Higher Secondary Course

# Electronics

XII

Government of Kerala

DEPARTMENT OF EDUCATION

State Council of Educational Research and Training (SCERT) Kerala

2015

## THE NATIONAL ANTHEM

Jana-gana-mana adhinayaka, jaya he
Bharatha-bhagya-vidhata.
Punjab-Sindh-Gujarat-Maratha
Dravida-Utkala-Banga
Vindhya-Himachala-Yamuna-Ganga
Uchchala-Jaladhi-taranga
Tava subha name jage,
Tava subha asisa mage,
Gahe tava jaya gatha.
Jana-gana-mangala-dayaka jaya he
Bharatha-bhagya-vidhata.
Jaya he, jaya he, jaya he,
Jaya jaya jaya, jaya he!

## PLEDGE

India is my country. All Indians are my brothers and sisters.

I love my country, and I am proud of its rich and varied heritage. I shall always strive to be worthy of it.

I shall give my parents, teachers and all elders respect, and treat everyone with courtesy.

To my country and my people, I pledge my devotion. In their well-being and prosperity alone lies my happiness.

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## Foreword

Dear Students,

It is with immense pleasure and pride that State Council of Educational Research and Training (SCERT) Kerala brings forth its first textbook in Electronics for higher secondary students of XII$^{th}$ standard. Our endeavor to revise the syllabus and prepare a textbook of Electronics has fulfilled a long cherished dream of teachers and students of the subject at higher secondary level.

Electronics is a very highly demanded discipline in the modern era. It is a growing science which aims to make a difference in the world by dealing with technology advanced in all places and at all times. The rapid development of Electronics extremely influences the lifestyle of every human being and causes to improve the facilities in all fields of life. The tremendous career and research opportunities in the field of Electronics engineering and technology worldwide encourage many of the science students to go for their higher studies in this field. Moreover it is worthwhile if the study of science and technology in school level supports students in skill acquisition and develops aptitude towards entrepreneurship. These are some of the facts which led to the introduction of Electronics as an optional subject for higher secondary level. The students can utilize the knowledge acquired from the classroom to design and develop innovative devices to be used in various situations in their day to day life. I am sure that the study of Electronics at higher secondary level will be a stepping stone for students to enter the vast and amazing world of technology.

This textbook is the result of a combined effort of a team of practicing teachers and experts in Electronics in the state of Kerala. I hope that the students of this subject would make full use of the inputs offered in the book.

Wish you all success.

**Dr. S Raveendran Nair**
Director
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# 1 POWER SUPPLIES AND VOLTAGE STABILISERS

## Significant Learning Outcomes

*After completing this chapter the learner:*

- explains the need of regulated power supply
- explains the basic principles of series and shunt voltage regulators
- sketches the circuit diagrams of fixed voltage regulators
- identifies various IC voltage regulators
- designs and constructs a power supply

In general, all electronic circuits require a source of DC power. The amplifier and oscillator circuits you studied previous year employ a DC power supply named $V_{cc}$. Batteries can be used for providing DC supply. But they are costly and require frequent replacement. Also the electric power supply available is AC. So the DC power required for electronic circuits is obtained from 230 V AC lines by using a rectifier filter system followed by a regulator. This system which is used to convert AC supply to the required DC power is known as DC power supply.

The DC voltage from power supply remains constant so long as either AC mains voltage or load is constant. In most of the applications, it is desired that DC voltage should remain constant irrespective of the changes in AC mains or load. To make the DC voltage constant, voltage regulating devices are used. A power supply including a voltage regulator circuit is called a regulated DC power supply. Its output is a DC voltage which is fairly constant. The process of rectification and filtering were discussed previous year. In this chapter, we shall focus our attention on various voltage regulating circuits used to obtain regulated power supply.

## 1.1 Need for regulated power supply

Consider the block diagram of a DC regulated power supply. It consists of a rectifier, filter and a voltage regulator as shown in Fig 1.1.

Use your school laboratory to assemble the above circuit and verify the output waveforms using a CRO. The output from the rectifier is a pulsating DC. These pulsations are due to the presence of AC components in the rectifier output. We have seen that a filter circuit helps to obtain steady DC voltage across the load. But the DC voltage may change due to variations in the input voltage and load current.

**Limitations:** An ordinary DC power supply has the following drawbacks.

1) The DC output voltage decreases as the load current increases. This is due to the voltage drop in the transformer winding, rectifier and filter circuits. When the load current increases, the internal drop increases and the output voltage decreases.

2) In practice, there are considerable variations in the AC line voltage caused by outside factors beyond our control. We know that the DC output voltage of a fullwave rectifier is given by the equation $V_{DC}=2V_m/\pi$, where $V_m$ is the peak value of input AC voltage and it is quite clear that the output DC voltage depends on the input AC voltage. Most of the electronic circuits refuse to work satisfactorily on such output voltage fluctuations. This necessitates the use of regulated DC power supply as the DC voltage source of electronic circuits.

3) The internal resistance of ordinary power supply is relatively large (>30 $\Omega$). Therefore, the output voltages will be considerably affected by the amount of load current drawn from the supply.

These variations in DC voltage may cause erratic operation of electronic circuits. Therefore, regulated DC power supply is essential in most electronic applications.

## 1.2 Regulated Power supply

A DC power supply which maintains the output voltage constant irrespective of AC mains fluctuations or load variations is known as a regulated DC power supply.



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A regulated power supply consists of an ordinary power supply and a voltage regulating device. Fig 1.1 shows the block diagram of a regulated power supply. You have studied about rectifier, capacitor, filter and zener voltage regulator the previous year. Now you can set up the circuit as shown in the block diagram in your lab. You may observe the output of each section using

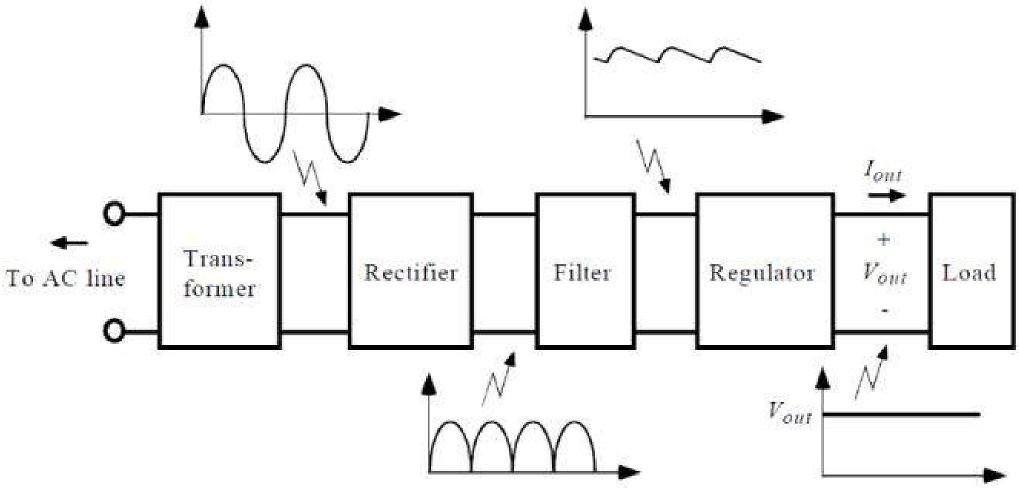


Fig 1.1 Block diagram of a regulated power supply

a CRO and ensure that what you are seeing is similar to the output shown in the figure. The output of ordinary power supply is fed to the voltage regulator which produces the final output. The output voltage remains constant irrespective of the load current changes or fluctuations in the input AC voltage. The performance of a voltage regulator is measured using the following parameters.

### i) Load Regulation

The DC voltage available across the output terminals of a given power supply depends upon the load current. If the load current $I_L$ is increased by decreasing $R_L$ the terminal voltage drops by a small amount. It is expected that the regulator minimizes this variation in the terminal voltage. The load regulation is a measure of the ability of the power supply to reduce the variation in output voltage with the change in load current. The percentage load regulation is represented

$$\text{as \% of Voltage regulation} = \frac{V_{NL} - V_{FL}}{V_{FL}} \times 100$$

Where $V_{NL}$ is the DC output voltage at no load (zero load current), that is, the voltage at the minimum load. The minimum load is the one that draws the least current, i.e. at the highest specified load resistance.

$V_{FL}$ is the DC output voltage at full-load (maximum load current), that is, the voltage at the maximum load. The maximum load is the maximum current that can be drawn from the power supply as per its current rating. As an example, for a 5 V, 1 A power supply, the maximum current or current rating is 1 A and hence the full load is 1 A. In order to obtain this condition a 5 Ω resistance (5 V/1 A) should be connected as the load resistance.

In a well designed power supply, the full-load voltage is only slightly less than the no-load voltage, that is the voltage regulation approaches zero. Lower the voltage regulation, lesser is the difference between full-load and no-load voltages and better is the power supply. Power supplies used in practice have a voltage regulation of 1 % that is the full-load voltage is within 1 % of the no-load voltage.

### ii) Line Regulation

Any change in the line voltage out of the normal value (i.e., 230 V AC) will affect the performance of the power supply. We know that the DC output voltage of a full wave rectifier is given by $V_{dc} = 2V_m/\pi$. Here, $V_m$ is the peak voltage of the AC input. So any change in AC input voltage will change the output DC voltage also. Line regulation is a measure of the ability of the power supply to maintain its output voltage constant against the changes in the input line voltage. Line regulation is expressed as the percentage change in the output voltage relative to the change in the input line voltage. The line regulation is represented as:

$$\text{Line regulation} = \frac{V_{HL} - V_{LL}}{V_{LL}} \times 100$$

Where $V_{HL}$ = load voltage with high line voltage

$V_{LL}$ = load voltage with low line voltage

The smaller the line regulation, the better is the power supply. A well-regulated power supply can have a line regulation of less than 0.1%.

### Types of Voltage Regulators

A device which maintains the output voltage of an ordinary power supply constant irrespective of load variations or changes in input AC voltage is known as voltage regulator. A voltage regulator generally employs electronic devices to achieve this objective. There are basically two types of voltage regulators.

(i) Shunt voltage regulator

(ii) Series voltage regulator

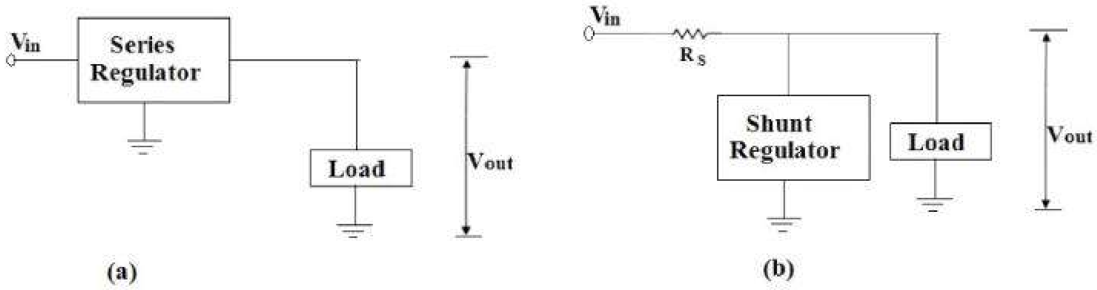


Fig 1.2 Series and shunt voltage regulators

The series regulator is placed in series with the load as shown in Fig 1.2(a). On the other hand, the shunt regulator is placed in parallel with the load as shown in Fig 1.2 (b). Each type of regulator provides output voltage that remains constant even if the input voltage varies or the load current changes.

## 1.3 Zener diode Voltage Regulator

We have studied that, when the zener diode is operated in the break down or zener region, the voltage across it is constant for large change of current through it. This characteristic of zener diode helps it to be used as voltage regulator. Fig 1.3 shows the circuit of a zener diode regulator. As long as the input voltage $V_{in}$ is greater than the zener voltage $V_Z$, the zener operates in the breakdown region and maintains constant voltage across the load. The zener diode will operate in the breakdown region only if the current through it is greater or equal to a minimum current as its specifications. Also a zener diode cannot withstand current above the maximum value. Hence, for voltage regulators, zener diodes of appropriate voltage and current ratings should be selected. The series resistance $R_S$ is used for limiting the input current.

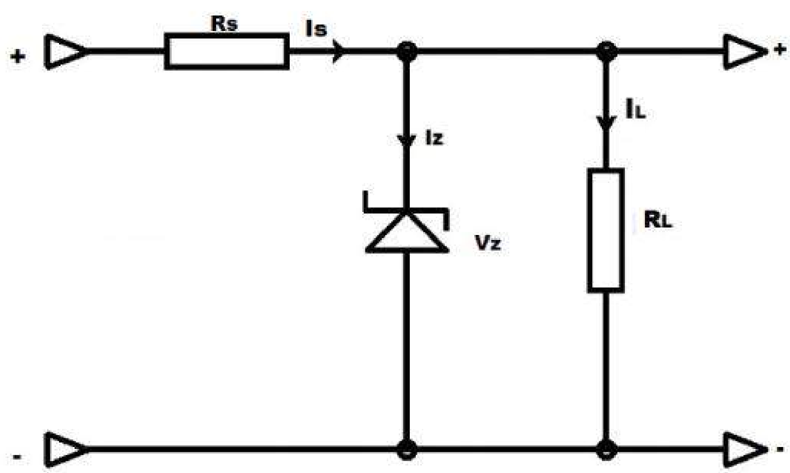


Fig 1.3 Zener diode regulator



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The zener will maintain constant voltage across the load in spite of the changes in the load current or the input voltage. Now, let us see how this regulator maintains constant output voltage when the input voltage $V_{in}$ varies. Assume that $V_{in}$ varies above a specific voltage so that the zener diode is always in break down condition.

If $V_{in}$ increases, then the input current $I_S$ also increases because

$$I_S = (V_{in} - V_Z)/R_S \qquad (1.1)$$

Also we have

$$I_S = I_Z + I_L \qquad (1.2)$$

$$V_O = V_Z = I_L R_L \qquad (1.3)$$

From the equation 1.3, it is seen that $I_L$ cannot vary since $V_Z$ and $R_L$ do not vary. So from the equation 1.2, it is clear that when $I_S$ increases, $I_Z$ also increases. For a zener diode it is possible that the current through it can vary without changing the voltage across it, if the diode is in the break down condition. Thus the output voltage remains constant even when the input voltage increases. The increased input voltage causes $I_S$ to increase and hence this voltage drops across the series resistor $R_S$.

When $V_{in}$ decreases, the $I_S$ also decreases, so that the voltage across $R_S$ decreases and the output voltage $V_Z$ remains constant. Thus any change in the input voltage changes the voltage across $R_S$ and not the output voltage.

What happens to the regulator circuit when load varies? We shall discuss now the regulation against the load variation. Let us assume that the line voltage or input voltage remains constant. So $V_{in}$ is constant. Hence the total current drawn from the circuit, $I_S$ should be constant according to the equation 1.1.

If $R_L$ increases, $I_L$ also decreases. Then the current $I_Z$ flowing through the zener diode increases, so that, $I_S$ remains constant. Similarly when $R_L$ decreases, $I_L$ increases. So the zener current $I_Z$ decreases in order to keep $I_S$ constant. Thus if the regulator can keep $I_S$ constant, then the output voltage $V_Z$ can be made constant as per equation 1.1. In order to keep $I_S$ constant, $I_Z$ should vary according to equation 1.2. Now it is clear that the regulator circuit keeps the output voltage constant, since the zener diode under breakdown condition can vary the current flowing through it without varying the voltage across it.

**Limitations**

A zener diode regulator has the following drawbacks

(i) It has low efficiency because of the considerable power loss in the series



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resistance. We have seen that the regulator circuit draws a constant current $I_S$ from the supply irrespective of the required load current $I_L$. So even if a small load current is required, the circuit will draw constant $I_S$ and large power will be dissipated in $R_S$.

(ii) The output voltage slightly changes due to the slope in zener characteristic.

Hence, the changes in the load current produces changes in the zener current. Consequently, the output voltage also changes. Therefore, the use of this circuit is limited only to such applications, where the variations in the load current and the input voltage are small.

**Know your progress**

1. Discuss the property of zener diode which makes it suitable for voltage regulation.

## 1.4 Transistor Series Voltage Regulator

Fig 1.4 shows a simple series voltage regulator using a transistor and a zener diode. The circuit is called a series voltage regulator because the load current passes through the series transistor Q1 connected in series with the load as shown in Fig 1.4. The unregulated DC voltage is fed to the input terminals and the regulated output is obtained across the load. The zener diode provides the reference voltage.

The base voltage of transistor Q1 is held to a relatively constant voltage with the voltage across the zener diode. For example, if 8 V Zener (that is Vz= 8 V) is used, the base of Q1 will remain approximately at 8 V. Referring to Fig 1.4.

$$V_{Out} = V_Z - V_{BE}$$

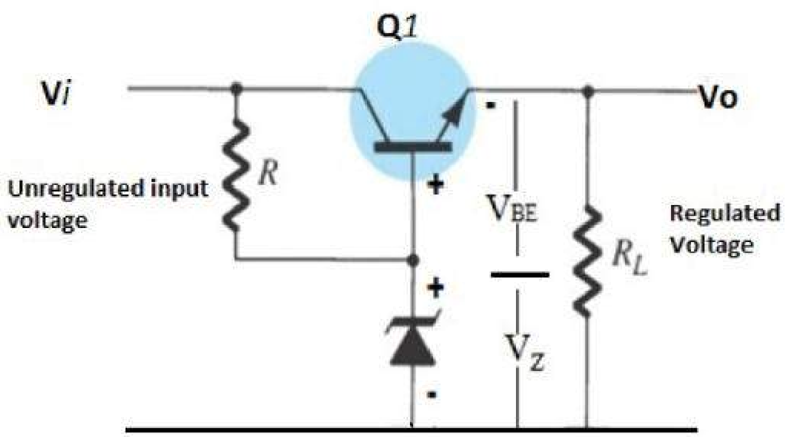


Fig 1.4 Transistor Series voltage regulator

i) If the output voltage decreases, $V_{BE}$ increases since $V_Z$ is a constant. The increased base-emitter voltage causes transistor Q1 to conduct more so that, load current flowing through $R_L$ increases thereby raises the output voltage ($V_O = I_L R_L$). As a result, the output voltage is maintained at a constant level.

ii) If the output voltage increases, the decreased base-emitter voltage causes the transistor Q1 to conduct less, thereby reduces the output voltage. Consequently, the output voltage is maintained at a constant level.

The advantage of this circuit is that, the changes in the zener current are reduced by a factor β. This is because in order to affect large change in the load current ($I_L=I_C$), small change in the zener current is sufficient ($I_Z=I_B=I_C/\beta$). Therefore, the effect of zener impedance is greatly reduced and much more stabilized output is obtained.

In addition to this, the circuit draws current from the supply according to the load requirement. It makes the system highly energy efficient.

**Limitations:**

i) Although the changes in zener current are much reduced, the output is not absolutely constant. It is because both $V_{BE}$ and $V_Z$ decrease with the increase in the room temperature.

ii) The output voltage cannot be changed easily as no such means are provided.

**Know your progress**

1. Find the reasons for the energy efficiency of transistor series regulator over the zener voltage regulator.

## 1.5. IC Voltage regulators

Voltage regulators comprise a class of widely used ICs. Regulator IC unit contains the circuitry for reference source, comparator amplifier, control device and over load protection circuit all within a single IC. Although the internal construction of IC is somewhat different from that of a discrete voltage regulator circuit, the external operation is more or less the same. IC unit provides regulation of a fixed positive voltage, a fixed negative voltage or a variable voltage. They are of low cost, high reliability and reduced size and give excellent performance. IC voltage regulators are of the following types:

i) Fixed output voltage regulator :Positive and Negative output voltage

ii) Adjustable output voltage regulators: Positive and negative output voltage

## 1.6 Fixed Voltage Regulators

### i) Positive voltage regulator

The 78XX (7800) series consists of three terminal positive voltage regulators with eight voltage options. This series of regulators provides fixed regulated voltages from 5 V to 24 V. Table 1.1 shows the voltage options of 78XX series. These 78XX series regulators give a maximum output current of about 1.5 amperes at fixed stabilized voltages of 5, 6, 8, 10, 12, 15, 18 and 24V respectively.

| IC Part | Output Voltage (V) | Minimum $V_i$ (V) |
|---|---|---|
| 7805 | +5 | +7.3 |
| 7806 | +6 | +8.3 |
| 7808 | +8 | +10.5 |
| 7810 | +10 | +12.5 |
| 7812 | +12 | +14.5 |
| 7815 | +15 | +17.7 |
| 7818 | +18 | +21.0 |
| 7824 | +24 | +27.1 |

Table 1.1 Positive voltage regulators in the 78XX Series

The minimum $V_i$ shown in the third column of the above table indicates the minimum input voltage required for the IC to provide the regulated voltage as required. These ICs are designed as fixed voltage regulators and with adequate heat sinking and they can deliver high output currents say, 0.5 A or more. Fig 1.5 below shows the typical package and pin out configuration of 78XX series regulator.

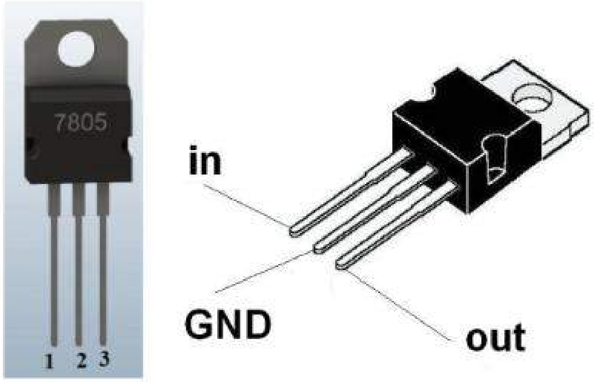


Fig 1.5 Typical package and pinout diagram of IC 7805



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The proper operation of 7805 requires a common ground between input and output voltages as shown in Fig 1.6. The only additional components required are capacitors across the input and the output.

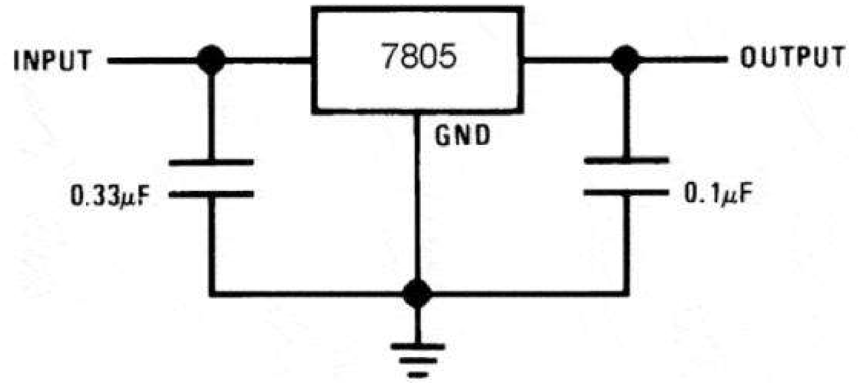


Fig 1.6 Connection diagram of IC-7805

The input capacitor bypasses any spike voltage in the input so as to make the operation of the IC stable. The output capacitor avoids the sudden changes in the output voltage when the load is suddenly changed. It can be typically between 0.1µF and 0.33µF.

The unregulated input voltage is connected to the IC's IN terminal (Pin-1). The ICs OUT terminal (Pin – 3) provides a regulated output which is filtered by the output capacitor. The third terminal is connected to the ground GND.

Now let us see the regulator using 7815.

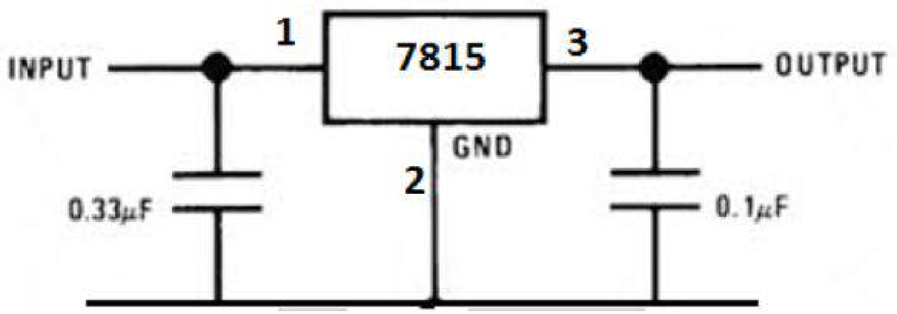


Fig 1.7 Connection of 7815 Voltage regulator

Here in this circuit IC-7815 is used which is a fixed positive voltage regulator so the output will be +15 V.

**ii) Negative Voltage regulator**

The 79XX (7900) series of fixed output negative voltage regulators are similar to the 78XX series devices. The negative regulators are available in different voltage options as the 78XX devices. In addition to these two extra voltage

| IC Part | Output Voltage (V) | Minimum $V_i$ (V) |
|---|---|---|
| 7905 | -5 | -7.3 |
| 7906 | -6 | -8.4 |
| 7908 | -8 | -10.5 |
| 7909 | -9 | -11.5 |
| 7912 | -12 | -14.6 |
| 7915 | -15 | -17.7 |
| 7918 | -18 | -20.8 |
| 7924 | -24 | -27.1 |

Table 1.2 Negative voltage regulators in the 79XX series

options, -2V and -5.2V are also available in the 79XX series. Table 1.2 shows the voltage options of 79XX devices and fig1.8 shows the package type and the connection diagram of 79XX series.

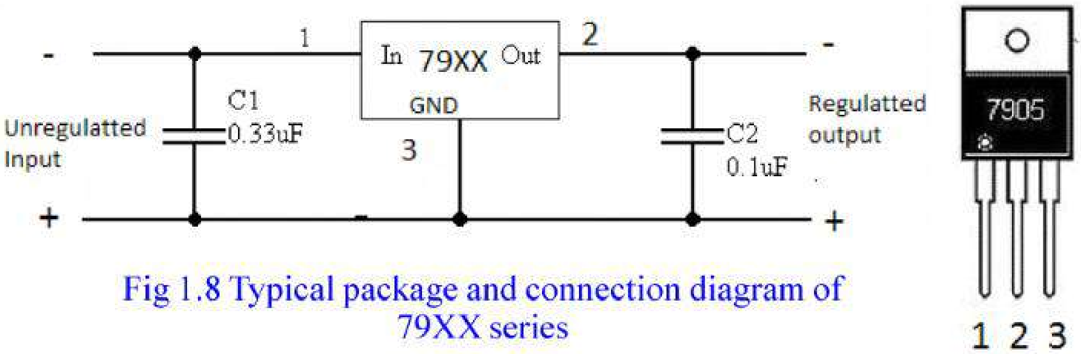


Fig 1.8 Typical package and connection diagram of 79XX series

## 1.7 Adjustable voltage regulators

Voltage regulators are also available in circuit configurations that allow the user to set the output voltage to a desired regulated value. The adjustable voltage regulators have become more popular because of versatility, performance and reliability. The LM 317 series is the most commonly used general purpose adjustable voltage regulators. Fig 1.9 shows typical package of LM317. The three terminals are $V_{in}$, $V_{out}$ and Adjustment (ADJ).

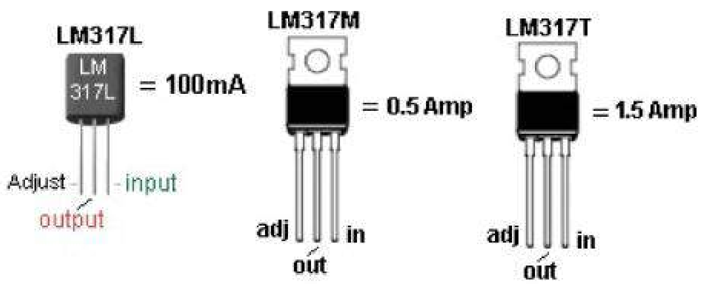


Fig 1.9 Typical package of LM317

The LM317T is a fully adjustable 3-terminal positive voltage regulator capable of supplying 1.5 A with heat sink, with an output voltage ranging from 1.25 to 37 V. By using two resistances, one of a fixed value and the other, variable the output voltage can be set to the desired level. Fig 1.10 shows typical connection diagram of the LM 317 regulator. From this diagram it is obvious that the LM 317 requires only two external resistors to set the output voltage. The output voltage of the LM317 is determined by the ratio of these two external resistors $R_1$ and $R_2$ which form a potential divider network across the output terminal as shown below.

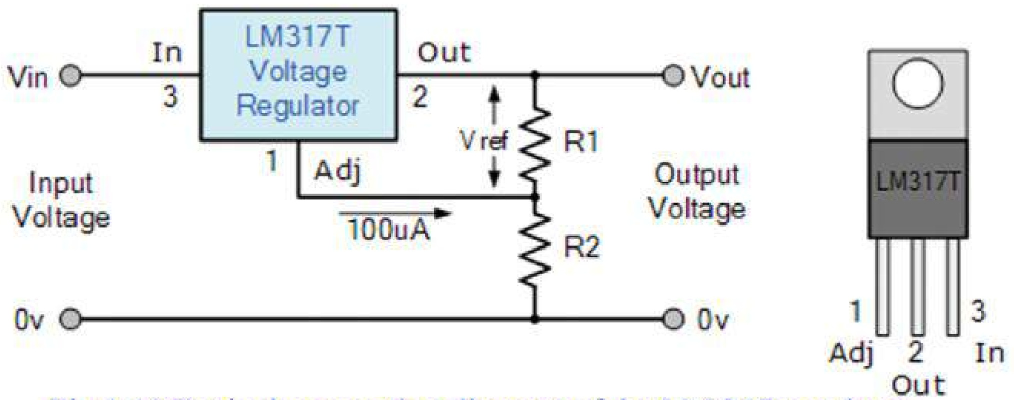


Fig 1.10 Typical connection diagram of the LM 317 regulator

The voltage produced between the 'output' and the 'adjustment' terminals is a constant and its value is 1.25 V. This voltage is taken as the reference voltage by the IC and hence it is denoted as $V_{ref}$. Let $I_1$ be the current flowing through $R_1$. The current from pin 1 of LM317, $I_{adj}$ is negligible compared to $I_1$. So we can consider that the current through $R_2$ is also $I_1$. Now the equation for output voltage is

$$V_o = I_1 (R_1 + R_2)$$

$$\text{But } I_1 = V_{ref} / R1$$

$$\text{So } V_o = [V_{ref} / R_1](R_1 + R_2)$$

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

From the figure, it is clear that the voltage across $R_1$ is equal to $V_{ref}$. Since this voltage is constant, the current $I_1$ is also constant for a given value of $R_1$. Since the resistor $R_1$ sets the current $I_1$, it is called current set or program resistor. In addition to the current $I_1$, the current $I_{ADJ}$ from the adjustment terminal also flows through the output set resistor $R_2$. The adjustment terminal current is a constant current of 100 µA and is comparatively very small and so



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that we can neglect it. Since the reference voltage across resistor $R_1$ is a constant, a constant current I will flow through the other resistor $R_2$, resulting in an output voltage of:

$$V_o = 1.25\left(1+\frac{R_2}{R_1}\right)$$

This indicates that the output voltage $V_o$ is a function of $R_2$ for a given value of $R_1$ and can be varied by adjusting the value of $R_2$. The current set resistor $R_1$ is usually 240 Ω.

### Adjustable Negative Voltage Regulator

LM 337 series of adjustable negative voltage regulators is similar to the LM 317 series devices. These negative regulators are available in the same voltage and current options as the LM 317 devices.

**Know your progress**

1. List out the electronic equipments which use DC power supply.
2. Discuss how the output voltage is fixed in an LM 317 regulator.

## 1.8 Typical DC regulated power supply design

As an example, let us discuss the design of regulated 5 V, 500 mA power supply. Let's start with the choosing of components

**Component List:**

1. Step down transformer
2. Voltage regulator
3. Capacitors
4. Diodes

Let's discuss about the ratings of the required devices.

**Voltage regulator:**

As we require a 5 V, we need LM7805 Voltage Regulator IC.

7805 IC Rating :

- Input voltage range 7 V- 35 V
- Current rating $I_c = 1$ A

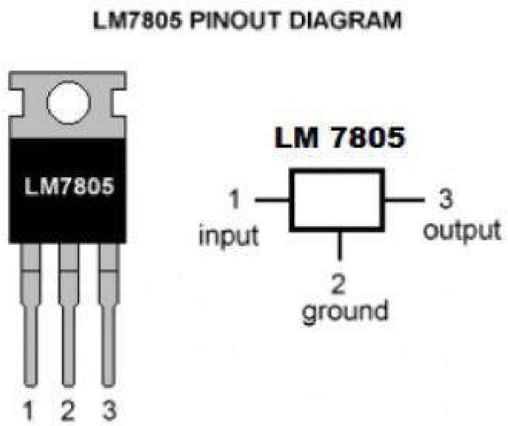


Fig 1.11 LM7805 - Pin Diagram



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**Transformer**

Selecting a suitable transformer is of great importance. The current rating and the secondary voltage rating of the transformer are the factors to be considered.

- The current rating of the transformer depends upon the current required for the load to be driven.
- The input voltage to the 7805 IC should be at least 2 V greater than the required output; therefore, it requires an input voltage at least to 7 V.
- 6-0-6 transformer with a current rating of 500 mA (Since $6 \times \sqrt{2} = 8.4$ V) can be chosen.

*Note* : Any transformer which supplies secondary peak voltage up to 35V can be used but as the voltage increases, the size of the transformer and the power dissipation across the regulator increases.

**Rectifying circuit:**

It is best to use a full wave rectifier here considering its efficiency and ripple factor.

- 1N4007 diodes can be used as it is capable of withstanding a higher reverse voltage of 1000 V, whereas the maximum reverse voltage of 1N4001 is 50 V.

**Capacitors**

To avoid ripples, it is better to use a large value capacitor as filter. Let us choose the capacitor 2200 µF.

The Datasheet of 7805 prescribes the use of a 0.01 µF capacitor at the output side to avoid transient changes in voltages.

The complete circuit showing the power supply is in fig. 1.13

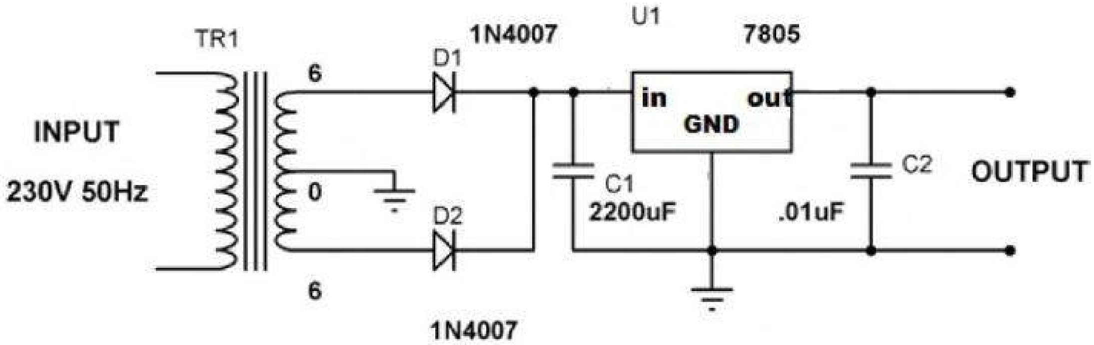


Fig 1.13 5V Power supply circuit using voltage regulator IC 7805


Fig.1.14. A typical regulated power supply unit

## Let us consolidate

The variation in DC voltage may cause malfunctioning of electronic circuits. So regulated DC power supply is essential in most electronic applications. A regulated power supply contains a rectifier, filter and a regulator. The load regulation is a measure of the ability of the power supply to reduce the variation of output voltage with the change in load current. Line regulation is a measure of the ability of a power supply to maintain the output voltage constant against the changes in the input line voltage. The two basic types of voltage regulators are shunt voltage regulator and series voltage regulator. A zener diode regulator circuit keeps the output voltage constant. Since the zener diode can vary the current flowing across it. In a transistor series voltage regulator the load current passes through the transistor connected in series with the load the zener diode provides the reference voltage.

A regulator IC unit contains the circuit for reference source, comparator amplifier, control device and overload protection circuit, all in a single chip. Fixed IC voltage regulators and adjustable voltage regulators are available for positive and negative output voltages. For designing a typical DV regulated power supply, the transformer, IC voltage regulator diodes and capacitors with appropriate rating have to be selected.

The contents of this chapter were learned through general discussion, drawing circuit diagrams, designing circuits and conducting practical experiments.



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## Let us asses

1. The output from the rectifier is a pulsating dc. These pulsations are due to the presence of ac components in the rectifier output.

   a) Which of the following component is generally used to remove the ac ripples?

      i) inductor  ii) capacitor  iii) resistor  iv) diode

   b) What happens to the output when the value of this component is increased?

2. Voltage regulation is essential in all power supplies.

   a) In an unregulated power supply, if input ac voltage increases, the output voltage ...........

   (i) increases (ii) decreases (iii) remains the same (iv) none of the above

   b) Discuss the limitations of an unregulated power supply

3. A dc power supply which maintains the output voltage constant irrespective of the fluctuations of the ac mains or load variations is known as regulated dc power supply.

   a) Draw the block diagram of a regulated power supply.

   b) The performance of a voltage regulator is measured using two parameters. Mention the names of these parameters. What are their significance?

   c) If the dc output voltage is 15 V with no-load attached to the power supply and decreases to 14.6 V at full load. Find the percentage voltage regulation.

4. Zener diode can be used as both dc and ac voltage regulator

   a) A zener diode utilizes .......... characteristics for voltage regulation.

   b) The zener diode will maintain constant voltage across the load in spite of the changes in load current or input voltage. Draw and explain the circuit of zener diode voltage regulator.

   c) Zener operates in the breakdown region and maintains constant voltage across the load. Explain the limitations of zener voltage regulator.

5. The series voltage regulator employs a transistor in the circuit as the regulating element.
   a) Discuss the role of transistor in the circuit.
   b) How does the transistor make the operation of the regulator power supply efficient?
6. Fixed output voltage regulator can be divided into positive and negative voltage regulator.
   a) IC 7805 has …….. output voltage.
   b) Draw the circuit arrangement of fixed voltage regulator IC 7805.
   c) Draw the pin diagram of IC 7905.
7. Commonly used variable voltage regulator IC is LM 317.
   a) IC LM 337 is ………………. Voltage regulator.
      (i) Fixed negative    (ii) Variable Positive
      (iii) Fixed positive    (iv) Variable Negative
   b) The output voltage range of LM 317 is ——— to ——— .
   c) Draw the circuit arrangement of IC - LM 317 as voltage regulator.
8. Describe the steps to design a +12 V, 500 mA regulated power supply.

# 2 WAVESHAPING CIRCUITS

## Significant Learning Outcomes

*After completing this chapter the learner:*

- explains different types of clipping circuits.
- demonstrates the working of clamping circuits.
- sketches the input and output waveforms of RC differentiator and integrator.
- identifies different types of RC filter circuits
- explains the working of different types of filters.
- sketches the circuit diagrams of buffer, adder and subtractor using op-amps.
- explains the working of comparator and its applications.

In signal processing, quite often, the signal wave form has to be properly shaped for various applications. For example, we know that a sawtooth waveform is required in CRT for deflecting the beam. Wave shaping is the process of modifying the shape of a signal to obtain a signal of desired shape. It becomes also necessary to generate one waveform from another, like sharp narrow pulses generated from a rectangular waveform. For wave shaping, we use various circuits like clipper, clamper, differentiator and integrator etc.

Wave shaping circuits that make use of only linear circuit elements, such as inductor, capacitor and resistor are known as the linear wave shaping circuits. Such circuits can perform differentiation, integration and summation. For clipping and clamping circuits, we have to use a non-linear element-diode. Wave shaping circuits using diodes in conjunction with other linear circuit elements, are called non linear wave shaping circuits.

### 2.1 Clipping Circuits

Clipping circuits ( known as limiters, amplitude selectors, or slicers), are used to remove or clip off the part of a signal that is above or between some defined reference levels. The circuit with which a wave form is shaped by removing (or clipping) a portion of the applied wave is known as clipping circuit. Depending on the orientation of the diode, the positive or / and negative region of the input signal is clipped off.

There are two general categories of clippers: series and parallel. The series configuration is defined as one in which the diode is in series with the load, while a parallel configuration has the diode in a branch parallel to the load. The main diode clippers are

I) Positive clipper

II) Negative clipper

III) Biased clipper

IV) Combinational clipper

## I) Positive clipper

A positive clipper is that which removes the positive half cycles of the input voltage. Fig 2.1 shows the circuit diagram of a positive clipper using a diode. As shown, the output voltage has all the positive half-cycles removed or clipped off.

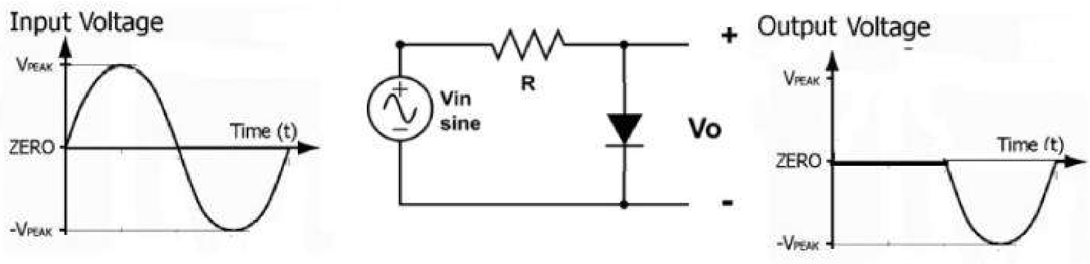


Fig 2.1 *Positive Clipper*

The circuit action is as follows. During the positive half cycle of the input voltage, the diode is forward biased and conducts heavily. Therefore, the voltage across the diode is almost zero (which behaves as short circuit). Hence, the output voltage during positive half cycle is zero.

During the negative half cycle of the input voltage, the diode is reverse biased and behaves as open circuit. Therefore, the input voltage is dropped across the diode. That is, the negative half cycle of the input voltage appears across the diode. The input and output waveforms are shown in Fig 2.1. In practice, the output voltage during positive half cycles will not be zero since there is a voltage drop of 0.7V across a forward biased diode (silicon). So the output voltage during positive half cycles will be 0.7V.

### I) Negative clipper

A negative clipper is that which removes the negative half cycle of the input voltage. Fig 2.2 shows the circuit diagram of a negative clipper.

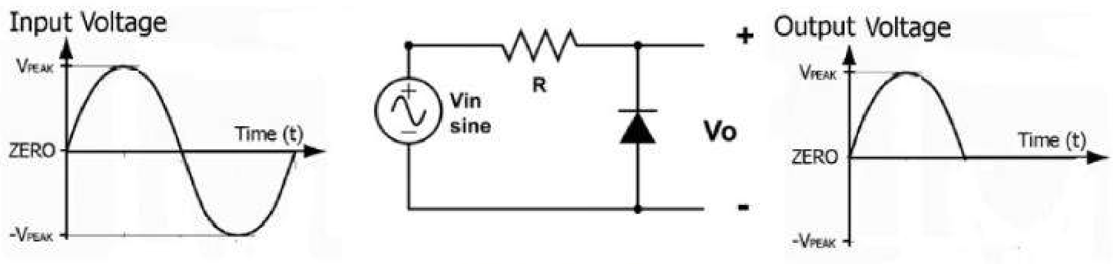


Fig 2.2 Negative clipper

During positive half cycle of the input voltage, the diode is reverse biased and it acts as an open circuit. So the input voltage appears at the output. But during the negative half cycle, the diode is forward biased and it conducts. Therefore, the voltage across the diode is approximately zero. So the output voltage at the terminals is also zero. The input and output waveforms are also shown in Fig 2.2

## Know your progress

Draw the output wave form of Fig 2.3 (a) and (b)

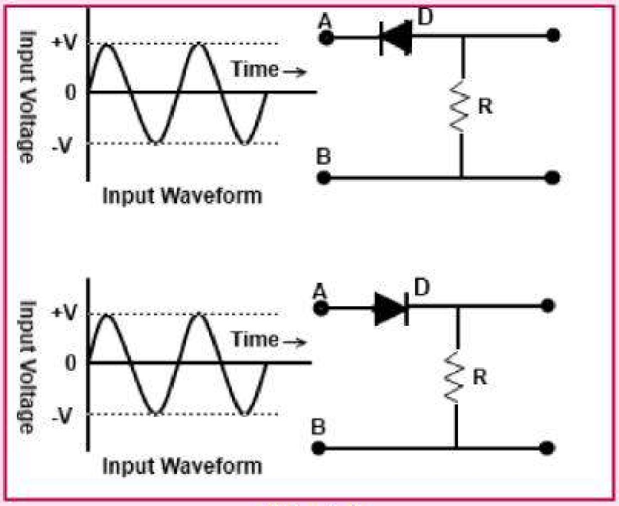


Fig 2.3

### III) Biased Clipper

Sometimes, it is required to remove a small portion of positive or negative half cycles of the signal voltage. For this purpose biased clipper is used.

#### a) Positive clipper with positive biasing

Fig 2.4 shows the circuit of a biased clipper using a diode and with a battery of $V_{dc}$ volts. With polarities of the battery shown, a portion of each positive half cycle will be clipped. However the negative half cycle will appear as such across the load. Such clipper is called positive clipper with positive biasing.

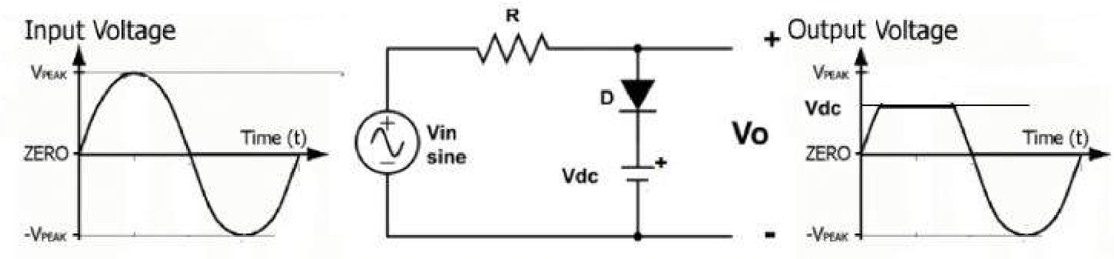


Fig 2.4: Positive clipper with positive biasing

The circuit action is as follows. The diode will conduct heavily as long as input voltage is greater than the battery voltage $V_{dc}$. When the input voltage is greater than the battery voltage $V_{dc}$, the diode behaves as a short and the output voltage equals the battery voltage $V_{dc}$. The output will stay at battery voltage $V_{dc}$ as long as the input voltage is greater than battery voltage $V_{dc}$. During the period the input voltage is less than battery voltage $V_{dc}$, the diode is reverse biased and behaves as an open circuit. Therefore, most of the input voltage appears across the output. In this way, the biased positive clipper removes the input voltage which is above the battery voltage $V_{dc}$.

During the negative half cycle of the input voltage, the diode remains reverse biased. Therefore, the entire negative cycle appears across the load.

#### b) Negative clipper with negative biasing

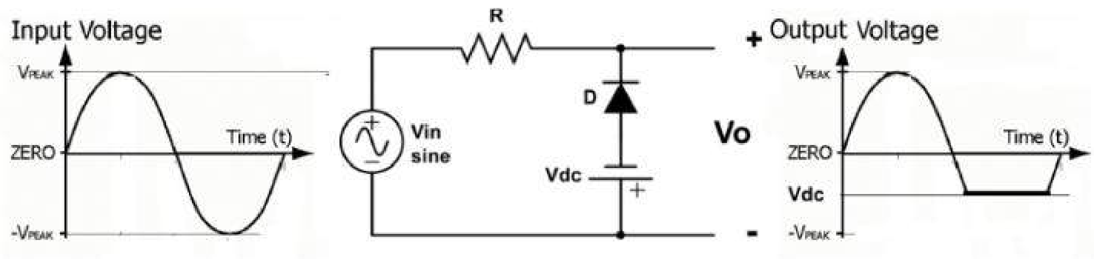


Fig 2.5 Negative clipper with negative biasing

If it is required to clip a portion of the negative cycles of the input voltage, the only thing to be done is reverse the polarities of diode and battery of positive clipper with positive biasing. Such circuit is then called a biased negative clipper.

### IV) Combinational clipper

It is a combination of biased positive and negative clippers. With a combination clipper, a portion of both positive and negative half cycles of the input voltage can be removed or clipped as shown in Fig 2.6.

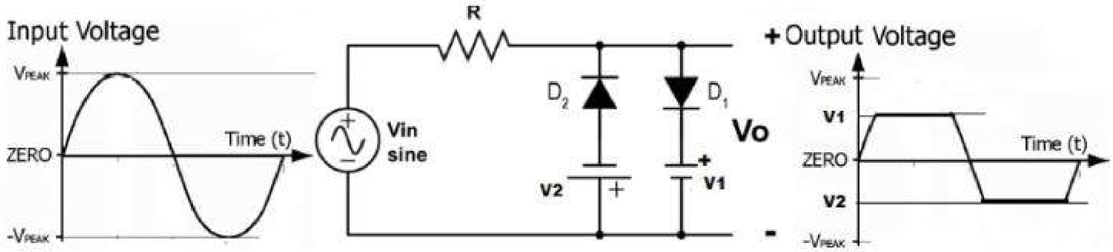


Fig 2.6 Combinational clipper

The circuit action is as follows. When positive input voltage is greater than $+V_1$, the diode $D_1$ conducts heavily while diode $D_2$ remains reverse biased. Therefore a voltage $+V_1$ appear across the load. Thus the output stays at $+V_1$ as long as the input voltage exceeds $+V_1$.On the other hand, during the negative half cycle, the diode $D_2$ will conduct heavily and the output stays at $-V_2$ as long as the input voltage is greater than $-V_2$. Note that $+V_1$ and $-V_2$ are less than $+V_m$ and $-V_m$ respectively.

Between $+V_1$ and $-V_2$ neither of the diodes is ON. Therefore, in this condition, most of the input voltage appears across the load. It is interesting to note that this clipping circuit can give a square wave output if $V_m$ is much greater than the clipping levels.

**Know your progress**

Draw the output wave form of the given circuit.

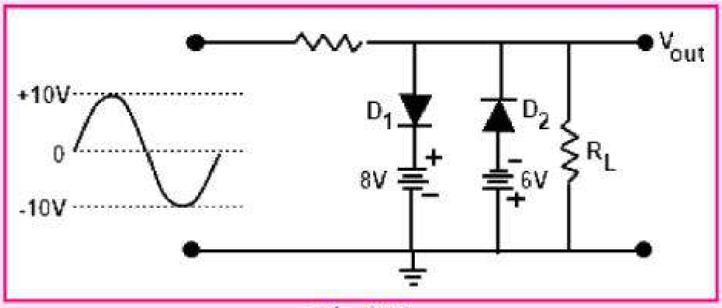


Fig 2.7

## 2.2 Clamping Circuits

A circuit that places either the positive or negative peak of a signal at a desired level by shifting its dc value is known as clamping circuit. Clamping circuits are also known as dc restorers. The network consists of a capacitor, diode and a resistive element, but it employs an independent DC supply to introduce an additional shift. The values of R and C must be chosen such that the time constant T= RC is large enough to ensure that the capacitor cannot discharge significantly during the interval when the diode is not conducting. The important clamping circuits are

I) Positive clamper

II) Negative clamper

III) Biased clamper

Fig 2.8 shows the key idea behind clamping. The input signal is sine wave having a peak-to-peak value of 2 V. The positive clamper in Fig 2.8(a) adds the dc component and pushes the signal upwards, so that, the negative peaks falls on the zero level. As you can see, the waveform now has peak values of +2 V and 0 V. It may be seen that the shape of the original signal has not changed; there is only a vertical positive shift in the signal. Such clamper is called a positive clamper. The negative clamper does the reverse. It will push the signal downwards, so that the positive peak falls to the zero level. It is shown in Fig 2.8 (b).

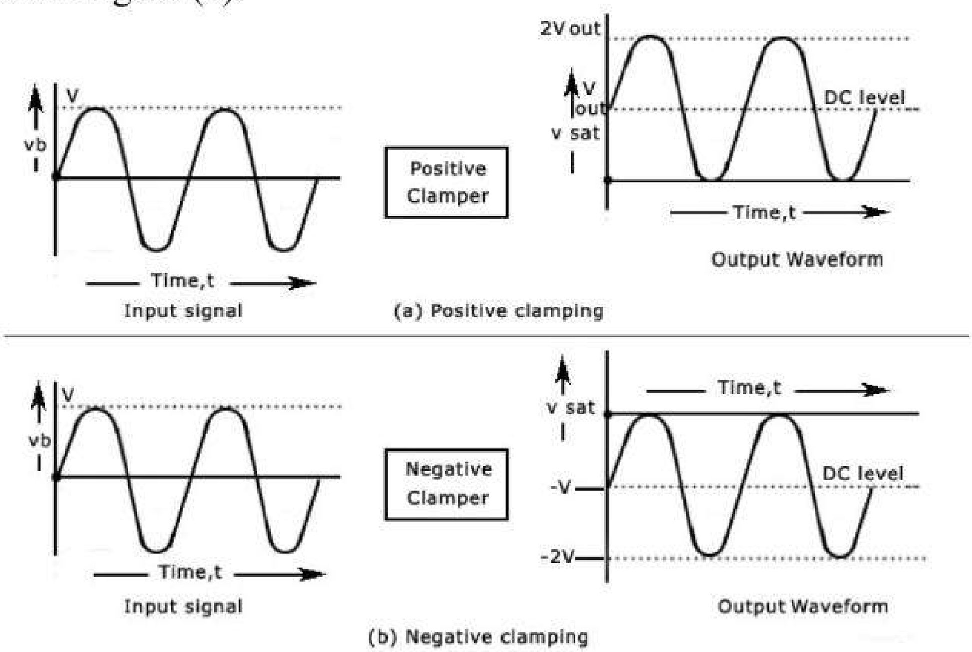


Fig 2.8 Positive and negative clamping



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### I) Negative Clamper

The circuit for a negative clamper is shown in the Fig 2.9.

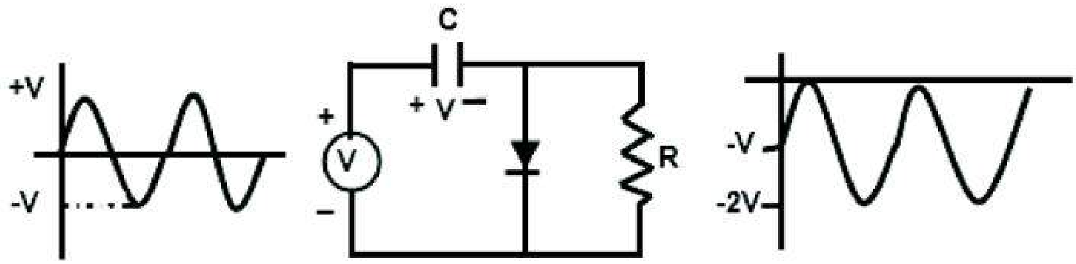


Fig 2.9 Negative clamper

Do you know that during the positive half cycle the diode conducts and acts like a short circuit as shown in Fig 2.10? The capacitor charges to peak value of input voltage $V_m$ so that the capacitor voltage is $V_C = V_m$. During this interval, the output $V_o$ is taken across the short circuit and therefore, will be equal to zero.

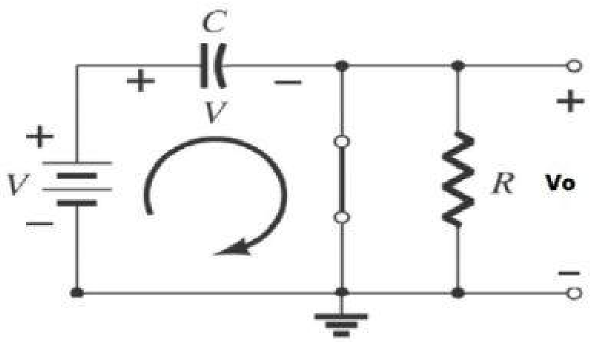


Fig 2.10

During the negative half cycle, the diode becomes reverse-biased and acts as an open-circuit as shown in Fig 2.11. Thus, there will be no effect on the capacitor voltage. The resistance R, being a very high value, cannot discharge C during the negative partion of the input wave-form. Thus during the negative input, the output voltage will be the sum of the input voltage and the capacitor voltage. The output voltage can be found by applying KVL and is equal to

$$-V_m - V_m - Vo = 0$$

$$V_O = -2V_m$$

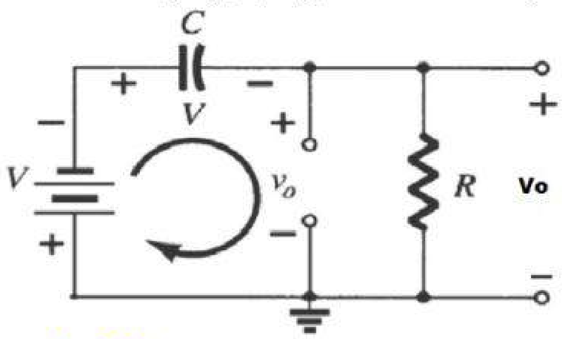


Fig 2.11

### II) Positive Clamper

The circuit for a positive clamper is shown in the Fig 2.12. During the negative half cycle of the input signal, the diode conducts and acts like a short circuit. The output voltage $V_O = 0V$. The capacitor is charged to the peak value of input voltage $V_m = V_c$ and it behaves like a battery. During the positive half of the input signal, the diode does not conduct and acts as an open circuit. Hence the output voltage, $V_O = V_m + V_m = 2V_m$. This gives a positively clamped output voltage.

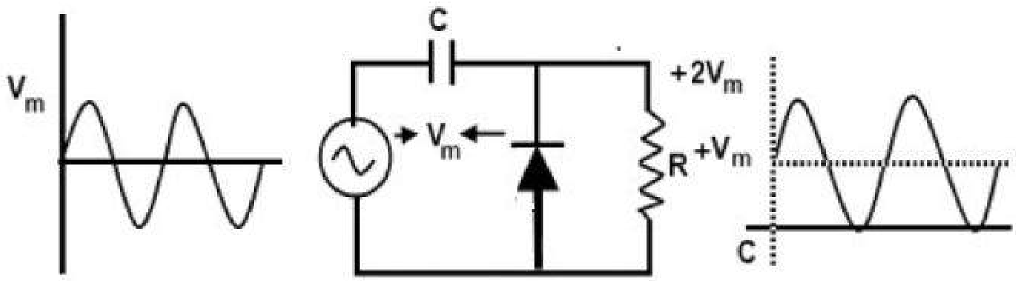


Fig 2.12 Positive Clamper

### III) Biased Clamper

The circuit of a positively biased clamper is shown in the Fig 2.13. During the negative half cycle of the input signal, the diode is forward biased and acts like a short circuit. The capacitor charges to $V_m + V_1$. Applying KVL to the input side

$$-V_m + V_C - V_1 = 0V$$

$$V_c = V_m + V_1$$

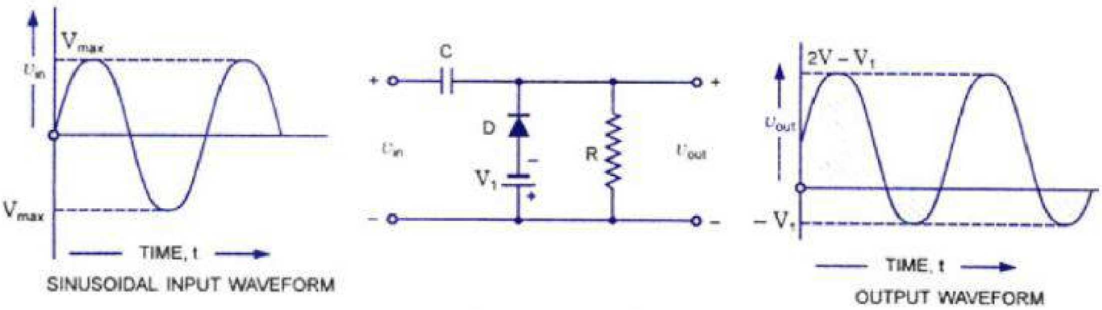


Fig 2.13 Biased positive clamper

The voltage across the resistor will be equal to the source voltage $V_{in}$. During the positive half cycle of the input signal, the diode is reverse biased and it acts as an open circuit. Hence $V_s$ has no effect on $V_o$. Applying KVL around the outside loop

$$V_m + V_c - V_O = 0\ V$$

$$V_O = V_m + V_c = V_m + V_m + V_1 = 2V_m + V_1$$

**Know your progress**

Set up a positive clamper and its output is given to a CRO. What change will you notice in the output signal, when the AC/DC switch of the CRO is kept in AC position first and then in DC position.

## 2.3 Differentiating Circuit

A circuit in which the output voltage is directly proportional to the derivative of the input is known as a differentiating circuit.

$$\text{Output} \ \alpha \ \frac{d}{dt}\text{(Input)}$$

It is a simple RC series circuit with output taken across the resistor R and it can be used as a differentiating circuit. If a DC or constant input is applied to such circuit, the output will be zero. It is because the derivative of a constant is zero.

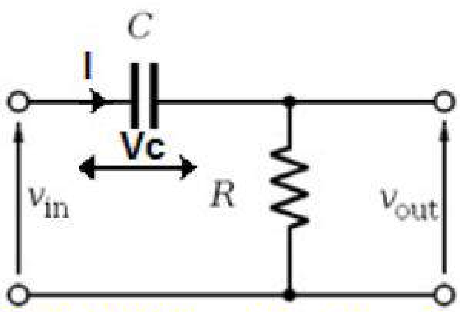


Fig 2.14 Differentiating Circuit

Fig 2.14 shows a typical differentiating circuit. The output across R will be the derivative of the input. It is important to note, that merely using voltage across R does not make the circuit a differentiator: it is also necessary to set proper circuit values for R and C. In order to achieve good differentiation, the following two conditions should be satisfied:

(i) The time constant RC of the circuit should be much smaller than the time period of the input wave.

(ii) As a thumb rule the value of $X_C$ $(1/2\pi fC)$ should be 10 or more times larger than R at the operating frequency f.

Fulfilling these conditions, the output across R in fig 2.14 will be the derivative of the input.

Let $V_i$ be the input alternating voltage and let I be the resulting alternating current. The charge Q on the capacitor at any instant is

$$Q = C\,Vc$$

$$I = \frac{dQ}{dt} = \frac{d}{dt}(Q) = \frac{d}{dt}(CVc)$$

$$I = C\frac{d(Vc)}{dt}$$

Since the capacitive reactance is very much larger than R, the input voltage can be considered to be equal to the capacitor voltage with negligible error ie: Vc=Vi

$$I = C\frac{d(Vi)}{dt}$$

Output voltage $V_o$=IR

$$= RC\frac{d(Vi)}{dt}$$

$$\alpha \frac{dVi}{dt} \ (\therefore \text{RC is constant})$$

$$\text{Output voltage } \alpha \ \frac{d}{dt}(\text{input})$$

The output waveform from a differentiating circuit depends upon the time constant and the shape of the input wave. If the input fed to a differentiating circuit is a square wave, the output will consist of sharp narrow pulses (spikes) as shown in fig 2.15,

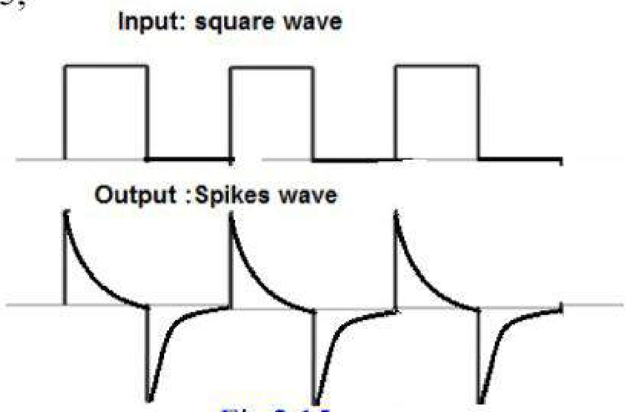


Fig 2.15



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## 2.4 Integrating Circuit

A circuit in which the output voltage is directly proportional to the integral of the input is known as integrating circuit.

$$\text{Output} \propto \int \text{input}$$

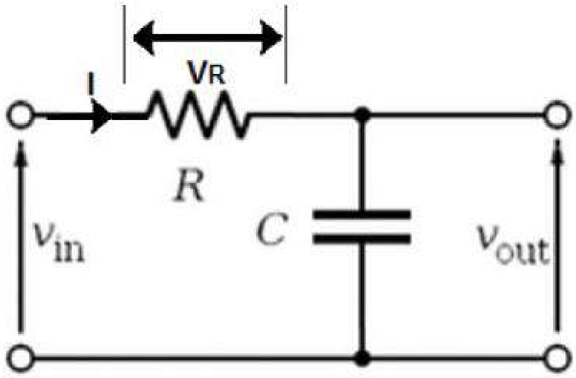


Fig.2.16 Integrating Circuit

An integrating circuit is a simple RC series circuit with output taken across the capacitor C as shown in fig 2.16. It can be seen that, R and C of the differentiating circuit have interchanged their positions. In order that the circuit renders good integration, the following conditions should be fulfilled:

i) The time constant RC of the circuit should be very large as compared to the time period of the input wave.

ii) The value of R should be 10 or more times larger than Xc.

Let $V_i$ be the input alternating voltage and let I be the resulting alternating current. Since R is very large as compared to capacitive reactance Xc of capacitor, it is reasonable to assume that the voltage across R (ie: $V_R$) is equal to the input voltage. That is

$$V_i = V_R$$

$$\text{Now } I = \frac{V_R}{R} = \frac{V_i}{R}$$

The charge Q on the capacitor at any instant is

$$Q = \int I dt$$

$$\text{Output voltage } V_o = \frac{Q}{C} = \int \frac{I dt}{C}$$



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$$= \frac{\int \frac{Vi}{R} dt}{C}$$

$$= \frac{1}{RC} \int Vi \ dt$$

$\alpha \int$ Vi dt (RC is constant)

$\therefore$ Output $\alpha \int$ input

The output waveform of an integrating circuit depends upon the time constant and shape of the input wave. When the input fed to an integrating circuit is a square wave, the output will be triangular wave as shown in Fig 2.17.

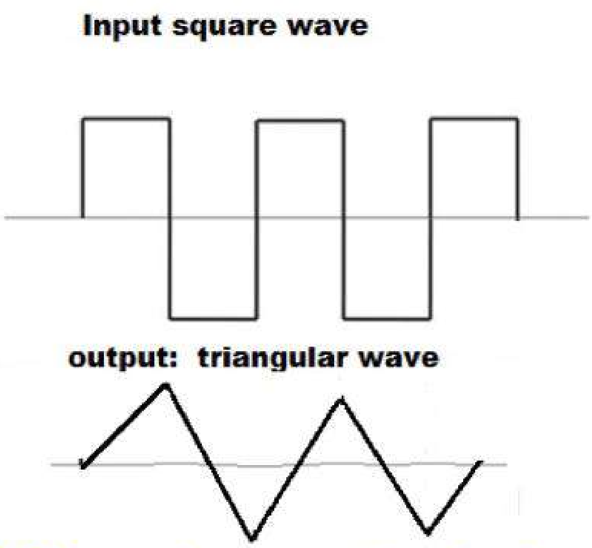


Fig 2.17 Input and output waveform of an integrator

**Know your progress**

1. Predict the output of a differentiator if the input signal is a sine wave.
2. Set up a differentiator circuit in which a sine wave is to be given as input. Observe both the input and output of this circuit simultaneously in a CRO using a dual input mode. Find out the difference.

   Repeat this for an integrator also.

## 2.5 OP-AMP Circuits

This section shows how the inverting and non inverting configurations of op-amp are useful in applications such as voltage follower, adder, subtractor, integrator, differentiator and comparator.

### i) Voltage Follower (Buffer)

The lowest gain that can be obtained from a non inverting amplifier with feedback is 1.When the non inverting amplifier is configured for unity gain, it is called ***voltage follower*** because the output voltage is equal in amplitude and phase with the input. In other words, in the voltage follower the output voltage follows (tracks) the input voltage.

Although it is similar to the discrete emitter follower, the voltage follower is preferred to because it has a much higher input resistance, and the output amplitude is exactly equal to the input.

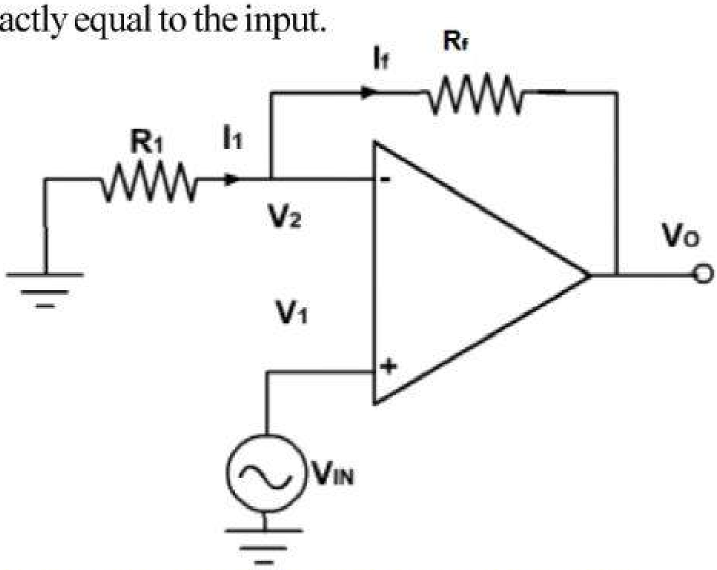


Fig 2.18 Non inverting amplifier configuration

We have studied about the non inverting configuration the previous year. Fig 2.18 shows the non inverting amplifier configuration and we know that the voltage gain

$$A = 1 + \frac{R_F}{R_1}$$

To obtain the voltage follower from the non inverting amplifier, simply remove $R_1$ and short $R_F$ ($R_F$ =0). When the voltage gain of the amplifier is A=1, the amplifier is called **unity gain buffer.** The resulting circuit is shown in Fig 2.19. In this figure all the output voltage is fed back into the inverting terminal of the op-amp; consequently, the gain of the feedback circuit is 1.

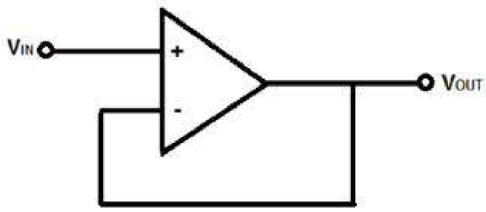


Fig 2.19 Voltage follower

A voltage buffer amplifier is used to transfer voltage from one circuit, having high output impedance level, to the following circuit with a low input impedance level. The interposed buffer amplifier prevents the second circuit from loading the first circuit unacceptably and interfering with its desired operation. This is also called impedance matching which is done to ensure maximum power transfer from the preceding stage to the next stage. For example, impedance matching should be applied between a power amplifier stage and loud speaker for maximum power transfer to the loud speaker. This is because the output impedance of the power amplifier is comparatively large and the impedance of the loudspeaker is small. In the ideal voltage buffer, the input resistance is infinite and the output resistance is zero.

**i) Summing Amplifier (Adder Circuit)**

Fig 2.20 shows the inverting configuration with three inputs Va, Vb and Vc. Depending on the relationship between the feedback resistor $R_F$ and the input resistors Ra, Rb and Rc ,the circuit can be used as a summing amplifier. The circuit function can be verified by examining the expression for the output voltage, Vo.

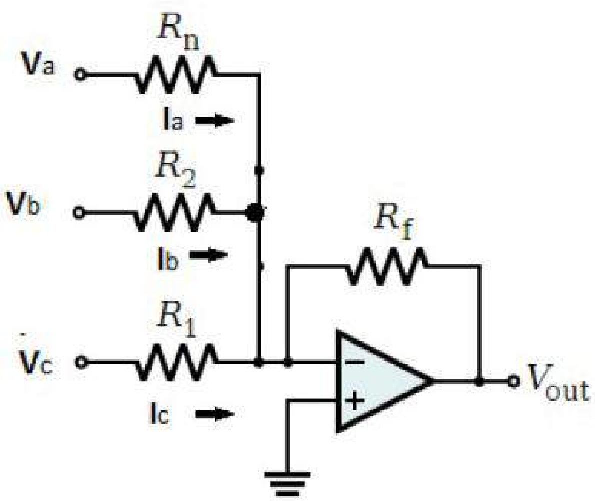


Fig 2.20 Summing Amplifier

$$Vo=-\left(\frac{Rf}{Ra}Va+\frac{Rf}{Rb}Vb+\frac{Rf}{Rc}Vc\right)$$

If Ra = Rb = Rc = R then the equation can be rewritten as

$$Vo=\frac{-Rf}{R}(Va+Vb+Vc)$$



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This means that the output voltage is equal to the negative sum of all the inputs times the gain of the circuit Rf / R; hence the circuit is called a summing amplifier. Obviously, when the gain of the circuit is 1 that is Ra = Rb = Rc = Rf, the output voltage is equal to the negative sum of all the input voltages. Thus

Vo = - (Va + Vb + Vc)

We see that the output is the sum of input voltages. The negative sign can be avoided by using an inverting amplifier with unity gain ($R_f=R_1$) at the output of the summing amplifier.

**Know your progress**

Find the output of a summing amplifier with three input voltages Va = 2 V, $V_b$ = 1V and $V_c$ = 3V.

### ii) Subtractor Circuit

A basic differential amplifier can be used as a subtractor as shown in Fig 2.21.Here, all the external resistors are equal in value, so the gain of the amplifier is equal to 1.From this figure, the output voltage of the differential amplifier with gain of 1 is

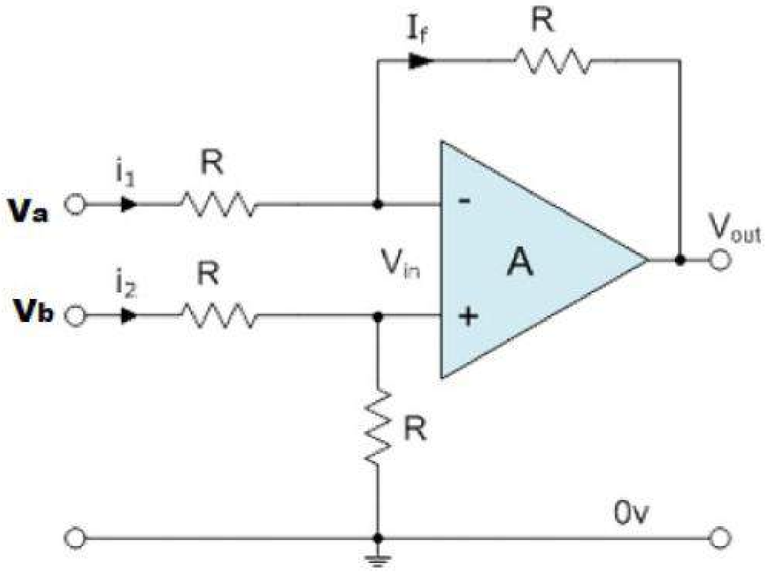


Fig 2.21 Subtractor Circuit using OPAMP

$$Vo = \frac{R}{R}(Vb - Va)$$

That is, Vo = Vb - Va

Thus the output voltage Vo is equal to the voltage $V_b$ applied to the non inverting terminal minus the voltage $V_a$ applied to the inverting terminal; hence, the circuit is called a subtractor.

### iii) The Integrator

A circuit in which the output voltage waveform is the integral of the input voltage waveform is called an integrator or integration amplifier. Such a circuit is obtained by using a basic inverting amplifier configuration and the feedback resistor $R_F$ is replaced by a capacitor $C_F$. The expression for the output voltage $V_O$ is

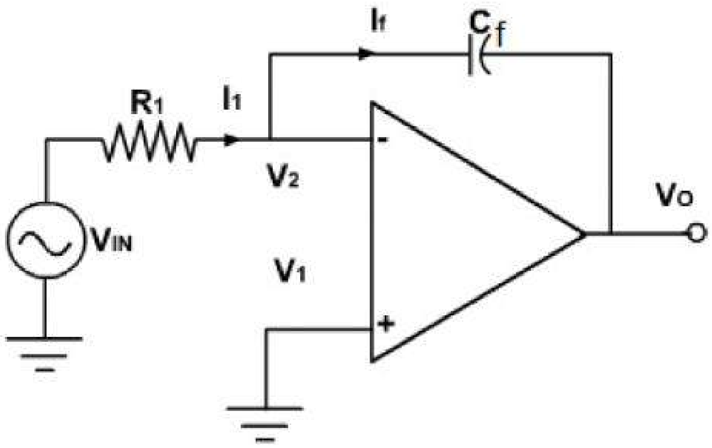


Fig 2.22 Integrator Circuit using OPAMP

$$Vo = \frac{-1}{R_1 C_F} \int Vin\, dt$$

The equation indicates that the output voltage is directly proportional to the the negative integral of the input voltage and inversely proportional to the time constant $R_1 C_F$. For example, if the input is a sine wave, the output will be a cosine wave. If the input is square wave, the output will be a triangular wave as shown in Fig 2.23.

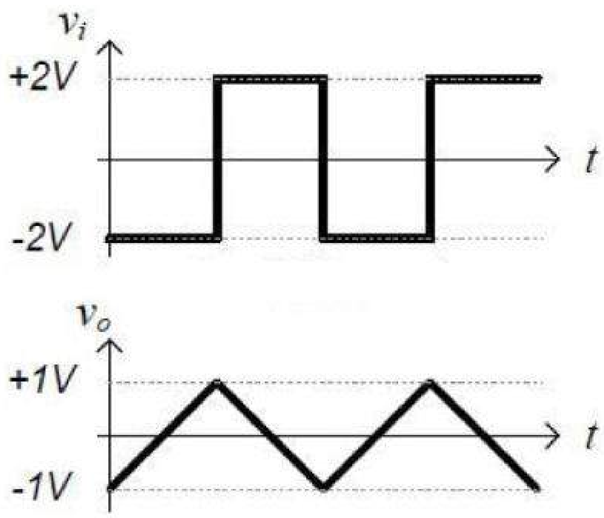


Fig 2.23 Output waveform of integrator

The advantage of op- amp integrator over the simple RC integrator is that integration will be good for a large range of frequencies in an op-amp integrator. But in a simple RC integrator good integration can be obtained only for a very small range of frequencies.

**iv) The Differentiator**

Fig 2.24 shows the differentiator or differentiating amplifier. As its name implies, the circuit performs the mathematical operation of differentiation; that is the output wave form is the derivative of the input waveform. The differentiator may be constructed from a basic inverting amplifier, if the input resistor $R_1$ is replaced by a capacitor $C_1$. The expression for the output voltage $V_O$ can be expressed as

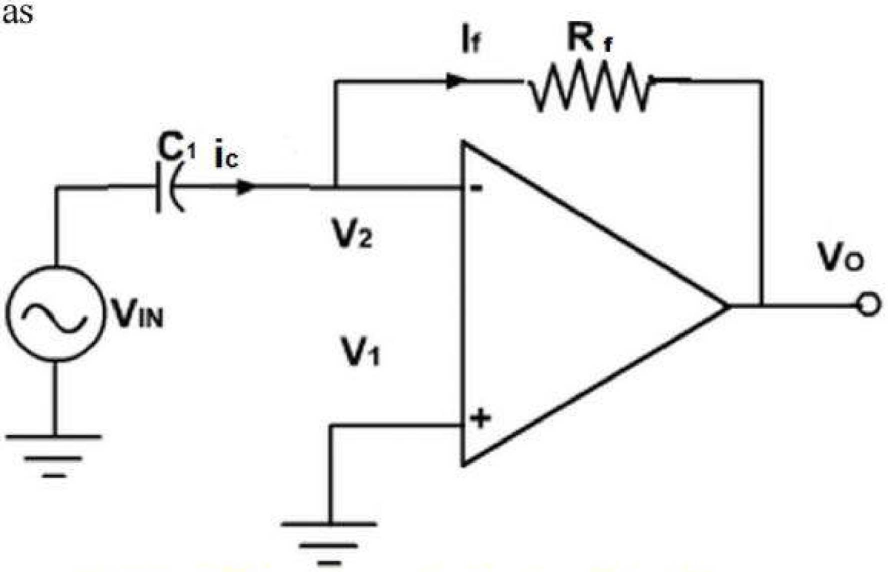


Fig 2.24 Differentiator circuit using OP-AMP

$$Vo = -R_F C_1 \frac{dVin}{dt}$$

Thus output $V_o$ is equal to $R_F C_1$ times the negative instantaneous rate of change of the input voltage $V_{in}$ with time. If the input fed to a differentiating circuit is a square wave, the output will consist of sharp narrow pulses (spikes) as shown in fig 2.25.

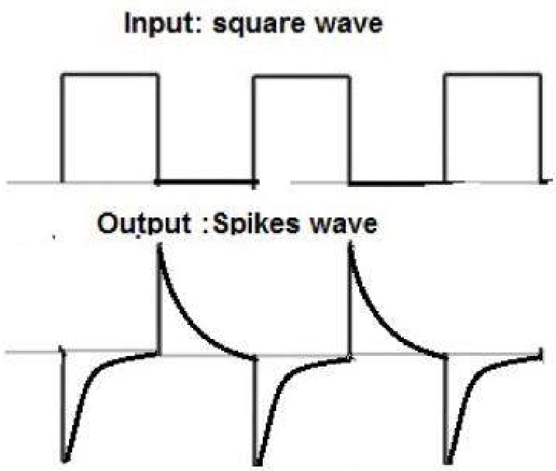


Fig 2.25 Differentiator- output waveform

In op amp differentiator also, the differentiation will be good for a wide range of frequencies compared to a simple RC differentiator.

### vi) Comparator

A comparator, as its name implies, compares a signal voltage of one input of an op-amp with a known voltage called the reference voltage of the other input. Comparators are used in circuits such as digital interfacing, Schmitt triggers, voltage level detectors and oscillators.

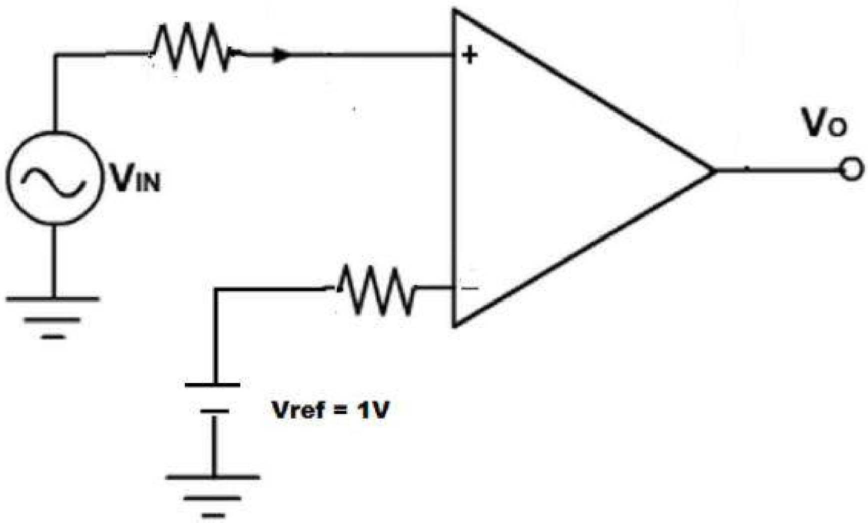


Fig 2.26 Comparator circuit

Fig 2.26 shows an op-amp used as a comparator. Here open loop configuration (no negative feedback) is used. A fixed reference voltage $V_{ref}$ of 1V is applied to the negative input and the other time varying signal voltage $V_{in}$ is applied to the positive input. Because of this arrangement, this circuit is called the non inverting comparator. When $V_{in}$ is less than $V_{ref}$, the output voltage $V_o$ is at $-V_{sat}$ ($\cong -V_{EE}$) because the voltage at the negative input is higher than that at the positive input. On the other hand, when $V_{in}$ is greater than $V_{ref}$, the positive input becomes positive with respect to the negative input, the Vo is $+V_{sat}$ ($\cong +V_{cc}$). Thus Vo changes from one saturation level to the other whenever $V_{in}$ oscillates above and below $V_{ref}$, as shown in fig 2.27.

In short, the comparator is a type of analog-to-digital converter. At any given time the $V_o$ waveform shows whether $V_{in}$ is greater or lesser than $V_{ref}$. The comparator is sometimes called a voltage-level detector because for a desired value of $V_{ref}$, the voltage level of the input $V_{in}$ can be detected.

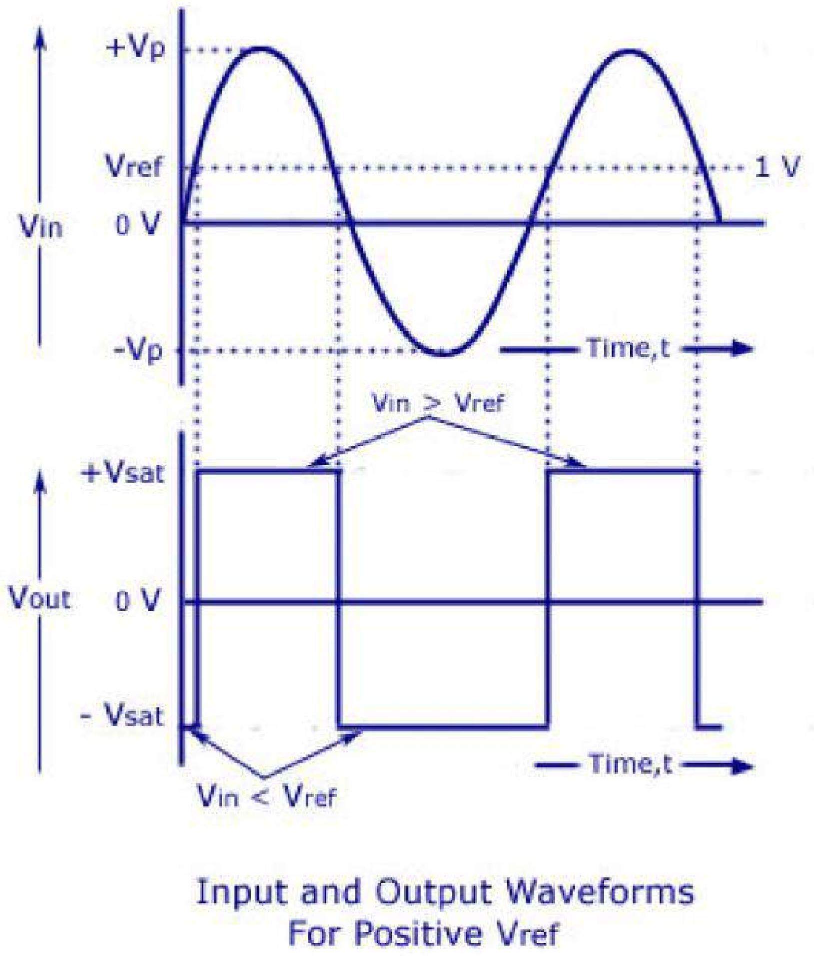


Fig 2.27 Output waveform of comparator

## 2.6 Filters

In electronic systems the desired signal is often affected by unwanted signals called noise. Signals are distinguished by their frequency characteristics and some form of frequency selective circuit or filter can accomplish the extraction of signals from the noise. A filter is a frequency selective circuit that allows a band of frequencies to pass through it and blocks or attenuates signals of frequencies outside this band. A general classification of filters is shown in Fig. 2.28.

Depending on the type of elements used in their construction, filters may be classified as passive and active. Elements used in passive filters are resistors, capacitors and inductors. Active filters on the other hand, employ transistor or op-amps in addition to resistors and capacitors. Depending upon the frequency range, filters are divided into audio frequency and radio frequency filters.

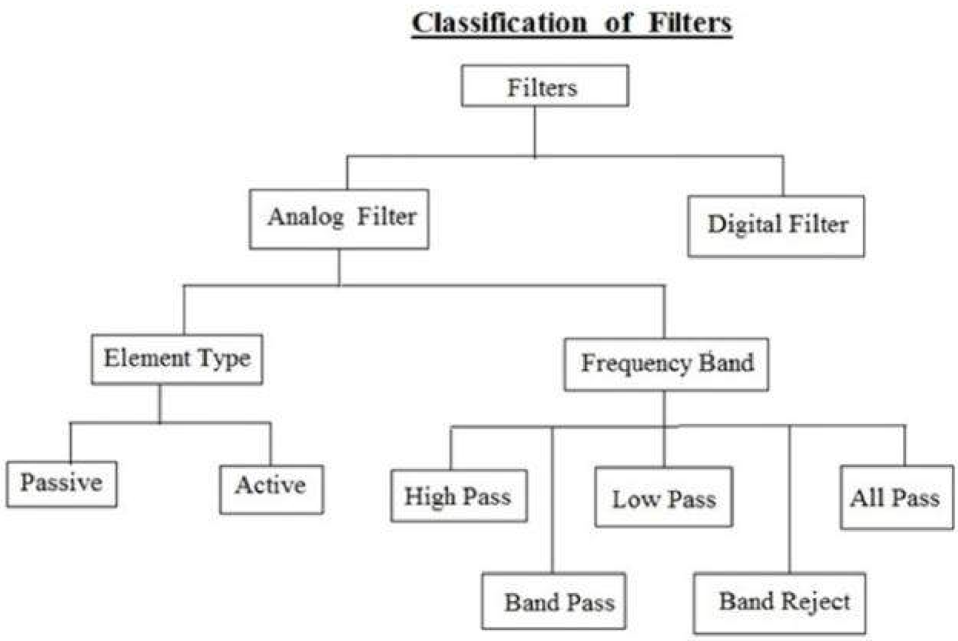


Fig 2.28 Filter types

Audio frequency filters are low frequency filters which commonly use resistors and capacitors, while radio frequency filters use inductor, capacitor or crystals. The most commonly used filters are

a) Low pass filter
b) High pass filter
c) Band pass filter
d) Band stop (rejection) filter

The figure 2.29 shows the ideal frequency response of these filters. However in practice, this response cannot be attained. The band of frequencies that are allowed to pass through a filter is called its *pass-band*. The band of frequencies that are not allowed to pass through them is called *stop band* or *attenuation band*. The frequency that differentiates between pass and stop band is called cut-off frequency. An ideal filter will have an amplitude response that is unity or a fixed value in the pass band and zero in the stop band.

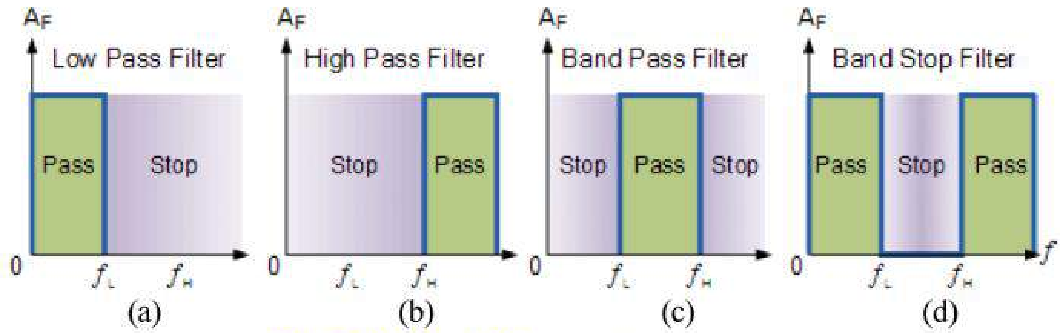


Fig 2.29 Ideal filter response curves

Figure 2.29 (a) shows frequency response of an idealized low pass filter. In this filter the low frequencies are in the pass band and the higher frequencies are in the stop band. Figure 2.29 (b) shows the idealized high-pass filter frequency response. Here, the low frequencies are in the stop-band, and the high frequencies are in the pass band. If a high-pass filter and a low-pass filter are cascaded, a band pass filter is created. The band pass filter passes a band of frequencies between a lower cutoff frequency, $f_L$ and an upper cutoff frequency, $f_H$. Frequencies below $f_L$ and above $f_H$ are in the stop band. An idealized band pass filter frequency response is shown in Figure 2.29(c).

Figure 2.29(d) shows the idealized frequency response of the *band-reject* or *band-stop* or *notch* filter. The pass bands include frequencies below $f_L$ and above $f_H$. The band from $f_L$ to $f_H$ is in the stop band.

### i) Low pass filter (LPF)

A low pass filter (LPF) is a filter which passes low-frequency signals and blocks high frequency signals. In other words, low-frequency signals go through the filter much easier and with less resistance and high-frequency signals have a much harder getting through, which is why it is a low pass filter.

Low pass filters can be constructed using resistors or with either capacitors or inductors. A low pass filter composed of a resistor and a capacitor is called a low pass RC filter. And a low pass filter with a resistor and an inductor is called a low pass RL filter.

A low pass RC filter, again, is a filter circuit composed of a resistor R and capacitor C which passes low-frequency signals, while blocking high frequency signals. A low pass RC filter is shown in the circuit below:

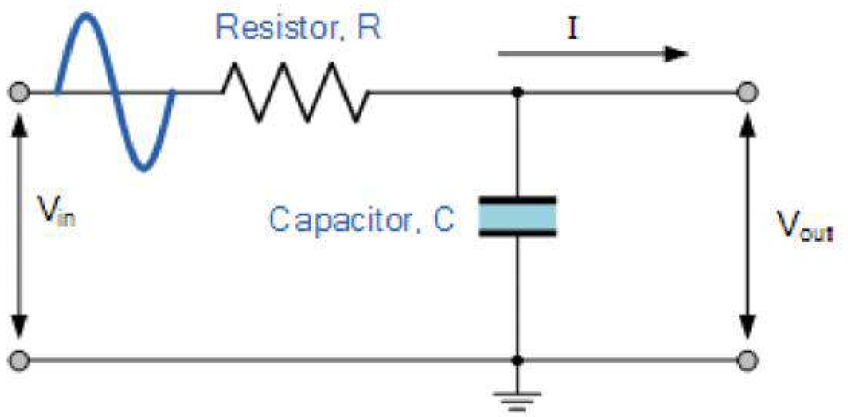


Fig 2.30 RC Low Pass Filter Circuit



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As capacitor is a reactive device, it offers differing resistance to the signals of different frequencies passing through it. It offers very high resistance to low-frequency, or DC, signals and low resistance to high-frequency signals. So the DC as well as low frequency signals will be dropped across the capacitor and will be available at the output. The high-frequency signals will go through the capacitor, the capacitor offers them a very low-resistance path. So high frequency signal will be dropped across the resistor and not reaches the output. Fig 2.31 shows frequency response of low pass filter.

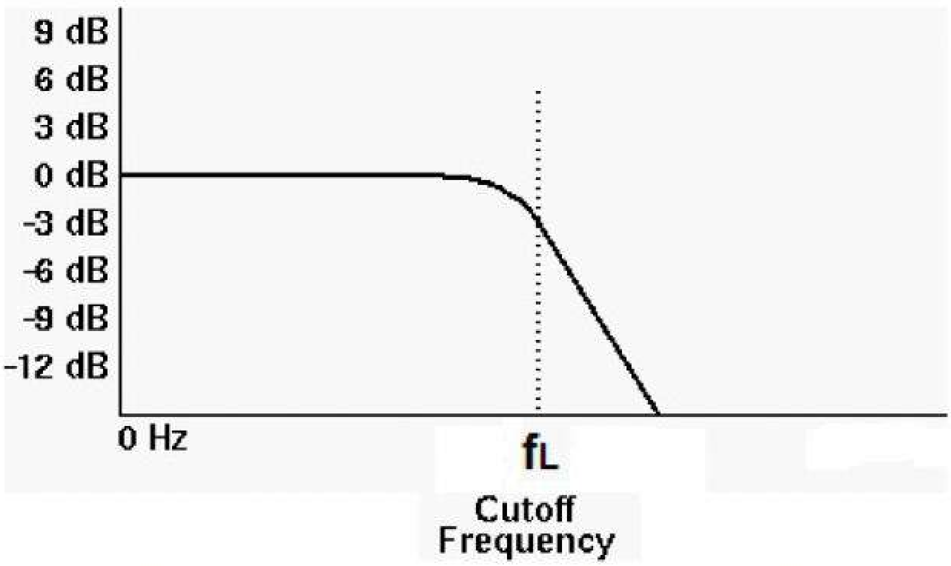



Fig 2.31 Frequency Response of a Low Pass Filter.

The cut-off frequency of low pass filter is given by $f_L = 1/2\pi RC$

Applications of passive low pass filters are seen in audio amplifiers and speaker systems where they are used to direct the low frequency bass signals to the large bass speakers or to reduce any high frequency noise or "hiss" type distortion.

### ii) High Pass Filter

A high pass filter (HPF), is obtained by interchanging the positions of the components R and C of the low pass filter and the output signal ( $V_{out}$ ) is taken across the resistor as shown in Fig 2.32.

A low pass filter allows signals with frequency below its cut-off frequency $fc$ to pass, whereas a high pass filter circuit as its name implies, only passes signals above the selected cut-off point, $fc$ eliminating any low frequency signals. Consider the circuit shown in Fig.2.32.

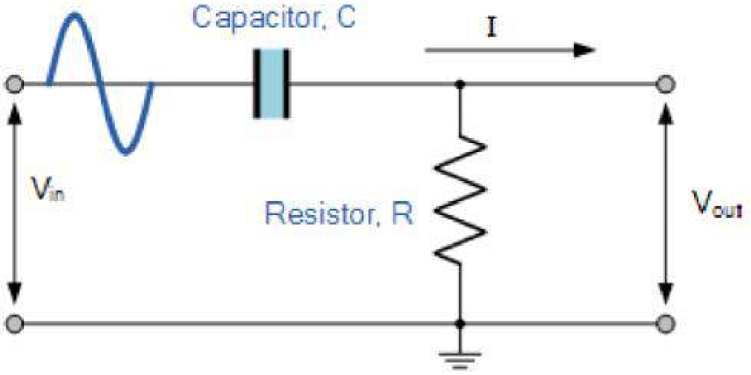



Fig 2.32 .The High Pass Filter Circuit

In this circuit arrangement, the reactance of the capacitor is very high at low frequencies so the capacitor acts like an open circuit and blocks any input signals until the cut-off frequency point ( $f_H$ ) is reached. Above this cut-off frequency point, the reactance of the capacitor is reduced sufficiently so that only a small amount of input signal is dropped across the capacitor and the remaining input reaches the output. When the frequency is very large, the reactance of the capacitor becomes almost zero and hence, most of the input signal reaches the output.

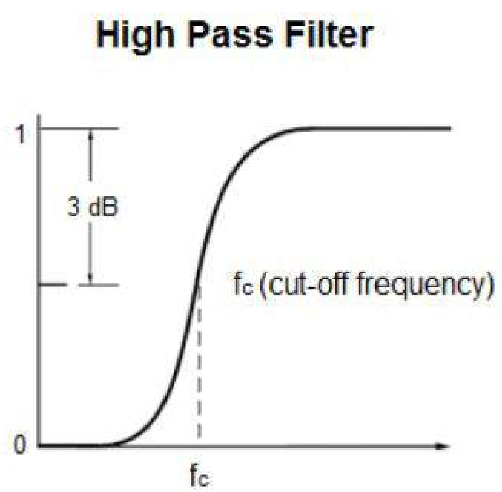



Fig 2.33 Frequency Response of a High Pass Filter

The cut-off frequency of high pass filter is given by $f_H = 1/2\pi RC$

A very common application of a passive high pass filter is in speaker systems where it is used to direct the high frequency signals to the small "tweeter" type speakers while blocking the low bass signals or is also used to reduce any low frequency noise or 'rumble' type distortion. When used like this in audio applications, the high pass filter is sometimes called a "low-cut", or "bass cut" filter.

### iii) Band Pass Filter

Sometimes it is necessary to pass signals of certain range of frequencies and this range can be anywhere between 0 Hz and infinity.

By connecting or 'cascading' together a single **Low Pass Filter** circuit with a **High Pass Filter** circuit, we can produce another type of passive RC filter that passes a selected range or 'band' of frequencies that can either be narrow or wide while attenuating all those outside this range. This new type of passive filter arrangement produces a frequency selective filter commonly known as a **Band Pass Filter** (BPF). Fig 2.34 shows the circuit diagram of band pass filter.

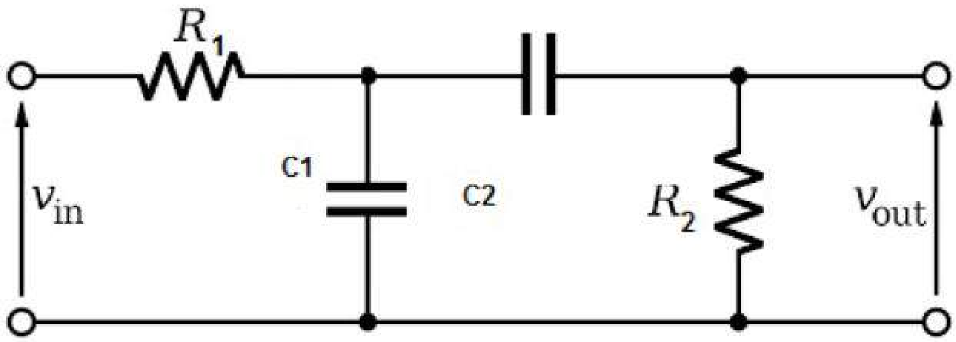



Fig 2.34 Band Pass Filter Circuit

**Know your progress**

Fig 2.35 shows the practical circuit of band pass filter. Draw its frequency response after finding the lower and upper cut off frequencies.

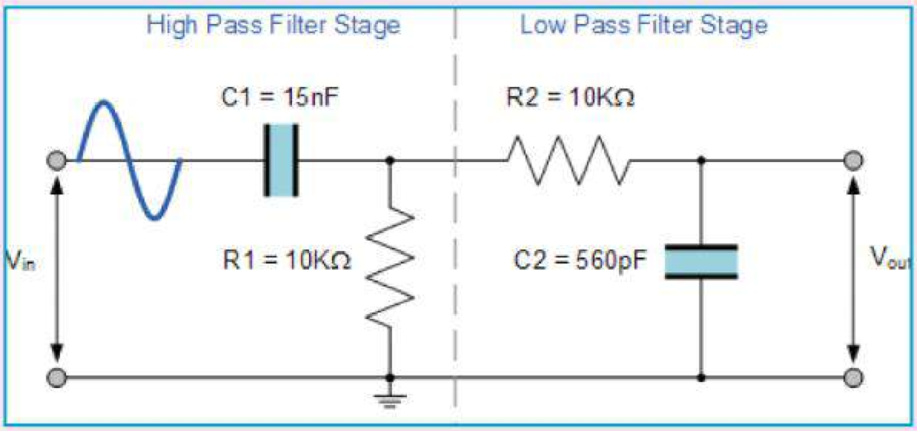



Fig 2.35 Practical Band pass filter circuit

Fig 2.36 shows how the frequency response of a band pass filter is arranged using high pass and low pass filter responses and Fig 2.37 shows the frequency response of a practical band pass filter. The high pass filter removes all frequencies below the cut off frequency $f_L$ and low pass filter removes all frequencies above the cut off $f_H$.

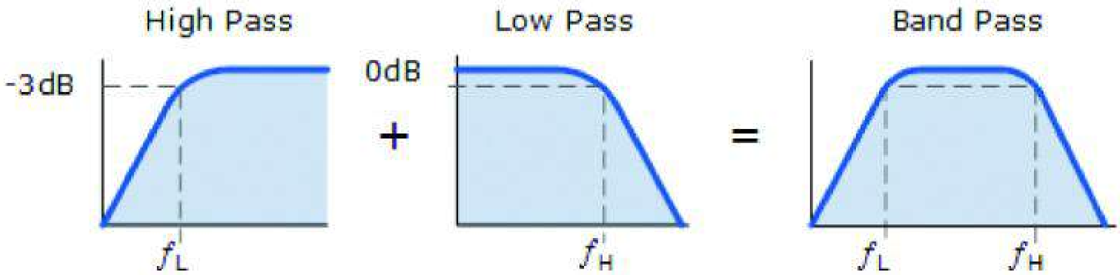



Fig 2.36 Combining frequency responses of LPF and HPF

Here, the cut off frequency of the LPF is high compared to the cut off frequency of the HPF. The difference between their cut off frequencies is the width of the pass band of the band pass filter.

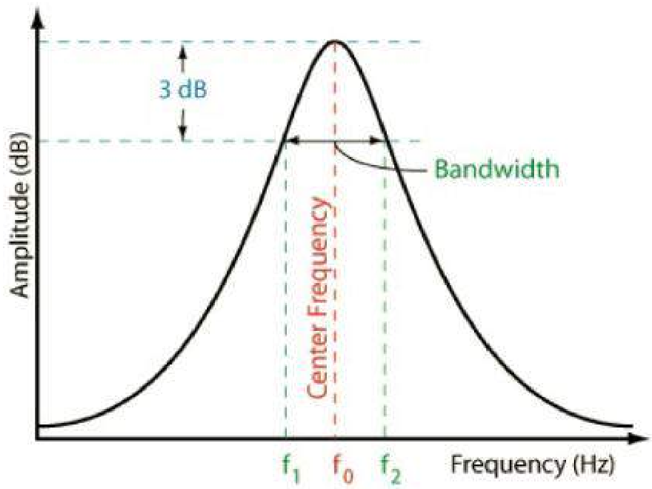



Fig 2.37 Frequency Response of a Band Pass Filter

### iv) Band Reject Filters

A filter that allows all the frequencies other than a band of frequency is called band rejection filter. Such filters are also called as band stop filter. Its frequency response is opposite to that of a band pass filter. Fig 2.38 shows the block diagram of band reject filter. Fig 2.39 shows the frequency responses of a band rejection filter.

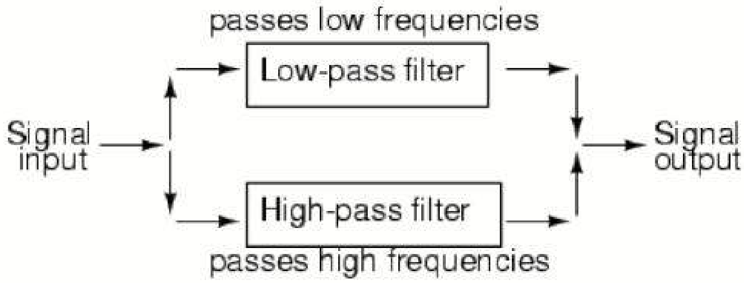



Fig 2.38 Block diagram of band reject filter

Here, the cut off frequency of the LPF should be small compared to the cut off frequency of the HPF. The difference between their cut off frequencies is the width of the stop band of the band reject filter.

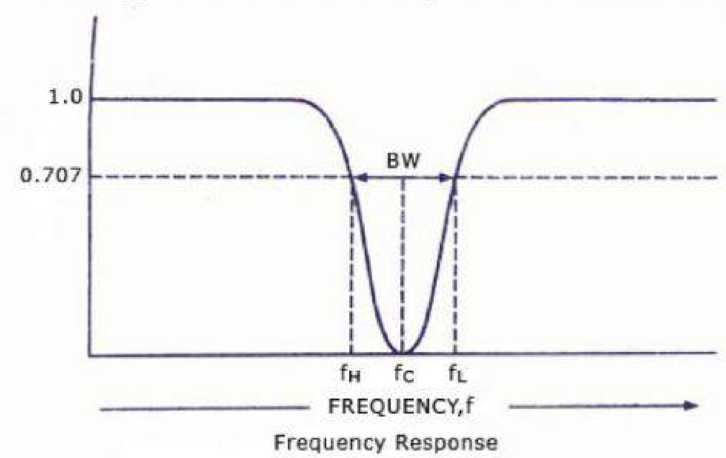



Fig 2.39 Frequency Response of a Band Reject Filter

## Let us consolidate

Wave shaping is the process of modifying a signal to obtain it in desired shape. It includes differentiation, integration, clipping, clamping, etc. The components like resistor, capacitor, inductor, diode etc are used for this purpose. Clipping circuits are used to remove a part of the signal. The different types of clippers are positive clipper, negative clipper, biased clipper and combinational clipper. A positive clipper removes positive half of a signal and a negative clipper removes the negative half. A biased clipper removes a portion of either positive half or negative half of the signal depending on the bias voltage. A combinational clipper removes a portion of both positive and negative halves and the level of removal depends on the bias voltages. Clamping circuits add a DC voltage to a signal so that it will be shifted up or down depending on the polarity of the added DC voltage. This addition of DC voltage is achieved not with a battery but with a large value capacitor. If the signal is shifted up, it is positive clamping and if it is shifted down, it is negative clamping. In biased clamping the level of shifting up or down can be decided with a DC power supply of the given voltage.

A differentiator circuit produces the differential or derivative of the input signal at its output. It is a simple RC network. The quality of differentiation depends on the values of R and C and the frequency of the input signal. An integrator is also a RC network for which the output is the integral of the input signal. Here also the quality of integration depends on the values of R and C and the frequency of the input signal. A voltage follower or buffer circuit is usually used for impedance matching. It has unit voltage gain or the output signal is same as the input signal. A summing amplifier or adder circuit can be used to obtain the sum of input voltages at the input terminals of the circuit. An op-amp based circuit is used for this purpose. The basic differentiator and integrator circuits can be modified using op-amp circuits so that their performance will be improved. A comparator is an op-amp circuit in which the voltage level of an input signal is compared with a fixed reference voltage.

Filter circuits are used to separate signals of different frequencies. Depending on the frequency characteristic, the filters can be broadly classified as LPF, HPF, BPF and Band reject filters.

The contents of this unit were learned through general discussion, sketching wave forms, designing circuits and experimentation.

## Let us asses

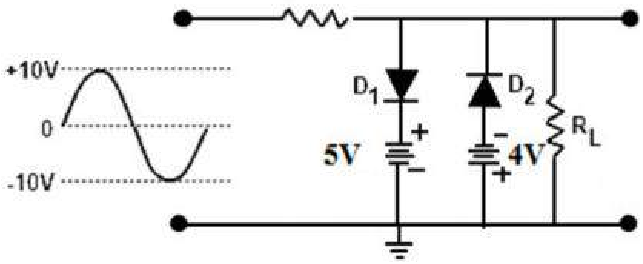
1. Clipping circuits are used to remove or clip off a part of the signal.

   a) A positive clipper is that which removes the …….. half cycles of the input signal :

   i) negative ii) positive iii) both positive and negative

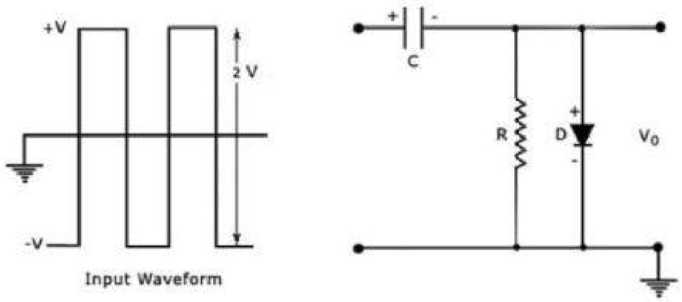
   b) Describe the working of a negative clipper with neat diagram.

   c) Draw the output waveform for the clipper circuit shown below.




2. A circuit that places either the positive or negative peak of a signal at a desired DC level is known as a clamping circuit.

   a) Draw the output waveform for the clamping circuit shown below.




3. A circuit in which output voltage is directly proportional to the derivative of the input is known as a differentiating circuit. Sketch the output waveform of a differentiating circuit when the input is a square wave.

4. A circuit in which output voltage is directly proportional to the integral of the input is known as integrating circuit.

   a) An integrating circuit is a simple RC series circuit with output taken across………

   i) both R and C ii) R iii) C iv) None of the above

   b) Show that the output from an integrating circuit is the integral of the input.

5. In a voltage follower, the output voltage follows (tracks) the input voltage.

   a) In the ideal voltage buffer, the input resistance is ..........

   b) Draw the voltage follower circuit using OPAMP.

   c) Voltage follower circuit is also called unity gain buffer. Why?

6. In summing amplifier the output voltage is equal to the negative sum of all the inputs times the gain of the circuit $Rf/R$. For a summing amplifier the three inputs voltages are $V_a$ = 2.5V , $V_b$ =2 Vand Vc= 1.5V. Find the output voltage.

7. A comparator, compares a signal voltage of one input of an op-amp with a known voltage called the reference voltage of the other input.

   a) Draw and explain the working of comparator circuit.

   b) Mention the important applications of the comparator.

8. An electric filter is a frequency selective circuit that allowing a band of frequencies and blocks or attenuates signals of frequencies outside this band.

   a) Draw the low pass filter circuit and describe its working.

   b) List down the applications of LPF.

9. ......... filter is used to reduce any low frequency noise or "rumble" type distortion.

10. Sketch the practical frequency response of a high pass filter.

11. A filter that allows all the frequencies other than a band of frequency is called band rejection filter.

    a) Compare band pass and band rejection filter.

    b) Describe the working of BPF with help of a circuit diagram.

# 3 DIGITAL ELECTRONICS

## Significant Learning Outcomes

*After completing this chapter the learner:*

- explains the difference between combinational and sequential circuits.
- demonstrates the operation of multiplexer and demultiplexer.
- sketches the circuits of multiplexer and demultiplexer.
- explains the operation of encoder and decoder.
- designs and sets up a one bit comparator.
- explains the operation of the basic flip flops.
- sketches the circuit and prepares the truth tables of these flip flops.
- explains the operation of a counter.
- sketches the circuit of a shift register with D FFs.
- explains the types of shift registers.

Digital systems are gaining widespread acceptability in electronic circuits and designs because of the developments in microcontrollers and sophisticated computers. These systems process digital data. Digital signal processing and digital image processing techniques are utilised in various communication and entertainment systems. The ability to reduce the effects of noise makes digital communication superior to analog communication. The TV transmission has almost fully changed to digital system. Thus it is very important to know the fundamentals of digital systems and their basic circuits. As we discussed the previous year, there are two types of digital circuits namely, combinational circuits and sequential circuits. In a combinational circuit at any moment the output entirely depends on the inputs present at that moment. In the design of combinational circuits AND, OR, and NOT operations are required. Examples of combinational circuits are adder, subtractor, encoder and decoder. We have already studied adder circuits, the previous year. In a sequential circuit, at any moment the output depends on the past outputs as well as the input present at that moment. The past outputs depend on the past or previous inputs.

Thus we need 'memory' to store the past outputs in a sequential circuit. In other words, sequential logic has memory while combinational logic does not. Combinational logic is used in computer circuits to perform Boolean algebra on input signals and on stored data. *Practical computer circuits normally contain a mixture of combinational and sequential logic. For example, the part of an arithmetic logic unit, or ALU, that does mathematical calculations is constructed using combinational logic. Other circuits used in computers, such as half adders, full adders, half subtractors, full subtractors, multiplexers, demultiplexers, encoders and decoders are also made by using combinational logic. But the circuits such as Flip flops, counters and shift registers which have memory are made using sequential logic circuits. In this chapter let us discuss all these combinational and sequential logic circuits in detail.*

## 3.1 Combinational Circuits

The circuits, which are made up from basic gates (AND, OR, NOT) or universal gates (NAND, NOR) that are 'combined' or connected together to produce more complicated switching circuits are called combinational logic circuits. An example of a combinational circuit is a decoder, which converts the binary code data present at its input into a number of different output lines, one at a time producing an equivalent decimal code at its output.

- In these circuits 'the outputs at any instant of time depends on the inputs present at that instant only.'

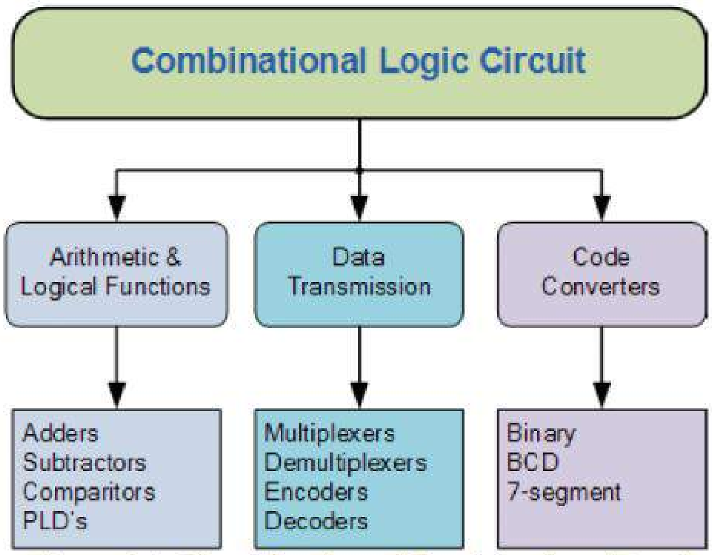



Figure 3.1 Classification of Combinational Logic

- For the design of combinational digital circuits, basic gates (AND, OR, NOT) or universal gates (NAND, NOR) are used. Examples for combinational digital circuits are Half adder, Full adder, Half subtractor, Full subtractor, Code converter, Decoder, Multiplexer, Demultiplexer, Encoder, ROM etc.

In a combinational circuit, as long as input values are maintained, the output values are also maintained. No feedback path exists between the input and the output.

## 3.2 Multiplexer

A multiplexer or mux is a device that allows digital information from several sources to be routed to a single line for transmission over that line to a common destination. It selects one of several analog or digital input signals and forwards the selected input into a single line. A multiplexer of $2^n$ inputs has n select lines, which are used to select the input line which has to be sent to the output.

- Multiplexers can be used for the implementation of Boolean functions and combinational circuits. They can also used for parallel to serial conversion.
- Multiplexer is also called data selector or universal circuit.
- Boolean functions can be implemented by using multiplexer without using any additional gates.

Figure 3.2 shows the logic symbol for 4:1 multiplexer. It consists of four input terminals and two select variables A and B.

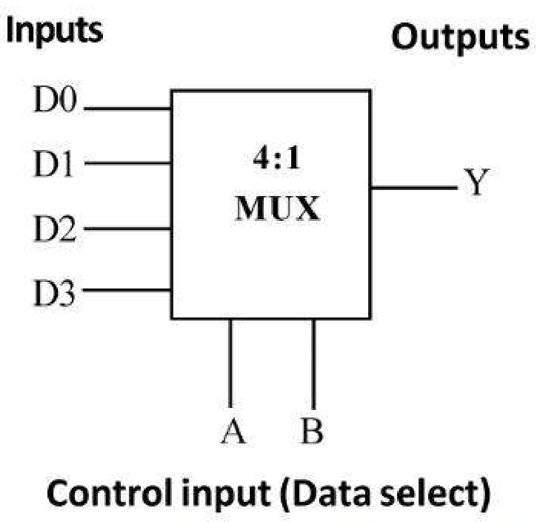



Figure 3.2 Logic Symbol for 4 input multiplexer

The select lines A and B determine which input line is to be selected from D0 to D3 to the output Y at a time. The values of A and B and the correspondingly selected input line as shown in the following table 3.1.

| INPUTS | | OUTPUT |
|---|---|---|
| A | B | Y |
| 0 | 0 | D0 |
| 0 | 1 | D1 |
| 1 | 0 | D2 |
| 1 | 1 | D3 |

Table 3.1 Truth table of a multiplexer

The table indicates that when A=0, B =0, the input D0 is connected to the output Y and when A=0, B=1, the input D1 is connected to the output and so on.

Now we will see the internal structure of a 4:1 multiplexer. It is shown in figure 3.3. Here the select lines A and B are made available in both direct and compliment forms, by using inverter gates. Each data source input is connected to AND gate. An AND gate will pass data to its output when its two other

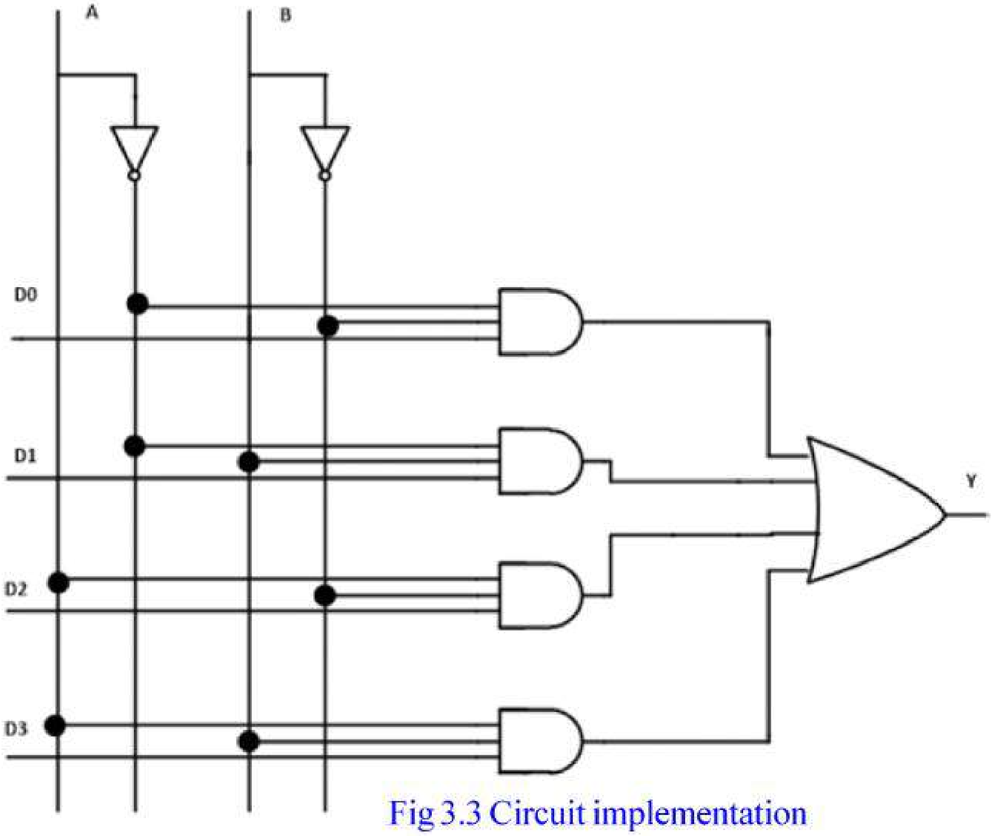



Fig 3.3 Circuit implementation

inputs are '1'. The output of the AND gates reach the final output of the multiplexer Y through the OR gate.

When AB = 00, the upper AND gate is enabled while all other gates are disabled. Therefore data bit D0 is transmitted to the output, giving Y=D0.If the control input is changed to AB =11, all gates are disabled except the bottom AND gate. In this case, D3 is transmitted to the output and Y=D3.

**Applications of multiplexers**:

1. Communication system - communication system is a set of systems like transmission system, relay and tributary station and communication network that enables communication. The efficiency of communication system can be increased considerably using multiplexers. It allows transmitting different types of data such as audio and video at the same time using a single transmission line.
2. Telephone network - In telephone network, multiple audio signals are integrated on a single line for transmission with the help of mux.
3. Computer memory - Mux is used to implement the huge amount of memory into the computer. At the same time it reduces the number of copper lines required to connect the memory to the other parts of the computer circuit.
4. Multiplexers help to transmit signals from the computer system to a satellite.

**Know your progress**

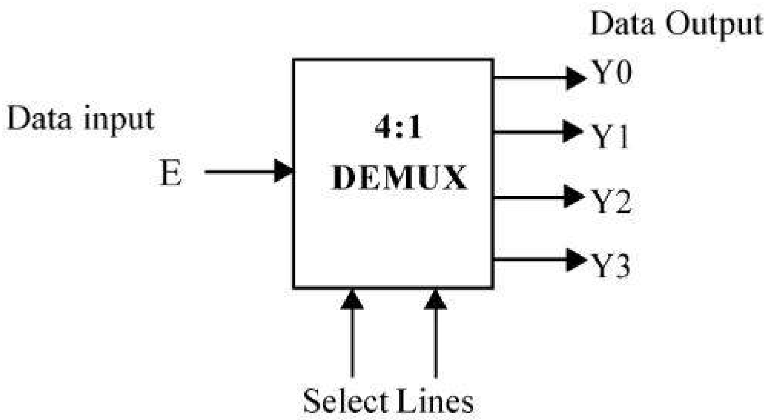
Draw the circuit of a 8:1 multiplexer. How many select lines are required in this case?

## 3.3 Demultiplexer

'Demultiplexer' is a logical circuit that takes the input from a single source and sends it to one of the several $2^n$ possible output lines. The selection of specific output line is controlled by the bit values of 'n' selection lines. The function of a demultiplexer is reverse of that of a multiplexer. We have to note the following facts about a demultiplexer.

- It performs the inverse operation of a multiplexer.

- It is a combinational circuit that receives input from a single line and transmits it to one of the $2^n$ possible output lines.
- The selection of the specific output is controlled by the bit combination of n selection lines.




**Control input (Data select)**

Fig. 3.4 Logic Symbol for 4 input multiplexer

| A | B | Y0 | Y1 | Y2 | Y3 |
|---|---|---|---|---|---|
| 0 | 0 | E | 0 | 0 | 0 |
| 0 | 1 | 0 | E | 0 | 0 |
| 1 | 0 | 0 | 0 | E | 0 |
| 1 | 1 | 0 | 0 | 0 | E |

Table 3.2 *Truth table for 1:4 demultiplexer*

The table shows that when the select lines A= 0, B= 0, the input goes to the output Y0 and when A = 0, B=1, the input goes to the output Y1 and so on.

In the circuit implementation of the 1:4 demultiplexer, we have the select lines A and B and their complements produced with the help of NOT gates. It is shown in figure 3.5. The input data is made available to the input of all the four AND gates. The AND gate passes the input data to the output only if its other two inputs are 1's. Thus the data reaches the output Y0 if A=0 and B=0 and data reaches the output Y1 if A=0 and B=1 and so on.

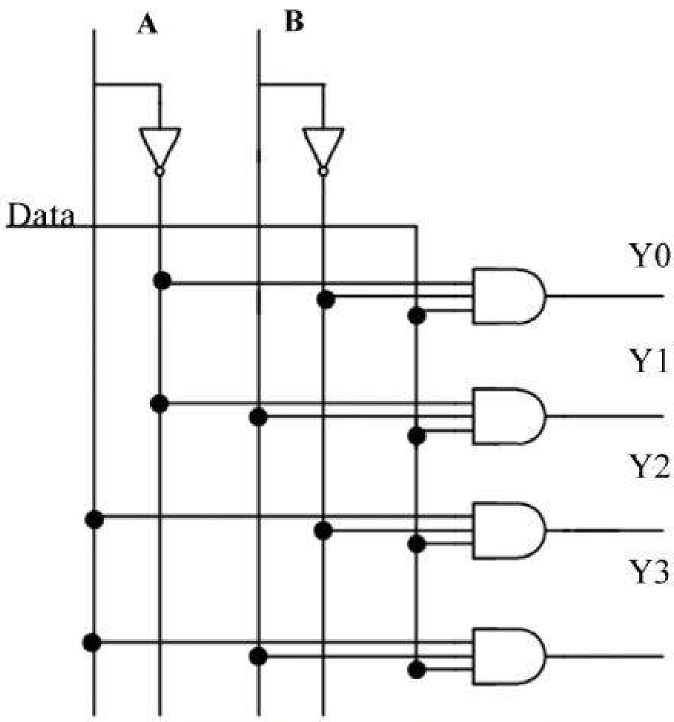



Fig 3.5 Circuit implementation

## 3.4 Encoder

An Encoder has a number of inputs ($2^n$) of which only one is active or in a high state, and an n-bit code is generated depending on the input which is excited. From the code generated at the output, we can identify the input which is presently active. Otherwise, we can say that an encoder assigns a code for each input. An encoder has $2^n$ input lines and n output lines. The output lines generate the binary code for the input variables. Let us consider an example in which the encoder has16 inputs $I_0 - I_{15}$ and the position of each of these input is encoded into binary by the encoder. If the input line $I_0$ is active then the code 0000 is generated on the four output lines. If the input line $I_5$ is active then the code 0101 is generated in the output and so on.

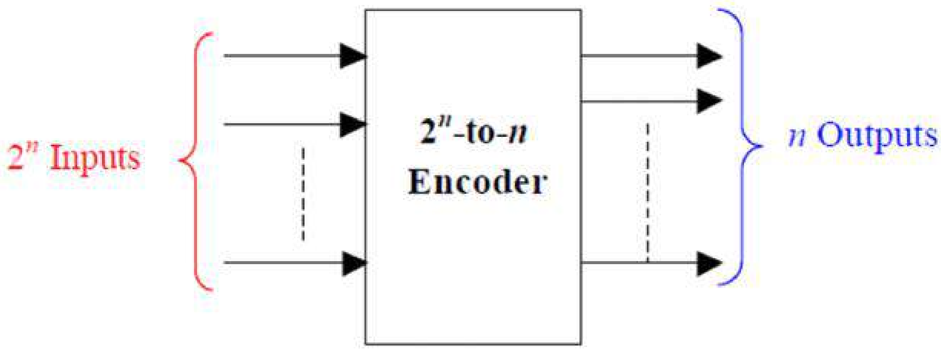



Fig 3.6 Typical encoder



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- An encoder has
  - $2^n$ inputs
  - $n$ outputs
- Outputs the binary value of the selected or active input.
- Only one input can be logic 1 at any given time (active input). All other inputs must be 0's.

The circuit diagram of a 4X2 encoder is shown below:

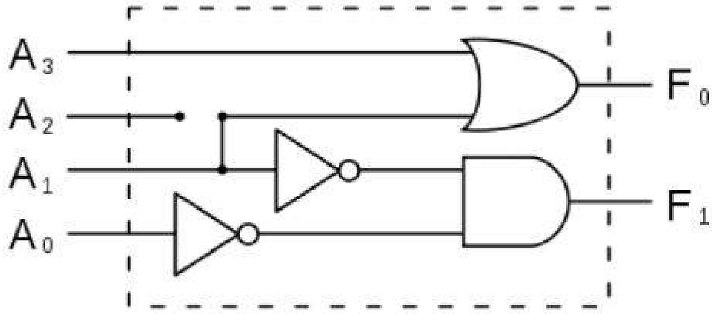



Fig. 3.7 Circuit of an encoder

The operation of this encoder is described in the following table. The reader can verify the following table by giving input to the above circuit.

| INPUT | | | | OUTPUT | |
|---|---|---|---|---|---|
| A3 | A2 | A1 | A0 | F1 | F0 |
| 0 | 0 | 0 | 1 | 0 | 0 |
| 0 | 0 | 1 | 0 | 0 | 1 |
| 0 | 1 | 0 | 0 | 1 | 0 |
| 1 | 0 | 0 | 0 | 1 | 1 |

Table 3.3 Truth table of 4 x 2 encoder

The example of an octal to binary encoder is discussed below.

### Octal-to-binary encoder

The octal to binary encoder consists of eight inputs, one for each of the digits (0-7), and three outputs that generate the corresponding binary number. Any octal number has digits from 0 to 7 and these digits can be converted into binary with the help of this encoder.

We have eight inputs, one for each of the octal digits, and three outputs that generate the corresponding binary number. Thus, in the truth table, we can see eight input variables on the left side and three variables on the right side.

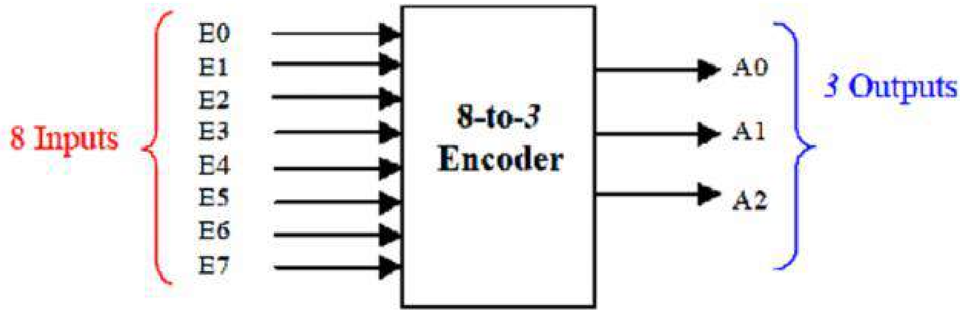



Fig 3.8 Octal to binary encoder

If the octal digit is 0, then the input line E0 is active or HIGH and all other input lines are LOW. Now the code 000 is generated in the output which is the binary equivalent of the octal number 0. If the octal digit is 6, then only E6 is HIGH and output 110 is obtained at the output. If an octal number 53 is to be converted into binary, then the first digit 3 is given to the encoder first and its binary is obtained. Then the next digit 5 is input and its binary is obtained. This is shown in the table 3.3.

| Inputs | | | | | | | | Outputs | | | Decimal Code |
|---|---|---|---|---|---|---|---|---|---|---|---|
| E7 | E6 | E5 | E4 | E3 | E2 | E1 | E0 | A2 | A1 | A0 | |
| 0 | 0 | 0 | 0 | 0 | 0 | 0 | 0 | 0 | 0 | 0 | |
| 0 | 0 | 0 | 0 | 0 | 0 | 0 | 1 | 0 | 0 | 0 | 0 |
| 0 | 0 | 0 | 0 | 0 | 0 | 1 | 0 | 0 | 0 | 1 | 1 |
| 0 | 0 | 0 | 0 | 0 | 1 | 0 | 0 | 0 | 1 | 0 | 2 |
| 0 | 0 | 0 | 0 | 1 | 0 | 0 | 0 | 0 | 1 | 1 | 3 |
| 0 | 0 | 0 | 1 | 0 | 0 | 0 | 0 | 1 | 0 | 0 | 4 |
| 0 | 0 | 1 | 0 | 0 | 0 | 0 | 0 | 1 | 0 | 1 | 5 |
| 0 | 1 | 0 | 0 | 0 | 0 | 0 | 0 | 1 | 1 | 0 | 6 |
| 1 | 0 | 0 | 0 | 0 | 0 | 0 | 0 | 1 | 1 | 1 | 7 |

Table 3.4 Truth table for Octal to Binary encoder

## 3.5 Decoder

A decoder is a logic circuit that converts an n bit binary input code into $2^n$ output lines such that each output line will be activated for only one of the possible combinations of inputs.

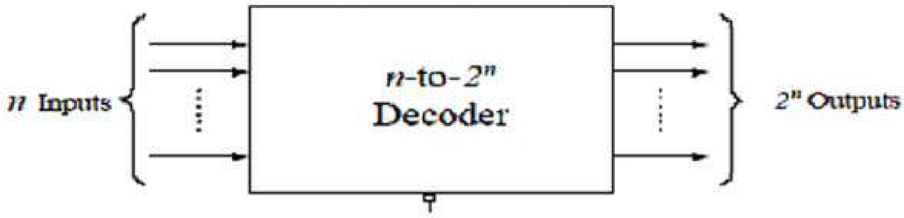



*Fig 3.9* Typical decoder

For example, if there are 4 input lines, then there will be 16 output lines. For any input combination from 0000 to 1111, the output lines $I_0$ to $I_{15}$ are selected; one output line for each input combination. If the input lines carry the number 0000, then the output line $I_0$ is selected at the output and if the input is 1010, then the output line $I_{10}$ is selected. As an example, a 3 to 8 decoder is explained below.

### 3-to-8 Decoder

In a 3 to 8 decoder, there are three inputs and eight outputs as shown in the figure 3.10. Here A0 is the least significant variable, while A2 is the most significant variable. The three inputs are decoded into eight outputs. That is, binary values at the input form a combination, and based on this combination, the corresponding output line is activated.

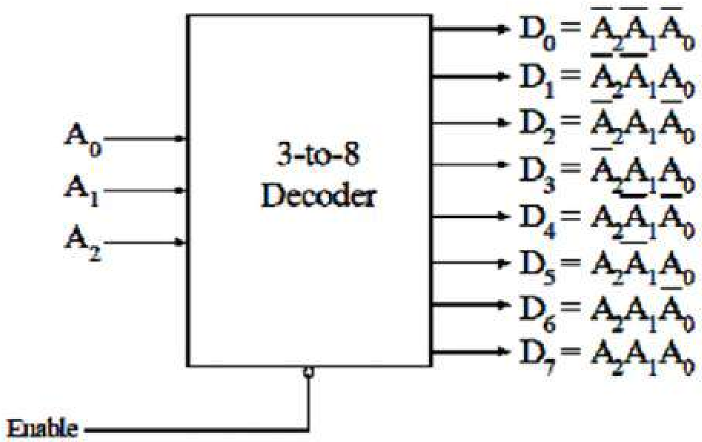



Fig 3.10 3 to 8 decoder

If the data in the input line is 000, then the output line $D_0$ will only be activated and if the code in the input line is 011, then the output line $D_3$ will be HIGH. This is shown in the table below.

| Dec. Code | Inputs | | | Outputs | | | | | | | |
|---|---|---|---|---|---|---|---|---|---|---|---|
| | $A_2$ | $A_1$ | $A_0$ | $D_0$ | $D_1$ | $D_2$ | $D_3$ | $D_4$ | $D_5$ | $D_6$ | $D_7$ |
| 0 | 0 | 0 | 0 | 1 | 0 | 0 | 0 | 0 | 0 | 0 | 0 |
| 1 | 0 | 0 | 1 | 0 | 1 | 0 | 0 | 0 | 0 | 0 | 0 |
| 2 | 0 | 1 | 0 | 0 | 0 | 1 | 0 | 0 | 0 | 0 | 0 |
| 3 | 0 | 1 | 1 | 0 | 0 | 0 | 1 | 0 | 0 | 0 | 0 |
| 4 | 1 | 0 | 0 | 0 | 0 | 0 | 0 | 1 | 0 | 0 | 0 |
| 5 | 1 | 0 | 1 | 0 | 0 | 0 | 0 | 0 | 1 | 0 | 0 |
| 6 | 1 | 1 | 0 | 0 | 0 | 0 | 0 | 0 | 0 | 1 | 0 |
| 7 | 1 | 1 | 1 | 0 | 0 | 0 | 0 | 0 | 0 | 0 | 1 |

Table 3.5 Truth table of 3 to 8 decoder

The above decoder circuit using gates is shown below.

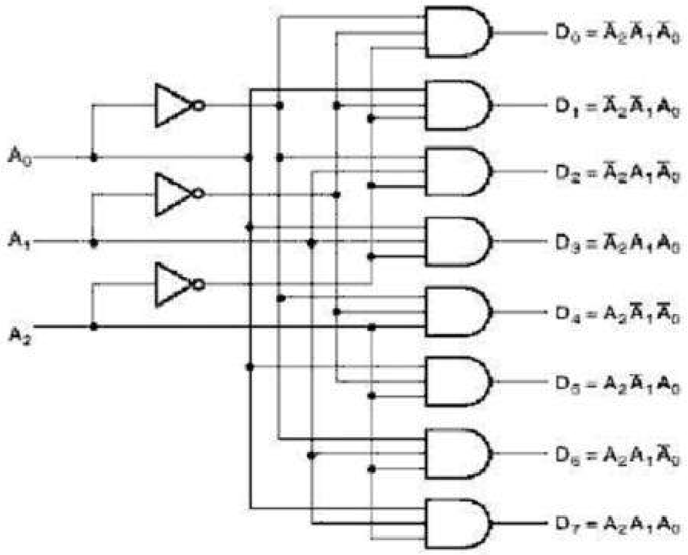



Fig 3.11 Circuit implementation of 3 to 8 Decoder

## 3.6 Comparator

Digital comparators are used to compare two digital numbers. An AND gate can be used to detect whether a number (bit) is greater than another number (bit) or not. An EX-NOR gate is used to compare the equivalence of two bits. Thus combining EX-NOR gates with AND gates, we can construct a digital comparator.

### One bit comparator

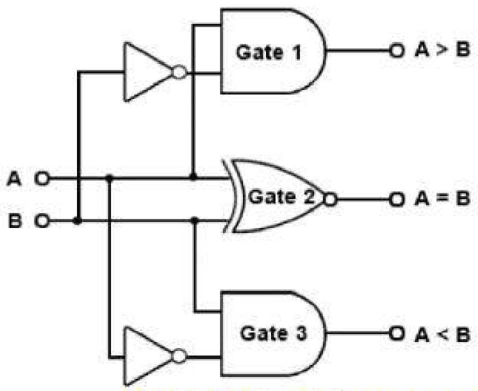



Fig 3.12 One bit comparator

Here, we make use of two AND gates and one EX-NOR gate. The top AND gate has output 1, when A>B. The output of this AND gate becomes 1 if only A = 1 and B = 0. Similarly the bottom AND produces 1when A<B (A=0 and B=1).The EX-NOR will produce an output of '1' if both A and B are equal (i.e. either if A=B=0 or if A=B=1)

**Know your progress**

Design a 2- bit comparator by upgrading the concept of the above 1-bit comparator.

## 3.7 Sequential Circuits

Sequential circuits have memory to remember the effects of previous inputs. These circuits use previous output as well as the present input to form the present output. Usually there will be a feedback from output to input in such circuits. We have a basic unit of memory called flip flop and the sequential circuits are mainly made up of flip flops. Some of the sequential circuits we study here are counters and shift registers. Before going to the details of these circuits we will familiarise with the operation and properties of basic flip flops.

## 3.8 Flip Flops

Flip flops are actually derived from logic gates. With the help of Boolean logic flip flops can be used to memorize data. Flip flops can also be considered as the most basic unit of Random Access Memory [RAM]. If certain input values are given to a flip flop, it will be stored accurately. Flip flops have two stable states; HIGH and LOW.

There are mainly four types of flip flops that are used in electronic circuits. They are

1. S-R Flip Flop
2. J-K Flip Flop
3. Delay Flip Flop [D Flip Flop]
4. Toggle Flip flop [T Flip Flop]

### 1. S-R Flip Flop

The SET-RESET flip flop is designed with the help of two NOR gates or with two NAND gates. These flip flops are also called S-R Latch.

#### S-R Flip Flop using NOR Gate

This flip flop has two inputs, called the SET [S] and RESET [R]. There are also two outputs, Q and Q' which are complementary to each other. The diagram and the truth table are shown below.

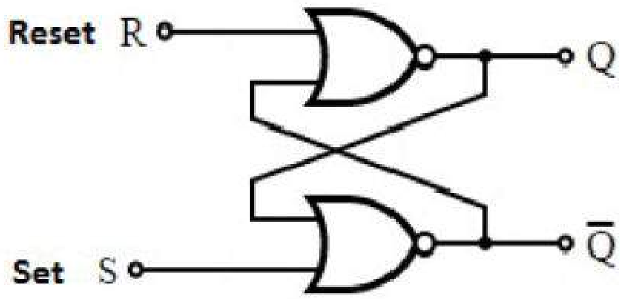



Fig. 3.13 S-R Flip flop using NOR Gate

| INPUTS | | OUTPUT |
|---|---|---|
| R | S | Qn+1 |
| 0 | 0 | Qn |
| 0 | 1 | 1 |
| 1 | 0 | 0 |
| 1 | 1 | Invalid |

Table 3.6 Truth table of S-R Flip flop

$Q_{n+1}$ indicates the output at the n+1$^{th}$ instant and $Q_n$ is the output of the previous state. From the truth table, it is evident that the flip flop has four states. They are

**S=1, R=0 — Q=1, Q'=0**

This state is called the SET state.

**S=0, R=1 — Q=0, Q'=1**

This state is known as the RESET state.

In both the states you can see that the outputs are just the compliments of each other and that the value of Q follows the value of S.

**S=0, R=0 — Q & Q' = previous state.**

If both the values of S and R are switched to 0, then the output remains in the previous state or we can say that the circuit remembers the previous state.

**S=1, R=1 — Q=0, Q'=0 [Invalid]**

This is an invalid state because the values of both Q and Q' are 0. They are supposed to be compliment of each other. Normally, this state must be avoided without giving S=1, R=1.

## S-R Flip Flop using NAND Gate

The circuit of the S-R flip flop using NAND Gate and its truth table are shown below.

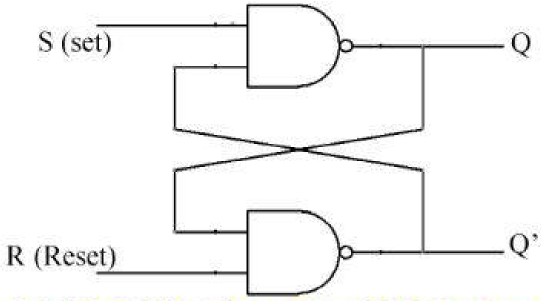



Fig. 3.14 S-R Flip Flop using NAND Gate

| INPUTS | | OUTPUT |
|---|---|---|
| **R** | **S** | **Qn+1** |
| 0 | 0 | Invalid |
| 0 | 1 | 0 |
| 1 | 0 | 1 |
| 1 | 1 | Qn |

Table 3.7 Truth table of S-R flip flop

Like the NOR gate S-R flip flop, this one also has four states. They are

**S=1, R=0 — Q=0, Q'=1**

This state is also called the SET state.

**S=0, R=1 — Q=1, Q'=0**

This state is known as the RESET state.

In both the states you can see that the outputs are just the compliments of each other and that the value of Q follows the compliment value of S.

**S=0, R=0 — Q=1, & Q' =1 Invalid**

If both the values of S and R are switched to 0 it is an invalid state because the values of both Q and Q' are 1. They are supposed to be the compliment of each other. Normally, this state must be avoided.

**S=1, R=1 — Q & Q'= previous state**

If both the values of S and R are switched to 1, then the circuit remains in its previous state or it remembers the previous state.

## Clocked S-R Flip Flop

It is also called a gated S-R flip flop.

The problems with S-R flip flops using NOR and NAND gate is related to its state. This problem can be solved by using a bistable SR flip-flop that can change outputs when certain invalid states are met, regardless of the condition of either the Set or the Reset inputs. For this, a clocked S-R flip flop is designed by adding two AND gates to the basic NOR Gate flip flop. The circuit diagram and the truth table are shown below.

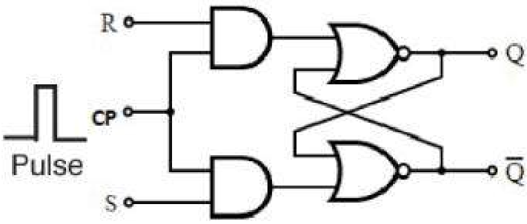



Fig. 3.15 Clocked S-R Flip Flop

| CP | S | R | Qn+1 | STATE |
|---|---|---|---|---|
| 1 | 0 | 1 | 0 | RESET |
| 1 | 1 | 1 | 1 | SET |
| 1 | 1 | 1 | X | INVALID |
| 1 | 0 | 0 | Qn | NO CHANGE |

Table 3.8 Truth table of clocked S-R flip flop

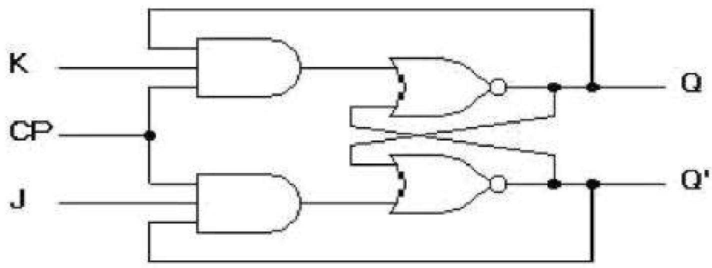
A clock pulse CP is given to the inputs of the AND Gate. When the value of the clock pulse is '0' , the outputs of both the AND gates remain '0' . As soon as a pulse is given, the value of CP turns '1' . This makes the values at S and R to pass through the NOR gate flip flop.

**Know your progress**

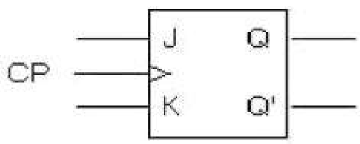
Clock pulses control the operation of S-R flip flop. Explain.

## 2. J-K Flip Flop

The circuit diagram and symbol of a J-K flip flop is shown below.




(a) Logic diagram




(b) Graphical symbol

*Fig. 3.16 J-K Flip Flop- circuit and symbol*



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A J-K flip flop can also be considered as a modified S-R flip flop. The only difference is that the invalid state in S-R Flip flop can be avoided in J-K Flip flop.

The behavior of the inputs J and K is same as the S and R inputs of the S-R flip flop. The letter J stands for SET and the letter K stands for CLEAR. The truth table of a JK flip flop is shown below.

| J | K | Qn+1 |
|---|---|---|
| 0 | 0 | Qn |
| 0 | 1 | 1 |
| 1 | 0 | 0 |
| 1 | 1 | $\overline{Qn}$ |

Table 3.9 Truth table of J-K flip flop

The reader can verify the truth table by giving values to J and K in the above circuit. When both the inputs J and K have a HIGH state, the flip-flop switches to the complement state. So, for a value of $Q_n = 1$, it switches to $Q_{n+1}=0$ and for a value of $Q_n = 0$, it switches to $Q_{n+1}=1$.

The output may be repeated in transitions once they have been complemented for J=K=1 because of the feedback connection in the JK flip-flop. So the output at a given time cannot be determined whether it is 0 or 1. We can overcome this problem by setting the pulse duration lesser than the propagation delay through the flip-flop. The restriction on the pulse width can be eliminated with a master-slave or edge-triggered construction. These techniques are not discussed here.

## 3. D Flip Flop

The circuit diagram and the symbol are given below.

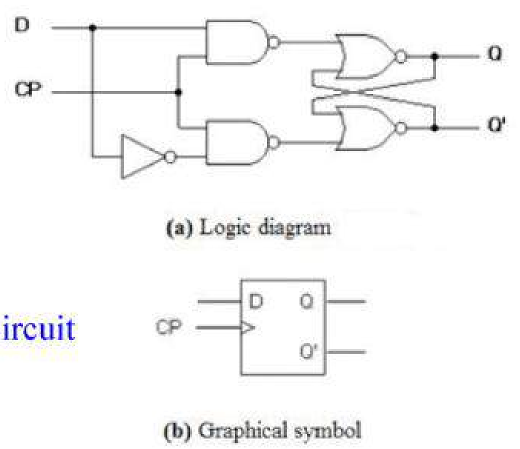



Fig. 3.17 D Flip Flop- circuit and symbol

The truth table of the D Flip flop is given below.

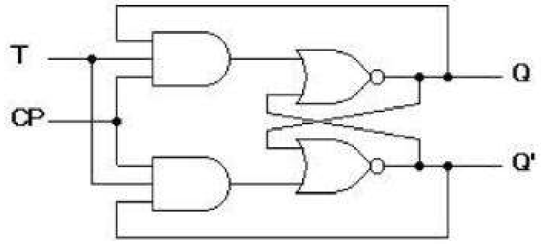
| D (input) | Q (output) |
|---|---|
| 0 | 0 |
| 1 | 1 |

Table 3.10 Truth table of D flip flop

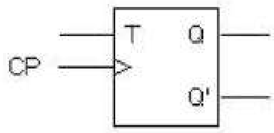
D flip flop is actually a slight modification of the above explained J-K flip-flop. From the figure you can see that the D input is connected to the J input and the complement of the D input is connected to the K input. The D input is passed on to the flip flop when the value of CP is '1' . When CP is HIGH, the flip flop moves to the SET state. If it is '0' , the flip flop switches to the CLEAR state. From the above table, it is seen that the output Q follows the input D after a delay of one clock pulse. Hence the D Flip flop is known as delay Flip flop. The D Flip flops are used as the basic building block of shift registers or memories. Shift registers are used in microprocessors.

## 4. T Flip Flop

A T Flip flop is made by modifying a JK Flip flop in which both the J and K inputs are connected together and thus it is also called a single input J-K flip flop. When clock pulse is given to the flip flop and T = HIGH, the output begins to toggle or the output continuously changes from HIGH to LOW and back. This is same as the case with JK Flip flop when J=K=1. The circuit diagram and the symbol are shown below.




(a) Logic diagram




(b) Graphical symbol

Fig. 3.18 T Flip Flop- circuit and symbol

The truth table of the T Flip flop is shown below.

| T (input) | Qn+1 (output) |
|---|---|
| 0 | Qn |
| 1 | $\overline{Qn}$ |

Table 3.11 Truth table of T flip flop

From the table it is seen that when input T=0, the output continues in the previous state. If T is made HIGH and a clock pulse is given, then output complements or changes its state. Thus in a T Flip flop, the output changes its state for each clock pulse or it toggles if T=1. The T Flip flops are the basic unit of a digital counter circuit.

## 3.9 Binary counters

**Know your progress**

Verify the truth table of above four flip flops by giving inputs and find the output of each gate in their circuit.

Counters are mainly used in counting applications, where they either measure the time interval between two unknown time instants or measure the frequency of a given signal. Basically the digital counter can count the number of pulses given as its clock. So the counter can count any event if it is converted into electric pulses. For example, we can count the number of persons entering an auditorium or the number of items to be put into a packet if electric pulses are produced with the help of necessary electronic circuits during such events. Based on the direction of counting, counters are classified as up counter, down counter or up-down counter. According to the number of states the counter counts, they can be classified into binary counter, decade counter and so on.

- The counter is a special case of sequential circuits.
- Its function is to count the input pulses (clock signal) and store the result till the arrival of the next signal.
- The counting process consists of a series of storage and addition operations.
- The counters are built from T flip-flops and gate circuits.

## Asynchronous or Ripple counter

A ripple counter is a cascaded arrangement of flip-flops where the output of one flip-flop drives the clock input of the following flip-flop. In a ripple counter, which is also called an asynchronous counter or a serial counter, the clock input is applied only to the first flip-flop, or the input flip-flop, in the cascaded arrangement. The clock input to any subsequent flip-flop comes from the output of its immediately preceding flip-flop. For instance, the output of the first flip-flop acts as the clock input to the second flip-flop and the output of the second flip-flop feeds the clock input of the third flip-flop and so on. The three stage ripple counter using JK flip-flops uses the maximum counting capability of the eight stages and can thus be classified as mod 8. The flip flops are connected in toggle mode (J=K=1) and can change the state at the positive going transitions of the pulse waveform applied to the clock input. (it is indicated by the arrow symbol at the clock input of each flip flop) If N flip flop stages are used in a counter, then it is termed as N- bit counter and it has $2^N$ counting states. Then it is a mod $2^N$ counter. The figure of a 3- bit counter is given below. The clock input and output of each flip flop are also shown.

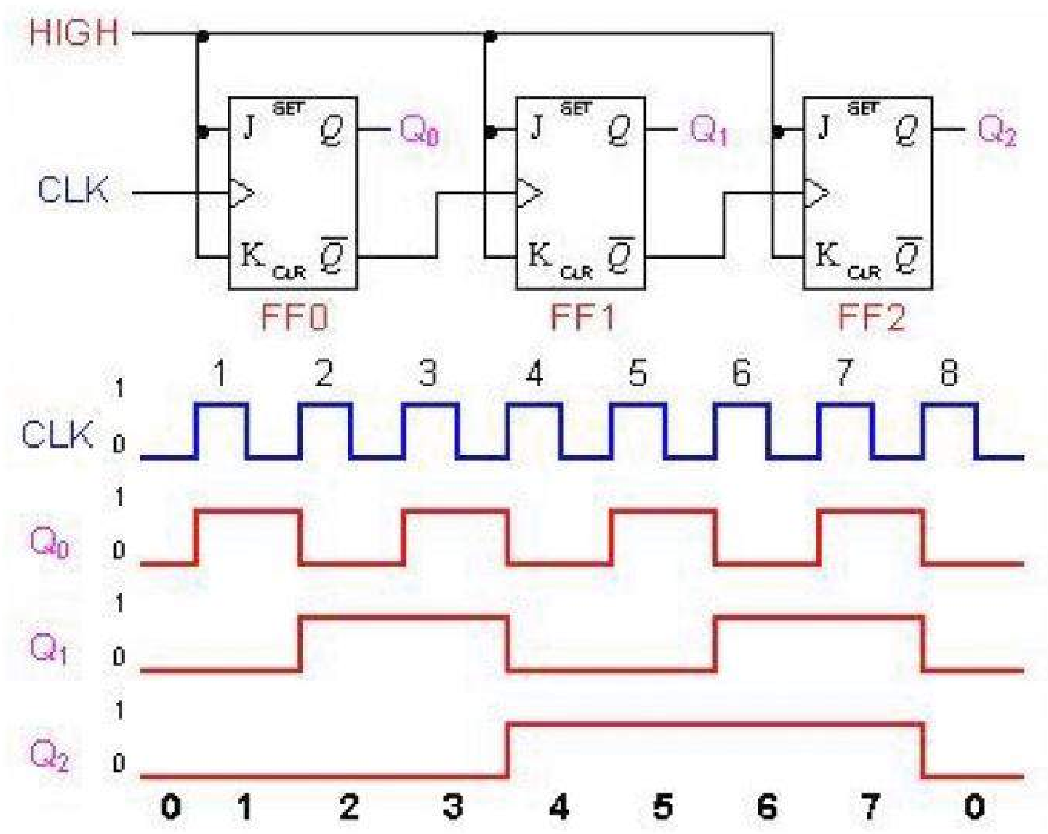



*Fig. 3.19 3- bit counter and output waveforms*

The count is obtained by observing the sequence $Q_2\ Q_1\ Q_0$ which starts at 000 (=decimal 0). When the first clock pulse is applied, the flip flop FF0 switches from 0 to 1 at the positive edge of the clock or $Q_0$ becomes 1. Now the state is $Q_2\ Q_1\ Q_0 = 001$. When the second clock pulse is given, FF0 switches from 1 to 0. At this time its 'Q' which is given as clock of FF1 goes from 0 to 1 and at this positive transition FF1 switches from 0 to 1. Now the count is 0 1 0. For the third clock pulse, FF0 changes state from 0 to 1. At this time FF1 does not change state and the count is 011. Now the reader can find out next successive states of the counter for each input clock pulse. When seven clock pulses are given, the counter reaches the state 111. What will happen when the eighth clock pulse is given?

When the eighth pulse is given, FF0 switches from 1 to 0. At this time FF1 switches from 1 to 0 and hence FF2 switches from 1 to 0 or the new state is 000. Thus we see that the counter automatically resets to 000 from 111 and it counts again for the succeeding clock pulses.

The number of clock pulses and the counter output are shown in the following table.

| No.of pulses | $Q_2$ | $Q_1$ | $Q_0$ |
|---|---|---|---|
| 0 | 0 | 0 | 0 |
| 1 | 0 | 0 | 1 |
| 2 | 0 | 1 | 0 |
| 3 | 0 | 1 | 1 |
| 4 | 1 | 0 | 0 |
| 5 | 1 | 0 | 1 |
| 6 | 1 | 1 | 0 |
| 7 | 1 | 1 | 1 |
| 8 | 0 | 0 | 0 |

Table 3.12 Table showing counting states

It is to be noted that a 4-bit counter counts from 0000 to 1111 and then automatically resets to 0000. It has a total of 16 counting states. Can we design a decade counter which counts from 0 (0000) to 10 (1010)? Yes, this is possible with a 4- bit counter having a simple additional circuitry with gates to reset all flip flops to 0000 when the count reaches 1010.



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**Know your progress**

Draw the circuit of a 4-bit counter and draw the waveforms of clock, Q0, Q1, Q2 and Q3.

## 3.10 Shift registers

A shift register is a clocked sequential circuit which can store a binary word. The bits stored can be shifted towards left or right. A shift register can cause a delay to the input data. The input data is delayed by 'n' discrete clock times, where 'n' is the number of shift register stages. Thus, a four stage shift register delays 'data in' by four clocks to 'data out'. The stages in a shift register are *delay stages*, typically **D** flip-flops. Basic shift registers can be classified into the following types depending on how data bits enter the register and how they are retrieved from it.

- **Serial in-serial out**
- **Parallel in-serial out**
- **Serial in-parallel out**
- **Parallel in-parallel out**

We will discuss what the serial in- serial out shift register is at present and the other three are beyond the scope of this textbook.

### Serial In – Serial Out (SISO) Shift Register

In this type of register, the binary data to be stored is accepted serially that is, one bit is accepted at a time. The stored information is also produced at the output in a serial form, i.e. bit by bit. A 4- bit shift register which has four D flip flops is shown below. A 4- bit data may be stored in this register. The first bit is given to the D input of FFA and the clock pulse is applied. Then this bit goes to the output $Q_A$ or it is registered. Now the second bit is applied to the D input of the flip flop FFA and the clock pulse is applied. This time the input bit reaches $Q_A$ and the previous bit at $Q_A$ now moves to $Q_B$. When the third bit is given and the clock pulse is applied, this bit goes to $Q_A$ and the bit at $Q_A$ moves to $Q_B$ and the previous bit at $Q_B$ moves to $Q_C$. When the fourth bit is given and the clock pulse is applied, the fourth bit reaches $Q_A$ and the three previous bits will be at $Q_B$, $Q_C$ and $Q_D$ respectively.



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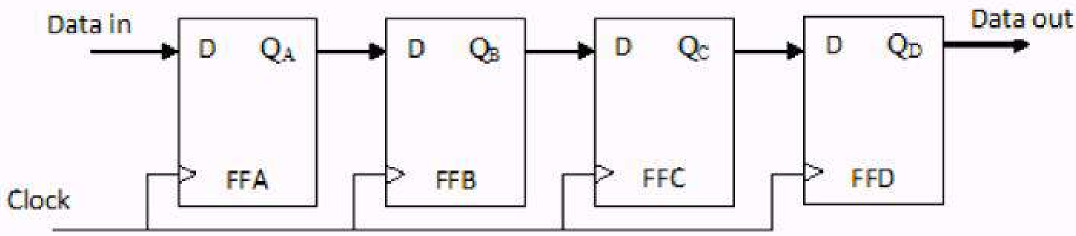


*Fig. 3.20 Serial in- serial out shift register*

When the data is to be read out, the first entered bit should readily available at the output $Q_{D.}$ If we apply one clock pulse, the second bit at $Q_C$ reaches the output. At this time the third bit reaches $Q_C$ and the fourth bit reaches $Q_B$. If we apply the next clock, then the third bit at $Q_C$ reaches the output and the fourth bit reaches at $Q_{C.}$ For the next clock, the fourth bit reaches the output and thus all the four bits can be read out serially.

## Let us consolidate

In a combinational logic circuit the outputs at any instant depends on the inputs present at that instant only. They are made up of basic logic gates or universal gates. A multiplexer allows digital information from several sources to be routed to a single line. Boolean functions can be implemented using multiplexers. They find applications in communication systems, telephone network, computer memory, transmission of signals from computers to satellites etc. Demultiplexer performs inverse operation of a multiplexer. An encoder generates an n-bit code depending on the input which is excited. An encoder has $2^n$ inputs and n outputs. A decoder converts an 'n' bit binary input code into $2^n$ output lines such that each output line will be activated for only one of the possible combinations of the inputs. A digital comparator is used to compare two digital numbers. A digital comparator is used to compare two digital numbers. A digital comparator can be constructed by combining EX-NOR gates with AND gates. In a sequential logic circuit the output depends on the past output as well as the present input. Flip flops, counters, shift resisters etc., are sequential logic circuits. S-R flip flop is designed with two NOR gates or two NAND gates. Clocked S-R flip flop overcome the problem of invalid states in S-R flip flop.

Here AND gates are used to provide clock pulses. A J-K flip flop is a modified form of S-R flip flop and the D flip flop is a slightly modified form of J-K flip flop. A T flip flop is also called a single input clock pulses and stores the result till the arrival of the next signal. It is used for counting applications. The counters are built from T flip flops and gate circuits. A ripple counter or asynchronous counter is a cascaded arrangement of flip flops. An n bit counter used n flip flop states and it has $2^n$ counting stages. A shift resister is a clocked sequential circuit that stores a binary bit which can be shifted towards left or right. In a serial In-serial Out (SISO) shift resister, the binary data to be stored is accepted serially and is produced at the output in serial form.

The contents of this unit were learned through general discussion, group discussion, circuit design, truth table creation and chart preparation.

## Let us asses

1. Digital circuits can be broadly classified into combinational and sequential circuits.
   a) Distinguish between them.
   b) Write examples for combinational and sequential circuits.
2. A multiplexer is used to combine many input lines into a single output line.
   a) Draw the circuit of a 4:1 multiplexer.
   b) How many control lines are required for a 16:1 multiplexer?
   c) Draw an 8:1 multiplexer using two 4:1 multiplexer.
3. In an 8 to 3 encoder, write the code words for each of the eight input lines. Also list out a few applications of an encoder.
4. A digital comparator can be implemented using logic gates.
   a) Why do we use an Ex-NOR gate here?
   b) Discuss the function of the two AND gates in the circuit.
5. S-R flip flop can be designed using NOR gates as well as NAND gates.
   a) What is the difference between NOR gate based S-R flip flop and NAND gate based S-R flip flop?

b) Write the truth table of this flip flop.
c) Why do we call the input S=R=1 as forbidden?
d) Discuss the role of clock in the functioning of S-R flip flop.

6. The main difference between J-K flip flop and R-S flip flop is that J=K=1 input can be given to J-K flip flop.
   a) What modification is done in J-K flip flop to achieve this?
   b) What happens to J-K flip flop when J=K=1 is given?
7. The D flip flop is a one bit memory.
   a) Why is it known as delay flip flop?
   b) If a square wave of frequency 'f' is given to the input terminal D of this flip flop, find the frequency of the square wave obtained at its output.
8. The T flip flop is known as the toggling flip flop.
   a) What is toggling?
   b) Where do we utilise toggling?
9. Shift registers use D flip flop as its basic unit.
   a) How many D flip flops are used in a 8 bit shift register?
   b) Draw the circuit diagram of this register.
10. Counters are designed using T flip flops.
   a) Draw the circuit of a 3 bit counter.
   b) How many counting states are possible with this counter?
   c) Can we modify this counter to count up to 6 only and then reset to zero?



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# 4 RADIO BROADCASTING

## Significant Learning Outcomes

*After completing this chapter the learner:*

- identifies the need for modulation.
- explains the concept of AM.
- explains the concepts of spectrum, bandwidth and power of AM.
- explains the operation of AM generator and AM demodulation.
- explains FM.
- sketches the spectrum of FM
- describes the noise immunity of FM
- compares the performance of AM and FM.
- explains the operation of TRF receivers.
- explains the merit of superheterodyne receiver.

You are familiar with different radio broadcasting stations in your locality. Also you might have heard about AM and FM in connection with radio stations. Have you ever thought about this? The audio signal corresponding to the radio programs are produced in a radio station and it is then transmitted to the radio receivers located all around. It is a very good example for one directional wireless communication. Wireless communication is carried out with the help of antennas. Can we transmit an audio signal directly using an antenna? The answer is no and the reason for it will be discussed in this chapter. We use an indirect method to transmit a low frequency signal using antenna and this technique is called modulation. The AM and FM are actually two different modulation techniques.

## 4.1 Modulation

Modulation is an important step of communication system. Modulation is defined as the process whereby some characteristics (amplitude, frequency and phase) of a high frequency signal wave (carrier wave) is varied in accordance with the instantaneous value of the voltage of low frequency signal wave (modulating wave). So the information which is a small frequency signal is added to a high frequency carrier signal by the process of modulation. Depending on which characteristic of the carrier is to be varied after the addition of information, we have different types of modulation schemes.

We shall see the process of modulation in detail. Two signals are involved in the modulation process. The baseband signal and the carrier signal. The baseband signal is the information signal or message which is to be transmitted to the receiver. The frequency of this signal is generally low. In the modulation process, this baseband signal is called the modulating signal. In radio broadcasting, radio program is the modulating signal and in TV, it is video signal. In telephone, our speech signal is the modulating signal.

The other signal involved in the modulation process is a high frequency sinusoidal wave. This signal is called the carrier signal or carrier. The frequency of the carrier signal is always much higher than that of the baseband signal. After modulation, the baseband signal of low frequency is transferred to the high frequency carrier, which carries the information in the form of some variations. After completing the modulation process, some characteristic of the carrier is changed such that the resultant variations carry the information.

The carrier signal is represented by the equation:

$$e_c = E_c \sin(\omega_c t + \theta) \quad \text{---------------- (4.1)}$$

In Equation (4.1), the subscript 'c' indicates the carrier signal. The components of this equation are as follows:

- $e_c$: Instantaneous amplitude of the carrier
- $E_c$: Maximum amplitude of the carrier
- Angular frequency of the carrier, $\omega_c = 2\pi fc$, where fc is the frequency of the carrier, also called the carrier frequency.
- $\theta$ : Initial phase of the carrier signal

Equation (4.1) has three parameters namely, amplitudes ($E_c$), frequency ($\omega_c$), and phase. In principle, these parameters have constant values for a particular sinusoidal wave. According to the definition of modulation, some characteristics of the carrier signal is varied in accordance with the modulating signal. After modulation any one of the three parameters of the carrier signal, namely, amplitude, frequency, or phase is varied keeping the remaining two constant. The baseband signal is then carried by these variations. The type of the modulation is decided by the parameter chosen to be varied.

For example, if the amplitude of the carrier is chosen to be varied in accordance with the instantaneous amplitude of the baseband signal, keeping frequency and phase constant, the resulting modulation is called amplitude modulation (AM). Frequency modulation (FM) and phase modulation (PM) are also obtained in a similar way.

Low-frequency baseband signal is thus translated to a high frequency carrier such that the information is coded in the variations in one of the parameters of the carrier. At the receiver's side, these variations are detected through the demodulation process to recover the original baseband signal.

The following can be summarized with reference to modulation.

- The baseband signal is known as the modulating signal.
- The baseband signal is a low-frequency signal.
- The carrier signal is always a high frequency sinusoidal wave.
- During the modulation process, the modulating signal changes the frequency, amplitude, or phase of the carrier in accordance with its instantaneous amplitude.
- After modulation, the carrier is said to be modulated by the modulating signal.
- The output of the modulator is called the modulated signal.

The process of modulation in a communication system increases its cost and complexity. However, modulation is extensively used in most communication systems. There is a definite need for using modulation.

## 4.2 Need for modulation: -

(i) **To separate signal from different transmitters**

Audio frequencies are within the range of 20 Hz to 20 kHz. Without modulation all signals of same frequencies from different transmitters would be mixed up. It will create an impossible situation to tune any one of them or separate them. In order to separate the various signals, the radio stations must broadcast at different frequencies. Each radio station must be given its own carrier frequency and frequency band. This can be achieved by a process called frequency translation. This frequency translation is the result of modulation process.

(ii) **Size of the antenna: -**

For efficient transmission, the transmitting antennas should have a length at least equal to a quarter of the wavelength of the signal to be transmitted. For an electromagnetic wave of frequency 15 kHz, the wavelength is 20 km and one-quarter of this will be equal to 5 km. Obviously a vertical antenna of this size is impractical. On the other hand, for a frequency of 1 MHz, the height is reduced to 75m. Thus, instead of transmitting the message signal directly using an antenna, we can give the modulated signal to the antenna which is a high frequency signal containing the message. In this way the size of the antenna can be reduced by modulation.

**(iii) Power of radiation:-**

Also, the power radiated by an antenna of length $l$ is proportional to $(l/\lambda)^2$. This shows that for the same antenna length, power radiated is large for shorter wavelength. Thus, the signal which is of low frequency, must be translated to the high frequency spectrum of the electromagnetic wave so that, more power can be effectively radiated. This is achieved through the process of modulation.

**Know your progress**

1. *Find the length of the antenna required to radiate a signal of frequency 3MHz effectively.*
2. *Two different messages having the same frequency range cannot be transmitted through a channel at the same time. Comment.*

## 4.3 Amplitude Modulation (AM)

As explained above, if the amplitude of the carrier is varied according to the instantaneous voltage of the message signal, the process is called amplitude modulation or AM. An example of an amplitude varying signal is shown at in Fig. 4.1.

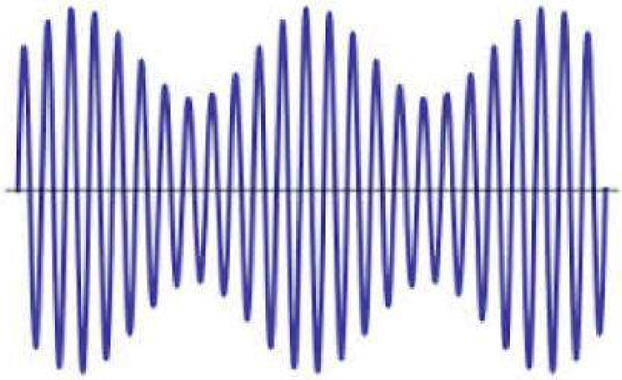

Fig. 4.1 Amplitude modulated signal

The carrier, message signal and the resulting AM signal is shown below.

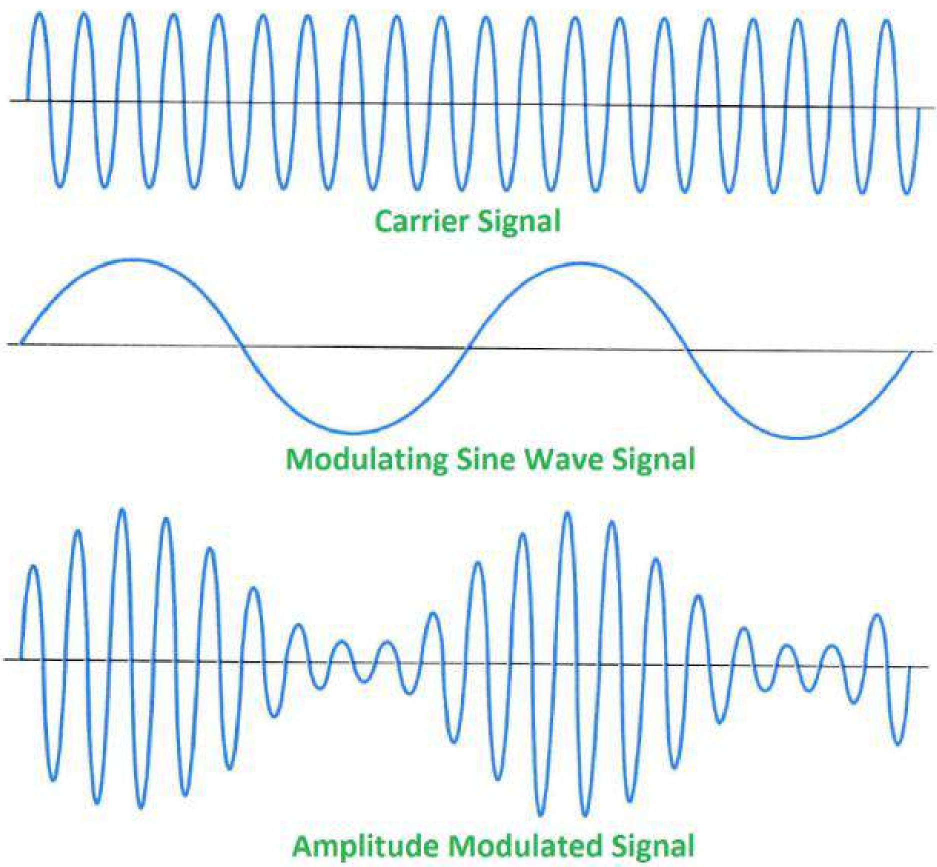


Fig. 4.2 Amplitude modulation

In the AM signal, we can see that its amplitude variation is similar to the voltage variation of the modulating signal. Otherwise, we can say that the envelope of the AM signal is the message. The amplitude of the modulating signal determines the degree of modulation or change in the amplitude variation of the carrier. If the voltage of the message signal is large, it will cause large amplitude variation of the carrier. The degree of modulation or strength of modulation is indicated by the term modulation index.

## 4.4 Modulation index of AM

The strength of modulation is known as modulation index (m) and it is given by the equation, $m = V_m/V_c$. Here $V_m$ is the amplitude of the message signal and $V_c$ is that of the carrier. The modulation index also represents the depth of modulation. Thus if $V_m$ is large, the modulation index is large. The range of value of modulation index is 0-1. The maximum value of this index occurs when $V_m=V_c$. Also a 50% modulation indicates that $V_m = V_c/2$.

The AM signal for m= 0.5, 1 and 1.5 is shown in the following figure.

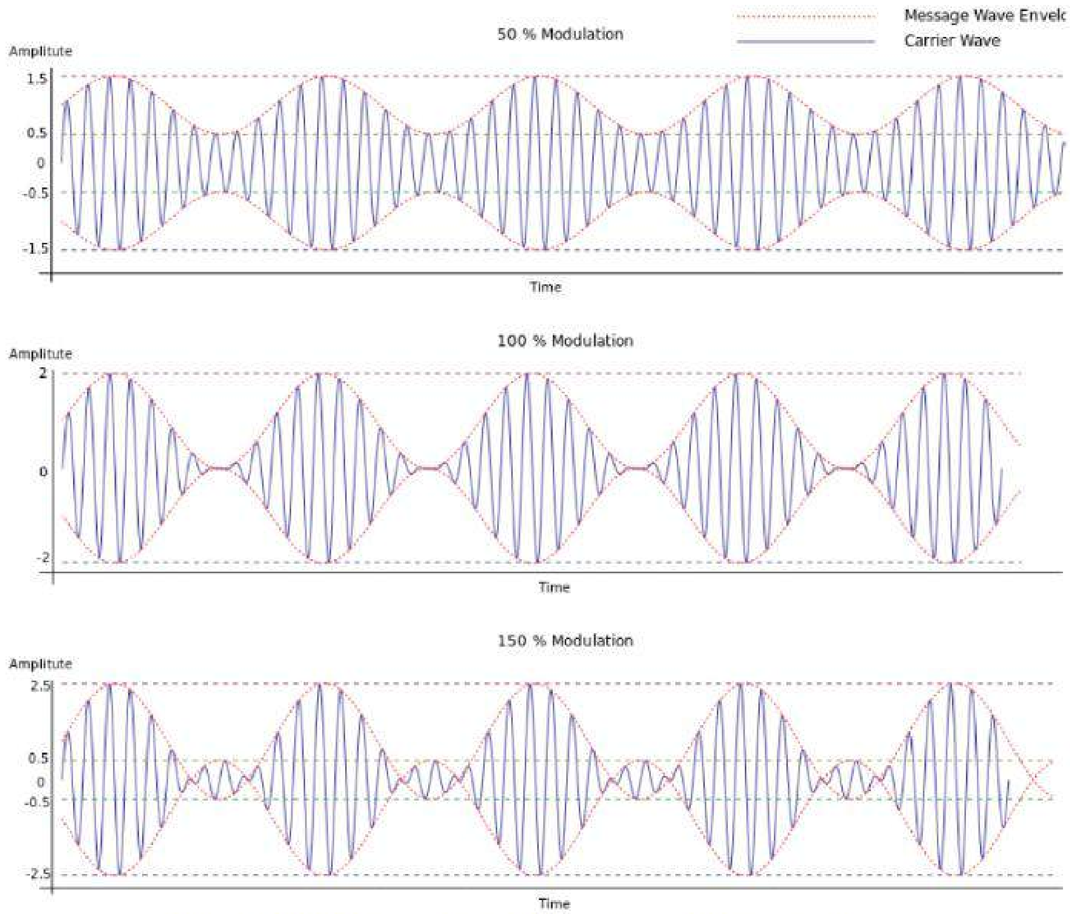


Fig. 4.3 AM signal for different modulation indices

The modulation index of an AM signal can be calculated by displaying the signal on a CRO as follows.

We can observe an AM signal on a CRO as in Fig.4.4.

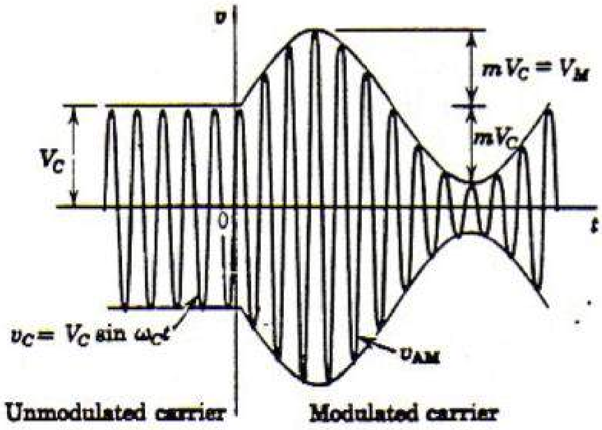


Fig 4.4 AM signal with maximum and minimum voltage levels

Now note down the maximum and minimum voltage values of the AM signal as $V_{max}$ and $V_{min}$. We can see that

$$V_{max} = V_c + V_m \quad \ldots\ldots\ldots\ldots \quad 4.2$$

$$V_{min} = V_c - V_m \ldots\ldots\ldots\ldots \quad 4.3$$

Or we get $\quad V_{max} - V_{min} = 2V_m$

$$V_m = (V_{max} - V_{min})/2$$

Also $\quad V_{max} + V_{min} = 2V_c$

$$V_c = (V_{max} + V_{min})/2$$

Or the modulation index, $m = V_m/V_c = (V_{max} - V_{min}) / (V_{max} + V_{min}) \quad (4.4)$

Thus by measuring the values of Vmax and Vmin from the CRO, we can find the value of the modulation index of an AM signal.

Also it can be noted from the above figure and the equation (4.3) that when $V_m$ becomes equal to $V_c$, the minimum value of the AM signal becomes zero or that point touches the X axis. So if Vm exceeds $V_c$, then distortion will occur in the AM signal or the message signal cannot be retrieved properly at the receiver. In other words, the maximum value of Vm can be equal to $V_c$ or the modulation index, m cannot exceed unity.

## 4.5 Frequency spectrum of AM

The term frequency spectrum refers to a graph representing all frequency components present in a signal with amplitude of each component. Usually frequency is represented along X axis and amplitude along Y axis. Thus the spectrum of AM indicates all frequency components present in the AM signal and the strength or amplitude of each component.

Now, we will form a mathematical expression for the AM signal from which we can get an idea about the frequencies present in the AM signal.

Let the carrier and message signals are respectively represented by

$$v_c = V_c \sin\omega_c t \quad \ldots\ldots\ldots\ldots\ldots\ldots \quad (4.5)$$

$$v_m = V_m \sin\omega_m t \ldots\ldots\ldots\ldots\ldots\ldots \quad (4.6)$$

We know that the amplitude of the carrier is constant and it will be varied when the carrier is modulated by the message. The envelope of the AM signal will look like the message. Thus the amplitude of the unmodulated carrier Vc will get modified to Vc + vm when modulated.(Refer fig 4.3) . The amplitude of the AM signal is no longer a constant quantity but a variable, $A = V_c + v_m$.

$$A = Vc + vm. \; = V_c + Vm \sin \omega_m t.$$

$$= V_c + mVc \sin \omega_m t.$$

$$= V_c( 1+m \sin \omega_m t)$$

Now the instantaneous voltage of the AM signal is given by

$$v = A \sin\omega_c t \quad = V_c( 1+m \sin \omega_m t) \sin \omega_c t$$

$$= V_c \sin\omega_c t + mVc \sin \omega_m t \sin \omega_c t. \; (4.7)$$

This equation can be expanded using trigonometric relation

sinA.sinB = [cos(A- B) - cos(A+B)]/2 to give

$$v = V_c \sin\omega_c t + ½mV_c \cos(\omega_c - \omega_m)t - ½mV_c \cos(\omega_c + \omega_m)t \qquad (4.8)$$

It is thus seen that the AM signal contains three terms having different frequencies. The first term is the unmodulated carrier signal. If we look at the equation 4.8, it becomes clear that the process of AM modulation has the effect of adding something to the unmodulated carrier rather than changing it. The two terms added are of frequencies higher and lower than the original



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carrier frequency. These two terms are known as sidebands. The frequency of the lower sideband (LSB) is $f_c - f_m$ and that of upper sideband (USB) is $f_c + f_m$ and both the terms are having same amplitude $mV_c/2$. The frequency spectrum of an AM signal is shown below.

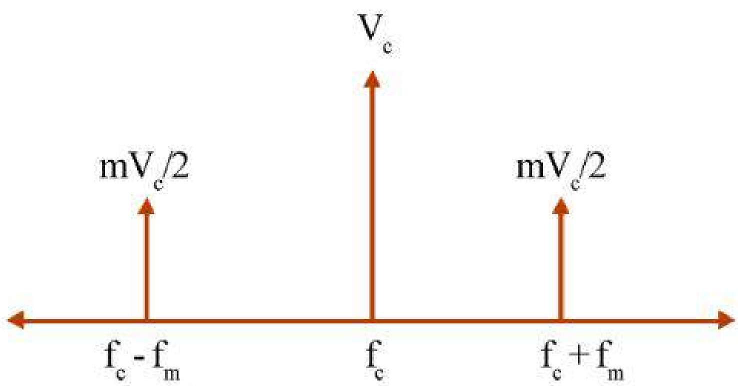


Fig 4.5 Frequency spectrum of AM

## 4.6 Bandwidth

The bandwidth of a signal is defined as the range of frequencies contained in it or it is the difference between the highest frequency and the lowest frequency contained in that signal. In the case of AM, the highest frequency is $f_c + f_m$ and the lowest frequency is $f_c - f_m$.

So the bandwidth, $BW = (f_c + f_m) - (f_c - f_m) = 2\ f_m$ ........ (4.9)

*It is clear that the bandwidth required for an AM signal is twice the modulating frequency.*

So far we have considered that the message contains only a single frequency $f_m$. Now how will the spectrum, be if the message contains two frequencies: $f_{m1}$ and $f_{m2}$? Considering $f_{m2} > f_{m1}$ and if we derive the equation for the AM signal as we did previously, we will get four different frequency terms other than the original carrier frequency term. These four frequencies are $f_c + f_{m1}$, $f_c + f_{m2}$, $f_c - f_{m1}$ and $f_c - f_{m2}$. If the message signals are $v_{m1} = V_{m1} \sin\omega_{m1} t$ and $v_{m2} = V_{m2} \sin\omega_{m2} t$, then the spectrum will be as shown below.



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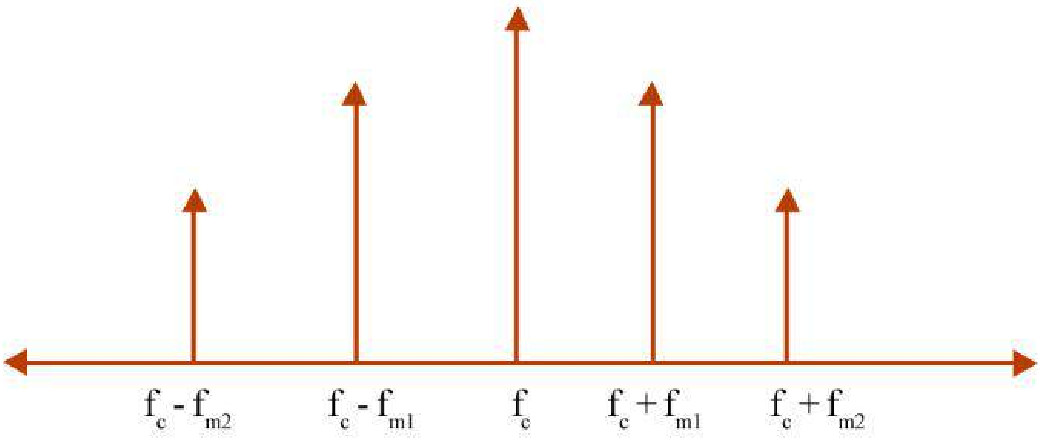


Fig 4.6 spectrum of AM with two modulating signals

Here $m_1 = V_{m1}/V_c$ and $m_2 = V_{m2}/V_c$. From the above concept it can be concluded that if the message contains several frequencies, then the bandwidth of the AM will be twice the highest frequency in the message. If we have a signal with continuous spectrum occupying the range 0- $f_M$, then this frequency spectrum will be shifted to $(f_C - f_M) - f_C$ range (LSB) or $f_C - (f_C + f_M)$ range(USB) when we modulate a carrier with frequency $f_C$ by this signal. If we use a BPF to select USB only, then any spectrum can be shifted to any frequency location by amplitude modulation with a suitable frequency carrier. This is called frequency translation.

## 4.7 Power relations in AM

We know that the AM signal contains one term which is same as the unmodulated carrier and two other terms which represent the sidebands. So the power of AM signal will be greater than the original carrier. The total power of the AM signal is equal to the sum of powers of each term in it. So the total power of AM

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

where $P_C$ is the power of unmodulated carrier and $P_{LSB}$ and $P_{USB}$ are the powers of lower sideband and upper sideband terms respectively.

$$\text{Now } P_T = (V_C/\sqrt{2})^2/R \text{ (rms value)}$$

$$= V_C^2/2R$$

$$P_{LSB} = P_{USB} = [(mV_C)/2\sqrt{2}]^2/R = (m^2 V_C^2)/8R = m^2/4 * V_C^2/2R$$



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$$\text{So } P_T = V_C^2/2R + m^2/4 * V_C^2/2R + m^2/4 * V_C^2/2R$$

$$= PC + m^2/4\,P_C + m^2/4\,P_C = P_C(1+m^2/2)$$

$$P_T = P_C(1+m^2/2) \qquad (4.10)$$

It is interesting to note that the maximum power contained in AM signal is 1.5 $P_C$ when m = 1.

**Example 4.1**

A 400W carrier is modulated to a depth of 60%. Calculate the power of the modulated signal and the power in one sideband.

Solution:

Power of the modulated signal, $P_T = P_C(1+m^2/2)$

Substituting $P_C = 400W$ and $m = 0.6$, we get $P_T = 472W$

Power in one sideband = $m^2/4\,P_C = 36W$

So $P_{LSB} = P_{USB} = 36W$ and $P_T = P_C + P_{LSB} + P_{USB} = 400+36+36 = 472W$.

**Know your progress**

1. *Modulation index of AM signal cannot exceed one. Discuss the reason.*
2. *Draw the spectrum of an AM signal generated when a carrier $5\sin 10^5 t$ is modulated by a message $2\sin 2\times 10^4 t + 3\sin 3\times 10^4 t$.*
3. *Comment on the terms; bandwidth of a signal and bandwidth of a channel.*
4. *Discuss the relationship between the power of message in an AM signal and the modulation index.*

## 4.8 AM generation

In an AM signal, the amplitude of the carrier varies in the same manner as that of the modulating signal. One simple circuit for obtaining such a signal is shown below.



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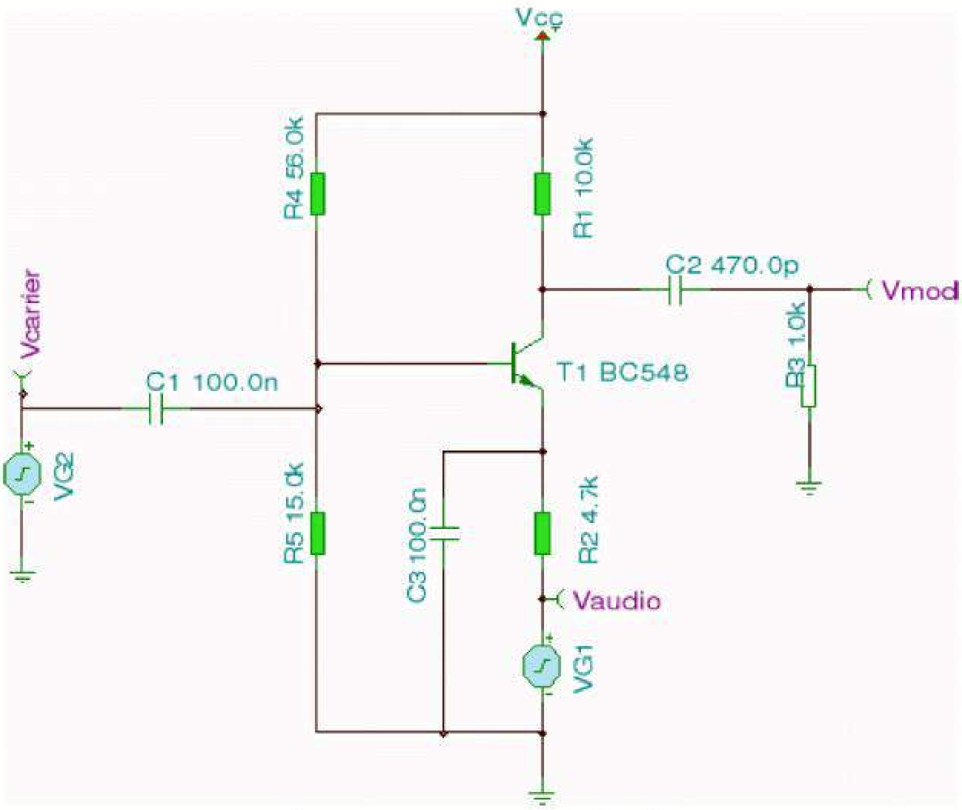


Fig 4.7 AM generator circuit

The above circuit is a simple CE amplifier meant for amplifying the carrier signal. Here the emitter bias voltage is not a constant, but is the sum of the DC bias voltage and the modulating signal. When the emitter bias voltage varies, the gain of the amplifier will vary and hence the amplitude of the output signal also varies. Thus the amplitude of the output carrier signal will change according to the amplitude of the message signal. So the output we get will be an AM signal.

## 4.9 AM Demodulation

The process of separating the message from the AM signal is known as AM demodulation or detection. We know from the frequency spectrum of an AM signal that it contains the carrier and message signals. If we are able to remove the high frequency carrier from the AM signal, then what we are left with is the low frequency message signal. A low pass filter whose cut off frequency is fairly above the highest frequency of the message can effectively filter out the carrier. A block diagram of the circuit of an AM diode detector is shown below.

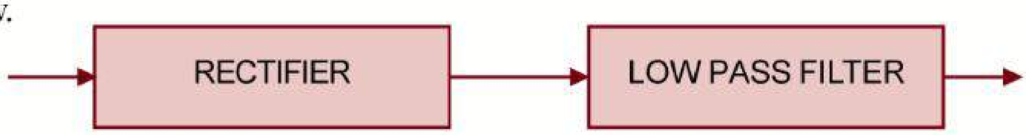


Fig 4.8 AM detector

The diode acts as a half wave rectifier and removes the lower part of the AM signal. Then the signal is passed through a low pass filter to obtain the message signal. If the lower part of the AM signal is not removed before it is passed through the filter, then the message signal on either side of the AM signal will add up to cancel out, each other so that no message will be obtained at the output of the filter.

## 4.10 Types of AM

AM signal transmission is of different types. These types are developed on the basis of the power to be transmitted and the bandwidth availability. If one sideband or a part of the sideband is removed, we can save bandwidth and if the carrier is removed or reduced partly, we can save power. The various types are discussed in brief below.

**DSB-FC (Double Side Band - Full Carrier) :** This is nothing but the AM transmission in which both the sidebands and the full carrier is transmitted.

**SSB (Single Side Band):** The required bandwidth to transmit AM signal is $2f_M$. In order to save the bandwidth, the single sideband (SSB) technique is developed in which the carrier and only one sideband (either LSB or USB) is transmitted. This is possible as we know, both the sidebands in AM carry the same information. By transmitting either of the sidebands, we can transmit the entire message. A band pass filter can be used to remove the unwanted sideband. This will reduce the band width from $2f_M$ to $f_M$ or the band width is reduced to half compared to AM. The disadvantage of SSB is that the power contained in the message will be reduced to half as one sideband is removed. We know that in AM, the message is contained in the sidebands and both sidebands contain the same message and equal power. The spectrum of SSB is shown below.

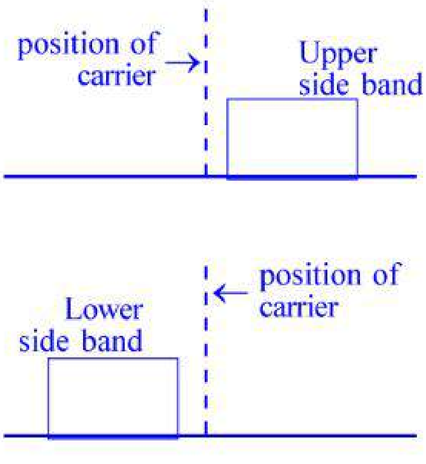


Fig. 4.9 SSB technique

SSB technique is used in frequency division multiplexing (FDM) where different messages are frequency translated so that they are transmitted through a single channel. FDM is discussed in chapter 6. The various types of SSBs are mentioned below.

**Single sideband suppressed carrier, SSBSC:** SSBSC is the form of single sideband modulation that is most widely used for communication applications on the HF portion of the radio spectrum. There is only one sideband - the other sideband and the carrier are removed. But the carrier needs to be re-inserted within the receiver for demodulation purpose. Any slight difference in the re-inserted carrier frequency and the original carrier frequency changes the pitch of the audio. However it gives the most efficient spectrum and power usage.

**Single sideband reduced carrier:** This form of single sideband modulation removes one sideband, but retains a small amount of the carrier. The reduced carrier element can then be used to generate local carrier at the receiver for the demodulation of the sideband with the correct audio pitch. Naturally the power efficiency of this form of single sideband modulation is not as high as the single sideband suppressed carrier.

**Vestigial sideband:** Vestigial sideband is a form of single sideband modulation where one sideband is present, but the other has been partly cut off or suppressed. Vestigial sideband is used for analog AM television transmission and it helps to reduce the overall bandwidth. For TV transmission, $2f_M$ is too large a bandwidth and at the same time the low frequency components of the video signal are very significant and need to be considered. Thus to include the low frequency components of both the sidebands while saving the bandwidth, the vestigial sideband transmission is the only solution.

**Independent sideband, ISB:** This form of single sideband is not strictly "single" because it has two sidebands. However, each sideband carries different modulation, and therefore doubles spectrum efficiency. In other words, within the spectrum of AM, LSB contains one message and USB has a different message. Thus in ISB, two different messages can be transmitted using a single carrier.

**Know your progress**

*Prepare a table of comparison of the merits and demerits of different types of AM.*

### 4.11 Frequency modulation (FM)

The main disadvantage of AM is that it is easily affected by noise. The noise affects the amplitude of a signal and in AM signal the message is contained in the amplitude. So if the noise alters the amplitude of AM signal during transmission, then it alters the message or the noise is added to the message. When such an AM signal is demodulated we get only a noise added message signal. How can we reduce the effect of noise during signal transmission? It is possible if we do not add the message to the amplitude of the carrier.

We can add the message to a carrier signal as its frequency varies with the amplitude variations of the message. Such a modulation is called frequency modulation or FM. Here the amplitude of the carrier is kept constant. A FM signal is shown below.

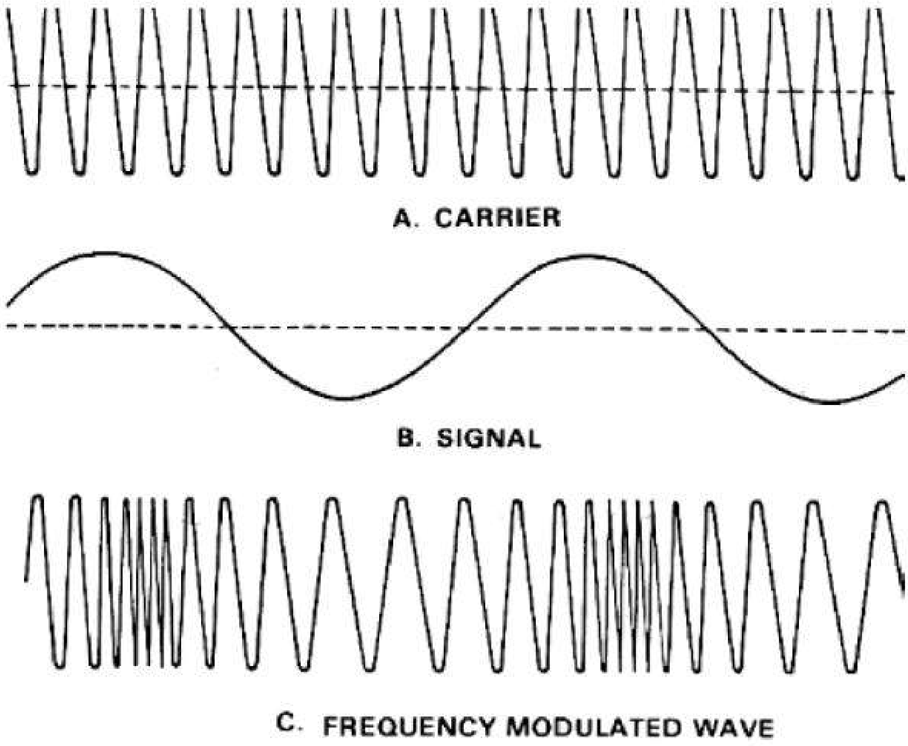


Fig 4.10 FM Modulation

In FM, the frequency of the original carrier will shift according to the amplitude of the message. The deviation of the carrier frequency is proportional to the amplitude of the modulating voltage. The modulation index in FM is defined as the ratio of the frequency deviation of the carrier to the highest frequency in the message signal.

$m_f = \Delta f / f_m$

As an example if a modulating signal of 15kHz causes a frequency deviation of 75kHz, the modulating index is 5. The modulation index in FM varies between zero and infinity.

The instantaneous frequency of FM signal is given by

$f = f_c(1 + kV_m \cos\omega_m t)$

where $f_c$ = unmodulated carrier frequency

k= constant

$V_m \cos\omega_m t$ = instantaneous modulating voltage.

The maximum frequency is given by

$$f = f_c(1 + kV_m)$$

The maximum frequency deviation is $\delta = kV_m f_c$

The modulation index, $m_f = \delta / f_m$

## 4.12 Frequency spectrum of FM

The mathematical expression of FM signal is given by

$$v = A\sin(\omega_c t + m_f \sin\omega_m t)$$

On analysing this expression, we can understand that it is a complicated one and involves sine of a sine term and on expansion, it results in infinite number of terms with different frequencies. From this, we can get an idea that an FM signal has infinite number of sidebands. These sidebands are separated from the carrier on either side by $f_m$, $2f_m$, $3f_m$, $4f_m$ etc.

The amplitude of the sidebands decreases gradually and after a few terms they become insignificant. The frequency spectrum of an FM signal for a specific modulation index is shown below.

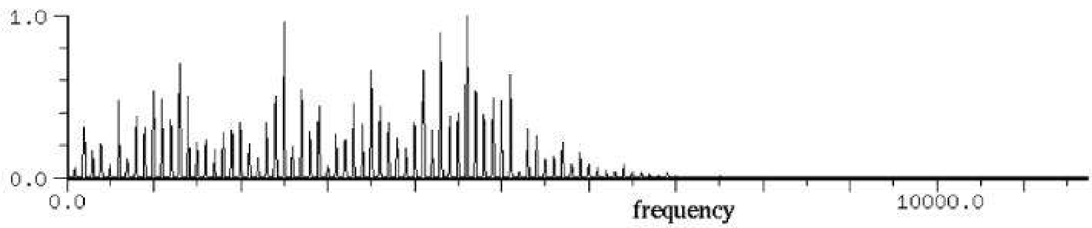


Fig 4.11 spectrum of FM



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It is very important to state that the modulation index determines how many sidebands are having significant amplitudes. When modulation index increases, the number of significant sidebands increases or we can say that in FM, the bandwidth is determined by the modulation index.

Also it is interesting to note that the total transmitted power in FM remains constant independent of the depth of modulation or modulation index. This is because the amplitude of the FM signal is a constant and does not depend on modulation. But in AM, as the depth of modulation increases the sideband power and the total transmitted power increases. In FM, the increased modulation index does not increase the transmitted power, but it increases the number of significant sidebands and hence the bandwidth required to transmit the FM signal increases.

## 4.13 Bandwidth of FM

Theoretically FM has an infinite number of sidebands and hence it has infinite bandwidth. But for practical calculation of bandwidth, we need to consider only the significant sidebands. How many sidebands are required to be considered as significant? A solution for this question came from the scientist Carson and according to him we need to consider ($m_f$+1) number of sidebands on either side of the carrier as significant and thus the total bandwidth required is;

$$\begin{aligned} BW &= 2(m_f + 1)f_m \\ &= 2(\delta/f_m + 1)f_m \\ &= 2(\delta + f_m) \end{aligned}$$

Suppose $m_f = 5$, then BW for FM is $12f_m$. But in AM it is only $2f_m$. It shows that bandwidth requirement in FM is very large compared to that in AM.

## 4.14 Comparison between AM and FM

Did you notice any difference in the quality of sound of AM and FM stations? Obviously the answer is that the quality of FM is superior to that of AM. Let us compare the characteristics of AM and FM below.

1. In FM, almost all the transmitted power is useful whereas, in AM most of the transmitted power is in the carrier which contains no useful information.

2. In FM, the transmitted power is constant. In AM, the transmitted power varies with the depth of modulation. So amplifiers in FM can be designed to handle constant power and hence they are more efficient.
3. FM receivers use amplitude limiters to remove the amplitude variations caused by noise and it will not affect our message as it is not contained in the amplitude. This makes FM reception more immune to noise than AM.
4. FM broadcasts operate in the upper VHF and UHF frequency ranges at which noise is found to be less compared to the MF and HF ranges which are used for AM broadcast.
5. Large bandwidth is required for FM channel than AM (as large as 10 times).This is the most significant disadvantage of FM.
6. The equipment for FM modulation and demodulation is more complex compared to AM.
7. The area of reception of FM is much smaller than that of AM. This is because AM transmission is done at relatively lower frequencies.

The FM modulator and demodulator circuits are not explained in this textbook as they are out of the scope of this textbook.

## 4.15 Radio receivers

The two practical radio receivers are tuned radio frequency (TRF) receiver and superheterodyne receiver. Though the second one is widely used these days, let us discuss TRF receiver first and the operation of the superheterodyne receiver will be dealt with, after citing the drawbacks of the TRF receiver.

**Tuned Radio Frequency (TRF) Receiver**

The block diagram of a TRF receiver is shown below in fig 4.12

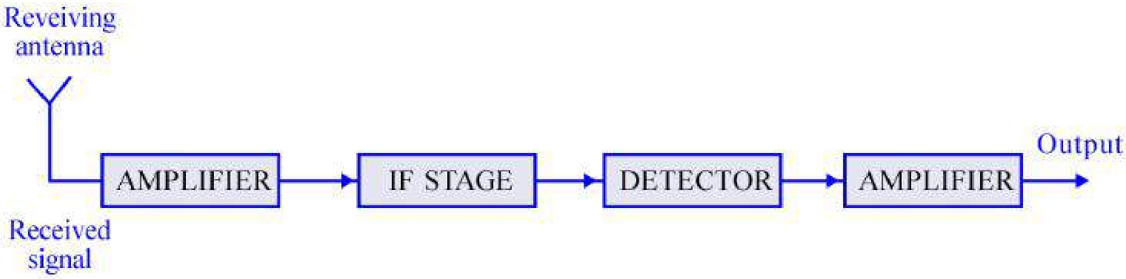


Fig 4.12 TRF receiver

A tuned radio frequency receiver (or TRF receiver) is a type of radio receiver that is composed of one or more tuned radio frequency (RF) amplifier stages followed by a detector (demodulator) circuit to extract the audio signal and then an audio frequency amplifier. Though popular in the 1920s, it was difficult to operate because each stage must be individually tuned to the station's frequency. By the mid 1930s, it was replaced by the superheterodyne receiver. The classic TRF receivers consisted of three sections:

- one or more tuned RF amplifier stages. These amplify the signal of the desired station to a level sufficient to drive the detector, while rejecting all the other signals picked up by the antenna.
- a detector, which extracts the audio signal from the radio carrier signal by rectifying and filtering it.
- one or more audio amplifier stages which increase the power of the audio signal.

The disadvantages of TRF are poor selectivity and low sensitivity. TRF has now become practically obsolete. Selectivity requires narrow bandwidth or high Q, but the bandwidth of a filter with a given Q factor increases with frequency ($Q=f_0/BW$). So to achieve a narrow bandwidth at a high radio frequency high Q or many filter sections are required. In contrast, a superheterodyne receiver translates the incoming high radio frequency to a lower intermediate frequency where it is easier to achieve selectivity.

The major problem with the TRF receiver is its complicated tuning. All the tuned circuits need to be tracked to attain the narrow bandwidth tuning. Keeping multiple tuned circuits align to a particular frequency band while tuning over a wide frequency range is rather difficult. In the early TRF sets, the operator had to perform that task. A superheterodyne receiver only needs to track the RF amplifier and local oscillator stages; the tedious selectivity requirements are confined to the IF (intermediate frequency) amplifier which is fixed-tuned and hence easy. Here the received signals from various stations will be converted to a common IF frequency before applying to the IF stage. Thus IF amplifier always gets signals of constant frequency band and hence need not be tuned while receiving signals from different stations.

## Superheterodyne Receiver

One of the most common forms of radio receiver is the superhet or superheterodyne radio receiver. Virtually all broadcast radio receivers, as well as televisions, short wave receivers and commercial radios use the superheterodyne principle as the basis of their operation.

The superheterodyne radio technique is used in most radios found in households. The basic concept for it was developed way back in 1918, and its invention is credited to a brilliant American engineer Edwin Armstrong who constructed the first superhet radio.

The superheterodyne radio receiver, although more complicated than TRF, offers many advantages in terms of performance, particularly its selectivity. In this way it is able to remove unwanted signals more effectively than other forms like TRF.

## Basic superheterodyne concepts

The superheterodyne radio operates by taking the incoming signal of a transmitting station, mixing it with a locally generated signal. This signal is generated by a local oscillator (LO). After mixing, a lower frequency called intermediate frequency (IF) signal is generated. Now our message is there in this IF signal and it is then passed through a high performance fixed frequency filter and amplifier. This amplified signal is then demodulated to extract the required message signal.

The main process in the superheterodyne radio is mixing.

**Mixers and Mixing:**

The mixing process used in RF circuits is not similar to that used in audio mixers where signals are added together. For RF mixers a totally different concept is used, the circuit is non-linear and this has the effect of multiplying them together. When two signals of frequencies f1 and f2 are mixed, two new signals with frequencies f1 + f2 and f1 - f2 are produced.. In other words if signals of 3MHz and 4 MHz enter a mixer, then new signals at frequencies of 1 MHz (difference) and 7 MHz (sum) will be generated.

The tuning of the receiver is simply accomplished by changing the frequency of the local oscillator. The frequency of local oscillator is adjusted such that when it mixes with the incoming signal frequency IF signal is obtained.

It is often helpful to look at an example to illustrate how the process works. Consider two signals from two different transmitting stations, one at 1.0 MHz and another at 1.1 MHz. If the IF filter is centered at 0.25 MHz, and the local oscillator is set to 0.75 MHz, then the two signals generated by the mixer for the 1.0 MHz input signal are 0.25 MHz (1MHz-0.75MHz) and 1.75 MHz (1MHz+0.75MHz). Naturally the 1.75 MHz signal is rejected by the filter, but the one at 0.25 MHz passes through the IF stages because the passband of the filter is 0.25MHz.

The signal at 1.1 MHz produces a signal at 0.35 MHz (1.1-0.75) and another at 1.85 MHz (1.1+0.75). Both these signals fall outside the bandwidth of the IF filter so that the only signal to pass through IF is that from the input signal 1MHz or we can say that the input signal 1MHz is selected.

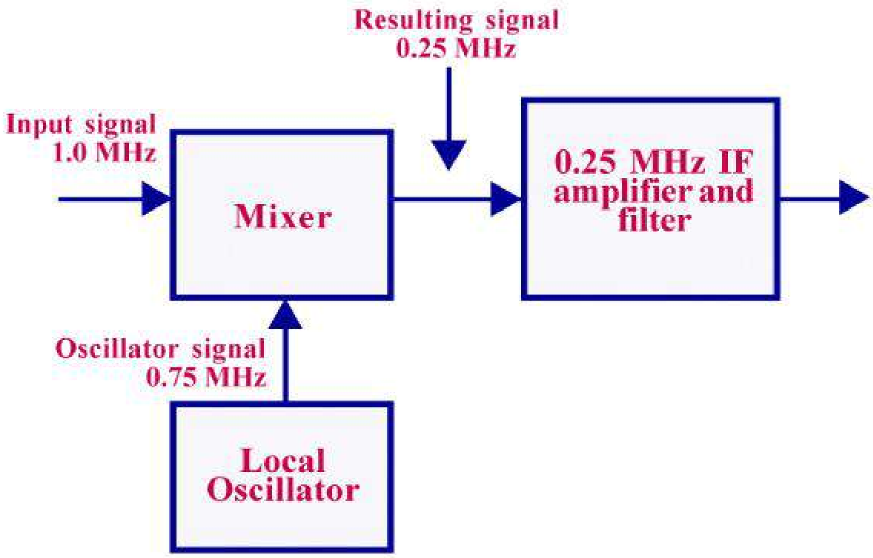


Fig. 4.13 heterodyning

By tuning, if the local oscillator frequency is moved to 0.85 MHz then the signal at 1.1 MHz will give rise to a signal at 0.25 MHz (1.1-0.85) and another at 1.95 MHz (1.1+0.85). The IF filter passes the 0.25MHz signal obtained from the input signal of 1.1MHz or this 1.1MHz input signal is selected by the receiver. The signal at 1.0 MHz will give rise to a signal of 0.15 MHz (1-0.85) and another at 1.85 MHz(1+0.85) and both will be rejected. So the input station frequency 1.1 MHz is selected by the receiver and 1MHz is rejected. In this way the receiver acts as a variable frequency filter, and any input station can be tuned by varying the frequency of the local oscillator.

The advantage of the superheterodyne radio process is that very selective fixed frequency IF filters can be used and these far out perform any variable frequency ones.

***Image frequencies in the superheterodyne radio receiver***

The basic concept of the superheterodyne receiver appears to be fine, but there is a problem. The major limitation of super heterodyne receiver is the problem of image frequency. The image frequency results in two stations being received by the receiver at the same time. This leads to interference. With the local oscillator set to 0.75 MHz and with an IF of 0.25 MHz, it has already been seen that a signal at 1.0 MHz mixes with the local oscillator to produce a signal at 0.25 MHz that will pass through the IF filter. However, if a signal at 0.5 MHz enters the mixer it produces the sum frequency 1.25 MHz and the difference frequency of 0.25 MHz. This can be proved to be a problem because it is perfectly possible that the two signals of completely different input frequencies (1 MHz and 0.5 MHz) enter the IF. The unwanted frequency (0.5MHz) is known as the image. However it is possible to place a tuned circuit before the mixer to prevent the image signal entering the mixer, or reduce its level to an acceptable value.

The image signal will be separated from the wanted signal by a frequency equal to twice the IF. $F_I = Fs + 2F_{IF}$. In other words with an IF at 0.25 MHz, the image will be 0.5 MHz away from the wanted frequency.

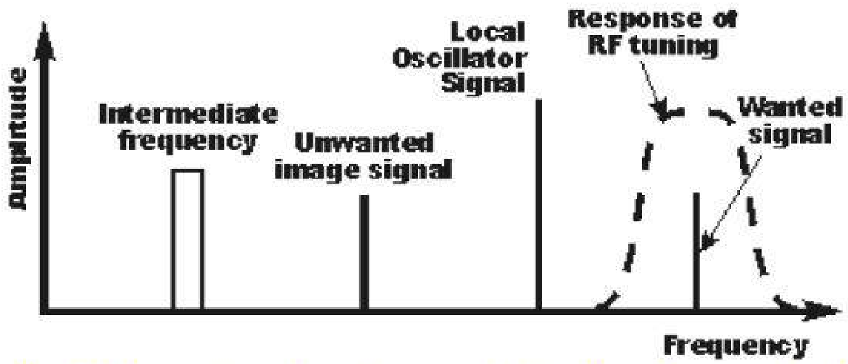


Fig. 4.14 spectrum showing wanted signal and image signal

## The working of a Superheterodyne Receiver

Having looked at the concepts behind the superheterodyne receiver, it is helpful to look at a block diagram of a basic superheterodyne receiver. Signals enter the tuner circuitry from the antenna. This tuning of the superheterodyne receiver removes the image signal and often includes an RF amplifier to amplify the signals before they enter the mixer. This tuning selects only one input signal and rejects the image frequency. For example, in the previous case when tuner is tuned to select the 1MHz input signal, it rejects the image frequency of 0.5MHz. The amplification is to ensure that a good signal to noise ratio is achieved before mixing.

The tuned and amplified signal then enters one input of the mixer. The output of local oscillator is connected to the other input. The local oscillator consists of a variable frequency oscillator that can be tuned with the help of a variable capacitor.

Once these signals leave the mixer, they enter the IF stages. Most of the amplification within the receiver takes place in this stage. Filtering which enables signals of one frequency to be separated from others also takes place in this stage.

Once these signals pass through the IF stages of the superheterodyne receiver, they need to be demodulated. The output from the demodulator is the recovered audio. This is passed through the audio stages where they are amplified and presented to the headphones or loudspeaker.

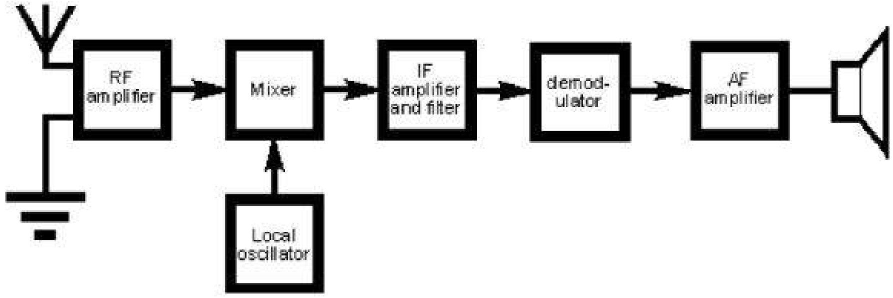


Fig. 4.15 Block diagram of a basic superheterodyne receiver

The IF stage is very important as it provides maximum gain to the receiver. The stages of an IF amplifier are tuned to a fixed frequency that does not change as the receiving signal frequency changes. This is achieved by mixing. The fixed frequency simplifies the optimization of the the parameters of IF amplifier. The IF amplifier is selective about its center frequency $f_{IF}$. The fixed center frequency allows the stages of the IF amplifier to be carefully tuned for the best performance (this tuning is called 'aligning' the IF amplifier). If the center frequency is changed with the receiving frequency, then the IF stages would have to track their tuning. That is not the case with the superheterodyne.

When used at high frequencies, many amplifiers show a constant gain-bandwidth product characteristic. If an amplifier has a gain-bandwidth product of 100 MHz, then it will have a voltage gain of 100 at 1 MHz but only 10 at 10 MHz. If the IF amplifier needs a voltage gain of 10,000, then it requires only two stages with an IF at 1 MHz, but four stages at 10 MHz. Thus it is better for the amplifier to operate at a constant frequency where it has a maximum gain. The superheterodyne receiver actually does this by converting different station frequencies into a common IF frequency.

In order to avoid interference in receivers, common IF frequencies are not assigned to different types of transmitting stations. Standard intermediate frequencies used are 455 KHz for medium-wave AM radio, 10.7 MHz for broadcast FM receivers, 38.9 MHz (Europe) or 45 MHz (US) for television, and 70 MHz for satellite and terrestrial microwave equipment.

The received signal is now processed by the demodulator stage where the audio signal (or other baseband signal) is recovered and then further amplified. AM demodulation requires the simple rectification of the RF signal and a simple RC low pass filter to remove remnants of the intermediate frequency. FM signals may be detected using a discriminator, ratio detector, or phase-locked loop. The resulting audio signal is then amplified which then can be used to drive a loudspeaker.

### Know your progress

1. Discuss the challenge involved in designing the IF amplifier of a TRF receiver. How can we overcome it in a superheterodyne receiver?
2. In a superheterodyne receiver, the IF is 200 KHz and local oscillator frequency is 800 KHz. Find the tuned station frequency and the image frequency.
3. Prepare a list of AM and FM radio stations in your locality and their frequency. Identify the main difference.

## Comparison between AM and FM bandwidths

The AM band of frequencies occupies between 535 kHz and 1605 kHz. Each AM channel is assigned a range of frequencies, typically about 10 kHz wide. The common frequency identification (such as AM 1190 for America AM 1190) represents the midpoint of this operating channel range. The audible range of the human ear is upto 20 kHz, and AM has only 5 KHz. For sounds other than human speech (eg. sound of musical instruments) this is inadequate and that is the reason why AM sounds imperfectly. Then there is the FM band of frequencies. These correspond to the electromagnetic waves with frequencies between 88 MHz and 108 MHz. Each FM channel is assigned a bandwidth of 200 kHz, and the midpoint of this operating channel range is the carrier frequency and this frequency is used for the identification purposes of the channel. (Club FM 94.3, Radio Mango 91.9). The FM bandwidth can easily cover the audible range of the human ear which is about 20 kHz, and that is why FM radio sounds better than AM radio. In fact, the bandwidth allows FM to be broadcast in stereo.

The AM signal is subject to interference from electrical storms (lightning) and other electromagnetic interference. But FM is relatively immune to such noise. At the same time, greater fidelity is possible by spacing station frequencies further apart. Instead of 10 KHz apart, as in the AM band, FM channels can be 200 kHz (0.2 MHz) apart.

## Let us consolidate

Modulation is the process by which amplitude, frequency or phase is varied in accordance with the instantaneous value of the modulating signal. Modulation becomes necessary to separate signals from different transmitters so as to avoid them from interfering one another, to reduce the size of antenna and to increase the power of signal radiation. The modulation index (m) of AM is the ratio of modulating signal voltage to the carrier voltage. Its value ranges from 0 to 1. AM wave contains the fundamental carrier frequency, the lower side band frequency (fc - fm) and the upper side band frequency (fc + fm). The band width of a signal is defined as the range of frequencies contained in it. So for AM signal, the band width is 2fm. The total power contained in AM wave is

$pt = pc\left(1+\frac{m^2}{2}\right)$. The maximum power contained is equal to 1.5 pc. A CE amplifier circuit can be used for constructing an AM generator. A rectifier with a LPF is used as AM demodulator. On the basis of power to be transmitted and the band width availability AM is of different types.

DSBFC, SSB, SSBSC, VSB and ISM are these types. In FM, the frequency is varied in accordance with the instaneous value of the modulating signal. The modulation index of FM is δ/fm. FM contains fininite member of side bands which are fm, 2fm, 3fm etc. Practically the BW of FM can be calculated by the formula BW = 2(δ + fm). FM is more immune to noise than AM. But larger band width is needed for FM than that for AM. Tuned radio frequency receiver contains one or more amplifier stages followed by the detector. It has poor selectivity and sensitivity. These drawbacks are overcome by super heterodyne receiver. SHR generates a local oscillator frequency which is mixed with the incoming signal to produce the intermediate frequency signal (IF). This is amplified and demodulated to extract the message signal. It can be seen that the image frequency $F_I = Fs + 2F_{IF}$.

The contents of this unit were learned through general discussion, group discussion, sketching wave forms and preparing charts.

## Let us asses

1. In communication, a message signal is transmitted from source to destination. The message signal can be a

   a) video signal   b) audio signal   c) digital signal   d) any of these

2. In modulation any one parameter of a carrier is varied according to the message. In FM, which of the following parameters of the carrier is varied?

   a) amplitude   b) time period   c) phase   d) none of these

3. During AM modulation, the distortion of the carrier increases when the modulation index 'm'

   a) increases   b) decreases   c) independent of 'm'

4. The message is retrieved from the modulated signal using demodulation. During AM demodulation, the distorted message is obtained if

   a) m=0   b) m=0.5   c) m=1   d) m=1.1

5. A message of bandwidth 0-5 kHz is used to modulate a carrier of 50 kHz. The USB falls in the range of

   a) 0-5kHz   b) 45-50 kHz   c) 25-30 kHz   d) 50-55 kHz

6. The noise immunity of a system determines the quality of communication. FM has superior noise immunity because

   a) Noise affects amplitude of a signal
   b) Message is kept in frequency
   c) Amplitude limiter can be used in FM
   d) All of the above

7. The main gain providing stage in superhet receiver is

   a) RF amplifier  b) IF amplifier  c) mixer  d) none of these

8. Modulation is a process of adding message signal to the carrier signal.

   a) Mention three methods of doing this.
   b) Draw AM, FM, PM signals when message is a binary data.
   c) How does modulation help to reduce the length of the antenna?

9. The band width of a signal is the range of frequencies contained in it.

   a) Write the expression for band width of AM.
   b) Show that it is twice the band width of the message signal.

10. It is said that modulation index should not exceed unity in AM. Considering message as a sine wave

    a) Draw an AM signal with index greater than one.
    b) Draw the demodulated message in this case.
    c) Give reason for the distortion of the message.

11. Compare the characteristics of AM and FM.

12. The superhet is the widely used receiver.

    a) Discuss the drawback of the earlier TRF receiver.
    b) What modification is done in superhet receiver?
    c) Explain the working of superhet receiver.
    d) What is image frequency?

13. In order to save bandwidth, AM is modified to obtain different types.

    a) Name three types of AM.
    b) Discuss the merits of SSB.
    c) Discuss the reason for developing VSB.
    d) Mention the advantages of ISB.



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# 5 COMMUNICATION SYSTEMS

## Significant Learning Outcomes

*After completing this chapter the learner:*

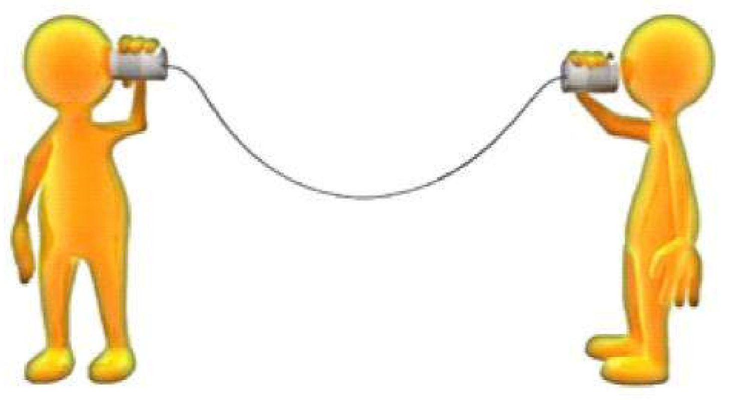
- explains the origin and history of development of communication system.
- points out the allocation of different frequency bands.
- explains the invention of radio waves and its propagation.
- points out the differences between long, medium and short wave propagation.
- differentiates ground wave and sky wave.
- identifies the different layers of Ionosphere.
- explains critical frequency and maximum usable frequency.

Ever since the ancient times, people have been continuously devising new techniques and technologies for communi cating their ideas, needs, and desires to others. Thus, many forms of increasingly complex communication systems have appeared over the years. One of the earliest known communication systems, for example, was the use of fire signal by the Greeks in the eighth century B.C. for sending alarms, calls for help, or announcements of certain events. When the receiver was the human eye, line-of-sight transmission paths were required, and the atmospheric effects such as fog and the rain made the transmission path unreliable. Improvements of these systems were not possible very actively because of the limitations of technology at that time.

The invention of the telegraph by Samuel F. B. Morse in 1838 (using Morse code, ie.dot and dash) opened a new world in communications,and a new era in electrical communications. In the later years, increasingly sophisticated and more reliable electrical communication systems with progressively larger information capacities were developed and deployed.

## 5.1 Basic Communication System

Let us see the framework of a basic communication system. The figure 5.1 shows the block diagram of the system. In the transmitter section, the microphone converts the audio signals into electric signals. As we have studied



*Try the above method using two cups and a long cotton thread. Voice can travel long distance through a medium (cotton thread) than through free space.*

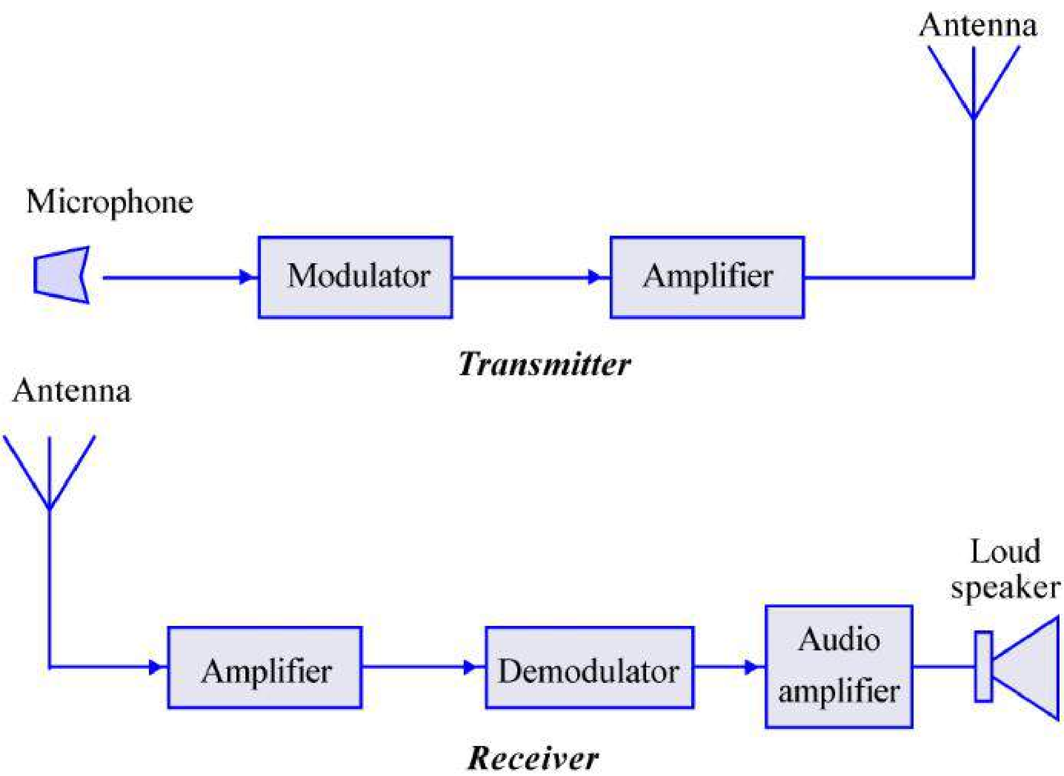



Fig. 5.1 *A basic communication system.*

in the previous chapter, the audio signal can be transmitted only after modulation. After modulation, the signal is amplified and transmitted through an antenna. In receiver, the antenna receives the modulated signal. This signal is very week, so it is amplified using an amplifier and demodulated to separate the message signal. This signal is amplified using an audio amplifier and is reproduced using a loud speaker. In radio and TV, we know that different channels are allotted different carrier frequencies. The carrier frequencies refer to high frequency sinusoids, which are generally referred to as radio waves.

## 5.2 Radio waves

Radio waves are a form of electromagnetic radiation. The velocity (or speed) of a radio wave radiated into free space by a transmitting antenna is equal to the speed of light—186,000 mile per second or 300,000,000 metre per second. A radio wave can travel large distances with very less attenuation and hence, can be used for communication without the need of a wired medium. The transmission of radio waves was first predicted in the mathematical work done in 1867 by Scottish mathematical physicist **James Clerk Maxwell**. He proposed equations that described the possibility of the transmission of electromagnetic waves. In 1887, **Heinrich Hertz** experimentally proved Maxwell's equation, by demonstrating the transmission of electromagnetic waves through a short distance in his laboratory. Different frequency bands are used for various communication purposes, such as AM broadcasting, amateur radio, cellular telephony, satellite communication, radar etc. The table 5.1 shows the details of frequency band allotment.

| Frequency Band | Designation | Propagation Characteristics | Typical Uses |
|---|---|---|---|
| 3-30 kHz | Very Low Frequency (VLF) | Ground wave: low attenuation during day and night; high atmospheric noise level. | Long-range navigation; submarine communication |

| Frequency Band | Designation | Propagation Characteristics | Typical Users |
|---|---|---|---|
| 30-300 kHz | Low Frequency (LF) | Similar to VLF, less reliable; absorption during daytime | Long-range navigation and marine communication radio beacons |
| 300-3000 kHz | Medium Frequency (MF) | Ground wave and night sky wave; attenuation low at night and high in day; atmospheric noise | Maritime radio, direction finding, and AM broad casting. |
| 3-30 MHz | High Frequency (HF) | Ionospheric reflection varies with the time of day,season,and frequen cy; low atmos pheric noise at 30 MHz | Amateur radio; inter national broadcasting, military communication, long-distance aircraft and ship communication, telephone, telegraph, facsimile |
| 30-300 MHz | Very High Frequency (VHF) | Nearly line-of-sight (LOS) propagation with scattering because of temperature inversions; cosmic noise | VHF television, FM two-way radio, AM aircraft communication, aircraft navigational aids. |
| 0.3-3 GHz | Ultrahigh Frequency (UHF) | LOS propagation, cosmic noise | UHF television, cellular telephone, navigational aids, radar, GPS, microwave links, personal communication systems |
| 3-30 GHz | Super high Frequency (SHF) | LOS propagation; rainfall attenuation above 10 GHz, atmospheric attenuation because of oxygen and water vapour, high water vapour absorption at 22.2 GHz | Satellite communication, radar microwave links |
| 30-300 GHz | Extremely High Frequency (EHF) | Same; high water-vapour absorption at 183 GHz and oxygen absorption at 60 and 119 GHz | Radar, satellite, experimental |

Table 5.1 Frequency Bands for Communication

## 5.3 Long, medium and short waves

The long wave refers to those parts of the radio spectrum which lies in the longer wavelength region for wavelength greater than 1000 metres. The frequency of long wave is very low because the wavelength is very high and goes up to 300 kHz. The long wave signal travels between two points by means of surface wave or sky wave propagation. These waves are attenuated to a large extent when covering large distances and hence suitable only for short distance communications.

The frequencies ranging from **300 kHz to 3000 kHz are known as medium frequencies and the band is known as Medium Wave band.** The medium wave frequency is larger than long wave frequency because the wave length of the medium wave is smaller than that of the long wave.

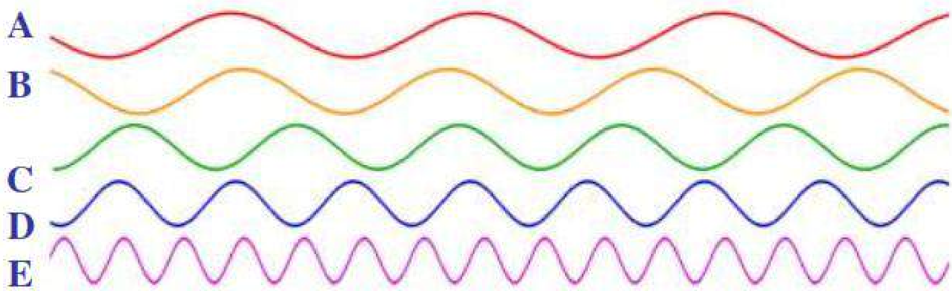



Fig 5.2
(A &B) Long wave length and low frequency wave
(C&D) Medium wave length and medium frequency wave
(E) Shorter wave length and high frequency wave

The short wave refers to those parts of the radio spectrum which lies in the short wavelength region. A short wave has the property of travelling longer distance with lesser attenuation than that of a long and medium radio wave. These waves can travel to any distance on the earth. It is not unusual to pick up a radio station of a distant country when a person is tuning a radio set on short waves. The short wave was first discovered in 1921 and the amateur radio started using them since 1922.

(i) Find out how many radio broadcasting stations are there in Kerala?

(ii) Do you find any difference between them?

(iii) List the transmission frequencies of different radio stations in Kerala.

(iv) Listen to different AM and FM stations in Kerala and observe the differences in the quality of sound between AM and FM stations.

## 5.4 Propagation of radio waves

The radio waves of different frequencies have different propagation characteristics and they can travel through vacuum, air, or other transmission media. In Earth's atmosphere, radio waves can either reflect off from the ionosphere and thus travel to long distance (sky wave), or bend or reflect very little and travel on a line of sight (ground wave). Now, let us see how electromagnetic waves of different frequencies are propagated through various means.

**Know your progress**

- Describe the frequency range and uses of VHF and UHF bands.
- What is the speed of a radio wave?

## 5.5 Ground Waves

A ground wave can be either a **Surface Wave** or a **Space Wave**. A surface wave travels along the surface of the Earth while a space wave travels over the surface. Ground wave propagation is generally used in TV, radio broadcasting etc.

### Surface wave

A surface wave reaches the receiving site by traveling along the surface of the ground as shown in figure 5.3. When a surface wave meets an object and the dimensions of the object do not exceed its wavelength, the wave tends to be curved or bent around the object. Thus the waves emanating from the transmitting antenna travel to the receiving antenna as shown in fig. 5.3.

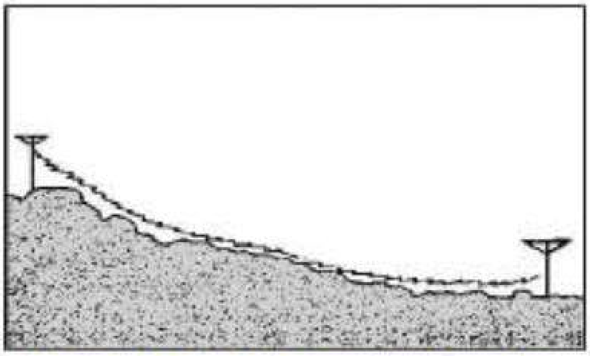


Figure 5.3 Surface wave propagation.

### Space wave

The space wave follows two distinct paths from the transmitting antenna to the receiving antenna, one through the air directly and the other reflected from the ground to the receiving antenna. This is illustrated in figure 5.4. The primary path of the space wave is direct from the transmitting antenna to the receiving antenna. So, the receiving antenna should be located within the radio horizon of the transmitting antenna.

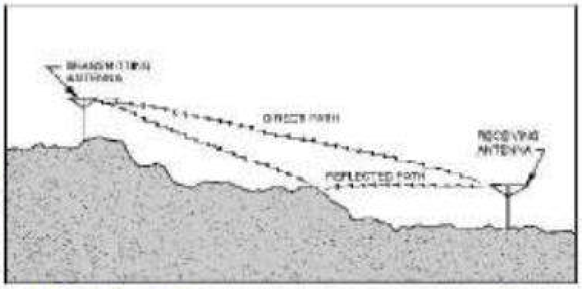


Fig 5.4 The space wave propagation

Although space waves suffer little ground attenuation, they are susceptible to fading. This is because space waves actually follow two paths of different lengths (direct path and ground reflected path) to the receiving site and, therefore, may arrive in or out of phase. If these two waves are received in phase, the result is a reinforced or stronger signal. Likewise, if they are received out of phase, they tend to cancel each other, resulting in a weak or fading signal.

## 5.6 Sky Wave

The sky wave, often called the ionospheric wave, is radiated in an upward direction and returned to Earth at some distant location because of the reflection from the ionosphere. This form of propagation is relatively unaffected by the Earth's surface and can propagate signals over great distances. Usually the high frequency (HF) band is used for sky wave propagation. Let's study the various layers of ionosphere and its effect on sky waves to understand the nature of sky wave propagation.

## 5.7 Ionosphere layers

Ionosphere is a region of the upper atmosphere extending from a height of about 50 km upto 500 km. Within this region, some molecules of air become ionized by solar radiations from the sun. Two kinds of solar radiations cause ionization in the atmosphere, namely x-rays and ultraviolet radiations.

Because of the heavy temperature of the sun the upper portion of the atmosphere gets heated and the particles become charged or ionized. This ionized layer is called ionosphere. This ionization causes electrons to be free to move about. The electrons that have been stripped off are free to move. The free electrons under the influence of an electromagnetic wave will absorb and radiate energy, and modify the direction of the electromagnetic wave front. The free electrons in the ionosphere cause radio waves to be refracted (bent) and reflected back to earth eventually. The greater the density of electrons, higher the frequencies that can be reflected by the ionosphere. According to the ionization density and the distance from the earth the ionospheric layer can be split into four sub-layers.

The four layers of ionosphere and their approximate height ranges are:

- **D** region 50 to 90 km;
- **E** region 90 to 140 km;
- **F1** region 140 to 210 km;
- **F2** region greater than 210 km.

Out of these layers D and E layers do not help the communication significantly. But the F1 and F2 layers aid communication.

During daylight hours, the F region separates into distinct regions called F1 and F2. The ionization levels in these regions can be quite high, and vary greatly depending on the degree of ultraviolet radiation from the sun, which in turn is primarily determined by the altitude of the sun (ie. time of day). At night, the ionisation density of the atmosphere is low, and because of this, the recombination of ions occurs slowly after sunset. Hence F1 region merges with F2 region. The F regions are responsible for long distance communications due to their ability to reflect signals up to 30 MHz.

## 5.8 Critical frequency

The critical frequency of a given layer is the highest frequency of the wave that will return to Earth due to the reflection, from the layer in the ionosphere, if the beam is directed straight at it. If the frequency of the sky wave is increased

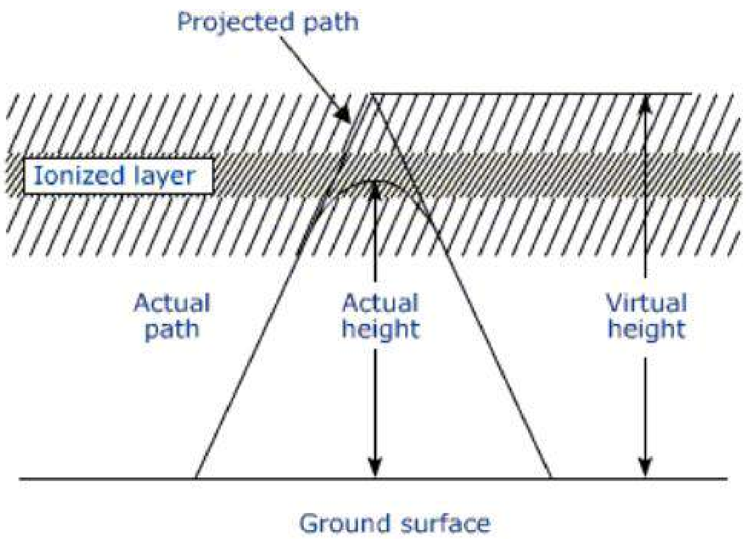



Fig 5.5 Actual and virtual heights of an ionized layer

beyond the critical frequency, it will not come back to earth after reflection. The critical frequency of a layer is determined by its ionization density.

## 5.9 Maximum usable frequency

Although the critical frequency of any layer represents the highest frequency that will be reflected back from that layer at vertical incidence, it cannot be the highest frequency that can be reflected from the layer. The highest frequency that can be reflected depends also upon the angle of incidence, and hence, for a given layer height, it depends upon the distance between the transmitting and receiving points. The maximum frequency that can be reflected back for a given distance of transmission is called the maximum usable frequency (MUF) for that distance.

In radio wave transmission, **maximum usable frequency** (**MUF**) is the highest radio frequency that can be used for transmission between two points via reflection from the ionosphere at a specified time, independent of the transmitter power. This index is especially useful with regard to shortwave transmissions. It is seen that the MUF is related to the critical frequency and the angle of incidence.

$$\textbf{MUF} = \textit{Critical frequency} / \cos \theta$$

Where "θ" is the angle of incidence.

## 5.10 Skip Distance

The skip distance is the shortest distance from a transmitter, measured along the surface of the Earth, at which, a sky wave of a particular frequency returns to Earth. Fig 5.6 illustrates the signals reflected from the ionosphere and the skip distance. A receiver placed at a distance within the skip distance will not be able to receive the transmitted signal.

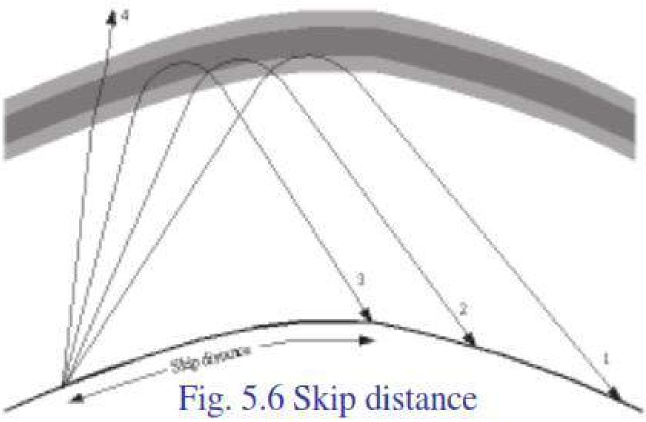



Fig. 5.6 Skip distance



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The skip distance depends up on the frequency of the transmitting wave, the angle of transmission and the transmission path.

## Know your progress

1. Observe how many channels are there in your TV and note down their respective frequencies.
   Write down the name of the channel and respective frequency in a table.
2. Find how many mobile network providers are there in your city and corresponding frequency of transmission.

## Let us consolidate

Communication is a process in which data is transferred in the form of signals from one point to another. Radio waves are a type of electromagnetic radiation that propogates in different frequencies in communication system. For different applications different frequencies are used. Ground wave propagation is of two types: surface wave and space wave. Surface wave reaches the receiving end by traveling along the surface of the ground. The space wave reaches the receiving antenna through the air and reflected from the ground. The sky wave, often called the ionospheric wave, is radiated in an upward direction and reflected to the Earth station from some distant location due to refraction in the ionosphere. The four layers of ionosphere are D layer, E layer, F1 layer and F2 layer. Critical frequency of sky wave is the maximum frequency in which the sky wave after reflecting from the ionosphere return to earth. In radio wave transmission the maximum usable frequency (MUF) is the highest radio frequency that can be used for transmission between two points via reflection from the ionosphere at a specified time, independent of transmitter power.The skip distance, the shortest distance from a transmitter measured along earth's surface at which a sky wave of a particular frequency returns to earth.

The contents of this unit were learned through general discussion, group discussion and with the help of ICT.

## Let us asses

1. The transmission of radio waves was first predicted in the mathematical work done in 1867 by Scottish mathematical physicist James Clerk Maxwell. He desired equations that described the possibility of the transmission of electromagnetic waves. Name the scientist who experimentally proved Maxwell's equation in 1887.

2. Radio waves have the frequencies ranging from 3 kHz to 300 GHz. This frequency range is further divided into (eight) different frequency bands, with various propagation system and use.
   (a) What are the eight different radio frequency bands?
   (b) Give the propagation details of different frequency bands.
   (c) Point out the typical uses of different radio frequency bands.

3. Surface wave and space wave are also called ground waves. A surface wave travels along the surface of the earth, while the space wave travels over the surface.
   (a) What are the applications in which the ground wave propagation is used?
   (b) Differentiate the propagation methods of surface wave and space wave.

4. Ionosphere is a region of the upper atmosphere extending from a height of about 50 km up to or greater than 500 km. Within this region some molecules of air become ionized by solar radiations.
   (a) What kind of radiations causes ionization in the atmosphere?
   (b) Explain the process of ionization that occurs in the ionosphere due to the solar radiations.
   (c) Give the different layers of ionosphere with its range.
   (d) What happens to the ionospheric layers during day time and at night time?

5. The sky wave is radiated in upward direction and returned to earth at some distant location because of refraction and reflection from the ionosphere. Usually the high frequency band is used for sky wave propagation.
   (a) Give the significance of critical frequency in sky wave propagation.
   (b) Differentiate the critical frequency and maximum usable frequency in the sky wave propagation.
   (c) Explain the term skip distance in the sky wave propagation and give the different factors that affect the skip distance.

6

# DATA COMMUNICATION

## Significant Learning Outcomes

*After completing this chapter the learner:*

- explains the difference between continuous modulation and pulse modulation.
- describes the concept of sampling.
- explains the differences between PAM, PCM,PWM and PPM.
- explains the need for multiplexing.
- describes the concept and difference between TDM and FDM.
- demonstrates various digital modulation.
- explains different modulation schemes such as ASK, FSK, and PSK

We know that a computer processes digital signals or digital data. When you print picture saved in your computer using a printer the data corresponding to the picture is sent by the computer to the printer. Therefore, whenever there is a communication between computers or between computers and their input or output devices digital data corresponding to the communication is transferred. So data communication takes place when you print a document, when you download some information from the internet or when you send an email. The data stream is similar to a square wave form with rapid transitions from one voltage level to the other. From the binary data, the samples and from the samples the continuous signal can be reproduced at the receiver. We know that a voice or music can be stored in computer with the help of a microphone connected to it. Here the continuous signal produced by the microphone is sampled and then converted into digital so that data can be stored in the computer.

## 6.1 Sampling

We have already discussed the continuous modulation schemes in which the information or message signal is continuous. The examples are AM and FM. Most of the naturally occurring messages are continuous. Examples are audio signal and video signal. In digital TV transmission, this video signal has to be converted into digital. In the present system of Telephone communication our voice is converted into digital form and transmitted. The first step in the conversion of continuous signal to digital signal is sampling. The sampling of a continuous message is done at regular intervals. A continuous signal and its samples taken at regular intervals are shown below.

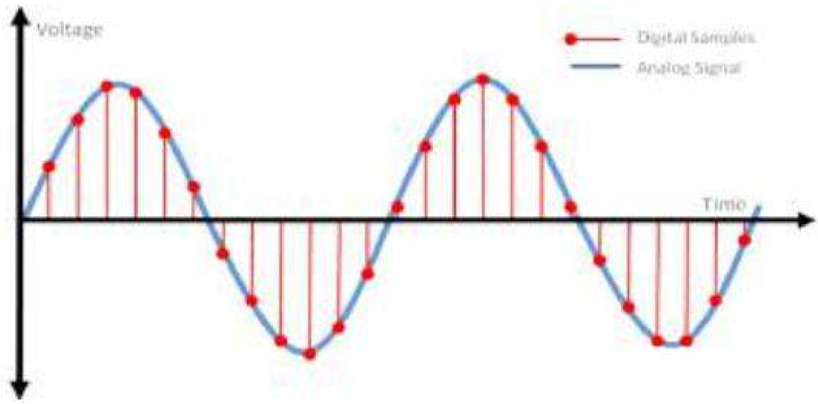


Fig 6.1 Sampling a continuous signal

The sampling interval in the above figure is 'T'. For a continuous signal x(t), its samples are indicated as x(nT); where n = 0,1,2,3, ... For example, x(5T) is the 5th sample value of x(t). Sample value means height of the sample. Instead of transmitting the continuous message, its samples are transmited only or in other words, the information at the time of samples is transmitted. At the receiver side, the original continuous message can be reconstructed from these samples. The quality of the reconstructed signal or the closeness of the reproduced signal to the original signal depends on the frequency of the samples. If the number of samples taken in one period of the signal is large, then the quality of the reconstructed signal will be good. But large number of samples will create large amount of data to be transmitted.

**Know your progress**

*Find the sample values of a sine wave with amplitude 5V and time period 10msec when sampling starts at the origin and sampling period is 2.5msec.*



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## 6.2 Sampling theorem

What should be the minimum sampling rate, so that the message signal can be reconstructed with minimal error? The sampling theorem states that if the sampling rate (fs) exceeds twice the maximum frequency contained in the message, (fm) then the signal can be reconstructed at the receiver with minimal distortion.ie, $fs \geq 2fm$. This theorem is used to determine the minimum sampling rates. As an example, consider the transmission of speech over telephone channel. The channel supports a frequency range of 300 Hz to 3400 Hz . For this application a sampling rate of 8000 samples per second is the standard. Note that, this rate is satisfactorily more than twice the highest audio frequency According to sampling theorem the minimum sampling rate required here is $2 \times 3400 = 6800$ Hz

Pulse modulation can be divided into two categories; analog and digital. In the analog system, sample amplitude can be of any value. In the latter case, the sample value is rounded to the nearest predefined sample values. Each of these sample voltages is assigned a digital code and it will be transmitted instead of the actual sample. Pulse amplitude and pulse position modulation are both analog and pulse code modulation is a digital modulation scheme. We will discuss pulse code modulation (PCM) in this chapter, since it is the one used mostly in practice.

## 6.3 Pulse code modulation (PCM)

Here, the continuous signal or message is first passed through a low pass filter in order to remove any unnecessary high frequency noise that may be contained in the message. The samples are generated using sampler which is a switch. This switch will be closed, whenever the samples are to be taken. The samples are quantized to the predetermined voltage levels using quantiser. Each level is assigned with a binary number by the encoder in an ADC. In other words, the quantized sample is converted to a digital number using analog to digital converter (ADC).This binary code represents the sample value. Thus, instead of transmitting the samples, its binary code is transmitted. This modulation is called PCM.

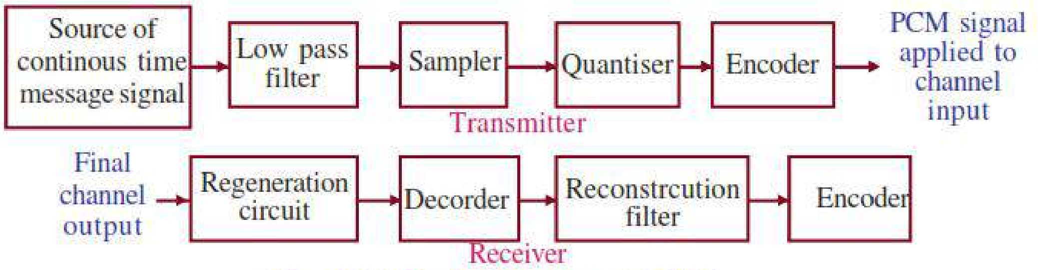


Fig. 6.2 Block diagram of PCM

### Quantization

The samples can have infinitely different values and to convert such different values to binary, we need infinite number of bits. In order, to encode the samples to binary with a few numbers of bits, the sample values have to be restricted to a finite number of levels. Thus the sample values can be rounded off to predetermined fixed number of levels. So we will round off the sample values to predetermined fixed number of levels. This process of rounding off is called quantization.

### A/D conversion

If there are N quantized voltage levels in a PCM system, then we need n number of bits to encode them into binary such that $2^n = N$. These binary data is transmitted through the channel.

At the receiver side, there is regeneration circuit to make the shape of pulses obtained from the channel back to its original form as the shape of pulses would be distorted when they are transmitted through a channel for a very long distance. There are digital to analog converter (DAC) or decoder and reconstruction filter. The DAC converts the binary stream back to the samples. From the samples, the continuous signal is retrieved by using the reconstruction filter. The reconstruction filter is basically a low pass filter.

**Know your progress**

*One of the important steps of PCM is quantization. What is the need of quantization?*

Now let us see an example. Suppose, we have a message signal to be transmitted using PCM method. Consider that the samples produced by the sampler for that signal in a particular time interval are of voltage 2.1, 4.3, 5.8, 7.4 and 0.9. The quantiser converts these samples into 2, 4, 6, 7 and 1. The encoder produces the corresponding binary data as 010, 100, 110, 111 and 001. This bit stream is passed through the channel.

At the receiver side, this bit stream is decoded back to 2,4,6,7 and 1. The reconstruction filter produces the continuous signal from these samples.

## 6.4 Multiplexing

When data from different sources are transmitted through a single channel simultaneously from source to destination, they will get mixed up in the channel and they cannot be separated then. Otherwise, we need a number of separate channels equal to the number of different sources to transmit data from each source separately through individual channel. It is difficult as well as uneconomic to use many channels between a common source and destination. In such a situation, we can use the technique of multiplexing. The two widely used multiplexing techniques are time division multiplexing and frequency division multiplexing.

## 6.5 Time division multiplexing (TDM)

When data is transmitted in the form of pulses, we can think of sending data from different sources through the same channel. When we transmit data from different sources through the same channel in the form of packets, it is called time division multiple access. Here, before the completion of sending data from one source, we will transmit data from other sources. The time is divided into slots and successive slots are allotted for transmission of data from different sources. Consider that there are n sources required to transmit data through a single channel and the corresponding receivers cannot wait until the data from other sources is transmitted completely.

Now we will take a time period T and let it be divided into n slots so that time duration of one slot is T/n. We can transmit data from the first source within the duration of the first T/n time. The second T/n time is used for the transmission of first packet of data from the second source and the third T/n time for sending the first packet of data from the third source and so on. When the first packet of data from the nth source is transmitted, the time T would be elapsed. Then during the next time duration 'T', second packet of data from all the 'n' sources is transmitted.

Consider the delay between successive packets received in a receiving station is T and if this delay is not noticeable there, then the effect of continuous transmission is felt. Or the transmission between all the n sources and their corresponding receivers will happen simultaneously through a single channel. The TDM is illustrated below.

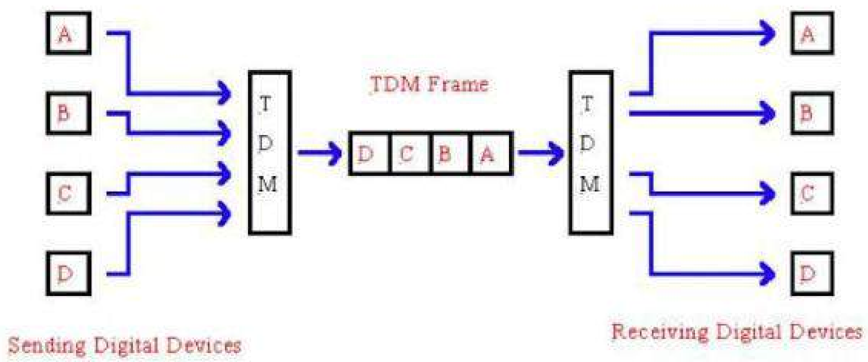


Fig. 6.3 TDM

If the sampling interval for all sources is T, then the time between successive samples is T. In other words we will get time T after sending one sample and before next sample arrives. If we transmit the first samples of all other n-1 sources within this time T, then it will not affect the transmission of the samples of the first source at all. The advantage of TDMA is that we can transmit data from n sources simultaneously using a single channel. Otherwise, we require n different channels for simultaneous transmission.

## 6.6 Frequency division multiplexing (FDM)

Here, a number of signals are transmitted through a single channel simultaneously without dividing the time. But the available bandwidth of the channel is divided and frequency slot is allocated for the transmission of each message. If the message signals are occupying the same frequency range, then those signals are shifted to different frequency slots by using AM modulation. Now, these signals can be transmitted through a single channel at the same time without being mixed up.

At the receiver side these signals can be separated with the help of band pass filters whose pass band matches with frequency slot of the corresponding signals. The output of each band pass filter is demodulated to obtain the individual messages. The block diagram of FDM is shown below.

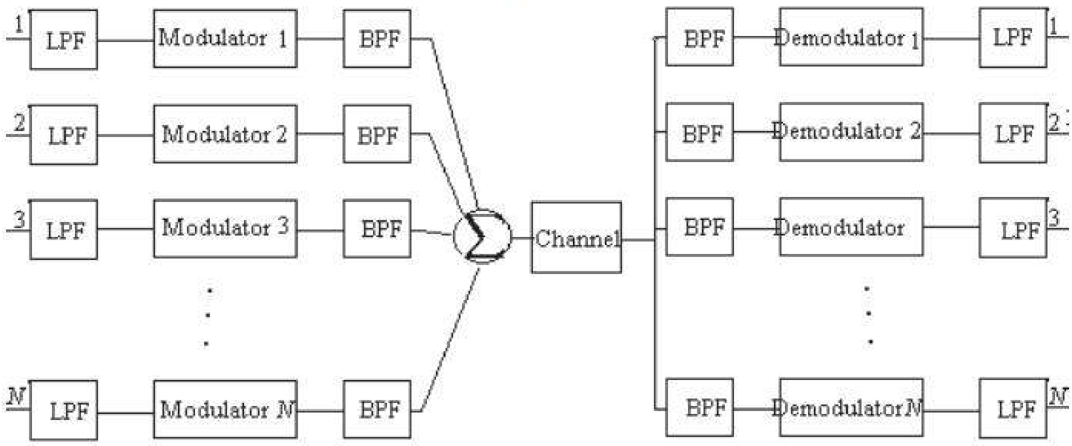


Fig. 6.4 FDM

If frequency translation does not take place here, then the individual messages whose frequency ranges are the same will be mixed in the channel and they can never be separated then.

The operation of FDM can be illustrated with the following example. Suppose we have a channel with a bandwidth of 200kHz and this band falls within the range of 100kHz to 300kHz. Now, we have a message signal of band width

20KHz. Using FDM technique, we can transmit 10 such messages through this channel simultaneously without mixing.

Now let us do frequency translation using AM modulation(SSB). The first message is modulated with a carrier of frequency of 100kHz, so that one sideband falls within the range 100-120kHz. The second message is modulated with 120kHz carrier so that this message is translated into the range 120-140kHz and so on. The tenth message is modulated with a carrier of 280kHz and the message falls within 280-300kHz range. Now, all the ten messages whose baseband falls in the same range are translated into ten different frequency slots and then transmitted through the channel.

At the receiver side, we can use band pass filters to separate these messages. A band pass filter with a pass band of 100-120kHz will select the first message and BPF of passband of 120-140kHz will select the second message and so on. After each message is separated, the baseband message is obtained after demodulation.

FDM requires large bandwidth than TDM but data can be transmitted fast.

**Know your progress**

*Four voice signals of band 0-15 kHz are to be transmitted through a channel with available frequency slot of 250kHz – 320kHz. Find the suitable carrier frequencies to set up FDM.*

## 6.7 Digital modulation techniques

What do we mean by **digital modulation?** Typically the objective of a digital communication is to transport digital data between two or more nodes. The digital data or binary pulses have a large bandwidth. So a low bandwidth channel like telephone lines cannot carry this digital data without causing large distortion. Also an antenna cannot radiate digital pulses as the signal does not continuously vary. The solution to the problem is to modulate an analog signal (sine wave) to include the digital data in it. In radio communications, this is usually achieved by adjusting a physical characteristic of a sinusoidal carrier, the frequency, phase or amplitude. This is performed in real systems with a modulator at the transmitting end to impose the physical change to the carrier and a demodulator at the receiving end to detect the resultant modulation on reception. In computer communication, a modem does digital modulation and demodulation, in order to transmit data through low bandwidth channels such as telephone lines.



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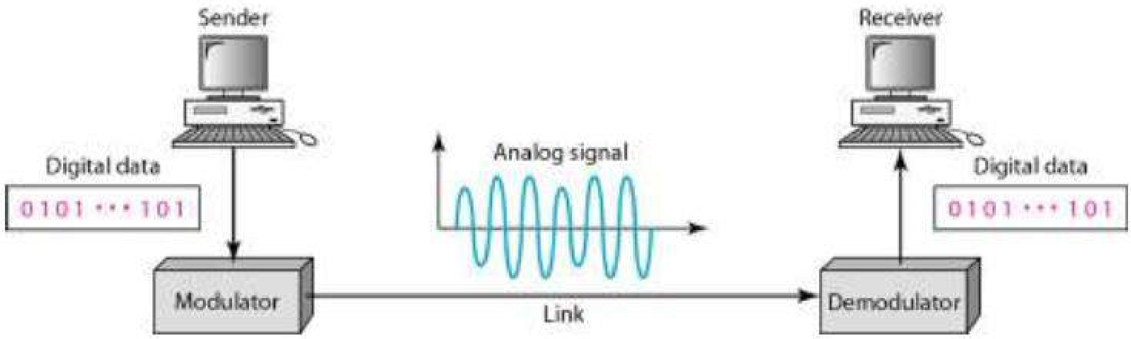


Fig. 6.5 Example of data communication

We begin our discussion by introducing three basic types of digital modulations:

ASK [Amplitude Shift Keying]

FSK [Frequency Shift Keying]

PSK [Phase Shift Keying]

All of these techniques vary the parameter of a sinusoid to represent the information we wish to transmit.

A sinusoid has three parameters that can be varied; they are amplitude, phase and frequency.

## 6.8 Amplitude Shift Keying (ASK)

In ASK the amplitude of a carrier changes with response to the information. Bit '1' is transmitted by a carrier of specific amplitude. To transmit '0' we should change the amplitude to a different level. Here the frequency and phase of the carrier remains constant. Commonly the bit '0' is represented by zero amplitude carrier. An ASK signal can be represented mathematically as

$$s(t) = \begin{cases} A_0 \cos(2\pi f_c t), & \text{binary } 0 \\ A_1 \cos(2\pi f_c t), & \text{binary } 1 \end{cases}$$

If $A_o$ is zero, then the ASK signal for a given baseband data is shown below.

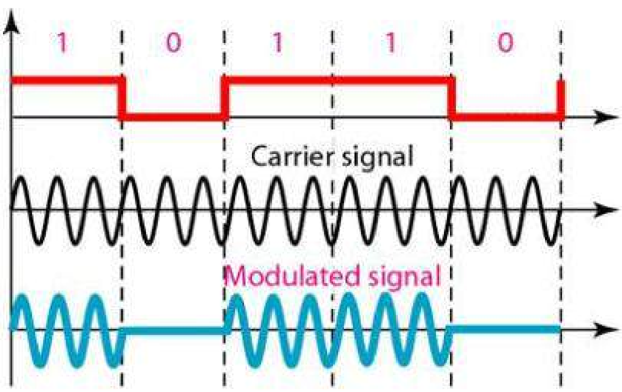


Fig. 6.6 ASK modulation

An ASK signal can be generated by multiplying a carrier signal with the data as shown here.

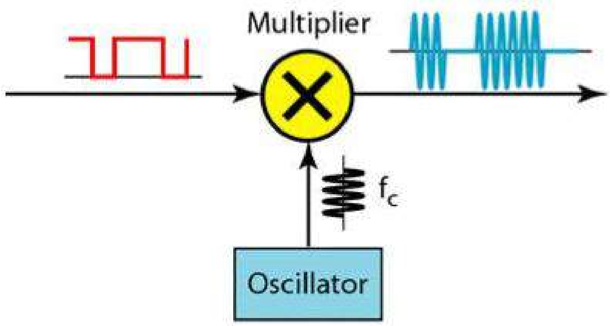


Fig. 6.7 ASK modulator

The demodulation of ASK is very simple. Here the presence or absence of a given sinusoid during a particular interval needs to be determined. If sinusoid is present, data is '1'; otherwise it is '0'. Another advantage of ASK is that both the transmitter and receiver are simple in construction. The main disadvantage of ASK is that the signal is highly susceptible to noise since the data is kept in the amplitude of the signal. Noise usually affects the amplitude of a signal.

ASK is used to transmit data over an optical fiber.

## 6.9 Frequency Shift Keying (FSK)

In **FSK**, we change the frequency in response to information. One particular frequency is selected to represent '1' and another frequency for '0'. Here the amplitude and phase of the carrier remain unchanged. A FSK signal is represented mathematically as

$$s(t) = \begin{cases} A\cos(2\pi f_1 t), & \text{binary } 0 \\ A\cos(2\pi f_2 t), & \text{binary } 1 \end{cases}$$

Here $f_1$ and $f_2$ are different frequencies. A FSK signal is shown below.

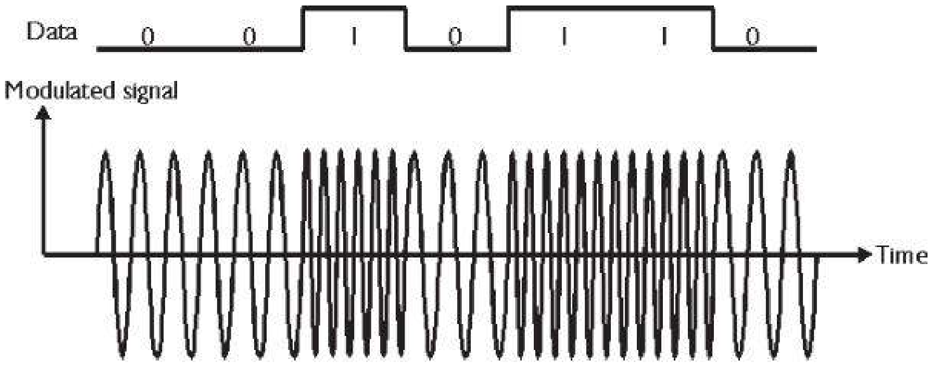


Fig. 6.8 FSK signal

In FSK modulator, two different frequencies are selected to represent bits '0' and '1' and transmitted in the corresponding bit duration. This is shown below.

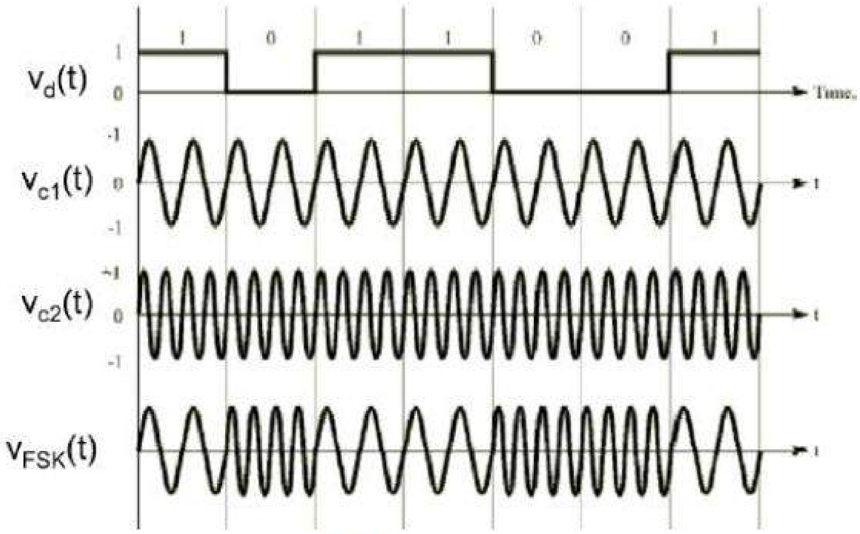


Fig 6.9 FSK generation

The FSK demodulator works on the principle that the two different frequencies which are reaching the receiver at a given interval are distinguished.

The FSK discussed above is known as binary FSK or BFSK because only two different frequencies are used to represent data. Usually, there can be four different frequencies to represent pair of bits 00,01,10,11.

| Frequency | Data |
|---|---|
| $f_0$ | 00 |
| $f_1$ | 01 |
| $f_2$ | 10 |
| $f_3$ | 11 |

Here the advantage is that two bits can be transmitted at a time or speed of data transmission is doubled. But the receiver must be sensitized to distinguish between four different frequencies. This idea can be extended, so that large number of bits can be transmitted at a time. In general 'n' bits can be sent at a time if we use $2^n$ different frequencies. Meanwhile it will add the complexity at the receiver. This is known as M-ary FSK where $M= 2^n$.

FSK is less susceptible to noise or errors compared to ASK as data is kept in frequency of the carrier and not in the amplitude. The main disadvantage of FSK is that it requires large bandwidth which is twice that of ASK. FSK is used in a variety of applications. One example is FAX.

## 6.10 Phase Shift Keying (PSK)

In PSK, we change the phase of the carrier signal to indicate information. Binary '0' is represented by carrier of zero phase and '1' is represented by a carrier of phase $180^0$. The amplitude and frequency of the carrier remains unchanged during the interval of both '0' and '1'. A phase shift of $180^0$ means the signal becomes negative. [ $A\sin(\omega t+180) = -A\sin\omega t$] . So PSK is equivalent to multiplying a carrier with +1 when data is '0' and with -1 when data is '1'. The mathematical representation of PSK signal is given below.

$$s(t) = \begin{cases} A\cos(2\pi f_c t), & \text{binary 1} \\ A\cos(2\pi f_c t + \pi), & \text{binary 0} \end{cases}$$

$$s(t) = \begin{cases} A\cos(2\pi f_c t), & \text{binary 1} \\ -A\cos(2\pi f_c t), & \text{binary 0} \end{cases}$$

The change in phase shift of a PSK signal is shown below.

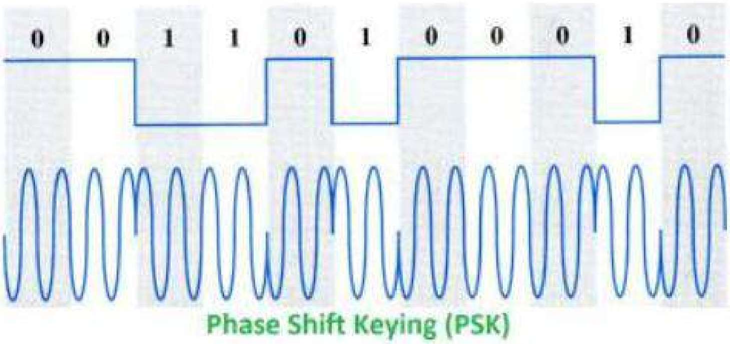


Fig. 6.10 PSK signal

A PSK demodulator must be able to differentiate the phase shift in the carrier to retrieve the data bit. As in the case of FSK, PSK is less susceptible to errors compared to ASK, but PSK requires the same bandwidth as ASK. This makes PSK very popular even though its demodulation is more complex than ASK and FSK. The PSK modulation is shown below.

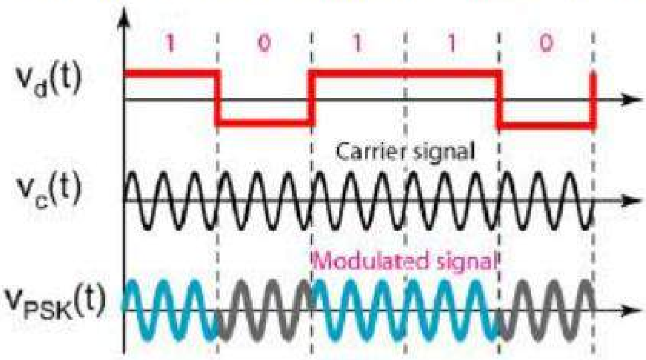


Fig. 6.11 PSK generation

The technique discussed above is known as binary PSK (BPSK) since there are only two different phases.

As in FSK, two bits can be transmitted at a time using carrier with four different phases.

$$s(t) = \begin{cases} A\cos(2\pi f_c t), & \text{binary } 00 \\ A\cos(2\pi f_c t + \frac{\pi}{2}), & \text{binary } 01 \\ A\cos(2\pi f_c t + \pi), & \text{binary } 10 \\ A\cos(2\pi f_c t + \frac{3\pi}{2}), & \text{binary } 11 \end{cases}$$

| Data | Phase |
|---|---|
| 00 | 0 |
| 01 | 90 |
| 10 | 180 |
| 11 | 270 |

Here, high data rate is possible and it is called quadrature PSK (QPSK) and this method can be extended to transmit n bits together, so that there should be $2^n$ different phases for the carrier. This is called M-ary PSK where $M=2^n$. But the main difficulty here is that the receiver must be able to distinguish small phase differences.

## Let us consolidate

Modern communication uses digital transmission and reception almost everywhere. It helps to reduce the effect of noise and enables storage of data. It also makes processing of data possible. To achieve digital transmission A/D conversion is done at the transmitter and D/A conversion at the receiver. The first step in the process of A/D conversion is sampling in which voltage samples of the continuous message signal is taken at regular intervals. According to sampling theorem, the samples should be taken at a rate greater than twice the highest frequency of the signal contained in the message. PCM is a digital data transmission scheme which involves sampling, quantising and encoding. From the transmitted PCM signal, the original message is retrieved at the receiver with minimum distortion. Multiplexing helps to reduce the number of physical channels between the transmitter and receiver when there are a number of sources and destinations existing. The two basic multiplexing schemes are TDM and FDM. In TDM, time is divided into fixed interval slots and data from different sources are transmitted during successive slots. But in FDM, the bandwidth of channel is divided into slots and each slot is allocated to different users and all users are transmitting their data continuously. The digital data cannot be transmitted as such using antennas or with low bandwidth channels. In such cases we use digital modulation schemes such as ASK, FSK, PSK etc. In ASK, amplitude of an analog carrier is shifted between two levels to denote '1' and '0'. In FSK, frequency of the carrier is shifted between two values to represent '1' and '0'. In PSK, the phase of the analog signal is changed between two values.

The content of this unit were learned through general discussion, experimentation, drawing, circuit diagrams and wave forms.

## Let us asses

1. The samples of a signal can represent the continuous signal.
   a) What is the minimum rate of sampling?
   b) How can we reconstruct the original signal from its samples?
2. Write the disadvantages of PAM.
3. Quantization is unavoidable in PCM.
   a) Write the need of quantization.
   b) What is the problem with quantization?
4. Multiplexing allows the transmission of different data through a single channel.
   a) What are the two widely used multiplexing schemes?
   b) Which scheme requires large bandwidth for the channel?
   c) Which factor restricts the number of messages which can be time division multiplexed?
   d) Which factor restricts the number of messages that can be frequency division multiplexed?
5. Write the differences between an AM signal and ASK signal?
6. Discuss the advantage of FSK over ASK.
7. Find out the different phases in PSK when three bits are to be transmitted simultaneously.

# 7 OPTICAL FIBER AND SATELLITE COMMUNICATION

## Significant Learning Outcomes

*After completing this chapter the learner:*

- Points out the advantages of optical fiber communication system
- Classifies optical fibers
- Explains the structure of optical fiber cable and light propagation in optical fibers
- Explains the advantages of optical fiber communication
- Differentiates various types of fiber optic cables
- Describes the light phenomena like reflection and refraction
- Describes the features of different light sources used in optical fiber communication
- Describes the block diagram of an optical communication system
- Explains light detection in optical fibers
- Explains the principles of satellite communication
- Calculates the orbital velocity and time period of a satellite
- Derives the relationship between orbital radius and time period
- Points out various satellites launched by India.

So far we have learned various communication technologies. As we know, nowadays, enormous data are being transmitted and received through different communication media. This is possible due to the increased data rate in the communication system and it wouldn't have become possible if the communication channel were only conventional electrical wires and wireless communication links. Now the most commonly used communication channel is optical fiber cable. The field of fiber optics communications has exploded over the past two decades. Optical fibres are the backbone of telephone communication in our country. There are many desirable properties of optical fibers for carrying information. They have enormous information-carrying capacity, are of low cost, and possess immunity from the many disturbances that can affect electrical wires and wireless communication links. The superiority of optical fibers for carrying information from place to place is leading them to replacing older technologies rapidly. Fiber is an integral part of modern day communication infrastructure and can be found along roads, in buildings, hospitals and machinery. The fiber itself is a strand of silica based glass, its dimensions are similar to those of a human hair, surrounded by a transparent cladding. Light can be transmitted along the fiber over great distances at a very high data rate providing an ideal medium for the transport of information.

Since the beginning of the long distance telephone network, there has been a need to connect the telecommunication networks of one country to another. This has been accomplished in several ways. Submarine cables have been used most frequently. However, there are many occasions, where need to establish a satellite based link to connect to transoceanic points, geographically remote areas or poor countries that have little communication infrastructure. Groups like the international satellite consortium (Intelsat) have fulfilled much of the world's need for this type of service. Satellites are able to provide communications in many instances, where other forms of communications technology may not provide a feasible alternative.

In this unit we will discuss the fundamentals of optical fiber and satellite communication technologies.

## 7.1 Basic Fiber Optic Communication System

Optical fiber is a medium for carrying information from one point to another in the form of light. A basic fiber optic system consists of a transmitting device that converts an electrical signal into a light signal, an optical fiber cable that carries the light, and a receiver that accepts the light signal and converts it back into an electrical signal. The complexity of a fiber optic system can range from very simple network (i.e., local area network) to an extremely sophisticated and expensive system (i.e., long distance telephone or cable television network). A simple system is shown in Figure 7.1. It can be built very inexpensively using a visible LED, plastic fiber, a silicon photodetector, and some simple electronic circuitry. The basic question is 'how much information is to be sent and how far it has to go?' With this in mind we will examine the various components that make up a fiber optic communication system and the considerations that must be taken into account in the design of such systems.

We have already discussed the block diagram of a simple communication system in chapter 5. The difference here is that the channel is optical fiber.

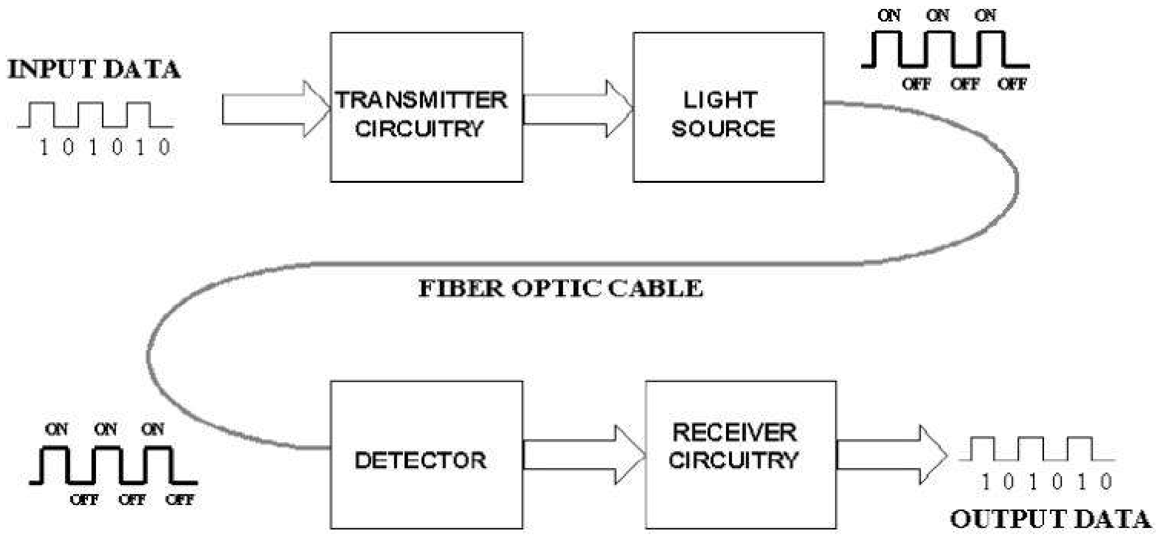


Fig. 7.1 Basic optical fiber communication system

Optical fiber can carry the signal only if it is in the form of light. The transmitter circuitry includes transducer, amplifiers, modulator etc. Then the electrical signal is converted to light with help of a light source. It can be a LED or LASER diode. At the other end the light signal is converted in to electrical form using a detector. This detector can be a photodiode or a photo transistor. The receiver circuitry includes amplifier, demodulator and transducer.

## 7.2 Advantages of optical fiber communication

1. *Long transmission distance*. Optical fibers have lower transmission losses compared to copper wires. This means that data can be sent over longer distances, with minimum the number of intermediate repeaters needed for these spans. This reduction in equipment and components decreases the system cost and complexity.

2. *Large information capacity*. Optical fibers have wider bandwidths than copper wires, which means that more information can be sent over a single physical line. This property results in decrease in the number of physical lines needed for sending a certain amount of information. The large bandwidth of optical fibers enables us to send large amount of data at a time. It means that we can transmit huge amount of data in a short time.

3. *Small size and low weight*. The low weight and the small dimensions of fiber offer a distinct advantage over heavy, bulky wire cables in crowded

underground city ducts or in ceiling-mounted cable trays. This is also of importance in aircraft, satellites, and ships, where small, lightweight cables are advantageous and in tactical military applications, where large amounts of cable must be unreeled and retrieved rapidly.

4. *Immunity to electrical interference*. A specially important feature of optical fibers is that they consist of dielectric materials, which means they do not conduct electricity. This makes optical fibers immune to the electromagnetic interference effects seen in copper wires, such as inductive pickup from other adjacent signal-carrying wires or coupling of electrical noise into the line from any type of nearby equipment.

5. *Enhanced safety*. Optical fibers do not have the problems of sparks, and potentially high voltages inherent in copper lines. However, precautions with respect to laser light emissions need to be observed to prevent possible eye damage.

6. *Increased signal security*. An optical fiber offers a high degree of data security, since the optical signal is well confined within the fiber and any signal emissions are absorbed by an opaque coating around the fiber. This is in contrast to copper wires, where electric signals often can be tapped off easily. This makes fibers attractive in applications, where information security is important, such as in financial, legal, government, and military systems.

**Know your progress**

1. Draw the block diagram of optical fiber communication system and explain the function of each block.
2. Compare optical fiber transmission with conventional cable system of communication.

## 7.3 Optical fiber - structure

To understand the process of making a fiber, consider the schematic of a typical fiber structure, shown in Fig. 7.2. A fiber consists of a solid glass cylinder called the *core*. This is surrounded by a dielectric *cladding*, which has a different material property from that of the core. Surrounding these two layers is a polymer buffer coating that protects the fiber from mechanical and environmental effects.

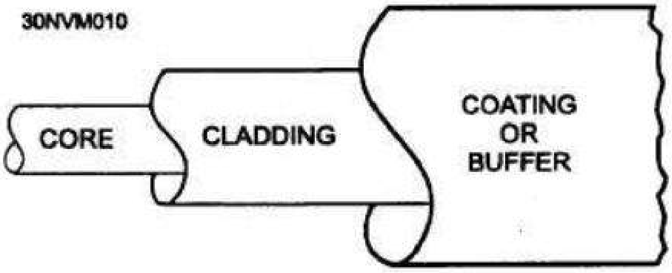


Fig. 7.2 A typical fiber structure

Core and cladding have different refractive indices, with the core having a refractive index, $n1$, which is slightly higher than that of the cladding, $n2$. It is this difference in refractive indices that enables the fiber to guide the light. Because of this guiding property, the fiber is also referred to as an "optical wave guide."

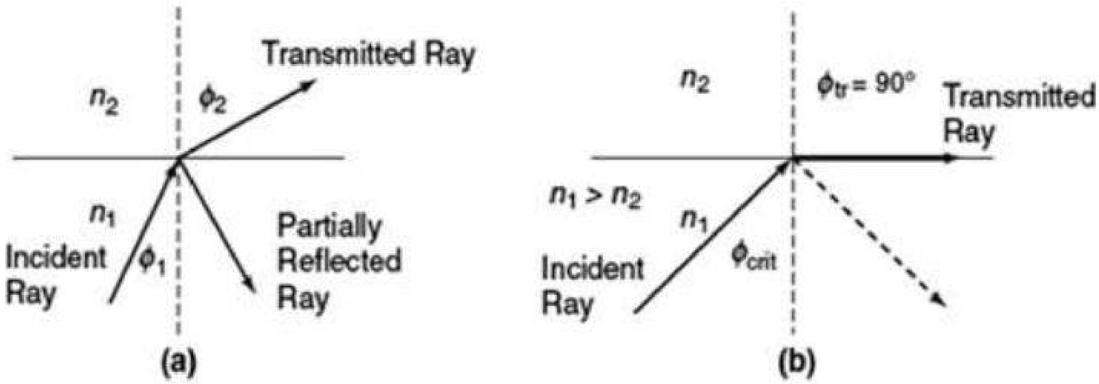


Fig. 7.3 Snell's law

The basics of light propagation can be discussed with the use of geometric optics. The basic law of light guidance is the Snell's law (Figure 7.3). Consider two dielectric media with different refractive indices $n_1$ and $n_2$ such that $n_1 > n_2$ and suppose they are in perfect contact. At the interface between the two dielectrics, the incident and refracted rays satisfy Snell's law of refraction, ie.

$$n_1 \sin \phi_1 = n_2 \sin \phi_2$$

Or

$$\frac{\sin \phi_1}{\sin \phi_2} = \frac{n_2}{n_1}$$

At a particular incident angle $\phi_1$, angle $\phi_2 = 90^o$ and the ray will pass through the core cladding interface. This incident angle is known as critical angle ($\phi_c$).

$$\sin \phi_c = \frac{n_2}{n_1}$$

### Total Internal Reflection (TIR)

When the incident angle is increased beyond the critical angle, the light ray does not pass through the interface but is reflected back. This gives the effect of mirror at the interface with no possibility of light escaping outside the medium. If the angle of the incident ray is greater than the critical angle, the ray is reflected back into the medium with refractive index $n_1$. This process is called total internal reflection. This basic idea can be used to propagate a light ray in a structure with $n_1 > n_2$, and Figure 7.4 illustrates this idea.

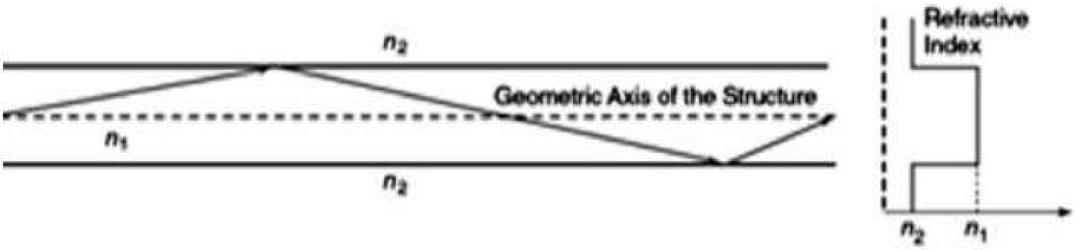


Fig. 7.4:Light guidance using Snell's law

The two conditions necessary for total internal reflection to occur are:

1. The refractive index of the first medium must be greater than the refractive index of the second one.
2. The angle of incidence must be greater than (or equal to) the critical angle.

**Know your progress**

1. Why is total internal reflection useful for signal transmission through fiber optic cables?
2. What is the relation between the ratio of refractive indices and critical angle? Obtain this relation from snell's law. Derive.

## 7.4 Types of fibers

Three basic types of fiber optic cable used in communication systems are:

1. Step-index multimode
2. Step-index single mode
3. Graded-index

This is illustrated in the figure shown below.

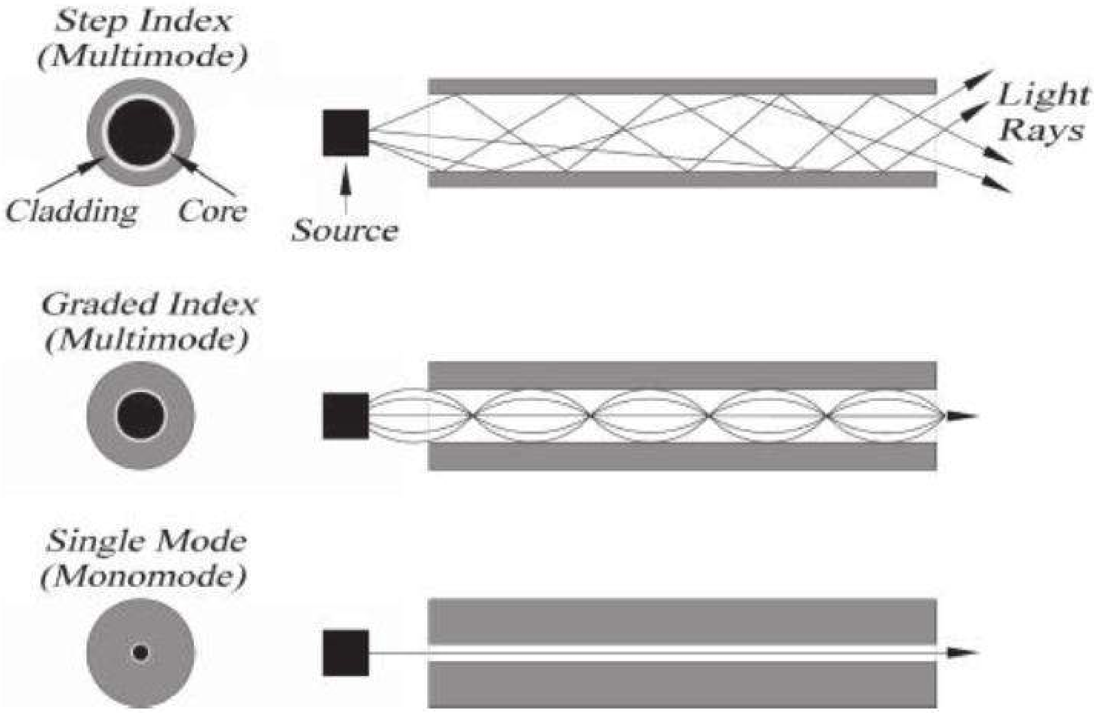


Fig. 7.5 Types of fibers

Single mode fiber has a very small core causing light to travel in a straight line and has a typical core size of 8 or 10 microns. It has unlimited bandwidth and the signal can go unrepeated for over 80 km, depending on the type of transmitting equipment. Single mode fiber has enormous information capacity, more than that of a multimode fiber.

A multimode fiber supports multiple paths of light and has a much larger core and has a core size of 50 or 62.5 microns. The light travels through a much larger path in a multimode fiber, allowing the light to go through several paths or modes.

A multimode fiber can be manufactured in two ways: In step-index form or graded index form. A step-index fiber undergoes an abrupt change or step between the index of refraction of the core and the index of refraction of the cladding.Multimode step-index fibers have lower bandwidth than the other two fibers mentioned.

Graded index fiber was designed to reduce modal dispersion inherent in step index fiber. Modal dispersion occurs as light pulses travel through the core along higher and lower order modes. Graded index fiber is made up of multiple layers with the highest index of refraction at the core. Each succeeding layer has a gradually decreasing index of refraction as the layers move away from the center. High order modes enter the outer layers of the cladding and are reflected back towards the core. Multimode graded index fibers have less attenuation (loss) of the output pulse and have higher bandwidth than multimode step-index fibers.

## 7.5 Dispersion

Dispersion is the spreading of pulses in an opticalfiber. As a pulse of light propagates through a fiber, elements such as numerical aperture, core diameter, refractive index profile, wavelength, and laser linewidth cause the pulse to broaden.This causes a limitation on the overall bandwidth of the fiber as demonstrated in Figure 7.6.

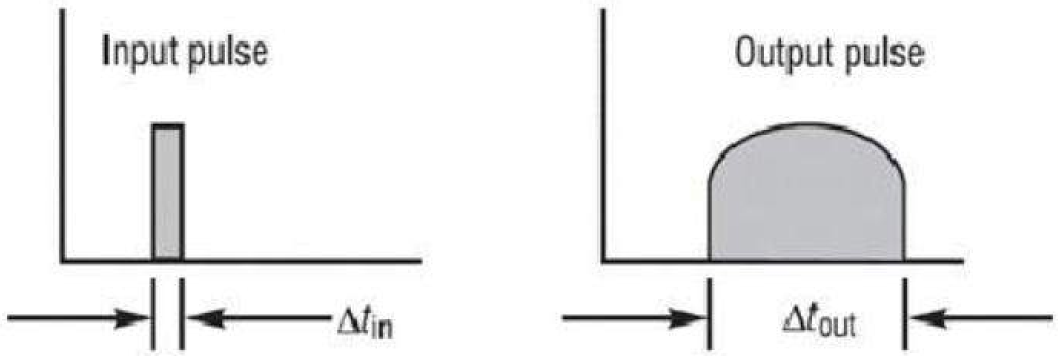


Fig. 7.6 Pulse broadening caused by dispersion

The overall effect of dispersion on the performance of a fiber optic system is known as *Intersymbol interference* (Figure 7.7). Intersymbol interference

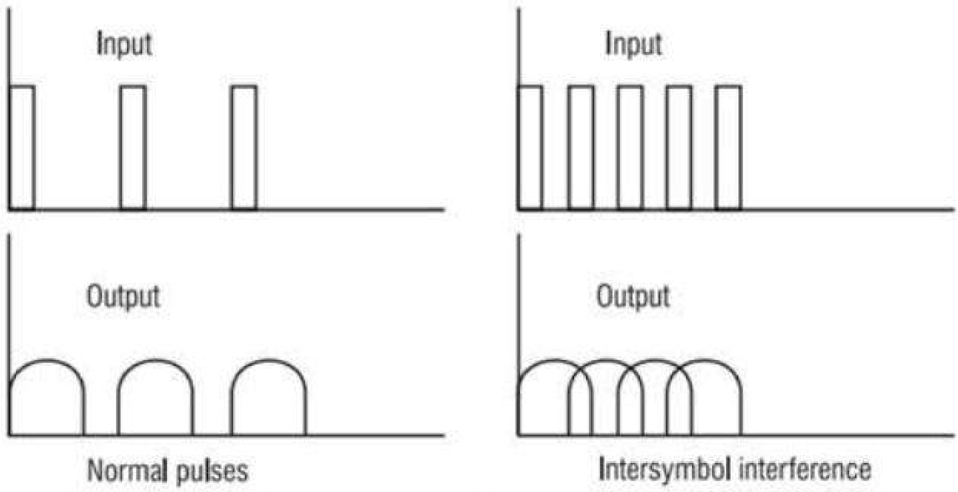


Fig. 7.7Intersymbol interference

occurs when the pulse spreading caused by dispersion causes the output pulses of a system to overlap, rendering them undetectable. If an input pulse is caused to spread such that the rate of change of the input exceeds the dispersion limit of the fiber, the output data will become unidentifiable.

**Know your progress**

1. What are the features of a single mode fiber?
2. Dispersion causes reduction in overall bandwidth of the fiber. Do you agree with this? Justify.

## 7.6 Light sources

When you go to a large hardware store, you will see dozens of different lights- bulbs ranging in size and power from flashlight to floodlights. A similar situation holds for light sources used in optical communications. In this case the sources are much smaller, but they also range from simple, inexpensive light-emitting diodes (LEDs) to costly, high-power laser diodes with complex semiconductor structures.

### LEDs

If we look around us, we notice LEDs everywhere. They can be seen glowing in green, yellow, or red in vehicles, computer equipment, kitchen appliances, telephones, cameras, and in every piece of electronic equipment. They are inexpensive and highly reliable light sources. The LEDs used in optical communications are much smaller and emit in the infrared region, but compared to the other telecommunication light sources used, they are much less expensive and easier to use in transmitter designs. However, because of their relatively low power output, broad emission pattern, and slow turn-on time, their use is limited to low-speed (less than 200-Mbps), short-distance (up to a few kilometers) applications using multimode fibres. Structure and working of LEDs were already discussed in our first year text book.

### Laser Diodes

Semiconductor-based laser diodes are the most widely used optical sources in fiber communication systems. The four main laser types are the Fabry-Perot (FP) laser, the distributed-feedback (DFB) laser, tunable laser, and the vertical cavity surface-emitting laser (VCSEL). Key properties of these lasers include high optical output powers (greater than 1mW), narrow linewidths (a fraction of a nanometer, except for the FP laser), and highly directional output beams for efficient coupling of light into fiber cores.

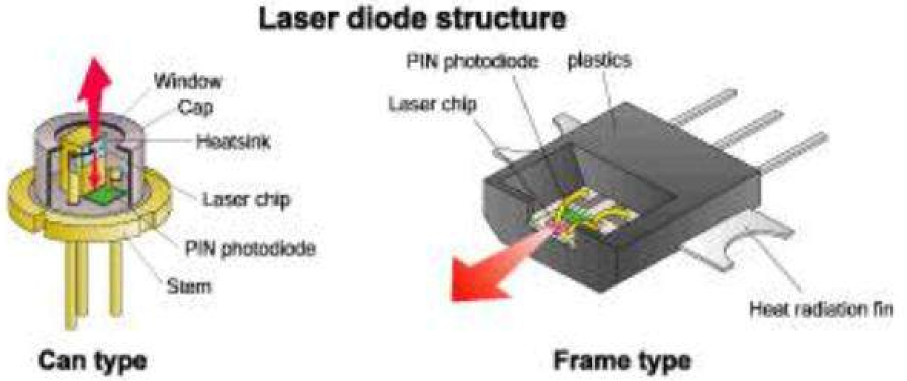


Fig. 7.8: Laser diode

The term laser actually is an acronym for the phrase 'light amplification by stimulated emission of radiation.' So what is stimulated emission of radiation and how does it result in light amplification? First, let us look at the possible photon emission and absorption processes for a two-level atomic system. An electron can move from an energy level E1 in the valence band (called the ground state) to a higher state (called an excited state) at the energy level E2 in the conduction band either by being pumped externally to that level or by absorbing the energy h1 from a passing photon. The latter process is called stimulated absorption. In either case, an excited electron then can return to the ground state either spontaneously or by stimulation, thereby emitting a photon. The corresponding photon generation processes are called spontaneous emission and stimulated emission respectively. Spontaneous emission occurs randomly 'at the will' of the excited electron. Consequently, spontaneously generated photons have random phases and frequencies (or equivalently, random wavelengths). Therefore, this type of light has a broad spectral width and is called incoherent.

When an incoming photon of specific frequency interacts with an excited electron, stimulated emission occurs and it causes an excited electron to drop to the ground state. The photon emitted in this process has the same energy (i.e., the same wavelength) as the incident photon and is in phase with it. Here both the amplitudes add to produce a brighter light. Thus this type of light is called coherent light. Under normal conditions the number of excited electrons is very small, so that the stimulated emission is essentially negligible. For laser action through stimulated emission to occur there must be a population inversion of carriers. The term population inversion simply means that there are more electrons in an excited state than in the ground state. Since this is not a normal condition, population inversion is achieved by supplying additional external energy to pump electrons to a higher energy level. The 'pumping' techniques can be optical or electrical. Laser action normally takes place within a region called the gain medium or laser cavity. To achieve lasing action within this region, the photon density needs to be built up, so that the stimulated emission rate becomes higher than the rate at which photons are absorbed by the semiconductor material. A variety of mechanisms can be used either at the ends or within the cavity to reflect most of the photons back and forth through the gain medium. While passing each time through the cavity, the photons stimulate more excited electrons to drop to the ground state, thereby emitting more photons of the same wavelength. This process thus builds up the photon density in the gain region.

## 7.7 Light Detectors

As we know, in order to detect the signal there must be a receiver at the end of an optical transmission link. The function of this device is to interpret the information contained in the optical signal. An optical receiver consists of a photodetector and various other associated electronic circuits as per the requirements of signal processing.

### The PIN Photodiode

As a detector, the PIN photo diode has several advantages over the p-n photodiode. The working of a general photodiode was discussed the previous year. The diode is operated in reverse bias. As light falls on the pn junction carriers are generated and there by the reverse saturation current is increased.The increase in current is proportional to the amount of light falling on the junction. A p-i-n diode is a p-n junction with an intrinsic (usually lightly doped) layer sandwiched between the p and n layers. It may be operated under the variety of bias conditions discussed in the preceding section. The energy-band diagram, charge distribution, and electric field distribution for a reverse-biased p-i-n diode are illustrated in Fig. 7.9. This structure serves to extend the width of the region supporting an electric field, in effect widening the depletion layer.

Photodiodes with the p-i-n structure offer the following advantages:

1. As the width of the depletion layer is increased (where the generated carriers can be transported by drift) it increases the area available for capturing light.
2. As the width of the depletion layer is increased it reduces the junction capacitance thereby reducing the RC time constant. On the other hand, the transit time increases with the width of the depletion layer. It makes this suitable for high frequency application.

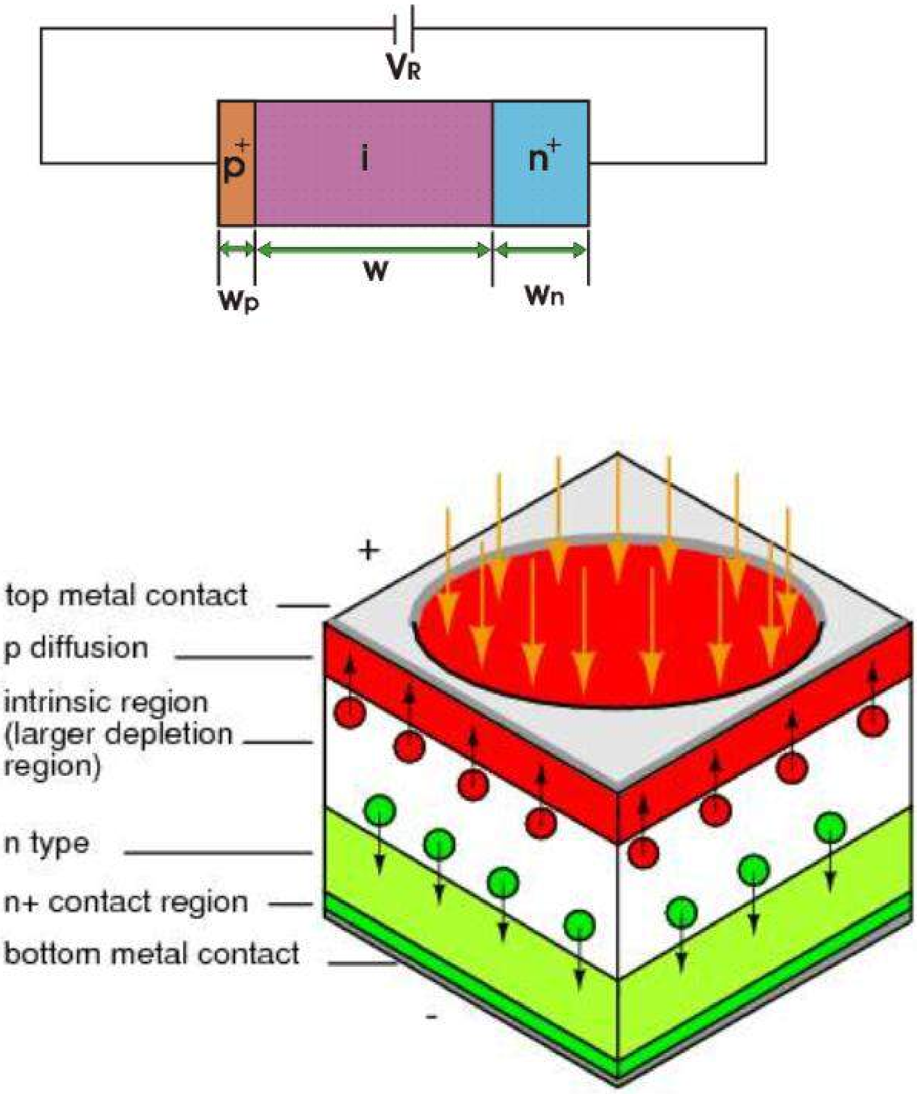


Fig. 7.9 PIN photodiode

## Avalanche photodiode

An avalanche photodiode is a semiconductor based photodetector (photodiode) which is operated with a relatively high reverse voltage (typically tens or even hundreds of volts).

These avalanche photodiodes (APDs) are silicon photodiodes with an internal gain mechanism. As with a conventional photodiode, absorption of incident photons creates electron-hole pairs. A high reverse bias voltage creates a strong internal electric field, which accelerates the electrons through the silicon crystal lattice and produces secondary electrons by impact ionization(avalanche effect). By applying a high reverse bias voltage (typically 100-200 V)APDs show an internal current gain effect. The resulting electron avalanche can produce gain factors up to several hundreds.

Advantages of avalanche photo diode are mainly

- High Sensitivity and Low Noise
- High-Speed Response

Si APDs are often used in high-speed applications since the excess noise from the avalanche process is still lower than the noise that would be generated in connecting an external amplifier to a conventional photodiode operated at high frequencies. Typical applications include low-light level measurement, spectroscopy, range finding and spatial/fiber optic communication.

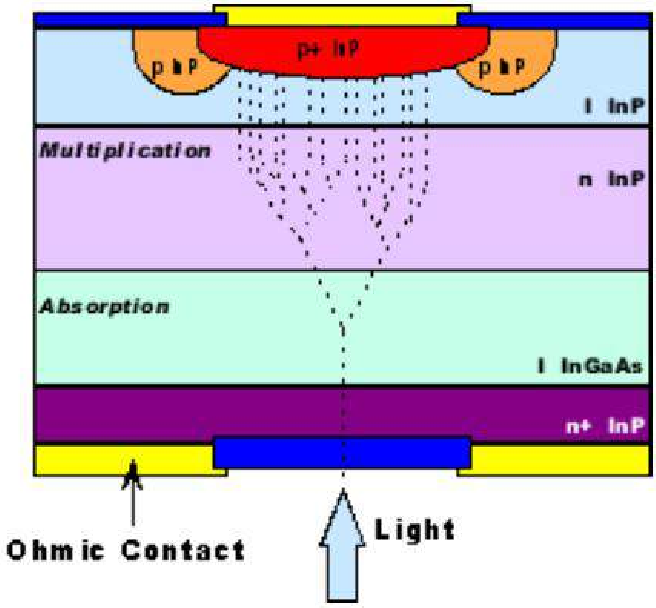


Fig. 7.10 : Avalanche photodiode

**Know your progress**

1. Name the light sources used for optical fiber transmission.
2. What are the advantages of avalanche photo diode?

## 7.8 Satellite Communication

Satellites are are used for a number of puposes. One of the major roles of satellite is communication. Here, the satellite enables communication to be established over large distances - well beyond the line of sight. Communication satellites may be used for many applications, including relaying telephone calls, providing communication to remote areas of the earth, providing satellite communication to ships, aircraft and other mobile vehicles, and there are many more ways in which communication satellites can be used.

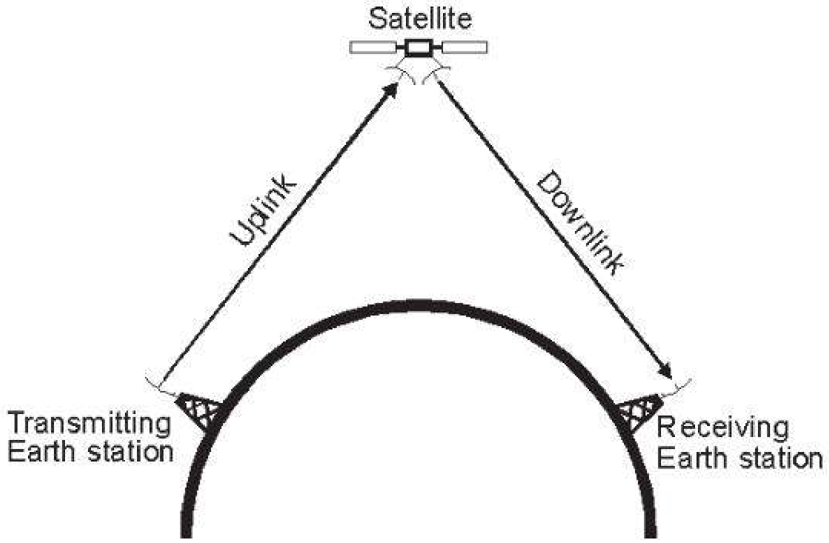


Fig. 7.11 Satellite communication

**The detials of some of the Satellite launched by India were given below:**

| Satellite | Launch Date | Remarks |
|---|---|---|
| Aryabhatta | 19.05.1975 | Active technological experience in building and operating a satellite system. |
| Bhaskara-I | 07.06.1979 | First experimental remote sensing satellite. Carried TV and microwave cameras. |
| INSAT-1A | 10 .04.1982 | First operational multipurpose communication and meteorology satellite. |

| | | |
|---|---|---|
| INSAT-1B | 30.08.1983 | Identical to INSAT-1A. Served for more than design life of seven years. |
| IRS-1A | 17.03.1988 | Earth observation satellite. First operational remote sensing satellite. |
| INSAT-1D | 12.06.1990 | Identical to INSAT-1A. Still in service. |
| INSAT-2C | 07.12.1995 | Has additional capabilities such as mobile satellite service, business communication and television outreach beyond Indian boundaries. Still in service. |
| Kalpana-1 (METSAT) | 12.09.2002 | First meteorological satellite built by ISRO. Originally named METSAT. Renamed after Kalpana Chawla who perished in the Space Shuttle Columbia. |
| EDUSAT | 20.10.2004 | Also designated GSAT-3. India's first exclusive educational satellite. |
| CARTOSAT-1 | 05.05.2005 | Earth observation satellite. Provides stereographic in-orbit images with a 2.5-meter resolution. |
| Chandrayaan-1 | 22.10.2008 | Unmanned lunar probe. Carries 11 scientific instruments built in India, USA, UK, Germany, Sweden and Bulgaria. |
| RISAT-2 | 20.04.2009 | Radar imaging satellite used to monitor India's borders and as part of anti-infiltration and anti-terrorist operations. |
| INSAT-3D | 26.07.2013 | INSAT-3D is the meteorological Satellite with advanced weather monitoring payloads. |
| Mars Orbiter Mission (MOM) | 05.11.2013 | The Mars Orbiter Mission (MOM), informally called Mangalyaan is India's first Mars orbiter. |
| IRNSS-1C[9] | 10.11.2014 | IRNSS-1C is the third satellite in the Indian Regional Navigation Satellite System (IRNSS). |
| GSAT-16 | 07.12.2014 | GSAT-16 is twenty fourth communication satellite of India configured to carry a total of 48 communication transponders. |
| IRNSS-1D | 07.03.2015 | IRNSS-1D is the fourth satellite in the Indian Regional Navigation Satellite System (IRNSS). |



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## 7.9 Satellite Orbits

Satellites orbit around the earth. Depending on the application, these orbits can be circular or elliptical.The only force on a satellite is the force of gravity. A satellite remains in the orbit because this force of gravity ($F_g$) provides the centripetal force ($F_c$) necessary for it to move in a circle. Of course satellite orbits may be ellipses, but we shall restrict our attention to satellites in circular orbits. This keeps the radius and the speed constant. For a satellite with mass in revolving in a circular orbit of radius r with a velocity v, we have

$$F_g = F_c$$
$$F_g = m\,v^2 / r$$
$$m\,g = m\,v^2 / r$$
$$g = v^2 / r$$
$$v^2 = g\,r$$

The Space Shuttle is an excellent example of a satellite in a low-Earth orbit. The Space Shuttle orbits about 100 km to 200 km above Earth's surface. Earth's radius is about 6 000 km, so this is an increase of only about 2% or 3%. That means, the force of gravity is only about 4% to 6% less than that at the Earth's surface.

Therefore, we can use the approximation that the force of gravity is the same on the orbiting satellite as it is on the Earth's surface. That is,

$$F_g = m\,g$$

We know the radius; to be in a low-Earth orbit, the radius of the orbit must be nearly equal to Earth's radius of about 6 000 km or $6 \times 10^6$ m. g is the acceleration due to gravity at the surface of the Earth, g is about 10 m/s$^2$. Of course, we could use 9.8 m/s$^2$ but our entire calculation is a reasonable approximation so we will use 10 m/s$^2$.Using these data now let us calculate the orbital velocity and orbital period..

$$v^2 = g\,r$$
$$v^2 = (10 \text{ m/s})\,(6 \text{ x } 10^6 \text{ m}) = 6 \times 10^7 \text{ (m/s)}^2$$
$$v = 7.7 \times 10^3 \text{ m / s} \quad \text{which is equal to 28000 km/h}$$

Now let us find the orbital period

If 'D' is the diameter of the earth which is the distance travelled by the satellite for one revolution and T is its time period then, The orbital velocity, $v = D / T$

$T = D / v$

$T = 2\pi r / v$

$T = 2 (3.14) (6\times10^6 \text{ m}) / (7.7\times10^3 \text{ m/s})$

$T = 4\,900 \text{ s} = 81 \text{ min}$

## Geo Synchronous orbit

Geosynchronous orbit (sometimes abbreviated GSO) is an orbit around the Earth with an orbital period of one side real day, intentionally matching the Earth's side real rotation period (approximately 23 hours 56 minutes and 4 seconds).

A geostationary satellite is an earth-orbiting satellite, placed at an altitude of approximately 35,800 kilometers (22,300 miles) directly over the equator, that revolves in the same direction, the earth rotates (west to east). At this altitude, one orbit takes 24 hours, the same length of time as the earth requires to rotate once on its own axis. The term geostationary comes from the fact that such a satellite appears nearly stationary in the sky as seen by a ground-based observer.

Now let us calculate the orbital radius required for such a geostationary satellite. As already discussed, $F_g = F_c = m v^2 / r$

$$\text{i.e., } F_g = m v^2 / r$$

However, we can no longer use an approximation like $F_g = mg$. Now we must use the "real" gravitational force,

The real gravitations force between two objects is directly proportional to the product of their masses and inversely proportional to the square of the distance between them. So if M is the mass of the earth, 'm' is the mass of the satellite and 'G' is the gravitational constant.

$$\begin{aligned} \text{Then } F_g &= G M m / r^2 \\ \text{Since } F_g &= m v^2 / r \\ G M m / r^2 &= m v^2 / r \\ G M / r^2 &= v^2 / r \end{aligned}$$

As said earlier, the orbital velocity v = D/T where D is the diameter of the earth and T is the period of revolution of the Satellite.

$$\text{ie., } v = 2\ r / T$$
$$GM/r^2 = (2\ r/T)^2/r$$
$$GM/r^2 = (4\ ^2 r^2/T^2)/r$$
$$GM/r^2 = 4\ ^2 r^2 / T^2 r$$
$$GM/r^2 = 4\ ^2 r / T^2$$
$$GMT^2/4^2 = r^3$$
$$r^3 = GMT^2/4\ ^2$$
$$r^3 = [GM/4\ ^2]\, T^2$$

Therefore it is clear that $r^3 / T^2$ = constant and this is Kepler's third Law of planetary motion.

**Know your progress**

1. What are the approximate values of orbital velocity and the time period of a satellite?
2. What do you mean by a geo synchronous satellite?

## 7.10 Applications of Satellites

**1. Weather Forecasting**

Certain satellites are specifically designed to monitor the climatic conditions of the earth. They continuously monitor the assigned areas of earth and predict the weather conditions of that region. This is done by taking images of earth from the satellite. These images are transferred using assigned radio frequency to the earth station. (Earth Station: it is a radio station located on the earth and is used for relaying signals from satellites). These satellites are exceptionally useful in predicting disasters like hurricanes, and monitor the changes in the Earth's vegetation, sea state, ocean color, and ice fields.

**2. Radio and TV Broadcast**

These dedicated satellites are responsible for making hundreds of channels across the globe available for everyone. They are also responsible for broadcasting live matches, news, world-wide radio services. These satellites require a 30-40 cm sized dish, to make these channels available globally.



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### 3. Military Satellites

These satellites are often used for gathering intelligence, as a communication satellite used for military purposes, or as a military weapon.

### 4. Navigation Satellites

Navigation satellite is a satellite designed to enable operators of aircraft, vehicles, or vessels to determine their geographical position. The system allows for precise localization world-wide, and with some additional techniques, the precision is in the range of some meters.

### 5. Global Telephone

One of the first applications of satellites for communication was the establishment of international telephone backbones. Instead of using cables it is sometimes faster to launch a new satellite. But, fiber optic cables are still replacing satellite communication across long distance as in fiber optic cable, light is used instead of radio frequency, making the communication much faster. Using satellites, in order to reach a distance approximately 10,000 km away, the signal needs to travel almost 72,000 km, that is, sending data from ground to satellite and (mostly) from satellite to another location on earth. This causes substantial amount of delay and this delay becomes more prominent for users during voice calls.

## Connecting Remote Areas

Due to their geographical location, many places all over the world do not have direct wired connection to the telephone network or the internet (e.g., researchers on Antarctica) or because of the current state of the infrastructure of a country. Here, the satellite provides complete coverage and (generally) atleast one satellite is always present across the horizon.

## Global Mobile Communication

The basic purpose of satellites for mobile communication is to extend the area of coverage. Cellular phone systems, such as GSM (and their successors) do not cover all parts of a country. Areas that are not covered usually have low population, where it is too expensive to install a base station. With the integration of satellite communication, however, the mobile phone can switch to satellites offering world-wide connectivity to a customer. Satellites cover a certain area of the earth. This area is termed as a 'footprint' of that satellite. Within the footprint, communication with that satellite is possible for mobile users. These users communicate using a Mobile-User-Link (MUL). The base-stations communicate with satellites using a Gateway-Link (GWL).



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Sometimes, it becomes necessary for the satellite to create a communication link between users belonging to two different footprints. Here, the satellites send signals to each other and this is done using Inter-Satellite-Link (ISL).

## Frequency allocation for satellites

Allocation of frequencies to satellite services is a complicated process which requires international coordination and planning. This is done as per the International Telecommunication Union (ITU) standards. To implement this frequency planning, the world is divided into three regions:

Region 1 : Europe, Africa and Mongolia

Region 2 : North and South America and Greenland

Region 3 : Asia, Australia and south-west Pacific.

Within these regions, the frequency bands are allocated to various satellite services.

The name and range of major fequency bands used for satellite communication are given below.

| Name of the band | Fequency range (GHz) |
|---|---|
| L | 1-2 |
| S | 2-4 |
| C | 4-8 |
| X | 8-12 |
| Ku | 12-18 |
| K | 18-27 |
| Ka | 27-40 |
| O | 40-50 |
| V | 50-70 |

Among these bands, C, Ku and Ka are the most commonly used.



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## Let us consolidate

Optical fiber is a medium for carrying information from one point to another in the form of light. The transmitter circuitry includes transducer, amplifiers, modulator etc. Then the electrical signal is converted to light with help of a light source. It can be a LED or LASER diode. Optical fibres have many advantages over conventional cables. Long distance communication without loss, large information capacity, small size, low weight, immunity to electrical interference, good safety and increased signal security are some of them. A fiber consists of a core, a cladding and a polymer buffer coating. The principle of transmission of signals through OFC cables is total internal reflection.Step index multimode, step index single mode and graded index multimode fibres are the three basic types of fiber optic cables. The overall effect of dispersion on the performance of a fiber optic system is known as inter symbol interference. The light detectors used for fiber optic communication are pin diode and avalanche diode.

A satellite enables communication over long distances. Communication satellites are used for relaying telephone calls, providing communication to remote areas, ships, air craft and other mobile vehicles. The orbital velocity of a satellite can be calculated to be approximately 28000 km and its period of rotation to be 81 minutes. It can be seen that for a geo synchronous satellite $r^3 = [\,GM/4\ ^2]\ T^2$ or $r^3\ \alpha\ T^2$. The various uses of satellite include weather forecasting, radio and TV broadcasting, as military and navigational satellites, for global telephone communication and global mobile communication.

The contents in this module are to be learned through general discussion, group discussion, data collection, chart preparation and with the help of ICT where ever is applicable.

## Let us asses

1. A) In optical fiber communication electrical signal is converted into light using
   a) Photo diode b) Avalanche photo diode c) LASER diode d) Photo transistor
   B) Draw and explain the basic block diagram of an optical communication system.
2. Explain the structure of an optical fiber cable..
3. A) Spreading of pulses in optical pulses is called
   a) refraction b) dispersion c) reflection d) polarization
   B) What do you mean by inter symbol interference? Explain.
4. A) What are the features of a single mode fiber?
   B) Compare step index and graded index fibers.
5. A) Name two light sources which are used in optical fiber communication system.
   B) Explain their working and compare them.
6. A) Name two detectors used in optical fiber communication
   B) What are the advantages of avalanche photo diode?
7. A) What do you mean by a geo synchronous orbit?
   B) Derive the relation between orbital radius and time period of a geo-synchronous satellite.
8. A) The orbital velocity of a satellite is approximately
   a) 52 km/h b) 52000 km/h c) 28 km/h d) 28000 km/h
   B) Why is a satellite able to continue rotating in its orbit without failure?
9. A) The orbital time period of a satellite is approximately
   a) 81 minutes b) 81 s c) 81 h d) 24 h
   B) Find out the approximate values of orbital velocity and time period of a satellite. Take ‘ g’ as 10 $m/s^2$ and the radius of earth as 6000 km.
10. List the various applications of satellites.

8

# TELEVISION

## Significant Learning Outcomes

*After completing this chapter the learner:*

- explains the history and development of TV systems.
- explains the scanning process of television picture.
- explains the bandwidth of TV transmission.
- draws the block diagram and thereby explains the working of monochrome TV receiver.
- identifies the basics of colour TV.
- explains additive colour mixing.
- explains the reception process of dish antenna.
- points out the uses of co-axial cable.
- explains the arrangement of cable TV network.

Do you know how we see world cup cricket/football match or republic day parade in our Television (TV) set? The television system enables us to sit at home and watch any program, which is taking place several thousand kilometers away. Let us study the basic concept of TV in this chapter.

Television is a telecommunication medium that is used for transmitting and receiving moving images and sound. In ancient Greek, tele means "far" and in Latin, vision means "sight". The history of television starts in the early years of the twentieth century when many scientists experimented with the idea of using selenium photosensitive cells for converting light from pictures to electrical signals and transmitting them through wires.

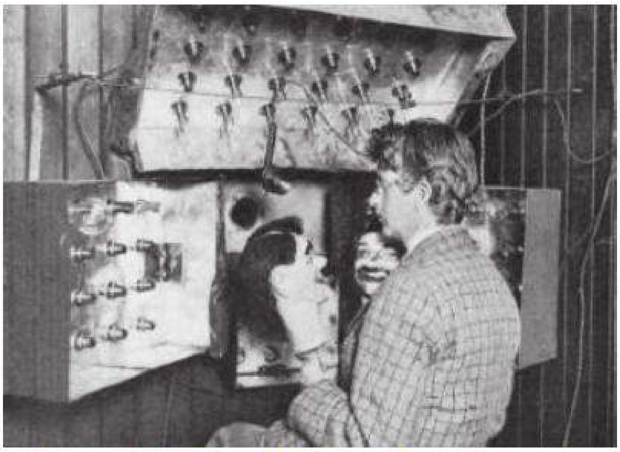

Fig 8.1 Baird in 1925 with his television equipment

The first demonstration of actual television was given by J.L. Baird in UK(see Fig.8.1) and C.F. Jenkins in USA around 1927. Though television broadcast started in 1935, with the end of the Second World War, it rapidly grew into a popular medium for dispersion of news and mass entertainment.

In 1960s colour broadcasting became popular. The replacement of bulky, high-voltage cathode ray tube (CRT) screen displays, with compact, energy-efficient, flat-panel alternatives such as LCDs, plasma displays, and OLED displays were a major hardware revolution in the consumer computer monitor market in the late 1990s, and it soon revolutionalised TV sets also. Internet television was introduced in 2010. In 2013, 87% of televisions sold were with colour LCD screens.

## 8.1 Basic Principles of Television

The television signal consists of two main parts: light and sound components. Video signal creates moving images in the screen. The camera captures the image frame by frame. When these image frames are displayed on the screen at a sufficient rate, we see moving images or video. The image appears to be moving because of persistence of vision of human eye.

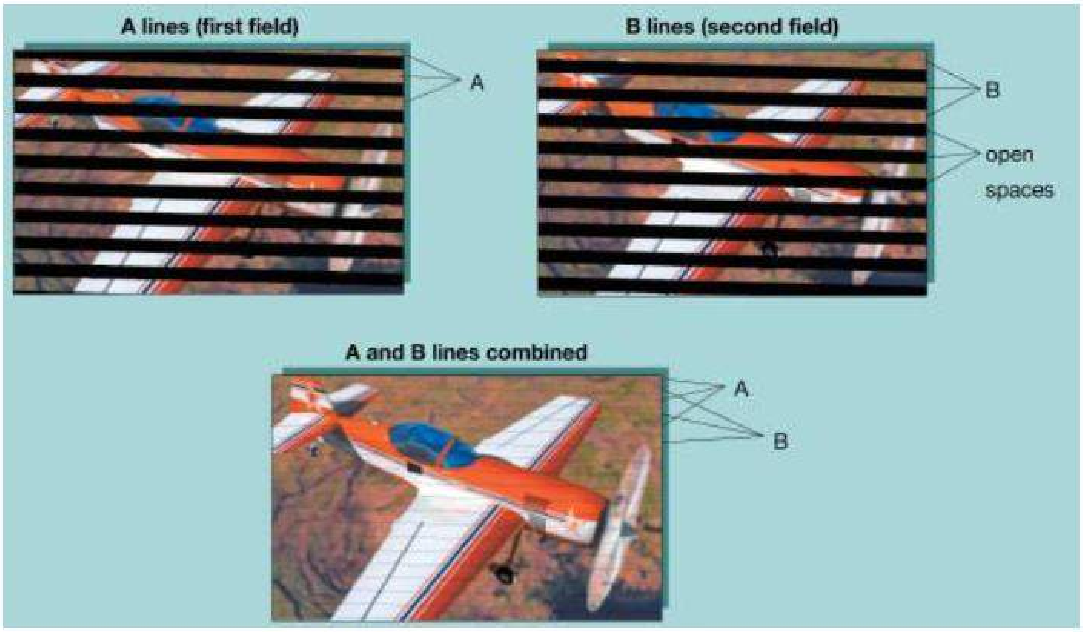


Fig.8.2 Illustration of interlaced scanning

The standard TV scene dimensions are in the ratio 4:3; that is, the scene width is 4 units for every 3 units of height. This width to height ratio is called aspect ratio. To create a picture, the scene is subdivided into many fine horizontal lines called scan lines. The light intensity variation in a picture is captured by the camera by capturing the intensity variation over each scan line. This is similar to reading a text. We scan through each line of the text from left to right (tracing). To read the next line, we move our eye quickly to the right (retracing). As we understand a complete text by combining all the lines we read, a picture is captured and recreated by combining the intensity values of all the scan lines. This process is called scanning. During scanning of each line, the video camera produces the electrical signal corresponding to the light intensity variation of that line or we can say that the variation of voltage of this electrical signal is similar to the variation of light intensity along that line. This electrical signal is called video signal. When all lines of the frame are scanned, the electrical signal of all those lines is generated. This corresponds to the video signal of a frame. Then this process repeats for the next frame and so on. In video camera, when the camera is focussed on an image, the light intensity variations in the image is first converted into electrical resistance variation in a photoconducting plate. Actually, this resistance variation is then converted into video signal by the process of scanning.

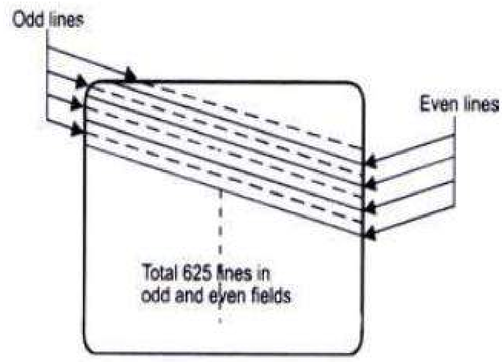

Fig.8.3 Interlaced Scanning

The sound signal or audio signal is captured using a microphone and transmitted in the form of mono or stereo along with the video signal.

## 8.2 Principles of Scanning

For converting a picture to an electrical signal, an electron beam has to explore the picture point by point. The scanning is done at a fast rate creating an illusion of continuity, due to persistence of vision. The complete image is called a frame. Each frame is divided into 625 lines and 25 such frames are scanned per second. The scanning is done line by line from left to right (trace). When one line has been scanned, the beam moves onto the next line. The retrace of the beam from the one end of the line to the beginning of next line is very fast. The greater the number of scan lines, higher is the resolution and more details of the picture can be observed. The task of the TV camera is to convert this scene to an electric signal.

## Interlaced scanning

Although the scanning of 25 frames per second in television pictures is enough to cause an illusion of continuity, they are not rapid enough to allow the brightness of one picture or frame to blend smoothly with the next causing the screen is blanked between successive frames. This results in flickering of light that is very annoying to the observer, when the screen is alternatively bright and dark between successive frames. The flickering can be avoided if the frame rate is increased. But it will result in increased bandwidth of the video signal. The method used in TV to avoid flickering without increasing the bandwidth is interlaced scanning. Before getting in to the details of interlaced scanning process, let us do a simple exercise to understand what scanning process is. Take a picture from the news paper or magazine and cut it horizontally to equal size (0.5 cm). Then number the pieces from the top (1, 2, 3 etc.). Arrange the odd ones and even ones separately. These separated parts are the first and second field of the frame. The scanning is similar to this ( see figure 8.2). In TV, each frame is split into two fields- odd field and even field. The odd field contains odd numbered lines and even field contains even numbered lines. Each field is scanned alternatively ( see figure 8.3).

That means every alternate line gets scanned instead of every successive line. When the beam reaches the bottom of the picture frame, it quickly returns to the top to scan those lines that were missed during the previous scanning. It reduces flicker to an acceptable level since the area of the screen is covered at twice the rate.

In the 625 line monochrome system, for successful interlaced scanning, the 625 lines of each frame or picture are divided into sets of 312.5 lines and each set is scanned alternately to cover the entire picture area. This is illustrated in figure 8.4. Since the first field ends in a half line and the second field commences at the

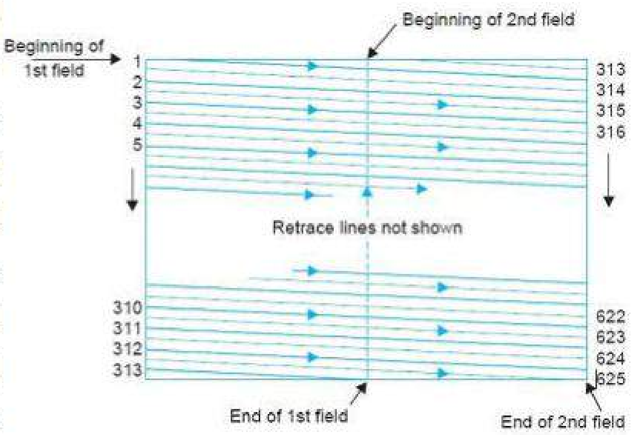


Fig.8.4 The two fields of interlaced scanning



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middle of the line on the top of the target plate or screen (see Fig. 2.4), the beam is able to scan the remaining 312.5 alternate lines during its downward journey. Hence, the beam scans 625 lines ($312.5 \times 2 = 625$) per frame at the same rate of 15625 lines ($312.5 \times 50 = 15625$) per second. Therefore, with interlaced scanning the flicker effect is eliminated without increasing the speed of scanning, which in turn does not need any increase in the channel bandwidth.

## Scanning periods

The wave shapes of both horizontal and vertical sweep signals are shown in Fig. 8.5. As shown, the retrace time involved (both horizontal and vertical) is very less. The time period of the horizontal sweep waveform as shown in Fig. 8.5 (a) is 64μs (1/15625 = 64 μs), out of which the active line period is 52 μs and the remaining 12μs is the line blanking period. The beam returns during this short interval to the extreme left side of the frame to start tracing the next line.

Similarly with the field frequency set at 50 Hz, the time period of the vertical trace (see Fig. 8.5 (b)) is 20 ms (1/50 = 20 ms). Out of this 18.720 ms are spent in bringing the beam from top to bottom and the remaining 1.280 ms is

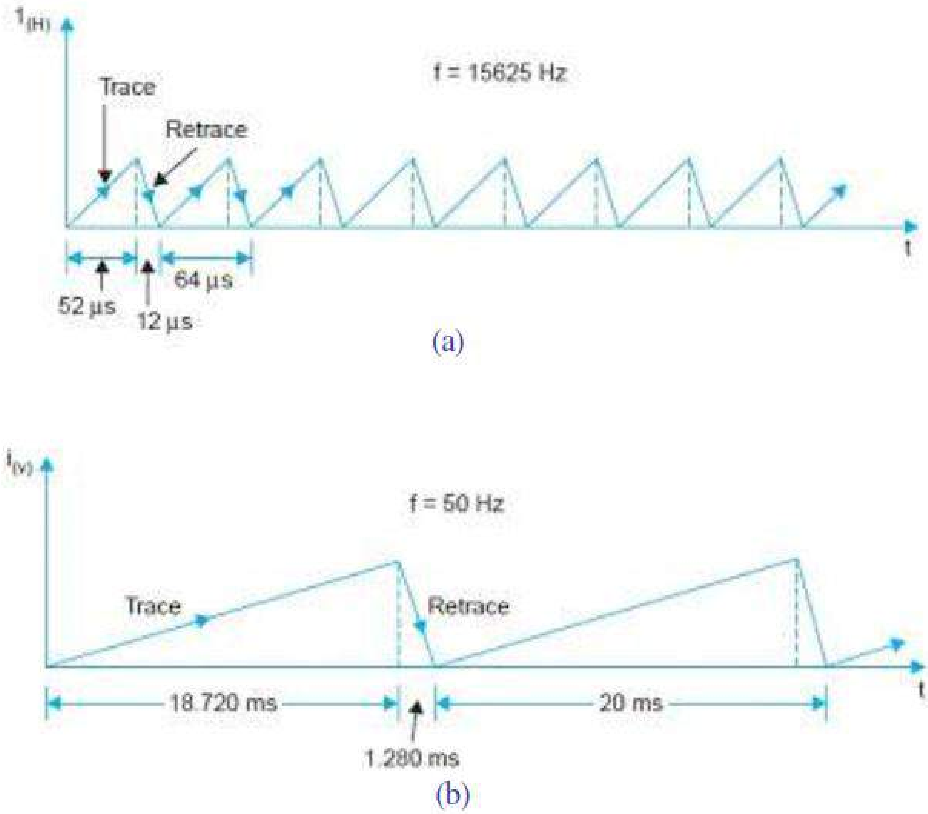


Fig.8.5 Details of horizontal and vertical sweep waveforms

taken by the beam to return to the top to commence the next cycle. Since the horizontal and vertical sweep oscillators operate continuously to achieve the fast sequence of interlaced scanning, 20 horizontal lines (1.280 ms /64 μs =20 lines) get traced during each vertical retrace interval. Thus 40 scanning lines are lost per frame as blanked lines during the retrace interval of two fields. So the active number of lines for scanning the picture details becomes equal to 625 - 40 = 585, instead of the 625 lines actually scanned per frame.

**Know your progress**

1. What do you understand by interlaced scanning? Explain how it reduces flicker.
2. Define the terms aspect ratio, frame and field.

## 8.3 Bandwidth of Television Transmission

In TV, video signal is modulated by vestigeal sideband amplitude modulation (VSB) and sound signal by FM modulation. In this VSB modulation the frequency range of (0-1.25 MHz) of the lower sideband is retained. A sound carrier is always positioned at the extreme end of the upper sideband and hence is 5.5 MHz away from the picture carrier. The 5 MHz video signal will extend up to 5.5 MHz because of the limitation of a practical filter. The FM sound signal occupies a frequency spectrum of about ± 75 kHz around the sound carrier. However, a guard band of 0.25 MHz is allowed on the sound carrier side of the television channel to allow adequate inter-channel separation. The total channel bandwidth thus occupied is 7 MHz. Figure 8.6 shows the complete channel. The frequency axis is scaled relative to the picture carrier, which is marked as 0 MHz. This makes the diagram very informative, as details such as the width of the upper and lower sidebands and the relative position of the sound carrier can easily be read.

Amplitude modulation of a carrier results in a sum and a difference frequency, creating two symmetrical side-bands. The symmetry means that one of the sidebands is redundant, so removing one sideband does not affect the information content of the signal.

VSB is similar to SSB but it retains a small portion (a vestige) of the redundant sideband. This reduces DC distortion since the low frequencies are not affected by filtering one of the sidebands. The video signal in TV requires a bandwidth of 5 MHz.

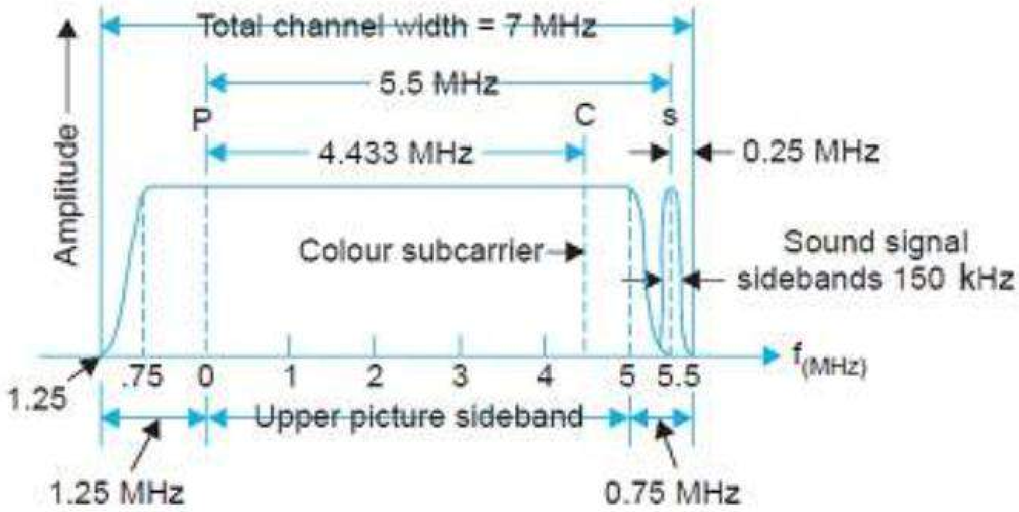


Fig.8.6 The spectrum of a TV channel

**Know your progress**

1. Draw the complete channel bandwidth in Indian TV system.
2. Which modulation scheme is used for TV transmission?

## 8.4 Block Diagram of Monochrome Television Receiver

A simplified block diagram of a black and white TV receiver is shown in Fig. 8.7. The receiving antenna intercepts the radiated RF signals and the tuner selects desired channel's frequency band and converts it to the common intermediate frequency (IF) band . The receiver employs two or three stages of intermediate frequency (IF) amplifiers. The output from the last IF stage is demodulated to recover the video signal and then it is fed into a video amplifier. A part of the video signal is the synchronising pulses. The synchronising pulses are separated from the video signals. These synchronising pulses synchronise the scanning process at the receiver and at the transmitter. The signal that carries picture information is then amplified and coupled to the picture tube which converts the electrical signal back to picture elements.

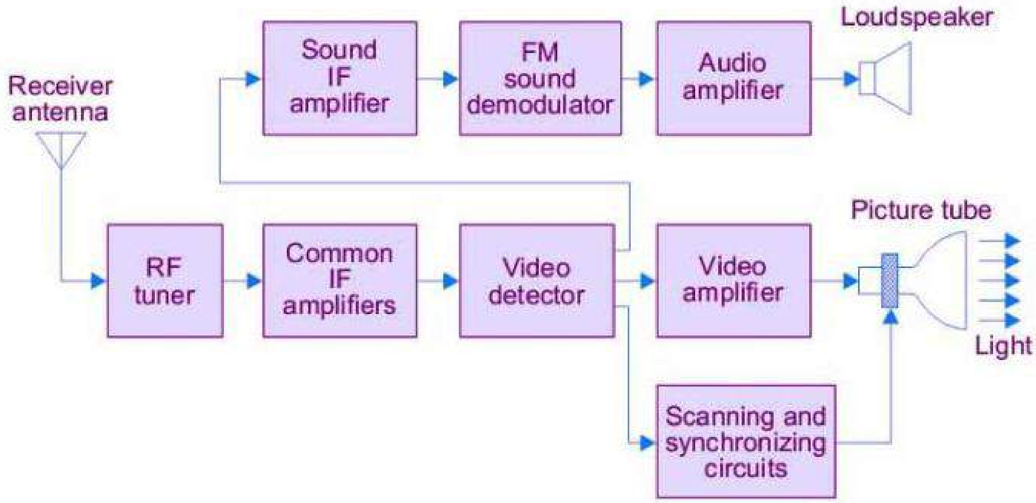


Fig.8.7
Block diagram of monochrome television receiver

### Sound Reception

The path of the sound signal is same as that of a picture signal travelling from antenna to video detector section of the receiver. Here the two signals are separated and fed to the corresponding channels. The frequency modulated audio signal is demodulated after one stage of amplification. The audio output from the FM detector is given due amplification before feeding it to a loudspeaker.

## 8.5 Colour TV Receiver

A colour TV receiver is similar to a monochrome receiver. The main difference is that the colour TV receiver needs a colour subsystem. It receives the colour signal, process it and feeds it to the colour picture tube. A colour picture tube has three electron guns corresponding to three pick-up tubes in the colour camera. The screen of the picture tube has red, green and blue phosphorus dots arranged in alternate stripes. Each gun produces an electron beam to illuminate the corresponding colour phosphor distinctly on the fluorescent screen. The eye then integrates the red, green and blue colour information and their luminance to perceive the actual colour and brightness of the picture being televised. By varying the intensity of the colour beams, the dot triads can be made to produce any colour. The dots are so small that the eye cannot recognize them individually at a distance. What the eye sees is a colour picture swept out on the face of the tube. Fig.8.8 shows how all the signals come together at the picture tube to produce the colour picture. The $R$, $G$ and $B$ signals are mixed with the $Y$ (luminance) signal to control the cathodes of the CRT. Thus the beams are properly modulated to reproduce the colour picture.

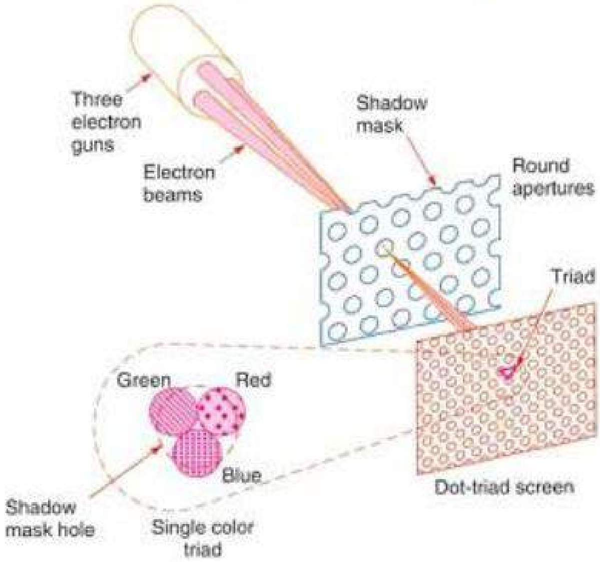


Fig.8.8 Details of colour picture tube.

Computer monitors and televisions use additive colour mixing. CRT, LCD and other types the colour video displays are composed of red, green and blue sub-pixels, the lights from which combine in various proportions to produce all the other colours as well as white and the shades of grey. In additive mixing which forms the basis of a colour television, lights from two or more colours, obtained either from independent sources or through filters, can create a combined sensation of a different colour. Thus different colours are created by mixing pure colours. The additive mixing of three primary colours-red, green and blue in adjustable intensities can create most of the colours seen in everyday life. The impression of white light can also be created by choosing suitable intensities of these colours. Red, green and blue are called primary colours. These are used as basic colours in television. By pair wise additive mixing of the primary colours, the following secondary colours can be produced:

**Red + Green = Yellow**

**Red + Blue = Magenta (purplish red shade)**

**Blue + Green = Cyan (greenish blue shade)**

This is illustrated in Fig. 8.9 where each circle corresponds to one primary colour.

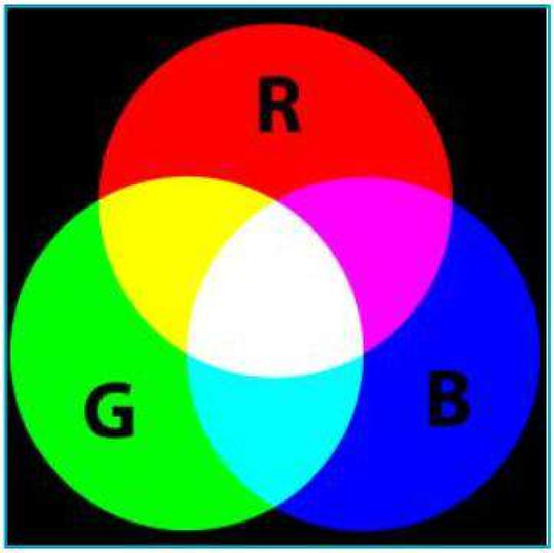


Fig 8.9 Additive colour mixing: adding red and green yields yellow; adding all the three primary colours together yields white.

## Luminance, Hue and Saturation

Take the paint application in a computer and select the colour selection button. Now, select different colour options and thus notice various properties of colour. Refer Fig.8.10.

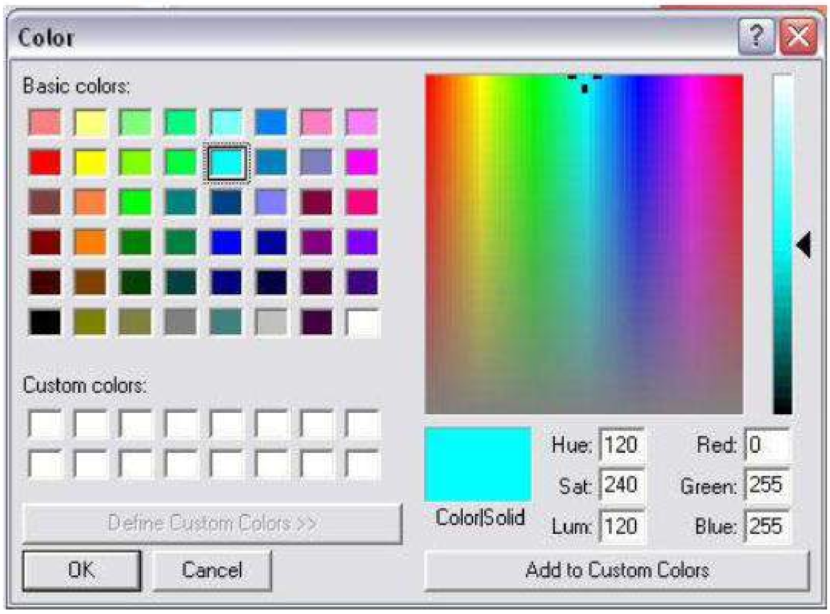


Fig.8.10 Computer screen to show the effect of variance of luminance,hue and saturation

- What do you find in the colour chart?
- What happens to the colour while changing luminance, hue and saturation?

Any colour has three characteristics to specify its visual information. These are (*i*) luminance, (*ii*) hue or tint, and (*iii*) saturation. These are defined as follows:

**Luminance or Brightness**

This is the amount of light intensity as perceived by the eye regardless of colour. In black and white pictures, better lightened parts have more luminance than the dark areas. Different colours also have different shades of luminance in the sense that though equally illuminated they appear with more or less luminance/brightness. Thus on a monochrome TV screen, dark red colour will appear as black, yellow as white and a light blue colour as grey.

**Hue**

This is the predominant spectral colour of the received light. The colour of any object is distinguished by its hue or tint. The green leaves have green hue and red tomatoes have red hue. Different hues result from different wavelengths of spectral radiation and are perceived as such by the sets of cones in the retina.

**Saturation**

This is the spectral purity of the colour light. This indicates the amount of other colours present. Thus saturation may be taken as an indication of how little the colour is diluted by white. A fully saturated colour has no white. As an example, vivid green is fully saturated and when diluted by white it becomes light green. The hue and saturation of a colour put together is known as chrominance. Note that it does not contain the brightness information. Chrominance is also called chroma.

Most applications allow picking colour either by RGB (red, green, blue) values or HSL (hue, saturation and luminance). The ways of scaling can vary with the applications - Paint Shop Pro uses 0 to 255 for all the three qualities, Photoshop has an HSL plug-in, but uses HSB (hue, saturation, brightness) in its colour picker. Hue is expressed in the degree around the colour wheel, while saturation and brightness are set as percentage.

## 8.6 Television Displays

We know that CRT display and many flat panel TV displays are available. The CRT display was already discussed the previous chapter. The CRT has



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two main drawbacks. It is bulky in size and heavy. Also it consumes large electric power to operate. Another display system used in TV is plasma display. The TV which uses this display is called plasma TV. A plasma display is an array of tiny gas cells sandwiched between two sheets of glass. Each cell acts like a mini fluorescent tube emitting UV light which then strikes Red, Green and Blue spots on the screen to produce images.

Another display which almost replaced plasma display is the LCD display. An LCD TV is a flat panel TV that utilizes the basic liquid crystal display technology. LCD panels are made up of two layers of glass like material which are polarized and glued together. One of the glass layers is coated with a special polymer that holds the individual liquid crystals. The property of liquid crystals can be altered using electricity so that light can be either transmitted through or blocked by the polarized glass layers- liquid crystal assembly. This technique is used to produce display in LCDs. Unlike CRT and plasma TVs, there are no phosphors that light up in LCD and hence it requires less power to operate.

A slight modification to the LCD TV came in the form of LED TV in which the required backlight for the LCD is LED. But the LCD TV use cold cathode fluorescent lamps (CCFLs) as the source of backlight. It means that both LCD TV and LED TV use the same LCD display screen but they have different sources of backlight.

The OLED TV (organic LED TV) is a very recent development in which organic material in the form of carbon is used to provide a natural light source to light the display. This allows for OLED screens to be larger, lighter and retain consistent color from even the widest viewing angles.

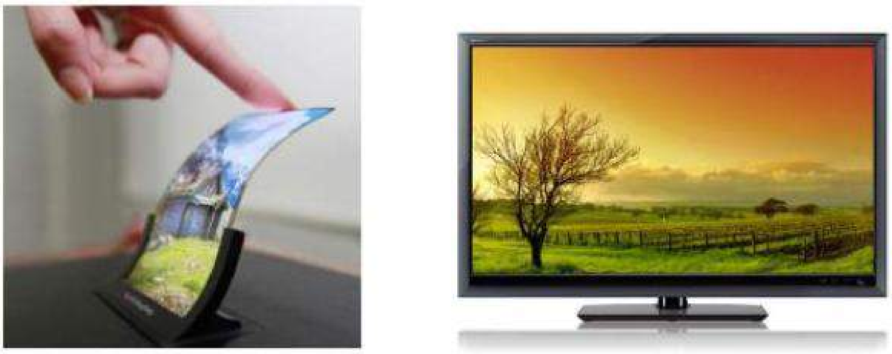

Fig.8.11 OLED and LCD displays

## Know your progress

1. Draw the block diagram of a monochrome TV receiver.
2. State the differences between a colour TV receiver and a monochrome TV receiver.
3. What you mean by additive colour mixing?

## 8.7 Satellite Television

One of the most common methods of TV signal distribution is via communication satellite. They rotate in synchronism with the earth and therefore appears to be stationary. The satellite is used as a radio relay station (Fig.8.11). The TV signal to be distributed is used to modulate a microwave carrier, and then it is transmitted to the satellite. The path from earth to the satellite is called the *uplink*. The satellite translates the signal to another frequency and then retransmits it back to earth. This is called the *downlink*. A receiver site on earth picks up the signal. The receiver site may be a cable TV company or an individual consumer. Satellites are used widely by TV networks, premium channel companies, and cable TV industry for distributing signals nationally. A newer form of consumer satellite TV is *direct broadcast satellite (DBS)* TV. The DBS system is designed specifically for consumers to receive signals directly from the satellite. The new DBS system features digitally encoded video and audio signals, which make transmission and reception more reliable and provides outstanding picture and sound quality. By using higher-frequency microwaves, higher-power satellite transponders, and very low-noise GaAs FETs in the receiver, the customer's satellite dish can be made very small. These systems typically use 18-inch dish while 5- to 12-ft-diameter dishes are still used in older satellite TV systems.

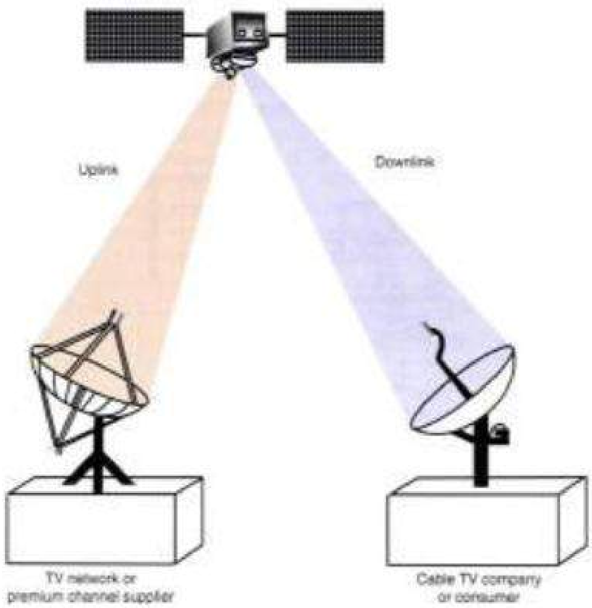


Fig.8.12 Satellite communication

## Satellite dish

Do you have dish antenna at home?

If you can see a dish antenna anywhere, observe the different parts of the antenna and also the shape of the antenna. Then note the position and the direction of the antenna.

A **satellite dish** is a dish-shaped type of parabolic antenna designed to receive electromagnetic signals from satellites.

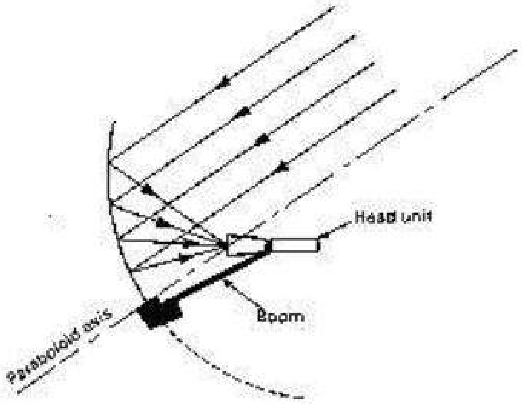


Fig.8.13 Dish antenna

The parabolic shape of a dish reflects the signal to the dish's focal point. The most common type is shaped like a dish and is popularly called a **dish antenna** or **parabolic dish**. The main advantage of a parabolic antenna is that it has high directivity. It functions similar to a searchlight or flashlight reflector to direct the radio waves in a narrow beam, or receive radio waves only from one particular direction. The main parts of a dish antenna receiver system is parabolic reflector, feed antenna, feeder cable and coaxial cable.

### Parabolic reflector

A wire grid-type parabolic antenna is fed by a vertical dipole under a small aluminium reflector. The reflector can be of sheet metal, metal screen, or wire grill construction, and it can either be a circular "dish" or of any other shape to create different beam shapes. A metal screen reflects radio waves like a solid metal surface when the holes are smaller than one-tenth of the wavelength. Screen reflectors are often used to reduce weight and wind loads on the dish. To achieve the maximum gain, it is necessary that the shape of the dish should be accurate within a small fraction of a wavelength, to ensure that the waves from different parts of the antenna arrive at the focus in phase. Large dishes often require a supporting structure behind them to provide the required stiffness.

### Feed antenna

The feed antenna at the reflector's focus is a low-gain type antenna such as a half-wave dipole or more often a small horn antenna called a feed horn. In more complex designs, such as the Cassegrain and Gregorian, a secondary reflector is used to direct the energy to the parabolic reflector from a feed antenna located away from the primary focal point. The feed antenna is connected to the associated radio-frequency (RF) transmitting or receiving equipment by means of a coaxial cable transmission line or waveguide.

Parabolic antennas are used as high-gain antennas for point-to-point communication. They are used in applications such as microwave relay links that carry telephone and television signals between nearby cities, wireless WAN/LAN links for data communication, satellite communication and spacecraft communication antennas. They are also used in radio telescopes.

### Feeder cables

In telecommunication and electronics, an **antenna feed** refers to the components of an antenna which collect the incoming radio waves, convert them to electric currents and transmit them to the receiver. The feed consists of a dipole driven element, which converts the radio waves into electric current, and a coaxial cable or twin lead transmission line which conducts the received signal from the driven element to the television receiver.

### Coaxial cable

Coaxial cable, or coax, is a type of cable that has an inner conductor surrounded by a tubular insulating layer, surrounded by a tubular conducting shield. Many coaxial cables also have an insulating outer sheath or jacket. The term coaxial comes from the inner conductor and the outer shield sharing a geometric axis. The coaxial cable was invented by an English engineer and

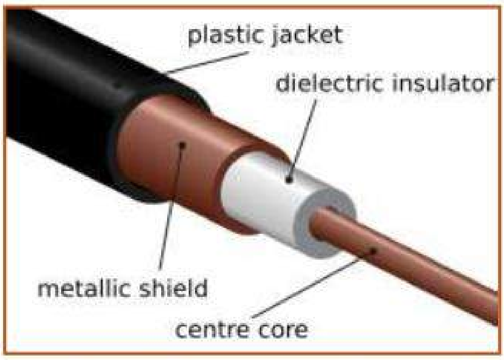


Fig.8.14 Internal structure of a coaxial cable



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mathematician Oliver Heaviside, who patented the design in 1880. A coaxial cable differs from other shielded cables used for carrying low-frequency signals, such as audio signals, in that the dimensions of the cable are controlled to give a precise, constant conductor spacing, which is needed for it to function efficiently as a radio frequency transmission.

A coaxial cable conducts electrical signal using an inner conductor (usually a solid copper, stranded copper or copper plated steel wire) which is surrounded by an insulating layer and enclosed by a shield, typically one to four layers of woven metallic braid and metallic tape. The cable is protected by an outer insulating jacket. Normally, the shield is kept at ground potential and a voltage is applied to the centre conductor to carry electrical signals. The advantage of coaxial design is that the electric and magnetic fields are confined to the dielectric with little leakage outside the shield. Conversely, the electric and magnetic fields outside the cable are largely kept from causing interference to signals inside the cable. Larger diameter cables and cables with multiple shields have less leakage. This property makes coaxial cable a good choice for carrying weak signals that cannot tolerate interference from the environment or for higher electrical signals that must not be allowed to radiate or couple into adjacent structures or circuits.

Common applications of coaxial cable include video and Community Antenna Television (CATV) distribution, RF and microwave transmission, and computer and instrumentation data connections.

**Know your progress**

1. Which are the main parts of a dish antenna?
2. Which kind of cable is used for dish antenna to connect with TV?
3. Explain the parts of a co-axial cable.

## 8.8 Cable TV

If you are using a cable TV system in your home, what are the facilities that you get from the distributer? Check if there are internet facilities, radio, local channels etc. List down the extra services you get from the cable TV other than TV channels.

*Cable TV,* sometimes called *CATV,* is a system for delivering TV signals to home receivers using a coaxial cable rather than over the air by radio wave propagation. A cable TV company collects all the available signals and programs and frequency-multiplexes them on a single coaxial cable that is fed to the homes of the subscribers. A special cable decoder box is used to receive the cable signals, to select the desired channel, and to feed a signal to the TV set. Today, most TV reception uses a cable connection instead of an antenna.

Many companies offer TV signals by cable. They put up very tall high-gain TV antennas. The resulting signals are amplified and fed to the subscribers by cable. Similar systems are developed for apartments and other high rise buildings. Single master antenna systems are installed in buildings by means of which the signals are amplified and distributed to each apartment or unit by using a cable.

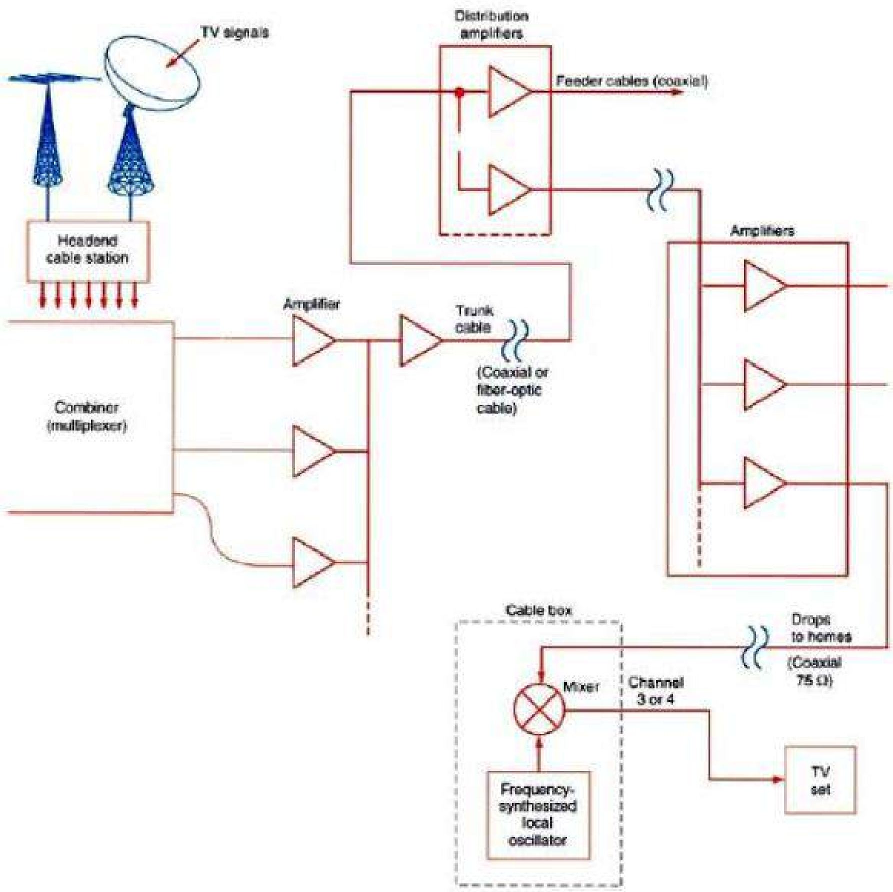


Fig.8.15 The modern cable TV system.

Today, cable TV companies, generally referred to as multiple (cable) systems operators (MSOs), collect signals and programs from many sources, multiplex them, and distribute them to the subscribers (see Fig. 8.14). The main building



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or facility is called the *head end*. The antennas receive local TV stations and other nearby stations in addition to the special cable channel signals distributed by satellite. The cable companies use parabolic dishes to pick up the so-called premium cable channels. A cable TV company uses many TV antennas and receivers to pick up the stations, whose programs will be redistributed by it. These signals are then processed and combined or frequency-multiplexed onto a single cable. The main output cable is called the *trunk cable*. In older systems it was a large, low loss *coaxial cable*. Newer systems use fiber-optic cable. The trunk cable is usually buried and extended to surrounding areas. A junction box containing amplifiers takes the signal and redistributes it to smaller cables, called *feeders,* which go to specific areas and neighbourhoods. From there, the signals are again rejuvenated with amplifiers and sent to individual homes using coaxial cables called *drops*. The overall system is referred to as a *hybrid fiber cable (HFC) system*. The coaxial cable (usually) comes to a home and is connected to a cable decoder box, which is essentially a special TV tuner that picks up the cable channels and provides a frequency synthesizer and mixer to select the desired channel. The mixer output is heterodyned and then fed to the TV set antenna terminals. The desired signal is frequency-translated by the cable box to the channel that the TV set can receive. This service eliminates the need for antennas. And because of the direct connection of amplified signals, there is no such thing as poor, weak, noisy, or snowy signals. In addition, many TV programs are available only via cable, e.g., the specialized content and premium movie channels. The only downside to cable TV is that it is more expensive than connecting a TV to a standard antenna.

## Let us consolidate

Transmission of TV signal through a single channel requires scanning of image on the photo sensitive target plate of a camera tube from left to right and top to bottom. Reception of these signals requires identical scanning. Interlaced scanning is used to reduce flicker. Two sequences of scanning in interlaced scanning are called fields. The number of fields transmitted- per second is called frequency. The video signal representing the picture and the sound signals are transmitted to the receiver. The

TV receiver is a special super heterodyne that receives the sound and picture information. The picture is displayed on a picture tube.

Today, most of the TV signals are transmitted via a hybrid fiber optic co-axial cable system. A converter box at the TV set converts the cable signal to a format compatible with the TV receiver. One of the most common means of TV signal distribution is via communication satellite (dish TV). Now a days CRT monitors are replaced by LCD and LED TV with high definition picture.

## Let us asses

1. Scanning process is used to transmit video information in TV. Name the television scanning process.
2. In television scanning, the combination of two fields is called ________ (frame, vertical trace, horizontal trace, retrace)
3. In order to avoid flicker ________ scanning is used in TV. (interlaced scanning, progressive scanning, document scanning)
4. In a colour TV system, ________ type of colour mixing is used.
5. Synchronising pulses are transmitted along with the picture signal. Explain.
6. A TV signal consists of both picture and sound information. With the help of the complete channel bandwidth, explain the arrangement.
7. Super heterodyne receiver is a part of monochrome TV receiver. Draw the block diagram of a TV receiver and explain the process of separating sound and picture information.
8. With the help of a picture, explain the parts of a dish antenna.
9. List the major services provided by the modern cable TV system.

9

# FUNDAMENTALS OF COMPUTERS

## Significant Learning Outcomes

*After completing this chapter the learner:*

- sketches the block diagram of a computer and explains its structure.
- explains the functions of various input and output devices.
- differentiates various printing technologies.
- identifies various units of memory storage.
- explains the characteristics of primary memory.
- explains the characteristics of different secondary storage devices.
- distinguishes between static and dynamic RAM.
- classifies computers on the basis of speed and computing power.
- explains the functions and structure of motherboard.
- identifies different computer ports.
- classifies computer softwares.
- explains various system and application softwares.
- differentiates various computer languages.

Now a days, every one is familiar with computer and it has become an inevitable part of our daily life. In our lower classes, we have studied the very basics of computers. So we are familiar with basic block diagrams, applications, software, hardware and so on. As an electronics student, one has to know much more about computers and so to begin with, let us memorise the very basics and then enter into a detailed study.

A computer is a fast and accurate electronic machine which stores data, processes it and produces output results under the direction of stored program of instructions. Its wide spread applications can be seen in the field of education, business, entertainment, communication, instrumentation, medical field, defence etc. We are also much familiar with the internet, E mail, social networking sites and so on. So in this chapter we shall learn more about computer hardware, software and networking of computers.

## 9.1 Basic block diagram of a computer

The basic building block of a computer can be represented as shown in Fig 9.1.

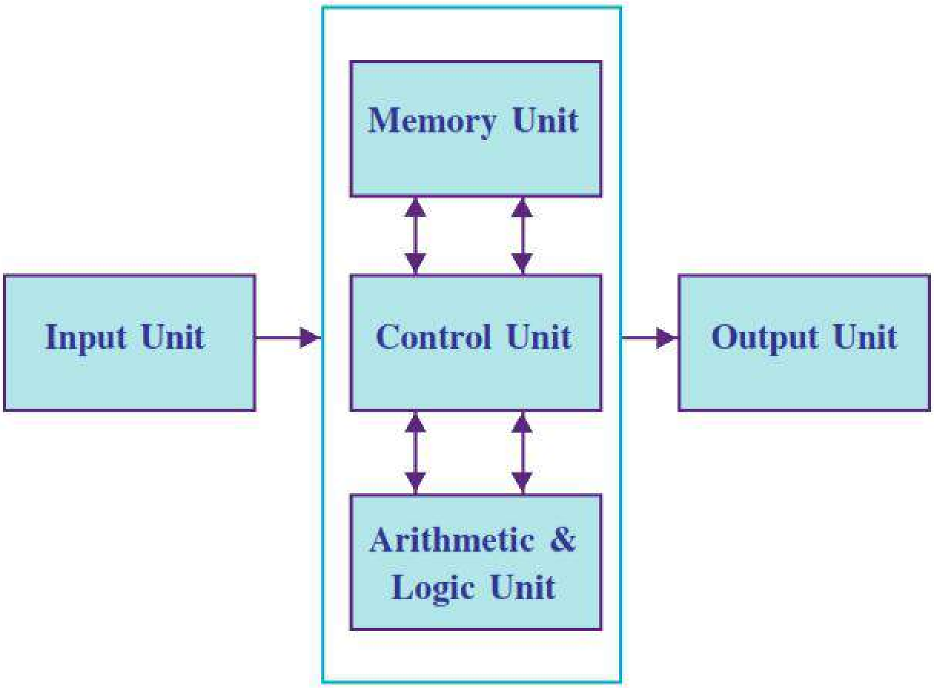


Fig 9.1 : Basic building block of a computer

## Input Unit

This unit contains devices with the help of which, we can enter data into a computer. This unit helps to make a link between the user and the computer. The input devices translate the information into a form which can be understood by the computer.

## CPU (Central Processing Unit)

CPU is considered as the brain of the computer. CPU performs all types of data processing operations. It stores data, intermediate results and instructions (program). It controls the operation of all parts of the computer.CPU itself has the following three components

- ALU (Arithmetic Logic Unit)
- Memory Unit
- Control Unit

## Output Unit

The output unit consists of devices with the help of which we get information from the computer. This unit acts as a link between the computer and the users. The output devices translate the computer's output into a form understandable by the users.

**Know your progress**

*What are the basic functions of a CPU ?*

## 9.2 Input devices

Following are some of the important input devices which are used in a computer:

- Keyboard
- Mouse
- Joy Stick
- Light pen
- Track Ball
- Scanner
- Graphic Tablet
- Microphone
- Magnetic Ink Card Reader (MICR)
- Optical Character Reader (OCR)
- Bar Code Reader
- Optical Mark Reader (OMR)

### Keyboard

The keyboard is the most common and very popular input device which helps in inputting data to the computer. The layout of the keyboard is like that of traditional typewriter, although there are some additional keys provided for performing additional functions. Keyboards are of two sizes 84 keys or 101/102 keys, but now keyboards with 104 keys or 108 keys are also available for Windows and Internet. The keys on the keyboard are as follows:

| Sl.No | Keys | Description |
|---|---|---|
| 1 | **Typing Keys** | These keys include the letter keys (A-Z) and digit keys (0-9) which generally has the same layout as that of typewriters. |
| 2 | **Numeric Keypad** | It is used to enter numeric data or for cursor movement. Generally, it consists of a set of 17 keys that are laid out in the same configuration as used by most adding machines and calculators. |

| | | |
|---|---|---|
| 3 | **Function Keys** | Twelve function keys are present on the keyboard which are arranged in a row at the top of the keyboard. Each function key has a unique meaning and is used for some specific purpose. |
| 4 | **Control keys** | These keys provide cursor and screen control. It includes four directional arrow keys. Control keys also include Home, End, Insert, Delete, Page Up, Page Down, Control(Ctrl), Alternate(Alt), Escape(Esc) keys. |
| 5 | **Special Purpose Keys** | Keyboard also contains some special purpose keys such as Enter, Shift, Caps Lock, Num Lock, Space bar, Tab, and Print Screen. |

Fig 9.2 : Keyboard

## Mouse

Mouse is the most popular pointing device. It is a very famous cursor-control device having a small palm size box with a round ball at its base which senses the movement of the mouse and sends corresponding signals to CPU when the mouse buttons are pressed.

Generally it has two buttons, left and right button and a wheel is present between the buttons. A mouse can be used to control the position of cursor on screen, but it cannot be used to enter text into the computer.

Fig : 9.3 Mouse

**Advantages**

- Easy to use
- Not very expensive
- Moves the cursor faster than the arrow keys of the keyboard.

## Joystick

Joystick is also a pointing device which is used to move the cursor position on the monitor screen. It is a stick having a spherical ball both at the upper and lower ends. The lower spherical ball moves in a socket. The joystick can be moved in all the four directions.

Fig : 9.4 Joystick

The function of joystick is similar to that of a mouse. It is mainly used in Computer Aided Designing (CAD) and for playing computer games.

## Light Pen

Light pen is a pointing device which is similar to a pen. It is used to select a displayed menu item or draw pictures on the monitor screen. It consists of a photocell and an optical system placed in a small tube. When the tip of a light pen is moved over the monitor screen and pen button is pressed, its photocell sensing element detects the screen location and sends the corresponding signal to the CPU.

Fig : 9.5 Lightpen

## Track Ball

Track ball is an input device that is mostly used in notebook or laptop computer, instead of a mouse. This is a ball which is half inserted and by moving fingers on the ball, the pointer can be moved. Since the whole device is not moved, a track ball requires only less space than a mouse. A track ball comes in various shapes like a ball, a button and a square.

Fig : 9.6 Trackball

## Scanner

Scanner is an input device which works more like a photocopy machine. It is used when some information available on the paper is to be transferred to the hard disc of the computer for further manipulation. Scanner captures images from the source which are then converted into the digital form that can be stored on the disc. These images can be edited before they are printed.

Fig 9.7 : Scanner

## Digitizer

Digitizer is an input device which converts analog information into digital form. Digitizer can convert a signal from the television or camera into a series of numbers that could be stored in a computer. They can be used by the computer to create a picture of whatever the camera is being pointed at. Digitizer is also known as Tablet or Graphics Tablet because it converts graphics and pictorial data into binary inputs. A graphic tablet as digitizer is used for doing fine works of drawing and image manipulation applications.

Fig 9.8 : Digitizer

## Microphone

Microphone is an input device to input sound that is then stored in digital form. The microphone is used for various applications like adding sound to a multimedia presentation or for mixing music.

Fig 9.9 : Microphone

## Magnetic Ink Character Recognizer (MICR)

MICR input device is generally used in banks to process a large number of cheques every day. The bank's code number and the cheque number are printed on the cheques with a special type of ink that contains particles of magnetic material that are machine readable. The process of reading these ink characters is called Magnetic Ink Character Recognition

Fig 9.10 : MICR

(MICR). The main advantages of MICR is that it is fast and the errors are very few.

### Optical Character Reader(OCR)

OCR is an input device used to read a printed text. OCR scans the text optically character by character, converts them into a machine readable code and stores the text on the system memory.

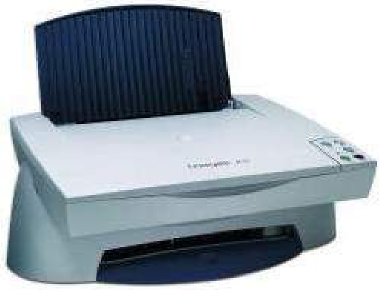

Fig 9.11 :
Optical character reader

### Bar Code Readers

Bar Code Reader is a device used for reading bar coded data (data in the form of light and dark lines). Bar coded data is generally used in labelling goods, numbering the books etc. It may be a hand held scanner or embedded in a stationary scanner. Bar Code Reader scans a bar code image, converts it into an alphanumeric value which is then fed to the computer to which bar code reader is connected.

Fig 9.12 : Barcode reader

### Optical Mark Reader (OMR)

OMR is a special type of optical scanner used to recognize the type of mark made by pen or pencil. It is used where one out of a few alternatives is to be selected and marked. It is specially used for checking the answer sheets of examinations having multiple choice questions.


Fig 9.13 : Optical mark reader



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**Know your progress**

- *Give the functions of a scanner and digitizer.*
- *What are the applications of a bar code reader ?*

## 9.3 Output devices

The following are a few of the important output devices which are used in a computer.

- Monitors
- Graphic Plotter
- Printer

### Monitors

A monitor, commonly called as the Visual Display Unit (VDU), is the main output device of a computer. It forms images from tiny dots, called pixels that are arranged in a rectangular form. The sharpness of the image depends upon the number of pixels.

There are two kinds of viewing screens for monitors.

- Cathode-Ray Tube (CRT)
- Flat- Panel Display

### Cathode-Ray Tube (CRT) Monitor

The CRT display is made up of small picture elements called pixels. The smaller the pixels, the better is the image clarity, or resolution. It takes more than one illuminated pixel to form the whole character. A finite number of characters can be displayed on the screen all at once. The screen can be divided into a series of character boxes which are fixed locations on the screen where standard character can be placed. Most screens are capable of displaying 80 characters of data horizontally and 25 lines vertically. The disadvantages of CRT are:

- Large in Size
- High power consumption

Fig 9.14 : CRT monitor

## Flat-Panel Display Monitor

The flat-panel display refers to a class of video devices that have reduced volume, weight and power requirement in comparison to CRT. You can hang them on walls or wear them on your wrists. Current uses of flat-panel displays include calculators, video games, monitors, laptop computer, graphics display.

The flat-panel display is divided into two categories:

- **Emissive Displays** - The emissive displays are devices that can convert electrical energy into light. Examples are plasma panel and LED (Light-Emitting Diodes).
- **Non-Emissive Displays** - The non-emissive displays use optical effects to convert sunlight or light from some other source into graphic patterns. Example is LCD (Liquid-Crystal Device).

Fig 9.15 :
Flat panel display monitor

## Printers

A printer is an output device, which is used to print information on paper.

There are two types of printers:

- Impact Printers
- Non-Impact Printers

### Impact Printers

The impact printers print the characters by striking them on the ribbon which is then pressed on to the paper.

The characteristics of Impact Printers are:

- Very low consumable costs
- Very noisy
- Useful for bulk printing due to low cost
- There is physical contact with the paper to produce an image

Dot matrix printer is an example of an impact printer.

## DOT MATRIX PRINTER

In the market one of the most popular printers is Dot Matrix Printer. These printers are popular because of their ease of printing and economical price. Each character printed is in the form of pattern of dots which come out to form a character. That is why it is called Dot Matrix Printer.

**Advantages**

- Inexpensive
- Widely Used
- Other language characters can be printed

**Disadvantages**

- Slow Speed
- Poor Quality

Fig 9.16 : Dot matrix printer

## NON-IMPACT PRINTERS

Non-impact printers print the characters without using ribbon. These printers can print a complete page at a time, so they are also called Page Printers.

These printers are of three types

- Laser Printers
- Inkjet Printers
- Thermal Printers

Characteristics of Non-impact Printers

- Faster than impact printers.
- They are not noisy.
- High quality.
- Support many fonts and different character size.

### Laser printers

Laser printers are the fastest and most popular printers on the market today. They produce extremely high quality images – close to photo quality.

**Main Principle of Laser Printer**

The main principle in the working of laser printer is static electricity i.e., they use electro photography, or an electrophotostatic process, to form images on a paper. The basic principles involved here is the science of atoms – oppositely-charged atoms are attracted to each other, so opposite static electricity fields cling together.

**Parts of a laser printer**

The basic parts of a laser printer are toner cartridges, photosensitive drum, erase lamp, primary corona, transfer corona, fuser assembly. Each of these parts has a very important role to play in the printing process.

**How does it work?**

The drum is the main component of a laser printer and is often located near the center. It is usually made of a highly photoconductive material that can be charged or discharged by light. The drum interfaces directly with the paper and places the toner at the right locations to produce the image.

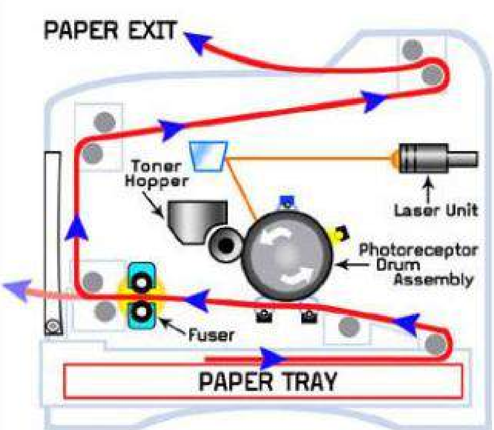


Fig 9.17 : Components of Laser printer

The way that the drum works is that, it is given an initial charge to begin with. As the drum rotates in circles, the laser shines upon certain areas of the drum. The parts of the drum that get exposed to the laser experience a change in charge. For example, in certain laser printers, the drum is initially given strong negative electro-static charge and the laser causes the exposed areas to change from a negative to a positive electro-static charge. In this way, the laser generates an electrostatic image on the drum.

Then, the printer exposes the rotating drum to negatively charged toner particles. The toner particles are attracted to the positive areas of the drum that are exposed by the laser. As a result, an electrostatic image is developed on the drum surface that will get transferred to the paper at a later stage.

Now, the paper is given a strong positive charge (much stronger than that of the drum) and is slid beneath the drum. Since the paper has a stronger positive charge than the drum, it takes the toner off the drum so that the pattern from the drum is translated to the paper. Then, the paper goes through the fuser and the toner particles are fused into the paper.

**In brief, the steps involved in the working of a laser printer are given below:**

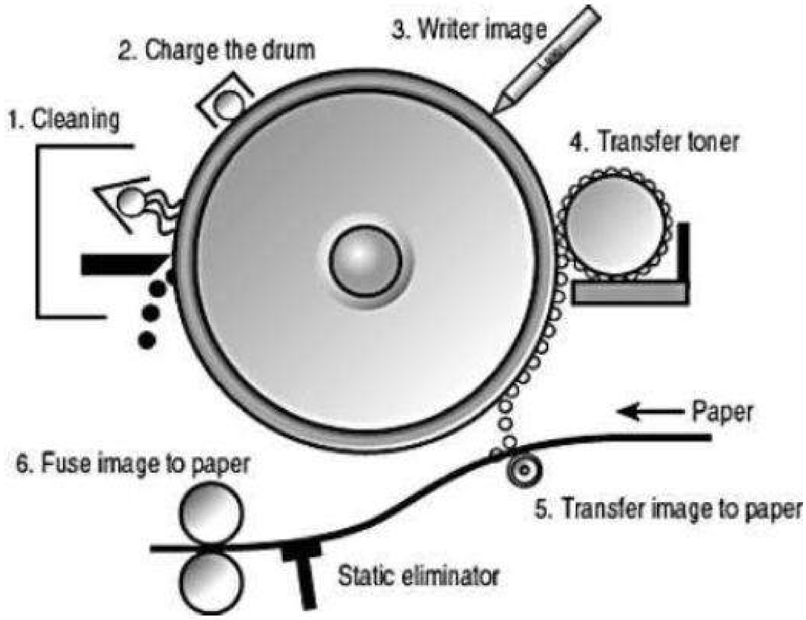


Fig9.18 : Functions of Laser printer

1. **Paper feeding**

   The printer moves a sheet of paper from the proper tray onto a series of rollers, through the imaging and fixing areas, and to the output hopper.

2. **Drum Cleaning and Charging**

   Any residual toner from past jobs is scraped from the printer's photosensitive drum. A fine wire (the primary corona) produces a negative electrical charge across the entire face of the drum. The image is set in raster lines as a series of fine dots on the drum.

3. **Imaging the Drum**

   The information from the raster-image processor is read from the memory and sent to the print engine, one line at a time. The laser sets a positive charge in the areas of the image to be filled with toner.

4. **Transferring Toner to the Drum**

   A film of fine plastic is placed on the toner transfer roller, which turns close to the photosensitive drum. This toner is then attracted to the positively charged areas of the drum.

5. **Transferring Toner to the Paper**

   The corona wire places a positive electrical charge on the paper as it moves close to the drum. The toner is attracted to the page, forming an image.

6. **Fusing the Toner**

   The page passes through a pair of rollers. The heating roller heats the toner that has been placed on the page to melt the plastic toner particles onto the page without smearing. The roller on the other side supplies the needed pressure.

### Advantages and Disadvantages

Laser printers have a number of advantages over the rival inkjet technology. They produce much better quality text documents than inkjets, and they turn out more pages per minute (100 to 200 pages per minute are typical) at a lower cost per page than inkjets.

Laser printers are well known for their speed and they can handle large volumes and another advantage is that they are not messy as inkjet that is, there is no ink spillage as the ink is created from powder and they can print on any type of paper and the disadvantage is that laser printers are expensive.

### Advantages

- Very high speed
- Very high quality output
- Give good graphics quality
- Support many fonts and different character size

### Disadvantages

- Expensive.
- Cannot be used to produce multiple copies of a document in a single printing.

## Inkjet Printers

Fig 9.19 : Inkjet Printer

Inkjet printers are non-impact character printers based on a relatively new technology. They print characters by spraying small drops of ink onto paper. Inkjet printers produce high quality output with presentable features.

They make less noise because no hammering is done and these have many styles of printing modes. Colour printing is also possible. Some models of Inkjet printers can produce multiple copies of printing also.

### Advantages

- High quality printing
- More reliable

### Disadvantages

- Expensive as cost per page is high
- Slow as compared to laser printer

## THERMAL PRINTER

Thermal printer technology is different from inkjet or laser printer, thermal printer uses heat produced by heating resistors in the printer. Basically thermal printer prints by applying heat on a chemical coated paper. The coating turns black in the areas where it is heated, producing an image.

The printing mechanism in thermal printer are not similar to the regular type of printing mechanism found in Inkjet or Laser printers. This printing technology was widely used in FAX machines earlier. Thermal printers are used in restaurants, in service buses for issuing bus tickets and where,printing on small paper is required.

### Working principle

Fig 9.20 : Thermal printer

Thermal printer operates on the application of heat. A thermal print head is a single component which has the embedded circuit, heating resistors etc.

A roll of thermal sensitive paper is placed in a container provided in the machine. The end of the roll is fed into a slot over a pressure roller. When the data to be printed is sent to printer, it heats appropriate heating resistors in the printer which in turn heats and burns the chemical coating in the paper thereby forming a black impression. This kind of continuous black impression forms the printed data or image on the paper. This kind of printing is simple in terms of maintenance and is also economical. If any one of the heating resistors is damaged, the total printing on the paper related to that damaged resistor is affected. This produces vertical white lines on the paper. To rectify this problem, full thermal print head is to be replaced. It cannot be used like other regular printers because the printing process is very slow, the print head cannot withstand high volume of printing and color printing is also not possible.

One of the main disadvantages of thermal printer is that the printed data on the paper vanishes after some days. So we need to take a photo copy in order to preserve the document.

**Know your progress**

*Distinguish between the impact and nonimpact printers.*

## 9.4 Memory

A memory is just like a human brain. It is used to store data and instructions. Computer memory is the storage space in computer where data is to be processed and instructions required for processing can be stored. The memory is divided into a large number of small parts called cells. Each location or cell has a unique address which varies from zero to memory size minus one. For example if computer has 64k words, then this memory unit has 64 * 1024=65536 memory locations. The address of these locations varies from 0 to 65535.

The unit of memory is the amount of data that can be stored in the storage unit or that in which storage capacity is expressed in terms of Bytes.

Following are the main memory storage units:

| Sl.No. | Unit | Description |
|---|---|---|
| 1 | Bit (Binary Digit) | A binary digit is logical 0 and 1 representing a passive or an active state of a component in an electric circuit. |

| | | |
|---|---|---|
| 2 | Nibble | A group of 4 bits is called a nibble. |
| 3 | Byte | A group of 8 bits is called a byte. A byte is the smallest unit which can represent a data item or a character. |
| 4 | Word | A computer word, like a byte, is a group of fixed number of bits processed as a unit which varies from computer to computer but is fixed for each computer.<br>The length of a computer word is called word-size or word length and it may be as small as 8 bits or may be as long as 96 bits. A computer stores the information in the form of computer words. |

| Sl.No. | Unit | Description |
|---|---|---|
| 1 | Kilobyte (kB) | 1 KB = 1024 Bytes |
| 2 | Megabyte (MB) | 1 MB = 1024 kB |
| 3 | GigaByte (GB) | 1 GB = 1024 MB |
| 4 | TeraByte (TB) | 1 TB = 1024 GB |
| 5 | PetaByte (PB) | 1 PB = 1024 TB |

Memory is primarily of three types

- Cache Memory
- Primary Memory/Main Memory
- Secondary Memory

## Cache Memory

Cache memory is a very high speed semiconductor memory which can speed up CPU. It acts as a buffer between the CPU and the main memory. It is used to hold those parts of data and program which are most frequently used by CPU. The parts of data and programs are transferred from disk to cache memory by the operating system, from where the CPU can access them.

### Advantages

The advantages of cache memory are as follows:

- Cache memory is faster than the main memory.
- It consumes less access time as compared to main memory.
- It stores the program that can be executed within a short period of time.
- It stores data for temporary use.

### Disadvantages

The disadvantages of cache memory are as follows:

- Cache memory has limited capacity.
- It is very expensive.

## Primary Memory (Main Memory)

Primary memory holds only those data and instructions on which the computer is currently working. It has limited capacity and the data is lost when power is switched off. It is generally made up of semiconductor device. These memories are not as fast as registers. The data and instructions required to be processed reside in the main memory. It is divided into two subcategories RAM and ROM.

### Characteristics of Main Memory

- These are semiconductor memories
- Usually volatile memory.
- Data is lost in case, the power is switched off.
- It is the working memory of the computer.
- Faster than secondary memories.
- A computer cannot run without primary memory.

RAM (Random Access Memory) is the internal memory of the CPU for storing data, program and program result. It is read/write memory which stores data until the machine is working. As soon as the machine is switched off, the data is erased.

Access time in RAM is independent of the address that is, each storage location inside the memory is as easy to reach as other locations and takes the same amount of time. Data in the RAM can be accessed randomly but it is very expensive.

RAM is volatile, i.e. data stored in it is lost when the computer is switched off or if there is a power failure. Hence a backup uninterruptible power supply (UPS) is often used with computers. RAM is small, both in terms of its physical size and the amount of data it can hold.

Fig 9.21 : RAM

- Static RAM (SRAM)
- Dynamic RAM (DRAM)

## Static RAM (SRAM)

The word **static** indicates that the memory retains its contents as long as power is being supplied. However, the data is lost when the power gets down because it is volatile in nature. SRAM chips use a matrix of 6-transistors and has no capacitors. Transistors do not require power to prevent leakage, so SRAM need not be refreshed on a regular basis.

Because of the extra space in the matrix, SRAM uses more chips than DRAM for the same amount of storage space, thus making the manufacturing costs higher. So SRAM is used as cache memory and has very fast access.

Characteristic of the Static RAM

- It has long life
- There is no need to refresh
- It is faster
- It can be used as cache memory
- It is large in size
- It is expensive
- It needs high power consumption

## Dynamic RAM (DRAM)

DRAM, unlike SRAM, must be continuously **refreshed** in order to maintain the data. This is done by placing the memory on a refresh circuit that rewrites the data several hundreds of times per second. DRAM is used for most of the system memory because it is cheap and small. All DRAMs are made up of memory cells which are composed of one capacitor and one transistor.

**Characteristics of the Dynamic RAM**

- It has short data lifetime
- Need to be refreshed continuously
- Slower as compared to SRAM
- Used as RAM
- Lesser in size
- Less expensive
- Less power consumption

## Secondary Memory

This type of memory is also known as external memory or non-volatile memory. It is slower than the main memory. These are used for storing data/ information permanently. Contents of secondary memories are first transferred to main memory, and then CPU can access it. For example : Hard disk, CD-ROM, DVD etc.

There are several types of secondary storage media used in today's world, where each of these can be compared to the other in terms of portability, speed and capacity.

**Characteristic of Secondary Memory**

- These are magnetic and optical memories
- It is also known as backup memory.
- It is non-volatile memory.
- Data is permanently stored even if power is switched off.
- It is used for storage of data in a computer.
- Computer may run without secondary memory.
- Slower than primary memories.

## Magnetic Tape

Magnetic Tape is a recording medium consisting of a thin tape with a coating of a fine magnetic material, used for recording analogue or digital data. A device that stores computer data on magnetic tape is a tape drive. The capacity of tape media are generally of the same order as that of hard disk drives (The largest being about 5 Terabytes in 2011). Though magnetic tapes generally transfer data a bit slower than the hard drives, they are cheaper and are more durable.

### Floppy Disk

Floppy Disks were considered as a main form of data storage between 1980's and early 2000's. However, they are now superseded by data storage methods with much greater capacity, such as USB flash drives. Floppy disks come in 3 sizes: 8-inches, 5.5-inches and 3.5-inches. The capacities of Floppy disks vary between 1-250 Megabytes and these devices are very slow, reading data at rates ofbytes and kilobytes/second. However, most of them are very small and portable.

Fig 9.22 : Floppy disk

## 9.5 Hard Disk

The hard disk is the main, and usually the largest data storage device in a computer. It is a non-volatile, random access digital magnetic data storage device. A hard disk is made up of platters which store the data, and read/ write heads to transfer data. A hard disk is generally the fastest of the secondary storage devices, and has the largest data storage capacity, approximately the same as magnetic tapes. Hard disks however, are not very portable and are primarily used internally in a computer system. Some people use hard drives as a form of storage externally and as a substitute for portable storage, hard drives used for these purposes are called external hard drives.

Fig 9.23 : Hard disk

A hard disk is divided into tracks and sectors. The data on this hard disk is positioned into these tracks and sectors so that they can be easily read by the heads and also help reduce fragmentation on the hard disk. Data on a hard drive are accessed by two methods:

1. **Fixed Head**

   Hard Disks with fixed heads have a read/write head for each track on the hard disk, since there is no moving of heads to access data, the data access time is generally faster for fixed head hard disk.

2. **Moving Head**

   A moving head hard disk is one in which one or more read-write heads are attached to a movable arm which allows each head to cover many tracks of information.

## 9.6 Optical Disks

Optical disk is an electronic data storage medium from which data is read and written to by using a low-powered laser beam. It is a flat, circular plastic or glass disk on which data is stored in the form of light and dark pits. There are three basic types of optical disks: Read-only optical disks, Write once read many Optical disks and Rewritable Optical disks. Two main types of optical disks are:

**CD** - is an abbreviation of compact disk, and is a form of data storage that can transfer data up to the speed of 7800 kbps. A standard 120 mm CD holds up to 700 MB of data, or about 70 minutes of audio. There are two types of CD: CD-ROM and CD-RW. CD-ROM stands for CD-Read Only Memory and they function the same way as Read Only Memory does. CD-RW Stands for CD-Rewritable, these disks can be erased and rewritten at any time.

**DVD**: is an abbreviation of Digital Versatile Disc, and is an optical disc storage media format that can be used for data storage. The DVD supports disks with capacities of 4.7 GB to 17 GB and has access rates of 600 kbps to 1.3 Mbps. A standard DVD stores up to 4.7 GB of data. There are two types of DVD's: DVD-ROM and DVD-RW. DVD-ROM stands for DVD-Read Only Memory and they function in the same way as Read Only Memory does. DVD-RW (DVD - Rewritable) can be erased and rewritten at any time.

## Flash Drive

A flash drive is a small external storage device, typically of the size of a human thumb that consists of flash memory. USB flash drives are removable and rewritable flash memory. They are solid-state storage media that are both inexpensive and durable. Currently, USB 2.0 flash drives on the market are able to reach a data transfer speed of 480 Mbps and USB 3.0 has transmission speeds up to 5 Gbps. USB Flash drives vary in size from 8 Megabytes to 512 Gigabytes. More commonly used sizes vary from 2 Gigabytes to16 Gigabytes.

Fig 9.24 : Flash drive

## Flash Memory cards

Flash memory is an EEPROM non-volatile computer storage chip. These memory cards currently vary in size between 1 Gigabytes and 16 Gigabytes and they transfer data at a rate of approximately 14.65 MB/s. Flash memory cards have most of the characteristics of a flash drive in that they are inexpensive and durable, and are very small. However, flash memory cards are flat and have a size of about 1 inch x 0.75 inch with a thickness of about 2mm. Flash memory cards also have a smaller version which is used in cell phones. These smaller cards are about 6mm x 3mm in size having thickness less than 1mm .

Fig 9.25 : Flash drive

### Know your progress

1.Classify the following as input, output and memory devices.

Laser printer, monitor, scanner, digitizer, microphone, loudspeaker, RAM, joystick, hard disk, CDROM, flash drive.

2.The secondary memory which acts as a buffer between main memory and the CPU is.......................

## 9.7 Classification of computers

Computers can broadly be classified by their speed and computing power.

| Sl.No. | Type | Specifications |
|---|---|---|
| 1 | PC (Personal Computer) | It is a single user computer system having moderately powerful microprocessor |
| 2 | WorkStation | It is also a single user computer system which is similar to personal computer but having more powerful microprocessor. |
| 3 | Mini Computer | It is a multi-user computer system which is capable of supporting hundreds of users simultaneously. |
| 4 | Main Frame | It is a multi-user computer system which is capable of supporting hundreds of users simultaneously. Software technology is different from minicomputer. |
| 5 | Supercomputer | It is an extremely fast computer which can execute hundreds of millions of instructions per second. |

### PC (Personal Computer)

A PC can be defined as a small, relatively inexpensive computer designed for an individual user. PC's are based on the microprocessor technology that enable manufacturers to put an entire CPU on one chip. Businesses use Personal computers are used for word processing, accounting, desktop publishing, and for running spreadsheet and database management applications. At home, the most popular use of personal computers is for playing games and surfing Internet

Fig 9.26 : Personal computer



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Although personal computers are designed as single-user systems, these systems are normally linked together to form a network. In terms of power, now-a-days, high-end models of the Macintosh PC offer the same computing power and graphic capability, as the low-end workstations provided by Sun Microsystems, Hewlett-Packard, and Dell.

## Workstation

Workstation is a computer used for engineering applications (CAD/CAM), desktop publishing, software development, and other such types of applications which require a moderate amount of computing power and relatively high quality graphic capabilities.

Workstations generally come with a large, high-resolution graphics screen, large amount of RAM, inbuilt network support, and a graphical user interface. Most workstations also have a mass storage device such as a disk drive, but a special type of workstation, called a diskless workstation, comes without a disk drive.

Common operating systems for workstations are UNIX and Windows NT. Like PC, Workstations are also single-user computers but are typically linked together to form a local-area network, although they can also be used as stand-alone systems.

Fig 9.27 : Work station

## Minicomputer

It is a midsize multi-processing system capable of supporting nearly 250 users simultaneously.

Fig 9.28 Mini computer

## Mainframe

Mainframe is very large in size and is an expensive computer capable of supporting hundreds or even thousands of users simultaneously. Mainframe executes many programs concurrently and supports simultaneous execution of programs.

Fig 9.29 Mainframe computer

## Supercomputer

Supercomputers are one of the fastest computers currently available. Supercomputers are very expensive and are employed for specialized applications that require immense amount of mathematical calculations (number crunching). For example, weather forecasting, scientific simulations, (animations) graphics, fluid dynamic calculations, nuclear energy research, electronic design, and analysis of geological data (e.g. in petrochemical prospecting).

Fig 9.30 Supercomputer

**Know your progress**

Distinguish between personal and mainframe computers.

## 9.8 Motherboard of a computer

Fig 9.31 Motherboard

The motherboard serves to connect all the parts of a computer together. The CPU,memory, hard drives, optical drives, video card, sound card and other ports and expansion cards all connect to the motherboard directly or via cables. Motherboard is the piece of computer hardware that can be thought of as the 'back bone' of the PC. It is also known as mainboard, mobo (abbreviation), MB (abbreviation) system board, or logic board.

### Important Motherboard Facts

Desktop motherboards, cases and power supplies all come in different sizes called form factors. All the three must be compatible to work together properly.

Motherboards vary greatly with respect to the types of components they support. For example, each motherboard supports a single type of CPU and a short list of memory types. The motherboard manufacturer should provide clear guidance of component compatibilities. In laptops, tablets, and even desktops, the motherboard often incorporates the functions of the video card and sound card. This helps to keep the size of the computers small.

### Popular Motherboard Manufacturers

ASUS, AOpen, Intel, ABIT, MSI, Gigabyte, Biostar

## 9.9 Computer Ports

**Computer ports** are connecting points or interfaces with peripheral devices that work to communicate with your computer.

A port has the following characteristics:

- External devices are connected to a computer using cables and ports.
- Ports are slots on the motherboard into which a cable of external device is plugged in.

Examples of external devices attached via ports are mouse, keyboard, monitor, microphone, speakers etc.

**USB Port (Universal Serial Bus)**

USB port was created in the mid 1990's to standardize the communication between computers and peripheral devices. USB ports can also be used as power supply for different devices like cellphones, cameras, laptop coolers and more. There are four different types of USB computer ports: USB 1.0 and 1.1 released between 1996 and 1998 with a speed range starting from 1.5 Mbps up to 12 Mbps. USB 2.0 was released in 2000 with a maximum speed of 480 Mbps. Finally USB 3.0 was released in 2008 with a maximum speed of 5 Gbps.

**Ethernet / Internet Ports**

Ethernet ports were first introduced in 1980 to standardize the local area networks (LAN). Internet ports use RJ45 connectors and have speeds in various ranges such as 10 Mbps, 100 Mbps, 1 Gbps, 40 Gbps, 100 Gbps.

**IEEE 1394 Ports**

This technology was developed by Apple between 1980 and 1990 with the name 'FireWire' and it is the equivalent of the USB.

**TRS Ports**

Used for receiving and transmitting analog signals mainly audio.

**PS/2 Ports**

Introduced in 1987 to replace the serial mouse and keyboard.

**Serial Port**

Uses the DB9 socket connector and transfers information one bit at a time between the computer and external peripherals.

**VGA ports (Video Graphics Array)**

This port has 15 pins on three rows and it is used for connecting the monitor with the video adapter from the computer motherboard to display video on the monitor.

**HDMI (High Definition Multimedia Interface) ports**

These are ports on computer to transmit High Definition video from the computer video card to the monitor.

**DVI (Digital Visual Interface)**

Computer ports used to transmit uncompressed digital video data.

**Know your progress**

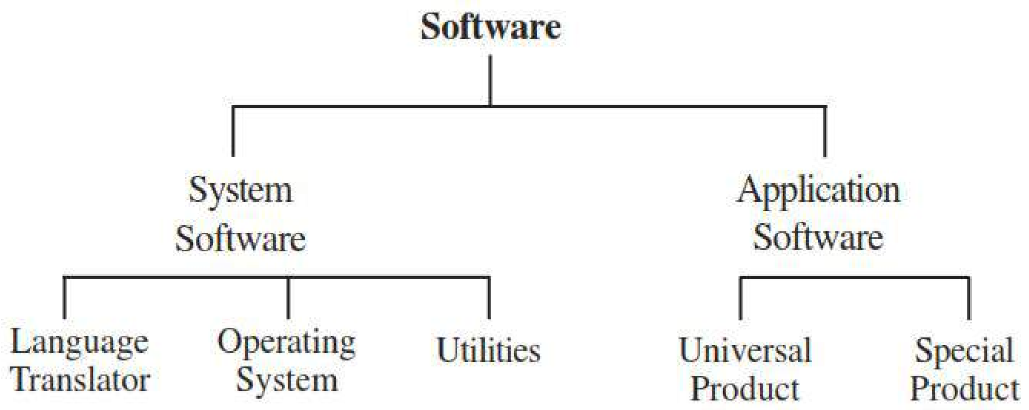
- Name any three names of motherboard manufacturers.
- The computer port used to replace serial mouse and keyboard is..............

## 9.10 Computer Software

A software is a set of programs, which are a set of instructions. These instructions are written in a special computer language that a computer can understand. These languages are known as Programming Languages.

A computer can neither think nor take any decision of its own. Also it is impossible for any computer to independently analyze the given data and follow its own method of solution. It needs a program to tell what to do with the data.

A computer software is classified in to two parts:




### System software

A system software consists of many programs for controlling the input/output operations. An operating system is an example of system software. A system software is further classified in to three parts

- **Language Translator:-** A program that convert programming source code to machine readable codes are known as language translator. There are three types of language translators.

1. **Interpreter**: This is a program that converts high level language to machine level language i.e. Machine language. The basic property of an Interpreter is that it first scans one line of a program or source code, if this is error free then it executes the program or it will stop the execution. So an interpreter will check a program line by line and execute it, if it is error free. This process takes more time in the execution of any program.

2. **Compiler**: A compiler is a program used to convert high-level language to machine level. The basic property of a compiler is that it first scans the entire file at a time and checks for any error, if no error is found then it changes the program to the machine level. Otherwise, it shows all the errors present in the program. So it takes very less time for execution.

3. **Assembler**: An assembler is a program written to convert assembly level language to machine level language.

- **Operating system:-**An operating system is a system software which is used to operate the computer. An operating system manages computer resources very efficiently, takes care of scheduling multiple jobs for execution and manages the flow of data and instructions between input/ output unit and the main memory. Windows, Unix, Linux, Macintosh etc. are few widely used operating systems. An operating system is classified into different categories based on the performance.

  1. *Single user operating system*: A single user operating system gives permission to run or execute one application or program at a time. That is one user can work at a time. e.g.- MS-DOS.

  2. *Multi-user Operating system*: A multi-user operating system gives permission to many users to work at same time. A transaction process system such as railway reservation system that needs hundreds of terminals under a single program is an example of multi-user operating system. e.g. Unix, Linux etc.

  3. *Network operating system*: A network operating system is a collection of software's which allow a set of computers which are interconnected by a computer network to be used together in a convenient and cost effective manner. In a network operating system the users are aware of the existence of multiple computers and so one can log onto a remote machine and copy files from one location to another like Unix, Windows-NT, Linux etc.

4. *Graphical User Interface (GUI) operating system*:- A GUI uses graphical components like small images and pictures to represent a program, so that instead of typing it we just select it using pointing devices like mouse etc. Eg. Windows 3.1, Windows-95/98/2000/ ME/XP, Linux etc.

- **Utilities:-** Utility program are the programs which are often used by application programs. These utility programs are created by the manufacturer. Ex. Text Editors, Sorting, Formatting etc.

## Application Software

Application software is written to enable the computer to solve a specific data processing task. It is used for solving various works. A program written for specific purpose could be termed as an application software. Example-Word processor, Database management system software, accounting software etc. Application softwares can be classified in to different classes like

1. **Database Management Software:-** Database management softwares are used for maintaining data, queries and reports. These softwares provide the facility for easy creation of databases of invoice, order and contact files. Some examples of database softwares are Dbase, FoxPro, Oracle, SQL etc.

2. **Accounting Packages:-** The accounting packages are important packages for an office. Few facilities available in an accounting package like Tally, Tata EX etc., are:
   - Tax Planner facility
   - Facility for producing charts and graphs.
   - Finding accounts payable
   - Payroll function etc.

3. **Desktop Publishing Packages:-** Desktop publishing packages are getting more popular in India. These are used for creating formats for brochures, newsletters etc. Some examples of these packages are PageMaker, Corel Draw, Microsoft Publisher etc.

4. **Word Processor:-** It is a software package that helps to create and edit a document. Editing a document involves correcting spelling mistakes, if any copy,paste and deleting words, sentence or paragraphs. Example:- WordStar, Microsoft Word, Word perfect etc.

5. **Spread sheet:-** A worksheet that provide a number of rows and columns for numeric data entry. They are widely used in accounting, scientific and business fields. Example Lotus 1-2-3, Microsoft Excel etc.
6. **Designing & Architecture:-** Auto-CAD ( Auto desk Computer Added Designing) is a package useful for engineering design offices. Using AutoCAD we can draw and store them in magnetic form. We can recall the drawing and plot them using the plotter. AutoCAD allows us to view a drawing in a solid model form like a three dimensional object.

## 9.11 ComputerLanguages

We know that a software is a collection of programs, and each program is written in a specific language. We can use any language to write a program, but the computer can execute it after changing these codes to binary format (i.e. 0 or 1) with the help of a language translator.

The languages used for writing programs are as follows:-

### Machine Language

This is the sequence of instructions written in binary numbers consisting of 0 or 1. The computer can read this code directly, we don't need to use any language translator. This language is also known as first generation language. A program in this language is difficult to understand. A user cannot communicate with a computer if he doesn't know this language. Eg. 0010001001.

### Assembly Language

In machine language,it is difficult to debug any program by an external user. In assembly language, special symbols are used as instruction codes. This language is known as second generation language. Writing a program in assembly language is more convenient than the machine language. It is written in the form of symbolic instructions. But it is specific for a particular machine. The codes written for any application on any computer can't be run in another computer if it is architecturally different.

### High level language

It is a programming language that uses grammatical and mathematical notations similar to everyday language. Writing program in this language is quite easier than that in assembly or machine language. For converting this language to machine readable format we need a compiler or Interpreter. High level languages are also known as third generation languages. Examples are given below.

**BASIC**:- Beginner's All purpose Symbolic Instruction Code - used for general purpose programs.

**FORTRAN:-** FORmula TRANslation - used for scientific purpose.

**COBOL:-** COmmon Business Oriented Language- used for business oriented programs.

**C**- It is a high level language. It runs on DOS and Unix operating system. Now a days it is most widely used for preparing Operating Systems.

### Fourth generation language (4GLs)

This language is mainly database oriented language. In this generation of language a programmer need not write much codes. Most people feel that a fourth generation language is a high level language that requires fewer instructions for a particular task compared to a third generation language. Fourth generation languages are mostly machine independent. Usually they can be used in more than one type of computer. They are mostly used for office automation or business purposes.

#### Know your progress

1. The language translator which converts assembly language to machine language is....................
2. Give examples for utility softwares.

### Let us consolidate

Central Processing Unit is the brain of the computer. Key board, mouse, joystick, trackball, scanner, digitizer, MICR, OCR, barcode reader, OMR etc. are various input devices of a computer. Monitors, printers etc. are output devices. A flat panel display monitor has reduced volume, weight and power requirements compared to CRT. Dotmatrix printer is an impact printer. Laser and inkjet printers are non impact printers. Laser printers are of very high speed and very high quality. The memory of a computer is divided into cells. The storage capacity is expressed in terms of Bytes. KB, MB, GB, TB etc. are higher units of data. Cache memory is a high speed memory which acts as a buffer between CPU and main memory. Main memory or primary memory is divided into RAM and ROM. RAM is the internal memory of CPU for storing data temporarily. Static and dynamic RAM's are the two types of RAM's. Static RAM uses transistors only and no capacitors. DRAM is used for system memory because it is cheap and small. Secondary memory are magnetic and optical memory devices. Floppy disks and hard disks are magnetic storage devices. Optical disk is the one in which data is read and written by using a low powered laser beam. CDROM, DVD, flash drive etc. are optical storage devices. Computers can be classified by their speed and computing power as personal computers, work station, mini computers, mainframe computers and supercomputers. A mother board serves to connect all the parts of a computer with one another. Computer ports are the connecting points or interfaces with peripherals. These help the peripherals to communicate with the computer. Machine language, Assembly language and High level language are the languages used for writing programs. Machine language uses binary numbers, assembly language uses special symbols and high level language uses notations similar to everyday language for writing programs. Computer software is classified into system software and application software. Language translators such as interpreter, compiler and assembler, operating systems such as windows, linux etc. and utility programs come under system software. Application software is written for specific processing task. Database management software, accounting packages, DTP packages etc. are application softwares.

All the concepts and learning out comes of this chapter were attained through general discussions, demonstration of computer hardware components, ICT enabled learning and data collections.

## Let us asses

1. Classify the following into input, output and memory devices.

   Scanner, RAM, VDU, joystick, CDROM, floppy disk, ROM, flash drive, laser printer, microphone, loud speaker.

2. A) Select a nonimpact printer from the following

   a) Dotmatrix printer b) Daisy wheel printer c) Drum printer d) Laser printer

B) What are the advantages of laser printer compared to other printing technologies?

3. A) 1 Kilo Byte represents

a) 8 byte b) 1000 byte c) 64 byte d) 1024 byte

B) A group of 4 bits is called.....................

4. A) ................is a primary memory

a) CDROM b) RAM c) Hard disk d) Cache memory

B) Distinguish between static RAM and dynamic RAM.

C) What are the advantages of cache memory ?

5. A) All parts of a computer such as CPU, memory, hard disk etc. are connected through...........of the computer.

a) Flash drive b) Motherboard c) Microprocessor d) USB port

B) Write the names of any two branded motherboards.

6. Give a brief comparison of personal, mini and mainframe computers.

7. A) A computer language in which instructions are written in binary numbers is

a) High level language b) Machine language c) Assembly language d) Interpreter

B) Expand BASIC, FORTRAN and COBOL

C) Distinguish between compiler and interpreter.

8. A) MS DOS is a

a) Single user operating system b) Multi user operating system

c) Network operating system d) GUI operating system

B) Give two examples each for Network operating system and Graphical user interface operating system.

C) What are application softwares ? Give examples.

9. A) What are computer ports ?

B) What are the uses of HDMI and VDI ports ?

10. A) An example of optical disk storage device is

a) Hard disk b) Magnetic disk c) Floppy disk d) CDROM

B) Mention the characteristics of any two optical disk storage devices.



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# 10 INTERNET TECHNOLOGY

## Significant Learning Outcomes

*After completing this chapter the learner:*

- explains the needs and advantages of computer networking
- differentiates different network protocols
- compares different data communication devices
- explains the functions of MODEM
- identifies different network topologies
- differentiates different types of networks
- identifies domain names and IP addresses
- explains the concept of client/ server computing

Now a days all of us are familiar with internet, and its applications have penetrated into every field of life. We use e-mail for sending letters, photographs etc. Booking of train tickets, sending applications for admissions, jobs etc. are being done by most of the people using internet. Even online exams have become a common practice for the students. Have you heard of online shopping websites such as Flip Cart, Amazon etc.? Through such websites we can purchase goods and services using internet. So internet is a worldwide network of computers. It is a very good example to understand the concept of networking.In this chapter we will learn about different aspects of networking and the basic concepts of network.

## 10.1 Computer networking

A computer network is a group of computer systems and other computing hardware devices that are linked together through communication channels to facilitate communication and resource-sharing among a wide range of users. Networks are commonly categorized based on their characteristics.

One of the earliest examples of a computer network was a network of communicating computers that functioned as part of the U.S.

military's Semi-Automatic Ground Environment (SAGE) radar system. In 1969, the University of California at Los Angeles, the Stanford Research Institute, the University of California at Santa Barbara and the University of Utah were connected as part of the Advanced Research Projects Agency Network (ARPANET) project. It was this network that evolved to become what we now call the Internet.

Functions of computer networks are to:

- facilitate communication via email, video conferencing, instant messaging, etc.
- enable multiple users to share a single hardware device like a printer or scanner.
- enable file sharing across the network.
- allow the sharing of software or operating programs on remote systems.
- make information easier to access and maintain among network users.

**Advantages of networking**

In the beginning of this chapter it was mentioned that the internet is a very good example of a computer network. We all know about the unlimited services that we get from the internet. Now we cannot imagine a situation without e-mail, online services, social networking sites etc. because with the use of internet all information are at our fingertips. So we are aware of the advantages of networking. Some of the general advantages of networking can be listed as follows.

- **File Sharing:** Networks offer a quick and easy way to share files directly. Instead of using a disk or USB to carry files from one computer or office to another, you can share files directly using a network.
- **Software Cost and Management:** Many popular software products are available for networks at a substantial savings in comparison to buying individually licensed copies for all of your computers. You can also load software on the file server which saves time compared to installing and tracking files on independent computers. Upgrades are also easier because changes only have to be done on the file server instead of the individual workstations.
- **Security:** Specific directories can be password protected to limit access to authorized users. Also, files and programs on a network can be

designated as "copy inhibit". So you don't have to worry about the illegal copying of programs.

- **Resource Sharing:** All computers in the network can share resources such as printers, fax machines, modems, and scanners.
- **Communication:** Even if internet is not available, those on the network can communicate with each other via electronic mail over the network system. When connected to the internet, network users can communicate with people around the world via the network.
- **Flexible Access:** Networks allow their users to access files from the computers throughout the network. This means that a user can begin work on a project on one computer and finish up on another. Multiple users can also collaborate on the same project through the network.

**Know your progress**

1. The most popular computer network is......
2. What are the functions of a computer network ?

## 10.2 Network Protocols

When two humans converse, they may have to use the same language but they generally understand each other without having to adhere to rigid rules of grammar or formal language frameworks. Computers, on the other hand, have to have everything explicitly defined and structured. If computers wish to communicate with one another, they have to know in advance exactly how information is to be exchanged and precisely what the format will have to be. Therefore, standard methods of transmitting and processing various kinds of information are used and these methods are called "protocols". Protocols are established by international agreement and ensure that computers everywhere can talk to one another. There are a variety of protocols for different kinds of information and functions. Let us discuss some of the common protocols that the average PC user is likely to encounter.

### TCP/IP

TCP/IP stands for Transmission Control Protocol/Internet Protocol.TCP (Transmission Control Protocol) and IP (Internet Protocol) are two different procedures that are often linked together. The linking of several protocols is common since the functions of different protocols can be complementary so that together they carry out a complete task. The combination of several protocols to carry out a particular task is often called a 'stack' because it has layers of operations. In fact, the term 'TCP/IP' is normally used to refer to a whole suite of protocols, each with different functions. This suite of protocols is what carries out the basic operations of the web. TCP/IP is also used in many local area networks. The details of how the web works are beyond the scope of this text book and so we will discuss some of the basics of this very important group of protocols.

When information is sent over the internet, it is generally broken up into smaller pieces or 'packets'. The use of packets facilitates speedy transmission, since different parts of a message can be sent through different routes and then reassembled at the destination. It is also a safety measure to minimize the chances of losing information in the transmission process. TCP is the means for creating the packets, putting them back together in the correct order at the end, and checking to make sure that no packet is lost during transmission. If necessary, TCP will request that a packet be resent.

Internet Protocol (IP) is the method used to route information to the proper address. Every computer on the Internet has to have its own unique address known as the IP address. Every packet sent will contain an IP address showing where it is supposed to go. A packet may go through a number of computer routers before arriving at its final destination and IP controls the process of getting everything to the designated computer. Note that IP does not make physical connections between computers but relies on TCP for this function. IP is also used in conjunction with other protocols that create connections.

**Hypertext Transfer Protocol**

Web pages are constructed using the language called Hypertext Markup Language (HTML). An HTML page is transmitted over the web in a standard way and format known as Hypertext Transfer Protocol (HTTP). This protocol uses TCP/IP to manage the Web transmission.

A related protocol is 'Hypertext Transfer Protocol over Secure Socket Layer' (HTTPS) which was first introduced by Netscape. It provides the transmission in encrypted form to provide security for sensitive data. A web page using this protocol will have *https:* in front of its URL.

**File Transfer Protocol**

File Transfer Protocol (FTP) as its name indicates, provides a method for copying files over a network from one computer to another. More generally, it provides some simple file management on the contents of a remote computer.

It is an old protocol which was used frequently before the introduction of world wide web and now its usage is comparatively less. Today, its primary use is to upload files to a website. It can also be used for downloading from the web. Sites that have a lot of data for downloading (software sites, for example) will often have an FTP server to handle the traffic. If FTP is involved, the URL will have *ftp:* at the front.

**Know your progress**

1. The whole suit of protocols that carries out the basic operation of the web is ............
2. An HTML page is transmitted over the web page using............protocol.

## 10.3 Network Topologies

A Network topology is the way in which the computer systems or network equipment are connected with each other. Topologies may define both physical and logical aspects of the network. Both logical and physical topologies could be the same or different in a same network.

### Point-to-point

Point-to-point networks contain exactly two hosts (computer or switches or routers or servers) connected back to back using a single piece of cable. Often, the receiving end of one host is connected to sending end of the other end and vice-versa. A computer to printer data transmission serves as point to point topology.

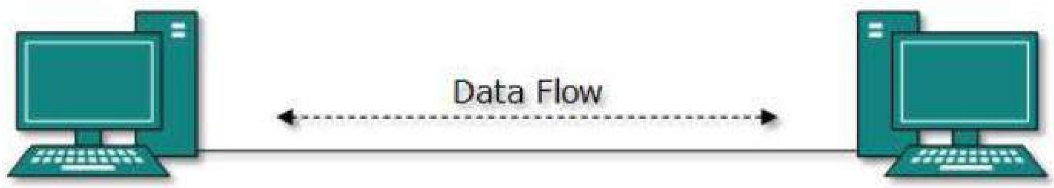


Fig 10.1 : *Image: Point-to-point Topology*

### Bus Topology

In contrast to point-to-point, in bus topology all the devices share single communication line or cable. All the devices are connected to this shared line. Bus topology may have problem when more than one host send data at the same time. Therefore, the bus topology recognizes one host as Bus Master to solve the issue. It is one of the simple forms of networking where the failure of a device does not affect others. But failure of the shared communication line make all other devices fail.

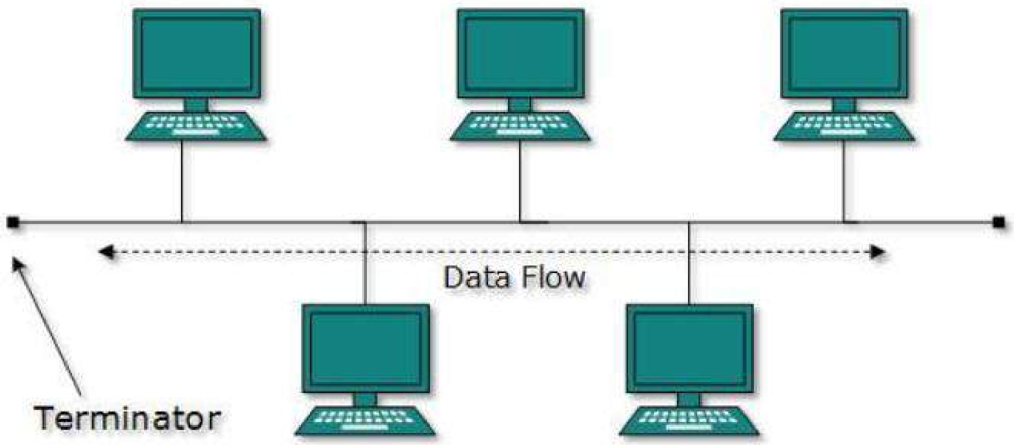


Fig 10.2 : *Bus topology*

Both ends of the shared channel have a line terminator. The data is sent only in one direction and as soon as it reaches the extreme end, the terminator removes the data from the line.

## Star Topology

All hosts in star topology are connected to a central device, known as Hub device, using a point-to-point connection. That is, there exists a point to point connection between hosts and Hub.

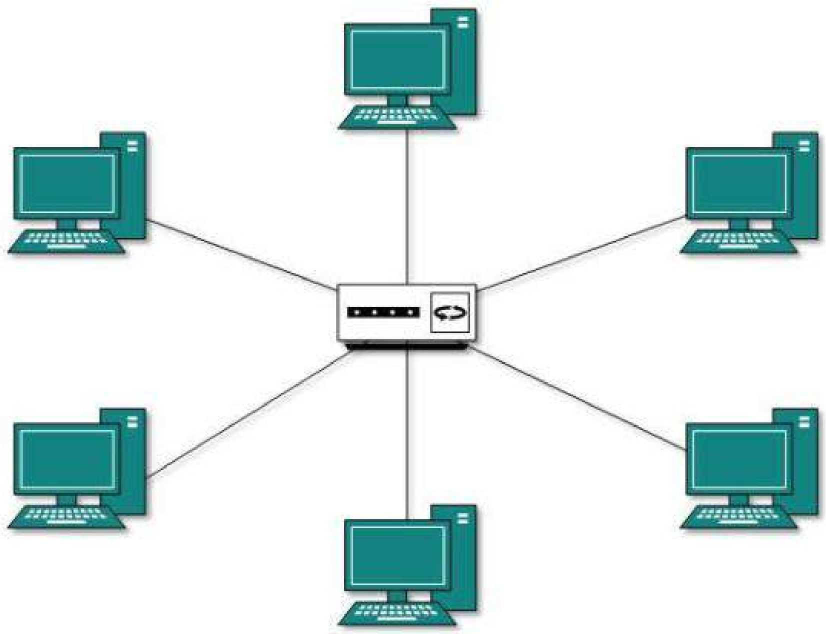

Fig 10.3 : *Star topology*



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Hub acts as a single point of failure. If hub fails, connectivity of all hosts to all the other hosts fails. All the communication between the hosts goes through the hub only. Star topology is not expensive as in order to connect one more host, only one cable is required and the configuration is simple.

### Ring Topology

In ring topology, each host machine connects exactly to two other machines, creating a circular network structure. When one host tries to communicate or send message to a host which is not adjacent to it, the data travels through all the intermediate hosts. To connect one more host in the existing structure administrator may need only one more extra cable.

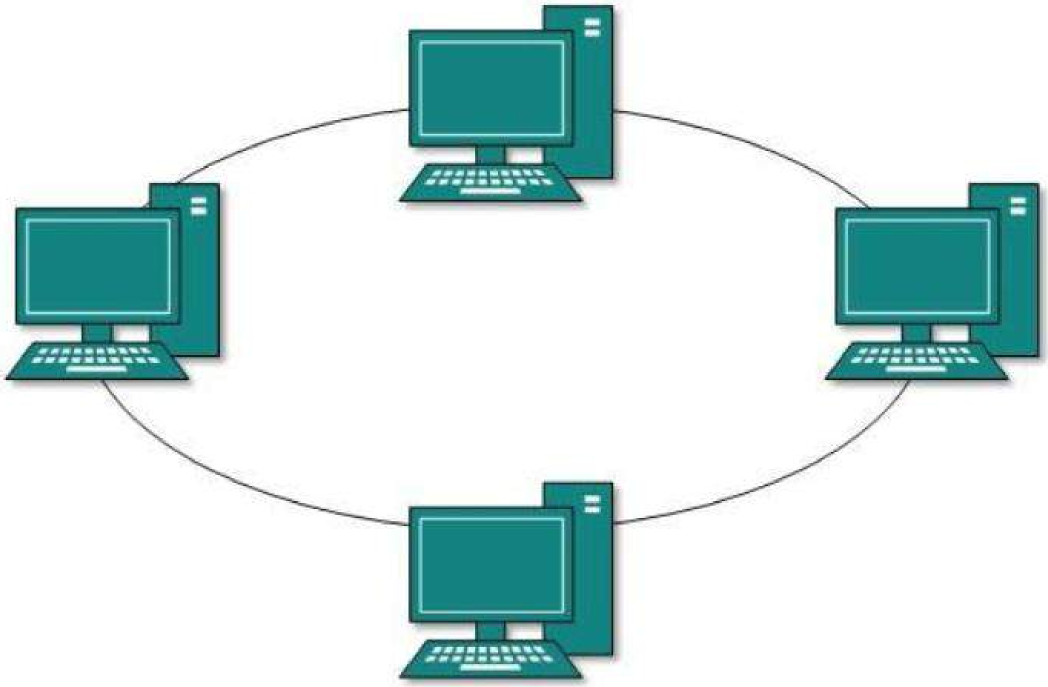

Fig 10.4 : *Ring topology*

Failure of any host results in the failure of the whole ring. Thus every connection in the ring is a point of failure. There exist methods which employ one more backup ring.

### Mesh Topology

In this type of topology, a host is connected to one or two or more than two hosts. This topology may have hosts having point-to-point connection to every other host or may also have hosts which are having point to point connection to a few hosts only.

Hosts in Mesh topology also work as relay for other hosts which do not have direct point-to-point links. Mesh technology comes into two flavors:

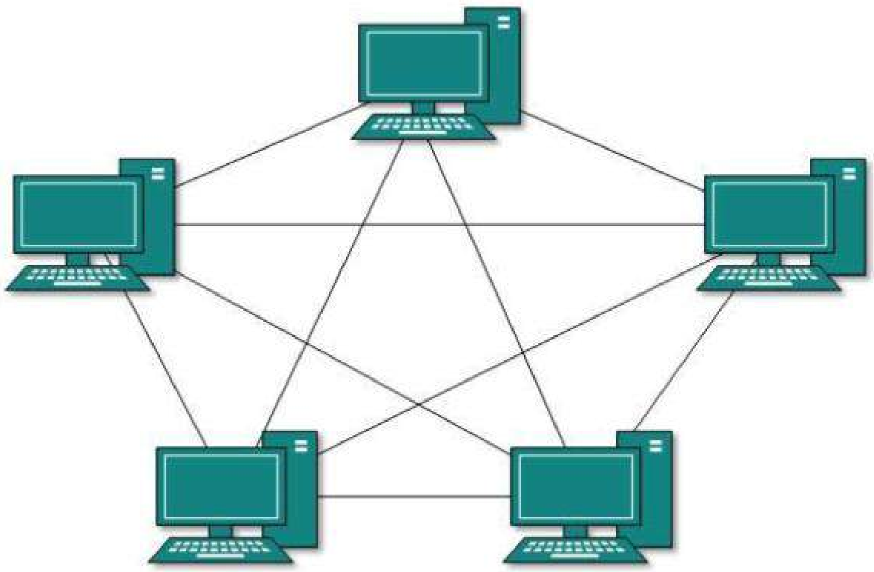

Fig 10.5 : *Mesh topology*

- **Full Mesh**: All hosts have a point-to-point connection to every other host in the network. Thus for every new host n(n-1)/2 cables (connections) are required. It provides the most reliable network structure among all the network topologies.
- **Partially Mesh**: Here all the hosts do not have point-to-point connection to every other host. Hosts connect with each other in some arbitrary fashion. This topology exists where there is we need to provide reliability to some host when it is not much necessary for others.

## Tree Topology

Tree topology which is also known as Hierarchical Topology is the most common form of network topology in use now. This topology imitates extended Star topology and inherits properties of Bus topology.

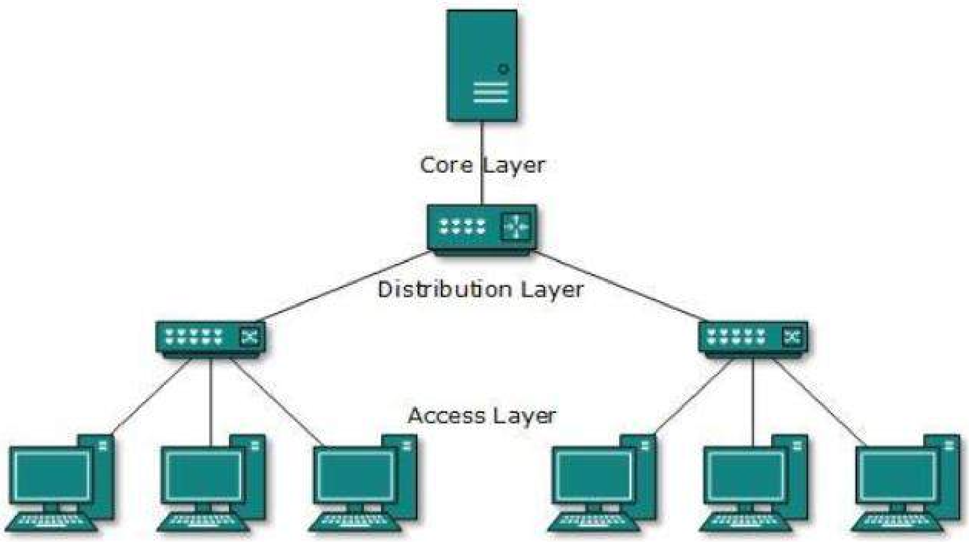


Fig 10.6 : *Tree topology*



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This topology divides the network in to multiple levels/layers of network. Mainly in LANs, a network is bifurcated into three types of network devices. The lowest is access-layer where users' computers are attached. The middle layer is known as distribution layer, which works as the mediator between upper and lower layer. The highest layer is known as Core layer, and is the central point of the network, i.e. root of the tree from which all nodes work.

All neighboring hosts have point-to-point connection between them. Like bus topology, if the root goes down, the entire network suffers. Every connection serves as a point of failure. The failing connection divides the network into unreachable segment and so on.

**Know your progress**

1. What do you mean by a network topology ?
2. Compare different network topologies.

## 10.4 Data Communication Devices

A data communication device provides an interface between computer and the communication channel. These devices are used to transmit, receive, amplify and route data signals across a network through various communication media.

### Network Interface Card

A Network Interface Card (NIC) is a computer hardware component that allows a computer to connect to a network. It can prepare, send,receive and control data on the network. It splits data into manageable units, translates the protocols of the computer to that of the communication medium and supplies address recognition capabilities. NICs may be used for both wired and wireless connections.

A NIC is known as network interface controller (NIC), network interface controller card, expansion card, computer circuit board, network card, LAN card, network adapter or network adapter card (NAC).

Most of the new computers have ethernet capabilities integrated into the motherboard chipset. A separate NIC is generally no longer needed. Wireless network cards or Wi Fi NIC's can transmit data at a speed of 1Gbps.

Typically, there is an LED next to the connector informing the user whether the network is active or whether the data is being transferred on to it. Depending on the card or motherboard, transfer rates may be 10, 100, or 1000 Megabytes per second.



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Hubs, Bridges, Switches and Routers are used to build networks. If you are trying to design your own LAN (Local Area Network) at home, then you probably need to know what they do and the main differences between them. Meanwhile we all have to remember that the internet is nothing more than a network of networks!

## Hub

**Hubs** are used to build a LAN by connecting different computers in a star/ hierarchal network topology. A hub is a very simple (or dumb) device, once it gets bits of data sent from computer A to B, it does not check the destination, instead, it forwards that signal to all other computers (B, C, D…) within the network. B will then pick it up while the other nodes discard it.

*There are mainly two types of hubs:*

1. Passive: The signal is forwarded as it is (so it doesn't need power supply).
2. Active: The signal is amplified, so they work as repeaters. In fact they have been called as multiport repeaters. (use power supply)

Hubs can be connected to other hubs using an uplink port to extend the network.

## Switches

**Switches** are more advanced than hub. Instead of broadcasting the frames everywhere, a switch actually checks for the destination MAC address and forward it to the relevant port to reach that computer only. This way, switches reduce traffic and divide the collision domain into segments. This is very sufficient for busy LANs and it also protects frames from being sniffed by other computers sharing the same segment. MAC address means Media Access Control address which is a unique identifier assigned to network interfaces for communications on the physical network segment. So MAC addresses are used as network addresses.

The switching mechanism build a table indicating which MAC address belongs to which segment. If a destination MAC address is not in the table the frame is forwarded to all segments except the source segment. If the destination is same as the source, frame is discarded.

Switches have built-in hardware chips solely designed to perform switching capabilities, therefore they are fast and include many ports. Sometimes they are referred to as intelligent bridges or multiport bridges.



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Different speed levels are supported by the switches. They can be 10 Mbps, 100 Mbps, 1 Gbps or more.

Most common switching methods are:

1. Cut-through: Directly forward what the switch gets.
2. Store and forward: receive the full frame before retransmitting it.

## Repeaters

As signals travel along a network cable (or any other medium of transmission), they degrade and become distorted in a process is called attenuation. If a cable is long enough, the attenuation will finally make a signal unrecognizable by the receiver.

A repeater enables signals to travel longer distances over a network. A repeater regenerates the received signals and then retransmits the regenerated (or conditioned) signals on other segments.

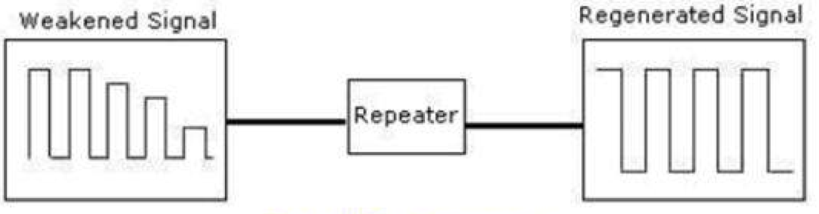


Fig 10.7 : *Repeater*

To pass data through the repeater in a usable fashion from one segment to the next, the packets and the Logical Link Control (LLC) protocols must be the same on each segment. This means that a repeater will not enable communication between segments with different protocols because repeaters do not translate anything.

## Bridges

Like a repeater, a bridge can join segments or workgroup LANs. However, a bridge can also divide a network to isolate traffic or problems. For example, if the volume of traffic from one or two computers or a single compartment is flooding the network with data and slowing down the entire operation, a bridge can isolate those computers or that compartment.

In the following figure, a bridge is used to connect two segments, say segment 1 and segment 2.

Bridges can be used to:

i. Expand the distance of a segment.
ii. Provide an increased number of computers on the network.

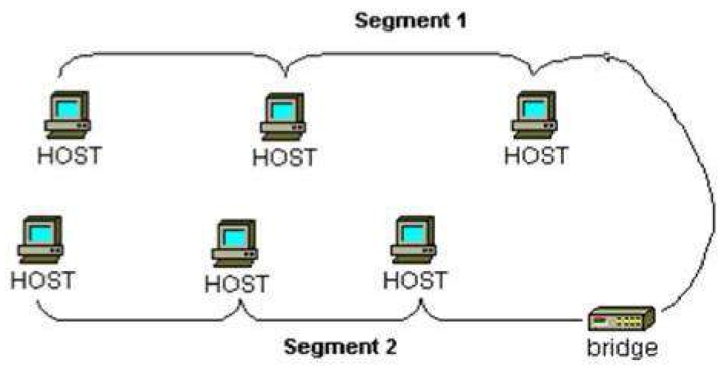


Fig 10.8 : *Bridge*

iii. Reduce traffic bottlenecks resulting from an excessive number of attached computers.

A bridge works on the principle that each network node has its own address. A bridge forwards the packets based on the address of the particular destination node.

As traffic passes through the bridge, information about the computer addresses are stored in the bridge's RAM. The bridge will then use this RAM to build a routing table based on source addresses.

## Routers

In an environment consisting of several network segments with different protocols and architecture, a bridge may not be adequate for ensuring fast communication among all of the segments. A complex network needs a device, which not only knows the address of each segment, but also can determine the best path for sending data and filtering broadcast traffic to the local segment. Such device is called a Router. Routers can switch and route packets across multiple networks. They do this by exchanging protocol-specific information between separate networks. Routers have access to more information in packets than bridges, and use this information to improve packet deliveries. Routers are usually used in a complex network situation because they provide better traffic management than bridges.

Routers can share status and routing information with one another and use this information to bypass slow or malfunctioning connections.

Routers do not look at the destination node address; they only look at the network address. Routers will only pass the information if the network address is known. This ability to control the data passing through the router reduces the amount of traffic between networks and allows routers to use these links more efficiently than bridge.

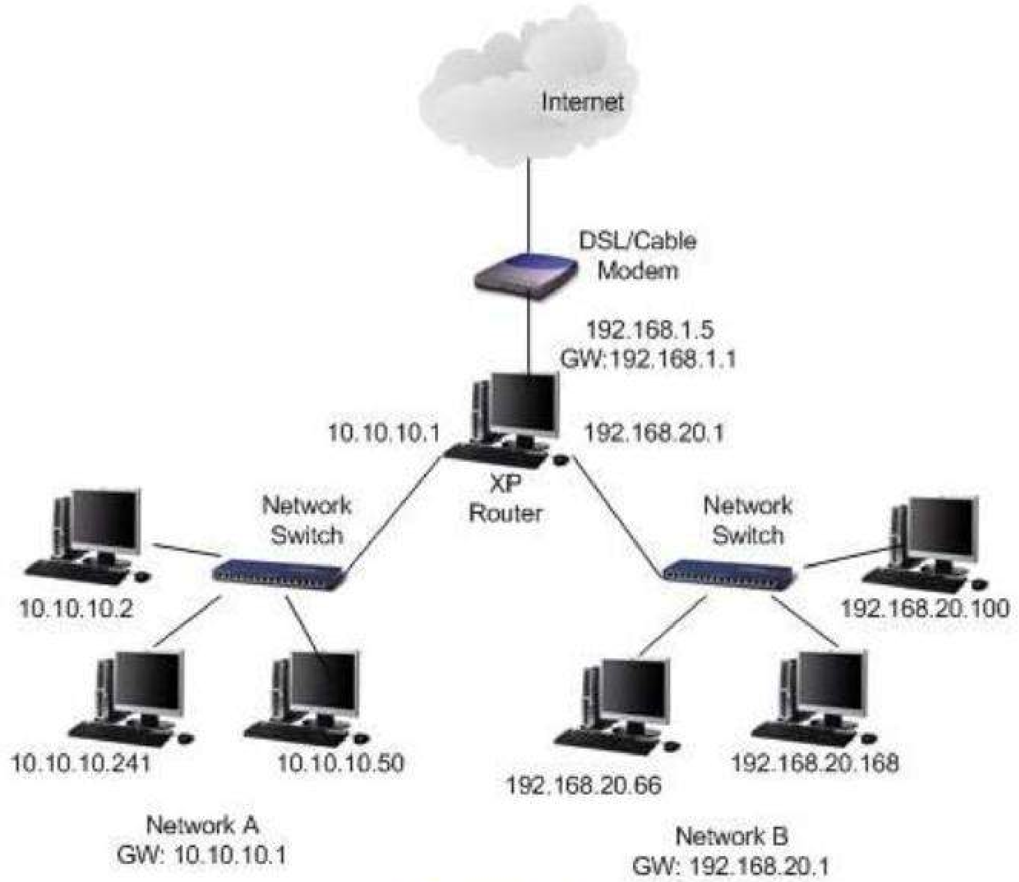


Fig 10.9 : *Router*

## Gateways

Gateways make communication possible between different architectures and environments. They repackage and convert data going from one environment to another so that each environment can understand the other's environment data.

A gateway repackages information to match the requirements of the destination system. Gateways can change the format of a message so that it will conform to the application program at the receiving end of the transfer.

A gateway links two systems that do not use the same:

i. Communication protocols

ii. Data formatting structures

iii. Languages

iv. Architecture

For example, electronic mail gateways, such as X.400 gateway, receive messages in one format, and then translate it, and forward in X.400 format used by the receiver, and vice versa.

To process the data, the gateway decapsulates the incoming data through the network's complete protocol stack. Then it encapsulates the outgoing data in the complete protocol stack of the other network to allow transmission.

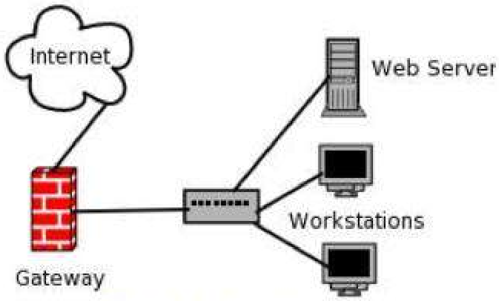


Fig 10.10 : *Gateway*

## MODEM

Modems are used to transmit digital information via analog systems. The word 'modem' is derived from the term 'modulator-demodulator.' The essential functions of a modem are to modulate an analog carrier signal to carry digital information; and to demodulate a similar signal so as to decode the digital information from the analog carrier signal. So the term MODEM was formed from these two words modulator and demodulator.

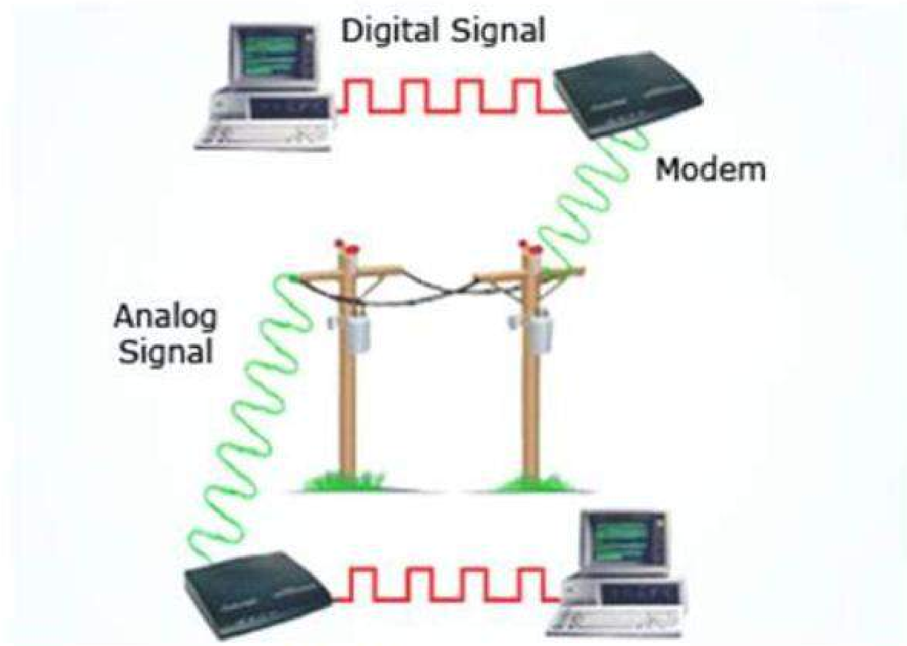


Fig 10.11 : *Modem*

### Functions of a MODEM

The essential function of a modem is to convert digital data into analog form so as to make it possible to be transmitted through analog transmitting channels without the loss of information and similarly to decode the analog signals transmitted into digital form at the receiving end.The most familiar use of modems is to send information over a telephone channel, but modems can be used to relay data over any system that provides a means of transmitting analog signals, including radio and optical networks.

### Data Compression

To reduce the time taken to send data and to cut down the amount of error in the signal, modems need to employ data compression. This was especially necessary in the early days of modem technology, since data had to be sent via conventional phone lines. Not being designed for digital information, phone lines placed heavy limitations on the size and speed of signals sent over them. Data compression techniques reduce the size of the signal needed to send the required data.

### Error Correction

When information is transmitted between modems, it can be sometimes damaged, which means that some parts of the data are altered or lost. To get over this, modems use error correction. Information is grouped into batches, called frames. Each frame is tagged with a checksum, a small piece of data derived from the information in the frame. A checksum can be thought of as a kind of fingerprint, unique to the data in a particular frame. The modem that receives the information derives its own checksum from the frame it has been sent, then compares its checksum data with the checksum sent by the transmitting modem. If the checksums match, the information is undamaged. If they don't match, the data has been corrupted in transmission; the recieving modem sends it back and waits for the transmitting modem to re-send that frame.

### Flow Control

Individual modems send information at different speeds. It's necessary for faster modems to slow down so that slower modems can catch up, otherwise the slower modem will receive more data than it can process. If this starts to happen, the slower modem transmits a character to the faster one. This character is a signal for the fast modem to pause in sending information until the slow modem gets caught up. When the slow modem is ready for more data, it sends a different character that signals to the fast modem that it can start transmitting again. In this way, the two modems can match their speeds.

### Modem Speed Classification

The speed of a modem is typically classified by the amount of data it can send in a specific length of time. This is generally expressed in terms of bits per second (bps). An alternative way of classifying modem speed is the change in the state of the signal per unit time, that is the number of times a modem sends a new signal in a given length of time. This is known as the symbol rate and is measured in units called baud (Bd).

### Internal vs. External Modems

An external modem is a discrete unit housed in a separate case. Typically, an external modem will be connected to the telephone line and the computer via cables. Internal modems are circuit boards that plug into a computer's motherboard. Modems can be dial-up or wireless (Wi-Fi). Dial-ups uses the telephone network to send and receive signals. They require authentication to connect. Dial-up is markedly slower than other kinds of modem connection. As of the date of publication, the fastest listed speed for a dial-up modem is 56.6 kbps.

### Broadband: Cable and ADSL Modems

Cable modems use the same radio frequency range as cable television. Cable modems have the advantage of using the existing cable television infrastructure, allowing cable TV companies to provide Internet services. Asymmetric Digital Subscriber Line (ADSL) modems use telephone lines to send and receive data but make use of a different frequency band than modems using the voiceband range of frequencies. ADSL modems are much faster than conventional voiceband modems. ADSL and cable modems are used to provide broadband Internet, which allows more data to be transmitted and thus makes using the Internet faster.

**Know your progress**

1. Name a data communication device which is capable of linking two systems with different protocols and architecture.
2. What are the functions of MODEM ?
3. Name two types of MODEM's used to provide broadband internet.

## 10.5 Types of networks

There are several different types of computer networks. Computer networks can be characterized by their size as well as their purpose.

The size of a network can be expressed by the geographic area they occupy and the number of computers that are part of the network. Networks can cover anything from a handful of devices within a single room to millions of devices spread across the entire globe.

Some of the different networks based on size are:

- Personal area network, or PAN
- Local area network, or LAN
- Metropolitan area network, or MAN
- Wide area network, or WAN

In terms of purpose, many networks can be considered for general purpose, which means they are used for everything from sending files to a printer to accessing the Internet. Some types of networks, however, serve a very particular purpose. Some of the different networks based on their main purpose are:

- Storage area network, or SAN
- Enterprise private network, or EPN
- Virtual private network, or VPN

Let's look at each of these in detail.

## Personal Area Network

A **personal area network**, or **PAN**, is a computer network organized around an individual person within a single building. This could be inside a small office or residence. A typical PAN would include one or more computers, telephones, peripheral devices, video game consoles and other personal entertainment devices.

If multiple individuals use the same network within a residence, the network is sometimes referred to as a home area network, or HAN. In a very typical setup, a residence will have a single wired Internet connection connected to a modem. This modem then provides both wired and wireless connections for multiple devices. The network is typically managed from a single computer but can be accessed from any device.

This type of network provides great flexibility. For example, it allows you to:

- Send a document to the printer in the office upstairs while you are sitting on the couch with your laptop.
- Upload the photo from your cell phone to your desktop computer.
- Watch movies from an online streaming service to your TV.

## Local Area Network

A **local area network**, or **LAN**, consists of a computer network at a single site, typically an individual office building. A LAN is very useful for sharing resources, such as data storage devices and printers. LANs can be built with relatively inexpensive hardware, such as hubs, network adapters and Ethernet cables.

The smallest LAN may use only two computers, while larger LANs can accommodate thousands of computers. A LAN typically relies mostly on wired connections for increased speed and security, but wireless connections can also be part of a LAN. High speed and relatively low cost are the defining characteristics of LAN.

LANs are typically used for single sites where people need to share resources among themselves but not with the rest of the outside world. Think of an office building where everybody should be able to access files on a central server or one should be able to print a document to one or more central printers. With LAN those tasks would be easy for everybody working in the same office. But if you want somebody just walking outside to be able to send a document to the printer from their cell phone a wireless network connection is needed. If a local area network, or LAN, is entirely wireless, it is referred to as a wireless local area network, or WLAN.

## Metropolitan Area Network

A **metropolitan area network**, or **MAN**, consists of a computer network across an entire city, college campus or a small region. A MAN is larger than a LAN, which is typically limited to a single building or site. Depending on the configuration, this type of network can cover an area from several miles to tens of miles. A MAN is often used to connect several LANs together to form a bigger network. When this type of network is specifically designed for a college campus, it is sometimes referred to as a campus area network, or CAN.

## Wide Area Network

A **wide area network**, or **WAN**, occupies a very large area, such as an entire country or the entire world. A WAN can contain multiple smaller networks, such as LANs or MANs. The Internet is the best-known example of a public WAN.

**Know your progress**

1. Point out an example of WAN.
2. Distinguish between LAN and WAN.

## 10.6 Domain name systems and IP addresses

Domain names are used to identify one or more *IP addresses*. For example, the domain name *microsoft.com* represents about a dozen IP addresses. Domain names are used in URLs to identify particular Web pages. For example, in the URL*http://www.pcwebopedia.com/index.html,* the domain name is *pcwebopedia.com.*

Every domain name has a suffix that indicates which top level domain (TLD) it belongs to. There are only a limited number of such domains. For example:

- **gov** - Government agencies
- **edu** - Educational institutions
- **org** - Organizations (nonprofit)
- **mil** - Military
- **com** - commercial business
- **net** - Network organizations
- **in** - India
- **ca** - Canada
- **th** - Thailand

Since the Internet is based on IP addresses and not the domain names, every Web server requires a Domain Name System (DNS) server to translate the domain names into IP addresses.

Every machine on a network has a unique identifier. Just as you need an address to send a letter via mail, computers use a unique identifier to send data to specific computers on a network. Most of the networks today, including all the computers on the Internet, use the TCP/IP protocol as the standard for communicating on the network. In the TCP/IP protocol, the unique identifier for a computer is called its IP address.

There are two standards for IP addresses: IP Version 4 (IPv4) and IP Version 6 (IPv6). All computers with IP addresses have an IPv4 address, and many have started to use the new IPv6 address system as well. Here's what these two address types mean:

- IPv4 uses 32 binary bits to create a single unique address on the network. An IPv4 address is expressed by four numbers separated by dots. Each number is the decimal (base-10) representation for an eight-digit binary (base-2) number, also called an octet. For example: 216.27.61.137
- IPv6 uses 128 binary bits to create a single unique address on the network. An IPv6 address is expressed by eight groups of hexadecimal (base-16) numbers separated by colons, as in 2001:cdba:0000:0000:0000:0000: 3257:9652. Groups of numbers that contain all zeros are often omitted to save space, leaving a colon separator to mark the gap (as in 2001:cdba::3257:9652).

**Know your progress**

1. The server which translates domain names to IP addresses is called...............
2. Write a URL and identify each term involved in it.

## Client-Server Computing

Restaurant service is an analogy which help to explain client/server computing. The customer (client)makes a series of requests for a specific set of services that may include an appetizer, meals and an ice cream. These requests are all typically made to one person, the waiter(server). The services may actually be provided by a number of other people in the restaurant including the helper, cleaner, and a variety of chefs. However, to the customer, these services are all provided by one person, the waiter. The customer doesn't want to know who performs what service. He would just like to have a high quality meal delivered in a timely fashion. The client, in client/server computing is much like the customer in a restaurant. The client requests a service, like running an application or accessing some information from a data base. The server becomes responsible for performing the service and returning the information to the client in a timely manner.The server is like the waiter in a restaurant responsible for handling the client's requests and delivering the products to the client.

Client/server computing generally refers to a computing model where two or more computers interact in such a way that one provides services to the other. This model allows customers to access information resources and services located anywhere within the customers information network.

As the term implies, client/server computing has two basic components, a client and a server. The client requests a service to be performed. This service might be to run an application, query a data base, print a document, or even perform a backup or recovery procedure. The server is the resource that handles the client's request. Clients are typically thought of as personal computers but a client can be a midrange system or even a mainframe. Servers are typically thought of as a midrange or mainframe system, however a server can also be another personal computer on the network. Client/server networks are like the example of a restaurant as said above where specific computers provide one or more services to other computers within a network. Today's networks have computers for file serving, data base serving, application serving, and communication serving. Each of these servers is dedicated to provide a specific service to all authorized users within a network.

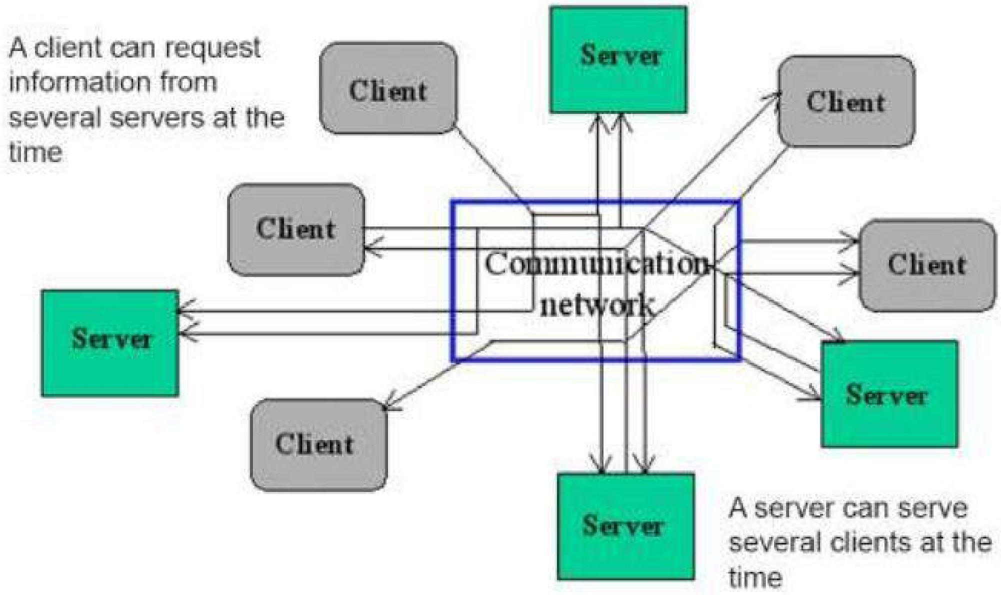


Fig 10.12 : *Client-Server Computing*

## File Servers

These servers focus on file management and are responsible for security, management and availability of stored files. These servers allow connections to the end-user for the purpose of sending and receiving files as per the request of the user.

### Mail Servers

Mail servers provide the services necessary to send, receive and store electronic mail (e-mail) in a local network, in a wide area network or on the Internet.

### Database Servers

The database server manages the database that is stored in that server using database management system. A client request is sent in the form of query to the server. The server in turn searches through the database for the requested information and sends the results to the client.

### Web Servers

Web servers provide access to the internet through the HyperText Transfer Protocol (HTTP). Files in a web server use HyperText Markup Language (HTML) to display content on the web browsers. A web server usually receives requests from a web browser and sends back the requested HTML file and related graphic files.

**Know your progress**

1. Diagrammatically represent and explain the concept of client / server computing.
2. Name the different types of servers.

### Let us consolidate

Computer networks facilitate communication and resource-sharing. File Sharing, resource sharing, software sharing, security, communication and flexible access are some of the advantages of networking. Standard methods of transmitting and processing various kinds of information are called 'protocols'. 'TCP/IP' is used to refer to a whole suite of protocols, each with different functions that carry out the basic operations of the web. An HTML page is transmitted over the web using a protocol known as Hypertext Transfer Protocol (HTTP). File Transfer Protocol (FTP) is used for copying files over a network from one computer to another. A data communication device provides an interface between computer and the communication channel. Repeaters, bridges, routers, gateways and modem are different data communication devices. A modem modulates an analog carrier signal to carry digital information and to demodulate a similar signal so as to decode the digital information from the analog carrier signal. There are internal modems, external modems, cable modems and ADSL modems. A Network Topology is the way the computer systems or network equipment are connected to each other. Computer networks are classified as PAN, LAN, MAN and WAN. Internet is a good example of WAN. Domain names are used in URL to identify particular web pages. Client / server computing refers to a computing model where two or more computers interact in such a way that one provides services to another. File server, mail server, data base server and web servers are various types of servers.

The contents of this chapter were learned through general discussion in the class room, group discussions, demonstration of diagrams, data collection and with the use of ICT.

## Let us asses

1. A) World wide network of computers is called..........

   i) LAN ii) MAN iii) Internet iv) WAN

   B) Mention the functions of computer networks.

2. A hard ware component that allows a computer to connect to a network is called

   a) Ethernet b) Router c) Bridge d) NIC

3. A) What is the working principle of MODEM ? Use necessary diagram.

   B) Explain the functions of MODEM.

4. Distinguish between cable and ADSL modems.

5. A) A single communication line or cable is shared by all devices in..........

   i) Ring topology
   ii) Star topology
   iii) Bus topology
   iv) Mesh topology

   B) Draw diagrams to represent Bus topology and Star topology.

6. A) Based on their size the networks PAN, LAN, MAN and WAN are indicated as shown. Which of these is correct?

a) PAN > LAN >MAN > WAN

b) LAN > PAN > MAN > WAN

c) WAN > MAN > LAN > PAN

d) WAN > MAN > PAN > LAN

B) Compare LAN, MAN, WAN

7. A) Write a URL and identify the domain name included in that URL.

B) The unique identifier of a computer is called.........

a) Domain name

b) Email

c) TCP / IP

d) IP address

8. Expand the following

a) TCP / IP

b) FTP

c) HTTP

d) HTML

9. Explain the client / server computing with the help of a diagram.

10. A) Give the names of four different servers.

B) What are the functions of data base servers and web servers?

# 11 BASICS OF TELEPHONE COMMUNICATION

## Significant Learning Outcomes

*After completing this chapter the learner:*

- explains the need of PSTN
- sketches the structure of PSTN
- identifies different switching offices
- differentiates centralized SPC and distributed SPC
- describes how cell system is a spectrum efficient system
- explains the concept of frequency reuse
- points out the requirements for a multiple access
- differentiates TDMA,FDMA and CDMA
- explains GSM
- explains the working of GPS
- explains GPRS technology

The transmission of information between people who are separated by a distance is known as telecommunication. In olden times, this was usually done by using visual signs such as smoke signals or flags which delivered a particular message or with the use of audio codes through particular drum beats or whistles which had different meanings. These days, electricity is being used a lot in telecommunication, and this has made the process of information transmission faster and possible over longer distances.

In the beginning of this way of telecommunication, wire was necessary for communication to take place. However, wireless communication became possible later on. This has made it possible for people to transmit large volumes of information over longer distances. Telecommunication is a key aspect of the global economy and developments in this sector aid the economic growth of any nation. The basic elements of telecommunication involve a transmitter which converts the message into a signal. The transmitter then transmits the information through a medium to the receiver which converts these signals into information that can be used and understood by people.

There are situations in which telecommunication can take place only in one direction, as in the case of a radio or television. However with the development of telephone technology, duplex telecommunication has become possible. So people can receive as well as transmit information through the same medium.The signals that are transmitted in digital communication can either be analog signals or digital signals. These two types of signals are different in the way in which they are coded as well as the quality of the sound received.

Telecommunication networks on the other hand represent all the systems that make telecommunication possible such as transmitters, receivers and the communication media. The telecommunication medium refers to the channel that carries the signal between a transmitter and a receiver. Examples of telecommunication media in our modern times are optical fibers as well as cables.

Telecommunication has significantly impacted our societies as well as our economies. For businesses, telecommunication was badly needed as this was the only way to transmit useful information about the products or services to potential customers.Telecommunication has affected social relationships because telephone made it easier for people to communicate with each other, regardless of their physical distances. So telecommunication recognizes the human need to communicate with each other, and it attempts to fulfill this need through the transmission of information as fast as possible between different people. Technology has advanced to phenomenal levels, and with the internet being used as a key telecommunication media these days, there is no limit to how far we can go and hence it caused to realize the concept of global village.

## 11.1 Public Switched Telephone Network

Public switched telephone networks are communication systems that are available to the public to allow users to interconnect communication devices. Public telephone networks within countries and regions are standard integrated systems of transmission and switching facilities, signaling processors, and associated support systems that allow communication devices to communicate with each other when they operate.

The public switched telephone network (PSTN) is the aggregate of the world's circuit-switched telephone networks that are operated by national, regional, or local telephony operators, providing infrastructure and services for public telecommunication.

The PSTN consists of telephone lines, fiber optic cables, microwave transmission links, cellular networks, communications satellites, and undersea telephone cables, all interconnected by switching centers, thus allowing any telephone in the world to communicate with any other.

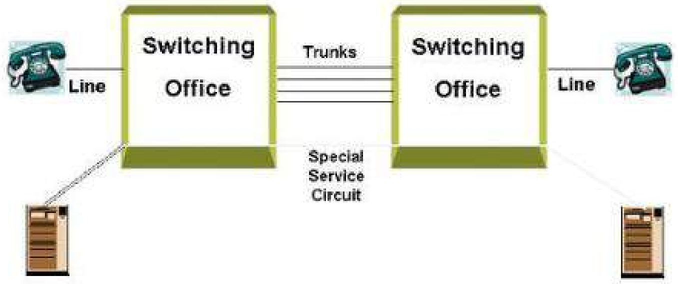


Fig. 11.1 Basic switching method in Telephone communication

In the field of telecommunication, a telephone exchange or telephone switch is a system of electronic components that connects telephone calls. Now let us familiarize the following terms.

- A central office is the physical building or switching office which is used to occupy the equipment including telephone switches which make proper telephone connections and relay speech information.
- The term exchange can also be used to refer to an area consisting of a particular switch.
- Telephone exchanges are connected together with trunks. The trunks are large band with channels having band width more than that of the lines used in local loop.

The basic structure of a PSTN consists of

- Regional offices (class 1)
- Sectional offices(class 2)
- Primary offices (class 3)
- Toll offices (class 4)
- End offices (class 5)

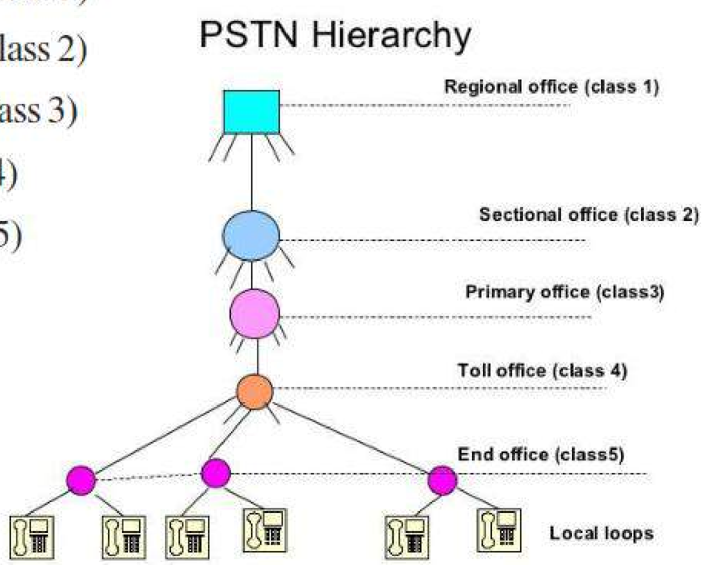


Fig. 11.2 Basic switching method in Telephone communication

The subscriber's telephone sets are connected to the end office using local loops. A small town may have only one end office. Large cities will have several end offices. If a subscriber wants to communicate with another attached to the same end office, the required switching is established within the end office. Many end offices are connected to one Toll office using trunks of more band width than the local loops. The connection between subscribers attached to different end offices will be established within the toll office in which these end offices are attached. Similarly several toll offices are connected to a Sectional office which normally serves more than a state. Several sectional offices are connected to a Regional office. All regional offices are connected using mesh technology.

**Know your progress**

1. *Sketch the block diagram of a PSTN.*
2. *What is meant by a local loop ?*

## 11.2 Electronic Exchange

**Stored Program Control Exchange**

The *Stored Program Control exchange* also written as SPC is the technical name used for the telephone exchanges controlled with the computer program stored in the memory of the system. In early years the exchanges such as the Strowger, the panel, the rotary, and the crossbar switches were electromechanical. Later on, electronic switching system came into existence with the introduction of SPC (Stored Program Control Exchange). The *SPC* allowed more sophisticated calling features. As the *SPC exchanges* evolved, the reliability and the versatility were increased. In 1980s the *SPC* was being completely used all over the industry. The modern digital computers use the *stored program control* concept. The program or the set of the instructions to the computer is stored in the memory and the instructions are executed automatically one by one by the processor. The *exchange* functions through the programs stored in the memory of the computer and hence the system is called the *stored program control system*.

The way the computer use to carry out the *control* functions of the *telephone exchange* is not as simple as using the computer for the commercial data processing. The telephone exchange should work without interruption for 24 hours a day, 365 days a year. It means the exchange must be tolerant to the faults. The *SPC* is the standard feature in all the electronic exchanges.

### Types of SPC

There are basically two types of the *stored program control*:

*i)* *Centralized*

*ii)* *Distributed.*

### The Centralized SPC

In the *centralized control*, all the control equipments are replaced by a single processor which is very powerful. It is able to process 10 to 100 calls per second and depends on the load to the system. The exchange resources and the memory module containing the programs for carrying out various control functions are controlled by the processor.

The dual processor architecture is the one in which two powerful processors are used and one will take over the control when the other fails. This architecture may be configured to operate in one of these modes:

1. The Standby mode
2. The Synchronous duplex mode
3. The Load sharing mode

### The *Distributed Stored program control*

In distributed SPC, the control functions are shared by many processors. Here much powerful processors are not required as in centralized SPC.

The advantages of *distributed stored program control* are

- Better availability
- More reliability

SPC enable sophisticated calling features. As SPC exchanges evolved, reliability and versatility increased.

**Know your progress**

1. *What do you mean by Stored program control system ?*
2. *Distinguish between centralized and distributed SPC's.*

## 11.3 Cellular Communication

Cellular telephone systems are a way of providing portable telephone services. Basically, the mobile phone or a cellular phone is a radio. It uses a radio signal in order to transmit and receive voice and data information. Previously, the radio device could only receive a signal from a commercial station making it a one way communication apparatus. However, by integrating the principles behind Bell's telephone, the simple radio became a communication device which can also serve as a small transmitter thus giving it the capability to become a mobile phone. Mobile phones are small radios embedded with mini transmitters. This means that it actually transmits radio signals when powered on.

The geographical area of service of the system is split up into several smaller regions, each covered by a different transmitter / receiver station.

These regions are conveniently known as cells, and has given rise to the name 'cellular' technology . Diagrammatically these cells are often shown as hexagonal shapes that conveniently fit together. In reality this is not the case. They have irregular boundaries because of the terrain over which they travel. Hills, buildings and other objects all cause the signal to be attenuated and to be diminished differently in each direction.It is also very difficult to define the exact edge of a cell. The signal strength gradually reduces and towards the edge of the cell, performance falls. As the mobiles themselves have different levels of sensitivity, this adds a further greying of the edge of the cell. Therefore it is never possible to have a sharp cut-off between cells. In some areas they may overlap, whereas in others there will be a 'hole' in coverage. In such a hole region no signal will be available from any of the cells.

Each phone is connected by a radio link to a base station. In turn, this is linked to the telephone network. With a cellular system, each base station covers a limited area, and if a phone moves away, the connection is passed across to an adjacent base station. This is called a hand-off, which allows mobility of phones.

Earlier schemes for radio telephones used a single central transmitter to cover a wide area. These radio telephone systems suffered from limited number of channels that were available. Often the waiting lists for connection were many times greater than the number of people who were actually connected. In view of these limitations this form of radio communication technology did not take off in a big way. Equipment was large and these radio communication systems were not convenient to use or carry around.

**The need for a spectrum efficient system**

To illustrate the need for efficient spectrum usage for a radio communications system, take an example where each user is allocated a channel. While more effective systems are now in use, let us take this example as a case of an analogue system. Each channel needs to have a bandwidth of around 25 kHz to enable sufficient audio quality to be carried as well as for including a guard band between adjacent signals to ensure there are no undue levels of interference. Using this concept it is only possible to accommodate 40 users in a frequency band of 1 MHz wide. Even if 100 MHz were allocated to the system this would only enable 4000 users to have access to the system. Today cellular systems have millions of subscribers and therefore a far more efficient method of using the available spectrum is needed.

**Know your progress**

1. *What is a 'Cell" ?*
2. *What is the need of efficient spectrum for radio communication?*

## 11.4 Cell system for Frequency Re-use

Cellular systems are being widely used today and cellular technology needs to offer a very efficient use of the available frequency spectrum. With billions of mobile phones in use around the globe today, it is necessary to re-use the available frequencies many times over without mutual interference of one cell phone with another. It is this concept of frequency re-use that is at the very heart of the cellular technology. However the infrastructure technology needed to support it is not simple, and requires a significant investment to bring the first cellular networks on line.

Now let us see the method that is employed to enable the frequencies to be re-used. Any radio transmitter will be having only a certain coverage area. Beyond this the signal level will fall to a limit below which it cannot be used and will not cause significant interference to the users associated with a different radio transmitter. This means that it is possible to re-use a channel once it is outside the range of the radio transmitter. The same is also true in the reverse direction for the receiver, where it will only be able to receive signals over a given range.

### Cell clusters

When designing the infrastructure technology of a cellular system, the interference between adjacent channels is reduced by allocating different frequency bands or channels to adjacent cells so that their coverage can overlap slightly without causing interference. In this way cells can be grouped together in what is termed a cluster.

It is necessary to limit the interference between cells having the same frequency. The topology of the cell configuration has a great impact on this. The larger the number of cells in the cluster, the greater the distance between cells sharing the same frequencies.

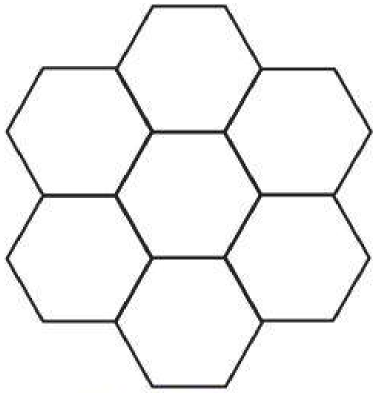
Fig 11.3 Cluster

Often these clusters contain seven cells, but other configurations are also possible. Seven is a convenient number, but there are a number of conflicting requirements that need to be balanced when choosing the number of cells in a cluster for a cellular system which are beyond the scope of this text book.

**Cell size**

The number of cells in a cluster of a cellular system can help to govern the number of users that can be accommodated. But by making all the cells smaller it is possible to increase the overall capacity of the cellular system. However a great number of transmitter receiver or base stations are required if cells are made smaller and this increases the cost to the operator. Accordingly in areas where there are more users, small low power base stations are installed.

The different types of cells are given different names according to their size and function:

- *Macro cells:* Macro cells are large cells that are usually used for remote or sparsely populated areas. These may be 10 km or possibly more in diameter.
- *Micro cells:* Micro cells are those which are normally found in densely populated areas which may have a diameter of around 1 km.
- *Pico cells:* Picocells are generally used for covering very small areas such as particular areas of buildings, or possibly tunnels where coverage from a larger cell in the cellular system is not possible. Obviously for the small cells, the power levels used by the base stations are much lower and the antennas are not positioned to cover wide areas. In this way the coverage is minimized and the interference to adjacent cells is reduced.

**Know your progress**

1. *What do you mean by the concept of frequency re-use ?*
2. *Classify cells according to their physical size.*

## 11.5 Multiple access shemes for cellular systems

### Requirements for a multiple access scheme

In any cellular system it is necessary to have a scheme so that it can handle multiple users at any given time. Advancement of cellular technology enable many other different techniques to be adopted for handling multiple users simultaneously.

The requirements that any multiple access scheme must be able to meet are

- Ability to handle several users without mutual interference.
- Ability to maximize the spectrum efficiency
- Reliability
- Flexibility to handover between cells.

Now let us discuss about the different multiple access schemes.

### FDMA - Frequency Division Multiple Access

FDMA is the simplest method of the multiple access schemes that have been used. As a subscriber comes onto the system, or swaps from one cell to the next, the network allocates a channel or frequency to each one. In this way the different subscribers are allocated a different slot and access to the network. As different frequencies are used, the system is naturally termed Frequency Division Multiple Access. This scheme was used by all analogue systems

### TDMA - Time Division Multiple Access

The second system came with the introudction of to digital schemes for cellular technology. Here digital data could be split up in time and sent as bursts when required. As speech was digitized it could be sent in short data bursts, any small delay caused by sending the data in bursts would be short and will not be noticed. In this way it became possible to organize the system so that a given number of slots are available on a given transmission. Each subscriber would then be allocated a different time slot in which they could transmit or receive data. As different time slots are used for each subscriber to gain access to the system, it is known as time division multiple access.Obviously this only allows a certain number of users to enter the system. Beyond this another channel may be used, so systems that use TDMA may have elements of FDMA operation as well.

### CDMA - Code Division Multiple Access

CDMA is a method for transmitting multiple digital signals simultaneously over the same carrier frequency (the same channel).CDMA transmits over the entire frequency range available. It does not assign a specific frequency to each user on the communications network.Here the base station allocates different codes to different users and when it receives the signal it will use one code to receive the signal from one mobile, and another spreading code to receive the signal from a second mobile. In this way the same frequency channel can be used to serve a number of different mobiles. When extracting the required data from a signal it is necessary to have the correct spreading or chip code, otherwise all the other data from sources using different chip codes would be rejected. It is therefore possible to allocate different users different codes, and thus different users are given access to the system.

The scheme has been likened to being in a room filled with people speaking different languages. Even though the noise level is very high, it is still possible to understand someone speaking in your own language. With CDMA, different spreading or chip codes are used. When generating a direct sequence spread spectrum, the data to be transmitted is multiplied with spreading or chip code.**Direct-sequence spread spectrum** (**DSSS**) is a modulation technique. As with other spread spectrum technologies, the transmitted signal takes up more bandwidth than the information signal that modulates the carrier. In CDMA, a single data bit is "spread" over a longer sequence of transmitted bits. These codes are known as chipcodes or spread codes.Such codes are also known as *orthogonal* codes. The name 'spread spectrum' comes from the fact that the carrier signals occur over the full bandwidth (spectrum) of a device's transmitting frequency. This widens the spectrum of the signal, but it can only be decoded in the receiver if it is again multiplied with the same spreading code. All signals that use different spreading codes are not seen, and are discarded in the process. Thus in the presence of a variety of signals it is possible to receive only the required one.

**Know your progress**

1. *What are the requirements of a multiple access scheme ?*
2. *How does CDMA technology differ from TDMA and FDMA ?*

## 11.6 GSM (Global System for Mobile Communications)

GSM (Global System for Mobile Communications) is a set of standard protocols for cellular networks and services developed by the European Telecommunications Standards Institute (ETSI). It was originally known as

Group Special Mobile. It was first introduced in 1987 for 2G mobile service. Now it is the default global standard for mobile communications and is available in over 219 countries and territories. Now let us discuss how does GSM works.

GSM is a time division multiple access system (TDMA), which allows multiple users and calls on the same frequency channel. When a user makes a call, voice is transformed into digital data which is assigned a timeslot in a channel. Let's term this timeslot 1. Let's also assume that there are two other calls on this channel, each with their own time slots; respectively, 2 and 3. The time slots would therefore follow this sequence over the channel: 123123123. Each call would pick up only its respective time slot. Therefore, the original call we spoke of would only pay attention to the data over its assigned timeslot, 1. Its sequence or stream of data would look like: 111—a steady stream of voice data or a call, in other words.

The TDMA method employed by GSM distinguishes it from its main competitor CDMA or Code Division Multiple Access. Instead of sending data in sequential timeslots, CDMA sends all data over the channel simultaneously. Each receiver has a key which corresponds to the sender (and vice versa), which allows them to draw only data relevant to them from the channel.

There are two generations of GSM: GSM and 3G GSM. As its name suggests, 3G GSM delivers 3G service while GSM delivers the older 2G service. However, 3G GSM is not, strictly speaking, GSM, as it does not use TDMA. Rather, it is a form of CDMA. This is because TDMA (and therefore GSM) is simply not powerful or flexible enough to deliver 3G services while the CDMA method is.

**Know your progress**

1. *The multiple access scheme used by GSM is.........*

## 11.7 GLOBAL POSITIONING SYSTEM (GPS)

Global Positioning System (GPS) is a space-based satellite navigation system that provides location and time information in all weather conditions, anywhere on or near the earth where there is an unobstructed line of sight to four or more GPS satellites. The system provides critical capabilities to military, civil and commercial users around the world. It is maintained by the United States government and is freely accessible to anyone with a GPS receiver.

The GPS project was developed in 1973 to overcome the limitations of previous navigation systems. GPS was created and realized by the U.S. Department of Defense (DoD) and was originally run with 24 satellites. It became fully operational in 1995.Advances in technology and new demands on the existing system have now led to efforts to modernize the GPS system.

In addition to GPS, other systems are in use or under development. The Russian Global Navigation Satellite System (GLONASS) was developed contemporaneously with GPS, but suffered from incomplete coverage of the globe until the mid-2000s.There are also the planned European Union Galileo positioning system, India's Indian Regional Navigation Satellite System, and the Chinese Beidou Navigation Satellite System.

The GPS (Global Positioning System) is a 'constellation' of 24 well-spaced satellites that orbit the Earth and make it possible for people with ground receivers to pinpoint their geographic location. The location accuracy is anywhere from 100 to 10 meters for most equipment. Accuracy can be pinpointed to within one (1) meter with special military-approved equipment. GPS equipment is widely used in science and has now become sufficiently low-cost so that almost anyone can own a GPS receiver.

The GPS is owned and operated by the U.S. Department of Defense but is available for general use around the world. Briefly, here's how it works:

- 21 GPS satellites and three spare satellites are in orbit at 10,600 miles above the earth. The satellites are spaced so that from any point on earth, four satellites will be above the horizon.
- Each satellite contains a computer, an automatic clock, and a radio. With an understanding of its own orbit and the clock, it continuously broadcasts its changing position and time. (Once a day, each satellite checks its own sense of time and position with a ground station and makes any minor correction.)
- On the ground, any GPS receiver contains a computer that 'determines' its own position by getting signals from three of the four satellites. The result is provided in the form of a geographic position - longitude and latitude, for most receivers, within 100 meters.
- If the receiver is also equipped with a display screen that shows a map, the position can be shown on the map.
- If a fourth satellite can be received, the receiver/computer can figure out the altitude as well as the geographic position.

- If you are moving, your receiver may also be able to calculate your speed and direction of travel and give you estimated times of arrival to specified destinations.

Scientists are using the GPS to measure the movement of the arctic ice sheets, the Earth's tectonic plates, and volcanic activity.GPS receivers are becoming consumer products. In addition to their outdoor use (hiking, cross-country skiing, ballooning, flying, and sailing), receivers can be used in cars to relate the driver's location with traffic and weather information. So the driver gets information about the changes in traffic and weather in his forward journey.

## 11.8 GPRS (General Packet Radio Service)

GPRS technology, General Packet Radio Service, provides the basic GSM upgrade technology used to provide packet data up to 172 kbps.

GSM was the most successful second generation cellular technology, but the need for higher data rates spawned new developments to enable data to be transferred at much higher rates. The first system to make an impact on the market was GPRS. GPRS technology enabled much higher data rates to be conveyed over a cellular network when compared to GSM that was voice centric.

GPRS technology became the first stepping-stone on the path between the second-generation GSM cellular technology and the 3G system. With GPRS technology offering data services with data rates up to a maximum of 172 kbps, facilities such as web browsing and other services requiring data transfer became possible. Although some data could be transferred using GSM, the rate was too slow for real data applications.

**What is GPRS? - packet switching**

The key element of GPRS technology is that it uses packet switched data rather than circuit switched data, and this technique makes much more efficient use of the available capacity. This is because most data transfer occurs in what is often termed a 'bursty' fashion. The transfer occurs in short peaks, followed by breaks when there is little or no activity.

In a traditional approach a circuit is switched permanently to a particular user. This is known as a circuit switched mode. With regard to the bursty nature of data transfer there are periods when it will not be carrying data.

To improve the situation the overall capacity can be shared between several users. To achieve this, the data is split into packets and tags inserted into the packet to provide the destination address. Packets from several sources can then be transmitted over the link. As it is unlikely that the data burst for different users will occur at the same time, by sharing the overall resource in this fashion, the channel, or combined channels can be used far more efficiently. This approach is known as packet switching, and it is at the core of many cellular data systems, and in this case GPRS. In packet switching, all the data packets may not travel along the same path or route. Instead different packets will move in different paths according to the availability of that path and all packets will finally reach the required destination as the address of the destination in added in each packet.

GPRS and GSM are able to operate alongside one another on the same network, using the same base stations.

### Know your progress

1. *What is the advantage of GPRS over GPS ?*
2. *Overall resourcesharing by GPRS technology improves the efficiency of transmission. Justify.*

### Let us consolidate

Public switched telephone network is (PSTN) is the aggregate of world's circuit switched telephone networks providing infrastructure and services for public tele communication. Telephone exchange or telephone switch is a system of electronic components that connects telephone calls. The basic structure of a PSTN consists of regional offices, sectional offices, primary offices, toll offices and end offices. The telephone exchanges controlled with computer program stored in the memory of the system is called Stored Program Control system. SPC's are of two types. Centralized SPC and Distributed SPC. Cellular telephone systems are a way of providing portable telephone services. It is possible to split up a communication area into several smaller regions each covered by a different transmitter/receiver station. These regions are called cells which give rise to the name cellular communication. According to the size, cells can be classified into macro cells, micro cells and pico cells. Multiple access schemes are used to handle several users to maximize spectrum efficiency and to enable ease of hand over between the cells. In FDMA the network allocates a channel to each subscriber. In TDMA each subscriber would be allocated a different time slot in which they could transmit or receive data. In CDMA the base station allocates different codes to different users and hence the same frequency channel can be used to serve a number of different mobiles. GSM describes protocols for 2G digital cellular networks used by mobile phones. GPS is a space based satellite navigation system that provides location and time information in all weather conditions. GPRS technology uses packet switched data rather than circuit switched data and makes much more efficient use of the available capacity.

All the concepts and learning outcomes of this unit were attained through general discussion, group discussion, chart preparation and using ICT.

## Let us asses

1. A) The subscriber's telephone sets are connected to the end office using............

   a) Trunk b) Inter toll trunks c) Local loops d) Co-axial cables.

   B) Write short notes about

   a) End office b) Toll office

   C) Sketch the structure of a PSTN

2. Differentiate Centralized SPC and Distributed SPC

3. What is a Cell? Give the significance of frequency reuse.

4. Differentiate macro, micro and pico cells.

5. A) The network allocates a dedicated channel to each subscriber in the case of

   a) TDMA b) FDMA c) In both TDMA and FDMA d) CDMA

   B) Distinguish between TDMA and FDMA technologies.

6. A) The standard developed to describe protocols for 2G digital networks is called

   a) GSM b) GPRS c) GPS d) CDMA

   B) Which of the following provides location and time information in all weather conditions?

   a) GSM b) GPRS c) GPS d) CDMA

7. A) The system that uses packet switching rather than circuit switching is

   a) GSM b) GPRS c) GPS d) CDMA

8. Explain how does GPS works?

## GLOSSARY

**Adder :** logic circuit which can perform binary addition.

**ADSL modem:** Asymmetric Digital Subscriber Line modem use telephone lines to send and receive data.

**ALU :** Arithmetic and Logic Unit which does mathematical calculations in a processor.

**AM :** Amplitude modulation which is a technique in which amplitude of the carrier is varied according to the message.

**Antenna :** A metallic device used to convert electric signal into electromagnetic signal and vice-versa.

**Application software :** A software formed for a specific application.

**Aspect ratio**: It describes the proportional relationship between width and height of an image.

**Assembler :** A program written to convert assembly language to machine language.

**Assembly language :** A computer programming language in which special symbols are used as instruction codes.

**Avalanche :** large mass of snow, ice, etc. detached from a mountain slope and sliding or falling suddenly downward.

**Backlit:** A backlight is a form of illumination used in liquid crystal displays (LCDs). Backlights illuminate the LCD from the side or back of the display panel, unlike front lights, which are placed in front of the LCD.

**Band Pass Filter:** It allows a specific frequency range to pass, while blocks lower and higher frequencies. It allows frequencies between two cut-off frequencies while attenuating frequencies outside the cut-off frequencies.

**Band Reject Filters :** It allows all the frequencies to pass through other than a band of frequencies called band rejection filter. Such filters are also called as band stop filter.

**Band width :** It is the range of frequencies that is contained in a signal.

**Baseband signal:** It is the message signal lying in its original band.

**BASIC :** Beginner's All purpose Symbolic Instruction Code used for general purpose programs.

**Bit :** Binary digit (A logic '0' or a logic '1')

**Bridge :** A device related to computer networks which can join segment or work group LANs.

**Byte :** A group of 8 bits.

**C :** A high level language which runs on DOS and Unix operating system.

**Cache :** A high speed memory placed as a buffer between CPU and main memory.

**Carrier :** A high frequency signal which carries the message signal to the destination from the source.

**CD :** Compact disk which is a secondary data storage device which can hold around 700 MB of data.

**Channel :** The path through which the signal travels between the source and destination.

**Clamping :** A process which places either the positive or negative peak of signal at a desired level by shifting its dc value.

**Client-server computing:** A computer model where two or more computers interact in such a way that one provides service to the other.

**Clipping:** A process in which a wave from is shaped by removing a portion of it.

**Clock :** The basic timing signal in a digital system.

**Coaxial Cable:** A type of cable that has an inner conductor surrounded by a tubular insulating layer, surrounded by a tubular conducting shield.

**COBOL :** Common Business Oriented Language used for business oriented programs.

**Combinational circuit :** combination of gate networks having no storage capability.

**Comparator :** Circuit which can compare two data and say whether they are equal or which one is larger.

**Compiler :** A program used to covert high level language to machine language. It scans the entire file at a time.

**Counter :** A digital circuit which can count electric pulses.

**CPU :** The Central Processing Unit which is the brain of Computer.

**Demultiplexer :** The device which puts different data in a line to different lines.

**Differentiator:** A circuit in which output voltage is directly proportional to the derivative of the input.

**Digitizer :** Computer input device which convert graphics and pictorial data into binary inputs.

**DNS:** Domain Name Server.

**Domain name:** Name used in URL to identify particular web page.

**DSB :** Double side band AM in which both the sidebands are retained.

**DVD :** Digital Versatile Disk is an optical storage device with a capacity up to 4.7 GB

**Encoder :** circuit which converts information into coded form

**Feedback :** A path from output to input which is used to control the performance of a circuit.

**Filters :** A frequency selective circuit that allows a band of frequencies to pass through and blocks or attenuates signals of frequencies out-side this band.

**Flash drive :** Small sized external storage device with a capacity of 2 Gigabytes to 16 Gigabytes.

**Flicker:** The blanking of a TV screen arises because the video display images are not generated continuously.

**Flip flop :** A logic circuit used to store one bit information.

**Floppy Disc:** A secondary storage device of low memory capacity used in earlier days.

**FM :** Frequency modulation in which frequency of the carrier in varied according to the message.

**FORTRAN :** Formula translation used for scientific purpose.

**Frequency spectrum :** It is an amplitude versus frequency graph in which amplitude of each frequency component of a signal is shown.

**FTP :** File Transfer protocol which is the method to copy files over a network from one computer to another.

**Gateway :** A network device which makes communication possible between different environments.

**Hard Disk :** The largest secondary memory in a computer.

**High level language:** A programming language that uses grammatical and mathematical notations similar to everyday language.

**High Pass Filter(HPF) :** It passes high-frequency signals and blocks low-frequency signals.

**HTTP :** Hyper Text Transfer Protocol which is a standard way for transmitting HTML pages.

**Hub :** A common connection point for devices in a network.

**Inkjet printer :** A non-impact character printer.

**Integrator :** A circuit in which output voltage is directly proportional to the integral of the input.

**Interlacing :** A technique for doubling the perceived frame rate of a video display without consuming extra bandwidth.

**Interpreter :** A program that convert high level language to machine language. It scans one line of a program at a time.

**IP address:** The unique identifier for a computer in internet.

**ISB :** Independent sideband in which two different messages are included at the LSB and USB

**LAN:** Local area of network of computers.

**Laser Printer :** A high speed and high quality non-impact printer which generates an electro-static image on the drum.

**Load regulation :** A measure of the ability of the power supply to reduce the variation in output voltage with the change in load current

**Low pass filter (LPF) :** It passes low-frequency signals and blocks high-frequency signals.

**Machine language :** A sequence of instructions written in binary numbers.

**Mainframe :** Very large and expensive computer capable of supporting thousands of users simultaneously.

**MAN:** Metropolitan area of network of computers.

**MICR :** Magnetic Ink Character Recognition Technology.

**Microns :** The millionth part of a meter.

**Mini Computer :** A midsize multi-processing system capable of supporting nearly 250 uses simultaneously.

**Mixer :** This circuit mixes two signals to produce two new signals with frequencies equal to sum and difference of the individual frequencies.

**Modem:** A device used to transmit digitized data via analog channels. A computer is connected to the internet via modem.

**Modulation :** A process in communication in which any one characteristics of a carrier is changed so as to add the message signal into it.

**Modulation index :** A term used to indicate the strength of modulation.

**Monochrome:** A picture, especially a painting, done in different shades of a single colour.

**Mother board :** All parts of a computers are connected together via mother board.

**Multiplexer :** A circuit which puts information from several sources on to a single line.

**Network Protocol :** Standard methods of transmitting and processing information.

**Network Topology :** The way in which the computer systems are connected with each other.

**Nibble :** A group of 4 bits.

**NIC :** Network Interface Card which is a hardware component that allows a computer to connect to a network.

**OCR :** Optical Character Reader.

**OMR :** Optical Mark Reader.

**Operating System :** A system software used to operate the computer.

**Optical fiber :** a long fine thread of glass used to carry light as a carrier in communication.

**Optics :** the branch of physical science that deals with the properties and phenomena of both visible and invisible light and with vision.

**PAN:** Personal area of network of computers.

**PC :** A personal computer which is designed for an individual user.

**Photosensitivity:** The amount to which an object reacts upon receiving photons, especially visible light.

**Pixel:** In digital imaging, a pixel or picture element is a physical point in a raster image, or the smallest addressable element in an all points addressable display device.

**PM :** Phase modulation in which phase of the carrier is varied according to the message.

**Quantiser :** A device which converts the sample values to the nearest prefixed values.

**RAM :** Random Access Memory is the internal memory (Primary memory) of computer which is a temporary storage device.

**Refractive index :** a measure of the extent to which radiation is refracted on passing through the interface between two media.

**Regulated Power supply:** A dc power supply which maintains the output voltage constant irrespective of ac mains fluctuations or load variations.

**Repeater :** A device used in electronic communication which regenerates the received signals and then re transmits.

**ROM :** Read only Memory which is a device to store binary data.

**Router :** A network device like bridge which can also determine the best path for sending data.

**Selectivity :** The property of a circuit (filter or amplifier) to select a particular range of frequencies.

**Sequential circuit :** Combination of gate networks having storage capability.

**Shift register :** A memory device in which stored bits can be shifted right or left.

**Sidereal :**A sidereal year is the time taken by the Earth to orbit the Sun once with respect to the fixed stars.

**Spectroscopy :** The science that deals with the use of the spectroscope and with spectrum analysis.

**SSB :** Single side band in which either LSB or USB is transmitted.

**Submarine :** A vessel that can be submerged and navigated under water, usually built for war fare and armed with torpedoes or guided missiles.

**Super computer :** The fastest computer for specialized applications such as weather forecasting, scientific simulations etc.

**Switch :** A data communication device similar to but more advanced that hub which checks the destination MAC address before sending data.

**TCP/ IP :** Transmission Control Protocol/ Internet Protocol.

**Thermal printer :** A printer which operates on the application of heat.

**Transducer :** A device that receives a signal in the form of one type of energy and converts it to a signal in another form.

**USB port:** Universal Serial Bus port which is used to standardise communication between computers and peripherals.

**Vestigial side band :** A side band with a part of which is cut off or suppressed.

**Voltage Follower (Buffer) :** The non inverting amplifier configured for unity gain. It is called ***voltage follower*** because the output voltage is equal in amplitude and phase with the input.

**WAN:** Wide area network of computers.

**Wave guide :** A tube, coaxial cable or strand of glass fibers, used as a conductor or directional transmitter for various kinds of electromagnetic waves.

**WLAN:** Wireless local area of network of computers.

**Work Station :** A computer used for engineering applications (CAD/ CAM)

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