LAGOS CITY POLYTECHNIC

LECTURE NOTE

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CONTENTS

CHAPTER 1.0

BASIC PRINCIPLES OF TELECOMMUNICATION SYSTEM

With advancements of every society regarding educational, business, manufacturing, computing, agricultural sectors etc, there became an intense desire for individuals (humans), as well as objects to communicate with other individuals such relatives, friends, associates, etc across the globe. Interactions such as such Human to human (H2H), Machine to Machine (M2M), or even human to machine (H2M) usually involves communication. The science of communication involving long distances is called Telecommunication derived from the word "tele", meaning long distance.

Telecommunication is the transmission of signs, signals, messages, words, writings, images and sounds (or intelligence) of any nature by wire, radio, optical or other electromagnetic systems over a distance.
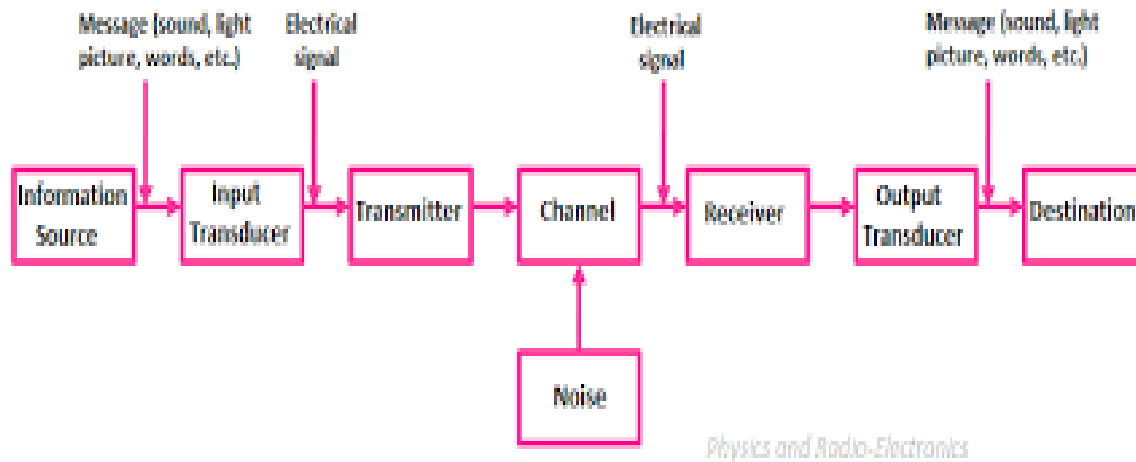

Figure 1.i: Block Diagram of a communication system

FUNCTIONS OF THE BLOCK DIAGRAMS ABOVE

- **Information Source**: the message to be communicated originates in the information source. There can be various messages in the form of words, group of words, code, symbols, sound signal etc. out of all these messages, only the desired message is selected and communicated. Therefore the function of the information source is to produced required message which has to be transmitted.
- **Input Transducer**: a transducer is a device which converts one forma of energy into another form. In a case where the message produced by the information source is not electrical in nature, an input transducer is used to convert it into a time varying electrical signal
- **Transmitter**: the function of the transmitter is to process the message signal into a form suitable for transmission over the communication channel.

- **The Channel & the noise**: this is the medium through which the message travels from the transmitter to the receiver. There are two types of channels: **point-to-point** and **broadcast channel.** Examples of point-to-point channels are wire lines, microwave links and optical fibers. Broadcast channel provides a capacity where several receiving stations can be reached simultaneously from a single transmitter. An example of the broadcast channel is a satellite in geostationary orbit, which covers about one-third of the earth' surface.
- **Receiver**: this reproduces the message signal in electrical form from the distorted received signal through the process known as demodulation or detection.
- **Output Transducer**: this is the conversion of electrical message signal into its original form. For example, in radion broadcasting, the output transducer is the loudspeaker which which works by converting the electrical signal in the form of original sound signal

In technical context, communication defines transmission, reception and processing of information by electronic means. The original communication systems viz: line telegraphy was invented in the eighteen forties. But radio communication became a reality in the beginning of the 20th century with the invention of triode devices. Radio communication has evolved immensely from the 2nd world war till date. This is as a result of the explosion in the invention of transistors, integrated circuits and other semi-conductor devices. Moore's law of transistor count still drives the electronic industries supporting telecommunication till date. In the recent years, communication has become more widespread with the use of satellites, fiber optics, and long term evolution networks (3-5Gs). Telemetry and radar systems play vital role in our military defense, navigation and even scientific research engagements.
Generally, a comprehensive learning of telecommunication systems as highlighted above will stimulate critical thinking on how to match human problems with technological solutions

Model of Communication System
To transfer information from point *A* to point *B*, another form of link is required between the two pints. A communication system therefore specifies the totality of mechanisms that offers the needed information link. For example, an electrical communication system could serve this purpose being the information link. The main idea behind any communication system is to successfully replicate the source information at the destination. As such, for successful communication to be achieved, the message received at the destination must remain identical to the original message evolving from the source. The different steps involved in the transmission of information are enumerated below.

i. Origin of information in the mind of the person who wants to communicate.
ii. Generation of message signal carrying the information.
iii. Converting the message signal into electrical form using a suitable transducer.
iv. Processing the message signal such that it will have the capability to travel for a long distance.
v. Transmission of the processed message signal to the desire destination
vi. Reception of the processed message signal at the desired destination
vii. Processing the received message signal in such a way to recreate the original nonelectrical form

viii. Finally, delivering the inform from the message signal to the intended person

A clear understanding of the basic issues in the above steps independent of the type of communication system will lead to comprehensive understanding of any communication system such as telephony, radio broadcasting, television broadcasting, radar communication, satellite communication, fiber optics communication, computer communication

ASSIGNMENT 1.0

1. Of what importance is telecommunication in our modern age?
2. Draw another block diagram of communication system apart from the one in this section and explain the effect of noise

QUIZ 1.0

1. Differentiate between transmitter and receiver
2. Differentiate between input and output transducers
3. State three challenges that may be encountered in telecommunication system
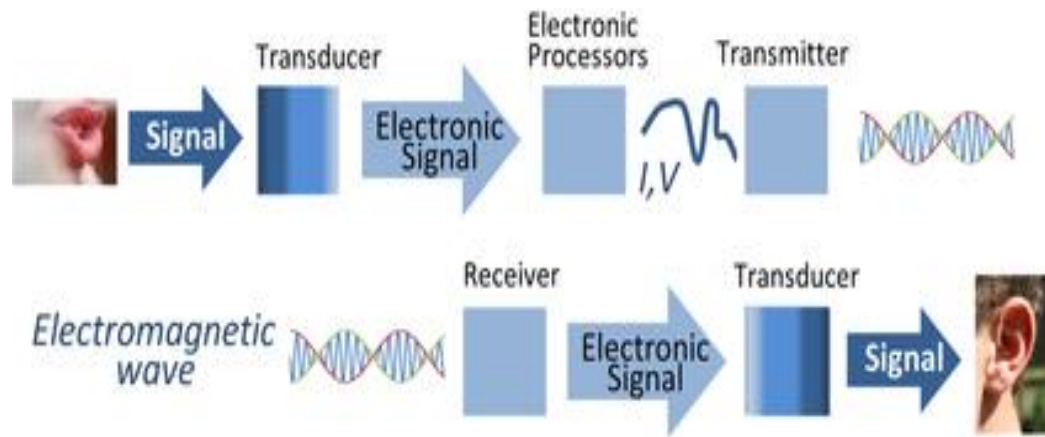
TRANSDUCERS

A **transducer** is a device that converts energy from one form to another. Usually a transducer converts a signal in one form of energy to a signal in another.[1]

Transducers are often employed at the boundaries of automation, measurement, and control systems, where electrical signals are converted to and from other physical quantities (energy, force, torque, light, motion, position, etc.). The process of converting one form of energy to another is known as transduction.

It is the first sensing element and is required only when measuring a non-electrical quantity, say, temperature or pressure. Its function is to convert the non-electrical physical quantity into an electrical signal. Of course, a transducer is not required if the quantity being measured is already in the electrical form.

Transducers are used in electronic communications systems to convert signals of various physical forms to electronic signals, and *vice versa*. In this example, the first transducer could be a microphone, and the second transducer could be a speaker.

Types of Transducers

1. Active Transducers
2. Passive Transducers
3. Bidirectional Transducers

- **Active Transducers**

Active transducers/ sensors generate an electric current in response to an external stimulus which serves as the output signal without the need of an additional energy source. Such examples are a photodiode, and a piezoelectric sensor, thermocouple

- **Passive Transducers**

Passive sensors/transducers require an external power source to operate, which is called an excitation signal. The signal is modulated by the sensor to produce an output signal. For example, a thermistor does not generate any electrical signal, but by passing an electric current through it, its resistance can be measured by detecting variations in the current or voltage across the thermistor

- **Bidirectional Transducers**

These convert physical phenomenal to electrical signals and also converts electrical signals into physical phenomena. Examples of inherently bidirectional transducers are **antennae**, which convert conducted electrical signal to or from propagating electromagnetic waves, and **voice coils,** which convert electrical signals into sound (when used in a loudspeaker) or sound into electrical signals (when used in a microphone).

APPLICATIONS OF VARIOUS TRANSDUCERS

- **Electromagnetic**:
  - Antennae – converts propagating electromagnetic waves to and from conducted electrical signals
  - magnetic cartridges – converts relative physical motion to and from electrical signals
  - Tape head, disk read-and-write heads – converts magnetic fields on a magnetic medium to and from electrical signals
  - Hall effect sensors – converts a magnetic field level into an electrical signal
- **Electrochemical**:
  - pH probes
  - Electro-galvanic oxygen sensors
  - Hydrogen sensors
- **Electromechanical (electromechanical output devices are generically called actuators):**
  - Accelerometers
  - Air flow sensors
  - Electroactive polymers
  - Rotary motors, linear motors
  - Galvanometers
  - Linear variable differential transformers or rotary variably differential transformers

- o Load cells – converts force to mV/V electrical signal using strain gauges
  - o Microelectromechanical systems
  - o Potentiometers (when used for measuring position)
  - o Pressure sensors
  - o String potentiometers
  - o Tactile sensors
  - o Vibration powered generators
  - o Vibrating structure gyroscopes
- Electroacoustic:
  - o Loudspeakers, earphones – converts electrical signals into sound (amplified signal → magnetic field → motion → air pressure)
  - o Microphones – converts sound into an electrical signal (air pressure → motion of conductor/coil → magnetic field → electrical signal)[2]
  - o Pickup (music technology) – converts motion of metal strings into an electrical signal (magnetism → electrical signal)
  - o Tactile transducers – converts electrical signal into vibration (electrical signal → vibration)
  - o Piezoelectric crystals – converts deformations of solid-state crystals (vibrations) to and from electrical signals
  - o Geophones – converts a ground movement (displacement) into voltage (vibrations → motion of conductor/coil → magnetic field → signal)
  - o Gramophone pickups – (air pressure → motion → magnetic field → electrical signal)
  - o Hydrophones – converts changes in water pressure into an electrical signal
  - o Sonar transponders (water pressure → motion of conductor/coil → magnetic field → electrical signal)
  - o Ultrasonic transceivers, transmitting ultrasound (transduced from electricity) as well as receiving it after sound reflection from target objects, availing for imaging of those objects
- Electro-optical (Photoelectric):
  - o Fluorescent lamps – converts electrical power into incoherent light
  - o Incandescent lamps – converts electrical power into incoherent light
  - o Light-emitting diodes – converts electrical power into incoherent light
  - o Laser diodes – converts electrical power into coherent light
  - o Photodiodes, photoresistors, phototransistors, photomultipliers – converts changing light levels into electrical signals
  - o Photodetector or photoresistor or light dependent resistor (LDR) – converts changes in light levels into changes in electrical resistance
  - o Cathode-ray tubes (CRT) – converts electrical signals into visual signals
- Electrostatic:
  - o Electrometers
- Thermoelectric:
  - o Resistance temperature detectors (RTD) – converts temperature into an electrical resistance signal

- o Thermocouples – converts relative temperatures of metallic junctions to electrical voltage
- o Thermistors (includes PTC resistor and NTC resistor)
- Radioacoustic:
  - o Geiger-Müller tubes – converts incident ionizing radiation to an electrical impulse signal
  - o Radio receivers converts electromagnetic transmissions to electrical signals.
  - o Radio transmitters converts electrical signals to electromagnetic transmissions.

- A **microphone**,– a transducer – that converts sound into an electrical signal. Microphones are used in many applications such as telephones, hearing aids, public address systems for concert halls and public events, motion picture production, live and recorded audio engineering, sound recording, two-way radios, megaphones, radio and television broadcasting, and in computers for recording voice, speech recognition, VoIP, and for non-acoustic purposes such as ultrasonic sensors or knock sensors.
- Several types of microphone are in use, which employ different methods to convert the air pressure variations of a sound wave to an electrical signal. The most common are the dynamic microphone, which uses a coil of wire suspended in a magnetic field; the condenser microphone, which uses the vibrating diaphragm as a capacitor plate; and the piezoelectric microphone, which uses a crystal of piezoelectric material. Microphones typically need to be connected to a preamplifier before the signal can be recorded or reproduced.

MICROPHONES

As stated earlier, Microphone is a transducer that converts sound into an electrical signal.

There are various types of microphones which include the following:

- Carbon microphone
- Dynamic microphone
- Ribbon microphone
- Crystal microphone
- Fiber optic microphone
- Laser microphone
- Etc.

Parameters used to describe the quality of a microphones:

- The first is the **output level**, which can be described either as an absolute output level in watts, when a reference level of sound pressure signal at 1000Hz is applied to the microphone; or in decibel referred to a standard power output level under similar input

conditions. The power output level so measured gives a measure of the sensitivity of ther microphone.

- The flat frequency response over the entire **audio range from 20HZ to 20kHz** with no response at all outside this range is the second parameter
- The third is the directional response in their pickup characteristics. This directionality is considered in much the same manner as the directionality of the antennas

In this section, we shall look into the carbon and crystal microphones

CARBON MICROPHONE:

The basic concept behind the carbon microphone is the fact that when carbon granules are compressed their resistance decreases. This occurs because the granules come into better contact with each other when they are pushed together by the higher pressure.

The carbon microphone comprises carbon granules that are contained within a small contained that is covered with a thin metal diaphragm. A battery is also required to cause a current to flow through the microphone.

Figure 2.1: Carbon Microphone

When sound waves strike the carbon microphone diaphragm it vibrates, exerting a varying pressure onto the carbon. These varying pressure levels are translated into varying levels of resistance, which in turn vary the current passing through the microphone
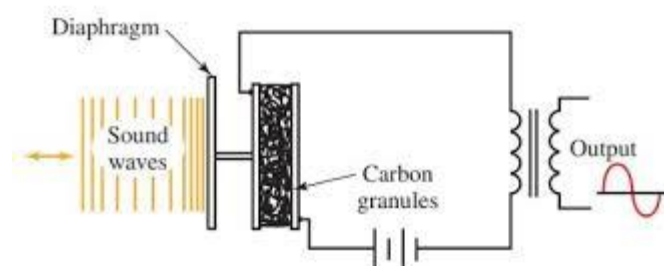

Figure 2.2: Schematic Diagram

As radio started to be used, the carbon microphone was initially used there as well – for broadcasting as well as communications purposes. However their use in broadcast applications soon came to end because of the drawbacks of noise and poor frequency response. Other types of microphone started to become available and their use was preferred because of the better fidelity that was available. The use of the carbon microphone persisted for many years for communications purposes as they gave a high output and they were robust. The poor frequency response was not an issue.

The carbon microphone was used for telephones up until the 1970s and 1980s, but even there it became possible to use other types of microphone more conveniently. Also the crackle and noise of the carbon microphone had always been an issue and when other types of microphone became available at a low cost they started t be used, despite the requirement for additional electronics needed.

Carbon microphones are now only used in a very few applications – typically only specialist applications. They are able to withstand high voltage spikes and this property lends itself to use in a small number of applications

Carbon microphone advantages

- High output
- Simple principle & construction
- Cheap and simple to manufacture

Carbon microphone disadvantages

- Very noisy - high background noise and on occasions it would crackle
- Poor frequency response
- Requires battery or other supply for operation

CRYSTAL MICROPHONE

A **crystal microphone** or **piezo microphone**[27] uses the phenomenon of piezoelectricity—the ability of some materials to produce a voltage when subjected to pressure—to convert vibrations into an electrical signal. An example of this is potassium sodium tartrate, which is a piezoelectric crystal that works as a transducer, both as a microphone and as a slimline loudspeaker component. Crystal microphones were once commonly supplied with vacuum tube (valve) equipment, such as domestic tape recorders. Their high output impedance matched the high input impedance (typically about 10 megohms) of the vacuum tube input stage well. They were difficult to match to early transistor equipment and were quickly supplanted by dynamic microphones for a time, and later small electret condenser devices. The high impedance of the crystal microphone made it very susceptible to handling noise, both from the microphone itself and from the connecting cable.

Figure 2.3 : Piezoelectric Rubber

Piezoelectric transducers are often used as contact microphones to amplify sound from acoustic musical instruments, to sense drum hits, for triggering electronic samples, and to record sound in challenging environments, such as underwater under high pressure. Saddle-mounted pickups on acoustic guitars are generally piezoelectric devices that contact the strings passing over the saddle. This type of microphone is different from magnetic coil pickups commonly visible on typical electric guitars, which use magnetic induction, rather than mechanical coupling, to pick up vibration.

Advantages:
There are some advantages of piezoelectric transducer which are given below,

o   The piezoelectric transducer is available in desired shape.
o   It has rugged construction.
o   It is small in size.
o   It has good frequency response.
o   It has negligible phase shift

Disadvantages:
There are some disadvantages of piezoelectric transducer which are given below,
o   The piezoelectric transducer is used for dynamic measurement only.
o   It has high temperature sensitivity.
o   Some crystals are water soluble and get dissolve in high humid environment.

Applications:
There are some important applications of piezoelectric transducer which are given below,

o   The piezoelectric transducer is used in spark ignition engines.
o   It can be used in electronic watches.
o   It is used in accelerometer.
o   It is used in record player.

LOUDSPEAKERS

A **loudspeaker** is an electroacoustic transducer; a device which converts an electrical audio signal into a corresponding sound. The most widely used type of speaker in the 2010s is the **dynamic speaker,** invented in 1924 by Edward W. Kellogg and Chester W. Rice.

DYNAMIC SPEAKER

The dynamic speaker operates on the same basic principle as a dynamic microphone, but in the off reverse, to produce sound from an electrical signal. When an alternating current electrical audio signal is applied to its voice coil, a coil of wire suspended in a circular gap between the poles of a permanent magnet, the coil is forced to move rapidly back and forth due to Faraday's law of induction, which causes a diaphragm (usually conically shaped) attached to the coil to move back and forth, pushing on the air to create sound waves. Besides this most common method, there are several alternative technologies that can be used to convert an electrical signal into sound. The sound source (e.g., a sound recording or a microphone) must be amplified or strengthened with an audio power amplifier before the signal is sent to the speaker.

TWEETER

A tweeter is the smallest type of loudspeaker that is also known as the treble speaker. The speaker is designed to reproduce the upper limit of the audible frequency range. It varies between tweeters, but typically the sound frequency it delivers ranges from 2,000 Hz to 20,000 Hz.

MID-RANGE DRIVER

A mid-range speaker is a driver that is also known as a squawker. It is designed to deliver sound from 250 to 2000 Hz frequency range.

LOW RANGE OR BASS

The low range frequency gets reproduced by woofers and sub-woofers. The word gets derived from the dog's barking or a 'woof', which uses lower frequency waves, compared to birds 'tweeting' that occupy the top of the audio spectrum.

The difference between woofers and sub-woofers is in the frequency range they are designed to reproduce with former typically working within the 40 Hz to 500 Hz range and latter occupying sub 100 Hz frequencies

Most consumer-grade loudspeakers combine woofers and sub-woofers into a single speaker, yet as you move up in the sound fidelity, those two get separated for purer, cleaner and more refined low-frequency sound.

WOOFER

A woofer, also called a bass speaker is a term for loudspeaker or a driver tasked with reproducing low frequency sounds.

Most of the time, it features a electrodynamic driver made of strong paper or various **polymers**. With the lowest end of human hearing being around 20 Hz, woofers don't typically exhaust human hearing capabilities working in 40 Hz and upwards range.

Figure 2.4: Piezoelectric speakers

A piezoelectric buzzer. The white ceramic piezoelectric material can be seen fixed to a metal diaphragm.

Piezoelectric speakers are frequently used as beepers in watches and other electronic devices, and are sometimes used as tweeters in less-expensive speaker systems, such as computer speakers and portable radios. Piezoelectric speakers have several advantages over conventional loudspeakers: they are resistant to overloads that would normally destroy most high frequency drivers, and they can be used without a crossover due to their electrical properties. There are also disadvantages: some amplifiers can oscillate when driving capacitive loads like most piezoelectrics, which results in distortion or damage to the amplifier. Additionally, their frequency response, in most cases, is inferior to that of other technologies. This is why they are generally used in single frequency (beeper) or non-critical applications.

Piezoelectric speakers can have extended high frequency output, and this is useful in some specialized circumstances; for instance, sonar applications in which piezoelectric variants are used as both output devices (generating underwater sound) and as input devices (acting as the

sensing components of underwater microphones). They have advantages in these applications, not the least of which is simple and solid state construction that resists seawater better than a ribbon or cone based device would.

Figure 2.5: Electrostatic loudspeaker
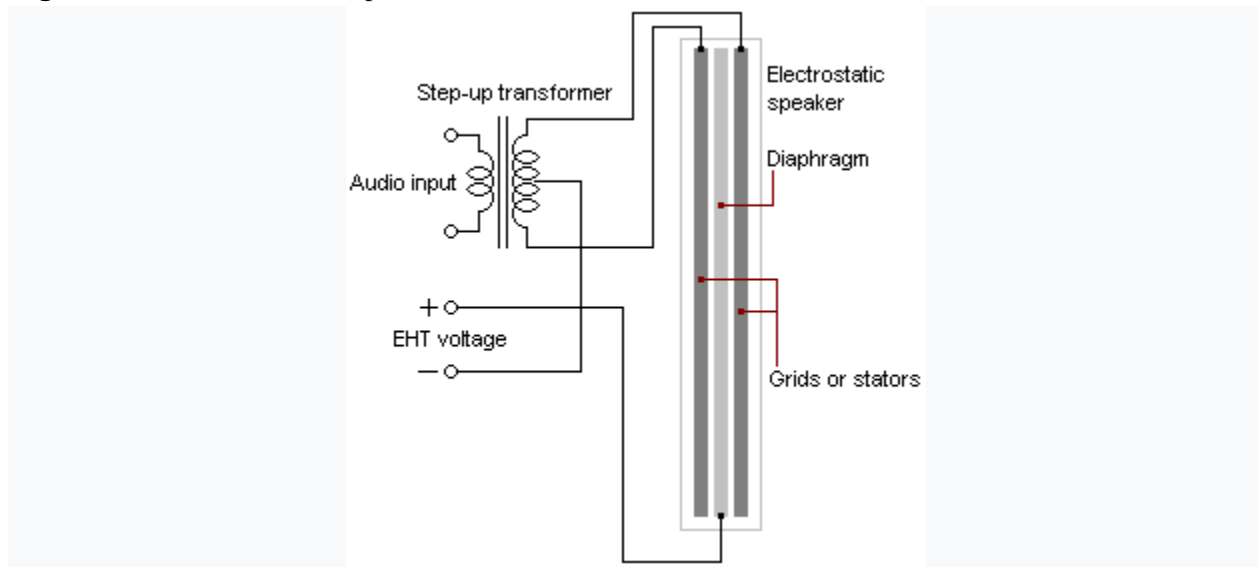
Schematic showing an electrostatic speaker's construction and its connections. The thickness of the diaphragm and grids has been exaggerated for the purpose of illustration.

Electrostatic loudspeakers use a high voltage electric field (rather than a magnetic field) to drive a thin statically charged membrane. Because they are driven over the entire membrane surface rather than from a small voice coil, they ordinarily provide a more linear and lower-distortion motion than dynamic drivers. They also have a relatively narrow dispersion pattern that can make for precise sound-field positioning. However, their optimum listening area is small and they are not very efficient speakers. They have the disadvantage that the diaphragm excursion is severely limited because of practical construction limitations—the further apart the stators are positioned, the higher the voltage must be to achieve acceptable efficiency. This increases the tendency for electrical arcs as well as increasing the speaker's attraction of dust particles. Arcing remains a potential problem with current technologies, especially when the panels are allowed to collect dust or dirt and are driven with high signal levels.

MOVING COIL LOUDSPEAKER

The moving coil loudspeaker is the most widely known and used form of loudspeaker. It can be found in many electronic items from radios to Bluetooth speakers and in public address systems - in fact anywhere that electrical waveforms need to be turned into audible sound.

The moving coil loudspeaker performs well and is able to be manufactured relatively easily. However, like all transducers converting electrical waveforms into sound, its operation can be complex as the moving coil loudspeaker links electrical and mechanical domains.

Figure 2.6: Moving Coil Speakers

The moving coil loudspeaker uses the magnetic effect generated by a flowing current as the basis of its operation.

When a current flows in a wire, a magnetic field appears around it. When the wire is wound into a coil, the effect is increased.

Wireless speakers are very similar to traditional (wired) loudspeakers, but they receive audio signals using radio frequency (RF) waves rather than over audio cables. There is normally an amplifier integrated in the speaker's cabinet because the RF waves alone are not enough to drive the speaker. This integration of amplifier and loudspeaker is known as an active loudspeaker. Manufacturers of these loudspeakers design them to be as lightweight as possible while producing the maximum amount of audio output efficiency.

Wireless speakers still need power, so require a nearby AC power outlet, or possibly batteries. Only the wire to the amplifier is eliminated.

ASSIGNMENTS 2 & 3

1. What are the factors that determine the qualities of a microphone and loudspeakers
2. State five applications of transducers
3. Discus on wireless microphone and wireless speakers

QUIZES 2 & 3

1. Sketch the schematic diagram of one type of microphone and one type of speaker and explain
2. Differentiate between active and passive transducers

CHAPTER 3.0

BASIC PRINCIPLES OF MODULATION & DEMODULATION

Introduction

For successful transmission and reception of intelligence (code, voice, music etc.) by the use of radio waves, two processes are essential :
(*i*) modulation and (*ii*) demodulation.

Speech and music etc. are sent thousands of kilometres away by a radio transmitter. The scene in front of a television camera is also sent many kilometres away to viewers. Similarly, a Moon probe or Venus probe checking its environments, sends the information it gathers millions of kilometres through space to receivers on earth. In all these cases, the carrier is the high-frequency radio wave. The intelligence i.e. sight, sound or other data collected by the probe is impressed on the radio wave and is carried along with it to the destination.

Modulation is the process of **combining the low-frequency signal with a very high-frequency radio wave called carrier wave (CW)**. The resultant wave is called **modulated carrier wave**. This job is done at the transmitting station.
Demodulation is the process of separating or **recovering the signal from the modulated carrier wave**. It is just the opposite of modulation and is performed at the receiving end.

It is the process of combining *an audio frequency* (*AF*) *signal with a radio frequency(RF) carrier wave*. The AF signal is also called a *modulating wave* and the resultant wave produced is called *modulated wave*.
Modulation is the **superimposing** of a low frequency (audio signal) on a high frequency (carrier wave). It is the process through which audio, video, image, or text information is added to an electrical or optical carrier signal to be transmitted over a telecommunication or electronic medium

What is a Carrier Wave?

It is a high-frequency undamped radio wave produced by radio-frequency oscillators , the output of these oscillators is first amplified and then passed on to an antenna. This antenna radiates out these high-frequency (electromagnetic) waves into space. These waves have constant amplitude and travel with the velocity of light. They are inaudible *i.e*. by themselves they cannot produce any sound in the loudspeaker of a receiver. As their name shows, their job is to **carry the signal** (audio or video) from transmitting station to the receiving station. The resultant wave is called **modulated** carrier wave as shown in **figure 3.1**
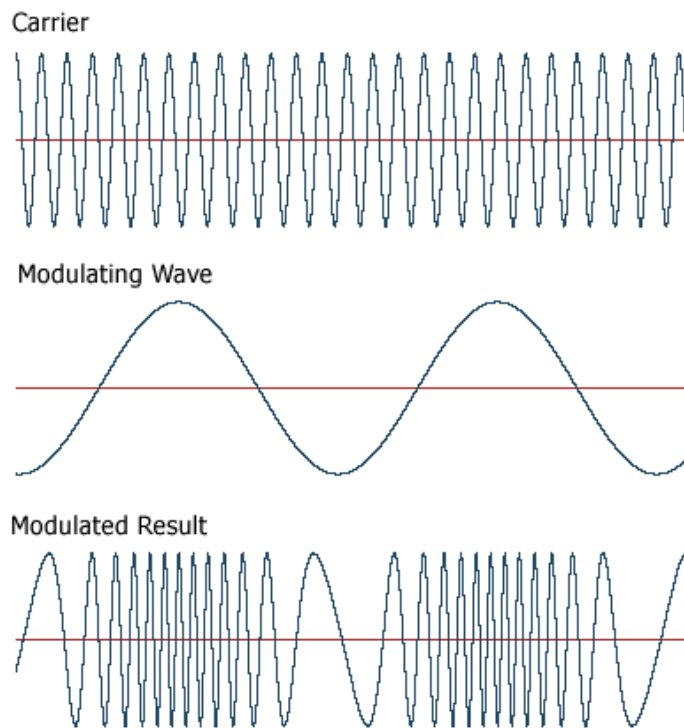
Figure 3.1 Carrier and modulating waves combined to form a modulated wave

Table 3.1:   Differences between Modulation and Demodulation

Modulation	Demodulation
Encoding of information	Decoding of information
Superimposing	Retrieving
Varying some parameters like amplitude, frequency and phase	Extraction of the audio signal from the transmitted modulated wave
Takes place at the source/transmitting end	Takes place at the destination/receiving end
Frequency transformation from low to high	Frequency transformation from high to low
Circuit modulator needed	Circuit demodulator needed
Operation is simple	Operation is complex

SOUND

It is a sort of disturbance which requires some physical medium for its propagation. Human voice consists of a series of compressions and rarefactions which travel through air with a velocity of about 345 m/s. The frequency range of human voice is from 20-4000 Hz which lies within the audible range of 20 to 20,000 Hz. Variations in human voice can be converted into corresponding variations in electric current with the help of a microphone shown in **figure 3.2**

When a sound wave strikes the microphone, it produces AF sound current. The positive halfcycles of sound current are produced by the compressions and negative half cycles by rarefactions.

As seen, human voice does not produce pure sinusoidal current because it does not consist of one frequency alone. It is quite complex and can be analysed to consist of a fundamental (or lowest frequency) and its integral multiple frequencies called **overtones** or **harmonics**.
We are interested in two main characteristics of sound :

(*i*) **Intensity**—It is the energy content of the wave. It depends on its amplitude. In fact, intensity of a wave is directly proportional to the square of its amplitude *i.e.* $I \propto a2$. Sensation of loudness felt by a listener depends directly on the intensity of the wave falling on his ears.

(*ii*) **Frequency**—It produces the sensation called pitch. Audible sounds have a frequency range from 20 Hz to 20,000 Hz. Though every sound has complex frequency structure, we will consider only single-frequency sound whose current wave is a pure sine wave .
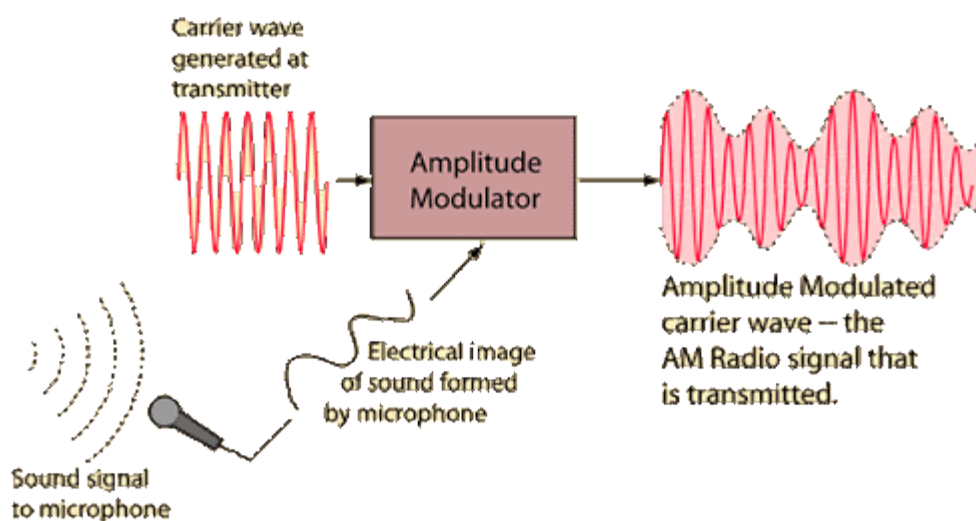

Figure 3.2: Transmission of sound wave

Need for Modulation

Sometimes, beginners question the necessity of modulation *i.e.* using a carrier wave to carry the low-frequency signal from one place to another. Why not transmit the signals directly and save lot of botheration? Unfortunately, there are three main hurdles in the process of such direct transmission of audio-frequency signals :

1. They have relatively short range,
2. If everybody started transmitting these low-frequency signals directly, mutual interference will render all of them ineffective
3. Size of antennas required for their efficient radiation would be large i.e. about 75 km

For efficient radiation of a signal, the minimum length of an antenna is one quarter wavelength **(λ/4).** The antenna length *L* is connected with the frequency of the signal wave by the relation

L = 75 × 106/*f* metres. For transmitting an audio signal of *f* = 1000 Hz, *L* = 75 × 106/103 = 75,000 m = 75 km ! In view of this immense size of antenna length, it is impractical to radiate audio-frequency signals directly into space.

Hence, the solution lies in modulation which enables a low-frequency signal to travel very large distances through space with the help of a high-frequency carrier wave. These carrier waves need reasonably-sized antennas and produce no interference with other transmitters operating in the same area.

Modulation is needed:
- For ease of radiation
- To reduce noise and interference
- To overcome equipment limitations

Radio Broadcasting

Let us see how radio broadcasting stations broadcast speech or music etc. from their broadcasting studios. First, the speech or music which consists of a series of compressions and rarefactions is translated into a tiny varying electric current with the help of a crystal microphone. The frequency of variations of this current lies in the audio-range, hence it is known as audio frequency signal. The audio-frequency signal cannot be radiated out from the antenna directly because transmission at audio-frequencies is not practical. For this purpose, oscillations of very high frequency or radio-frequency are produced with the help of any one of the oscillators

The electromagnetic waves so produced are of constant amplitude but of extremely high frequency. These waves, which are neither seen nor heard, travel through space with the velocity of light *i.e.* 3 × 108 m/s (approx). The audio frequency signal which is to be broadcast, is then superimposed on the *RF* waves, which are known as carrier waves (because they carry *A.F.* signal through space to distant places). In a way, the carrier waves can be likened to a horse and the audio-frequency signal to a rider. The process by which *AF* signal or information is impressed on the carrier wave is known as modulation. The horse and rider travel through space. At the receiving end, they strike the receiving aerial and enter the receiver which separates the horse from the rider. The horse *i.e.* carrier wave is returned and the rider *i.e.* audio-frequency signal is converted back into sound. This process by which the *R.F.* waves and *A.F.* waves are separated is known as **detection** or **demodulation** (because it is the reverse of modulation).
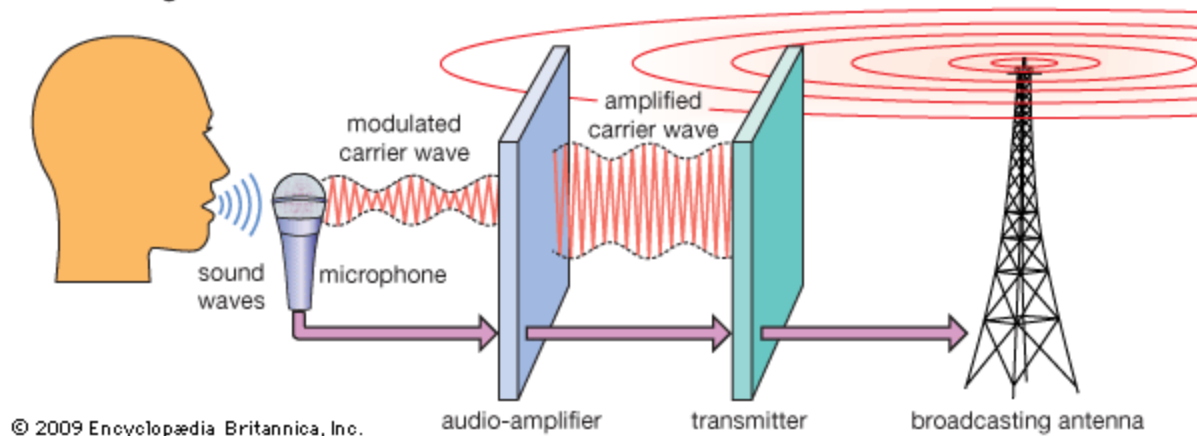
Transmitting Radio Waves

Figure 3.3: Transmission of radio waves

In sound transmission, the initial sound is first picked up by a microphone. The microphone generates a pulsating direct current called a black wave. Meanwhile an oscillator supplies a carrier wave. Electrical circuits combine the black wave and carrier wave into a modulated carrier pulse of alternating current. This pulse is amplified and used to radiate a carrier wave.

BENEFITS OF MODULATION

1. It helps to send a signal over a bandpass frequency range
2. It allows use of smaller antenna
3. It increases the range of communication
4. It improves quality of reception
5. It avoids mixing of signals
6. Multiplexing is possible

APPLICATIONS OF MODULATION

1. For broadcasting (radio and television), AM, FM, PM etc
2. GSM (global system for mobile communication)
3. Radar (Radio Detection and Ranging)
4. WiFi
5. Conversion of analog signal to digital signal

ASSIGNMENTS 4 & 5

1. State two different definitions of modulation
2. Differentiate between **carrier wave** and **modulating wave**
3. Show with a diagram how modulating wave and carrier wave combined to form a modulated wave
4. State five application areas of modulation

QUIZES 4 & 5 (Practical Session)

1. Explain the process involve in radio broadcasting
2. In groups, using the resources/equipment s available at the laboratory, demonstrate how sound signal can be transmitted , resulting in a modulated wave
3. What is the audio frequency range in broadcasting?
4. State three challenges you faced and how were you able to overcome them

CHAPTER 4-0

AMPLITUDE MODULATION

It is the process of combining **an audio frequency (AF) signal with a radio frequency(RF) carrier wave.** The AF signal is also called a **modulating wave** and the resultant wave produced is called **modulated wave.**

During modulation, some characteristic of the carrier wave is varied in time with the modulating signal and is accomplished by combining the two.

Methods of Modulation

The mathematical expression for a sinusoidal carrier wave is

$$e = EC \sin (\omega c \, t + \varphi) = EC \sin (2 \, \pi fc \, t + \varphi)$$

Obviously, the waveform can be varied by any of its following three factors or parameters :

1. EC — the amplitude, **2. fc** — the frequency, **3. φ** — the phase.

Accordingly, there are three types of sine-wave modulations known as :

1. Amplitude Modulation (AM)

Here, the information or *AF* signal changes the amplitude of the carrier wave without changing its frequency or phase.

2. Frequency Modulation (FM)

In this case, the information signal changes the frequency of the carrier wave without changing its amplitude or phase.

3. Phase Modulation (PM)

Here, the information signal changes the phase of the carrier wave without changing its other two parameters.

Amplitude Modulation

In this case, the **amplitude** of the carrier wave is varied in proportion to the **instantaneous amplitude** of the information signal or AF signal. Obviously, the amplitude (and hence the intensity) of the carrier wave is changed *but not its frequency.* Greater the amplitude of the AF signal, greater the fluctuations in the amplitude of the carrier wave.

The process of amplitude modulation is shown graphically in Figure. For the sake of simplicity, the *AF* signal has been assumed sinusoidal. The carrier wave by which it is desired to transmit the *AF* signal is shown. The resultant wave called modulated wave is also shown in the Figure.

The function of the modulator is to mix these two waves. All stations broadcasting on the standard broadcast band (550-1550 kHz) use *AM* modulation. If you observe the envelope of the modulated carrier wave, you will realize that it is an exact replica of the *AF* signal wave.
In summary
(*i*) **fluctuations** in the amplitude of the carrier wave depend on the **signal amplitude**,
(*ii*) **rate** at which these fluctuations take place depends on the **frequency** of the audio signal
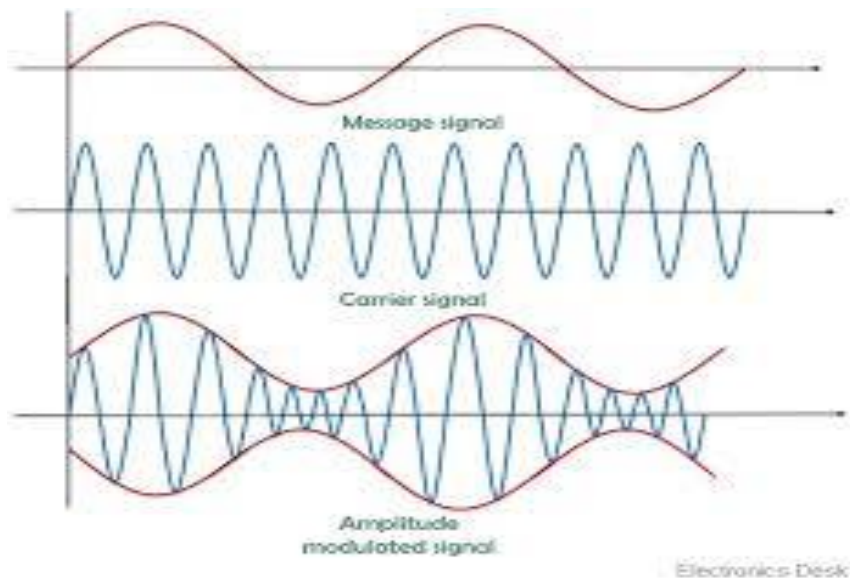

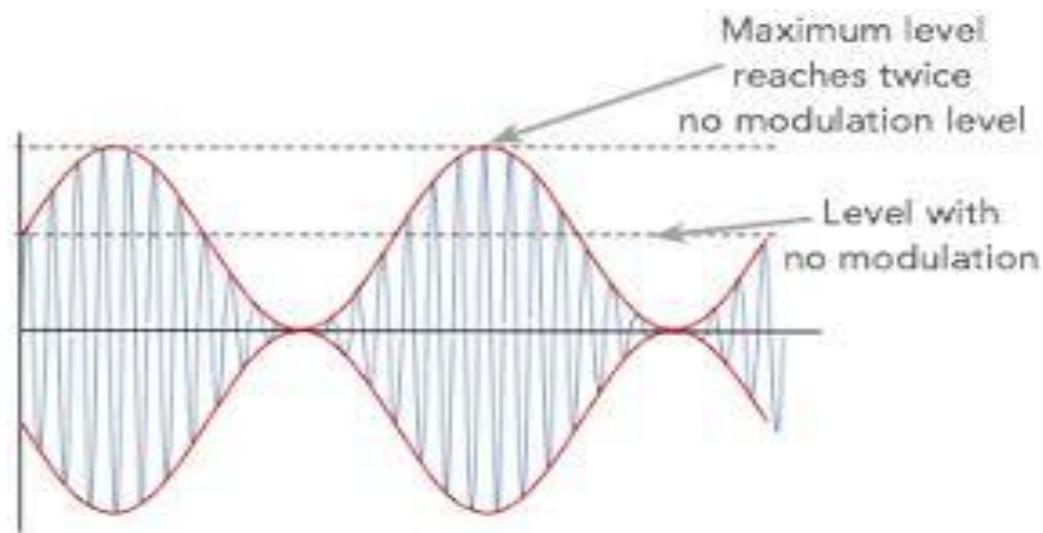**Figure 4.1**   Message signal, carrier wave and their mixing

Figure 4.2   **Modulated wave**

Percent Modulation

It indicates the degree to which the AF signal modulates the carrier wave

m = <u>maximum value of signal wave</u> X 100 = <u>Signal amplitude</u> X 100
maximum value of carrier wave Carrier amplitude

This is referred to as **modulation index** (MI)
i.e m = (modulation factor) = M.I X 100

In the image below for amplitude modulated sine wave:

- **0%** unmodulated, the sine envelope is not visible at all;

- **<** 100% modulation depth is normal AM use;

- **100%** modulation depth, the sine envelope touch at y=0. Maximum modulation that can be retrieved with an envelope detector without distortion;

- **>** 100% modulation depth, "overmodulation", the original sine wave can no longer be detected with an envelope detector.

Overmodulation:

1. The condition that prevails when the instantaneous level of the modulating <u>signal</u> exceeds the value necessary to produce 100% <u>modulation</u> of the <u>carrier</u>.
2. Overmodulation results in spurious emissions by the modulated carrier, and <u>distortion</u> of the recovered modulating signal.
3. Overmodulation in the sense of this definition is almost always considered a <u>fault</u> condition. **2.** The condition that prevails when the mean level of the modulating signal is such that peaks in the modulating signal exceed the value necessary to produce 100% modulation of the carrier.
4. Overmodulation in the sense of this definition, if not excessive, is sometimes considered permissible

- AM over modulation causes the carrier wave to invert it's phase when the modulating signal has an amplitude that is above a certain level. Broadcast AM typically never does this because the complexity of an accurate demodulator is too great for the thousands and millions of receivers.
- AM is just the mathematical multiplication of two signals and regular broadcast AM remains as a 2 quadrant multiplier whereas full modulation uses all four quadrants.
- 100% modulation is where the modulating signal drives the carrier to zero and is theoretically the maximum modulation that can be successfully demodulator by a regular AM envelope detector.
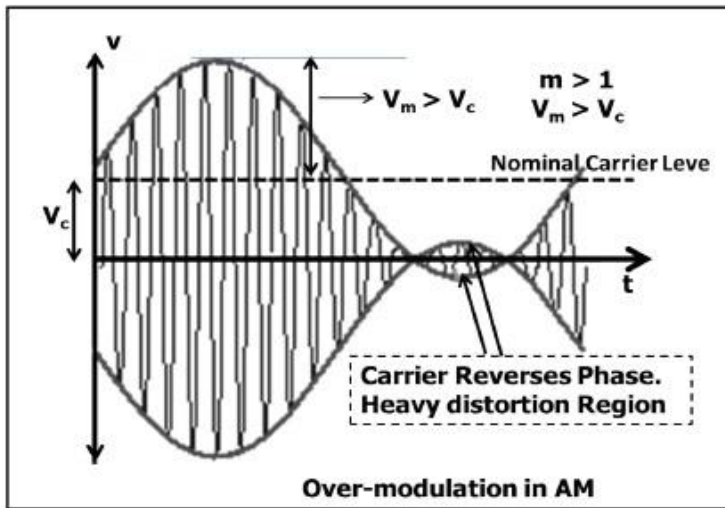
Figure 4.3

Over modulation isn't really of any significance to FM systems (unlike AM). If the modulation signal amplitude is too great, any decent frequency modulator will limit the signal so that it can't push the bandwidth of the modulated signal too wide in the frequency spectrum. In effect, the modulating signal becomes clipped.
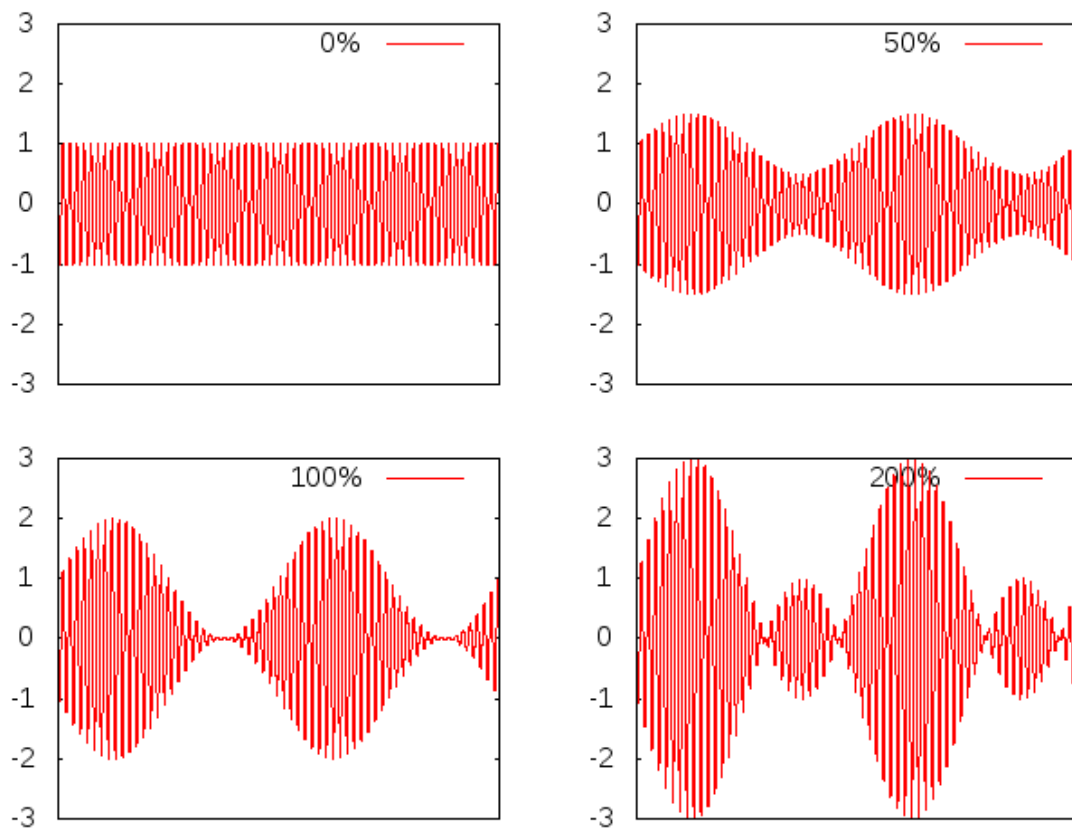

Figure 4.4: Various levels of modulation

Summary

- A carrier is used to make the wavelength smaller for practical transmission and to permit multiplexing.
- The spectrum is used to measure bandwidth (the range of frequencies) and the efficiency (the power in the side-bands compared to the total power)
- Bandwidth can be predicted using BW = 2 f_m where f_m = the maximum modulating frequency
- Efficiency depends only on the modulating index, m (the fraction of the carrier you modulate by)
- AM is limited to 33% efficiency because the modulation index cannot be increased to > 1.0 without introducing distortion in the receiver.

Upper and Lower Side Frequencies

An unmodulated carrier wave consists of only one single-frequency component of frequency *fc*. When it is combined with a modulating signal of frequency **fm**, heterodyning action takes place.
As a result, two additional frequencies called **side frequencies** are produced. The *AM* wave is found to consist of three frequency components :
1. The original carrier frequency component, *fc* .
2. A higher frequency component (*fc+ fm*). It is called the sum component.
3. A lower frequency component (*fc – fm*). It is called the difference component.

The two new frequencies are called the **upper-side frequency** (*USF*) and **lower-side frequency** *LSF*) respectively and are symmetrically located around the carrier frequency.
The modulating frequency remains unchanged but does not appear in the amplifier output because the amplifier's load presents practically zero impedance to this low frequency.
These are shown in time domain in Figure below
The amplitude of the side frequencies depends on the value of *m*. The amplitude of each side frequency = *mA*/2 where *A* is the amplitude of unmodulated carrier wave
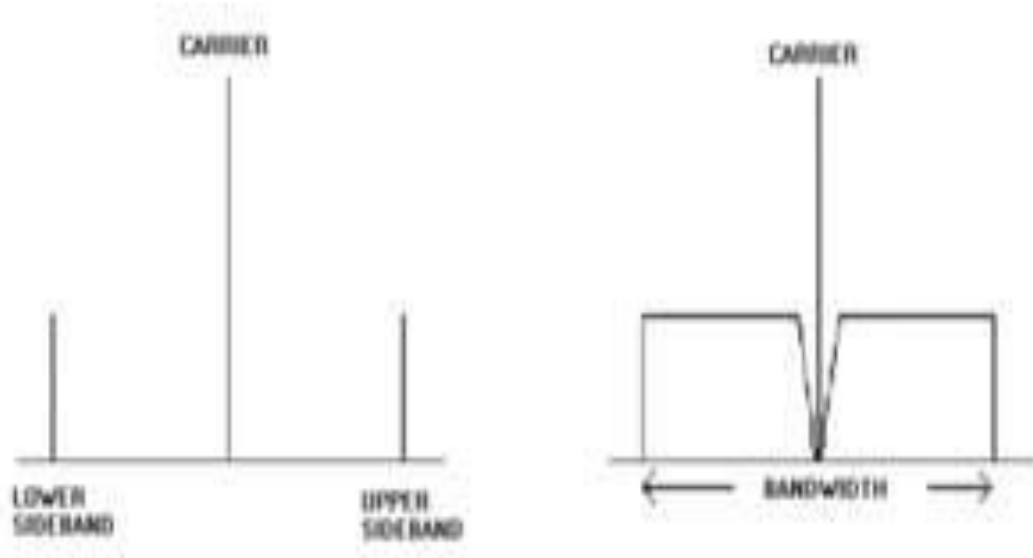
Figure 4.5

Example 4.1   *A 10-MHz sinusoidal carrier wave of amplitude 10 mV is modulated by a 5 kHz sinusoidal audio signal wave of amplitude 6 mV. Find the frequency components of the resultant modulated wave and their amplitudes.*

Solution. Here, fc = 10 MHz and fm = 5 kHz = 0.005 MHz. The modulated carrier contains the following frequencies :
1. original carrier wave of frequency fc = **10 MHz**
2. USF of frequency = 10 + 0.005 = **10.005 MHz**
3. LSF of frequency = 10 – 0.005 = **9.995 MHz**

Here,

$$\mathbf{m} = \frac{\text{maximum value of signal wave}}{\text{maximum value of carrier wave}} \times 100 = \frac{\text{Signal amplitude}}{\text{Carrier amplitude}} \times 100$$

$$= 6/10 = 0.6$$

Amplitude of $LSF = USF = mA/2 = 0.6 \times 10/2 =$ **3 mV**

It was assumed that the modulating signal was composed of one frequency component only. However, in a broadcasting station, the modulating signal is the human voice (or music) which contains waves with a frequency range of 20-4000 Hz. Each of these waves has its own *LSF* and *USF*. When combined together, they give rise to an upper-side **band** (*USB*) and a lower-side **band** (LSB). The *USB*, in fact, contains all sum components of the signal and carrier frequency whereas *LSB* contains their difference components.

The channel width (or bandwidth) is given by the difference between extreme frequencies i.e. between maximum frequency of *USB* and minimum frequency of *LSB*.
As seen, Channel width = 2 × maximum frequency of modulating signal = **2 × *fm (max)***

Example 4.2. *An audio signal given by 15 sin 2π (2000 t) amplitude-modulates a sinusoidal carrier wave 60 sin 2π (100,000) t.*
Determine :
(a) *modulation index,* *(b)* *percent modulation,* *(c)* *frequencies of signal and carrier,*
(d) *frequency spectrum of the modulated wave.*

Solution. Here, $B = 15$ and $A = 60$

> (*a*) *M.I. = B/A* = *15/60* = **0.25**
> (*b*) *m = M.I.* × 100 = 0.25 × 100= 25%
> (*c*) *fm* = **2000 Hz** — by inspection of the given equation
>   *fc* = **100,000 Hz** — by inspection of the given equation
> (*d*) The three frequencies present in the modulated *CW* are
>   (*i*) 100,000 Hz = **100 kHz**
>   (*ii*) 100,000 + 2000 = 102,000 Hz =**102 kHz**
>   (*iii*) 100,000 – 2000 = 98,000 Hz = **98 kHz**

Example 4.3. *A bandwidth of 15 MHz is available for AM transmission. If the maximum audio signal frequency used for modulating the carrier is not to exceed 15 kHz, how many stations can broadcast within this band simultaneously without interfering with each other?*

Solution. *BW* required by each station =2 *fm(max)* = 2 × 15 = 30 kHz
Hence, the number of station which can broadcast within this frequency band without interfering with one another is

$$= 15 \text{ MHz}/30\text{kHz} = \textbf{500}$$

Example 4.5. *In a broadcasting studio, a 1000 kHz carrier is modulated by an audio signal of frequency range, 100-5000 Hz. Find (i) width or frequency range of sidebands (ii) maximum and minimum frequencies of USB (iii) maximum and minimum frequencies of LSB and (iv) width of the channel.*

Solution: (*i*) Width of sideband = 5000 – 100   = **4900 Hz**
    (*ii*) Max. frequency of USB = 1000 + 5   = **1005 kHz**
        Min. frequency of *USB* = 1000 + 0.1   = **1000.1 kHz**
    (*iii*)Max. frequency of *LSB* = 1000 – 0.1   = **999.9 kHz**
        Min. frequency of *LSB* = 1000 – 5   = **995 kHz**
    (*iv*) Width of channel = 1005 – 995   = **10 kHz**

Example 4.6

A transmitter radiates 9kw without modulation and 10.125kw after modulation. Determine the depth of modulation.

Solution:

$$P_c = 9kw$$
$$P_{mod} = 10.125kw$$

Now $$P_{mod} = P_c (1 + m_a^2 /2)$$

$$(1 + m_a^2 / 2) = 10.125 / 9 = 1.125$$

$$m_a = 0.5$$

Advantages of AM over FM
1. Circuits for AM transmitter and receiver are simple and less expensive
2. AM signal can go so far in propagation
3. AM never suffers multi-path filtering
4. AM is easier to listen to with portable radios, not needing external wire antenna with local contents

Limitations of AM

1. It's power is not efficient
2. The bandwidth is equal to double of the highest audio frequency (the bandwidth required by AM is less (2 fm)
3. AM are sensitive to high level of noise
4. AM has poorer sound quality

APPLICATIONS OF AM

In amplitude modulation (AM), amplitude of carrier signal is varied in accordance with message signal. Amplitude modulation has many real world applications. Those are:

- AM is used for broadcasting in long wave or medium wave or short wave bands.
- The Very High Frequency (VHF) transmission is processed by AM. Radio communication uses VHF.
- A special type of AM is Quadrature Amplitude Modulation (QAM). It is used for data transmission from short range transmission to cellular communications.

Forms of Amplitude Modulation
As shown, **one carrier** and **two sidebands** are produced in AM generation. It is found that it is not necessary to transmit all these signals to enable the receiver to reconstruct the original signal.

Accordingly, we may attenuate or altogether remove the carrier or any one of the sidebands without affecting the communication process. The advantages would be

1. less transmitted power and **2.** less bandwidth required

The different suppressed component systems are :
(a)

FORMS OF AMPLITUDE MODULATION

- **DSB-SC**
  - It stands for double-sideband suppressed carrier system [Fig. below]. Here, carrier component is suppressed thereby saving enormous amount of power. Carrier signal contains 66.7 per cent of the total transmitted power for m = 1, Hence, power saving amounts to 66.7% at 100% modulation
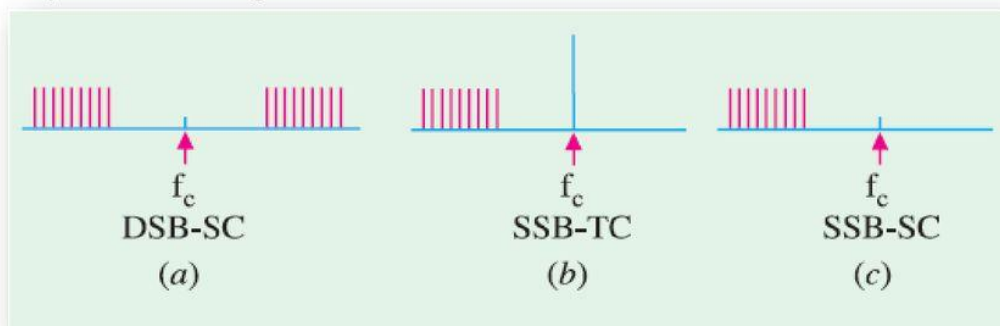
f_c
DSB-SC
(a)

f_c
SSB-TC
(b)

f_c
SSB-SC
(c)

Figure 4.6

20

(b) **SSB-TC**
In this case, one sideband is suppressed but the other sideband and carrier are transmitted. It is called *single sideband transmitted carrier* system. For *m* = 1, power saved is 1/6 of the total transmitted power

(c) **SSB-SC**
This is the **most dramatic suppression of all** because it suppresses one sideband and the carrier and transmits only the remaining sideband. In the standard or double-sideband full-carrier (*DSB.FC*) *AM*, carrier conveys **no information but contains maximum power**. Since the two sidebands are exact images of each other, they carry the same audio information. Hence, **all information is available in one sideband only**.
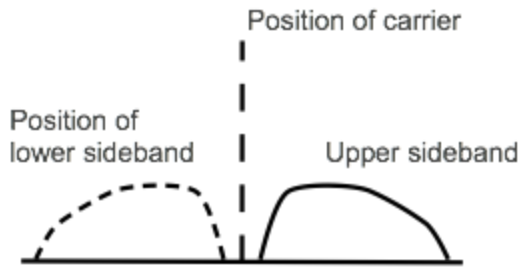
Figure 4.7

Obviously carrier is *superfluous* and one side band is *redundant.* Hence, one sideband and the carrier can be discarded with no loss of information. The result is *SSB* signal. The advantage of *SSB-SC* system are as follows :

1. Total saving of 83.3% in transmitted power (66.7% due to suppression of carrier wave and 16.6% due to suppression of one sideband). Hence, power is conserved in an SSB transmitter.

2. Bandwidth required is reduced by half *i.e*. 50%. Hence, twice as many channels can be multiplexed in a given frequency range.

3. The size of power supply required is very small. This fact assumes vital importance particularly in a spacecraft.

4. Since the SSB signal has narrower bandwidth, a narrower passband is permissible within the receiver, thereby limiting the noise pick up.

However, the main reason for wide spread use of *DSB-FC* (rather than *SSB-SC*) transmission in broadcasting is the relative simplicity of its modulating equipment

Methods of Amplitude Modulation

There are two methods of achieving amplitude modulation :

(*i*) Amplifier modulation,

(*ii*) Oscillator modulation.

.

Here, carrier and *AF* signal are fed to an amplifier and the result is an *AM* output

The modulation process takes place in the active device used in the amplifier.

Block Diagram of an AM Transmitter

Figure 4.6 shows the block diagram of a typical transmitter. The carrier wave is supplied by a crystal-controlled oscillator at the carrier frequency. It is followed by a tuned buffer amplifier and an *RF* output amplifier. The source of *AF* signal is a microphone. The audio signal is amplified by a low level audio amplifier and, finally, by a power amplifier. It is then combined with the carrier to produce a modulated carrier wave which is ultimately radiated out in the free space by the transmitter antenna as shown.
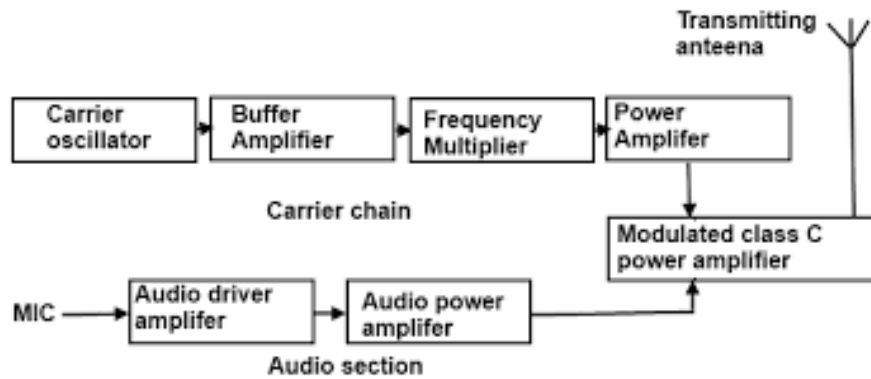
Figure 4.6

ASSIGNMENT 6

1. The antenna current of an AM transmitter is 8A when only carrier is sent out. It increases to 8.93A when the carrier is sinusoidally modulated. Find the percentage modulation?
2. A carrier wave of 600watts is subjected to 100% amplitude modulation. Determine
   (i)   Power in sideband   (ii) power of modulated wave

QUIZ 6

1. What are the consequences of overmodulation?
2. A 40kw carrier is to be modulated to a level of 100%.
   (i)   What is the carrier power after modulation
   (ii)   How much audio power is required if the efficiency of the modulated RF amplifier is 72%
3. Explain briefly the two methods of amplitude modulation (the amplifier modulation and oscillator modulation)

PRACTICAL SESSION

- Students are to experiment how AM is generated using the necessary laboratory equipment
- A report is to be submitted after

CHAPTER 5.0

FREQUENCY MODULATION

Frequency modulation (**FM**) is the encoding of information in a carrier wave by varying the instantaneous frequency of the wave. The term and technology is used in both **telecommunications and signal processing**.

In **analog** frequency modulation, such as FM radio broadcasting of an audio signal representing voice or music, the instantaneous **frequency deviation**, the difference between the frequency of the carrier and its center frequency, is proportional to the modulating signal.

In **frequency modulation,** the frequency of the carrier signal is varied in proportional to (in accordance with) the Amplitude of the input modulating signal. The input is a single tone sine wave. The carrier and the FM waveforms also are shown in the following figure.
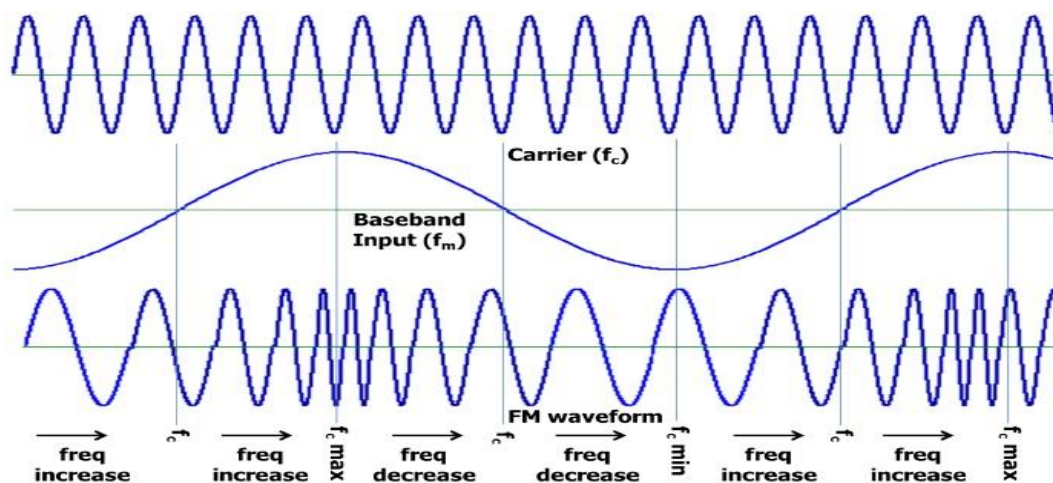

Figure 5.1: Frequency Modulation waveform

As the name shows, in this modulation, it is only the **frequency of the carrier which is changed and not its amplitude.** The **amount of change** in frequency is determined by the **amplitude** of the modulating signal whereas **rate of change** is determined by the **frequency** of the modulating signal. As shown in the figure above, in an *FM* carrier, information (or intelligence) is carried as variations in its frequency. As seen, frequency of the modulated carrier increases as the signal amplitude increases but decreases as the signal amplitude decreases. It is at its highest frequency (point *H*) when the signal amplitude is at its maximum positive value and is at its lowest frequency (point *L*) when signal amplitude has maximum negative value. When signal amplitude is zero, the carrier frequency is at its normal frequency *f*0 (also called *resting or centre* frequency.)**.** This **louder** signal causes **greater** frequency change in modulated carrier as indicated by *increased bunching and spreading* of the waves as compared with relatively weaker signal. The **rate** at which frequency shift takes place depends on the signal frequency if

the modulating signal is 1 kHz, then the modulated carrier will swing between its maximum frequency and lowest frequency 1000 times per second. If *fm* = 2 kHz, the rate of frequency swing would be twice as fast :

In short, we have established two important points about the nature of frequency modulation:

(i)   The **amount** of frequency deviation (or shift or variation) depends on the **amplitude** (loudness) of the audio signal. *Louder the sound, greater the frequency deviation and vice-versa.* However, for the purposes of *FM* broadcasts, it has been internationally agreed to restrict maximum deviation to **75 kHz** on each side of the centre frequency for sounds of maximum loudness. Sounds of lesser loudness are permitted *proportionately less frequency deviation.*

(*ii*) The **rate** of frequency deviation depends on the **signal frequency**.

Frequency Deviation and Carrier Swing

The frequency of an *FM* transmitterwithout signal input is called the **resting frequency** or **centre frequency** (*f0*) and is the allotted frequency of the transmitter. In simple words, it is the carrier frequency on which a station is allowed to broadcast.When the signal is applied, the carrier frequency deviates up and down from its resting value *f0*. This change or shift either above or below the resting frequency is called frequency deviation (Δf) The total variation in frequency from the lowest to the highest is called carrier swing (CS).

Obviously, **carrier swing** = 2 × frequency deviation of $CS = 2 \times \Delta f$

A maximum frequency deviation of 75 kHz is allowed for commercial *FM* broadcast stations inthe 88 to 168 MHz *VHF* band. Hence, *FM* channel width is 275 = 150 kHz. Allowing a 25 kHz guard band on either side, the channel width becomes = 2(75 + 25) = 200 kHz This guard band is meant to prevent interference between adjacent channels. However, a maximum frequency deviation of 25 kHz is allowed in the sound portion of the TV broadcast.

In *FM*, the highest audio frequency transmitted is 15 kHz. Consider an FM carrier of resting frequency 100 MHz. Since **(Δf)*max*** = 75 kHz, the carrier frequency can swing from the lowest value of 99.925 MHz to the highest value of 100.075 MHz. Of course, deviations lesser than 75 kHz corresponding to relatively softer sounds are always permissible.

Modulation Index

It is given by the ratio $mf = \dfrac{\text{frequency deviation}}{\text{modulation frequency}} = \dfrac{\Delta f}{fm}$

Unlike amplitude modulation, this **modulation index can be greater than unity**. By knowing the value of *mf*, we can calculate the number of significant sidebands and the bandwidth of the *FM* signal.

Deviation Ratio

It is the worst-case modulation index in which maximum permitted frequency deviation and maximum permitted audio frequency are used

$$\therefore deviation\ ratio = \frac{(\Delta f)}{fm(max)}$$

Now, for *FM* broadcast stations, (*Δf*)*max* = 75 kHz and maximum permitted frequency of modulating audio signal is 15 kHz

$$\therefore deviation\ ratio = \frac{75KHZ}{15KHZ} = 5$$

For sound portion of commercial TV deviation ratio $= \frac{25KHZ}{15KHZ} = 1.67$

Percent Modulation

When applied to *FM*, this term has slightly different meaning than when applied to *AM*. In *FM*, it is given by the ratio of actual frequency deviation to the maximum allowed frequency deviation

$$m = \frac{(\Delta f)actual}{(\Delta f)max}$$

Obviously, 100% modulation corresponds to the case when actual deviation equals the maximum allowable frequency deviation. If, in some case, actual deviation is 50 kHz, then

$$m = \frac{50}{75} = 0.667 = 66.7\%$$

Value of *m* = 0 corresponds to zero deviation *i.e.* unmodulated carrier wave. It is seen from the above equation that $m \propto (\Delta f)actual$. It means that when **frequency deviation (*i.e.* signal loudness) is doubled, modulation is doubled.**

Example 5.1. *What is the modulation index of an FM carrier having a carrier swing of 100 kHz and a modulating signal of 5 kHz ?*

Solution. *CS* = 2 ×Δ *f*

$$\therefore \Delta f = \frac{CS}{2} = \frac{100}{2} = 50Khz, \quad \therefore mf = \frac{(\Delta f)}{fm} = \frac{50}{5} = 10$$

Example. 5.2. *An FM transmission has a frequency deviation of 18.75 kHz. Calculate percent modulation if it is broadcast*

(i)   *in the 88-108 MHz band **(ii)** as a portion of a TV broadcast*

Solution. *(i)* For this transmission band,

$$(\Delta f)max = 75 \text{ kHz} \qquad\qquad \therefore\ m = \frac{18.75}{75}\ \text{X}\ 100\ = 25\%$$

(ii) In this case, $(\Delta f)max = 25 \text{ kHz} \ \therefore\ m = \frac{18.75}{25}\ \text{X}\ 100\ = 75\%$

Example 5.3. *An FM signal has a resting frequency of 105 MHz and highest frequency of 105.03 MHz when modulated by a signal of frequency 5 kHz. Determine*

*(i) frequency deviation, **(ii)** carrier swing, **(iii)** modulation index,*
*(iv) percent modulation, **(v)** lowest frequency reached by the FM wave.*

Solution. *(i)* Δf = 105.03 – 105= 0.03 MHz = **30 kHz**

    (ii) $CS = 2 \times \Delta f = 2 \times 30 = $ **60 kHz**

    (iii) $mf = \dfrac{30}{5}\ = 6$

    (iv) $mf = \dfrac{30}{5}\ \text{X}\ 100\ = 60\%$

    (v)   ***lowest frequency*** = 105 – 0.03 = **104.97 kHz**

FM Sidebands

In *FM*, when a carrier is modulated, a number of sidebands are formed.* Though theoretically their number is infinite, their strength becomes negligible after a few sidebands. They lie on both sides of the centre frequency spaced fm apart. Sidebands at equal distances from $f0$ have equal amplitudes. If $f0$ is the centre frequency and fm the frequency of the modulating signal, then *FM* carrier contains the following frequencies :

(i) $f0$ *(ii)* $f0 \pm fm$ *(iii)* $f0 \pm 2\ fm$ *(iv)* $f0 \pm 3\ fm$ and so on

The bandwidth occupied by the spectrum is $BW = 2nfm$ where n is the highest order of the significant sideband.
Another approximate expression for spectrum bandwidth is $BW = 2\ (1 + mf)fm$

$$Now,\ mf = \frac{(\Delta f)}{fm}, \quad \text{hence } BW = 2\ (\Delta f + fm)$$

This expression is based on the assumption that sidebands having amplitudes less than 5% of the unmodulated carrier wave are negligble or when *mf* is at least 6.

Modulation Index and Number of Sidebands

It is found that the number of sidebands
1. depends *directly* on the amplitude of the modulating signal,
2. depends *inversely* on the frequency of the modulating signal.

Since frequency deviation is directly related to the amplitude of the modulating signal, the above two factors can be combined in one factor called **modulation index**.

Hence, number of sidebands depends on $mf = \Delta f / fm$

Obviously, the number of pairs of sidebands

(***i***) **increases** as frequency deviation (or amplitude of modulating signal) **increases**.
(***ii***) **increases** as the modulating signal frequency **decreases**.

Example 5.4. *A 5 kHz audio signal is used to frequency-modulate a 100 MHz carrier causing a frequency deviation of 20 kHz. Determine*
  (***i***)   *modulation index* (***ii***) *bandwidth of the FM signal*.

Solution. (*i*) $mf = \dfrac{(\Delta f)}{fm} = \dfrac{20}{5}$ - 4

When mf = 4, and n= 7,

$$BW = 14fm = 14 \times 5 = \textbf{70 kHz}$$

Example 5.5. *In an FM circuit, the modulation index is 10 and the higest modulation frequency is 20 kHz. What is the approximate bandwidth of the resultant FM signal ?*

Solution. Since the value of *mf* is more than 6, we will use the expression
$$BW = 2\,(\Delta f + fm)$$

$$Now \; mf = \dfrac{(\Delta f)}{fm} \quad or \; 10 = \dfrac{(\Delta f)}{20}, \quad \therefore \quad \Delta f = 200 \text{ kHz}$$

$$\therefore \quad BW = 2\,(200 + 20) = \textbf{440 kHz}$$

Comparison Between AM and FM

Frequency modulation *(FM)* has the following **advantages** as compared to amplitude modulation **(AM)** :

1. All transmitted power in *FM* is useful whereas in *AM* most of it is in carrier which *serves no useful purpose*.

2. It has high signal-to-noise (S/N) ratio. It is due to two reasons : firstly, there happens to be less noise at *VHF* band and secondly, *FM* receivers are fitted with amplitude limiters which remove amplitude variations caused by noise.

3. Due to 'guard-band' there is hardly any adjacent-channel interference.

4. Since only transmitter *frequency* is modulated in *FM*, only fraction of a watt of audio power is required to produce 100% modulation as compared to high power required in *AM*.

However, FM has the following **disadvantages** :

1. It requires **much wider channel** —almost 7 to 15 times as large as needed by AM.

2. It requires complex and expensive transmitting and receiving equipment.

3. Since FM reception is limited to only line of sight, area of reception for FM is much smaller than for *AM*.

APPLICATIONS OF FM

There are four major areas of application for *FM* transmission:

1. First use is in **FM** broadcast band 88-108 MHz with 200 kHz channels in which commercial *FM* stations broadcast programmes to their listeners.

2. Second use is in TV. Though video signal is amplitude-modulated, sound is transmitted by a separate transmitter which is frequency-modulated.

3. Third use is in the mobile or emergency services which transmit voice frequencies (20-4000 Hz) only.

4. Fourth use is in the amateur bands where again only voice frequencies are transmitted.

ASSIGNMENT 7

1. Differentiate between AM modulation index and FM modulation index
2. A 5 kHz audio signal is used to frequency-modulate a 100 MHz carrier causing a frequency deviation of 20 kHz. Determine
   (i) modulation index **(ii)** bandwidth of the FM signal.

QUIZ 7

1. Compare AM and FM

2. An FM signal has a resting frequency of 95 MHz and highest frequency of 95.03 MHz when modulated by a signal of frequency 5 kHz. Determine
   (i) frequency deviation, **(ii)** carrier swing, **(iii)** modulation index,
   (iv) percent modulation, **(v)** lowest frequency reached by the FM wave.

CHAPTER 6.0

Demodulation or Detection of Am & FM Signals

When the *RF* modulated waves, radiated out from the transmitter antenna, after travelling through space, strike the receiving aerials, they induce very weak *RF* currents and voltages in them. If these high-frequency currents are passed through headphones or loudspeakers, they produce no effect on them because all such sound-producing devices are unable to respond to such high frequencies due to large inertia of their vibrating discs etc. Neither will such *RF* currents produce any effect on human ear because their frequencies are much beyond the audible frequencies (20 to 20,000 approximately). Hence, it is necessary to demodulate them first in order that the sound-producing devices may be actuated by audio-frequency current similar to that used for modulating the carrier wave at the broadcasting station.

This process of *recovering AF signal from the modulated carrier wave is known as demodulation or detection.*

The demodulation of an *AM* wave involves two operations :

(*i*) rectification of the modulated wave and
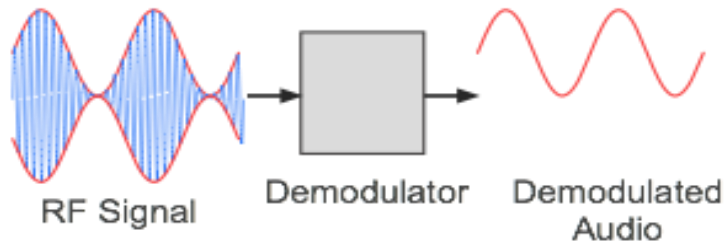(*ii*) elimination of the *RF* component of the modulated wave.

Figure 6.1: **AM Demodulation principle**

Essentials of AM Detection

For recovering the *AF* waveform from modulated wave (a mixture of *AF* wave and *RF* carrier), it is essential to find some way of reducing (or better, eliminating) one half of the modulated wave. The result of this elimination (or rectification) would be that the average value of the wave would not be zero because, now, the impulse would be all in one direction as shown in Fig. 66.23. If this new wave is now passed through a headphone shunted by a suitable capacitor, then *AF* wave will pass through the headphone whereas the *RF* wave will be by-passed by the capacitor (because the high inductance of magnet coils of the headphones will offer tremendous impendance to *RF* currents). Hence, two will become separated.

Amplitude modulation may be demodulated through the following means

- Diode detectors
- Transistor detectors

Diode Detector (Envelope Demodulator) for AM Signals

Diode detection is also known as **envelope-detection** or **linear detection**. In appearance, it looks like an ordinary half-wave rectifier circuit with capacitor input as shown in Fig. 6.1. It is called **envelope detection** because it recovers the *AF* signal envelope from the composite signal. Similarly, diode detector is called **linear detector** because its output **is *proportional to the voltage of the input signal*.**
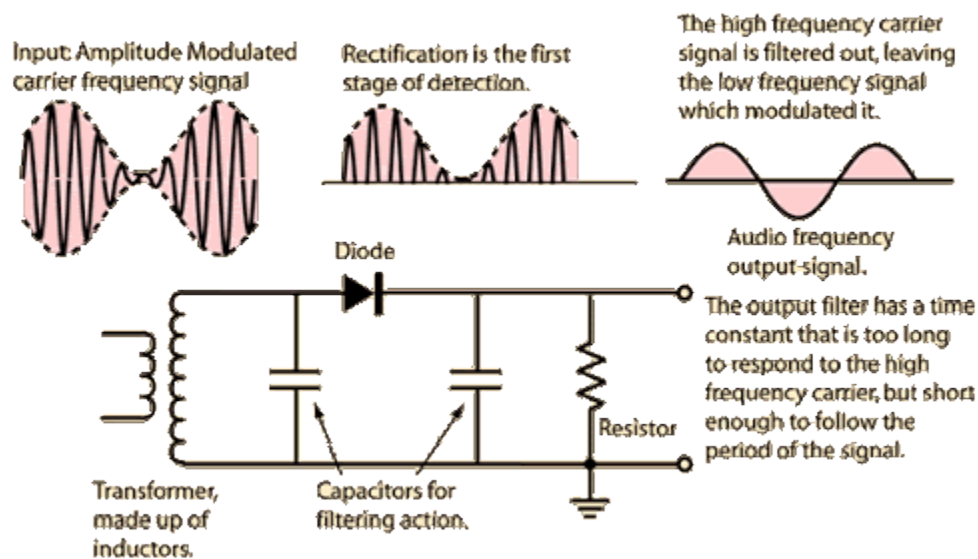

Figure 6.2: **Diode Detector circuit**

Advantages

Diode detectors are extensively used in AM broadcast receivers because they have the following advantages :

1. They can handle comparatively large input signals;
2. They can be operated as linear or power detectors;
3. They rectify with negligible distortion and, hence, have good linearity;
4. They are well-adopted for use in simple automatic-gain control circuits.

Disadvantages

However, the disadvantages are that

1. They do not have the ability to amplify the rectified signal by themselves as is done by a transistor detector. However, it is not a very serious drawback since signal amplification can be affected both before and after rectification;
2. while conducting, the diode consumes some power which reduces the Q of its tuned circuit as well as its gain and selectivity.

FM Detection

As discussed earlier, an *FM* carrier signal contains information (or intelligence we wish to convey) in the form of frequency variations above and below the centre frequency of the carrier. For recovering the information, we must first convert the *FM* signal in such a way that it appears as a modulated *RF* voltage across the diode. A simple method of converting frequency variations into voltage variations is to make use of the principle that reactance (of coil or capacitor) varies with frequency. When an *FM* signal is applied to an inductor, the current flowing through it varies in amplitude according to the changes in frequency of the applied signal. Now, changes in frequency of the *FM* signal depend on the amplitude of the modulating *AF* signal. Hence, the current in the inductor varies as per the amplitude of the original modulating signal. In this way, frequency changes in *FM* signal are converted into amplitude changes in current. These changes in current when passed through a resistor produce corresponding changes in voltage. Hence, we find that, ultimately, frequency variations in *FM* signal are converted into voltage changes. Also, there exists a linear relation between the two – something essential for distortion-less demodulation.
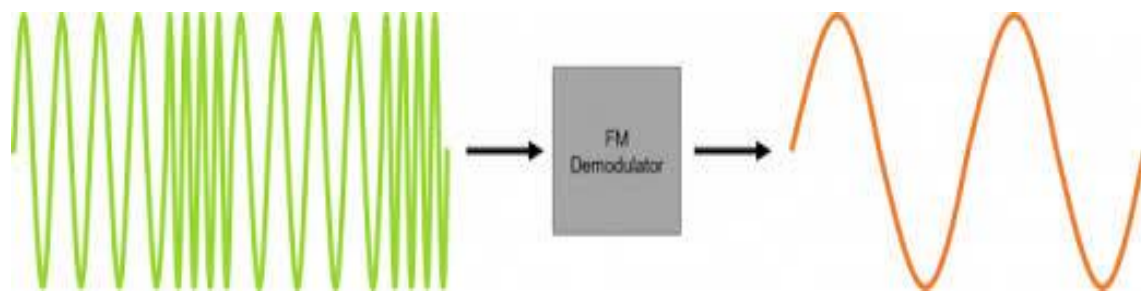

Figure 6.3: **FM Detector wzvefrom, seperating the AF signal**

FM demodulation may be carried out with the help of (*i*) ratio detector and (*ii*) quadrature detector.

Demodulation of an *FM* wave involves three operations (***i***) conversion of frequency changes produced by modulating signal into corresponding amplitude changes, (***ii***) rectification of the modulating signal and (***iii***) elimination of *RF* component of the modulated wave.

ASSIGNMENT 8

1. List the different methods of detecting AM and FM signals
2. Explain one method each form the list above
3. Sketch the circuit of the diode detector and explain

QUIZ 8

1. Define demodulation
2. State five differences between modulation and demodulation

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CHAPTER 7.0

Superhetrodyne AM Receiver

What Heterodyning is:

To heterodyne means to mix two frequencies together so as to produce a beat frequency, namely the difference between the two. Amplitude modulation is a heterodyne process: the information signal is mixed with the carrier to produce the side-bands. The side-bands occur at precisely the sum and difference frequencies of the carrier and information. These are beat frequencies (normally the beat frequency is associated with the lower side-band, the difference between the two).

What Superheterodyning is:

When you use the lower side-band (the difference between the two frequencies), you are superheterodyning. Strictly speaking, the term superheterodyne refers to creating a beat frequency that is lower than the original signal. Although we have used the example of amplitude modulation side-bands as an example, we are not talking about encoding information for transmission. What superheterodying does is to purposely mix in another frequency in the receiver, so as to reduce the signal frequency prior to processing.

It is also referred to as superhet and is extensively used in modern AM receivers. Its block diagram with signals is shown in Fig. 7.1. The operation of this receiver is as under :

(*i*) Let us assume that the incoming signal frequency is 1500 kHz. It is first amplified by the *R.F.* amplifier.
(*ii*) Next, it enters a mixer circuit which is so designed that it can conveniently combine two radio frequencies—one fed into it by the *R.F.* amplifier and the other by a local oscillator.
(*iii*) The local oscillator is an *RF* oscillator whose frequency of oscillation can be controlled by varying the capacitance of its capacitor. In fact, the tuning capacitor of the oscillator is ganged with the capacitor of the input circuit so that the difference in the frequency of the selected signal and oscillator frequency is always constant. Usually, the difference is maintained at 455 kHz. If signal frequency is 1,500 kHz, then oscillator frequency can be either 1,955 or 1,045 kHz. Let us suppose that it is 1,955 kHz. In fact, local oscillator frequency is always higher than the frequency of the incoming signal.
(*iv*) When two alternating currents of these two different frequencies are combined in the mixer transistors, then phenomenon of beats is produced. In the present case, the beat frequency is 1955-1500 = 455 kHz. Since this frequency is lower than the signal frequency but still above the range of audio frequencies, it is called *intermediate frequency* (*IF*)**.**
(*v*) The 455 kHz output of the mixer is then passed on to the *IF* amplifier which is *fixed tuned*

to 455 kHz frequency. In practice, one or more stages of *IF* amplification may be used.
(**vi**) The output of *IF* amplifier is demodulated by a detector which provides the audio signal.
(**vii**) This audio signal is amplified by the audio-frequency (*AF*) amplifier whose output is fed to a loud-speaker which reproduces the original sound
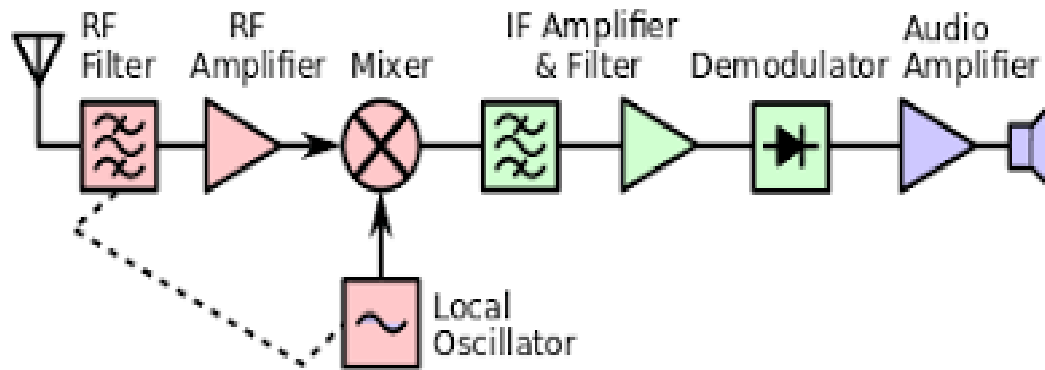

Figure 7.1: Block diagram of superheterodyne radio receiver

Signals enter the receiver from the antenna and are applied to the RF amplifier where they are tuned to remove the image signal and also reduce the general level of unwanted signals on other frequencies that are not required.

There are some key circuit blocks within the RF design of the basic superheterodyne receiver. Although more complicated receivers can be made, the basic RF circuit design is widely used – further blocks can add improved performance or additional functionality and their operation within the whole receiver is normally easy to determine once the basic block diagram is understood.

Below are the explanations of the block diagram of a superheterodyne radio receiver:

- ***RF tuning & amplification:*** This RF stage within the overall block diagram for the receiver provides initial tuning to remove the image signal. It also provides some amplification. There are many different approaches used within the RF circuit design for this block dependent its application.
  The RF circuit design presents some challenges. Low cost broadcast radios may have an amplifying mixer circuit that gives some RF amplification. HF radios may not want too much RF gain because some of the very strong signals received could overload later stages. The RF design may incorporate some amplification as well as RF attenuation to overcome this issue. Radios for VHF and above will tend to use more gain to have a sufficiently low noise figure to receive the signal.

  If noise performance for the receiver is important, then this stage will be designed for optimum noise performance. This RF amplifier circuit block will also increase the signal level so that the noise introduced by later stages is at a lower level in comparison to the wanted signal.

- ***Local oscillator:*** Like other areas of the RF circuit design, the local oscillator circuit block within the superhet radio can take a variety of forms.
  Early receivers used free running local oscillators. There was a considerable degree of RF circuit design expertise used with these oscillators in high performance superhet radios to ensure the lowest possible drift. High Q coils, low drift circuit configurations, heat management (because heat causes drift), etc . .

  Today most receivers use one or more of a variety of forms frequency synthesizers. The most common approach in the RF circuit design is to use a phase locked loop approach. Single and multi-loop synthesizers are used. Direct digital synthesizers are also being used increasingly. Whatever form of synthesizer is used in the RF design, they provide much greater levels of stability and enable frequencies to be programmed digitally in a variety of ways, normally using some form of microcontroller or microprocessor system.

- ***Mixer:*** The mixer can be one of the key elements within the overall RF design of the receiver. Ensuring that the mixer performance matches that of the rest of the radio is particularly important.
  Both the local oscillator and incoming signal enter this block within the superheterodyne receiver. The wanted signal is converted to the intermediate frequency.

  The actual implementation requires that the minimum number of spurious signals are generated. In some very low cost broadcast receivers, self oscillating mixers that provide RF amplification from a single transistor may be used, these do not offer high performance. For a high performance radio used for two way radio communications an the like, much better performance is required. To achieve this mixer circuits such as balanced mixers, double balanced mixers, and the like may be seen.

- ***IF amplifier & filter:*** This superheterodyne receiver block provides the majority of gain and selectivity. Often comparatively little gain will be provided int he previous blocks of the RF circuit design of the radio. The IF stages are where the main gain is provided. Being fixed in frequency, it is much easier to achieve high levels of gain and overall performance.
  Originally the IF stage might have included a number of different transistors, FETs or thermionic valves / vacuum tubes, but nowadays it is possible to obtain integrated circuits that contain a complete IF strip.

  This circuit block of the radio also provides the adjacent channel selectivity. High performance filters like crystal filters may be used, although LC or ceramic filters may be used within domestic radios. The type of filter will depend upon the radio RF design and its application.

- ***Demodulator:*** The superheterodyne receiver block diagram only shows one demodulator, but in reality many radio RF designs may have one or more demodulators dependent upon the type of signals being receiver.

Even many broadcast radios will have AM and FM, but professional radios used for monitoring and two way radio communications may require a larger variety in some instances. Having a variety of demodulators will enable many different signal modes to be received and increase the capability of the radio.

- *Automatic Gain Control, AGC:*   An automatic gain control is incorporated into most superhet radio block diagrams. The function of this circuit block is to reduce the gain for strong signals so that the audio level is maintained for amplitude sensitive forms of modulation, and also to prevent overloading.
  Although the basic concept is the same through all radio RF circuit designs, there are some variations in the implementation. Some of the key variations are the time constant of the AGC system. For AM and the like a relatively slow time constant is acceptable. For SSB, a shorter time constant is needed so that the envelope of the SSB signal is followed.

  There are also variations in the way the AGC voltage is derived, and where it is applied. Often it is applied to the IF circuit blocks first and then to the RF circuit block. In this way the best signal to noise ratio is preserved.

- *Audio amplifier:*   Once demodulated, the recovered audio is applied to an audio amplifier block to be amplified to the required level for loudspeakers or headphones. Alternatively the recovered modulation may be used for other applications whereupon it is processed in the required way by a specific circuit block.
  In many ways, this circuit block within the superheterodyne radio is the most straightforward. For many applications, the audio amplifier will involved some straightforward electronic circuit design, especially if the audio is applied to simple headphones or a loudspeaker. For two way radio communication applications, the audio bandwidth may need to be limited to the "telecommunications" bandwidth of about 300 Hz to 3.3 kHz. Audio filters could be employed as well.

  For applications requiring a higher quality output, more thought may need to be applied during the electronic circuit design to achieving high fidelity performance.

  Whatever the radio, there can be different requirements for this circuit block.

Advantages of Superheterodyne Receiver

So why are so many receivers using the superheterodyne method? There are three main advantages, depending on the application used for:

- It reduces the signal from very high frequency sources where ordinary components wouldn't work (like in a radar receiver).
- It allows many components to operate at a fixed frequency (IF section) and therefore they can be optimized or made more inexpensively.
- It can be used to improve signal isolation by arithmetic selectivity Reduction in frequency

At very extremely high frequencies, many ordinary components cease to function. Although we see many computer systems that work at previously unattainable frequencies like 166 MHz, you

certainly never see any system that works at radar frequencies like 10 GHz (try that Intel!). There are many physical reasons for this, but suffice it to say, it can't be done (yet). So the designer of a radar interceptor (fuzz-buster, et al.) is faced with a daunting circumstance unless he/she can use a superheterodyne receiver to knock down the frequency to an IF value. It is in fact, the local oscillator (a operating at radar frequencies) of the superheterodyne radar receiver that makes your radar detector detectable by the police (in VA for example, where the use of radar detectors are illegal).

Optimization of Components

It is a typical engineering dilemma: how to make components that have outstanding performance, but can also cover a wide range of frequencies. Again, the details aren't important, but the problem is very real. A possible solution to this, is to make as much of the receiver as possible always work at the same frequency (the IF). This is accomplished by using the superheterodyne method. The majority of components can be optimized to work at the IF without any requirements to cover a wide range of frequencies.

Arithmetic Selectivity

The ability to isolate signals, or reject unwanted ones, is a function of the receiver bandwidth. For example, the band-pass filter in the tuner is what isolates the desired signal from the adjacent ones. In real life, there are frequently sources that can interfere with your signal. The FCC makes frequency assignments that generally prevent this. Depending on the application, you might have a need for very narrow signal isolation. If the performance of your band-pass filter isn't sufficient to accomplish this, the performance can be improve by superheterodyning.

Frequently, the receiver bandwidth is some fraction of the carrier frequency. If your receiver has a bandwidth of 2 % and you are tuned to 850 kHz, then only signals within the range from 2 % above and below are passed. In this case, that would be from 833 to 867 kHz.

Arithmetic selectivity takes that fraction and applies it to the reduced frequency (the IF). For the fixed IF of 452 kHz, that means signals which are superheterodyned to the range of 443 to 461 kHz will pass. Taking this range back up into the carrier band, only carrier frequencies in the range of 841 to 859 kHz will pass. If this is confusing, recall that the local oscillator is set to reduce the 850 kHz to 452 kHz (i.e. must be set at 398 kHz). Thus, the 850 kHz is superheterodyned to 452 kHz. Any adjacent signals are also superheterodyned but remain the same above or below the original signal. An example might clear this up:

Suppose there is an interfering signal at 863 kHz while you are tuned to 850 kHz. A conventional 2 % receiver will pass 833 to 867 kHz and so the interfering signal also passes. The superheterodyne receiver mixes both signals with 398 kHz to produce the desired signal at 452 kHz and the interference at 465 kHz. At 2 %, the IF section only passes 443 to 461 kHz, and therefore the interference is now suppressed. We say that the superheterodyne receiver is more selective. With a little thought, the reason is simple: it operates at a smaller frequency, so the 2 % actually involves a smaller range. That is why it is called arithmetic selectivity. Bandwidths that

are expressed as a percentage are smaller when the center frequency is smaller (the same way that 2 % of $10 is less than 2 % of $10,000,000).

Whether or not, you need to take advantage of arithmetic selectivity depends on the application. If you have no problems with interference at your current bandwidth and/or it is not difficult or expensive to reduce the bandwidth of your receiver, then you don't need it. However, in cases where selectivity is important or the frequency is very high (like radar) then superheterodyning can greatly improve performance.

Summary

- Superheterodyne receivers reduce the signal frequency by mixing in a signal from a local oscillator to produce the intermediate frequency (IF).
- Superheterodyne receivers have better performance because the components can be optimized to work a single intermediate frequency, and can take advantage of arithmetic selectivity.

FM Receiver

These receivers operate in the *VHF* band of 88-108 MHz and have an *IF* of 10.7 MHz with a bandpass of 2×75 kHz $= 150$ kHz Like *AM* receivers, such receivers also have a frequency converter and *IF* amplifiers.

The function of the **limiter** is to remove all *amplitude variations* (caused by noise) from *IF* signal which might have crept into the *FM* signal. This removal of amplitude variations is necessary for distortionless demodulation. **Limiter is a sort of clipping circuit**.

The de-emphasis network *reduces the amplitude of high frequencies in the audio signal* which was earlier increased by the pre-emphasis network at the transmitting station. It serves to reestablish the **tonal balance** of the speech or music etc. lost in the pre-emphasis. Without it, the sound signal would have a heavy treble effect. It is, in fact, a low-pass filter having a time constant of 75 μs.

Comparison Between AM and FM

Frequency modulation *(FM)* has the following advantages as compared to amplitude modulation *(AM)* :

1. All transmitted power in *FM* is useful whereas in *AM* most of it is in carrier which *serves no useful purpose*.
2. It has high signal-to-noise (S/N) ratio. It is due to two reasons : firstly, there happens to be less noise at *VHF* band and secondly, *FM* receivers are fitted with amplitude limiters which remove amplitude variations caused by noise.
3. Due to 'guard-band' there is hardly any adjacent-channel interference.

4. Since only transmitter *frequency* is modulated in *FM*, only fraction of a watt of audio power is required to produce 100% modulation as compared to high power required in *AM*.

However, FM has the following **disadvantages** :

1. It requires **much wider channel** —almost 7 to 15 times as large as needed by AM.
2. It requires complex and expensive transmitting and receiving equipment.
3. Since FM reception is limited to only line of sight, area of reception for FM is much smaller   than for *AM*.

ASSIGNMENT 9 & 10

    1. What do you understand by superheterodyne principle?
    2. What are the functions of the following:
       (a) Mixer
       (b) Intermediate Frequency
       (c) Local Oscillator

    3. State three reasons why so many receivers are using superheterodyne method

QUIZES 9 & 10

    1. Sketch the block diagram of a superheterodyne receiver
    2. Explain the working principle of the superheterodyne receiver

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