



SRI G.C.S.R COLLEGE

(Affiliated to Dr. B. R. Ambedkar University, Srikakulam)

GMR Nagar, Rajam – 532127, Srikakulam (Dist.), A.P

T: +91 8941-251336, M: +91 89785 23866, F: +91 8941-251591,

www.srigcsrcollege.org



Department of ELECTRONICS

II B.Sc. III – Semester Electronics Paper-3

Analog Circuits & Communications

(Study Material)

Name of the Student : _____

Roll Number : _____

Group : _____

Year/ Semester : _____

Prepared by:

K. Venugopala Rao M.Sc.(Tech), M. Tech

Lecturer in Electronics

SRI GCSR COLLEGE, GMR Nagar, Rajam

venugopalarao.k@sgsrc.edu.in, +91-9603803168

Department of ELECTRONICS



Dr. B. R. AMBEDKAR UNIVERSITY-SRIKAKULAM
B.Sc. ELECTRONICS SYLLABUS
STRUCTURE UNDER CHOICE BASED CREDITS SYSTEM
REVIEWED SYLLABUS w.e.f. 2021-22
II B.Sc. Semester – III
Paper – III : Analog Circuits & Communication

UNIT – I (12hrs): OPERATIONAL AMPLIFIERS

Definition, Characteristics of Op-Amp, Block diagram of op-amp, inverting, noninverting, virtual ground, summing amplifier, subtractor, voltage follower, op-amp parameters, integrator, differentiator, Logarithmic amplifier with Diode.

UNIT- II (12hrs): OP-AMP CIRCUITS

voltage regulator, comparator, sine wave generator, square wave generator, Active filters (Basics)- low pass, high pass filters. IC-555 –functional block diagram and mention its applications

UNIT –III (12Hrs): AMPLITUDE MODULATION

Need for modulation, amplitude modulation-frequency spectrum of AM wave, representation of AM, power relations in the AM wave. Generation of AM- Balanced modulator. Detection of AM signals – Diode detector.

UNIT-IV (12hrs): FREQUENCY MODULATION

Theory of FM, Frequency deviation and carrier swing, modulation index, deviation ratio, percent modulation. Mathematical representation of FM, frequency spectrum and bandwidth of FM waves, Generation of FM signals – Reactance modulator. Detection of FM waves – Ratio detector.

UNIT-V (12hrs): RADIO BROADCASTING AND RECEPTION

Spectrum of electromagnetic waves, Radio broadcasting and reception, Transmitter, AM receiver- block diagram, Super heterodyne receiver. FM receiver- Block diagram

TEXT BOOKS:

1. Op Amp and Linear Integrated Circuits By Ramakant Gaykwad
2. Linear Integrated Circuits By Roy Chowdary
3. Unified Electronics Vol II – J.P. Agarwal and Amit Agarwal.
4. Electronic Communications - George Kennedy
5. Principles of communication system –Herbert Taub & D.L.Schillin

REFERENCE BOOKS :

1. Jacob Millan-Micro Electronics, McGraw Hill.
2. Mithal G K, Electronic Devices and Circuits Thana Publishers.
3. Electronic Communications – Roody & Colen
4. Communication Systems – Hayken--- 4th Edition
5. Modern digital and analog communication system –B.P. Lathi

UNIT – I**OPERATIONAL AMPLIFIERS****1.1 INTRODUCTION**

An operational amplifier (op-amp) is an integrated circuit (IC) that operates as a voltage amplifier. An op-amp has a differential input. That is, it has two inputs of opposite polarity. An op-amp has a single output and a very high gain, which means that the output signal is much higher than input signal. The op-amp symbol is shown in fig 1.1

These amplifiers are called "operation" amplifiers because they were initially designed as an effective device for performing arithmetic operations in an analog circuit. The op-amp has many other applications in signal processing, measurement, and instrumentation.

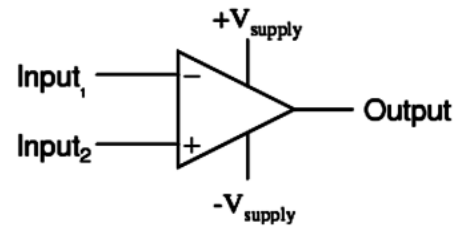


Fig 1.1: Op-amp symbol

An Operational Amplifier is basically a three-terminal device which consists of two high impedance inputs, one called the Inverting Input, marked with a negative or “minus” sign, (-) and if the voltage given to this , will produce out of phase signal at output. And the other one called the Non-inverting Input, marked with a positive or “plus” sign (+) and if the voltage given to this, will produce in phase signal at output. The third terminal represents the operational amplifiers output port.

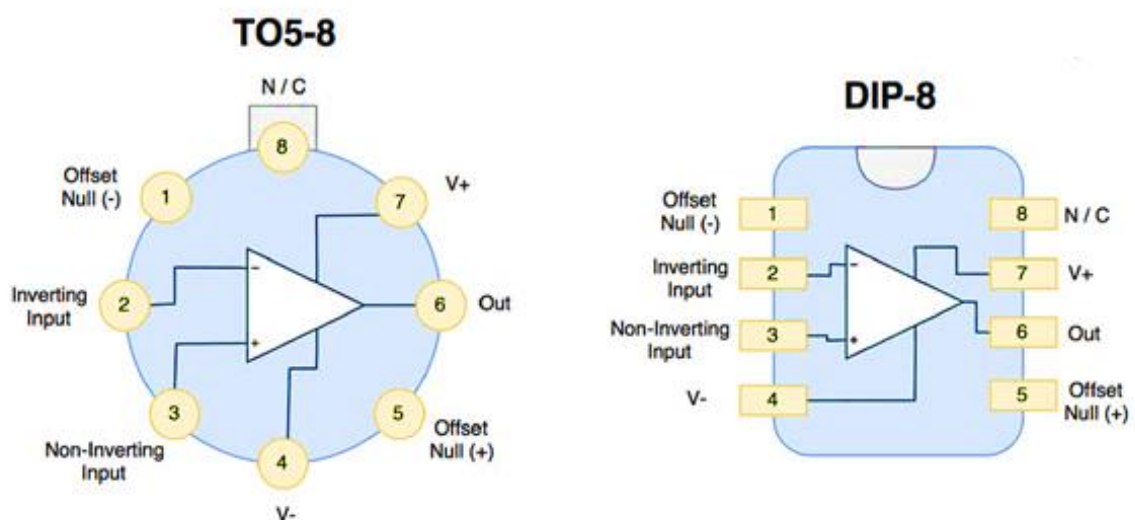
PIN DESCRIPTION

Fig 1.2: Pin-outs of IC 741

Pin4 & Pin7 (Power Supply): Pin7 is the positive voltage supply terminal and Pin4 is the negative voltage supply terminal. The voltage between these two pins can be anywhere between 5V and 18V.

Pin6 (Output): This is the output pin of IC 741. The voltage at this pin depends on the signals at the input pins and the feedback mechanism used.

Pin2 & Pin3 (Input): These are input pins for the IC. Pin2 is the inverting input and Pin3 is the non-inverting input.

If the input voltage at Pin2 (inverting input) is greater than Pin3 (non-inverting input) then the output voltage is negative. If the input voltage at Pin3 (non-inverting input) is greater than Pin2 (inverting input) then the output voltage is positive.

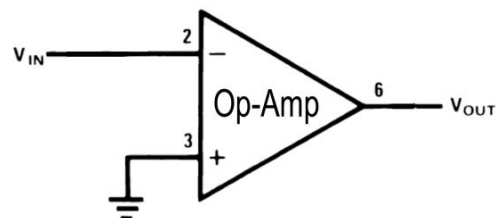
Pin1 & Pin5 (Offset Null): Because of high gain provided by 741 Op-Amp, even slight differences in voltages at the inverting and non-inverting inputs, caused due to irregularities in manufacturing process or external disturbances, can influence the output. To nullify this effect, an offset voltage can be applied at pin1 and pin5, and is usually done using a potentiometer.

Pin8 (N/C): This pin is not connected to any circuit inside 741 IC. It's just a dummy lead used to fill the void space in standard 8 pin packages.

TYPES OF OP-AMPS

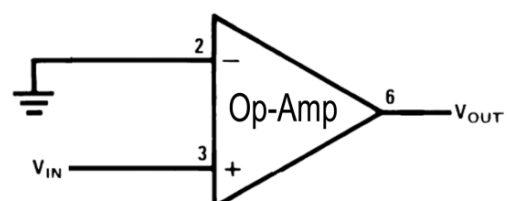
1. Inverting Op-Amp

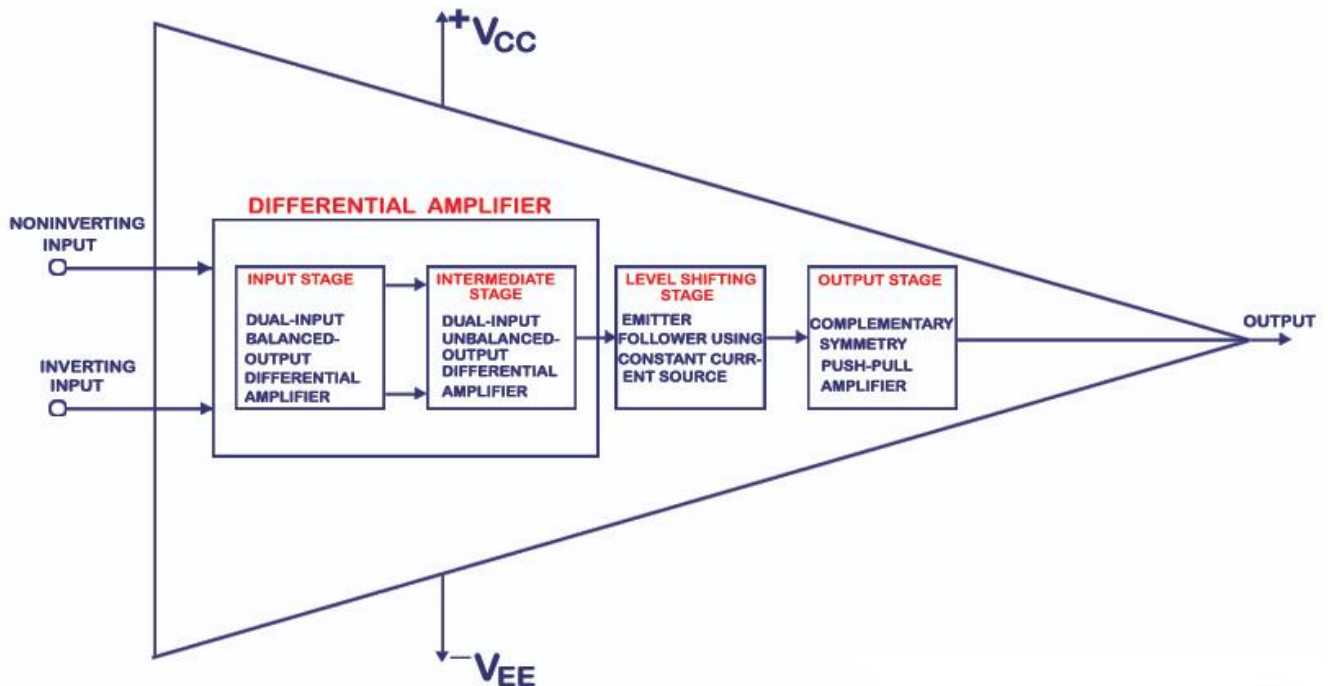
When input is applied to inverting terminal and non-inverting terminal connected to ground, then it is called Inverting Op-Amp.



2. Non inverting Op-Amp

When input is applied to non-inverting terminal and inverting terminal connected to ground, then it is called Non-inverting Op-Amp.



1.2 BLOCK DIAGRAM OF AN OP-AMP**Fig 1.3: Op-amp block diagram**

1. The Input Stage is a dual input balanced output differential amplifier which provides most of the voltage gain of amplifier and also establishes the input resistance of op-amp.

The i/p stage should have the following characteristics:

- High i/p resistance (typ. 10M ohm)
 - Low i/p bias current (typ. 0.5 micro-Amp.)
 - Small input offset voltage (typ. 10 mV)
 - Small input offset current (typ. 0.2 mA)
 - High CMRR (typ. 70 dB)
 - High Open-loop voltage gain (typ. 10⁴)
2. The Intermediate Stage is a dual input unbalanced output differential amplifier.
 - Driven by the o/p of 1st stage
 - Direct coupling → dc voltage well above ground level
 - Increases the overall gain of op-amp

3. DC voltage at the output of intermediate stage will be above ground potential. There is problem with direct coupling to output stage. Therefore, a Level Shifting Stage is used to shift the dc level to zero.
 - Dc voltage level is shifted to zero w.r.t ground
 - It is the emitter follower with constant current source
4. The Output Stage is usually a complementary push-pull amplifier which increases output voltage swing and current supplying capability of the op-amp. It is also responsible for establishing low output resistance of the op-amp.

It should have following characteristics:

- Large output voltage swing capability
- Large output voltage swing capability
- Low output resistance
- Short circuit protection

CHARACTERISTICS OF AN IDEAL OP-AMP

1. Infinite open-loop gain ($G = v_{out} / v_{in}$)
2. Infinite input impedance R_{in} , and so zero input current
3. Zero output impedance R_{out}
4. Zero input offset voltage
5. Infinite output voltage range
6. Infinite bandwidth with zero phase shift
7. infinite slew rate
8. Infinite common-mode rejection ratio (CMRR)
9. Zero noise
10. The characteristics of Op-Amp will not change with temperature.

1.3 INVERTING OPERATIONAL AMPLIFIER CONFIGURATION

Gain of inverting amplifier

Figure shows inverting amplifier with negative feedback. Now, calculate the overall voltage gain of an amplifier.

We know that input impedance of op amp is infinite. Hence, we can assume that the current flow through the R is equal to the current flows through R_f .

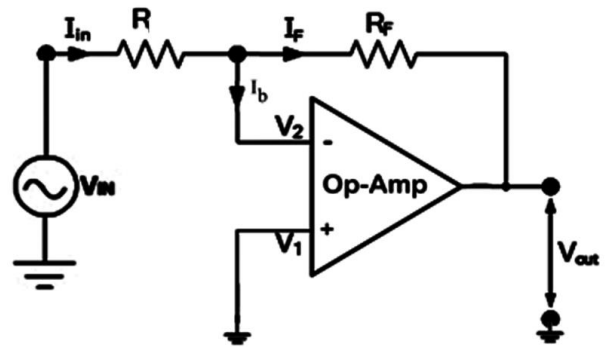


Fig 1.4: Op-amp inverting configuration

Current flows through the R is

$$I_{in} = \frac{V_{in} - V_2}{R}$$

Current passing through R_f is

$$I_f = \frac{V_2 - V_0}{R_f}$$

The closed loop current gain A_f of inverting op amp can be obtained by applying KCL at node V_2

$$I_{in} = I_b + I_f$$

$I_{in} = I_f$ since the input impedance is high, so I_b is neglected in above equation.

$$\frac{V_{in} - V_2}{R} = \frac{V_2 - V_0}{R_f} \quad \text{----- 1}$$

However, the open loop gain is

$$\Rightarrow V_0 = -AV_2$$

$$\Rightarrow V_2 = -\frac{V_0}{A} \quad \text{----- 2}$$

From 1 and 2

$$R_f(V_{in} - V_2) = R(V_2 - V_0)$$

$$R_f\left(V_{in} + \frac{V_0}{A}\right) = R\left(-\frac{V_0}{A} - V_0\right)$$

$$R_f V_{in} = -\frac{V_0 R_f}{A} - \frac{V_0 R}{A} - V_0 R$$

$$R_f V_{in} = -\frac{V_0 R_f}{A} - \frac{V_0 R}{A} - V_0 R$$

$$\frac{V_0(R_f + R + AR)}{A} = -R_f V_{in}$$

$$\frac{V_0}{V_{in}} = \frac{-R_f A}{(R_f + R) + AR}$$

$$A_f = \frac{V_0}{V_{in}} = \frac{-R_f A}{AR} \quad \because (R_f + R) \ll AR$$

$$A_f = \frac{-R_f}{R}$$

The negative sign indicates that input and output signal are in out of phase by 180°.

1.4 NON-INVERTING OPERATIONAL AMPLIFIER CONFIGURATION

Gain of non-inverting amplifier

Fig 1.5 shows non inverting amplifier with negative feedback. Now, calculate the overall voltage gain of an amplifier.

$$\text{Closed loop gain is } A_f = \frac{V_0}{V_{in}}$$

$$\text{Open loop gain is } A = \frac{V_0}{V_1 - V_2}$$

$$V_0 = A(V_1 - V_2) \quad \text{----- 1}$$

Now substitute the voltages V1, V2 in the above equation and from fig 1.5, V1 and V2 are given by

$$V_1 = V_{in}$$

$$V_2 = I_{in} R = \frac{V_0 R}{R + R_f}$$

From 1

$$V_0 = A(V_1 - V_2)$$

$$V_0 = A \left(V_{in} - \frac{V_0 R}{R + R_f} \right)$$

$$V_0 = AV_{in} - \frac{AV_0 R}{R + R_f}$$

$$V_0 + \frac{AV_0 R}{R + R_f} = AV_{in}$$

$$V_0 \left(1 + \frac{AR}{R + R_f} \right) = AV_{in}$$

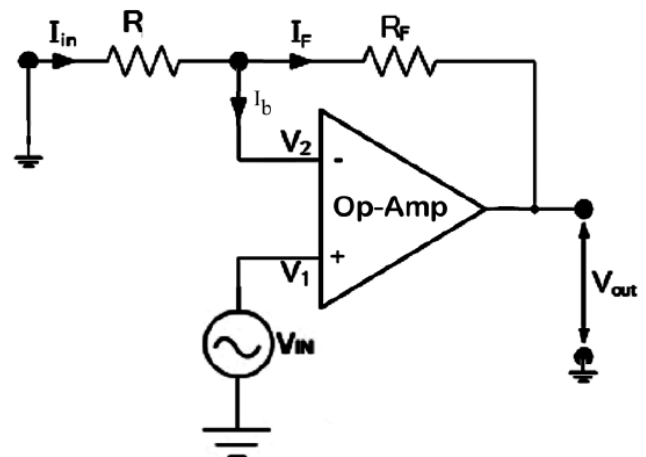


Fig 1.5: Op-amp non-inverting configuration

$$V_0 \left(\frac{R+R_f+AR}{R+R_f} \right) = AV_{in}$$

$$\frac{V_0}{V_{in}} = \frac{A(R+R_f)}{(R+R_f)+AR}$$

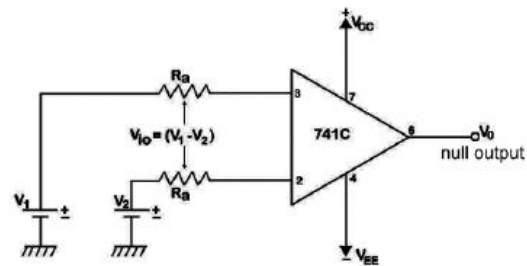
$$A_f = \frac{V_0}{V_{in}} = \frac{R+R_f}{R} \quad \because R+R_f \ll AR$$

$$A_f = 1 + \frac{R_f}{R}$$

1.5 OP-AMP PARAMETERS

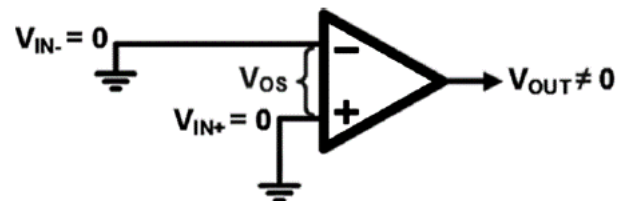
1. Input Offset Voltage

Input offset voltage is defined as the voltage that must be applied between the two input terminals of an OP-AMP to null or zero the output.



2. Output Offset Voltage

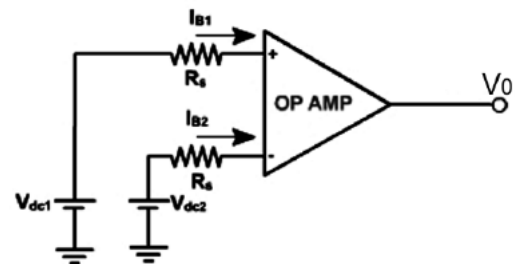
The output offset voltage is that the DC voltage present at the output terminal, when two input terminals are grounded.



3. Input offset Current

The input offset current I_{io} is the difference between the currents into inverting and non-inverting terminals of an op-amp.

$$I_{io} = I_{B1} - I_{B2}$$



4. Input Bias Current

The input bias current I_B is the average of the current entering the input terminals of an op-amp i.e.

$$I_B = \frac{I_{B1} + I_{B2}}{2}$$

5. Input Resistance (R_i)

R_i is the equivalent resistance that can be measured at either the inverting or non-inverting input terminal with the other terminal grounded. For the 741C the input resistance is relatively high $2M\Omega$.

6. Output Resistance

The output resistance is the equivalent resistance that can be measured between terminal and ground. It is 75 ohms for IC741

7. Input Capacitance (C_i)

C_i is the equivalent capacitance that can be measured at either the inverting and non-inverting terminal with the other terminal connected to ground. A typical value of C_i is 1.4 pf for the 741 IC.

1.6 VIRTUAL GROUND

Here the non-inverting terminal is grounded, the input is applied to inverting through R.

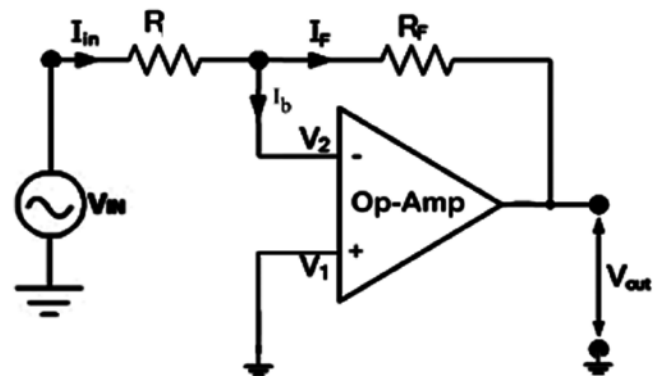
The open loop gain $A = \frac{V_0}{V_{id}}$

$$A = \frac{V_0}{V_1 - V_2}$$

$$V_1 - V_2 = \frac{V_0}{A}$$

$$V_1 - V_2 = 0 \quad (\text{Since } A \text{ is very large})$$

$$V_1 \approx V_2$$



The voltage at inverting terminal V_2 is almost equal to the voltage at non-inverting terminal V_1 . In the other words the inverting terminal voltage V_2 is approximately at ground potential. Therefore the inverting terminal is said to be at virtual ground. This concept is extremely useful in analysis of closed loop gain of inverting operation amplifier. For example, an ideal op-amp closed loop gain can be obtained by using the virtual ground concept.

From the given circuit $I_{in} = I_b + I_f$

$\Rightarrow I_{in} \approx I_f$ since the input impedance is high, so I_b is neglected in above equation.

$$\frac{V_{in} - V_2}{R} = \frac{V_2 - V_0}{R_f}$$

From virtual ground concept $V_1 = V_2$

$$\frac{V_{in}}{R} = \frac{-V_0}{R_f}$$

$$\frac{V_0}{V_{in}} = -\frac{R_f}{R}$$

$$\therefore A_f = -\frac{R_f}{R}$$

This is the gain of inverting op-amp and is same as gain of normal inverting amplifier.

1.7 SUMMING AMPLIFIER (ADDER)

In the summing amplifier circuit, the output voltage, (V_0) is the sum of the input voltages, V_A , V_B , V_C . Fig 1.6, shows inverting configuration summing amplifier.

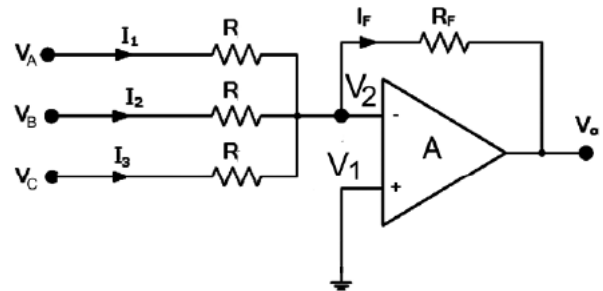


Fig 1.6: Op-amp as summing amplifier

Applying KCL at node V2

$$I_1 + I_2 + I_3 = I_f$$

$$\frac{V_A - V_2}{R} + \frac{V_B - V_2}{R} + \frac{V_C - V_2}{R} = \frac{V_2 - V_0}{R_f}$$

$$V_1 = V_2 = 0 \quad (\text{virtualgroundconcept})$$

$$\frac{V_A}{R} + \frac{V_B}{R} + \frac{V_C}{R} = -\frac{V_0}{R_f}$$

$$V_0 = -R_f \left(\frac{V_A}{R} + \frac{V_B}{R} + \frac{V_C}{R} \right)$$

$$V_0 = -\frac{R_f}{R} (V_A + V_B + V_C)$$

If $R = R_f$ in above circuit

$$V_0 = -(V_A + V_B + V_C)$$

Therefore the output is equal to -ve sum of all inputs.

1.8 SUBTRACTOR / DIFFERENTIAL AMPLIFIER USING OP-AMP

The circuit diagram of a subtractor/differential amplifier using one op-amp is shown below fig 1.7. R_1 and R_2 are the input resistors, R_f is the feedback resistor and R_L is the load resistor. Differential amplifier is just a combination of an inverting and non-inverting amplifier.

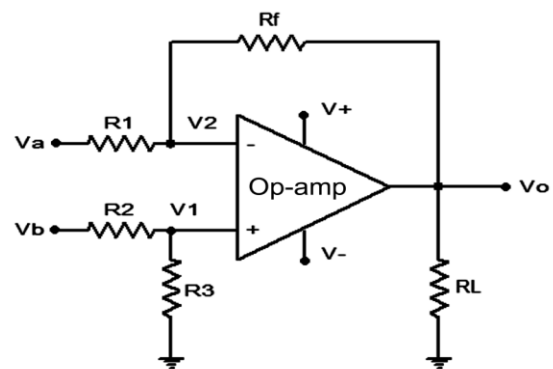


Fig 1.7: Op-amp as subtractor

Sum of the output voltages of these two configurations separately is the overall output voltage.

If V_b is made zero, the circuit becomes an inverting amplifier. The output voltage V_o is given by

$$V_{0a} = -\frac{R_f}{R_1}(V_a) \text{ ----- 1}$$

When V_a is made zero the circuit becomes a non-inverting amplifier. Let V_1 be the voltage at the non-inverting input pin. Relation between V_b and V_1 can be expressed using the following equation.

$$V_1 = \frac{V_b}{R_2 + R_3}(R_3) \text{ ----- 2}$$

Output voltage V_o for non-inverting amplifier is given by

$$V_{0b} = \frac{V_b}{R_2 + R_3}(R_3) \left(1 + \frac{R_f}{R_1}\right) \text{ ----- 3}$$

Let $R_1 = R_2$ and $R_3 = R_f$ then

$$V_{0b} = \frac{V_b}{R_1 + R_f}(R_f) \left(1 + \frac{R_f}{R_1}\right) = \frac{R_f}{R_1 + R_f} \left(\frac{R_1 + R_f}{R_1}\right)(V_b) = \frac{R_f}{R_1}(V_b)$$

$$V_{0b} = \frac{R_f}{R_1}(V_b) \text{ ----- 4}$$

Then overall output voltage is from eqn 1 and 4

$$V_o = V_{0a} + V_{0b} = -\frac{R_f}{R_1}(V_a) + \frac{R_f}{R_1}(V_b) = -\frac{R_f}{R_1}(V_a - V_b)$$

If $R_1 = R_f$ the

$$V_o = V_b - V_a$$

1.9 VOLTAGE FOLLOWER

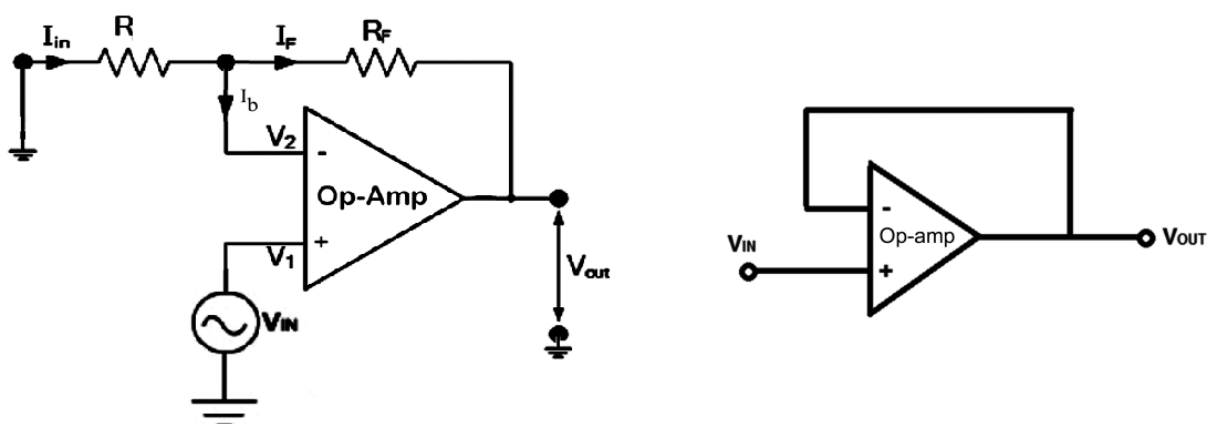


Fig 1.8: Op-amp as voltage follower

The voltage follower is a modification of non-inverting op-amp.

The gain of non-inverting op-amp is

$$A_f = 1 + \frac{R_f}{R}$$

$$A_f = \frac{V_o}{V_{in}} = 1 + \frac{R_f}{R}$$

To get minimum gain, modification of non-inverting op-amp as follows:

$R = \infty$ and $R_f = 0$ as shown in fig 1.8. R is infinity means that it is open circuited. R_f is zero means that it is short circuited.

$$A_f = \frac{V_o}{V_{in}} = 1 + \frac{0}{\infty}$$

$$\frac{V_o}{V_{in}} = 1$$

$$V_o = V_{in}$$

This shows that whatever the voltage is applied at input, the same voltage appears at output. This means that the output voltage is in phase with the input voltage. Hence, the circuit is called voltage follower.

1.10 INTEGRATOR CIRCUIT USING OP-AMP

Integrator is a circuit whose output is directly proportional to time integral of input voltage. Such a circuit is also termed as an integrating amplifier. The circuit is somewhat similar to an op-amp inverting amplifier but the feedback resistor R_f is replaced by a capacitor C_f . The circuit diagram of an op-amp as an integrator is shown below.

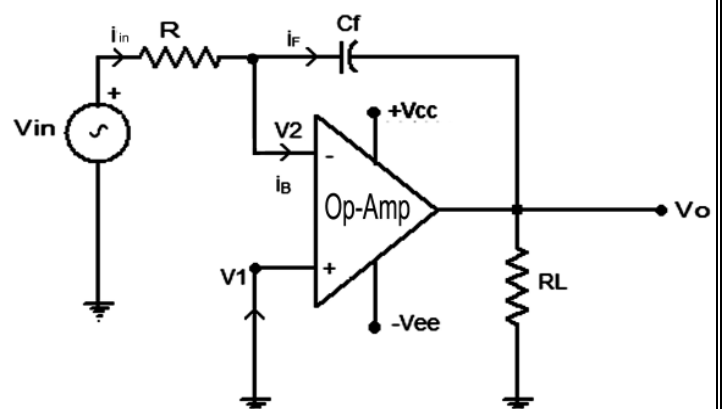


Fig 1.9: Integrator using Op-amp

Applying Kirchhoff's current (KCL) at node V_2

we get $I_{in} = I_f + I_b$

Since the input resistance of an op-amp is very high, I_b will be very small and it can be neglected. Therefore, $I_{in} = I_f$

The relation between the current through a capacitor and voltage across it is

$$Q = CV \Rightarrow \frac{d}{dt}Q = C_f \frac{d}{dt}V \Rightarrow I_f = C_f \frac{d}{dt}(V_2 - V_o)$$

And

$$I_{in} = \frac{(V_{in} - V_2)}{R}$$

Therefore, the equation $I_{in} = I_f$ can be rewritten as

$$\frac{(V_{in} - V_2)}{R} = C_f \frac{d}{dt}(V_2 - V_0)$$

$V_1 = V_2 = 0$ from the concept of virtual ground.

So, the above equation becomes

$$\frac{V_{in}}{R} = -C_f \frac{d}{dt}(V_0)$$

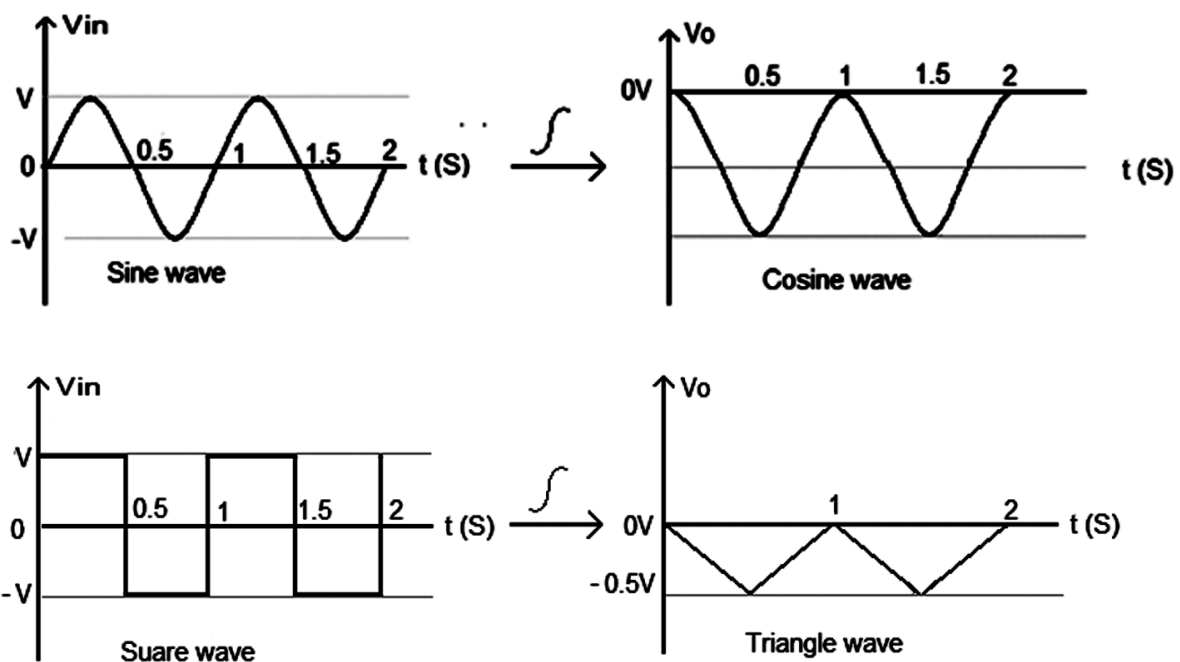
Integrating the both sides of the above equation with respect to time, we get

$$\int \frac{V_{in}}{R} dt = -C_f (V_0)$$

$$V_0 = -\frac{1}{RC_f} \int V_{in} dt \quad \Rightarrow V_0 \propto \int V_{in} dt$$

Thus, the output voltage is directly proportional to the time integral of input voltage. Where $-(1/RC_f)$ is the time constant of the circuit.

Integrating a square wave will result in a triangle waveform and integrating a sine wave will result in a Cosine waveform. It is shown in the figures shown below.



1.11 DIFFERENTIATOR CIRCUIT USING OP-AMP

Differentiator is a circuit whose output is directly proportional to derivative of input voltage. The circuit is somewhat similar to an op-amp inverting amplifier but the input resistor R is replaced by a capacitor C . The circuit diagram of an op-amp as differentiator is shown below.

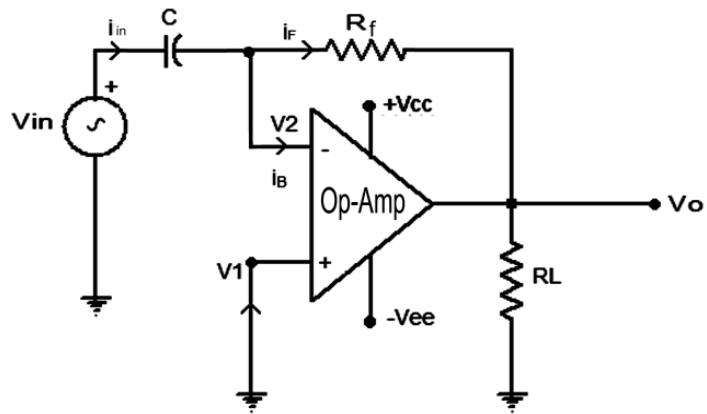


Fig 1.10: Differentiator using Op-amp

Applying Kirchhoff's current (KCL) at node V_2 we get $I_{in} = I_f + I_b$

Since the input resistance of an op-amp is very high, I_b will be very small and it can be neglected. Therefore, $I_{in} = I_f$

The relation between the current through a capacitor and voltage across it is

$$Q = CV \Rightarrow \frac{d}{dt}Q = C \frac{d}{dt}V \Rightarrow I_{in} = C \frac{d}{dt}(V_{in} - V_2)$$

And

$$I_f = \frac{(V_2 - V_o)}{R_f}$$

Therefore, the equation $I_{in} = I_f$ can be rewritten as

$$C \frac{d}{dt}(V_{in} - V_2) = \frac{(V_2 - V_o)}{R_f}$$

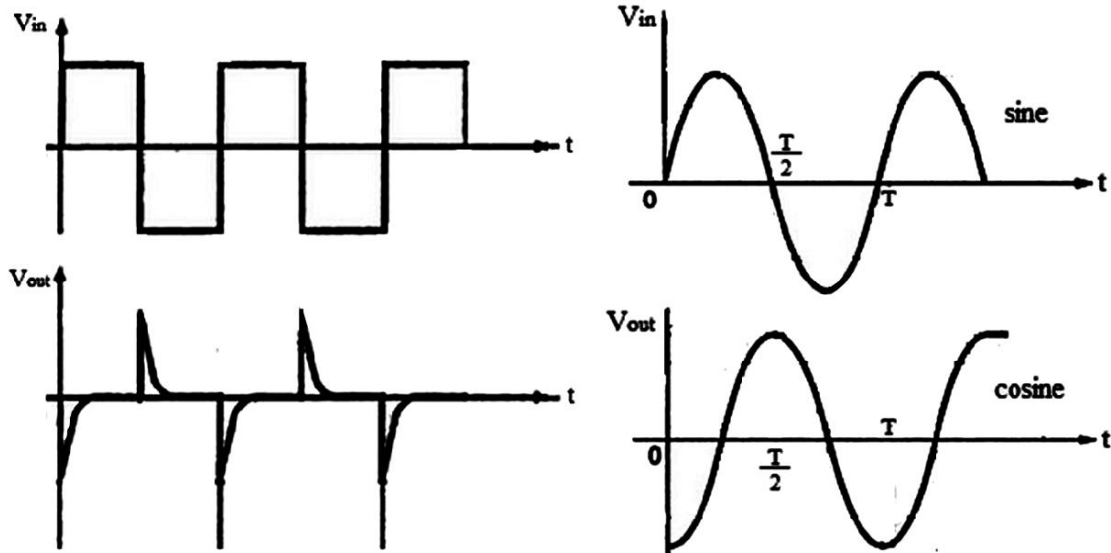
$V_1 = V_2 = 0$ from the concept of virtual ground. So, the above equation becomes

$$C \frac{d}{dt}(V_{in}) = -\frac{V_o}{R_f}$$

$$V_o = -R_f C \frac{d}{dt}(V_{in})$$

$$V_o \propto \frac{d}{dt}(V_{in})$$

Thus, the output voltage is directly proportional to the derivative of input voltage. Where $-RC_f$ is the time constant of the circuit. The negative sign indicates the output is out of phase by 180° with respect to the input. If the input to the differentiator is a square wave, the output will be a waveform consisting of positive and negative spikes, corresponding to the charging and discharging of the capacitor, the output of a differentiator for a sine wave input is a cosine wave and the input-output waveforms are shown in the figure below.



1.12 LOGARITHMIC AMPLIFIER (LOG AMPLIFIER)

Log amplifier is a linear circuit in which the output voltage will be a logarithm of the input. Log amplifier finds a lot of application in electronic fields like multiplication or division, signal processing, computerized process control, compression, decompression, RMS value detection etc. Basically, there are two log amp configurations: Op-amp-diode log amplifier and Op-amp-transistor log amplifier.

Logarithmic amplifier with Diode:

The circuit of a simple Op-amp diode log amplifier is shown above fig 1.11. This is an op-amp inverting configuration with a diode in the feedback path. The voltage across the diode will be always proportional to the log of the current through it and when a diode is placed in the feedback path of an op-amp in inverting mode, the output voltage will be proportional to the negative log of the input current. Since the input current is proportional to the input voltage, we can say that the output voltage will be proportional to the negative log of the input voltage.

The PN junction diode equation is

$$I_d = I_s \left(e^{\frac{V_d}{V_t}} - 1 \right)$$

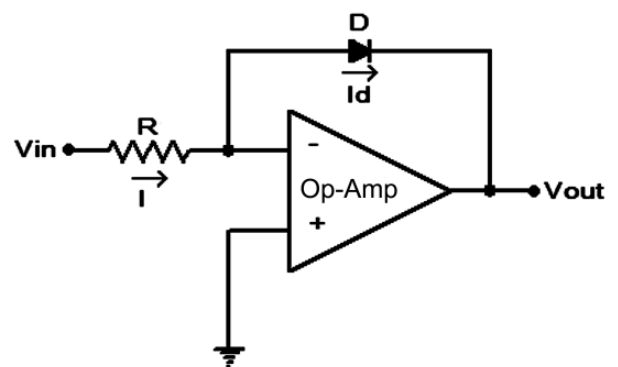


Fig 1.11: Logarithmic amplifier with diode

Where I_d is the diode current, I_s is the saturation current, V_d is the voltage across the diode and V_t is the thermal voltage. Since V_d the voltage across the diode is positive here and V_t the thermal voltage is a small quantity

$$\text{i.e. } e^{\frac{V_d}{V_t}} \gg 1$$

The above equation can be written as

$$I_d = I_s e^{\frac{V_d}{V_t}}$$

Since the inverting input pin of the op-amp is virtually grounded, we can say that

$$I = \frac{V_{in}}{R}$$

Since an ideal op-amp has infinite input resistance, the input current I has only one path, that is through the diode. That means the input current is equal to the diode current I_d .

$$I = I_d$$

$$\frac{V_{in}}{R} = I_s e^{\frac{V_d}{V_t}} \Rightarrow V_{in} = R I_s e^{\frac{V_d}{V_t}} \Rightarrow e^{\frac{V_d}{V_t}} = \frac{V_{in}}{R I_s}$$

Considering that the negative of the voltage across diode is the output voltage V_0 . we can rearrange the above equation to get

$$\frac{V_d}{V_t} = \log \frac{V_{in}}{R I_s} \Rightarrow V_d = -V_t \log \frac{V_{in}}{R I_s} \Rightarrow V_0 = -V_t \log \frac{V_{in}}{R I_s}$$

UNIT – II**OPERATIONAL AMPLIFIER CIRCUITS****2.1 VOLTAGE REGULATOR USING OP-AMP**

In voltage regulators, the output voltage should be independent of input voltage variations and the load current variations.

Fig. 2.1 shows a voltage regulator using op-amp with a fixed reference voltage source.

If V_{ref} is fixed and $R_1=R_2$, then the circuit acts as fixed voltage regulator.

The closed loop voltage gain is,

$$A_f = V_0 / V_{in} = 1 + R_2 / R_1$$

$$V_0 = V_{ref} (1 + R_2 / R_1)$$

If $R_1 = R_2$ then $V_0 = 2 V_{ref}$.

If $V_{ref} = 5V$, the $V_0 = 10V$. Any variations in V_+ and V_- are absorbed and the output voltage is fixed 10V.

Voltage regulator with Zener diode is shown in Fig. 2.2, here Zener diode connected to non-inverting terminal in reverse bias, that means it is made to operate only in break down region. Hence, the voltage applied to non-inverting terminal is Zener voltage V_z and is constant because the Zener diode is in reverse bias.

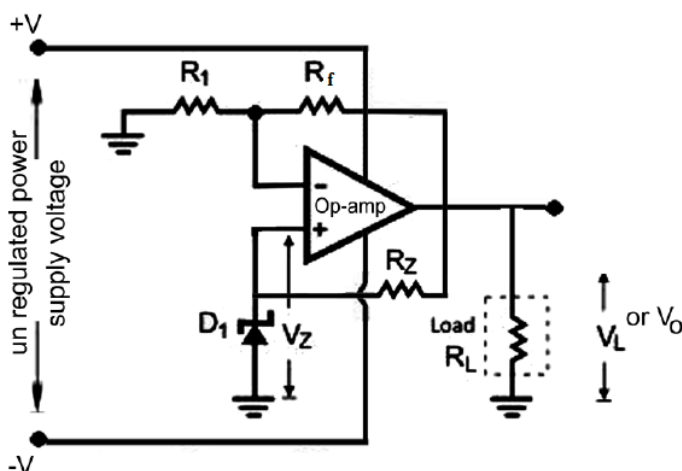


Fig 2.2: Voltage regulator with zener diode

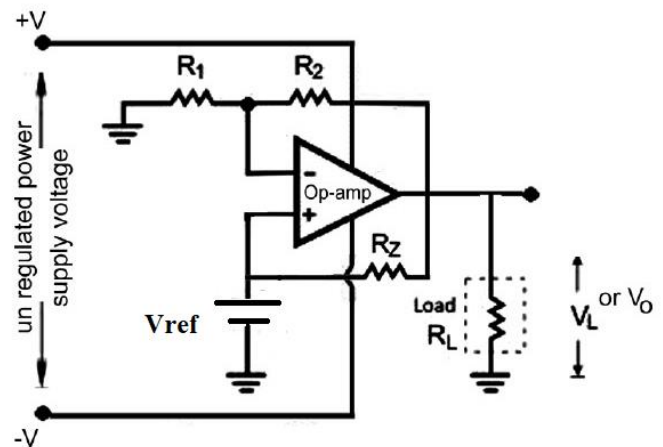


Fig 2.1: Voltage regulator with fixed reference voltage

The output voltage is given by,

$$V_0 = V_z \left(1 + \frac{R_f}{R_1} \right)$$

If $R_1 = R_f$

$$\text{Then } V_0 = V_z \left(1 + \frac{R_1}{R_1} \right) = V_z \left(1 + \frac{1}{1} \right) = 2V_z$$

Hence, the output voltage is fixed at $2V_z$ irrespective of input voltage variations.

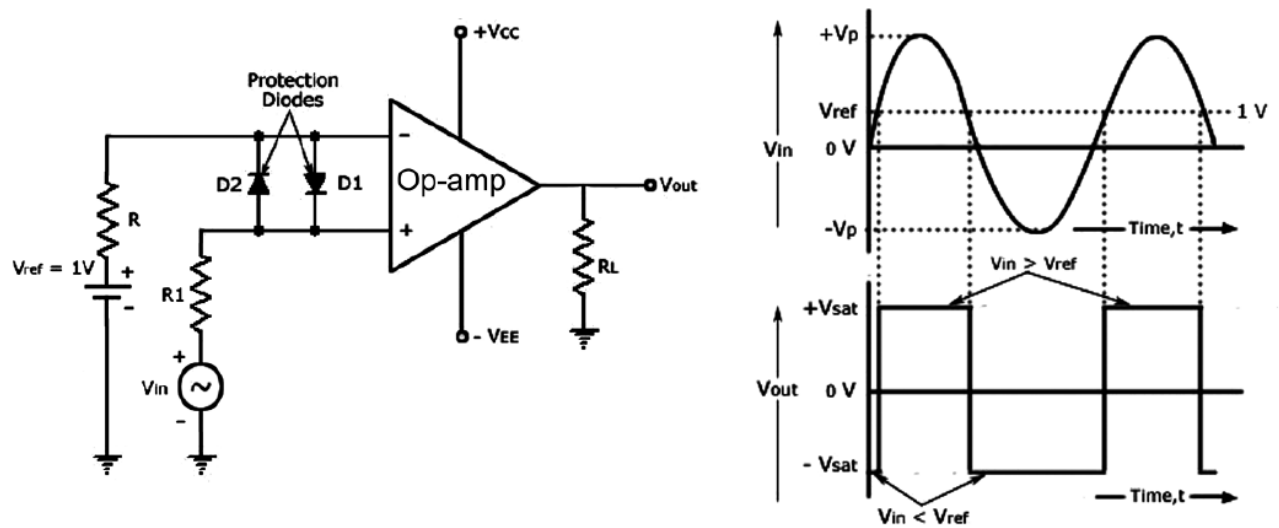
2.2 COMPARATOR USING OP-AMP

Fig 2.3: Comparator using op-amp and output wave forms

A comparator is a circuit where two voltage signals are to be compared and to be distinguished on which is stronger. An op-amp comparator circuit is shown in the fig. 2.3. It is called a non-inverting comparator circuit as the sinusoidal input signal V_{in} is applied to the non-inverting terminal. The fixed reference voltage V_{ref} is given to the inverting terminal (-) of the op-amp.

When the value of the input voltage V_{in} is greater than the reference voltage V_{ref} the output voltage V_o goes to positive saturation. This is because the voltage at the non-inverting input is greater than the voltage at the inverting input.

When the value of the input voltage V_{in} is lesser than the reference voltage V_{ref} , the output voltage V_o goes to negative saturation. This is because the voltage at the non-inverting input is smaller than the voltage at the inverting input. Thus, output voltage V_o changes from positive saturation point to negative saturation point whenever the difference between V_{in} and V_{ref} changes. This is shown in the waveform above. The comparator can be called a voltage level detector, as for a fixed value of V_{ref} , the voltage level of V_{in} can be detected.

The circuit diagram shows the diodes D_1 and D_2 . These two diodes are used to protect the op-amp from damage due to increase in input voltage. These diodes are called clamp diodes. Resistance R_1 is connected in series with input voltage V_{in} and R is connected between the inverting input and reference voltage V_{ref} . R_1 limits the current through the clamp diodes and R reduces the offset problem.

2.3 ZERO CROSSING DETECTOR USING OP-AMP

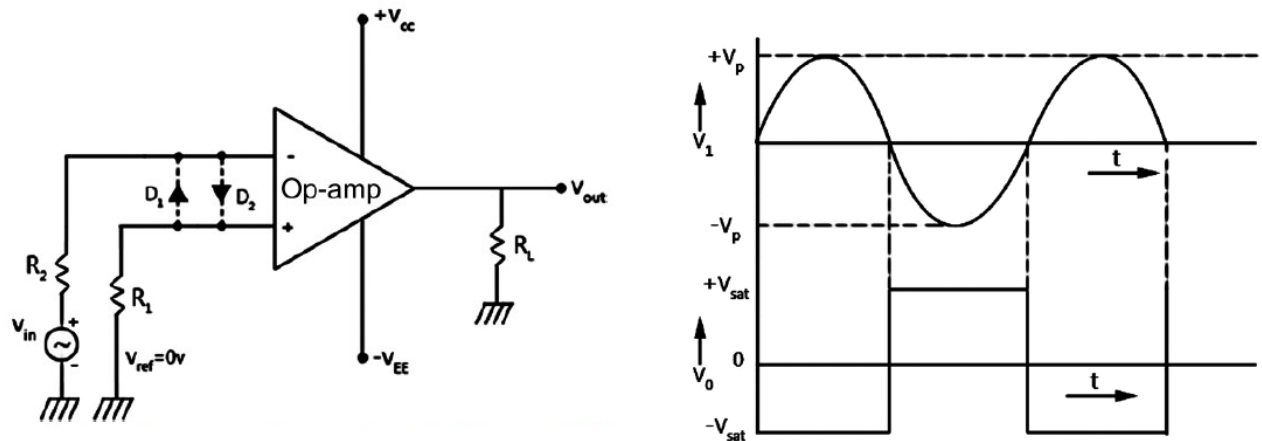


Fig 2.4: Zero crossing detector using op-amp and output wave forms

The zero-crossing detector circuit is an important application of the op-amp comparator circuit. It can also be called as the sine to square wave converter. Any one of the inverting or non-inverting comparators can be used as a zero-crossing detector. The only difference is that the reference voltage must be made zero ($V_{ref} = 0V$). An input sine wave is given as V_{in} . These are shown in the circuit diagram and input and output waveforms of an inverting comparator with a 0V reference voltage.

As shown in the waveform, for a reference voltage 0V, when the input sine wave passes through zero and goes in positive direction, the output voltage V_{out} is driven into negative saturation. Similarly, when the input voltage passes through zero and goes in the negative direction, the output voltage is driven to positive saturation. The diodes D_1 and D_2 are called clamp diodes. They are used to protect the op-amp from damage due to increase in input voltage.

2.4 SQUARE WAVE GENERATOR USING OP-AMP (Astable Multivibrator)

An astable multivibrator is a nonlinear circuit configuration using op-amp, which generates square waves without any external triggering. This circuit has no stable output state, only two quasi-states. The output oscillates continuously between these two quasi-stable states. An astable multivibrator is basically an oscillator, since it requires no external pulse to trigger it. For this reason, the circuit is often referred to as a free running oscillator.

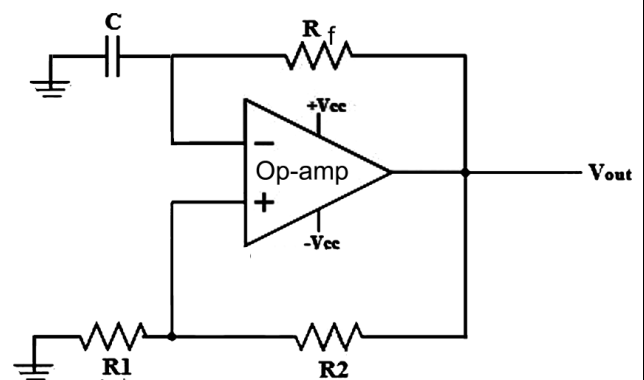


Fig 2.5: Square wave generator

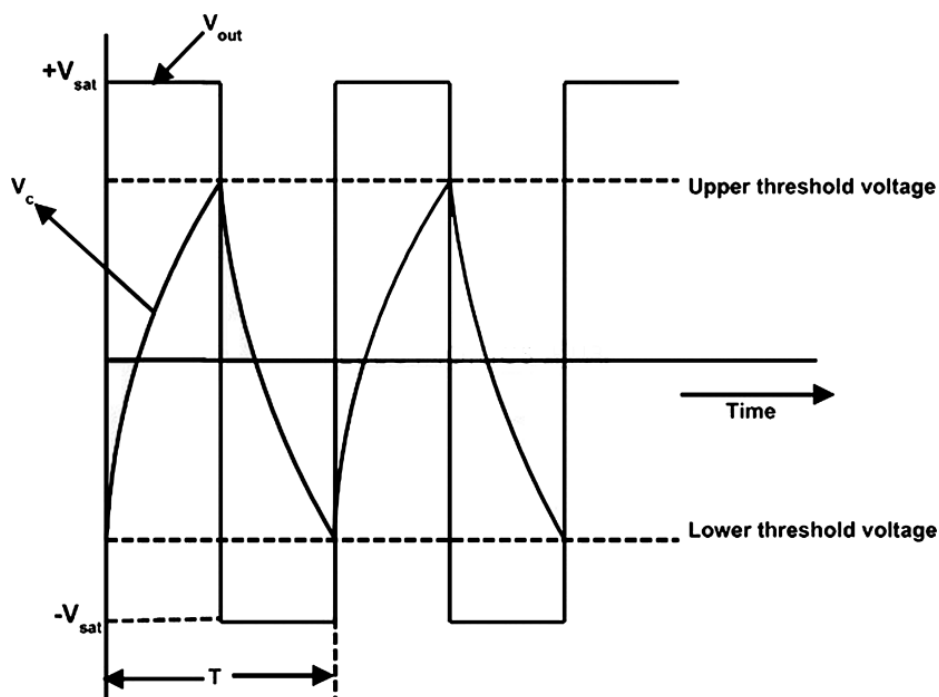
However, the circuit uses DC power supply for the op-amp. An astable multivibrator can be configured to produce square waves of required frequency, amplitude and duty cycle. The circuit diagram of an astable multivibrator using operational amplifier is as shown in the figure below. The circuit is a Schmitt trigger configuration, which has a feedback connection and includes an input capacitor at the inverting input terminal.

When the astable multivibrator circuit output is at its positive saturation level, current flows through the feedback resistor R_f into the capacitor C . This charges the capacitor, with the top plate positive. The capacitor gets charged until its voltage reaches the upper trigger voltage of the Schmitt trigger. At this point, the output of the circuit switches to its negative saturation level immediately. No current flows into the capacitor now and thus the capacitor starts discharging. The discharging of the capacitor continues till the capacitor voltage reaches the lower trigger voltage of the Schmitt trigger. The output switches to its positive saturation level and the cycle repeats.

It can be noted that the circuit is a square wave generator whose output swings between the op-amp positive and negative saturation voltage levels. The frequency of the output square wave depends on the capacitance C and the feedback resistor value, R_f .

$$T = 2R_f C \log \left(1 + \frac{2R_1}{R_2} \right)$$

The output and the capacitor voltage waveforms of an astable multivibrator are shown in the figure below.



2.5 SINE WAVE GENERATOR USING OP-AMP (Wien-Bridge Oscillator)

The Wien-bridge oscillator circuit is a sine-wave oscillator or generator. The frequency-selective Wien-bridge is constructed from the R1-C1 and R2-C2 networks. Normally, the Wien bridge is symmetrical, so that $C1 = C2 = C$ and $R1 = R2 = R$. When that condition is met, the phase relationship between the output and input signals varies from 90° to $+90^\circ$, and it reaches 0° at a center frequency f_0 .

$$f_0 = \frac{1}{2\pi RC} \text{ Hz}$$

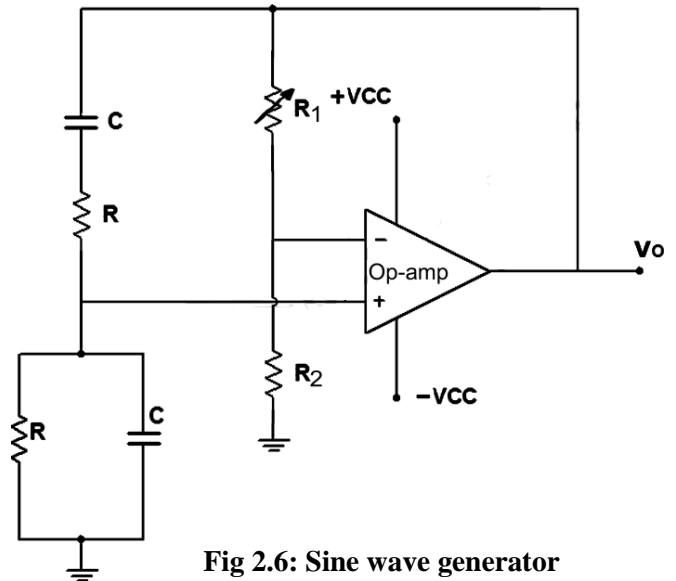
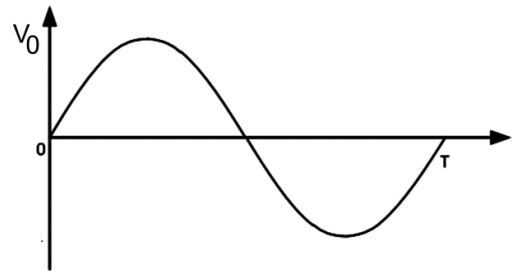


Fig 2.6: Sine wave generator

The Wien network is connected between the op-amp's output and the non-inverting input, so that the circuit gives zero overall phase shift at f_0 , the feedback network R1-R2, which gives an overall gain of unity by adjusting R1 for a low-distortion sine wave.



2.6 ACTIVE FILTERS

The main disadvantage of passive filters is that the amplitude of the output signal is less than that of the input signal, i.e., the gain is never greater than unity and that the load impedance affects the filters characteristics.

With passive filter circuits containing multiple stages, this loss in signal amplitude called "Attenuation" can become quite severe. One way of restoring or controlling this loss of signal is by using amplification through the use of Active Filters. Then the main difference between a "passive filter" and an "active filter" is amplification. The active filters are most extensively used in the field of communication and signal processing in radio, television, telephone, radar, space satellites, biomedical equipment ... etc.

2.6.1 Active Low Pass Filter:

The basic RC low pass non inverting filter circuit is shown in fig.2.7, The low pass active filter consists simply of a passive RC filter stage providing a low frequency path to the input of a non-inverting operational amplifier.

The frequency response of the circuit will be the same as that for the passive RC filter, except that the amplitude of the output is increased by the gain A_F of the amplifier.

For a non-inverting amplifier circuit, the voltage gain for the filter is given as

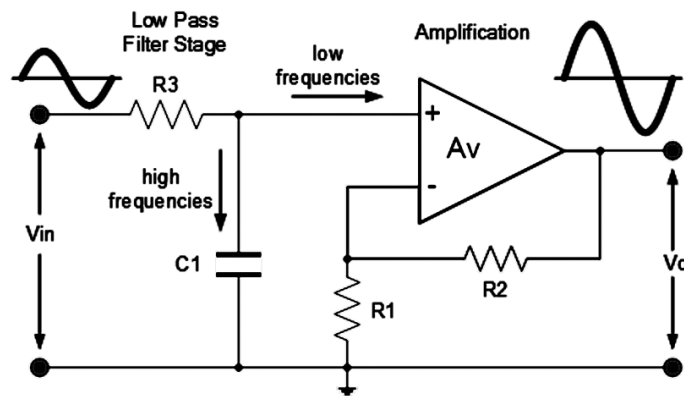


Fig 2.7: Active low pass filter

$$\text{Gain} = \frac{V_0}{V_1} = \left(1 + \frac{R_2}{R_1}\right) \Rightarrow V_0 = V_1 \left(1 + \frac{R_2}{R_1}\right) \Rightarrow V_0 = V_1 A_f$$

The output of the RC filter is given by

output of the filter (V_1) = RC filter gain (A) X RC filter input (V_{in})

$$A = \frac{1}{\sqrt{1 + \left(\frac{f}{f_c}\right)^2}}$$

$$V_1 = \frac{V_{in}}{\sqrt{1 + \left(\frac{f}{f_c}\right)^2}}$$

Now, the op-amp output is

$$V_0 = \frac{V_{in}}{\sqrt{1 + \left(\frac{f}{f_c}\right)^2}} A_f$$

Therefore, the gain of an active low pass filter as a function of frequency is given by

$$\text{Voltage gain } A_v = \frac{V_0}{V_{in}} = \frac{V_{in}}{\sqrt{1 + \left(\frac{f}{f_c}\right)^2}} A_f \times \frac{1}{V_{in}} = \frac{A_f}{\sqrt{1 + \left(\frac{f}{f_c}\right)^2}}$$

Where:

A_F = the pass band gain of the filter, $(1 + R_2/R_1)$

f = the frequency of the input signal (Hz)

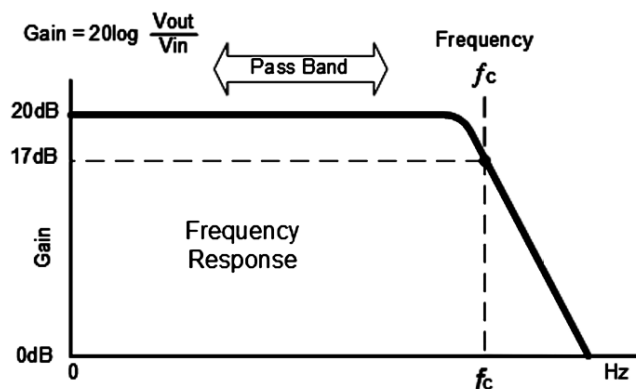
f_c = the cut-off frequency (Hz)

Thus, the operation of a low pass active filter can be verified from the frequency gain equation above as:

1. At very low frequencies, $f < f_c$ $\frac{V_0}{V_{in}} \cong A_f$
2. At the cut-off frequency, $f = f_c$ $\frac{V_0}{V_{in}} = \frac{A_f}{\sqrt{2}} = 0.707 A_f$
3. At very high frequencies, $f > f_c$ $\frac{V_0}{V_{in}} < A_f$

Thus, the Active Low Pass Filter has a constant gain A_f from 0 Hz to the high frequency cut-off point, f_c . At f_c the gain is $0.707A_f$, and after f_c it decreases at a constant rate as the frequency increases.

Frequency Response Curve



2.6.2 Active High Pass Filter:

The Active High Pass Filter can be constructed by reversing the positions of the resistor and capacitor in the circuit. This high pass filter consists simply of a passive filter followed by a non-inverting amplifier. The frequency response of the circuit is the same as that of the passive filter, except that the amplitude of the signal is increased by the gain of the amplifier.

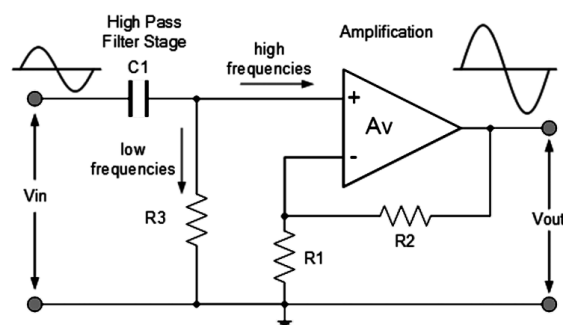


Fig 2.8: Active high pass filter

For a non-inverting amplifier circuit, the voltage gain for the filter is given as

$$\text{Gain} = \frac{V_0}{V_1} = \left(1 + \frac{R_2}{R_1}\right) \Rightarrow V_0 = V_1 \left(1 + \frac{R_2}{R_1}\right) \Rightarrow V_0 = V_1 A_f$$

The output of the RC filter is given by

output of the filter(V_1) = RC filter gain (A) X RC filter input(V_{in})

$$A = \frac{1}{1 + \frac{1}{J\omega CR}} = \frac{1}{\frac{J\omega CR + 1}{J\omega CR}} = \frac{J\omega CR}{1 + J\omega CR} = \frac{J2\pi fCR}{1 + J2\pi fCR} = \frac{J\left(\frac{f}{f_c}\right)}{1 + J\left(\frac{f}{f_c}\right)} = \frac{\left(\frac{f}{f_c}\right)}{\sqrt{1 + \left(\frac{f}{f_c}\right)^2}}$$

$$V_1 = V_{in} \times A = \frac{V_{in} \left(\frac{f}{f_c}\right)}{\sqrt{1 + \left(\frac{f}{f_c}\right)^2}}$$

Now, the op-amp output is
$$V_0 = \frac{V_{in} \left(\frac{f}{f_c}\right) A_f}{\sqrt{1 + \left(\frac{f}{f_c}\right)^2}}$$

Therefore, the gain of an active high pass filter as a function of frequency is given by

$$\text{Voltage gain } A_v = \frac{V_0}{V_{in}} = \frac{V_{in} \left(\frac{f}{f_c}\right) A_f}{\sqrt{1 + \left(\frac{f}{f_c}\right)^2}} \times \frac{1}{V_{in}} = \frac{A_f \left(\frac{f}{f_c}\right)}{\sqrt{1 + \left(\frac{f}{f_c}\right)^2}}$$

Where:

A_F = the pass band gain of the filter, $(1 + R_2/R_1)$

f = the frequency of the input signal (Hz)

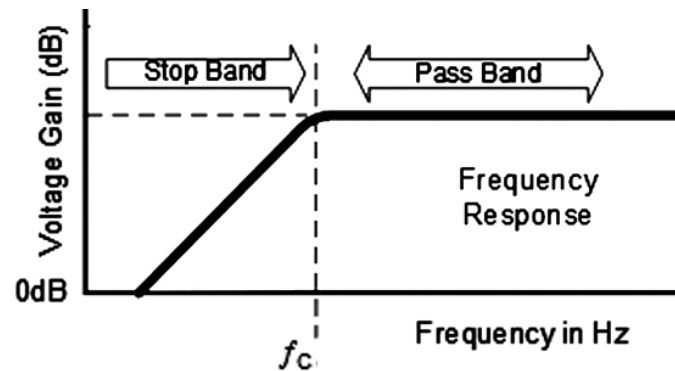
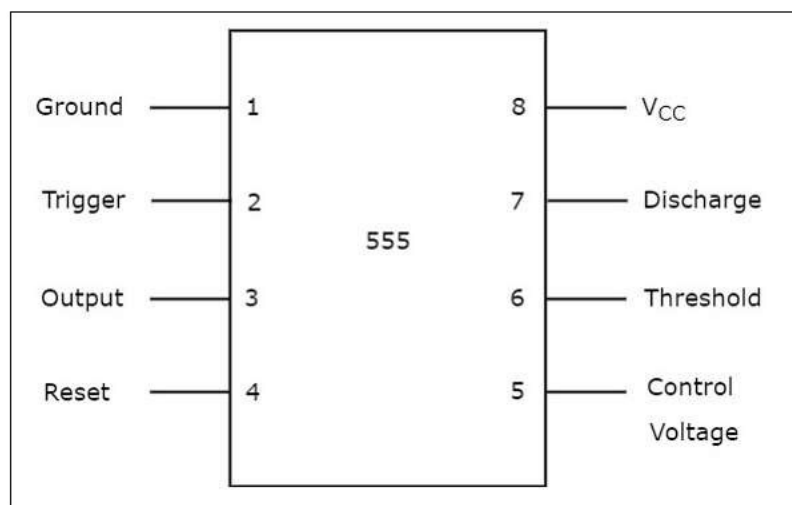
f_c = the cut-off frequency (Hz)

Thus, the operation of a high pass active filter can be verified from the frequency gain equation above as:

1. At very low frequencies, $f < f_c$ $\frac{V_0}{V_{in}} < A_f$
2. At the cut-off frequency, $f = f_c$ $\frac{V_0}{V_{in}} = \frac{A_f}{\sqrt{2}} = 0.707 A_f$
3. At very high frequencies, $f > f_c$ $\frac{V_0}{V_{in}} \cong A_f$

Then, the Active High Pass Filter has a gain A_F that increases from 0 Hz to the low frequency cut-off point f_c . At f_c the gain is $0.707A_F$, and after f_c all frequencies are pass band frequencies so the filter has a constant gain A_F .

Frequency Response Curve

**2.7 IC 555 TIMER****2.7.1 Pin Diagram****Fig 2.9: IC 555 Timer Pin diagram**

Pin 1: It is the ground pin directly connected to the negative rail.

Pin 2: It is the Trigger pin to activate the IC's timing cycle. It is generally low signal pin and the timer is triggered when voltage on this pin is below one third of the supply voltage. The trigger pin is connected to the Inverting input of the comparator inside the IC and accepts negative signals.

Pin 3: It is the output pin.

Pin 4: It is the reset pin. It should be connected to the positive rail to work the IC properly. When this pin is grounded, the IC will stop working.

Pin 5: Control pin – The $\frac{2}{3}$ supply voltage point on the terminal voltage divider is brought to the control pin. It requires to be connected to an external DC signal to modify the timing cycle.

Pin 6: It is the Threshold pin. The timing cycle is completed when voltage on this pin is equal to or greater than two-third of V_{CC}. It is connected to the non-inverting input of the upper comparator so that it accepts the positive going pulse to complete the timing cycle.

Pin 7: Discharge pin. It provides a discharge path for the timing capacitor through the collector of the NPN transistor, to which it is connected.

Pin 8: It is positive rail connected pin which is connected to positive terminal of the power supply. It is also known as V_{cc} . IC555 works in a wide range of voltage from 5V to 18 V DC

2.7.2 Functional Block Diagram

The pictorial representation showing the internal details of a 555 Timer is known as functional diagram. The functional diagram of 555 Timer IC is shown in the following figure.

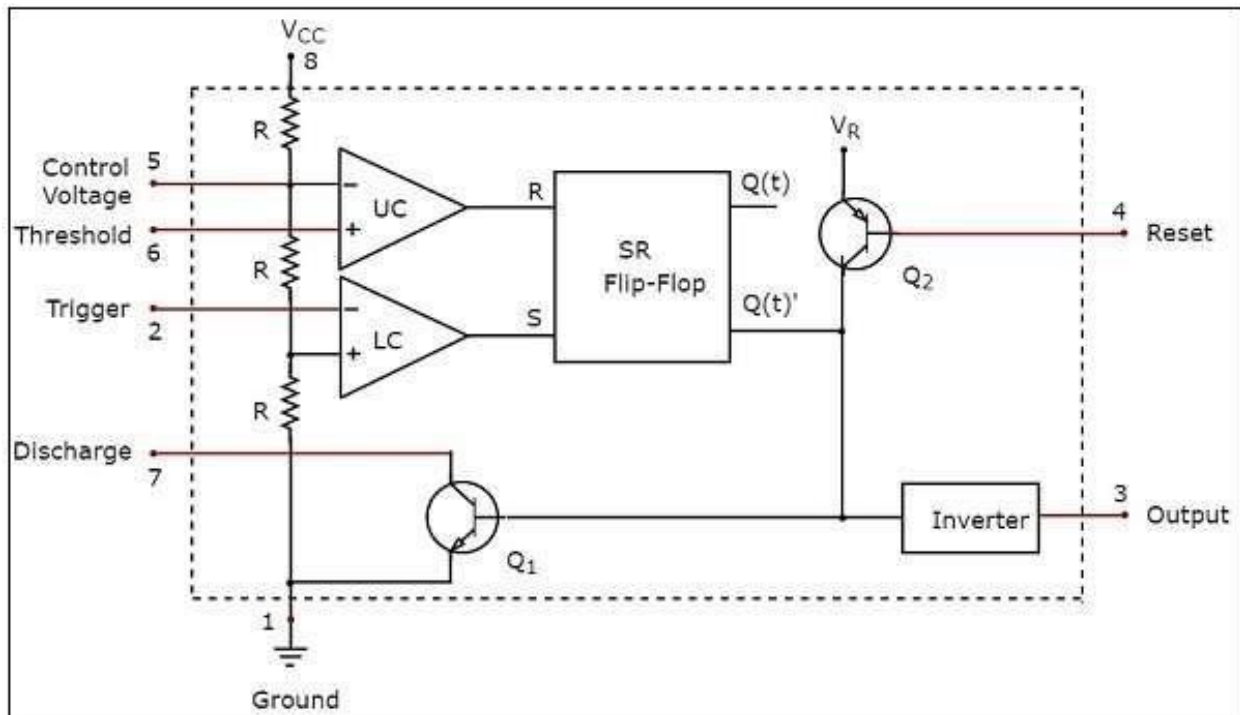


Fig 2.10: IC 555 Timer functional block diagram

Observe that the functional diagram of 555 Timer contains a voltage divider network, two comparators, one SR flip-flop, two transistors and an inverter. This section discusses about the purpose of each block or component in detail –

Voltage Divider Network:

1. The voltage divider network consists of a three $5K\Omega$ resistors that are connected in series between the supply voltage V_{cc} and ground.
2. This network provides a voltage of $V_{cc}/3$ between a point and ground, if there exists only one $5K\Omega$ resistor. Similarly, it provides a voltage of $2V_{cc}/3$ between a point and ground, if there exists only two $5K\Omega$ resistors.

Comparator:

1. The functional diagram of a 555 Timer IC consists of two comparators: An Upper Comparator (UC) and a Lower Comparator (LC).
2. Recall that a comparator compares the two inputs that are applied to it and produces an output.
3. If the voltage present at the non-inverting terminal of an op-amp is greater than the voltage present at its inverting terminal, then the output of comparator will be $+V_{sat}$. This can be considered as Logic High ('1') in digital representation.
4. If the voltage present at the non-inverting terminal of op-amp is less than or equal to the voltage at its inverting terminal, then the output of comparator will be $-V_{sat}$. This can be considered as Logic Low ('0') in digital representation.

SR Flip-Flop:

1. Recall that a SR flip-flop operates with either positive clock transitions or negative clock transitions. It has two inputs: S and R, and two outputs: Q(t) and Q(t)'. The outputs, Q(t) & Q(t)' are complement to each other.
2. The following table shows the state table of a SR flip-flop.

S	R	Q(t+1)
0	0	Q(t)
0	1	0
1	0	1
1	1	-

3. Here, Q(t) & Q(t+1) are present state & next state respectively. So, SR flip-flop can be used for one of these three functions such as Hold, Reset & Set based on the input conditions, when positive (negative) transition of clock signal is applied.
4. The outputs of Lower Comparator (LC) and Upper Comparator (UC) are applied as inputs of SR flip-flop as shown in the functional diagram of 555 Timer IC.

Transistors and Inverter:

1. The functional diagram of a 555 Timer IC consists of one NPN transistor Q1 and one PNP transistor Q2. The NPN transistor Q1 will be turned ON if its base to emitter voltage is positive and greater than cut-in voltage. Otherwise, it will be turned-OFF.
2. The PNP transistor Q2 is used as buffer in order to isolate the reset input from SR flip-flop and NPN transistor Q1.

3. The inverter used in the functional diagram of a 555 Timer IC not only performs the inverting action but also amplifies the power level.

2.7.3 Applications:

555 timers are most important integrated circuit (chip) used widely in digital electronics. Some common uses and application of 555 timer IC are as follow:

1. PWM (Pulse Width Modulation) & PPM (Pulse Position Modulation)
2. Duty Cycle Oscillator
3. Lamp Dimmer
4. To provide Accurate time delays
5. As a flip-flop element
6. Analog frequency meters
7. Pulse waveform and square wave generation
8. Tachometers & temperature measurement
9. It can be used as monostable and astable multivibrator
10. DC to DC Converters
11. DC Voltage Regulators
12. Voltage to Frequency Converter
13. Frequency Divider
14. Schmitt trigger
15. Pulse detector
16. Wiper speed control
17. Timer Switch

The 555 Timer IC are widely used in most of interesting electronic circuits and project like Traffic Light Circuit using 555 Timer, LED Flashing circuits, police siren, LED dice, Music Box, Metal detector, Joystick and game paddles, & low cost line receiver, Clap switch activated circuit and lots of other projects and circuits designs.

UNIT – 3

AMPLITUDE MODULATION

3.1 INTRODUCTION

The word communication arises from the Latin word *commūnicāre*, which means “to share”. Communication is the basic step for exchange of information.

For example, a baby in a cradle, communicates with a cry when she needs her mother. A cow moos loudly when it is in danger. A person communicates with the help of a language. Communication is the bridge to share.

Communication can be defined as the process of exchange of information through means such as words, actions, signs, etc., between two or more individuals.

3.2 TYPES OF SIGNALS

Conveying an information by some means such as gestures, sounds, actions, etc., can be termed as signaling. Hence, a signal can be a source of energy which transmits some information. This signal helps to establish a communication between the sender and the receiver.

An electrical impulse or an electromagnetic wave which travels a distance to convey a message, can be termed as a signal in communication systems. Depending on their characteristics, signals are mainly classified into two types: Analog and Digital. Analog and Digital signals are further classified, as shown in the following fig 3.1.

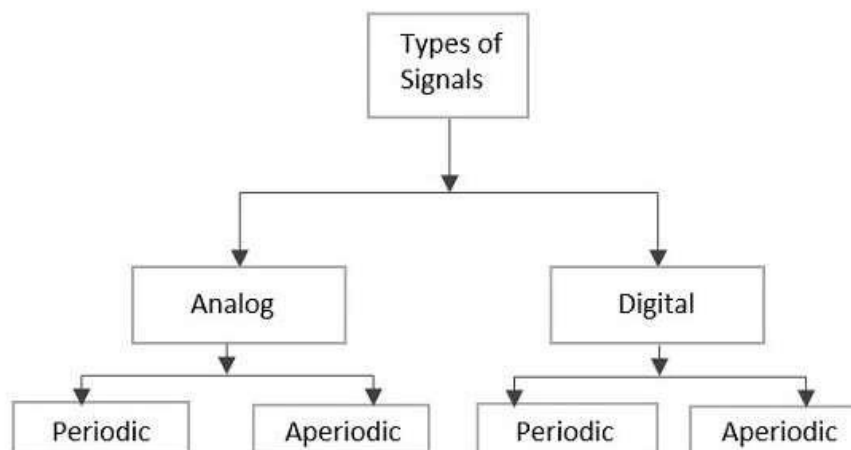


Fig 3.1 Types of signals

Analog Signal: A continuous time varying signal, which represents a time varying quantity can be termed as an Analog Signal. This signal keeps on varying with respect to time, according to the instantaneous values of the quantity, which represents it.

Example

Let us consider a tap that fills a tank of 100 liters' capacity in an hour (6 AM to 7 AM). The portion of filling the tank is varied by the varying time. Which means, after 15 minutes (6:15 AM) the quarter portion of the tank gets filled, whereas at 6:45 AM, 3/4th of the tank is filled.

If we try to plot the varying portions of water in the tank according to the varying time, it would look like the following fig 3.2.

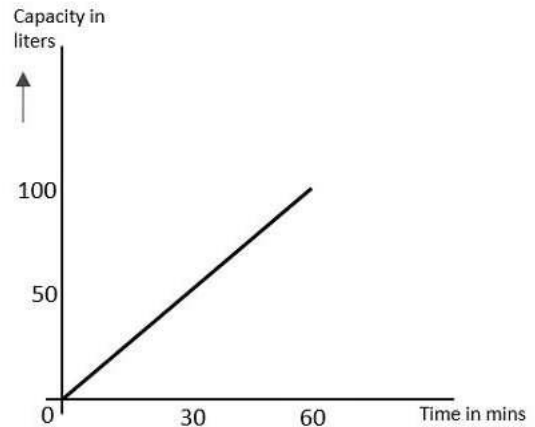


Fig 3.2: Water level variations in a tank with respect to time

As the result shown in this image varies (increases) according to time, this time varying quantity can be understood as Analog quantity. The signal which represents this condition with an inclined line in the figure, is an Analog Signal. The communication based on analog signals and analog values is called as Analog Communication.

Digital Signal: A signal which is discrete in nature or which is non-continuous in form can be termed as a Digital signal. This signal has individual values, denoted separately, which are not based on the previous values, as if they are derived at that particular instant of time.

Example

Let us consider a classroom having 20 students. If their attendance in a week is plotted, it would look like the following figure 3.3.

In this figure, the values are stated separately. For instance, the attendance of the class on Wednesday is 20 whereas on Saturday is 15. These values can be considered individually and separately or discretely, hence they are called as discrete values.

The binary digits which has only 1s and 0s are mostly termed as digital values. Hence, the signals which represent 1s and 0s are also called as digital signals. The communication based on digital signals and digital

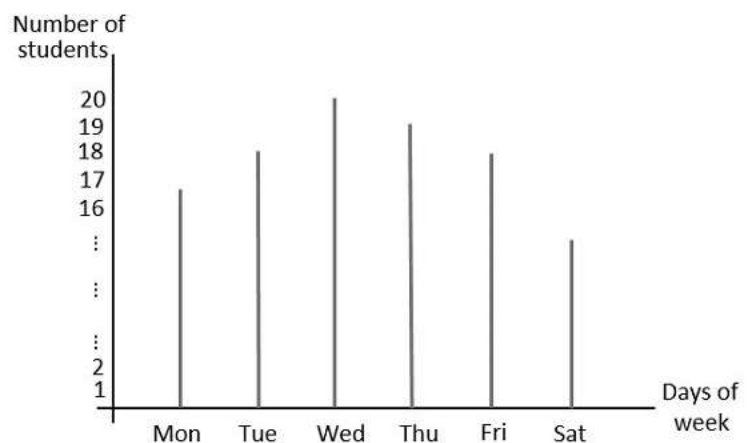


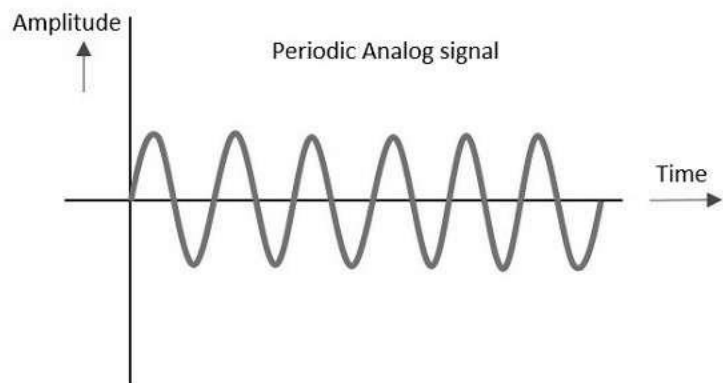
Fig 3.3: Digital signal

values is called as Digital Communication.

Periodic Signal: Any analog or digital signal, that repeats its pattern over a period of time, is called as a Periodic Signal. This signal has its pattern continued repeatedly and is easy to be assumed or to be calculated.

Example

If we consider a machinery in an industry, the process that takes place one after the other is a continuous procedure. For example, procuring and grading the raw material, processing the material in batches, packing a load of products one after the other, etc., follows a certain procedure repeatedly.

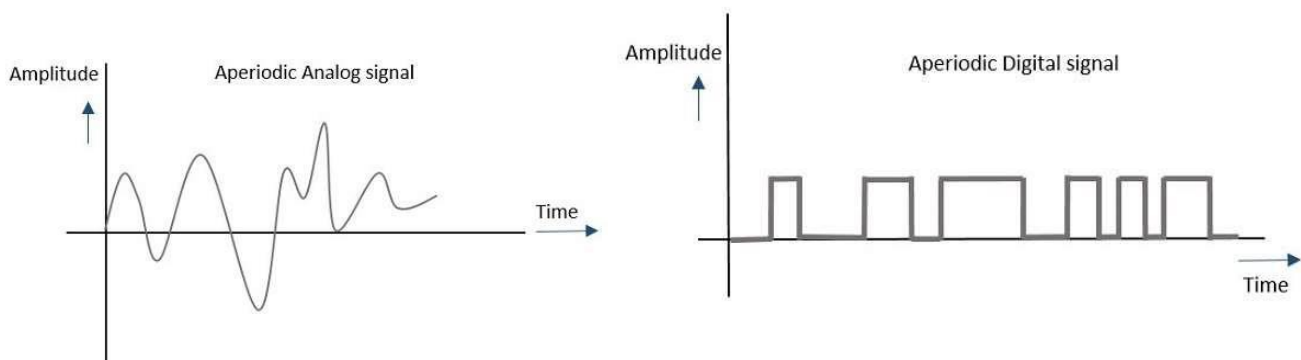


Such a process whether considered analog or digital, can be graphically represented as above.

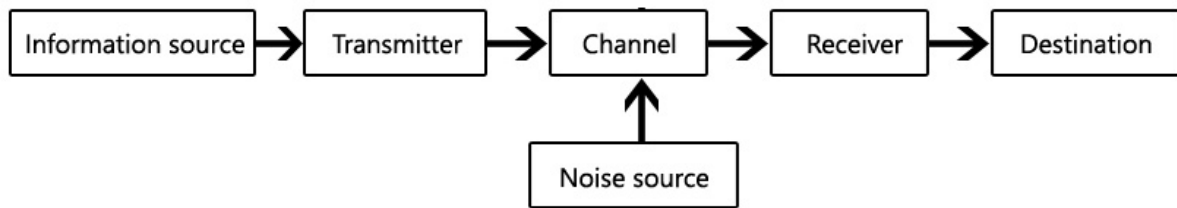
Aperiodic Signal: Any analog or digital signal, that doesn't repeat its pattern over a period of time is called as Aperiodic Signal. This signal has its pattern continued but the pattern is not repeated. It is also not so easy to be assumed or to be calculated.

Example

The daily routine of a person, if considered, consists of various types of work which take different time intervals for different tasks. The time interval or the work doesn't continuously repeat. For example, a person will not continuously brush his teeth from morning to night, that too with the same time period. Such a process whether considered analog or digital, can be graphically represented as follows.



In general, the signals which are used in communication systems are analog in nature, which are transmitted in analog or converted to digital and then transmitted, depending upon the requirement.

3.3 BASIC ELEMENTS (OR) COMPONENTS OF A COMMUNICATION SYSTEM**Fig 3.4 Components of a communication system**

Above is the basic block diagram of communication system. The basic components of a communication system are transmitter, communication channel or medium and transmitter.

Information source:

The various messages are in the form of words, group of word, code, symbol, sound signal etc. However, out of these messages one is selected & conveyed or communicated. It is used to produce required message which has to be transmitted.

Transmitter:

The transmitter is a collection of electronic circuits designed to convert the information into a signal suitable for transmission over a given communication medium.

Some message signals that comes from information source is non-electrical and it is not suitable for transmission. In-order to transmit the signal, it should be converted into electrical. The built-in circuitry such as decoders, encoders and transducers etc., in the transmitter makes incoming information suitable for transmission and subsequent reception.

Communication channel:

The communication channel is the medium by which the electronic signal is transmitted from one place to another. Depending on the type of medium, the communication system can be classified as

1. Wired communication (or) line communication
2. Wireless (or) radio communication

Noise:

Noise is an unwanted random signal which interfere with the message signal launched by the transmitter. This problem is particularly common in broadcasting, where two or more signals may be picked up at the same time by the receiver which in-turn will produce different message.

Noise in communication system can be classified as internal noise and external noise. Noise generated by components within a communication system such as resistor, diode and transistors are referred to as internal noise. The external noise results from resources outside a communication system such as atmospheric and man-made.

Receiver:

A receiver is a collection of electronic circuits designed to convert the signal back to the original information. It consists of amplifiers, decoders, mixer, oscillator, transducer and so on. The output transducer converts the electrical message signal into its original form.

3.4 MODULATION

For a signal to be transmitted to a distance, without the effect of any external interferences or noise addition and without getting faded away, it has to undergo a process called as **Modulation**. It improves the strength of the signal without disturbing the parameters of the original signal.

A message carrying a signal has to get transmitted over a distance and for it to establish a reliable communication, it needs to take the help of a high frequency signal which should not affect the original characteristics of the message signal.

The characteristics of the message signal, if changed, the message contained in it also alters. Hence, it is a must to take care of the message signal. A high frequency signal can travel up to a longer distance, without getting affected by external disturbances. We take the help of such high frequency signal which is called as a carrier signal to transmit our message signal. Such a process is simply called as Modulation.

Definition:

Modulation is the process of changing the characteristics (amplitude, frequency, phase) of the carrier signal (high frequency signal) in accordance with the instantaneous amplitude values of the message signal (modulating signal or low frequency signal).

Signals in modulation process:

Following are the three types of signals in the modulation process.

1. **Message or Modulating Signal:** The signal which contains a message to be transmitted, is called as a message signal. It is a baseband signal, which has to undergo the process of modulation, to get transmitted. Hence, it is also called as the modulating signal.
2. **Carrier Signal:** The high frequency signal, which has a certain amplitude, frequency and phase but contains no information is called as a carrier signal. It is an empty signal and is used to carry the signal to the receiver after modulation.
3. **Modulated Signal:** The resultant signal after the process of modulation is called as a modulated signal. This signal is a combination of modulating signal and carrier signal.

3.5 NEED FOR MODULATION

The need for modulation (or) advantages of modulation is given below:

1. Reduces the height of antenna.
2. Avoids mixing of signals.
3. Increases the range of communication.
4. Allows multiplexing of signals.
5. Improved quality of reception.

Reduces the height of antenna:

The height of antenna required for transmission and reception of radio waves in radio transmission is a function of wavelength of frequency used. The maximum height of the antenna required for transmission should be

$$h = \frac{\lambda}{4}$$

In addition $\lambda = \frac{C}{f}$, where C is the velocity of light and f is the frequency.

The above equation states that, at low frequencies wavelength of the signal is high and the height of antenna also be high.

For example, consider a signal with frequency $f = 15\text{K Hz}$

$$\text{Height of the antenna } h = \frac{\lambda}{4} = \frac{C}{4f} = \frac{3 \times 10^8}{4 \times 15 \text{KHz}} = 5000\text{m} = 5\text{Km}$$

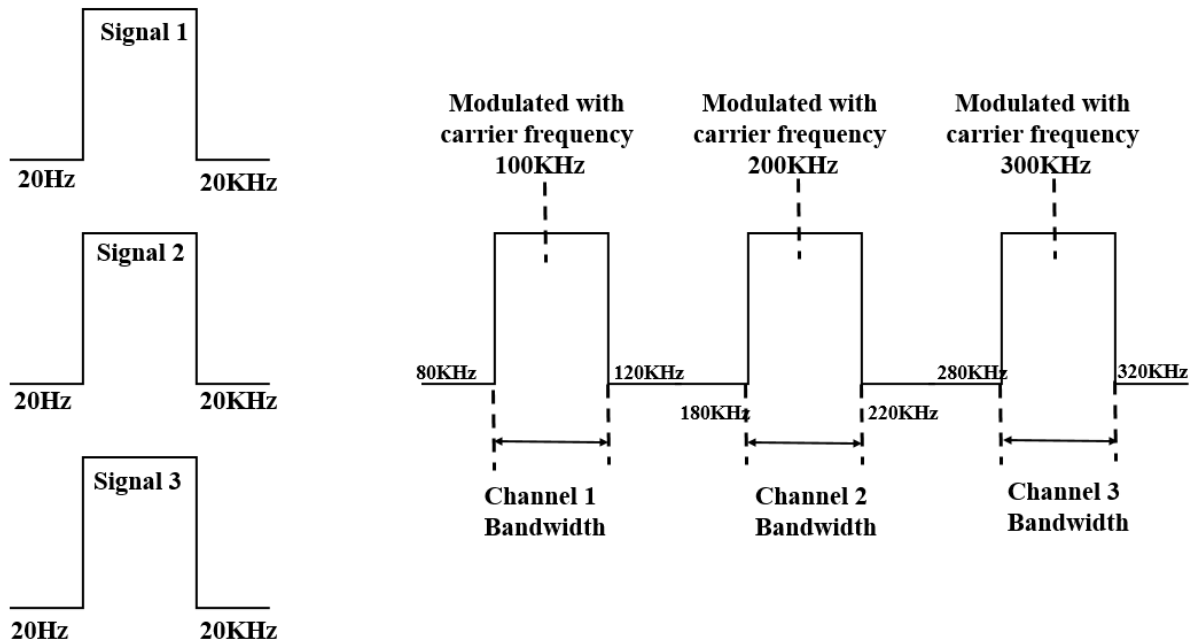
Now consider the same message signal is modulated with high frequency signal $f = 1\text{Mz}$

$$\text{Height of the antenna is } h = \frac{\lambda}{4} = \frac{C}{4f} = \frac{3 \times 10^8}{4 \times 1 \times 10^6 \text{Hz}} = \frac{300}{4} = 75\text{m}$$

Antenna height of 75m is practically possible and can be installed.

Avoids mixing of signals

All sound signals are concentrated within the range from 20Hz to 20KHz. There is a possibility of mixing of signals with same frequency range during transmission and it is difficult to separate at the receiver end. This problem can be overcome by modulating each signal with different carrier frequencies and then transmitted, at the receiving end a tuned circuit selects the desired signal.

**Increases the range of communication**

At low frequencies radiation is poor and signal gets highly attenuated. Therefore, baseband signals cannot be transmitted directly over long distances. Modulation effectively increases the frequency of the signal to be radiated and thus increases the distance over which signals can be transmitted faithfully.

Allows multiplexing of signals

Multiplexing means transmission of two or more signals simultaneously over the same channel. The different signals from different stations can be separated in the receiver since carrier frequencies for these signals are different. It is commonly known as tuning the receiver to the desired station. By tuning process, the desired signal is selected and at the same time, other unknown signals are rejected.

Improves quality of reception

The signal communication using modulation techniques such as amplitude, frequency, pulse modulation reduce the effect of the noise to great extent. Reduction in noise improves the quality of reception.

3.6 TYPES OF MODULATION

There are many types of modulations. Depending upon the modulation techniques used, they are classified as shown in the following figure.

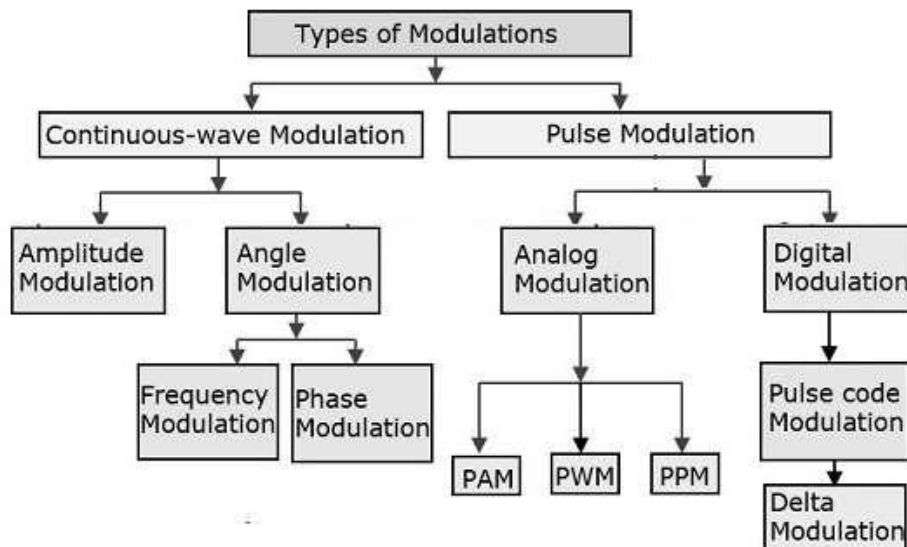


Fig. 3.5 Types of modulation

The types of modulations are broadly classified into continuous-wave modulation and pulse modulation.

Continuous-wave Modulation

In continuous-wave modulation, a high frequency sine wave is used as a carrier wave. This is further divided into amplitude and angle modulation.

1. If the amplitude of the high frequency carrier wave is varied in accordance with the instantaneous amplitude of the modulating signal, then such a technique is called as Amplitude Modulation.
2. If the angle of the carrier wave is varied, in accordance with the instantaneous value of the modulating signal, then such a technique is called as Angle Modulation. Angle modulation is further divided into frequency modulation and phase modulation.
 - If the frequency of the carrier wave is varied, in accordance with the instantaneous value of the modulating signal, then such a technique is called as Frequency Modulation.
 - If the phase of the high frequency carrier wave is varied, in accordance with the instantaneous value of the modulating signal, then such a technique is called as Phase Modulation.

Pulse Modulation

In Pulse modulation, a periodic sequence of rectangular pulses is used as a carrier wave. This is further divided into analog and digital modulation.

In analog modulation technique, if the amplitude or duration or position of a pulse is varied in accordance with the instantaneous values of the baseband modulating signal, then such a technique is called as Pulse Amplitude Modulation (PAM) or Pulse Duration/Width Modulation (PDM/PWM), or Pulse Position Modulation (PPM).

In digital modulation, the modulation technique used is Pulse Code Modulation (PCM) where the analog signal is converted into digital form of 1s and 0s. As the resultant is a coded pulse train, this is called as PCM. This is further developed as Delta Modulation (DM). These digital modulation techniques are discussed in Digital Communications.

3.7 AMPLITUDE MODULATION**Definition:**

According to the standard definition, “The amplitude of the carrier signal varies in accordance with the instantaneous amplitude of the modulating signal.” Which means, the amplitude of the carrier signal containing no information varies as per the amplitude of the signal containing information, at each instant. The frequency of the carrier signal remains constant during the modulation process but only the amplitude of the carrier signal varies in accordance with the message signal.

Time-domain Representation

The Instantaneous value of the message signal is

$$m(t) = A_m \cos(2\pi f_m t)$$

The Instantaneous value of the carrier signal is

$$c(t) = A_c \cos(2\pi f_c t)$$

Where, A_m is the maximum amplitude of the modulating signal.

A_c is the maximum amplitude of the carrier signal.

f_m is the frequency of the modulating signal.

f_c is the frequency the carrier signal.

The amplitude modulated signal is given by

$$S(t) = [A_c + m(t)] \cos(2\pi f_c t) \text{ ----- (1)}$$

$$S(t) = [A_c + A_m \cos(2\pi f_m t)] \cos(2\pi f_c t)$$

$$s(t) = A_c \left[1 + \frac{A_m}{A_c} \cos(2\pi f_m t) \right] \cos(2\pi f_c t) \text{ ----- (2)}$$

Modulation Index, μ or m :

A carrier wave, after being modulated, if the modulated level is calculated, then such an attempt is called as Modulation Index or Modulation Depth or Degree of Modulation. Modulation index is defined as the ratio of maximum amplitude of the message signal to the maximum amplitude of the carrier signal.

Mathematically, we can write it as

$$\text{Modulation index, } \mu = m = \frac{A_m}{A_c} = k_a A_m ; 0 < \mu \leq 1 \text{ ----- (3)}$$

Where, $K_a = \frac{1}{A_c}$ is a constant called the amplitude sensitivity of the modulator responsible for the generation of the modulated signal.

Substitute equation (3) in equation (2), we get

$$s(t) = A_c [1 + \mu \cos(2\pi f_m t)] \cos(2\pi f_c t) \text{ ----- (4)}$$

We can rearrange the modulated signal as;

$$s(t) = A_c [1 + K_a A_m \cos(2\pi f_m t)] \cos(2\pi f_c t)$$

$$s(t) = A_c [1 + K_a m(t)] \cos(2\pi f_c t)$$

This can be well explained by the following figures.

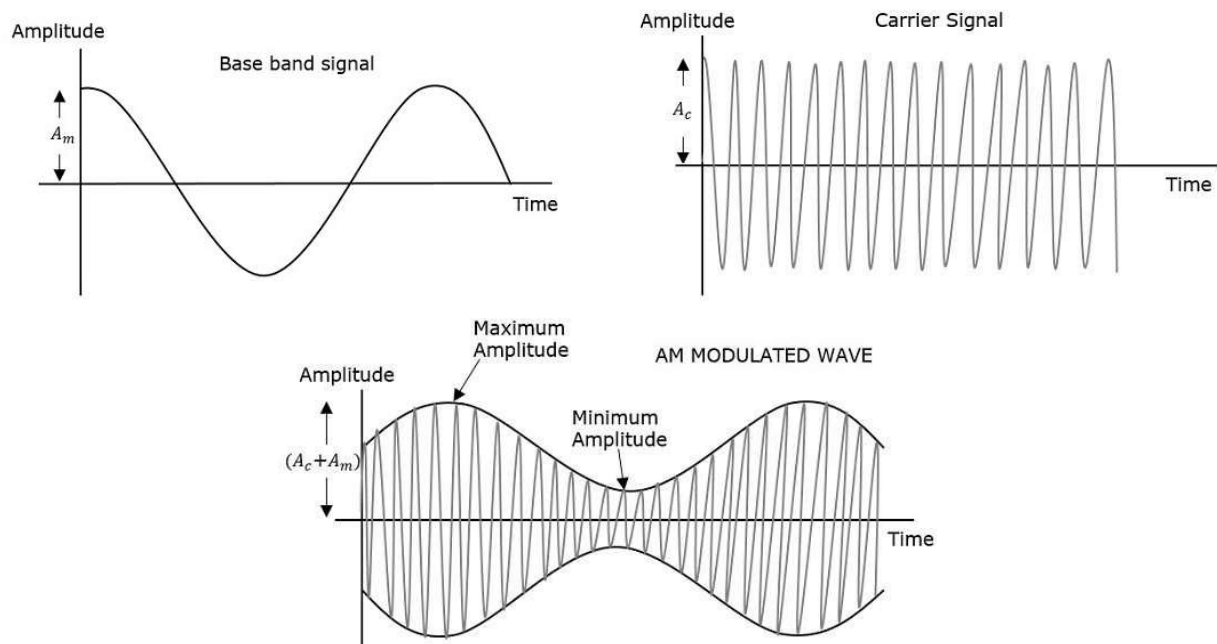


Fig 3.6: Time Domain Representation Waveforms of Amplitude Modulation

The first figure shows the modulating wave, which is the message signal. The next one is the carrier wave, which is a high frequency signal and contains no information. While, the last one is the resultant modulated wave. It can be observed that the positive and negative peaks of the carrier wave are interconnected with an imaginary line. This line helps recreating the exact shape of the modulating signal. This imaginary line on the carrier wave is called as Envelope. It is the same as that of the message signal.

One more formula for Modulation index:

Now, let us derive one more formula for Modulation index by considering Equation 1.

$$S(t) = [A_c + A_m \cos(2\pi f_m t)] \cos(2\pi f_c t)$$

We can use this formula for calculating modulation index value, when the maximum and minimum amplitudes of the modulated wave are known.

Let A_{max} and A_{min} be the maximum and minimum amplitudes of the modulated wave. We will get the maximum amplitude of the modulated wave, when $\cos(2\pi f_m t)$ is 1.

$$A_{max} = A_c + A_m$$

We will get the minimum amplitude of the modulated wave, when $\cos(2\pi f_m t)$ is -1.

$$A_{min} = A_c - A_m$$

Add Equations for A_{max} and A_{min} .

$$A_{max} + A_{min} = A_c + A_m + A_c - A_m = 2A_c$$

$$A_c = \frac{A_{max} + A_{min}}{2}$$

Subtract Equations for A_{max} and A_{min} .

$$A_{max} - A_{min} = A_c + A_m - (A_c - A_m) = 2A_m$$

$$\Rightarrow A_m = \frac{A_{max} - A_{min}}{2}$$

We know that the ratio of A_m and A_c gives modulation index.

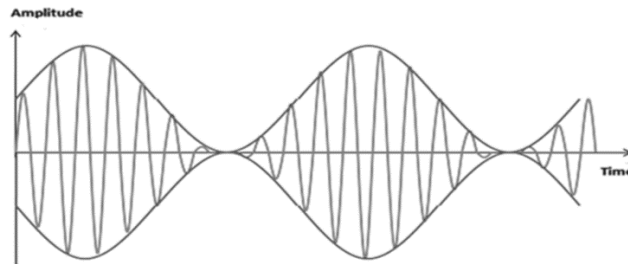
$$\frac{A_m}{A_c} = \frac{(A_{max} - A_{min})/2}{(A_{max} + A_{min})/2}$$

$$\text{Modulation Index } \mu = m = \frac{A_m}{A_c} = \frac{(A_{max} - A_{min})}{(A_{max} + A_{min})}$$

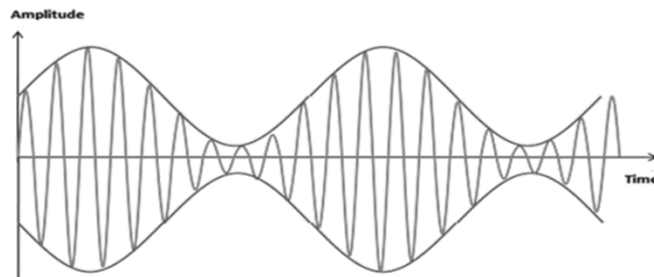
Percentage of Modulation: Modulation index or modulation depth is often denoted in percentage called as Percentage of Modulation. We will get the percentage of modulation, just by multiplying the modulation index value with 100.

$$\% \mu = \% m = \frac{A_m}{A_c} * 100$$

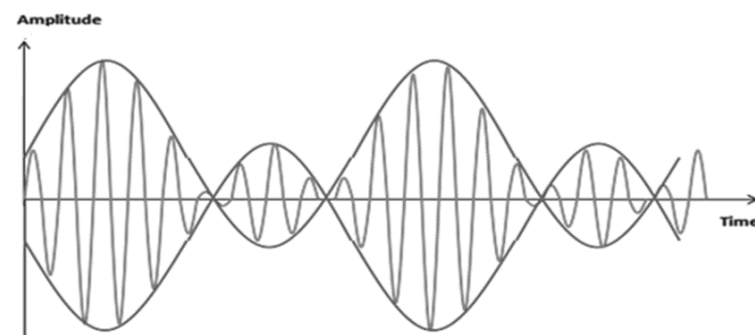
1. For Perfect modulation ($\mu = 1$), the value of modulation index should be equal to '1' i.e., $\mu = '1'$ then $A_m = A_c$, which implies the percentage of modulation i.e., $\% \mu = 100\%$.



2. For Under-modulation ($\mu < 1$), the value of modulation index is less than 1, i.e. $\mu < '1'$ then $A_m < A_c$, For Example $A_m = 1V$ & $A_c = 2V$, then modulation index is 0.5, which implies the percentage of modulation i.e. $\% \mu = 50\%$.



3. For Over-modulation ($\mu > 1$), the value of modulation index is greater than 1, i.e. $\mu > '1'$ then $A_m > A_c$, For Example $A_m = 1.5V$ & $A_c = 1V$, then modulation index is 1.5, which implies the percentage of modulation i.e. $\% \mu = 150\%$.



Frequency-domain Representation

Consider the following equation of amplitude modulated wave.

$$s(t) = A_c [1 + \mu \cos(2\pi f_m t)] \cos(2\pi f_c t)$$

$$\Rightarrow s(t) = A_c \cos(2\pi f_c t) + A_c \mu \cos(2\pi f_c t) \cos(2\pi f_m t)$$

$$\Rightarrow s(t) = A_c \cos(2\pi f_c t) + \frac{A_c \mu}{2} \cos[2\pi(f_c + f_m)t] + \frac{A_c \mu}{2} \cos[2\pi(f_c - f_m)t]$$

Hence, the amplitude modulated wave has three frequencies. Those are carrier frequency f_c , upper sideband frequency $f_c + f_m$ and lower sideband frequency $f_c - f_m$.

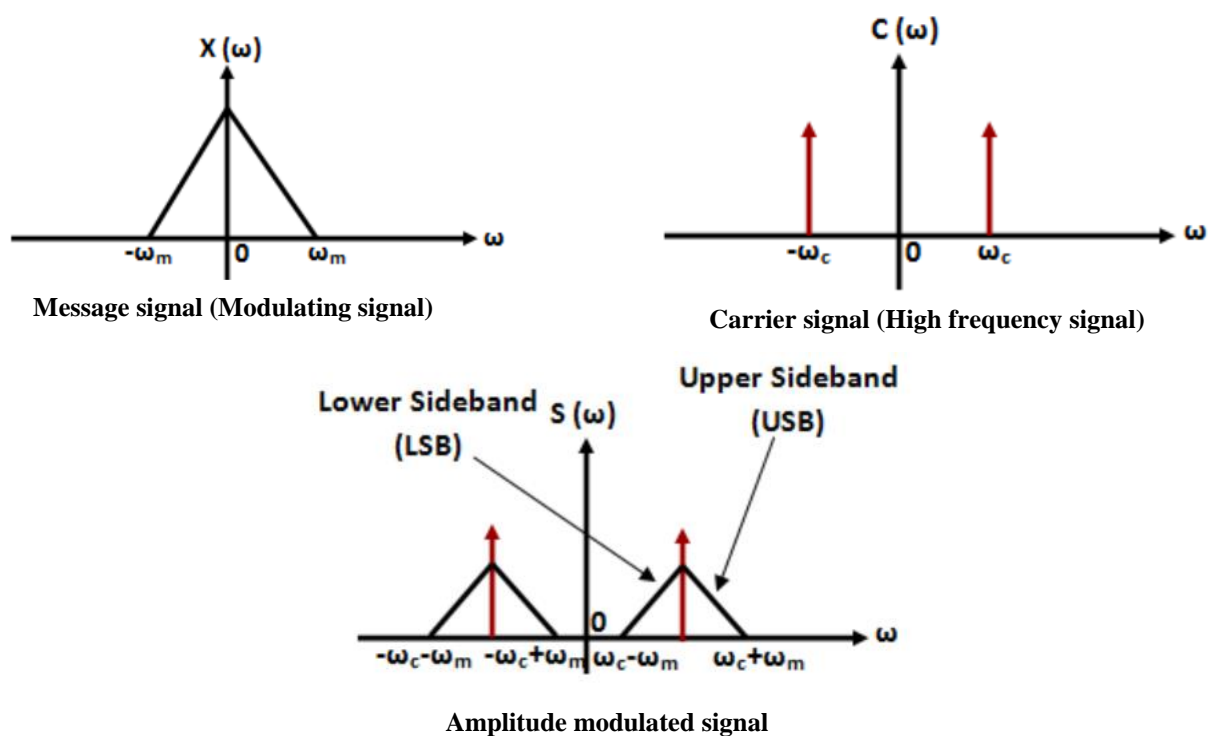


Fig 3.7: Frequency Domain Representation Waveforms of Amplitude Modulation

Bandwidth of AM wave:

Bandwidth (BW) is the difference between the highest and lowest frequencies of the signal. Mathematically, we can write it as

$$BW = f_{max} - f_{min}$$

Substitute, f_{max} and f_{min} values in bandwidth formula.

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

$$\Rightarrow BW = 2f_m$$

Thus, it can be said that the bandwidth required for amplitude modulated wave is twice the frequency of the modulating signal.

3.8 POWER RELATIONS IN AM WAVE

Consider the following equation of amplitude modulated wave.

$$s(t) = A_c \cos(2\pi f_c t) + \frac{A_c \mu}{2} \cos[2\pi(f_c + f_m)t] + \frac{A_c \mu}{2} \cos[2\pi(f_c - f_m)t]$$

Power of AM wave is equal to the sum of powers of carrier, upper sideband, and lower sideband frequency components.

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

We know that the standard formula for power of cos signal is

$$P = \frac{v_{rms}^2}{R} = \frac{(v_m/\sqrt{2})^2}{R}$$

Where, v_{rms} is the RMS value of cos signal, v_m is the peak value of cos signal.

First, let us find the powers of the carrier, the upper and lower sideband one by one.

Carrier power

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

Upper sideband power

$$P_{USB} = \frac{(A_c \mu / 2\sqrt{2})^2}{R} = \frac{A_c^2 \mu^2}{8R}$$

Similarly, we will get the lower sideband power same as that of the upper side band power.

$$P_{LSB} = \frac{A_c^2 \mu^2}{8R}$$

Now, let us add these three powers in order to get the power of AM wave.

$$\begin{aligned} P_t &= \frac{A_c^2}{2R} + \frac{A_c^2 \mu^2}{8R} + \frac{A_c^2 \mu^2}{8R} \\ \Rightarrow P_t &= \left(\frac{A_c^2}{2R}\right) \left(1 + \frac{\mu^2}{4} + \frac{\mu^2}{4}\right) \\ \Rightarrow P_t &= P_c \left(1 + \frac{\mu^2}{2}\right) \end{aligned}$$

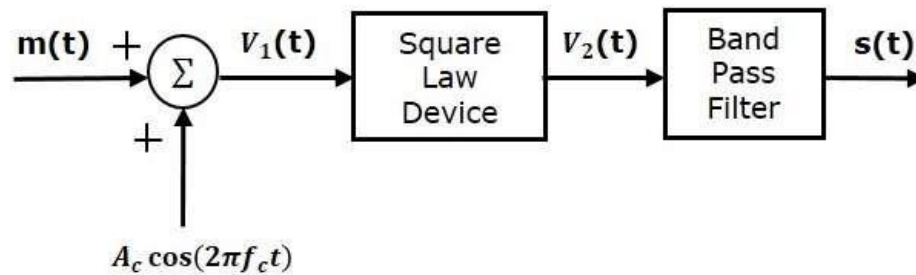
We can use the above formula to calculate the power of AM wave, when the carrier power and the modulation index are known.

3.9 GENERATION OF AM WAVES

Let us discuss about the modulators, which generate amplitude modulated wave. The following two modulators generate AM wave.

3.9.1 Square law modulator

Following is the block diagram of the square law modulator.



Let the modulating and carrier signals be denoted as $m(t)$ and $A_c \cos(2\pi f_c t)$ respectively. These two signals are applied as inputs to the summer (adder) block. This summer block produces an output, which is the addition of the modulating and the carrier signal.

Mathematically, we can write it as

$$V_1(t) = m(t) + A_c \cos(2\pi f_c t)$$

This signal $V_1(t)$ is applied as an input to a nonlinear device like diode. The characteristics of the diode are closely related to square law.

$$V_2(t) = k_1 V_1(t) + k_2 V_1^2(t)$$

Where, k_1 and k_2 are constants. Substitute $V_1(t)$ in the above equation $V_2(t)$,

$$\begin{aligned} V_2(t) &= k_1 [m(t) + A_c \cos(2\pi f_c t)] + k_2 [m(t) + A_c \cos(2\pi f_c t)]^2 \\ \Rightarrow V_2(t) &= k_1 m(t) + k_1 A_c \cos(2\pi f_c t) + k_2 m^2(t) + k_2 A_c^2 \cos^2(2\pi f_c t) + 2k_2 m(t) A_c \cos(2\pi f_c t) \\ \Rightarrow V_2(t) &= k_1 m(t) + k_2 m^2(t) + k_2 A_c^2 \cos^2(2\pi f_c t) + k_1 A_c \left[1 + \left(\frac{2k_2}{k_1} \right) m(t) \right] \cos(2\pi f_c t) \end{aligned}$$

The last term of the above equation represents the desired AM wave and the first three terms of the above equation are unwanted. So, with the help of band pass filter, we can pass only AM wave and eliminate the first three terms.

Therefore, the output of square law modulator is

$$s(t) = k_1 A_c \left[1 + \left(\frac{2k_2}{k_1} \right) m(t) \right] \cos(2\pi f_c t)$$

The standard equation of AM wave is

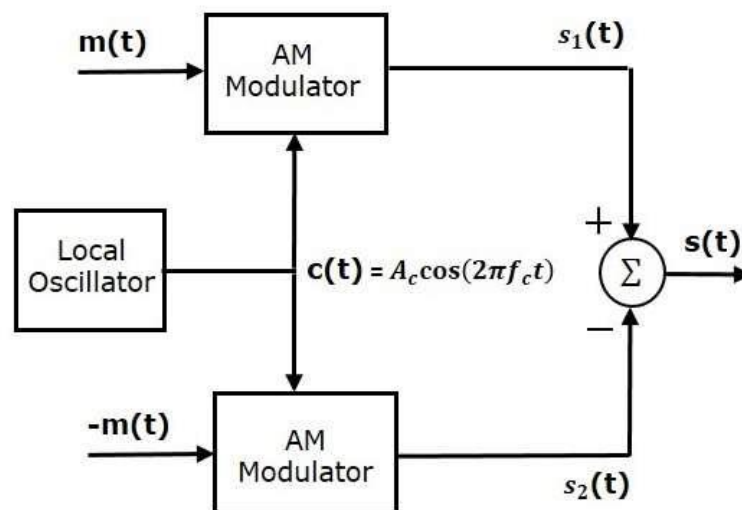
$$s(t) = A_c [1 + k_a m(t)] \cos(2\pi f_c t)$$

Where, k_a is the amplitude sensitivity.

By comparing the output of the square law modulator with the standard equation of AM wave, we will get the scaling factor as k_1 and the amplitude sensitivity k_a as $2K_2/K_1$.

3.9.2 Balanced modulator

Following is the block diagram of the balanced modulator.



Balanced modulator consists of two identical AM modulators. These two modulators are arranged in a balanced configuration in order to suppress the carrier signal. Hence, it is called as balanced modulator.

The same carrier signal $c(t) = A_c \cos(2\pi f_c t)$ is applied as one of the inputs to these two AM modulators. The modulating signal $m(t)$ is applied as another input to the upper AM modulator. Whereas, the modulating signal $m(t)$ with opposite polarity, i.e., $-m(t)$ is applied as another input to the lower AM modulator.

Output of the upper AM modulator is

$$s_1(t) = A_c [1 + k_a m(t)] \cos(2\pi f_c t)$$

Output of the lower AM modulator is

$$s_2(t) = A_c[1 - k_a m(t)] \cos(2\pi f_c t)$$

We get the DSBSC wave $s(t)$ by subtracting $s_2(t)$ from $s_1(t)$. The summer block is used to perform this operation. $s_1(t)$ with positive sign and $s_2(t)$ with negative sign are applied as inputs to summer block. Thus, the summer block produces an output $s(t)$ which is the difference of $s_1(t)$ and $s_2(t)$.

$$\Rightarrow s(t) = A_c[1 + k_a m(t)] \cos(2\pi f_c t) - A_c[1 - k_a m(t)] \cos(2\pi f_c t)$$

$$\Rightarrow s(t) = A_c \cos(2\pi f_c t) + A_c k_a m(t) \cos(2\pi f_c t) - A_c \cos(2\pi f_c t) + A_c k_a m(t) \cos(2\pi f_c t)$$

$$\Rightarrow s(t) = 2A_c k_a m(t) \cos(2\pi f_c t)$$

We know the standard equation of DSBSC wave is

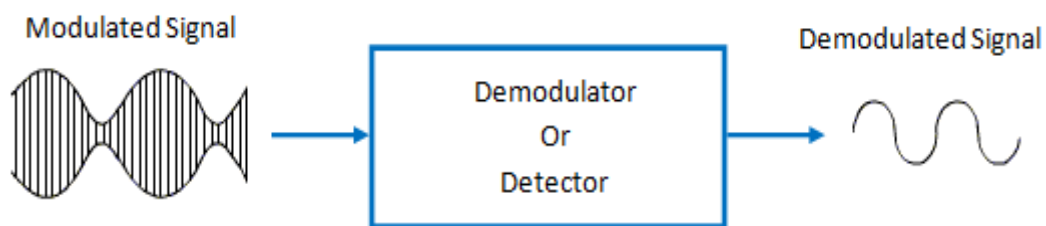
$$s(t) = A_c m(t) \cos(2\pi f_c t)$$

By comparing the output of summer block with the standard equation of DSBSC wave, we will get the scaling factor as $2k_a$.

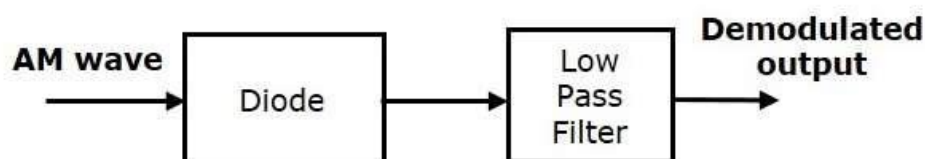
3.10 DETECTION (Demodulation) OF AM WAVES

3.10.1 Diode Detector or Envelop Detector

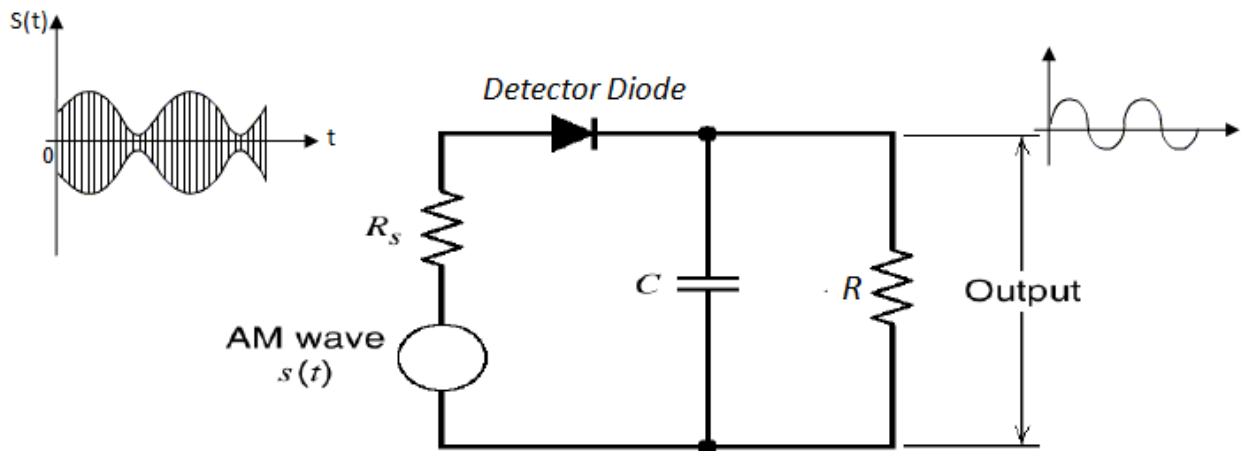
Demodulation: The process of recovering the message signal from the received modulated signal is known as demodulation. This process of detection is exactly opposite to that of modulation.



Envelope detector is used to detect (demodulate) high level AM wave. Following is the block diagram of the envelope detector.



This envelope detector consists of a diode and low pass filter. Here, the diode is the main detecting element. Hence, the envelope detector is also called as the diode detector. The low pass filter contains a parallel combination of the resistor and the capacitor.

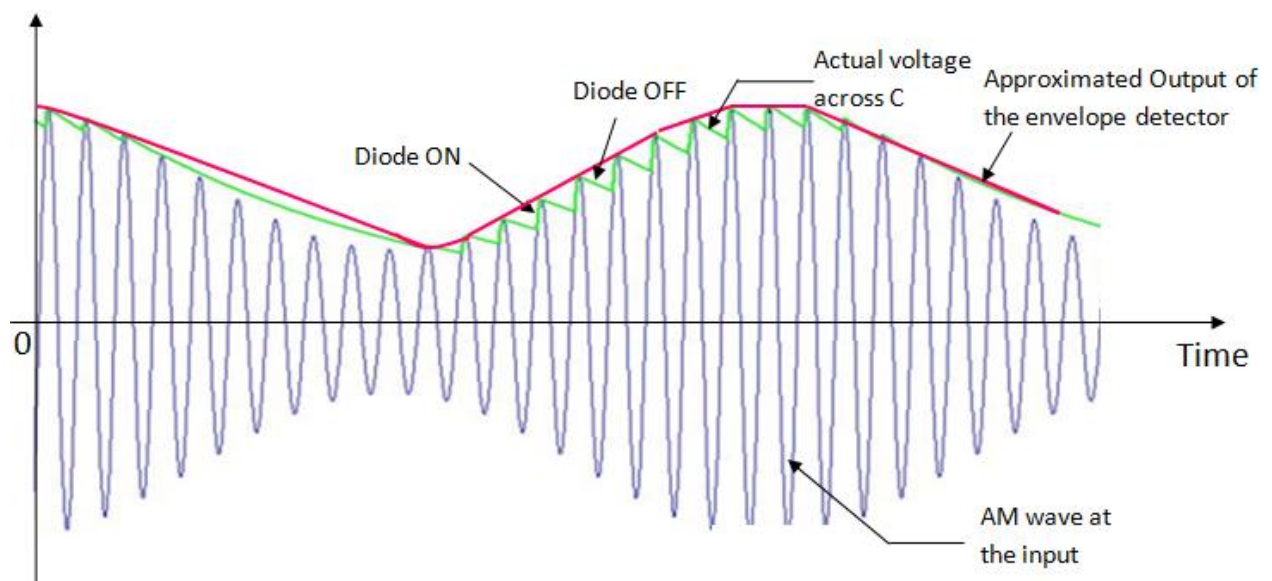


The AM wave $s(t)$ is applied as an input to this detector.

We know the standard form of AM wave is

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

In the positive half cycle of AM wave, the diode conducts and the capacitor charges to the peak value of AM wave. When the value of AM wave is less than this value, the diode will be reverse biased. Thus, the capacitor will discharge through resistor R till the next positive half cycle of AM wave. When the value of AM wave is greater than the capacitor voltage, the diode conducts and the process will be repeated.



UNIT – 4**FREQUENCY MODULATION****4.1 INTRODUCTION TO ANGLE MODULATION**

The other type of modulation in continuous-wave modulation is Angle Modulation. Angle Modulation is the process in which the frequency or the phase of the carrier signal varies according to the message signal.

The standard equation of the angle modulated wave is

$$s(t) = A_c \cos \theta_i(t)$$

Where,

A_c is the amplitude of the modulated wave, which is the same as the amplitude of the carrier signal.

$\theta_i(t)$ is the angle of the modulated wave.

Angle modulation is further divided into frequency modulation and phase modulation.

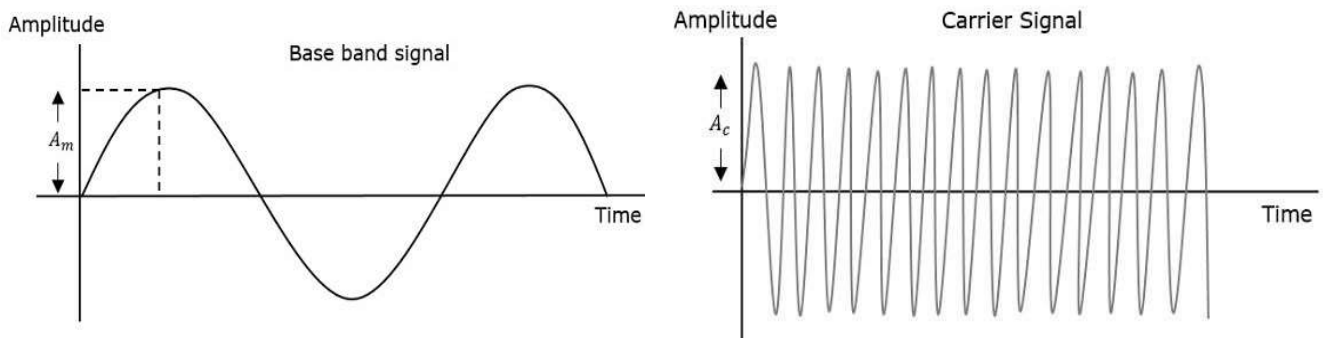
- **Frequency Modulation** is the process of varying the frequency of the carrier signal linearly with the message signal.
- **Phase Modulation** is the process of varying the phase of the carrier signal linearly with the message signal.

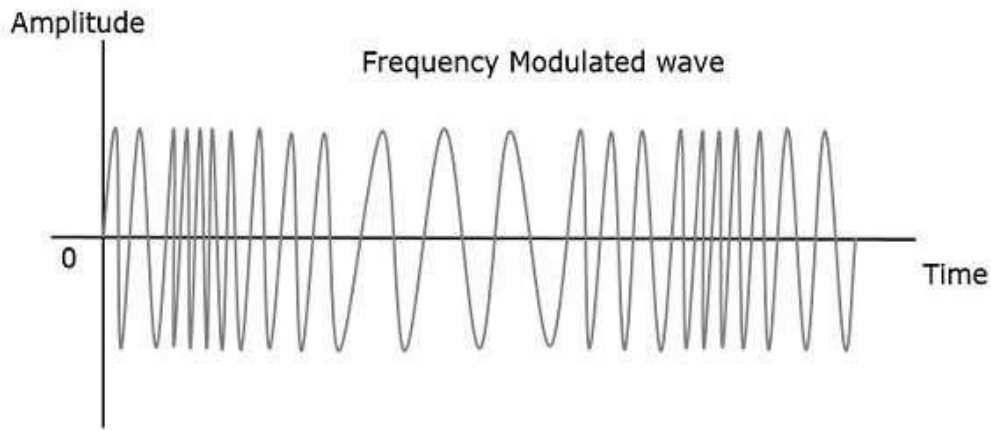
4.2 THEORY AND MATHEMATICAL REPRESENTATION OF FM

In amplitude modulation, the amplitude of the carrier signal varies. Whereas, in **Frequency Modulation (FM)**, the frequency of the carrier signal varies in accordance with the instantaneous amplitude of the modulating signal.

Hence, in frequency modulation, the amplitude and the phase of the carrier signal remains constant.

This can be better understood by observing the following figures.





The frequency of the modulated wave increases, when the amplitude of the modulating or message signal increases. Similarly, the frequency of the modulated wave decreases, when the amplitude of the modulating signal decreases. Note that, the frequency of the modulated wave remains constant and it is equal to the frequency of the carrier signal, when the amplitude of the modulating signal is zero.

Mathematical Representation

The equation for instantaneous frequency f_i in FM modulation is

$$f_i = f_c + k_f m(t)$$

Where,

f_c is the carrier frequency

k_f is the frequency sensitivity

$m(t)$ is the message signal

We know the relationship between angular frequency ω_i and angle $\theta_i(t)$ as

$$\begin{aligned}\omega_i &= \frac{d\theta_i(t)}{dt} \\ \Rightarrow 2\pi f_i &= \frac{d\theta_i(t)}{dt} \\ \Rightarrow \theta_i(t) &= 2\pi \int f_i dt\end{aligned}$$

Substitute, f_i value in the above equation.

$$\begin{aligned}\theta_i(t) &= 2\pi \int (f_c + k_f m(t)) dt \\ \Rightarrow \theta_i(t) &= 2\pi f_c t + 2\pi k_f \int m(t) dt\end{aligned}$$

Substitute, $\theta_i(t)$ value in the standard equation of angle modulated wave.

$$s(t) = A_c \cos\left(2\pi f_c t + 2\pi k_f \int m(t) dt\right)$$

This is the equation of FM wave.

If the modulating signal is $m(t) = A_m \cos(2\pi f_m t)$, then the equation of FM wave will be

$$s(t) = A_c \cos(2\pi f_c t + \beta \sin(2\pi f_m t))$$

Where,

$$\beta = \text{modulation index} = \frac{\Delta f}{f_m} = \frac{k_f A_m}{f_m}$$

The difference between FM modulated frequency (instantaneous frequency) and normal carrier frequency is termed as **Frequency Deviation**. It is denoted by Δf , which is equal to the product of k_f and A_m .

The deviation of the frequency of the carrier signal from high to low or low to high can be termed as the **Carrier Swing**.

$$\text{Carrier Swing} = 2 \times \text{frequency deviation} = 2 \times \Delta f$$

The **percent modulation** is defined as the ratio of the actual frequency deviation produced by the modulating signal to the maximum allowable frequency deviation.

Thus,

$$\% \text{ Modulation} = \frac{\text{Actual Frequency deviation}}{\text{Maximum allowed deviation}}$$

4.3 TYPES OF FM

FM can be divided into Narrowband FM and Wideband FM based on the values of modulation index β .

Narrowband FM

Following are the features of Narrowband FM.

- This frequency modulation has a small bandwidth when compared to wideband FM.
- The modulation index β is small, i.e., less than 1.
- Its spectrum consists of the carrier, the upper sideband and the lower sideband.
- This is used in mobile communications such as police wireless, ambulances, taxicabs, etc.

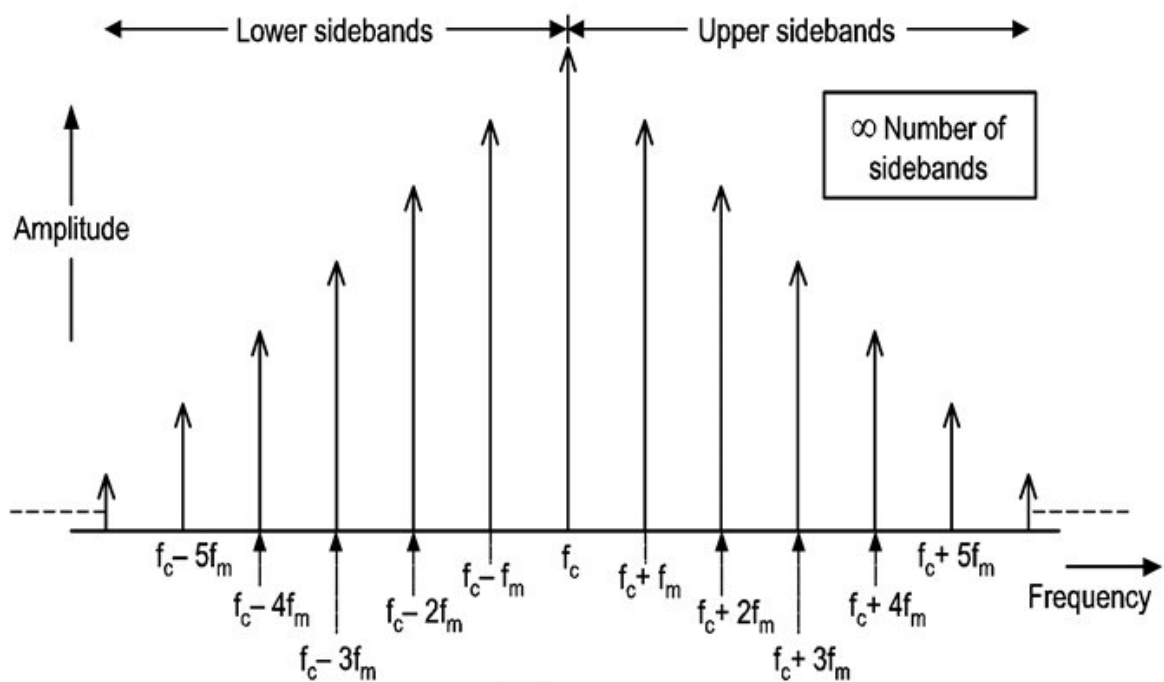
Wideband FM

Following are the features of Wideband FM.

- This frequency modulation has infinite bandwidth.
- The modulation index β is large, i.e., higher than 1.
- Its spectrum consists of a carrier and infinite number of sidebands, which are located around it.
- This is used in entertainment, broadcasting applications such as FM radio, TV, etc.

4.4 FREQUENCY SPECTRUM AND BANDWIDTH OF FM WAVES

Any modulation process produces sidebands. In FM, sum and difference sideband frequencies are produced. The spectrum of FM signal is usually wider than an equivalent AM signal. Although FM process produces an infinite number of upper and lower sidebands, only those with the largest amplitudes are significant in carrying the information. Typically, any sideband with an amplitude less than 1% of the unmodulated carrier is considered insignificant.



Knowing the modulation index, we can compute the number and amplitude of the significant sidebands. This is done through a complex mathematical process known as the Bessel functions. The spectrum of an FM signal varies considerably in bandwidth depending upon the modulation index. The higher the modulation index, the wider the bandwidth of the FM signal.

Bessel Functions

Modulation Index	Carrier	Sidebands															
		1	2	3	4	5	6	7	8	9	10	11	12	13	14	15	16
0.00	1.00	-	-	-	-	-	-	-	-	-	-	-	-	-	-	-	-
0.25	0.98	0.12	-	-	-	-	-	-	-	-	-	-	-	-	-	-	-
0.5	0.94	0.24	0.03	-	-	-	-	-	-	-	-	-	-	-	-	-	-
1.0	0.77	0.44	0.11	0.02	-	-	-	-	-	-	-	-	-	-	-	-	-
1.5	0.51	0.56	0.23	0.06	0.01	-	-	-	-	-	-	-	-	-	-	-	-
2.0	0.22	0.58	0.35	0.13	0.03	-	-	-	-	-	-	-	-	-	-	-	-
2.5	-0.05	0.50	0.45	0.22	0.07	0.02	-	-	-	-	-	-	-	-	-	-	-
3.0	-0.26	0.34	0.49	0.31	0.13	0.04	0.01	-	-	-	-	-	-	-	-	-	-
4.0	-0.40	-0.07	0.36	0.43	0.28	0.13	0.05	0.02	-	-	-	-	-	-	-	-	-
5.0	-0.18	-0.33	0.05	0.36	0.39	0.26	0.13	0.05	0.02	-	-	-	-	-	-	-	-
6.0	0.15	-0.28	-0.24	0.11	0.36	0.30	0.25	0.13	0.06	0.02	-	-	-	-	-	-	-
7.0	0.30	0.00	-0.03	-0.17	0.16	0.35	0.34	0.23	0.13	0.06	0.02	-	-	-	-	-	-
8.0	0.17	0.23	-0.11	-0.29	-0.10	0.19	0.34	0.32	0.22	0.13	0.06	0.03	-	-	-	-	-
9.0	-0.09	0.24	0.14	-0.18	-0.27	-0.06	0.20	0.33	0.30	0.21	0.12	0.06	0.03	0.01	-	-	-
10.0	-0.25	0.04	0.25	0.06	-0.22	-0.23	-0.01	0.22	0.31	0.29	0.20	0.12	0.06	0.03	0.01	-	-
12.0	-0.05	-0.22	-0.80	0.20	0.18	0.07	-0.24	-0.17	0.05	0.23	0.30	0.27	0.20	0.12	0.07	0.03	0.01
15.0	-0.01	0.21	0.04	0.19	-0.12	0.13	0.21	0.03	-0.17	-0.22	-0.09	0.10	0.24	0.28	0.25	0.18	0.12

Bandwidth

The total bandwidth can be determined by knowing modulation index and using the Bessel table.

The bandwidth can be determined using simple formula,

$$BW = 2 \times N \times f_{m(\max)}$$

Where N = Number of sidebands, $f_{m(\max)}$ = Maximum modulating frequency.

An alternative way to calculate the bandwidth of an FM signal is to use Carson's rule.

$$BW = 2 (f_{d(\max)} + f_{m(\max)})$$

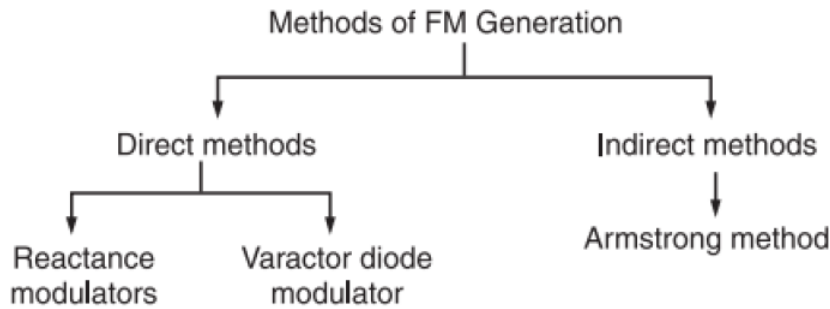
Where $f_{d(\max)}$ = maximum frequency deviation.

4.5 GENERATION OF FM SIGNALS

The FM modulator circuits used for generating FM signals can be divided into two categories such as:

- (i) The direct method or parameter variation method
- (ii) (ii) The Indirect method or the Armstrong method

The classification of FM generation methods is shown below:



4.5.1 The Direct Method or Parameter Variation Method

In direct method or parameter variation method, the baseband or modulating signal directly modulates the carrier. The carrier signal is generated with the help of an oscillator circuit. This oscillator circuit uses a parallel tuned L-C circuit. Thus the frequency of oscillation of the carrier generation is governed by the expression:

$$\omega_c = \frac{1}{\sqrt{LC}}$$

Now, we can make the carrier frequency ω_c to vary in accordance with the baseband or modulating signal $x(t)$ if L or C is varied according to $x(t)$. An oscillator circuit whose frequency is controlled by a modulating voltage is called voltage controlled oscillator (VCO). The frequency of VCO is varied according to the modulating signal simply by putting a shunt voltage variable capacitor with its tuned circuit.

4.5.2 Reactance Modulator

In direct FM generation shown in figure 4.1, the instantaneous frequency of the carrier is changed directly in proportion with the message signal. For this, a device called voltage controlled oscillator (VCO) is used. A VCO can be implemented by using a sinusoidal oscillator with a tuned circuit having a high value of Q.

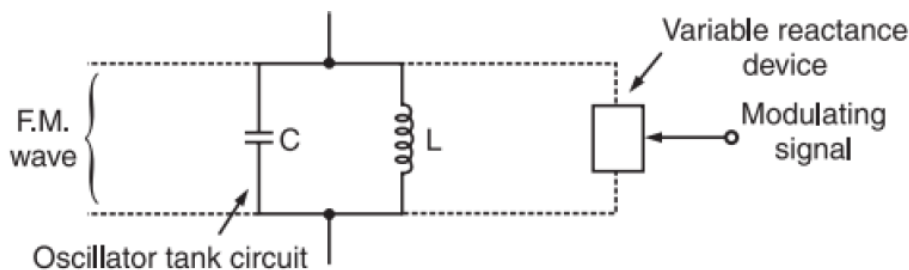


Figure.4.1 Principle of Reactance Modulator

The frequency of this oscillator is changed by changing the reactive components involved in the tuned circuit. If L or C of a tuned circuit of an oscillator is changed in accordance with the amplitude of modulating signal, then FM can be obtained across the tuned circuit as shown in figure 4.1 above.

A two or three terminal device is placed across the tuned circuit. The reactance of the device is varied proportional to modulating signal voltage. This will vary the frequency of the oscillator to produce FM. The devices used are FET, transistor or varactor diode. An example of direct FM is shown in figure 4.2 which uses a Hartley oscillator along with a varactor diode. The varactor diode is reverse biased. Its capacitance is dependent on the reverse voltage applied across it. This capacitance is shown by the capacitor C(t) in figure 4.2.

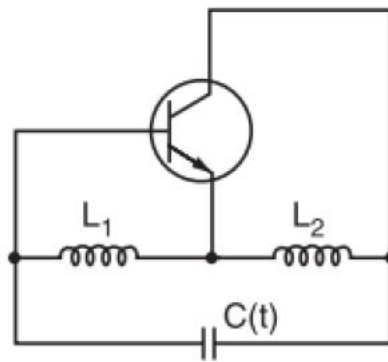


Figure.4.2 Hartley Oscillator

Frequency of oscillations of the Hartley oscillator is given by:

$$f_i(t) = \frac{1}{2\pi\sqrt{(L_1 + L_2)C(t)}}$$

where $C(t) = C + C(\text{varactor})$.

This means that C(t) is the effective capacitance of the fixed tuned circuit capacitance C and the varactor diode capacitance C(varactor).

Let the relation between the modulating voltage $x(t) = 0$ and the capacitance C(t) be represented as under:

$$C(t) = C - k_c x(t)$$

where C = total capacitance when $x(t)=0$.

k_c is the sensitivity of the varactor capacitance to change in voltage.

Substituting expression for C(t) in equation of $f_i(t)$, we get

$$f_i(t) = \frac{1}{2\pi\sqrt{(L_1 + L_2)(C - k_c x(t))}} = \frac{1}{2\pi\left[\sqrt{(L_1 + L_2)C - (L_1 + L_2)k_c x(t)}\right]}$$

or

$$f_i(t) = \frac{1}{2\pi\sqrt{(L_1 + L_2)C}\left[1 - \frac{k_c x(t)}{C}\right]^{1/2}}$$

But, let

$$\frac{1}{2\pi\sqrt{(L_1 + L_2)C}} = f_0$$

which is the oscillator frequency in absence of the modulating signal [$x(t) = 0$]. Therefore, we have,

$$f_i(t) = f_0 \left[1 - \frac{k_c}{C} x(t)\right]^{-1/2}$$

If the maximum change in the capacitance corresponding to the modulating wave is assumed to be small as compared to the unmodulated capacitance C , then the above equation for $f_i(t)$ can be approximated as under:

$$f_i(t) = f_0 \left[1 + \frac{k_c}{2C} x(t)\right]$$

or

$$f_i(t) = f_0 + \frac{f_0 k_c}{2C} \cdot x(t)$$

Now, let us define

$$\frac{f_0 k_c}{2C} = k_f$$

Therefore, we have

$$f_i(t) = f_0 + k_f x(t)$$

where k_f is called as the frequency sensitivity of the modulator.

4.6 DEMODULATION OF FM SIGNALS

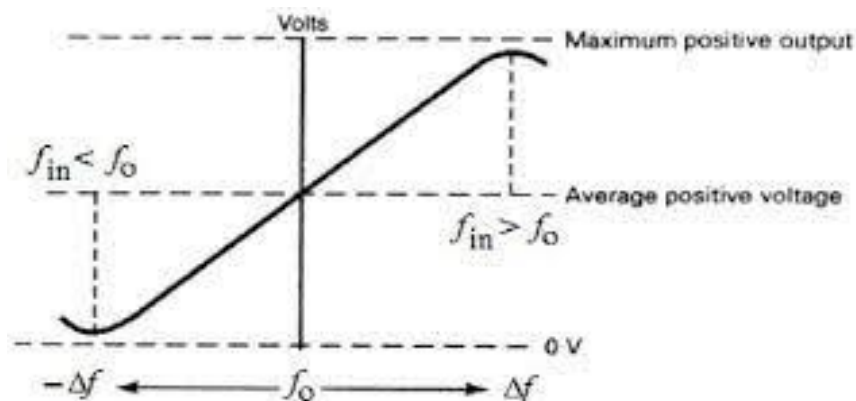
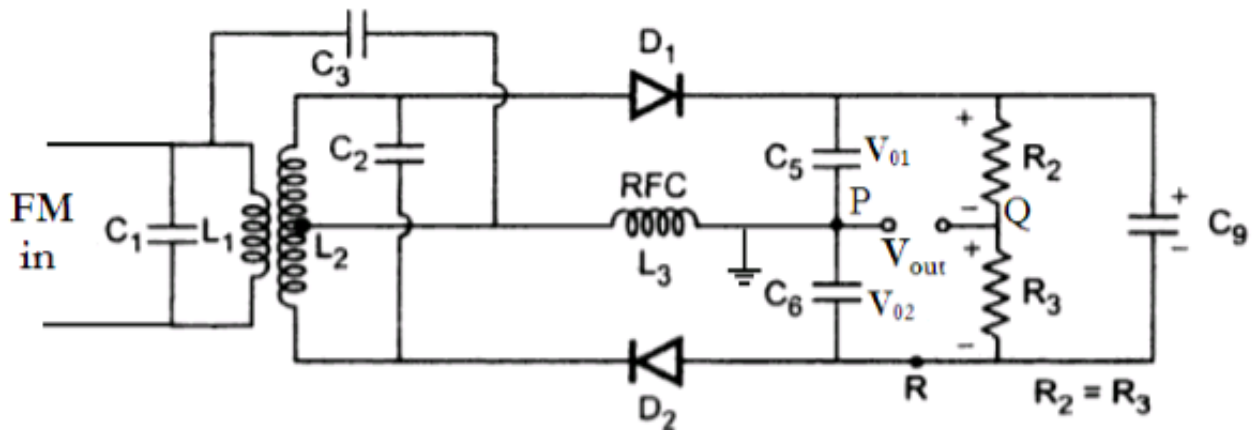
Demodulation process of FM waves is exactly opposite to that of frequency modulation.

FM demodulator is basically a frequency to amplitude convertor i.e., converting the frequency variations in FM wave at its input into amplitude variations at its output to recover original modulating signal.

Some requirements of FM demodulator / detector are:

1. It must convert frequency variations into amplitude variations.
2. Conversion must be linear.
3. Demodulator circuit must be insensitive to amplitude changes i.e. it must respond only to frequency variations.
4. Its operation and adjustment must not be critical.

4.6.1 RATIO DETECTOR



1. This detector demodulates FM signals and suppresses amplitude noise without the need of limiter stages.
2. The ratio detector uses to convert the instantaneous frequency variations into instantaneous amplitude variations.
3. C9 retains most of its charge because of the long time constant offered in combination with R2 and R3.
4. This slow charging and discharging helps to maintain the constant output.
5. The ratio detector is not affected by amplitude variations on the FM wave. The output of the detector adjusts itself automatically to the average amplitude of the input signal.

6. The polarity of voltage in the lower capacitor is reversed. Hence the voltages V_{D1} and V_{D2} across two capacitors add.
7. When $F_{in} = F_c$, at carrier frequency V_{D1} and V_{D2} are equal hence the net output of the discriminator will be zero.
8. When $F_{in} > F_c$, as the carrier moves off to one side of the carrier frequency the balance condition is destroyed. One diode conducts more than the other. This results in the voltage across one of the resistors being large than other. When input frequency increases above the carrier frequency (F_c) the phase shift between V_1 and V_2 reduces. $V_{D1} > V_{D2}$ hence $V_0 = V_{D1} - V_{D2}$ will be positive.
9. When $F_{in} < F_c$, input frequency reduces below F_c , the phase shift between V_1 and V_2 increases the $V_{D2} > V_{D1}$ hence $V_0 = V_{D1} - V_{D2}$ will be negative.

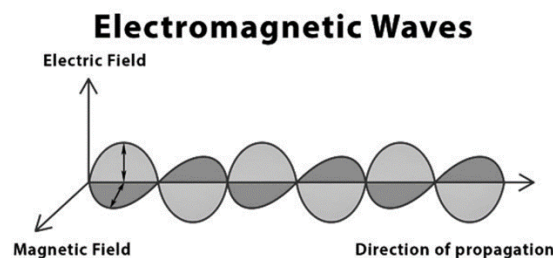
UNIT – 5**RADIO BROADCASTING AND RECEPTION****5.1 ELECTROMAGNETIC SPECTRUM**

The sun is our planet's principal source of energy, and its energy travels in the form of electromagnetic radiation. Electromagnetic energy moves across empty space at the speed of light in the form of waves of electric and magnetic fields with a range of frequencies or wavelengths.

Electromagnetic radiation is a common occurrence in our daily lives. All electromagnetic waves, from visible light, that our eyes can detect to microwave radiation that heats our meals or radio waves that power our radios, X-rays that enable doctors to identify any injury in our bones or UV radiation emitted by a hot surface, are EM waves.

Electromagnetic wave:

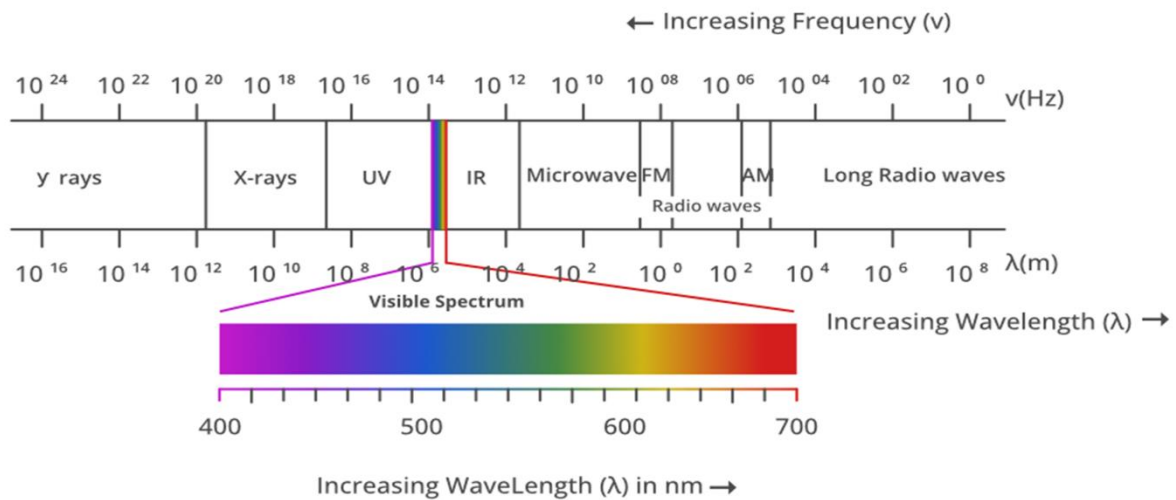
Waves created by the interaction of vibrating electric and magnetic fields are known as electromagnetic waves. An oscillating electric and magnetic field makes up EM waves.

**Electromagnetic Spectrum:**

1. The electromagnetic spectrum is a collection of frequencies, wavelengths, and photon energies of electromagnetic waves spanning from 1Hz to 1025Hz, equivalent to wavelengths ranging from a few hundred kilometers to a size smaller than the size of an atomic nucleus.
2. The electromagnetic spectrum can thus be described as the range of all types of electromagnetic radiation in basic terms.
3. In a vacuum, all electromagnetic waves travel at the same speed as light. For different forms of electromagnetic waves, however, the wavelengths, frequencies, and photon energy will vary.

Electromagnetic waves in electromagnetic spectrum:

Radio waves, microwaves, infrared radiation, visible light, ultraviolet radiation, X-rays, gamma rays, and cosmic rays make up the full range (electromagnetic spectrum) in increasing order of frequency and decreasing order of wavelength.



1. Radio Waves

- The rapid travel of charged particles across conducting wires causes these waves.
- Radio, television, and telecom signals are transmitted through them.
- These waves have a frequency range of around 3kHz to 300MHz.
- Radio picks up radio waves that are broadcast by radio stations. The majority of radio waves are used for TV and mobile communication.

2. Microwaves

- Microwaves are a type of electromagnetic radiation that has a frequency of a few gigahertz (GHz).
- Microwaves are commonly utilized in aviation navigation due to their short wavelengths.
- These rays are employed in microwaves, which aid in the heating of meals in homes and offices.
- It's also used by astronomers to figure out and understand the structure of surrounding galaxies and stars.

3. Infrared Rays

- Infrared waves are produced by hot bodies and molecules and are thus referred to as heatwaves.
- Infrared rays are near the low-frequency ($10^{13} - 10^{14}$) or long-wavelength end of the visible light spectrum.
- The greenhouse effect caused by these rays is critical for maintaining global warming and average temperatures.
- Night vision goggles make use of these radiations. Infrared light generated by objects in the dark can be read and captured by these devices.

4. Visible Rays

- Visible rays are electromagnetic waves that can be seen with the naked eye.
- These can be found in the frequency range of $4 \times 10^{14} \text{Hz} - 7 \times 10^{14} \text{Hz}$ or the wavelength range of 400nm–700nm.
- The visible light rays reflected or released from the objects around us assist us in seeing the world, and the range of visible radiation is different for different creatures.
- Devices that emit light in the visible area of the electromagnetic spectrum include bulbs, lamps, candles, LEDs, tube lights, and so on.

5. Ultraviolet Rays

- Although the sun is the primary source of ultraviolet radiation on Earth, the ozone layer absorbs the majority of UV energy before it reaches the atmosphere.
- UV radiation has a wavelength of 400nm–1nm.
- These radiations are emitted by special lamps and extremely hot bodies, and in big numbers, they can cause significant injury to humans. It tans the skin and creates burns.
- Because these radiations may be focused on tiny beams, they are used in high precision applications such as LASIK or laser-based eye surgery.
- UV lamps are used in water purifiers to eliminate microorganisms that may be present in the water.

6. X-Rays

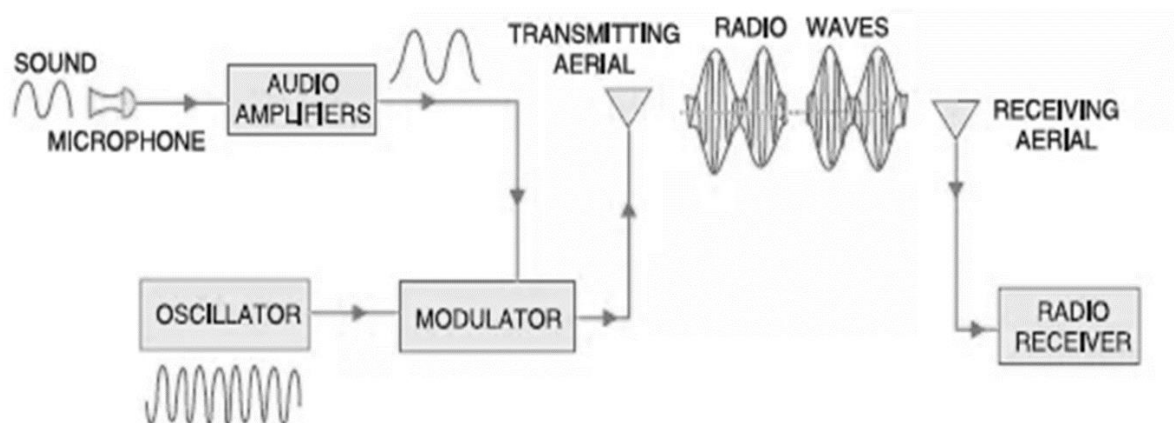
- This electromagnetic radiation is found outside of the ultraviolet (UV) region of the electromagnetic spectrum and is extremely valuable in the medical field.
- The wavelength range of X-ray radiation is 1nm– 10^{-3} nm.
- By blasting a metal target with high-energy electrons, X-rays can be produced.
- X-rays are a diagnostic technique in medicine that can be quite helpful in the treatment of some types of cancer. To find the source of the problem, a doctor utilizes an x-ray scanner to scan our bones or teeth. Overexposure to x-rays can cause harm or death to the organism's healthy tissues.
- At the airport checkpoint, security agents utilize it to search through passengers' luggage. X-rays are also emitted by the universe's heated gases.

7. Gamma-Rays

- The universe is the largest gamma-ray generator.
- These rays are in the electromagnetic spectrum's higher frequency region.
- Gamma rays have wavelengths ranging from 10^{-12}m to 10^{-14}m .
- Radioactive nuclei release high-frequency radiations, which are also created during nuclear processes.
- Gamma rays have a wide range of medical applications, including the destruction of cancerous cells. Gamma-ray imaging is a technique used by doctors to examine the insides of patients' bodies.

5.2 RADIO BROADCASTING AND RECEPTION

Radio communication or radio broadcasting is the radiation of radio waves by the transmitting station, the propagation of these waves through the space and their reception by the radio receiver.



The figure above shows the general principles of radio broadcasting, transmission and reception. We can divide the entire arrangement into 3 major sections, namely:

1. Transmitter
2. Transmission of radio waves
3. Radio receiver

1. Transmitter

This equipment is housed in the broadcasting station. It produces the radio waves for transmission into the space. The transmitter is made up of several components like:

Microphone – This is a device that converts soundwaves into electrical waves.

Audio amplifier – The audio signal from the microphone is usually quite weak and requires amplification. This is accompanied by cascaded audio amplifiers.

Oscillators – The purpose of the oscillator is to produce a high frequency signal, called a carrier wave. Normally, a crystal oscillator is used for this purpose. The power level of the carrier wave is raised to a sufficient level by radio frequency amplifier stages. Most of the broadcasting stations have a carrier wave power of several kilowatts.

Modulator - The amplified audio signals and carrier wave are fed to the modulator. In the modulator the audio signal is superimposed on the carrier wave in a suitable manner. The resultant waves are called modulated waves or radio waves and the process is called modulation. The process of modulation allows the transmission of audio signal at the carrier frequency. Since the carrier frequency is very high, the audio signal can be transmitted to large distances. The radio waves from the transmitter are fed to the transmitting antenna from where they are then radiated into the space.

2. Transmission of Radio Waves

The transmitting antenna radiates the radio waves in space in all directions. These radio waves travel with velocity of light that is 3×10^8 m/sec. The radio waves are electromagnetic waves and possess the same general properties. These are similar to light and heat waves except that they have longer wavelength.

3. Radio Receiver

On reaching the receiver antenna, the radio waves induce tiny e.m.f in it. This small voltage is fed to the radio receiver. Here, the radio waves are first amplified and then the signal is extracted from them by the process of demodulation. The signal is amplified by audio amplifiers and then fed to the speaker for reproduction into sound waves.

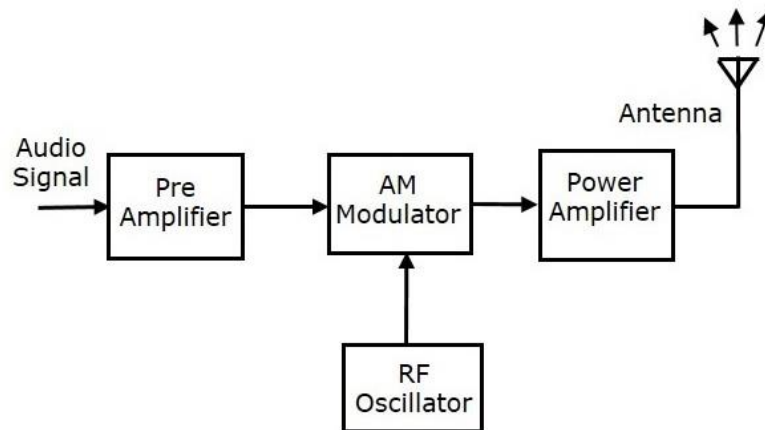
5.3 TRANSMITTERS

A transmitter consists of a precise oscillating circuit or oscillator that creates an AC carrier wave frequency. This is combined with amplification circuits or amplifiers. The distance a carrier wave travels is directly related to the amplification of the signal sent to the antenna.

Other circuits are used in a transmitter to accept the input information signal and process it for loading onto the carrier wave. Modulator circuits modify the carrier wave with the processed information signal.

5.3.1 AM Transmitter

AM transmitter takes the audio signal as an input and delivers amplitude modulated wave to the antenna as an output to be transmitted. The block diagram of AM transmitter is shown in the following figure.

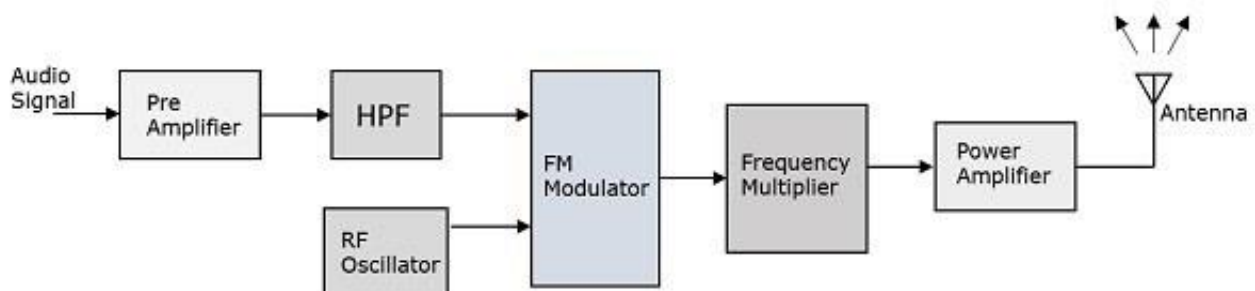


The working of AM transmitter can be explained as follows.

1. The audio signal from the output of the microphone is sent to the pre-amplifier, which boosts the level of the modulating signal.
2. The RF oscillator generates the carrier signal.
3. Both the modulating and the carrier signal is sent to AM modulator.
4. Power amplifier is used to increase the power levels of AM wave. This wave is finally passed to the antenna to be transmitted.

5.3.2 FM Transmitter

FM transmitter is the whole unit, which takes the audio signal as an input and delivers FM wave to the antenna as an output to be transmitted. The block diagram of FM transmitter is shown in the following figure.



The working of FM transmitter can be explained as follows.

1. The audio signal from the output of the microphone is sent to the pre-amplifier, which boosts the level of the modulating signal.
2. This signal is then passed to high pass filter, which acts as a pre-emphasis network to filter out the noise and improve the signal to noise ratio.
3. This signal is further passed to the FM modulator circuit.
4. The oscillator circuit generates a high frequency carrier, which is sent to the modulator along with the modulating signal.
5. Several stages of frequency multiplier are used to increase the operating frequency. Even then, the power of the signal is not enough to transmit. Hence, a RF power amplifier is used at the end to increase the power of the modulated signal. This FM modulated output is finally passed to the antenna to be transmitted.

5.4 RECEIVERS

5.4.1 Requirements of a Receiver

AM receiver receives AM wave and demodulates it by using the envelope detector. Similarly, FM receiver receives FM wave and demodulates it by using the Frequency Discrimination method. Following are the requirements of both AM and FM receiver.

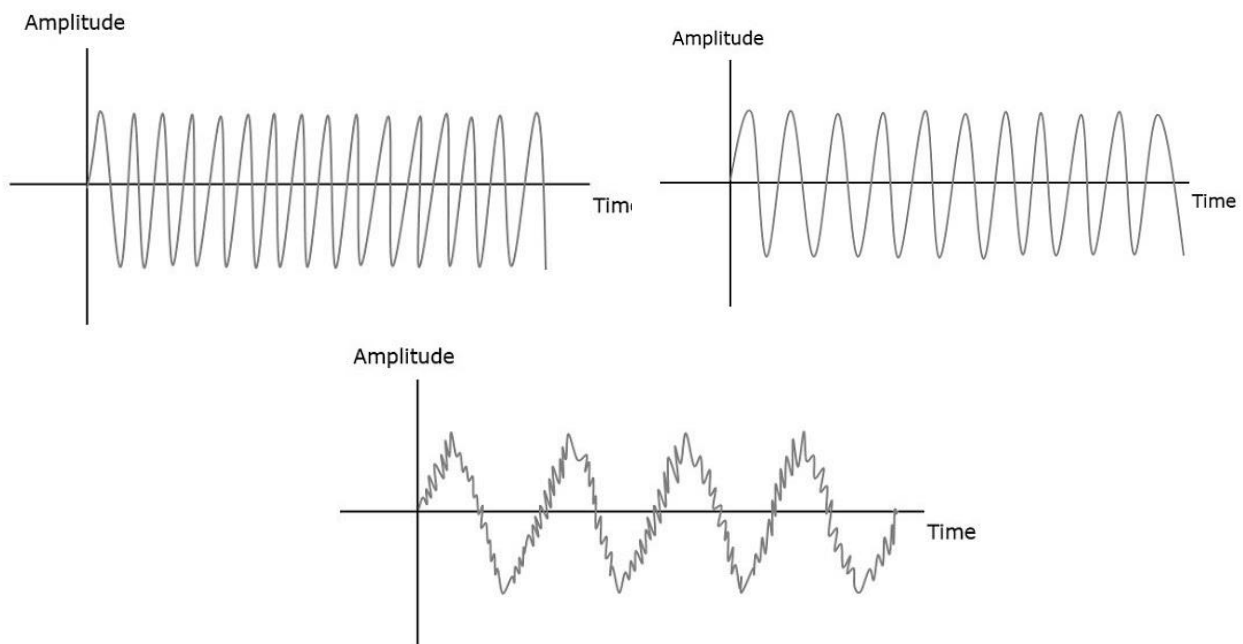
1. It should be cost-effective.
2. It should receive the corresponding modulated waves.
3. The receiver should be able to tune and amplify the desired station.
4. It should have an ability to reject the unwanted stations.
5. Demodulation has to be done to all the station signals, irrespective of the carrier signal frequency.

For these requirements to be fulfilled, the tuner circuit and the mixer circuit should be very effective. The procedure of RF mixing is an interesting phenomenon.

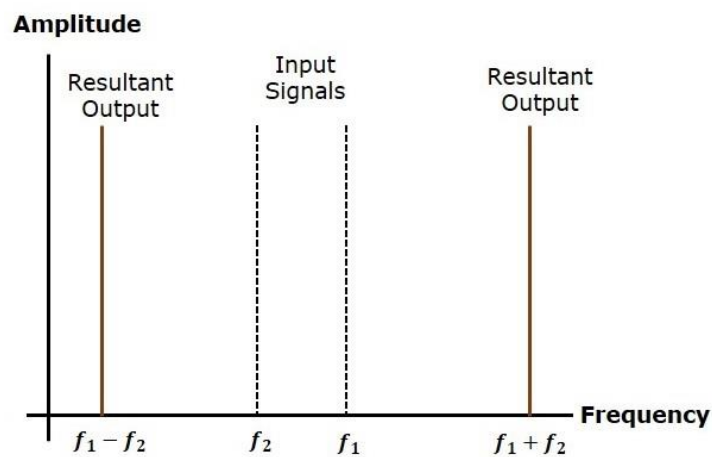
5.4.2 RF Mixing

The RF mixing unit develops an Intermediate Frequency (IF) to which any received signal is converted, so as to process the signal effectively.

RF Mixer is an important stage in the receiver. Two signals of different frequencies are taken where one signal level affects the level of the other signal, to produce the resultant mixed output. The input signals and the resultant mixer output is illustrated in the following figures.



Let the first and second signal frequencies be f_1 and f_2 . If these two signals are applied as inputs of RF mixer, then it produces an output signal, having frequencies of f_1+f_2 and f_1-f_2 . If this is observed in the frequency domain, the pattern looks like the following figure.



In this case, f_1 is greater than f_2 . So, the resultant output has frequencies f_1+f_2 and f_1-f_2 . Similarly, if f_2 is greater than f_1 , then the resultant output will have the frequencies f_1+f_2 and f_2-f_1 .

5.4.3 Characteristics of a Receiver

There are several characteristics commonly used to evaluate the ability of a receiver to successfully demodulate a radio signal. The important characteristics are:

1. Selectivity

Selectivity refers to the ability of a receiver to select a signal of a desired frequency while reject all others. Selectivity in a receiver is obtained by using tuned circuits. These are LC circuits tuned to resonate at a desired frequency. Selectivity shows the attenuation that the receiver offers to signals at frequencies near to the one to which it is tuned. A good receiver isolates the desired signal in the RF spectrum and eliminated all other signals.

2. Sensitivity

The sensitivity of a communication receiver refers to the ability to pick up weak signals and amplify it. It is often defined in terms of the voltage that must be applied to the receiver input terminals to give a standard output power, measured at the output terminals. The more gain that a receiver has, the smaller the input signal necessary to produce desired output power. Therefore, sensitivity is a primary function of the overall receiver gain. It is often expressed in microvolts or in decibels. Good communication receiver has sensitivity of 0.2 to 1 μ V.

3. Fidelity

Fidelity is the ability of a receiver to reproduce the original message signal after demodulation. The fidelity basically depends on the frequency response of the AF amplifier. High fidelity is essential in order to reproduce good quality music faithfully without any distortion. For this it is essential to have a flat frequency response over a wide range of audio frequencies.

5.5 TYPES OF RECEIVERS

The receivers can be classified into two types

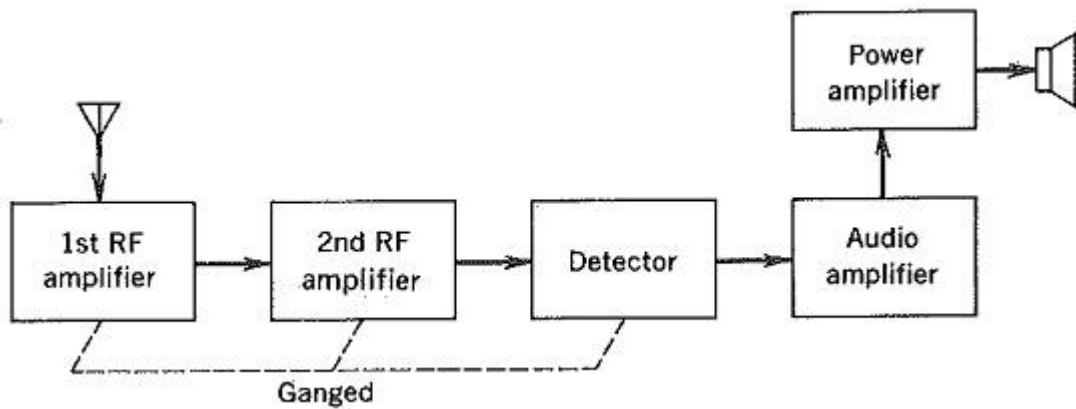
1. AM Receivers
2. FM Receiver

AM receivers are again classified into Two TYPES

- i. Tuned Radio Frequency (TRF) Receiver
- ii. Super heterodyne Receiver

5.5.1 Tuned Radio Frequency (TRF) Receiver

A TRF receiver is the simple form of AM receiver.



Receiving Antenna: A receiving antenna receives radio signals from various broadcasting stations and converts them into RF voltage. This RF voltage is then fed to the RF amplifier stage.

RF amplifiers: Two or perhaps three RF amplifiers, all tuning together, were employed to select and amplify the incoming frequency and simultaneously to reject all others. (such as noise).

Detector or Demodulator: It is used to recover the original message signal from the modulated signal.

Audio Amplifier: The detected audio signal is amplified by using audio amplifier.

Power Amplifier: A power amplifier is used at the final stage of receiver to amplify the power level of the signal to drive the loudspeaker.

Loudspeaker: A loudspeaker is a transducer, which is used to convert the electrical signal into sound signal.

Advantages:

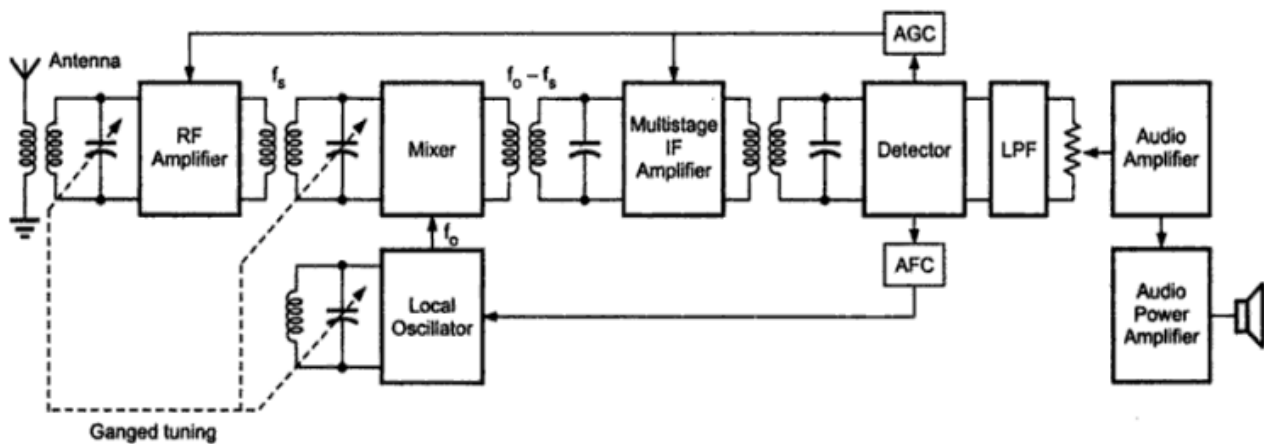
1. It is a basic and simple receiver.
2. These receivers can easily be implemented.
3. It requires less number of components.

Disadvantages:

1. Use of large number of RF amplifiers in the receiver circuit makes the receiver highly instable.
2. The selectivity of a receiver is its ability to distinguish between a desired signal and undesired signal. The selectivity of TRF receiver is poor, it is difficult to achieve sufficient selectivity at high frequencies.

5.5.2 Super Heterodyne Receiver

The block diagram of a basic super heterodyne receiver is shown below. It is the most widely used AM receiver.



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 detect 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 super heterodyne 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. shows the block diagram of super heterodyne receiver. As shown in the fig. 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. Final, 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 circuits. AGC is used to maintain a constant output voltage level over a wide range of RF input signal levels.

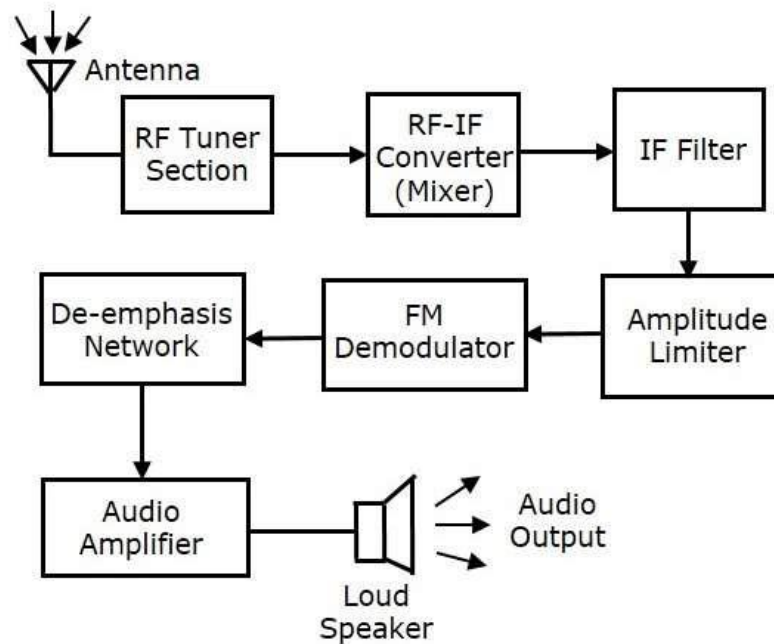
AFC circuit generates AFC signal which is used to adjust and stabilize the frequency of the local oscillator.

Advantages:

1. No variation in bandwidth over the entire frequency range.
2. It provides uniform gain over a wide range of frequencies.
3. High sensitivity and selectivity.
4. Improved frequency stability.
5. High adjacent channel frequency rejection
6. Good fidelity.

5.5.3 FM Receiver

The block diagram of FM receiver is shown in the following figure.



Receiving Antenna: A receiving antenna receives radio signals from various broadcasting stations and converts them into RF voltage. This RF voltage is then fed to the RF amplifier stage.

RF amplifier: An RF amplifier is always used in an FM receiver. Its main purpose is to reduce the noise figure, which could otherwise be a problem because of the large bandwidths needed for F.M.

RF Mixer: The signal from the tuner output is sent to the RF-IF converter, which acts as a mixer. It has a local oscillator, which produces a constant frequency. The mixing process is done here, having the received signal as one input and the local oscillator frequency as the other input. The resultant output is a mixture of two frequencies $[(f_1 + f_2), (f_1 - f_2)]$ produced by the mixer, which is called as the Intermediate Frequency (IF).

IF Filter: Intermediate frequency filter is a band pass filter, which passes the desired frequency. It eliminates all other unwanted frequency components present in it. This is the advantage of IF filter, which allows only IF frequency.

De-emphasis network: In FM transmitter, we have seen the pre-emphasis network (High pass filter), which is present before FM modulator. This is used to improve the SNR of high frequency audio signal. The reverse process of pre-emphasis is known as de-emphasis. Thus, in this FM receiver, the de-emphasis network (Low pass filter) is included after FM demodulator.

Amplitude limiter: We know that in FM modulation, the amplitude of FM wave remains constant. However, if some noise is added with FM wave in the channel, due to that the amplitude of FM wave may vary. Thus, with the help of amplitude limiter we can maintain the amplitude of FM wave as constant by removing the unwanted peaks of the noise signal.

This signal is passed to the audio amplifier to increase the power level. Finally, we get the original sound signal from the loudspeaker.