratio yielding an FM improvement which is not significantly deteriorated from the value predicted by the usual signal to noise formula assuming small noise.

Consider that the carrier is unmodulated. The signal at the output of FM discriminator is represented as,

$$x(t) = s(t) + n(t)$$

Here $s(t) = A_c \cos(2\pi f_c t)$ with no modulation, since $\phi(t) = 0$

Putting for $s(t)$ and $n(t)$ in above equation,

$$\begin{aligned} x(t) &= A_c \cos(2\pi f_c t) + n_c(t)\cos(2\pi f_c t) - n_s(t)\sin(2\pi f_c t) \\ &= [A_c + n_c(t)]\ \cos(2\pi f_c t) - n_s(t)\ \sin(2\pi f_c t) \end{aligned} \quad \cdots\textbf{(A)}$$

Fig. 5.25 shows the phasor diagram of above equation. The amplitude and phases of $n_c(t)$ and $n_s(t)$ change randomly with time. The point A wanders randomly around point B. The angle $\theta(t)$ varies approximately from $\frac{n_s(t)}{A_c}$ to multiples of 2π. Whenever $\theta(t)$ changes by $\pm 2\pi$, clicks are produced in the discriminator output. As the carrier to noise ratio is decreased further, the clicks per unit time increase. The threshold is said to occurred when these clicks are very large.

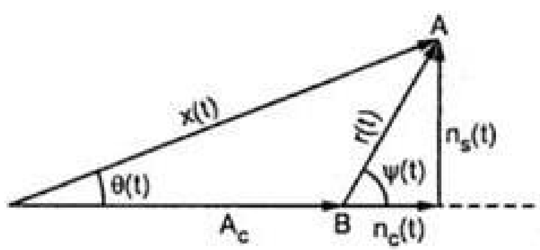


Fig. 5.25 Phasor representation of equation (A)

COMMUNICATIONS RECEIVERS

A communications receiver is one whose main function is the reception of signals used for communications rather than for entertainment. It is a radio receiver designed to perform the tasks of low- and high-frequency reception much better than the type of set found in the average household. In turn, this makes the communications receiver useful in other applications, such as the detection of signals from high-frequency impedance bridges (where it is used virtually as a high-sensitivity tuned voltmeter), signal-strength measurement, frequency measurement and even the detection and display of individual components of a high-frequency wave (such as an FM wave with its many sidebands). It is often operated by qualified people, so that any added complications in its tuning and operation are not necessarily detrimental, as they would be in a receiver to be used by the general public.

Fig.5.26 shows the block diagram of communication receiver.

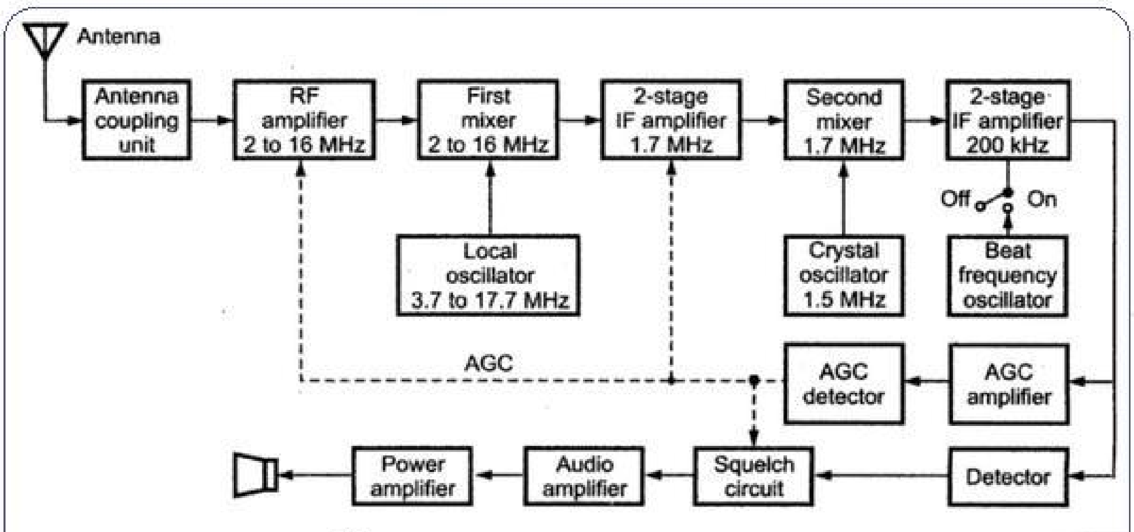


Fig. 5.26 Double communication receiver

Double Conversion

The front end selectivity of any receiver must reject the first image frequency at $2f_i$ above the desired signal frequency, and for this to occur, the intermediate frequency f_i should be high enough. However, as the f_i increases, its bandpass gets larger. Beyond the HF band (above 30 MHz), it becomes impossible to obtain the required bandpass with good image rejection using ordinary circuits. To avoid this problem two intermediate frequencies are used in the communication receiver. The first f_i is high, several megahertz or even higher. The second f_i is quite low, of the order of 1 MHz or even less. The procedure of employing two intermediate frequencies in one receiver is know as double conversion (Apart from ommunication receivers nowadays higher end FM receivers are also empoying these technique which are popularly known as Double conversion receiver)

The communication receiver allows the receiver to have good image rejection and also good adjacent channel selectivity. The image rejection is assured in the first conversion as image frequency is well outside the fixed-tuned RF amplifier bandpass. The narrow bandpass required for good adjacent channel rejection is obtained in the second intermediate frequency, which may be as low as 100 kHz.

Let us consider the receiver is tuned to receive signal of 150 MHz, first intermediate frequency is 10.7 MHz with the bandwidth of 150 kHz and second intermediate frequency is 465 kHz with bandwidth of 15 kHz. With these values, the image frequency of the first intermediate frequency can be given as

$$\begin{aligned} f_{si} &= f_s + 2 f_i \\ &= 150 + 2 \times 10.7 \\ &= 171.4 \text{ MHz} \end{aligned}$$

This image frequency is beyond the bandwidth limit of 150 kHz and hence it is rejected. But at the same time bandwidth of 150 kHz allows several 15 kHz wide adjacent channels to pass to the second mixer. The bandpass for the second intermediate frequency rejects these adjacent channels since its passband is only of 15 kHz.

As shown in the Fig. 5.26, communication receiver has number of modifications and added circuity. Let us see the added circuity.

Tuning calibration : Tuning calibration consists of a built-in crystal oscillator, usually nonsinusoidal, operating at 500 at 1000 kHz, whose output may be fed to the input of the receiver. With this, the calibration of the receiver may be corrected by adjustment of the pointer or cursor independent of the gang.

Beat frequency oscillator : The normal receivers are not capable of receiving transmission of Morse code, i.e., pulse-modulated RF carrier. In order to make Morse code audible, the communication receiver has a built-in Beat Frequency Oscillator (BFO), as shown in the Fig. 5.26 The BFO is a simple LC oscillator; usually Hartley oscillator, operating at a frequency of 1 kHz or 400 Hz above or below the last intermediate frequency. In case of nontelegraph reception, BFO is switched off to prevent interference.

Noise limiter : The noise limiter circuit is used to reduce interfacing noise pulses created by ignition systems, electrical storms or electrical machinery of various types. It does this by automatically silencing the receiver for the duration of a noise pulse. The typical limiter circuit provides a negative voltage as a result of noise impulse or any very sharp voltage pulse, and this negative voltage is applied to the detector. The diode detector remains cut off for the duration of the noise pulse.

Squelch circuit : When carrier is not present at the input i.e. in the absence of transmission on a given channel or between stations, a sensitive receiver will produce a disagreeable amount of loud noise. This is because AGC disappears in the absence of carrier ; the receiver acquires its maximum sensitivity and amplifies the noise present at its input. The squelch circuit detects the presence of carrier and enables receiver's output only when carrier is present. Thus avoids loud noise. The squelch circuit is also called muting or quieting circuit.

Fig.5.27 shows the squelch circuit.

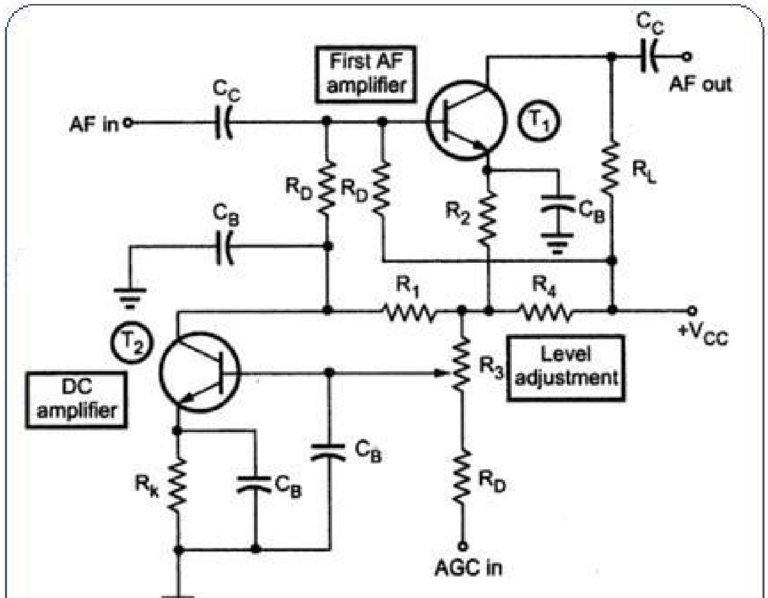


Fig. 5.27 Typical squelch circuit

As shown in the Fig.5.27, AGC is applied to the DC amplifier. When AGC voltage is low or zero, the dc amplifier T_2, draws current so that the voltage drop across its load resistance R_1 cuts off the audio amplifier T_1 ; thus no signal or noise is passed. On the other hand when AGC voltage is sufficiently negative, it cuts off T_2. Now T_2 does not draw any current and hence sufficient bias is provided to T_1 (First IF amplifier) to operate it is normal fashion.

Variable sensitivity and Variable selectivity

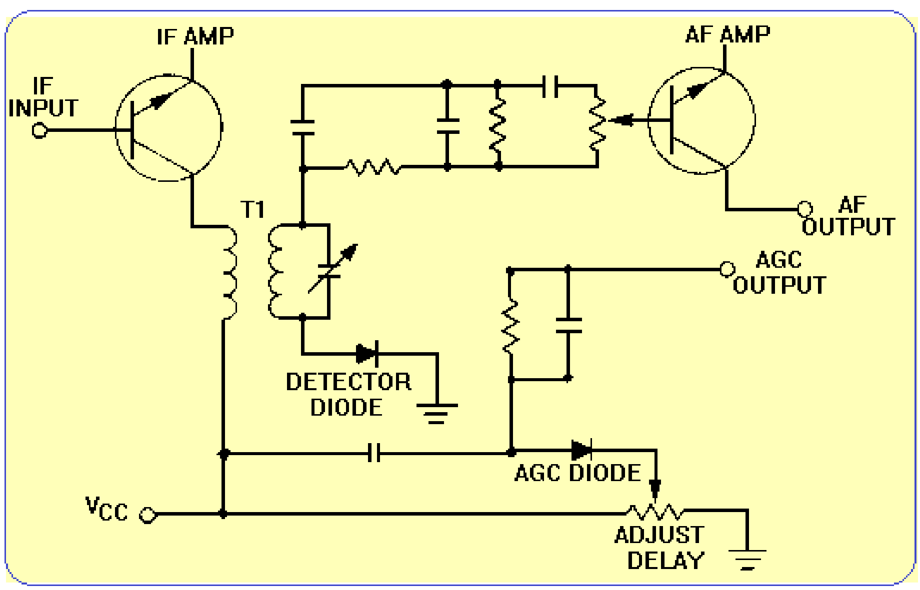
The ratio of the highest to the lowest signal strengths which a communications receiver may have to cope with could be as high as 10^5: 1. This means that the receiver must have sufficient sensitivity to amplify fully very weak signals, while also being capable of having its gain reduced by ACG action by a ratio of 10^5: 1, or 100 dB, so as not to overload on the strongest signal. Even the best ACG system is not capable of this performance. Apart from the alarming variations in output that could occur, there is also the risk of overloading several of the IF amplifiers, especially, the last one, and also the demodulator diode. To prevent the distortion which would follow ~as well as possible permanent damage, the most sensitive communications receivers incorporate a sensitivity control. This control generally consists of a potentiometer which varies the bias on the RF amplifier and is, in fact, an RF gain control. The AGe is still present, but it now acts to keep the sensitivity of the .receiver to the level determined by the setting of this control. The receiver is now considerably more versatile in handling varying input signal levels.

The selectivity, or to be more precise, the bandwidth, of the low-frequency I.F amplifier may be made variable over a range that is commonly 1 to 12 kHz. The largest *bandwidth* permits reception of high-quality broadcasts; whereas the smallest (although greatly impairs this quality) reduces noise and therefore increases intelligibility and will also reduce adjacent-channel interference. Variable selectivity is achieved in practice by switching in crystal, ceramic, or mechanical filters. A set is provided, any of which may be switched into the second IF stage to give bandwidths of 1, 2, 4, 6, 8, 10, and 12,kHz. Receivers designed for radiotelegraphy reception may have minimum bandwidths as low as 300Hz.

Delayed Automatic Gain Control

The disadvantage of automatic gain control, attenuating even the weak signal, is overcome by the use of delayed automatic gain control (DAGC). Let's take a look at the typical dagc circuitry in figure 5.28. This type of system develops no AGC feedback until an established received signal strength is attained. For signals weaker than this value, no agc is developed. For sufficiently strong signals, the delayed agc circuit operates essentially the same as ordinary agc.

Figure 5.28. - Delayed AGC action.



Our circuit uses two separate diodes; one is the detector diode and the other the agc diode. The AGC diode is connected to the primary of the last IF transformer and the detector diode to its secondary. A positive bias is applied to the cathode of the AGC diode. This keeps it from conducting until a prearranged signal level has been reached. The adjust delay control allows manual control of the AGC diode bias. Manual control allows you to select the signal level at which AGC is applied. If mostly weak stations are to be received, the setting should be high (no AGC until the signal level is high). However, you should set it as low as possible to prevent overloading of the last IF amplifier by stronger signals.

Finally, you must have two diodes to obtain delayed AGC. If only one diode were used, the AGC would be developed from the detector diode, and there would be no delayed action. Or, if a signal diode were biased to provide the delaying action desired, no signal would pass to the audio amplifier until the bias was exceeded by the input signal.

CCTV Concepts and Applications

Closed circuit television is a special application in which camera signals are made available only to a limited number of monitors or receivers. The particular type of link used depends on distance between the two locations, the number and dispersion of receivers and mobility of either camera or receiver. Figure 5.29 illustrates various link arrangements which are often used. The simplest link is a cable where video signal from the camera is connected directly through a cable to the receiver. A television monitor, which is a receiver, without RF and IF circuits, is only required for reception in such a link arrangement. About one volt peak-to-peak signal is required by the monitor. Since the video signal is normally delivered via cables and even when transmitted, it is over a limited region and for restricted use, CCTV need not follow television broadcast standards.

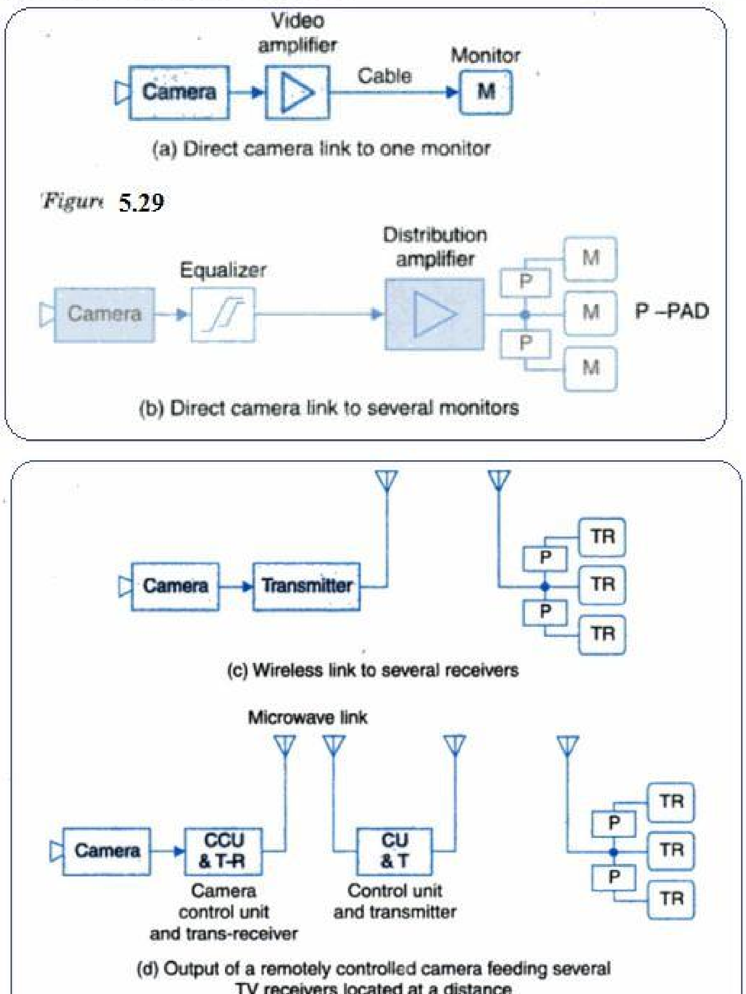


Fig. 5.29 Commonly used closed circuit television (CCTV) systems

CCTV Applications

There are numerous applications of CCTV and a few are briefly described here.

(*i*) *Education*. One instructor may lecture to a large number of students sitting at different locations. Similarly close-ups of demonstration experiments and other aids can be shwon on monitors during these lectures.

(*ii*) *Medicine*. Several monitors and camera units can be installed to observe seriously ill patients in intensive care units. In medical institutions, operations when performed can be shown to medical students without their actually gathering around the oepration table.

(*iii*) *Business*. Television cameras can be installed at different locations in big departmental stores to keep an eye over customers and sales personnel.

(*iv*) *Surveillance*. In banks, railway yards ports, traffic points and several other similar locations, closed circuit TV can be effectively used for surveillance.

(*v*) *Industry*. In industry CCTV has applications in remote inspection of materials. Observance of nuclear reactions and other such phenomena would have been impossible without television. Similarly television has played a great role in the scanning of earth's surface and probing of other planets.

(*vi*) *Home*. In homes a CCTV monitor finds its application in seeing the caller before opening the door.

(*vii*) *Aerospace and Oceanography*. Here a wireless link is used between the transmitter and receiver. In some applications camera is remotely controlled over a microwave radio link. As shown in Fig. 10.3 (*c*), for aerospace and oceanography a carrier is used for transmitting the signal and a complete receiver is then necessary for reception.

Cable Television (CATV- Community Antenna Television)

The CATV system is a cable system which distributes good quality television signal to a very large number of receivers throughout an entire community. In general, this system feeds increased TV programmes to subscribers who pay a fee for this service. A CATV system may have many more active (VHF and UHF) channels than a receiver tuner can directly select. This requires use of a special active converter in the head-end.

Formerly CATV system were employed only in far-fringe areas or in valleys surrounded by mountains where reception was difficult or impossible because of low level signal conditions. However, CATV systems are now being used in big cities where signal-level is high but all buildings render signals weak and cause ghosts due to multipath reflections. In either case, such a system often serves an entire town or city. A single antenna site, which may be on top of a hill, mountain or sky-scraper is chosen for fixing antennas. Several high gain and properly oriented antennas are employed to pick up signals from different stations. In areas where several signals are coming from one direction, a single broad based antenna (log-periodic) may be used to cover those channels. Most cable television installations provide additional services like household, business and educational besides commercial TV and FM broadcast programmes. These include news, local sports and community programmes, burgler and fire alarms, weather reports, commercial data retrieval, meter reading, document reproduction etc. Educational services include computer aided instructions, centralized library services and so on. Many of the above options require extra subscription fee from the subscriber.

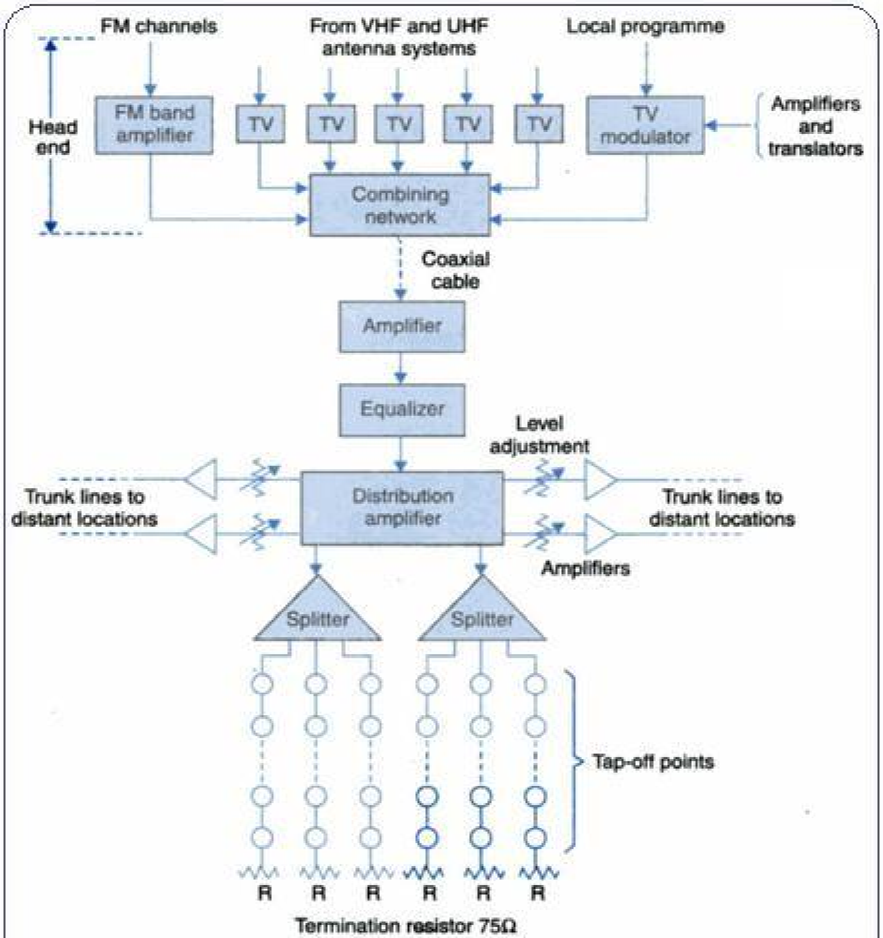


Fig. 5.30 Simplified block diagram of Cable (CA) TV system

When UHF reception is provided in addition to VHF, as often is the case, the signal from each UHF channel is processed by a translator. A translator is a frequency converter which hterodynes the UHF channel frequencies down to a VHF channel. Translation is advantageous since a CATV system necessarily operates with lengthy coaxial cables and the transmission loss through the cable is much greater at UHF than at VHF frequencies. various inputs including those from translators are combined in a suitable mixer. The set-up from the antennas to this combiner is called a head-end.

Further, as shown in the figure the CATV outputs from the combiner network are fed to a number of trunk cables through a broadband distribution amplifier. The trunk cables carry signals from the antenna site to the utilization site (*s*) which may be several kilometres away. Feeder amplifiers are provided at several points along the line to overcome progressive signal attenuation which occurs due to cable losses. Since cable losses are greater at higher frequencies it is evident that high-band attenuation will be greater than low-band attenuation. Therefore, to equalize this the amplifiers and signal splitters are often supplemented by equalizers. An equalizer or tilt control consists of a bandpass filter arrangement with an adjustable frequency response. It operates by introducing a relative low-frequency loss so that outputs from the amplifiers or splitters have uniform relative amplitude response across the entire VHF band. The signal distribution from splitters to tap-off points is done through multicore coaxial cables

In any case the signal level provided to a television receiver is of the order of 1.5 mV. This level provides good quality reception without causing acompanying radiation problems from the CATV system, which could cause interference to other installations and services.

Television Transmitter

An over simplified block diagram of a monochrome TV transmitter Is shown in Fig. 5.31.The luminance signal from the camera is amplified and synchronizing pulses added before feeding it to the modulating amplifier. Synchronising pulses are transmitted to keep the camera and picture tube beams in step. The allotted picture carrier frequency is generated by a crystal controlled oscillator. The continuous wave (CW) sine wave output is given large amplification before feeding to the power amplifier where its amplitude is made to vary (AM)in accordance with the modulating signal received from the modulating amplifier. The modulated output is combined (see Fig. 5.31) with the frequency modulated (FM)sound signal in the combining network and then fed to the transmitting antenna for radiation.

Colour Transmitter

A colour 'IV transmitter is essentially the same as the monochrome transmitter except for the additional need that colour (chroma) information is also to be transmitted. Any colour system is made compatible with the corresponding mono• chrome system. Compatibility means that the colour1V signal must produce a normal black and white picture on a monochrome receiver and a colour receiver must be able

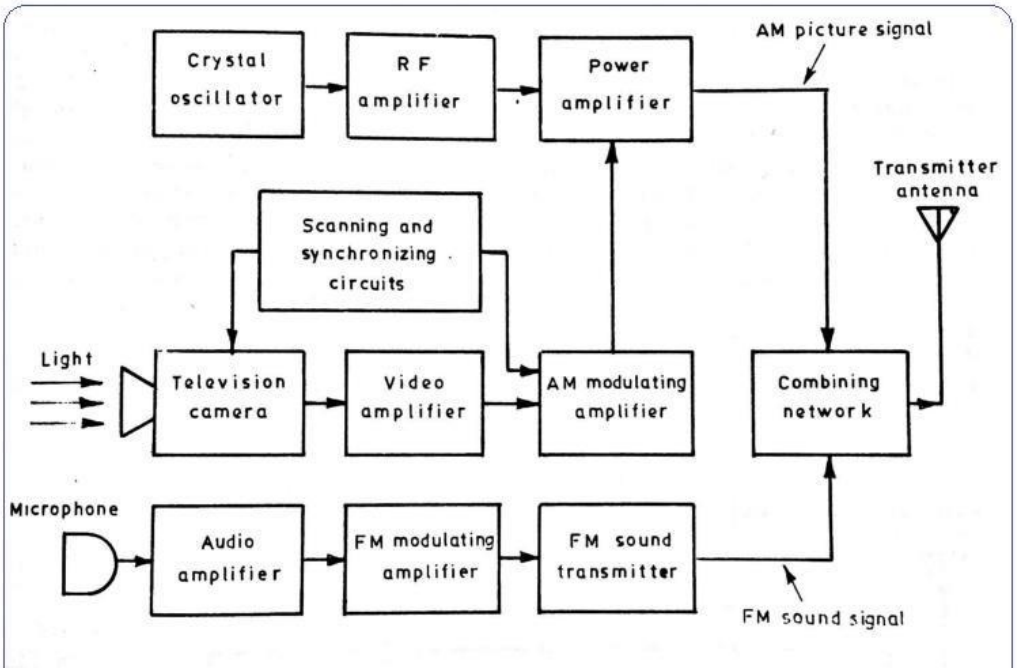


Fig. 5.31 Elementary block diagram of a monochrome television transmitter.

to produce a normal black and white picture from a monochrome TV signal. For this, the luminance (brightness) signal is transmitted in a colour system in the same way as in the monochrome system and with the same bandwidth. However, to ensure compatibility, the colour camera outputs are modified to obtain (B-Y) and (R-Y) signals. These are modulated on the colour sub-carrier, the value of which is so chosen that on combining with the luminance signal, the sidebands of the two do not interfere with each other i.e. the luminance and colour signals are correctly interleaved. A colour sync signal called 'colour burst' is also transmitted for correct reproduction of colours.

Sound Transmission

There is no difference in sound transmission between monochrome and colour TV systems. The microphone converts the sound associated with the picture being televised into proportionate electrical signal, which is normally a voltage. This electrical output, regardless of the complexity of its waveform, is a single valued function of time and so needs a single channel for its transmission. The audio signal from the microphone after amplification is frequency modulated, employing the assigned carrier frequency. In FM, the amplitude of carrier signal is held constant, whereas its frequency is varied in accordance with amplitude variations of the modulating signal. As shown in Fig. 5.31 output of the sound FM transmitter is finally combined with the AM picture transmitter output, through a combining network, and fed to a common antenna for radiation of energy in the form of electromagnetic waves.

Television Receiver

A simplified block diagram of a black and white TV receiver is shown in Fig. 5.32 The receiving antenna intercepts radiated RF signals and the tuner selects desired

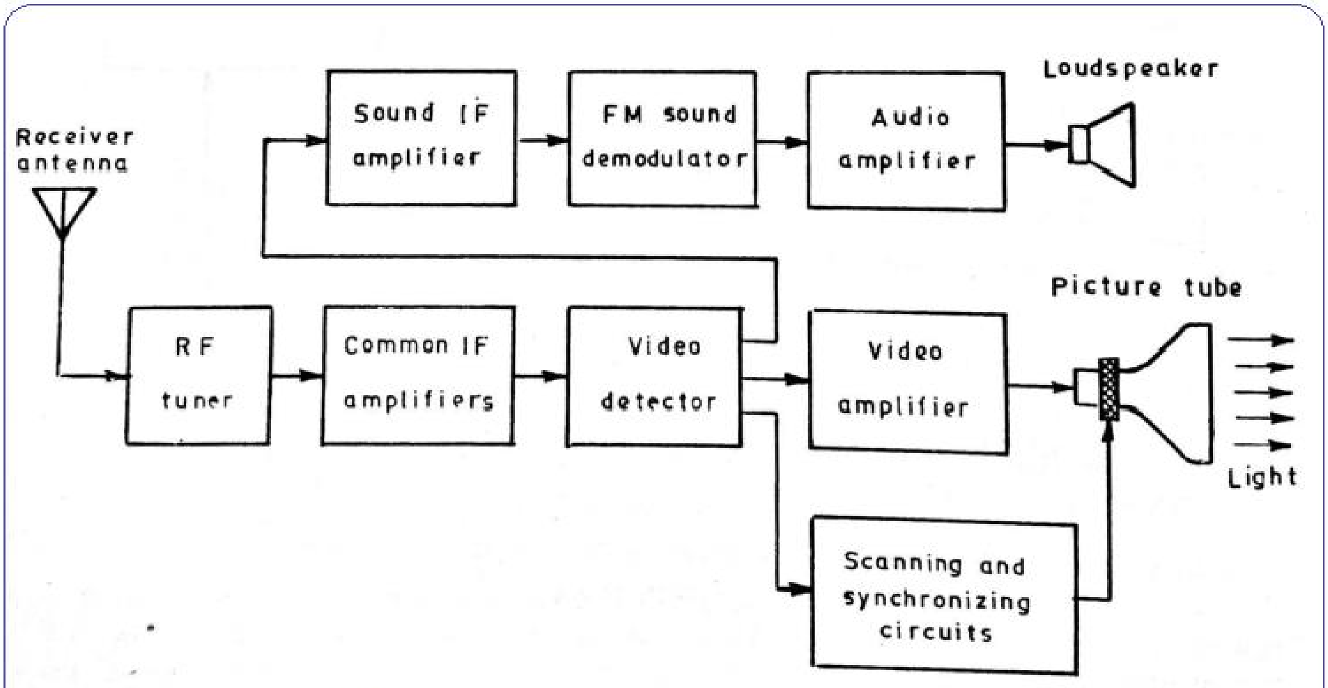


Fig. 5.32 Simplified block diagram of a black and white TV receiver.

channel's frequency band and converts it to the common IF band of frequencies. The receiver employs two or three stages of intermediate frequency (IF) amplifiers. The output from the last IF stage is demodulated to recover the video signal. This signal that carries picture information is amplified and coupled to the picture tube which converts the electrical signal back into picture elements of the same degree of black and white.

The picture tube shown in Fig.5.33 is very similar to the cathode-ray tube used in an oscilloscope. The glass envelope contains an electron-gun structure that produces a beam of electrons aimed at the fluorescent screen. When the electron beam strikes the screen, light is emitted. The beam is deflected by a pair of deflecting coils mounted on the neck of picture tube in the same way as the beam of camera tube scans the target plate. The amplitudes of currents in the horizontal and vertical deflecting coils are so adjusted that the entire screen, called raster, gets illuminated because of the fast rate of scanning.

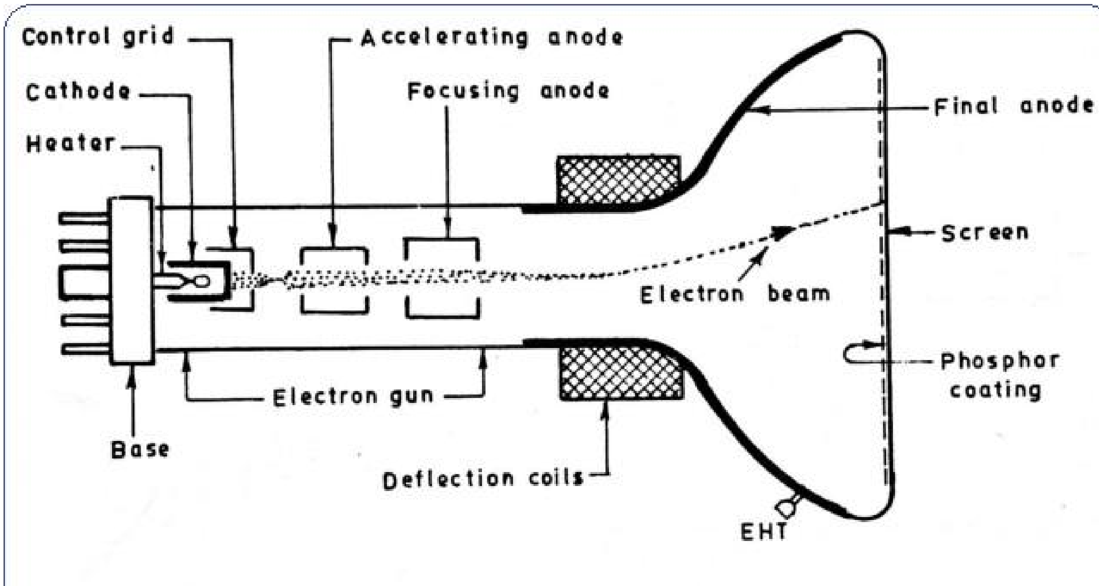


Fig. 5.33 Elements of a picture tube.

The video signal is fed to the grid or cathode of picture tube. When the varying signal voltage makes the control grid less negative, the beam current is increased, making the spot of light on the screen brighter. More negative grid voltage reduces brightness. If the grid voltage is negative enough to cut-off the electron beam current at the picture tube, there will be no light. This state corresponds to black. Thus the video signal illuminates the fluorescent screen from white to black through various shades of grey depending on its amplitude at any instant. This corresponds to brightness changes encountered by the electron beam of the camera tube while scanning picture details element by element. The rate at which the spot of light moves is so fast that the eye is unable to follow it and so a complete picture is seen because of storage capability of the human eye.

Sound *Reception*

The path of sound signal is common with the picture signal from antenna to Video detector section of the recetver. Here the two signals are separated and fed to their respective channels. The frequency modulated audio signal is demodulated after at least one stage of amplification. The audio output from the FM detector Is given due amplification before feeding it to the loudspeaker.

Overview of Telephony, Telegraphy

Basic Telephony:

As shown in the figure 5-34 all the landline telephones are connected with the use of twisted pair cables to the local exchange or central office(CO). Thousands of lines are connected to the central offices like this. All the COs are connected with the use of some medium either fiber optic or wireless link etc in the large telephone system.

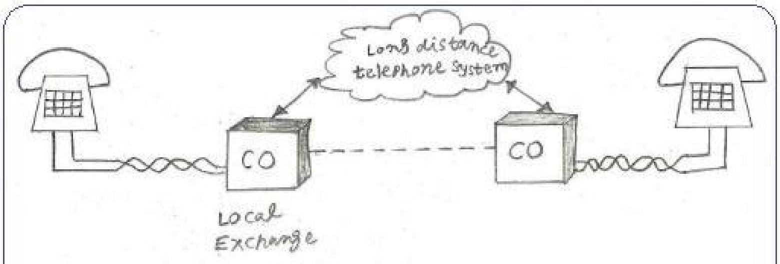


Fig 5-34 Basic Telephone System

Electronic Telephone

Usually any telephone set will have following basic functions in the transmit and receive direction:

In the transmit mode,

- Indication to the user that call can or cannot be made by way of dial tones and busy tones respectively.
- means to send the number to be dialled to the telephone CO or local exchange.

In the receive mode,

It should have ringer so that it rings the bell indicating call is received.

- Signal to the exchange(telephone system) that the call has been answered
- Device to convert voice to the electrical signal and vice versa, such device is called as transducer. In a telephone system all the telephones connected to the central offices are provided with group of basic electronic circuits referred as SLIC(Subscriber Line Interface Circuits). This SLIC has following basic functions referred as **BORSCHT**.

- Provision of **B**attery feed
- Provision of **O**ver voltage protection mechanism
- Provision of **R**inging circuit
- Provision of **S**upervision, which takes care of off-hook,ground start and ring trip signaling.
- Provision of **C**odec functionalities
- Provision of **H**ybrid and signal conditioning, which takes care of 2 wire to 4 wire conversion and vice versa.
- Provision of **T**esting the telephone system

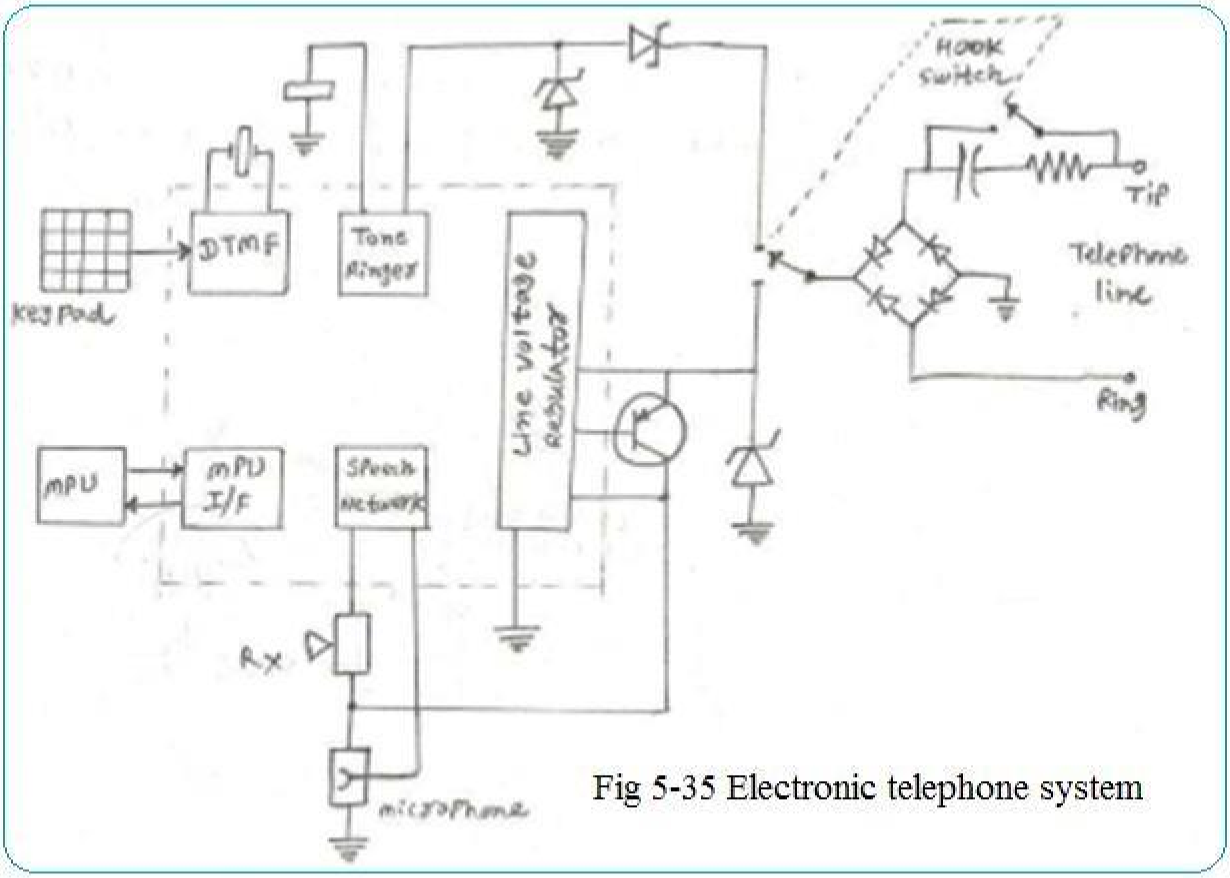


Fig 5-35 Electronic telephone system

Modern electronic telephones are designed based on single chip as shown in the block circuit diagram above.

A **telephone hook** or **switchhook** is an electrical switch which indicates when the phone is hung up. When the telephone was not in use, the receiver was hung on a spring-loaded hook; its weight would cause the hook to swing down and open an electrical contact. When the handset is on the cradle, the telephone is said to be "on-hook", or ready for a call. When the handset is off the cradle, the telephone is said to be "off-hook", or unable to receive any (further) calls. Pushing the switchhook quickly is termed a "hook flash".

Dual-tone multi-frequency signaling (**DTMF**) is an in-band telecommunication signaling system using the voice-frequency band over telephone lines between telephone equipment and other communications devices and switching centers. **Multi-frequency signaling** (MF) is a group of signalling methods that use a mixture of two pure tone (pure sine wave) sounds.

The DTMF system uses a set of eight audio frequencies transmitted in pairs to represent 16 signals, represented by the ten digits, the letters A to D, and the symbols # and *. As the signals are audible tones in the voice frequency range, they can be transmitted through electrical repeaters and amplifiers, and over radio and microwave links, thus eliminating the need for intermediate operators on long-distance circuits.

BASIC TELEGRAPH TRANSMITTER

There are various ways of modulating a radio wave. One method is to turn on and off a radio transmitter in accordance with a prearranged code like the dots and dashes of the Morse telegraph code. This system of radio telegraphy is the CW (continuous wave) system of radio transmission.

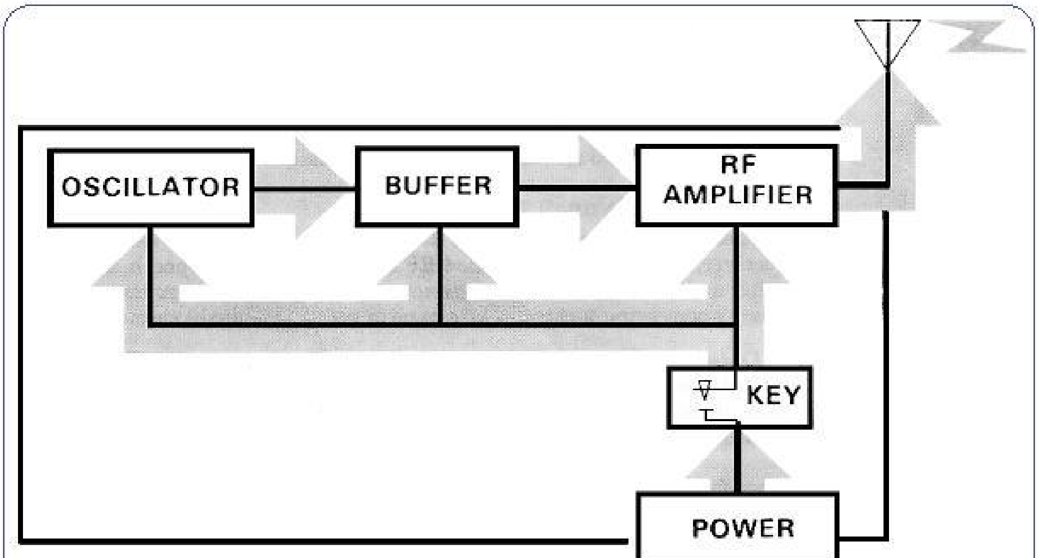


Fig 5-36 CW Telegraph Transmitter

Continuous wave (CW) transmitter: A radio transmitter is used to generate RF energy which is radiated into space. The transmitter may contain only a simple oscillator stage. Usually the output of the oscillator is applied to a buffer stage to increase its stability and to a power amplifier to produce high power output (refer fig 5-36). The telegraph key is used to control the energy waves produced by the transmitter. When the key is closed the transmitter produces its maximum output and when it is open no output is produced.